

APRIL 2007

# ProAudio Review

The Review Resource for Sound Professionals

## World Premiere! Apogee Symphony Ensemble



In This Issue:

### ■ SSL Duality Mixing Console

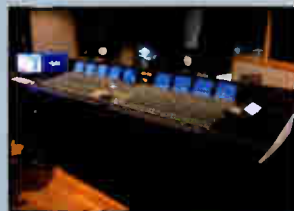
*Analog Fights back*

### ■ Trident 8T Analog Console

*A Lot Of Value For Under \$4,000*

### ■ From The Road

*The Analog Or Digital Dilemma*



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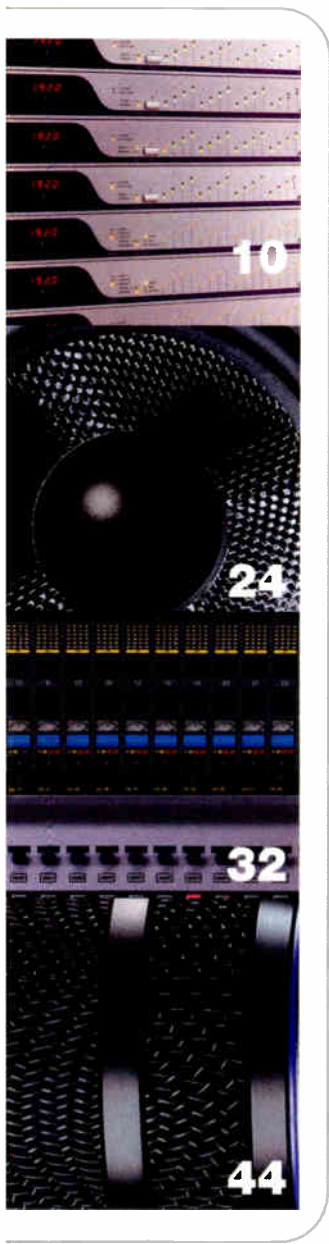
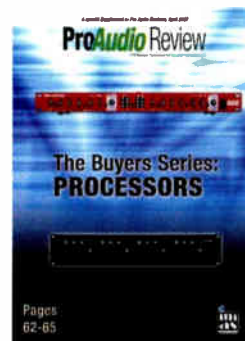
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Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

# NEW REALITIES IN AUDIO

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# AVIOM

# FROM THE PUBLISHER

John Gatski

## Music in Stereo

As I read Mike Rivers' detailed roundup of two-track editors for PCs in this issue (a Mac roundup is scheduled for June), I was reminded of the power of stereo. With all the products that bring us multitrack recording, playback, mixing and delivery, two-track stereo is still the way most people listen to music.

I'm the same way. With my reference high-resolution multichannel system and access to numerous SACD, DVD-A and Dual Disc multichannel titles, I still listen to 99 percent of music in stereo.

For movies, I have a kick-ass home theater HDTV receiver/multichannel system that gets maximum listening at 5.1, but if I play music on it, I usually listen in stereo.

And most mass music consumption is done the same way: iPods, MP3s, internet radio, HD Radio, CDs, FM — it's all stereo.

So why is that two-channel format that was developed more than 50 years ago still so dominant? I believe that stereo's steadfast adherence is a combination of habit, biology and simplicity.

First of all, two-channel stereo has been the standard way of listening for most people alive today. It is a habit and it is the reference. People are used to listening to two speakers in front of them or two speakers on their ears.

Biologically speaking, we are stereo creatures. We hear in stereo. We have two ears; we listen to two speakers. Makes sense, huh? We don't have 5.1 ears. Thus, stereo is pretty convincing in delivery of how we hear natural sound and live music. Like the wheel, which is simple, yet a most effective design for moving objects, stereo reproduction is amazingly simple and yet quite satisfying.

The other reason stereo is still so dominant in the music arena is economic simplicity. Why pay more for extra speakers and more complex amplification and processing when music sounds pretty good in stereo? Many folks have home theaters, but

that is for movie and TV sound. Their main music listening is likely to be more portable. Hence the iPod, computer, boomboxes, FM radios and other smaller audio stereo devices.

Though there is more music in multichannel delivery thanks to DVD and 5.1 broadcast, the music industry is still in stereo mode. Pre-recorded CDs are stereo as are all the portable delivery formats such as MP3 downloads. DVD 5.1 music discs are not as plentiful in comparison and those previously mentioned high-resolution formats are even less in number.

Now I am not saying that the mixed-to-stereo is better than discrete 5.1 channel audio. If properly positioned and set up, multichannel high-resolution audio brings out much more detailed spacial cues of music than stereo — especially from rear sound reflections. It sounds amazing, but stereo still sounds very good in comparison.

In today's music listening world, stereo is not going away any time soon.

### NINE MILLION PAGE VIEWS AND COUNTING

Having launched our computer-delivered digital edition of *Pro Audio Review* in August 2005, I am proud to report that, as of mid-March, more than nine million page views have been logged from the thousands of people who receive the monthly link. The nine million page views means that our readers treat the digital edition just like the paper mag; they read it, reread it and then read it again. However, the repeated viewings never tear up the pages like in the paper edition.

If you have not seen the digital edition, go to the [www.proaudioreview.com](http://www.proaudioreview.com) web site and check out a sample and sign up. It's free.

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# ProAudio Review

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# Letters

## FEELING GROOVY

Thank you for your thorough review of our Glory Comp compressor (*PAR* March 2007). I am always glad when a reviewer really "gets it", and Paul Richards really nailed the positive qualities that make the Glory unique and special.

We could only think of a few items of interest not specifically mentioned in the review.

First, up to seven Glory Comps can be daisy-chained together for stereo or 5.1 Mastering via easy TRS interconnections on the rear panel. These then can operate independently, or as 6 Slaves with a designated Master whose dynamic settings then control up to 7 channels of audio.

## Correction

From our March issue:

The picture of Shakira on page 17 of the "How the Pros Gear Up for Tour" feature should have included the following caption: Shakira proved again that hips don't lie and neither does the sonic performance of her Sennheiser SKM 3072 wireless mic at the 49th Annual Grammy Awards presentation. (Photo credit: Getty Images)

Second, we designed the Glory Comp as a variable transconductance tube compressor. BTW, this is the secret behind what might be the most coveted dynamics processor of all time, the Fairchild 670, another rare example of REAL tube compression.

Lastly, perhaps the most unique and expensive feature of Glory is not instantly seen; Glory has a fully differential signal path, commonly known as "balanced" path, from front to rear. This doubles the component count (and size!), and requires closely matching all components within 1%. But the results are well worth the extra time and cost evidenced by the incredible "air" in the sound of Glory which many have called "stunning". For producer/engineers who seek modern dynamic control options, that special smoothness that previously could only come from the Fairchild, but don't want to deal with the price tag and maintenance headaches that come along with a vintage processor, we feel the Glory won't disappoint anyone.

Thanks again and God bless.

Aspen Pittman  
Founder/CEO, Groove Tubes

Feedback: We want to hear from you. Send your comments to [letters@proaudioreview.com](mailto:letters@proaudioreview.com).



Pro Audio Review  
Aspen Pittman, Founder

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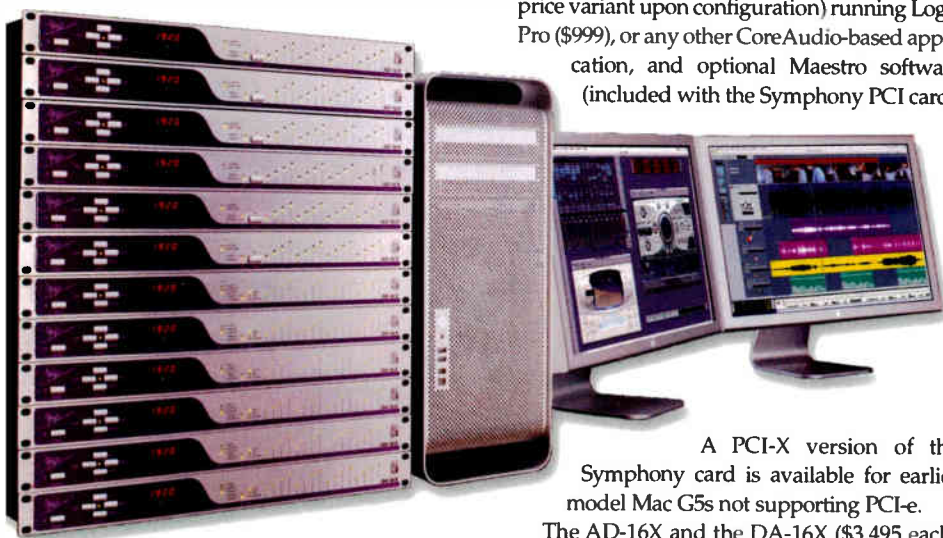
by Russ Long

# Apogee Symphony Recording System

This competitively priced, high-performance native DAW is a challenge to the industry pervasiveness of TDM systems.

Apogee Electronics has unveiled the Symphony System, perfectly timed with the rollout of ultra-fast, multi-core Intel-based

Cards (\$195); second, the Symphony PCI Express (PCI-e) card (\$795); and lastly, a compatible Apple computer (such as the Mac Pro, price variant upon configuration) running Logic Pro (\$999), or any other CoreAudio-based application, and optional Maestro software (included with the Symphony PCI card).



A PCI-X version of the Symphony card is available for earlier model Mac G5s not supporting PCI-e.

The AD-16X and the DA-16X (\$3,495 each) are Apogee's latest and most advanced conversion systems. They ramp up Apogee's legendary build quality by incorporating a redesigned power supply, standard 192 kHz sampling rates (the C777 clocking technology found in the Big Ben) and several optional expansion cards.

The power supply in the AD-16X and the DA-16X is a specially designed Synchronous Switching power that works in conjunction with Apogee's low jitter clock and filtering technology. It provides better heat and noise performance and an improved transient response, thus creating an ideal power system for first-rate conversion.

The AD-16X incorporates Apogee's classic SoftLimit and UV22HR features. SoftLimit is an analog peak limiter that enables the capture of an additional 4-6 dB of level without any overs. UV22HR is Apogee's industry standard bit-reduction/dithering technology for CD and DVD mastering; UV22HR can also be used to produce improved Internet and com-

puter audio content without increased file sizes or data rates.

The DA-16X incorporates Apogee's electronically balanced line drivers, which were originally designed for the Mini-DAC. These line drivers simulate transformer behavior. The circuitry is an ultra low output impedance/high current driver, capable of running levels up to 26 dBu to the most complex or low impedance loads. This translates into seamless integration with vintage equipment.

The Symphony PCI card was designed specifically for the Apple Mac. It is a 24-bit/192-kHz PCI-e card that provides 32 simultaneous channels of digital I/O. Up to three Symphony cards can be installed in one Mac, making it possible to have 96 simultaneous inputs and outputs running natively within a single computer. A 192-channel (96 in and 96 out) I/O setup requires six AD-16X boxes and six DA-16Xs. And configuring the AD-16X and DA-16X to "Advanced" routing allows additional hardware (converters, digital effects, etc.) to be connected via the AD/DA's digital I/O.

The heart of the Symphony System is the Maestro software. This software application, while not required, provides undeniably valuable control of Apogee interfaces by allowing the user to adjust hardware settings, configure routing between hardware I/O and audio software I/O, and to perform low-latency

**APOGEE** continues on page 12 ►

## FAST FACTS

### APPLICATIONS

Studio, broadcast, audio post

### KEY FEATURES

32-channels of I/O per card; 1.6 milliseconds of latency at 96 kHz (the lowest available in a native audio system); amazing sound; flexible configuration; reasonable price; built-in Big Ben-quality clock

### PRICE

Symphony PCI Series card: \$795  
AD-16X: \$3,495  
DA-16X: \$3,495  
X-Symphony Card: \$195  
Logic Pro: \$999

### CONTACT

Apogee Electronics Corp. |  
☎ 310-584-9394  
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Apple Mac Pros and the state of the art Logic Pro 7.2 DAW update. The Symphony System is a complete DAW that offers the greatest number of I/O channels per card of any CoreAudio compatible digital interface. It is actually the only solution capable of 96 channels of I/O for Mac without the need for an expansion chassis. The system maintains a low CPU load, thus conserving the computer's processing power for plug-ins. Connection between the computer and the audio hardware is completed via one high performance balanced cable that is capable of flawlessly transmitting digital audio over long distances. And the Symphony System is a competitively priced package, regardless of the exact configuration.

### | FEATURES

The Symphony System is made up of three components: first, a combination of X-Series converters (Rosetta Series converters are also supported) each equipped with X-Symphony

# Wireless Innovations.

Sony has a history of breakthroughs in wireless microphones, including the origination of synthesized UHF technology. Our newest models continue to lead the way. The WRR-862B camcorder-mounted receiver simplifies field production with two channels of reception. The MB-X6 tuner rack streamlines multi-channel sound with six channels in a single rack unit. The sleek WRT-88 body pack lets you choose high power for maximum distance or low power for maximum battery life. Sony's 800 Series also features diversity reception, legendary build quality and an expanded range of channels to help you navigate an increasingly crowded broadcast band. Outstanding simplicity and agility... that's wireless innovation.

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PRO AUDIO



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mixes of hardware I/O paths and software outputs. Maestro consists of two main windows, the "Settings" window and the "Routing/Mixer" window. Maestro also includes functionality for checking Ensemble software and firmware versions, personalizing "modifier" keys when setting the mixer, and saving and recalling routing and mixer configurations.

The Settings windows allows the user to toggle between the Apogee interfaces that are connected to the Symphony PCI card, set the clock source for the selected unit, configure the VBus, and configure the Performance Tuning. The V-Bus feature, employing Apogee's ultra-fast Symphony driver, provides unlimited routing possibilities across multiple applications in any Symphony System. This makes recording out of QuickTime or Final Cut Pro directly into your DAW a breeze. The Maestro Control Panel allows the virtual channels to be enabled. The V-Bus channel names are clearly labeled and the number of additional V-Bus channels is selectable within Apple's Audio Midi Setup. Logic users will love this feature because it makes it possible to record bus outputs within Logic Pro. VBus offers zero latency performance with minimal processing power on up to 32 buses per Symphony card, ensuring maximum audio performance.

Maestro's Performance Tuning feature provides the ability to fine tune the performance of the Symphony driver to allow super low latency for high performance platforms. This

gives the user the ability to maximize the performance of the Symphony system on all supported hardware.

The Routing/Mixer Window has three panes labeled "Input," "Output" and "Mixer." Like the Setting window, each of the Routing/Mixer panes has a "Unit Select" tab that allows the user to toggle between the Apogee interfaces that are connected to the Symphony PCI card.

The Input routing pane consists of a routing grid that plots the various connections between hardware inputs and signal paths to software applications. These connections appear as inputs within the software applications. It is also possible to send one hardware input simultaneously to several software signal paths. The hardware inputs are represented by a row at the top of the grid, while paths to software are represented by the column to the left of the grid.

Signal paths to software may be configured as Mono or Stereo within the "Matrix" column of drop down menus. The Mono setting allows for the greatest routing flexibility, but the Stereo setting offers the convenience of routing signals a pair at a time. I found that the Stereo generally worked best for all of my applications, but it was nice to have the option of configuring the paths as mono if required. The signal paths can be labeled in the Input column.

The Output routing pane is almost the same functionally as the Input pane but it is used to make connections between signal paths from software applications and hardware outputs. Hardware outputs are represented by a row at the top of the grid, while the column to the left of the grid represents paths from software. In the Matrix column of drop down menus, signal paths from software may be configured as Mono or Stereo. The Mono setting, like the Input pane, allows for the greatest routing flexibility, while the Stereo setting offers the convenience of routing signals a pair at a time. A path may be set to Off in this menu as well. Signal paths may be labeled in the Output text box. If the AD16X is set to Standard routing, there will be no paths listed in the Output pane as there is only one signal path (analog inputs to all digital outputs). When it is set to Advanced routing, the Input pane is used to route the AD16X's analog inputs to software inputs and the Output pane is used to route software app outputs to

the AD16X's AES and ADAT/SMUX outputs. There will likewise be no paths listed in the Input pane if a connected DA16X is set to "Standard" routing.

The "Maestro" mixer blends the hardware inputs with the software application playback and routes the mix directly to hardware outputs. The "Mixer Select" (A-B) drop down menu selects between the two available mixers per hardware interface. The "Input Channels" provide the ability to control inputs to the mixer. Hardware inputs of the selected interface are the source for these input channels. The mixer always displays 18 inputs; the unused



Apogee Symphony Card

mixer channels are grayed out if the selected device is equipped with less than 18 inputs. The mixer inputs each have "Pan," "Level," "Solo" and "Mute" controls. A bar graph style meter displays the pre-fader input level.

The "DAW Return" is a stereo input that provides level control, metering and mute/solo functions for the DAW Return signal, e.g. the mix of playback tracks from the software app. The "Main Output" is a stereo channel that provides a level fader, metering, and a routing selection drop down menu for controlling the stereo output of the mixer. The Main output drop down menu selects the hardware output to which the Main Output is routed.

## IN USE

The system that I reviewed consisted of a Macintosh dual 3-GHz Mac Pro with a beautiful 23-inch Apple Cinema HD Display, a Symphony PCI Express card, two Apogee AD16X interfaces, two DA16X interfaces, the Maestro software package (included with the Symphony card) and Logic Pro 7.2.

On the rear of the X-Symphony card are two self-latching multi-pin connectors labeled "Main" and "Through." Communication between the converter and the Symphony card is completed via Apogee's proprietary PC32 digital cable. The AD-16Xs and DA-16Xs have to be placed in the correct sequence for the rig to operate properly: first an AD-16X then a DA-16X, then the second AD-16X, then the second DA-16X, and so on. A PC32 cable runs from the Symphony card to the AD-16X's Main port and then from that unit's Through port into the Main port of the next unit, and so on.

I connected and configured the system (which took one call to tech support due to operator error), then launched Apple's Audio/MIDI Setup and set the default system I/O to the Symphony card, so all appli-

APOGEE continues on page 14 ►

## APPLE OF THE REVIEWER'S EYE

The Apple Mac Pro included with my review setup was, by a large margin, the best computer that I've ever used. It worked perfectly, was lightning fast and, quite frankly, made me drool. If I had the money (or even enough room on my credit card) it would have found a permanent home in my studio. The Symphony System, even considering the Mac Pro's price tag, is about half the price of a Pro Tools rig of similar specifications.

## HERE'S A CLOSER LOOK AT THE COMPUTER

- Mac Pro w/Dual-Core 3-GHz Intel Xeon processors
- L2 Cache (per processor): 4 MB
- Memory: 4 GB
- Bus Speed: 1.33 GHz
- 465 GB internal drive
- 1.36 TB internal raid

# LOUD and CLEAR



## WaveLab 6—Complete Native Mastering Suite

There's more music being produced these days, and in more formats than ever before. To get yourself heard, you need to make sure that your final mastered mixes have punch, clarity and sonic power. That's what WaveLab 6 is all about. Using WaveLab to master your CDs, DVDs and MP3s is the final step in making truly professional sounding productions. You've spent a lot of time and money on producing your music. Fortunately, with WaveLab 6, you don't have to spend a lot on mastering and making sure your music gets heard loud and clear...

[www.steinbergusers.com](http://www.steinbergusers.com) [www.steinbergupgrades.com](http://www.steinbergupgrades.com)



cations that use audio (including QuickTime and iTunes) played through the Symphony engine.

Now it was time to dig into Maestro. I knew the software was powerful, but if it weren't intuitive and easy to use it wouldn't translate well into the real world. I was impressed; the software was easy to navigate and manipulate.

I loaded in a multi-track work in progress and was ready to record my first Symphony System track. But first I went to the "Audio Hardware and Drivers" page and unchecking the "Software Monitoring," configuring Logic Pro so that my overdub track would mute when Record was engaged, yet would play back when Record was disengaged (even though the track was record-enabled). This would take full advantage of Symphony's low latency performance, and make the software behave like an analog tape-machine in auto-input.

I used the same configuration in a full tracking situation (22 tracks) and had flawless operation. Working on native systems in the past has always reminded me of taking pictures with a cheap camera. It seems like every time you try to take a picture there is a lag before anything happens. This is not the case with the Symphony System. It responds to any command instantly, making punching in or out or any quick adjustments a breeze. This is the first time ever I have found myself working on a native-based system that provides the same quality of performance that I've only been able to attain with Pro Tools.

I went on to route a single mic to all 32 inputs, just to see if I would

have any problems recording long audio files: 32 tracks, 96 kHz and 180 minutes and I didn't have a single problem.

I don't care how powerful or reasonably priced a recording system is, if it doesn't sound good, it's just not worth attention. I've been using the AD-16X and the DA-16X in various Nashville studios for years and I've always been a fan of their sound. When equipped with the X-HD Card instead of the X-Symphony card they act as seamless replacement for the Digidesign 192 within a Pro Tools rig (but they sound much better). If a studio has Apogee converters I'm confident enough to leave my DAW at home. This said, my assumption was that I would be pleased with the sonic performance of the Symphony System and I was right. In every instance the Symphony System sounded wonderful. The filters are focused, punchy and transparent, and they have a wonderful clarity.

I compared the clock in the AD-16X to my Lucid GENx96 Clock and I wasn't able to hear any difference. Wow, an internal clock that sounds as good as a stand-alone model. There's nothing wrong with that! I had great results using not only Logic Pro, but also Bias Peak, Nuendo, Garageband and Ableton Live with the Symphony System.

#### | SUMMARY

I'm a long time Pro Tools user and, quite frankly, I haven't considered switching platforms...until now. Logic Pro 7.2 coupled with the Symphony System is reasonably priced for any audio professional and it has no major faults that I can find.

Apple's Final Cut Pro has literally swiped Avid's golden crown of film and video editing over the last few years, and now it appears that Logic Pro teamed with the Apogee Symphony System could make it a double play. The Symphony System paired with Logic (or Cubase, or Digital Performer, etc.) matches the performance and surpasses the audio quality of Pro Tools | HD, while cutting the price virtually in half. Anyone in the market for a high-end DAW should give the Symphony System top consideration.

*Russ Long has done 5.1 DVD mixes for Allison Moorer and Mercy Me and is an in-demand engineer for live sound recordings.*

#### | SYMPHONY MOBILE: AN ON-THE-GO OPTION

Apogee Symphony Mobile, based on a new PCI ExpressCard, is built for on-the-go engineers using an Apple MacBookPro. The large number of MacBookPro laptops now used by professionals in the audio industry make this product a serious contender. It's 192 kHz capable and offers 32 channels of simultaneous I/O with nearly non-existent latencies at 96 kHz — a mobile symphony, indeed.

"The challenge was to make it easier for these professionals to get around and work at the same level that they could in a traditional environment," said Apogee Director of Sales Max Gutnik. "We believe Symphony Mobile meets this challenge exceedingly well, making it the ultimate solution for the professional on the go."

If the desktop review is any indication, engineers who can't sit still may find some extra grease for the wheels in the Symphony Mobile system — stay tuned.

— Strother Bullins

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by John Gatski

# Trident 8T 16-Channel Analog Console

Versatile, great-sounding small format mixer with a distinct Trident pedigree.

**W**ith all the advances in digital consoles and direct-to-computer recording these days, its refreshing to see companies, such as John Oram's Trident Audio Ltd. willing to make high-quality analog boards with traditional routing options. Many engineers love the sound and straightforward, linear approach of an analog board for recording to analog or digital sources.

The Trident 8T, priced at \$3,498 for 16-channels, is a classic example of a great sounding, relatively inexpensive analog board, still made in the UK — and it offers incredible flexibility. Yeah, it does not use discrete components in the audio path, like the original Trident Series 80, but this op-amp-based board has serious enough sonic quality to showcase the increased dynamic range and detail sub-

tleties of 24-bit/96-kHz recording.

## FEATURES

The Trident 8T is based on the original Trident Series 80 design, especially the EQ section. The console configuration range from eight to

32-channels.

The 8T is packed with plenty of features including eight aux sends, monitor buss, mono selection, optional meter bridge (\$1,050). Each channel also has peak and signal-present LEDs for those working without the meter bridge.

The console features such goodies as balanced direct outputs for each of the channels (16 in the test unit's case), eight track busses, eight aux sends and the same high quality mic preamps as the S20 standalone pre, which PAR tested and recommended highly. The entire console input and output structure is balanced. The microphone inputs are XLR while the rest of the I/Os are 1/4-inch TRS.

As mentioned, the EQ section is based on the original Trident Series 80 EQ, to which Trident Audio, Ltd. owner Mr. John Oram contributed circuitry and sonic design back in

the 1970s. The three-band EQ includes: high-shelf filter, high-mid level control, low-mid level control, and low-shelf control. The high-shelf turnover frequency is switchable to either 8 kHz or 12 kHz. The high-mid frequency control is adjustable from 1 kHz to 15 kHz. The low-mid frequency control ranges from 100 Hz to 1.5 kHz. The EQ section also includes 50 kHz low-cut filter; the low-shelf frequency can be set to either 60 Hz or 120 Hz.

This console has plenty of I/Os including the eight track busses, three sets of stereo outputs (L/R, main and closefield), as well as the array of aux send/returns. The board also has a talkback output and mic input for talkback.

Each channel has balanced, quarter-inch connections for line, tape, aux send/return (TRS), and the direct channel output. The balanced XLR input connects the microphone to the preamp and can also be used as a balanced



XLR line input with the "unity" switch engaged. The insert jack on each channel uses the common TRS unbalanced send/return scheme (also in the original Series 80).

There are a few interesting channel features not traditionally found on consoles in this price range including a "flip" switch, which swaps the tape inputs (on the rotary faders) with the live inputs (on the faders), allowing quick reconfiguration between recording and mixing modes.

For the money, the Trident 8T has a great subgroup section with the eight track (a.k.a. submix) busses. In addition to the L/R buss, each track can be assigned to any of the 4 track buss pairs; panning is used to isolate the signal to odd or even numbered busses of

## FAST FACTS

### APPLICATIONS

Studio, project studio

### KEY FEATURES

16 channels; eight-track busses; three-band EQ; eight aux sends; 16 direct outs; optional meter bridge

### PRICE

\$3,498

### CONTACT

Trident Audio Ltd. |  
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the pairs.. Each track buss has its own master output and buss solo button. An additional button on each pair allows discrete stereo soloing of each pair (as opposed to the traditional mono solo function).

Metering is done via two classic, oval-shaped, blue, backlit onboard VU meters,

## PRODUCTPOINTS

- Sonics
- Plenty of I/Os
- Eight track busses
- Quality build
- Same meter bridge components as high-end Tridents
- Wiggly rotary pots
- Needs more meter bridge backlighting
- Where's the manual?

## SCORE

*Incredibly versatile, great-sounding analog board for the serious professional home or recording studio*

which can monitor the track busses, the main output bus, and two track returns. The meters are set at 0 VU at +4 dBu, with a -10 dB attenuator switch designed to match with high-level digital signals. In addition to the aforementioned EQ and buss-assign controls, each channel also features a solo engage button and a flexible aux buss section that gives a maximum of options using a minimum of space. The first two of three aux send level knobs are switchable between auxes 1 & 3 and 2 & 4 respectively. The third level knob is a stereo send that is switchable between aux pairs 5/6 or 7/8 and can be configured pre- or post-fader. The 8T is equipped with smooth, tight 100-mm faders for the channels and the main stereo output. Output controls are available for monitor section, track busses, solo, talk back section and headphone amp.

The 8T board boast great specs including ultra-low noise from input to output and a 45 kHz (-3 dB point) frequency response to ensure an accurate top end past 20 kHz, which is good for those high resolution recordings. Other specs include 0.008-percent distortion.

The board is attractive looking; its wood trim and optional meter bridge give it a classy pedigree — even in a modest home studio decor. And with the power supply outboard and rack mountable, the 8T fits into quite a compact footprint. The 16-channel version, as tested, is only 29-inches wide and weighs a mere 40 pounds, making it ideal for tight-spaced pro home studios and live

remote recording gigs.

According to a U.S. Trident dealer, most of the 8Ts have been sold into home studios in the U.S.

### IN USE

The Trident was easy to set up and use, which was a good thing since a proper manual was still in production. Subsequently, most of this review's specs and control features were taken off the Trident web site.

I recorded a bunch of acoustic guitar/vocal tracks mixed straight to two-channel high resolution PCM. Using several microphones, including my trusty Audix SCX-25s, a pair of Heil PR-40s and Shure KSM 32s. I routed the main outputs of the Trident to the Benchmark ADC-1 converter, set at 24-bit/96 kHz, which then fed either an Alesis Masterlink or TASCAM DV-RA1000.

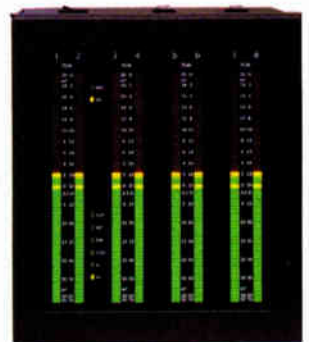
When recording to the two-track recorder **TRIDENT** continues on page 18 ►

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## STUDIO | REVIEW

TRIDENT Continued From Page 17

via the stereo mix busses, I not only routed audio to the main recorder at 96 kHz from the main stereo buss, but I simultaneously fed a second digital recorder running at 44.1 kHz from one of the other stereo outputs. This enabled me to make CD sample rate compatible references of the same recording without later having to sample rate convert from the reference 96 kHz. The 8T is a great board for feeding different sources. I also tracked to the recorder via the direct outputs and the track busses.

So how does it sound? The stereo bus output tracks that were recorded with the Audix SCX-25s on Martin and Guild acoustic guitars, plus voice and a electric Gibson L5 jazz guitar, recorded with Shure KSM-32s, were all very good to excellent sounding. From the stereo buss, the 96 kHz stereo recordings were very accurate and clean with a touch of that analog warmth but not so warm that the dynamics and transients were lost. Stereo imaging was quite good as well.

The direct channel outputs, as expected, produced the most transparent recording with slightly more transient energy and less warmth, but the summing buss acquitted itself quite well.

As a testament to good design, this board is really quiet, with nary any hiss — even with the controls cranked way up. It is super clean with plenty of headroom.

On the pair of Heil PR-40 microphones, a great dynamic with extended bottom end response, I used the EQ to add some top end and air to a Martin D35, and the EQ delivered audibly pleasing results with out adding harshness. Just what you want in an EQ.

The control and fader integrity is quite good — especially the faders. The “Fleximount” rotary controls are a bit wiggly feeling, but not noisy and very linear. The push-button switches engage okay, but the plastic buttons can fall off when transporting the board [*The company assures this has been addressed. — Ed.*].

Although the board can be used with its dual VUs, I believe those who do a lot of tracks are going to want the optional meter bridge. It contains so much more level information about multiple inputs and outputs versus the VUs. However, the meter bridge does make it more difficult to plug into some of the connections. The headphone jack and direct outputs, for example, were a bit of tight reach since they are the furthest in.

Also, the meter bridge legend illumination is too dim to easily read the reference level numbers. I think it could use a little bit more backlighting.

### | SUMMARY

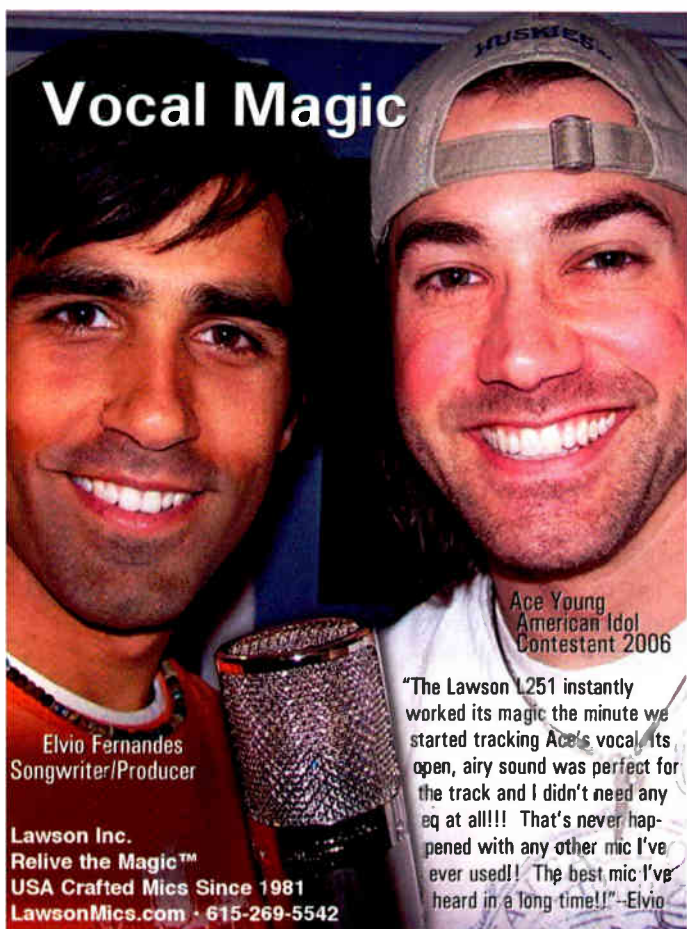
The Trident 8T is a quality, pure analog console that bridges the gap between the low-buck utility recording mixer and the high-end, without much compromise in audio quality of the latter. Its logical, easy-to-use layout and plethora of I/Os and adept monitoring and routing flexibility make it well suited for all kinds of recording tasks: studio, live recording, etc. Clearly a best buy console in my book.

### | EQUIPMENT SETUP

Audix SCX-25 microphones, Heil PR-40 dynamic microphones, Shure KSM-32 microphones, TASCAM DV-RA1000, Alesis Masterlink, Benchmark ADC-1 24-bit/192 kHz converter, Westlake Interconnects, Esoteric DV-50 Universal DVD-A/SACD player

John Gatski is the publisher/executive of Pro Audio Reivew.

## Vocal Magic



Elvio Fernandes  
Songwriter/Producer

Ace Young  
American Idol  
Contestant 2006

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by Stephen Murphy

# SSL Duality Analog Console/DAW Controller

Appropriately-named Duality adds fully featured digital audio workstation control to a fully analog large-format console.

The concept of combining a digital mixing console with computer workstation control is a natural match. In fact, it's not so much a match as the same thing. In both cases, the same hardware faders and controllers are communicating with digital mixing/processing software, whether residing locally (in the mixer) or remotely (in the workstation computer). At this point, I am not aware of any professional digital console — whether \$500 or \$500,000 — that does not offer external control capabilities. This is no longer news.

What is news is when one of the most respected high-end console manufacturers announces its latest fully analog large-format console is equipped with full-featured workstation integration and control. Combining a "best

of' set of features from its large-format analog consoles with the comprehensive control technology of its AWS 900+ workstation system, Solid State Logic's created just such a beast with its new Duality console.

## FEATURES

Despite stiff competition for a shrinking market, UK-based console manufacturer Solid State Logic could be successful well into the future based simply on the respect and momentum generated by its juggernaut analog consoles of the last three decades. But throughout its recent history, SSL has continued to develop its products and expertise, emerging as an early pioneer in analog surround, computer-controlled analog and fully digital console and workstation technology.

It could be easily argued that the Duality represents the epitome of all that SSL has produced, both conceptually and literally. I say "conceptionally" because Duality's dual functionality adeptly takes a head-on tack towards the large control surface arrays and I/O banks that have made their home where large-format consoles used to reside. I say "literally" because the console literally features the best bits of many of its most popular consoles and, in some cases, when the designers (and the vocal SSL user base, I'm sure) were faced with a devil's choice between two equally popular same-type elements, they included both.

As for specific pedigree, the Duality console features the sharp TFT channel displays of the C-series digital consoles, both E- and G-series

channel equalizer curves, SSL's peak-sensing and RMS/over-easy channel compression circuits, moving-fader and VCA-style (actually DCA) automation, multi-operator (!) Total Recall, the extended-range "SuperAnalogue" mic preamp (first introduced in the 9000 J), as well as a second "Variable Harmonic Drive" mic preamp path — quite the "Greatest Hits" playlist, complete with alternate versions!

The Duality is a 24 track-bus analog console with a central digital routing and monitoring control section. In addition to the 24 output busses, the Duality employs a six-buss main mix output path (divided into pairs A, B and C) that can be configured for 5.1 surround purposes or as three discrete stereo pairs.

Central section features include 5.1 (with flexible LFE source and bass management control), stereo and stereo down-mix output and



The Duality installed in Phoenix, MD's Sheffield Audio/Video Productions

Photo credit: David Fouda

monitoring, four stereo returns with full routing options, stereo analog VU and phase meters, and a LCD bar-graph section with 5.1 and stereo bus meters, 6-bus follow-monitor meters (what you are listening to), stereo solo/AFL meters, 24 track-bus meters and meters for the console's two stereo cue/aux and four mono fx/aux output busses.

The console maintains balanced signal paths throughout and, according to SSL literature, boasts a frequency bandwidth that exceeds 192 kHz.

## IN USE

I had the pleasure of spending a day in Sheffield Audio-Video Productions' (Phoenix, MD) Studio A, exploring their 48-channel SSL Duality console. Sheffield's Jake Mossman and SSL's Don Wershba quickly brought me up to speed with an overview of the console before diving into the finer points of this impressively innovative yet refreshingly easy-to-navigate console.

## FAST FACTS

### APPLICATIONS

Studio

### KEY FEATURES

24 track-bus analog console; TFT channel displays of the C-series digital consoles; E- and G-series channel equalizer curves; peak-sensing and RMS/over-easy channel compression circuits, moving-fader and VCA-style automation, multi-operator Total Recall, the "SuperAnalogue" mic preamp; second "Variable Harmonic Drive" mic preamp path; feature-packed central digital routing and monitoring control section.

### PRICE

Starting from \$240,000

### CONTACT

Solid State Logic | ☎ 212-315-1111  
 ☞ www.solid-state-logic.com

Users of SSL consoles — especially of the analog 9000 K- and digital C-series, and the AWS 900+ natch — will be right at home with many of the features found on the Duality, though there are plenty of new features and twists on some old ones to explore. Those who haven't worked on an SSL previously will face a steeper learning curve, but I can't imagine a professional independent engineer with a couple hours to spare before a session not being able to master the core console and control operations.

Let me first run down the channel strip before getting into the overall console/control-surface implementation — kind of like eating my desert before dinner, as the strip is always the sweetest part of an SSL.

The console breaks away from the traditional in-line model (full channel and monitor paths per strip) and moves to what SSL calls "Split-Mode" architecture. In this similar-but-different, streamlined approach, each channel has the traditional two input sources (channel and monitor) but one shared (via flip, split and routing functions) set of EQ, dynamics, filter, aux and cue paths/processors. Given that the Duality is weighted heavily towards use with a multichannel DAW system (with its own layer of channel and output controls etc.), this approach makes good sense from economic and console-complexity standpoint.

In its most direct configuration, the channel output (CHOP in SSL-speak) is derived straight from one of the two input preamplifiers and routed to any of the 24 track busses, which in turn feeds the DAW system inputs. The CHOP output section provides your "to tape" level control (+/- 20 dB), phase reverse and the option to move the strip's third-order 18 dB/octave high-pass filter (range: 20 to 500 Hz) and second-order 12 dB/octave low-pass filter (range: 3 kHz to wide open) out of the monitor path and into the record path.

The two preamp choices provide one of the most useful creative options I've seen/heard on a large-format console (or any size for that matter). The direct-coupled SuperAnalogue preamp satisfies the "SSL jones" in us all with its ultra-wide and quiet signature sound, while the variable harmonic drive (VHD) circuitry imparts a gentle tube-like quality when set conservatively, and pleasing-to-extreme overdrive can also be dialed in. A flip button routes the monitor ("from tape") signal from the buffered line input into the variable gain amp, allowing post-DAW preamp "processing."

The dynamics section reflects the schizophrenic (I mean it as a complement) nature of the Duality perfectly, providing both the classic RMS side-chain over-easy compressor (with selectable fast-attack mode for that trademark drum compression!) and an aggressive-sound-

ing peak-sensing compressor for overt effects. The gate/expander, uh, expands on the usability of the familiar three-control 4000-series section with an innovative Hold option that turns the release knob into a hold-time control and imposes a fixed release curve. Pushing the eminently useful hi-res TFT channel metering/routing screens (situated above each group of six channels) into a higher level of usefulness, compressor and gating gain reduction is displayed in two small simulated LED meters.

Each channel has its own dynamics section key input for linking gates, frequency-driven compression etc. Adjacent compression sections can be linked for preserving multichannel dynamic relationships (using the voltage-summing method). The aforementioned filter section can also be routed into the dynamics section sidechain, and the whole dynamics section can be placed pre- or post-EQ. The inclusion of a side-chain/key-input listen (routed to the PFL bus) is the cherry that sits on top of this feature-rich slice of SSL dynamics goodness. Mmmm.

With no treatment in sight for its split-personality disorder, the Duality insists on providing even more creative options in the EQ section, simultaneously providing the look and operations of the classic SL4000E EQ section, and the option to switch to the steeper curves and gain/bandwidth interdependency of the G-series. No complaints here! The four-band EQ features two fully parametric mid-band equalizers and high and low shelving equalizers (each switchable to a fixed-Q bell curve).

An Insert In button routes the balanced send/return insert points into the channel input path (post trim). Pressing the Post button cycles the insert point routing to two alternate locations: post EQ and post dynamic section. The Duality features a highly flexible stereo cue and fx sends implementation, including the ability for any fx or cue send to derive its source from a number of points including the default monitor path, the channel output (for latency-free phones mix), or an ALT alternative input (the input not assigned to the main channel path). Any send can be disengaged completely by pressing its knob.

I found the channel surround panning section to be intuitive, well implemented and highly flexible — certainly not the case on most analog consoles whose principal architecture is designed around stereo busses. For 5.1 mixes, the Duality uses XY-style panning, with LR (Left/Right) and FR (Front/Rear) controls plus a LFE bus feed/send and a Focus control, which alters the proportion of signal fed to the L/R busses versus the Center bus. A pan-to-track function routes the LR pan outputs to odd/even pairs of track busses.

SSL continues on page 22 ►

## PORTABLE ACOUSTICS



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Finishing out the traditional bits of the channel strip are two scribble strips (electronic and good-old analog grease pencil type), cut and solo switches (with options in behavior determined by central control and console mode), eight fader group assign switches (for VCA-style group control), a select button (for including that channel in central control routing operations), console and DAW automation controls and a 100mm motorized fader.

One of my favorite parts of the Duality design is its bank of TFT screens, which provide an immense amount of channel information in a logical and legible manner. A generous peak meter indicates analog or DAW levels (determined by a Focus button), and three DAW indicators display channel select, record and plug-in editor status. A channel processing order display indicates the placement order and on/off status of the EQ, filter and dynamics sections, as well as the routing of the channel and monitor path I/O. At the bottom of the TFT channel display is a mix-bus routing status display (bus pairs A, B & C), and routing status to any/all of the 24 track busses.

The central control panel of the console features an expanded version of the AWS 900's DAW master control section, as well as a unified channel routing panel with soft-switch control of many elements of one or several channels' routing, ordering and processing functions. Also in this section are the 24 track-bus output trim knobs (with smart AFL solo for single-bus centered or bus-pair stereo auditioning – very nice!), four stereo returns, cue/sends masters, the traditional dim, cut, mono, and CR volume controls. The monitoring, bus-summing, downmix and external input source routing functions are quite flexible but take a little more time to explore, for this is one of a very few places where a menu/layer system is employed. After some initial confusion on my part (partly due to the logic and simplicity employed throughout the rest of the console), I eventually came to recognize the Duality master section as the All-Knowing, All-Powerful Supreme Being it is.

Hmmm.... I have this nagging feeling like

I'm forgetting something. Oh yeah, the other half of why it's called Duality. SSL designed the Duality console with DAW systems at heart, with major consideration given not only to its direct mix-surface control of software applications, but also to the console's channel/monitor audio-path architecture, the seamless switching between DAW and audio console control modes, and the instant per-channel auditioning without the need to reconfigure all channels.

The Duality DAW control is compatible

The impressive Duality carves its own path straight through the clutter of linked controllers, computer peripherals and digital mixing engines

with any workstation that supports the HUI or MCU (Mackie Control Universal) protocols. Control templates exist for the most popular DAW systems in professional use including Pro Tools, Nuendo and Logic. The console connects to the DAW via 16 (eight in/eight out!) MIDI ports in order to emulate an array of linked HUI/MCU controllers and provide a dedicated MTC/MMC communication path.

The console focus button globally switches the Duality between controlling the analog console and the DAW. The central control section of the console features dedicated DAW navigation/motion controls, a plug-in editor, and DAW master control section (a la AWS). In addition to the fader, mute, solo and select functions expected of any control surface, each channel strip features its own "D-pot" rotary control and secondary solo/cut button set. In addition to providing typical rotary control functions (pan, sends, parameter adjust etc.) in DAW focus mode, the dedicated D-pots are also available to control DAW functions in Analog focus mode. A hybrid Digital In-Line mode allows the channel D-pots and associated solo/cut switches to control analog levels and channel status while the rest of the console is in DAW focus. This is a very cool and much appreciated design — no matter which focus mode you are in, you can always reach for and adjust the most immediately needed functions from the other mode.

In DAW focus, channel meters duplicate the DAW meter functions including pre/post

and mono/stereo settings. The electronic scribble strip switches from displaying fader number to DAW track names (albeit, highly condensed). The analog scribble strip continues to display your grease-pencil scrawl, regardless of focus mode.

One thing I think would be extremely helpful would be for the electronic scribble strip to steal the first two characters to display the DAW track number. When swapping between multiple banks of DAW tracks and scrolling the faders up and down by single tracks, it's very easy to become frustrated trying to quickly find a track that from strangely abbreviated names (a lot of my mix work comes from other studios, and the client usually doesn't want to pay me to rename 48 tracks for my control surface's limited display) may or may not be currently accessible on a fader. Just a general wish...

In addition to the meat-and-potatoes control functions, the Duality also provides plenty of in-depth control over DAW transport, channel routing, automation, edit window, mix window, plug-in editing, tempo and timeline, group enable/suspend, and I/O assignment functions. Between its dedicated (and excellent) plug-in editor, depth of functionality and fantastic focus mode switching, the Duality is easily the best control surface — dedicated or mixer-associated — that I have encountered. Download the Duality manual and check it out.

## | SUMMARY

With the introduction of its Duality console, Solid State Logic has very effectively leveraged its analog circuit design-, console architecture- and digital control-expertise, as well as its immense amount of goodwill in the industry to create a very comfortable and logical direction for the company. The impressive Duality carves its own path straight through the clutter of linked controllers, computer peripherals and digital mixing engines that have taken over many control rooms and mix stages. Duality elegantly and simultaneously provides SSL's unassailable analog recording paths and large-format stereo and surround mixing functionality, full-console automation, multitudes of non-virtual (i.e., real) SSL compressors and equalizers, Total Recall of all analog settings, comprehensive, high-resolution control of DAW mixers, comprehensive integration with workstation editing and processing functions, and — best of all — any combination thereof, at any time, with no reconfiguration.

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



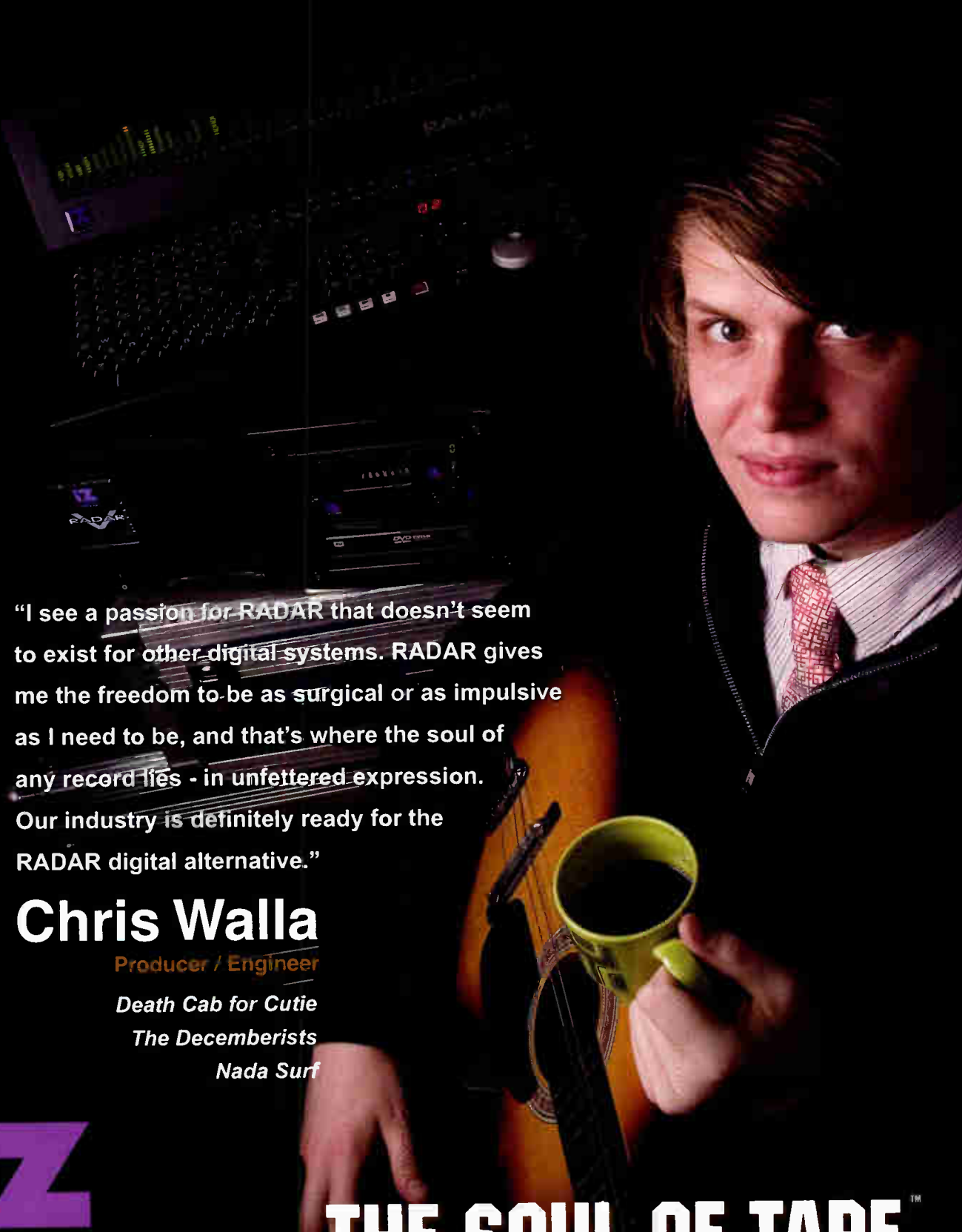


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World Radio History

Chris Walla at The Hive Creative Labs - Vancouver, BC



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by Strother Bullins

# ADAM A7

## Closefield Monitor

ADAM's signature A.R.T. ribbon is now within the reach of every working studio pro.

I had my first experience with ADAM studio monitors several years ago during a mix session at producer/engineer Michael Wagener's Nashville-area studio, Wireworld. Wagener, known for constantly improving his personal arsenal of recording gear, had recently acquired a 5.1 set of S3A active midfield monitors.

I suddenly became very intrigued with the ADAM brand, and specifically with their signature A.R.T. (Accelerated Ribbon Technology) tweeter. I was admittedly listening to Wagener's own tracks and mixes at his personal studio (and the man mixed *Master of Puppets*, folks); so lots of monitors would sound impressive in that environment, I reasoned.

Now I fully understand what Wagener was experiencing with his S3As. I've had ample time to evaluate the company's latest — the far-more-affordable A7 closefield monitor — in my own production/critical listening environment. I've listened to my own recorded tracks and mixes, as well as other all-time favorite mixes I know intimately and/or have heard in a variety of great acoustic environments

through great monitors. And, thanks to this pair of A7s, I have had a very impressive experience, indeed.

### FEATURES

The ADAM A7 (\$999 per pair) is a two-way, ported, active studio monitor featuring the A.R.T. ribbon tweeter and a 6.5-inch mid/bass woofer; each transducer has its own internal 50-watt RMS amplifier. Maximum power consumption of the A7 is 100 watts. Frequency response (+/- 3dB) is 46 Hz – 35 kHz, maximum SPL at 1 meter is 105 dB, and the internal crossover frequency is set at 2.2 kHz. The cabinet measures 7 inches x 13 inches x 11 inches and weighs just under 22 pounds.

The front of the A7 features the centered A.R.T. tweeter directly above the also-centered woofer. A bass reflex port is below the woofer and to the bottom left. Then, to the right, is a control panel featuring a power switch, blue LED on/off/standby indicator and a handy detented volume control. (Cheers amongst the "mouse crowd" for that last one — whoo-hoo!)

On the back of the A7 is the AC input, voltage selector switch, two signal input types — XLR and RCA — and a three-knob control panel. The panel offers tweeter voltage gain adjustment and two room EQs: shelving filters located at either end of the frequency, 150 Hz and 6 kHz cut-offs, respectively.

The most notable feature of the ADAM A7, as previously mentioned, is the A.R.T. ribbon

tweeter. This, truthfully, could warrant its own separate article. The A.R.T. ribbon, based on the original works of Dr. Oskar Heil in 1972, moves high-frequency sound waves by the design of its folded diaphragm, essentially squeezing air in and out of the monitor. (Visit ADAM's website for more information.)

### IN USE

The two DAW-based rock re-mixes I performed using the A7s — ones in which I made a few crucial EQ adjustments, mostly on vocals and acoustic string instruments, plus one "thick wall" rhythm guitar track — were clear improvements on my original mixes, and took a surprisingly short period of time to complete.

Elements of the mix that had originally struck me as potential "slackers" — things that simply could've been recorded better — were more obvious than ever, and I appreciated the tip. Those mixes in the end proceeded to translate well onto every other system where I played them (and, in direct A/B comparison, better than my original mix did).

Once I was sold on A7 production performance I brought out my trusty evaluation reference CD: it's a disc filled with music I love that also represents a few personal production standards on a variety of musical styles.

The fun began with U2's "Sister Havana" (from *Saturation*), which really stood tall as I listened to it three or four repeats in succession. "Sister Havana" is a great "kick drum" song, and to my ears my favorite "standard" modern rock mix/production of all time. The A7s nailed it, as expected, and it was just as I had recalled it in some of the best mastering studios and tracking rooms I've had the opportunity to use. Sometimes it was even better. The crisp, non-edgy, non-fatiguing and pleasant transients that the A.R.T. ribbons uniquely deliver carried the production honorably. And, despite their small size, the A7s didn't slouch on this particular song's full,

ADAM continues on page 26 ►



## FAST FACTS

### APPLICATIONS

Studio, audio-for-video, broadcast, edit suites

### KEY FEATURES

ADAM A.R.T. ribbon tweeters, two shelving filters, tweeter voltage gain adjustment, front panel detented volume control

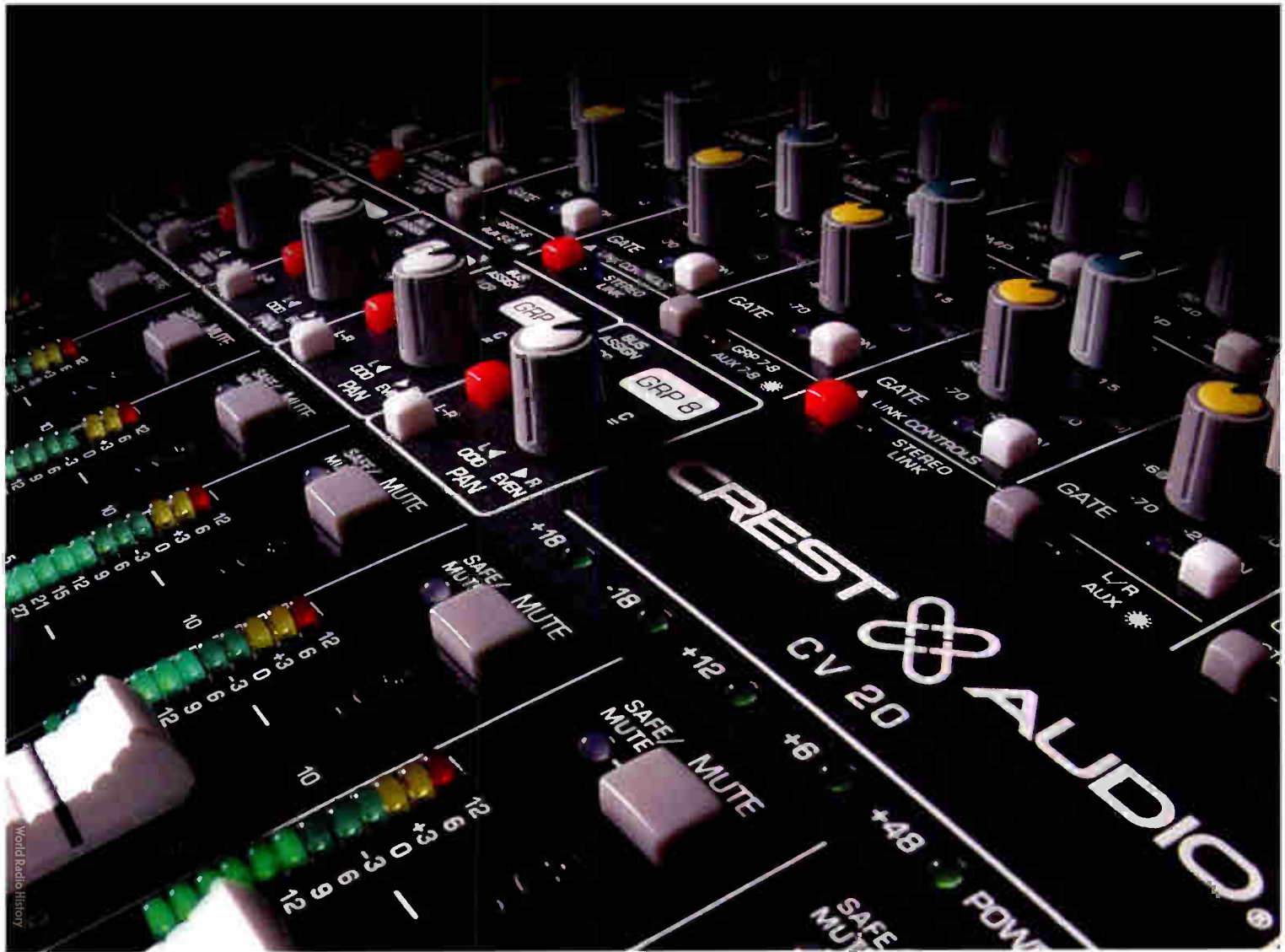
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\$999 per pair

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tight/punchy and well-balanced bottom end, either.

Yonder Mountain String Band's "Bloody Mary Morning" (from *Mountain Tracks, Vol. 3*) and Sergio Mendes' "Berimbau/Consolacao" (from *Timeless*) next showed the fine, unique acoustic detail of these live bluegrass and Latin music performances, respectively. Banjo, guitar, mandolin, bass and fiddle — with appropriately full, round bottom end and nice string transients throughout — sounded exceptionally detailed, not hyped. On the Mendes tune — the bass-heaviest/busiest cut on the disc — the A7s' tight bass performance was controlled and lots of fun to hear. I may have thought in this particular case for a moment or two about having a sub — a sub coupled with the A7s could likely make your grandmother dance spontaneously. A sub, however, was far from necessary.

#### | SUMMARY

Simply said, everything I heard during my A7 evaluations seemed to be the truth (and nothing but the truth). So, can you handle the truth? Some engineers can't and, most likely, they are the few that wouldn't enjoy what the A7s have to offer anyway. Therefore it is my opinion that constant seekers of acoustic truth

#### | SECOND OPINION

I set up the A7s in an equilateral 3.5-foot triangle and kept my ample 15-inch active subwoofer engaged initially, seeking to find differences from my normal 2.1 setup. The most obvious change was the sound of high-end transients, which the ADAM ribbons treated with little hype and negligible distortion. The attack portion of the envelope sounding clear, quick and open; the decay seemingly smoother than "normal" with less of that "after-ringing" or smearing we're so used to hearing, particularly with aluminum dome drivers.

I exclusively use cloth dome tweets, so it was unusual and nice to hear a new texture still gentle on my ears! I found these qualities to make for longer periods of fatigue-free listening at reasonable mix levels. Fatigue still set in quickly at higher levels, despite the more pleasant sound.

I always look to find a bump (or dip) of response in monitors, and here I had to look very closely. I only heard just a bit of emptiness around 700 - 800 Hz (a good bit lower than the 2200 Hz crossover point where I expected inaccuracy), and a touch of aggressiveness at about 4 kHz. These are only minor quibbles, as overall the mids were quite faithful and honest (noticeably with

## PRODUCTPOINTS



- Very accurate, non-fatiguing and pleasing performance
- Most affordable ADAM monitor available
- Versatile small size and front-panel volume control



- No negatives worth mentioning

## SCORE

*The ADAM A7 is a unique, top performer at a more-than-reasonable price*

who need a fairly affordable powered close-field monitor should give the ADAM A7 a try. They will work well for critical listeners in a variety of differently sized control rooms, recording environments, post-production and broadcast suites, or wherever more accurate, non-fatiguing and pleasant audio is considered a virtue.

*Strother Bullins used to blast Saturation: Track 1 over and over again from those really big Kinoshita/Hidley mains.*

guitars and vox), especially around 1 kHz where any hype can be disastrous.

The bottom end of the A7s had an unexpected fullness that was impressive. The woofer, with rigid carbon fibers stiffening up the cone, got quite animated for me, reaching impressive distances before finally reaching its max. I then disconnected my sub and was reminded why I have one, as the A7s put out admirable bass, but nonetheless needed reinforcement from about 80 Hz and below. Yet, these were louder, fatter and more "big" than I could ever reasonably expect with only two 50-watt amps within.

These monitors' extremely uncolored highs, largely accurate mids and some nicely punchy bass are an ideal choice for project studios, edit suites and video editors (who will really appreciate their articulate and trustworthy dialogue response). Larger music production rooms will love that forgiving top end, but would probably want to add a sub for the whole picture. Hmmm, I bet some of those bigger ADAM mains with the same accelerated ribbon technology would make for some great "ultra-loud/hype up the client" listening sessions...

— Rob Tavaglione

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# BENCH TEST

## ADAM A7 Closefield Monitor

### BENCH MEASUREMENT DATA

#### Frequency Response.

On-axis 70 Hz to 20 kHz +/- 3.1 dB

#### Bass Limit:

71 dB SPL @ 62 Hz @ 2 meters (<10% Distortion)

#### Control Action:

Room EQ < 150 Hz - 6

Actual Response -1 dB < 150 Hz

Room EQ < 150 Hz + 6

Actual Response +1 dB < 150 Hz

Room EQ > 6 kHz - 6

Actual Response -6 dB > 6 kHz

Room EQ > 6 kHz + 6

Actual Response +6 dB > 6 kHz

Tweeter Level +4 dB

Actual Response +3.4 dB > 1.5 kHz

Tweeter Level -4 dB

Actual Response -3.9 dB > 1.5 kHz

Front Panel Level +6

Actual Response +16 dB

Front Panel Level -60

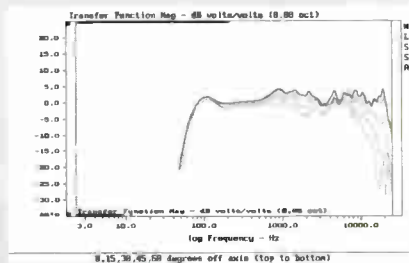
Actual Response -37 dB

**Note:** The Bass Limit of the speaker is the Sound Pressure generated at 2 meters. The figure of merit 10-percent distortion is used because operating characteristics of drivers (using DLC Design DUMAX) shows that when a speaker has reached the end of its linear operating range (BL product has fallen to 70-percent of the rest position value or the suspension compliance has stiffened by a factor of 4) the unit will still sound clean, but distortion increases exponentially with further drive. With powered speakers amplifier output or limiting may also limit sound pressure capability.

### BENCH MEASUREMENT COMMENTARY

Although the A7 is called a "closefield" monitor a more correct term would be "direct" field monitor where the listener will be in the direct acoustic field of the speaker. Closefield has a specific meaning when it comes to loudspeaker measurement where the microphone is placed within a half-inch of the radiating source. This technique is used when an anechoic chamber is not available. However the closefield technique, by its nature, will ignore front panel resonances and cabinet diffraction. While this is not significant at low frequencies combining near field measurements requires estimating the individual radiating

area of woofers and ports that often optimistically reports actual low frequency output. All measurement results here have been measured at a full two meters in a large room on a 6-foot stand. Using time windows then gives equivalent anechoic results above 200 Hz



ADAM A7 frequency response

including front panel reflections, cabinet diffraction and true acoustical summation of all drivers and passive radiating elements.

Because of its size, the A7 has limited

bass capability and frequency response exhibits a 4 dB elevation between 500 and 2000 Hz. The system is also relatively directional although the 19 kHz tweeter peak can be seen even at 60 degrees off-axis. An additional 6 dB of SPL at 62 Hz is available if the user is willing to accept 20-percent distortion. Optimal power and electronic protective circuitry is another advantage of active speakers; the system can be designed to prevent drivers from burning out and gross overload even when the speaker is used outside its proper operating range.

The EQ below 150 Hz has the specified turnover frequency but far less effect than anticipated. The front panel master level control has far greater effect in the upper half of rotation than specified so users may wish to use the control with care. The tweeter level and high frequency room EQ function as specified.

— Tom Nouisane

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by Dr. Frederick J. Bashour

# AEA TRP Ribbon Microphone Preamplifier

A high-gain, sweet-sounding preamp for non-phantom-powered ribbon microphones.

I bought and reviewed an AEA R84 ribbon microphone in 2003, and ever since I've been searching for its perfect preamp. I admittedly

of Manley MicEQ500s (usually dedicated to a pair of tube AKG C60s hanging over the Mason & Hamlin piano) and a Yamaha i88x



have a whole bunch of boutique preamps in my studio, but most of them are dedicated to stereo pairs of other microphones. My R84 needs a lot of gain and — even more importantly to the quiet acoustic sources I usually record — it seems not to tolerate long cable runs in my studio without picking up a small amount of hum.

I've found only two preamps in my studio to date that both amplified it without adding a lot of noise and did not change its perceived frequency response. They were one of my pair

(an mLAN unit I had here for a while during the time I was reviewing the Yamaha 01x mixer). The Yamaha preamp was dead quiet and made the mic sound good, but it didn't even have real analog outputs; I had to use an insert point. The Martech MSS-10 (PAR, 10/97), another seemingly obvious contender, makes my big Neumann M 249s or U 47 (or the new M-Audio Sputnik) sound punchy and silky despite its high gain and low noise. But it had a low enough input impedance characteristic to seriously reduce my R84's low end, despite flattering most of my other vocal mics.

I mentioned my preamp dilemma to AEA's Wes Dooley last spring, and he told me to just wait a month or so and he'd have a surprise for me. So I was elated come June when I received a prototype unit of "The Ribbon Pre," heretofore referred to as "TRP." It was small, stereo and (one channel of it) immediately became the R84's dedicated mic preamp; I purchased it on the spot, despite its prototype status.

## FEATURES

The AEA Ribbon Mic Preamplifier is a high-gain and high impedance, minimal-path FET mic preamp. It's designed by Fred Forssell, who's designed many well-known high-end vacuum tube and solid state preamps, plus equalizers for various manufacturers (as well as for his own company, whose

equalizer I reviewed in PAR, 9/97). AEA advertises that this low-noise two-channel preamp is intended for ribbon, moving coil and tube microphones that do not need phantom power circuitry. This "lack of a feature" allows for numerous electronic improvements that result in simpler circuitry. It also rids the unit of the ubiquitous phantom power blocking capacitors found in most other preamps. Zener diodes on the input protect the input stage if external phantom power is applied from elsewhere.

TRP is a two-stage transformerless solid state amplifier. The gain in its first stage is adjustable from +6 dB to +63 dB by a 12-position Grayhill switch. Another 20 dB of fixed gain is available after the output level potentiometer. (This is similar to the configuration of the MSS-10, though it uses a circuit with input and output transformers.) TRP also features polarity reversal and high-pass filter switches and LEDs that show operating and overload level metering. It has two sets

of outputs: balanced XLRs at +4 dBm and unbalanced 1/4-inch connectors at -10 dBm [AEA says shipping models will have outputs of +4 dB (balanced) and -2 dB (unbalanced); Fred's was a special prototype. — Ed.] A nice touch is its laser engraved legends and single line schematic on the top cover, reminiscent of old-school Roland equipment. TRP is a half-rack-wide unit, one rack unit high (8.5 inches x 8.5 inches x 1.7 inches), and weighs a mere two pounds. Its hefty external "line lump" power supply (which produces ±17.25 VDC) weighs another pound and a half.

## IN USE

I initially just used channel one of TRP with my R84, since it made it sound so good. The sound the R84 produced was as quiet as my quietest tube mics and, more remarkably, seemed more "hi-fi" than it had previously been (in that it appeared to have higher highs and lower lows). I thought to myself, "I haven't really heard this mic until now." I eventually started to get curious about how TRP, with its ultrahigh input impedance and special minimalist circuitry, would handle my other mics.

The only other passive ribbon mics I own are a pair of vintage Beyerdynamic M-500s, which I use live for female vocals. I find these hypercardioid Beyer ribbons to really excel in that application. The little bit of noise their low

AEA continues on page 30 ▶

## FAST FACTS

### APPLICATIONS

Studio, location

### KEY FEATURES

Two-channel; high-pass filter; phase reverse; Fred Forssell design

### PRICE

\$845

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output level brings out live in most preamps is rarely a problem, and their apparent built-in low-end rolloff is usually desirable (when it isn't it can always be cured by a little mid-bass boost). That's what I thought until I tried them with TRP. I just unplugged my R84, substituted one of the M-500s, and wow!

It had about the same output level as the

R84 and, although it obviously lacked that mic's large ribbon creaminess, it definitely sounded like a completely different M-500. It was big, bright, and had plenty of low-end punch. I wish I had a Neumann KMS 105 around with which to A/B it, because it really did sound like an M-500 on steroids. The sound in my monitors, meanwhile, was dead

quiet, and the Beyer still had that wonderful airy upper-midrange characteristic which many female vocalists love; it just sounded "bigger and better." I would bet that the still-available Beyer ribbons — like the 130, 260, etc. — would also shine with TRP. I no longer own any passive Royer ribbon mics (just their active versions), but I can't imagine an SF1 or stereo SF12 not sounding better (and quieter) with TRP than with just about any other preamp.

The only dynamic mic I own is a vintage Beyer M-88 and, besides requiring about 12 dB less gain from TRP than the ribbon mics, it sounded considerably better than an SM57 used with a cheap preamp, that's for sure!

## PRODUCTPOINTS



- Amazingly quiet
- High gain
- Sound quality



- No power switch

## SCORE

*A great value in a non-phantom-powered mic preamp*

I restored the R84 to TRP's Channel 1, and then started hooking up my tube mics to Channel 2. There was less of a difference between TRP and my other good preamps with tube mics than with the ribbon and dynamic mics. So I then started comparing preamps instead, keeping the mic invariant. One day I used my 3-micron U47; the next day (when I was writing the Sputnik article) I used the Sputnik.

Those who have been reading my mic preamp reviews since *Pro Audio Review* began publishing them might believe that I own only tube preamps. And, in fact, you'd be partly correct. I use my various Manley preamps and my D.W. Fearn VT-2 just about every day, but I also own three high-end solid state preamps, and it was those which I A/Bed with TRP. They are my "vintage" Millennium Media HV3C — the original model, with the built-in Apogee 500 A/D board (*PAR*, 11/95); my Dan Alexander stereo unit built with authentic Neve parts (*PAR*, 1/03); and the aforementioned single-channel Martech MSS-10. Here's where TRP stands in that lineup: the Dan Alexander preamp has the "heaviest" and most-colored sound (as would be expected) and the MSS-10, with its multiple modern transformers and specially-voiced circuitry, has a wonderful silky coloration that often

**AEA** continues on page 79 ►

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World Radio History



by Mel Lambert

# Mackie Digital X Bus X.200 Medium- Format Production Console

The Mackie X.200 Digital Medium-Format Production Console is definitely a product for our age. Digital audio workstations are ubiquitous — with current versions offering increasingly powerful user features and functionality — so a standalone console has to offer a lot more bang for the buck than it did a few years back. The Digital X Bus console, retailing at close to \$13,000, is a major winner in terms of simplicity of operation and front panel features. The console's ergonomic layout, co-developed by Mackie and SaneWave, puts assignable hardware controls directly in-line with the corresponding

rarely mounted at the right angle — leading to shoulder and/or wrist pain — and seldom offer much more functionality than I can achieve with a conventional mouse, track ball or graphics tablet. The X.200 changes all that. The combination of see-and-poke for rapid system setup, routing and function assignment is not only easy to master, but within just a few minutes you wonder how you used to do it in the olden days.

Both the X.200 console and its larger cousin, the X.400, accom-

or Windows PCs. The console's faders and rotary controls can additionally be mapped to command a variety of DAWs, including Pro Tools, Logic and Nuendo, using HUI and Universal Control protocols. This is useful considering the X-Series' PC-based internal mix engine supports VST plug-ins for signal processing — a neat feature if you have a favorite EQ or dynamics sound that you want to use on a remix session, while replaying edited tracks, for example, from a companion DAW.

The two Digital X Bus models — X.200 and X.400 — share a basic feature set, but there are interesting differences. The X.200 is a “meat-and-potatoes” tracking/mixing console, while the X.400 is positioned for surround sound projects and post production. The X.400's dual processors offer additional mixing bandwidth, while an extra card slot provides 24 additional I/O channels, for a total of 92 inputs and 100 outputs at a 96 kHz sample rate, plus extra surround functionality via up to 24 assignment busses. The X.400 also ships with a preloaded Universal Audio UAD-1 DSP card for instant access to the UA's processing plug-ins; the same card is an option for the X.200.

The Digital X Bus's modular architecture picks up added flexibility via the back panel chassis, which allows the analog/digital complement to be loaded according to the user's requirements. One is able to accommodate a busy tracking date

functions displayed on a pair of 15-inch touch-screens. (These, interestingly, display an inverted image during system boot. Its designers decided that the viewing angle and clarity would be enhanced by physically inverting the two TFT display panels in front of the user; a simple setup command simply flips the video feed once the Windows OS system is implemented. A neat trick!)

## THE LOOK

I have always shied away from touch-screens for a number of reasons. They are


moderate 192 kHz sample rates. The X.200 accepts 68 inputs and offers 76 outputs while running at 96 kHz; half these I/Os are available at the higher sampling rate. A modular rear panel I/O card frame supports as many as 64 analog mic/line inputs with digitally-controlled mic trims; digital I/O cards accept AES/EBU, TDIF and ADAT/optical format signals. Optional FireWire-capable I/O cards handle connection of audio to Mac

with additional mic sources, for example. (Two cards are fitted as stock: a Sync Card, which provides word clock and LTC I/O; and a Mix Out Card that provides control room and headphone feeds, plus analog and digital main mix bus outs.)

Currently available I/O cards include the Mic/Line 4, a four analog mic/line input card with digitally remote-controlled mic preamps and 48V phantom power per mic input; the Mic/Line 8, an eight analog mic/line input card with preamps and 48V power; an eight balanced analog line level I/O (jumper selectable (+4 dBu/-10 dBV)); eight ADAT optical digital I/Os at 96 kHz (four at 192 kHz) or







eight channels of TDIF digital I/O at 96 kHz; eight channels of AES/EBU digital audio I/O at 96 kHz; plus a FireWire Card that carries up to 24 I/Os at 44.1 kHz/48 kHz (eight I/O at 96) — all you need is an OSX Core Audio (Mac) or ASIO/WDM (Windows XP)-compatible DAW application.

#### DYNAMICS AND OTHER GOODIES

The X.200 Music Production Console boasts a powerful dynamics section — a full-feature compressor/limiter plus gate/expander — and four-band parametric EQ section on every I/O signal path, plus a total of 12 aux sends, selectable pre/post-fader, and a pair of DSP insert points per channel capable of accepting VST plug-ins. The unit supports full 7.1-channel surround mixing and monitoring. The automation system offers both event editing and a choice of dynamic/real-time or scene/snapshot storage and recall. Mic preamp level settings are stored with automation data, a useful feature. The X.200 behaves in all other aspects like a conventional console, with tape machine-like transport controls with auto locator memories — a rear-panel nine-pin connector supports common serial protocols, while MIDI Machine Control also is available for computer-based and similar transports. The console will also generate and/or chase-sync to external LTC and MTC. And the system includes a comprehensive control-room section with selectable monitoring, talkback and phones for interfacing with the immediate environment.

Internal assignment is via an array of eight mix/surround busses that, in turn, can be routed to main mix, channels or outputs. The console additionally boasts eight VCA-style virtual groups, plus channel linking. And onboard processing doesn't stop at EQ and dynamics; the X.200 incorporates built-in stereo reverb and delay effects that can be accessed via conventional aux sends and returns, although the optional UAD-1 plug-ins offer a very powerful alternative. (As your might expect from a power developer like Universal Audio).

Once a physical I/O configuration has been sorted out, the time comes to set inputs to virtual channels and busses, sends and monitor feed to output ports. This is when the X.200's dual 15-inch touchscreens really come into their own. The screens map all major features and functions as virtual controls, using large, button-style graphics and a minimum of pull-down menus. A combination of sensitive touch detection and Mackie's proprietary tracking algorithm produces screens that are remarkably accurate, enabling precise moves to be achieved via the GUI interfaces. Each of the 24

on-surface channels is equipped with a real-time rotary control, and a 100 mm Penny & Giles motorized fader. The rotary encoders are mapped to the corresponding on-screen graphics directly above them, including such everyday functions as mute, solo and pan, with faders and switches remaining in hardware, just the way your fingers like them. And for control of everyday functions — adding channel labels, editing mix titles and other chores — the console boasts standard USB ports for an

external mouse, keyboard or jog wheel.

But the \$64,000 Question is surely, "What's the big deal?" The market is full of assignable 72-channel/multibus/24-fader digital consoles. What sets the Mackie X.200 apart from the rest is its processing horsepower. Its designers have taken a stable Windows OS and linked it to a powerful mixing and processing engine. The latter is based on a proprietary architecture using a PCI-based accelera-

**MACKIE** continues on page 34 ►



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MACKIE Continued From Page 33

tor card that employs a Field Programmable Gate Array (FPGA) chip to handle the conversion of integer to floating-point math. The X.200's main processor, delegating the heavy lifting to the accelerator card, can handle complex mixing while simultaneously running an array of VST plug-ins. A big bonus is that such a topology ensures ultra-low total system latency from analog input to output - real-time overdubs are a viable feature of this console, unlike other medium-format digital consoles, and some DAWs.

Laid out in front of you on the business end of the X.200 are two banks of 12 input/output faders plus a master, with onscreen controls that fall readily to hand, plus easy-to-access stereo/surround assignment and monitoring functions. Running in-channel signals to outboards is easy to set up — "outboard" being off-console units and/or virtual processing within the box — and then to add the results to whatever outputs are being used, either tracking busses or a final mix. The transport section behaves as advertised, while the

jog/shuttle wheel is weighted just right for swift scrolling and tape/disc/DAW transport control. Metering is more than comprehensive, with appropriate displays where you need them, and user-adjustable ballistics on the console-wide meter bridge that offers a variety of meter-scale ranges. Control room monitoring can be set to feed a pair of headphones — really useful if you are tracking within a rehearsal room for example, or a dedicated overdub environment — plus ganged surround-level controls and a handy reference-level adjustment. (The X.200 features a built-in white/pink noise and variable-frequency generator with a spectrum analyzer for checking/tweaking system response.)

All in all, mixing a multichannel project on an X.200 Digital Production Console is a refreshing experience. The system is quick and agile and, in my experience at least, totally bulletproof. The control surface is compact but totally adequate with plenty of on-surface assignable controls and a wide landscape set of touch controls. The system is capable of

running at 192 kHz sample rates, extending its usefulness to classical music, mastering and scoring sessions — although many users would be pushed to hear any difference between 24/96 and 192! — with full connectivity to popular DAWs and recorders via flexible analog, digital and FireWire I/Os. The EQ and dynamics section are easy to learn and powerful in operation. The Mackie Digital X Bus X.200 is a console for all reasons.

My sincere thanks to Jeff Goodman, who until recently served as Mackie's X Bus Product Manager, for a great demo session, and Michael Warren, who provided access to an X.200 at the MW Audio showroom close to Marina del Rey, California.

For more information contact Mackie at 425-487-4333 or [www.mackie.com](http://www.mackie.com).

*Mel Lambert has been intimately involved with production and broadcast industries on both sides of the Atlantic for more years than he cares to remember. Now principal of Media&Marketing, a Los Angeles-based consulting service for the professional audio industry, he can be reached at [mel.lambert@MEDIAandMARKETING.com](mailto:mel.lambert@MEDIAandMARKETING.com); 818-753-9510.*



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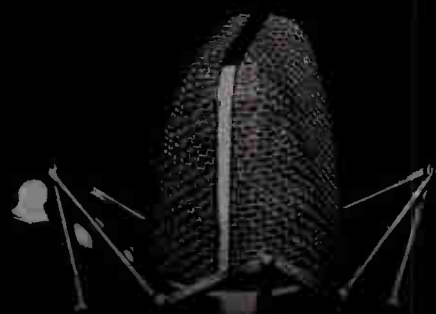
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by Dave Martin

# Crowley and Tripp Naked Eye Ribbon Microphone

A ribbon mic offering Crowley and Tripp quality with a lower price.

A decade ago some folks noticed the new millennium approaching, prompting them to make predictions about audio technology's future. Some of these predictions have come to pass (downloading, high sample rate/high bit rate recording technologies, digital workstations capable of recording and playing back more than 100 audio channels), while others have not (10.2 surround, anyone?). But no pundits predicted that the 21st century would bring in a renaissance of ribbon microphones.

Around the mid '90s pretty much the only ribbon microphone manufacturers were the UK-based Coles, whose 4038 was a '50s design, as well as Beyerdynamic. Though there were old ribbon mics aplenty around in most studios, they had been largely superceded by newer technologies with higher output and greater bandwidth. In 1998 Dave Royer introduced his R-121 — seemingly reintroduced a generation of engineers to the ribbon mic sound. The R121 was based largely on an old B&O microphone;

Mr. Royer has since introduced a number of new mics. In a similar vein Wes Dooley at Audio Engineering Associates (AEA) produced a faithful recreation of the venerable RCA 44 and has since come up with a number of his own designs.

The microphones made by Crowley and Tripp don't appear to be based as much on vintage design as they are a fresh approach to ribbon mic technology. The first three microphones released by the company — the Proscenium, Soundstage Image, and Studio Vocalist — are designed, despite their similar outward appearance, with quite different voicings. The Proscenium is tailored more towards a classic ribbon microphone sound, the Soundstage Image has a more natural low and mid-range (though with a more pronounced roll-off above 10 kHz), while the Studio Vocalist is voiced with a rising high frequency response. Now C&T has introduced the Naked Eye: a microphone combining the two flagship microphones' sounds at a much more affordable price of \$745.

## FEATURES

The Naked Eye departs from the earlier C&T models in a couple of areas: it's a bit smaller than the other product in their line and frequency response is asymmetrical. Each side of the Naked Eye has a distinctive sound. The front side is voiced similarly to the Proscenium, while the back of the mic is patterned after the Studio Vocalist. This gives the Naked Eye quite a bit of versatility. The Naked Eye ships with an "innovative low diffraction rotary mount" (discussed below), and was designed for low noise plus high output.

## IN USE

We record a lot of acoustic projects at my studio, Java Jive, so the Naked Eye saw use on numerous acoustic instruments. One of the first was an overdub session with dobro wizard Rob Ickes. Rob, like a number of Nashville session players, carries a favored microphone into sessions. But we both liked the Naked Eye on his Scheerhorn dobro. In general, we'd set up the Naked Eye beside the microphone usually used in a given application and compare. If the Naked Eye's front side sounded too dark, we'd rotate it to the back side.

Placed about a foot in front of a mandolin, the Naked Eye's front side gave a sound more reminiscent of Bill Monroe circa the '50s. This is not too surprising, since Monroe's recordings from that era were done with RCA ribbon mics. The back of the Naked Eye was closer to a contemporary mandolin sound. We achieved pretty much the same results when recording banjo; the front of the mic worked great for more "old school" sounds, while the back added a bit more high end definition. The Naked Eye's back wasn't as bright as the Neumann-Gefell M582 valve microphone I usually use in those applications, but it certainly worked well with the tracks we were recording. I mentioned earlier that the Naked Eye alone worked well on Rob Ickes' dobro. I also used the front side of the Naked Eye along with a M582 (with the two mics

panned left and right) when later recording another dobroist. The resulting sound was quite nice; the Naked Eye picked up the warmth of the instrument and the M582 captured the sparkle.

There's more to using the Naked Eye, however, than simply choosing either the mic's front or back side. Like most ribbons, the Naked Eye has a fairly pronounced proximity effect; judicious placement of the microphone allows for either using the low end boost (captured close to the source) or not (situating further from the source). We also recorded a bouzouki for two different songs on this project. For one of the songs the Naked Eye's back was put about 4" in front of the instrument to use proximity to add fullness. On the second track we backed the mic to somewhere around 18" from the instrument to capture more of a jangly sound that wouldn't



## FAST FACTS

### APPLICATIONS

Studio, remote recording

### KEY FEATURES

Various characteristics of Crowley & Tripp's top ribbon microphones; versatile on both acoustic and amplified instruments; handles high SPL well; affordable

### PRICE

\$745

### CONTACT

Crowley and Tripp | ☎ 508-231-4515  
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take up as much space in the track.

Fiddles can be fairly problematic instruments to record. Most folks want to close-mic them, and the resulting sound is usually pretty obnoxious — overly bright when not down-

## PRODUCTPOINTS



- Lower cost
- Sound quality
- Solid construction



- The microphone mount

## SCORE

An excellent product at an excellent price.

right “screechy” (this is exacerbated when using most large diaphragm condensers). The Naked Eye avoids this problem altogether. Fiddle (and viola) recorded with the Naked Eye were warm yet detailed. Typically, they would sit in the mix with minimal EQ, if any at all.

The Naked Eye stood up to some fairly

robust sound pressure levels. It was used on the low rotor of a Leslie cabinet (and I would have happily used it on the upper rotor if I'd a second one), against the grill of a Randall half stack amp and close to a trumpet player's bell with no problems. It also seems to have a higher output than most vintage mics; even with the Naked Eye on fairly quiet acoustic instruments, no preamps I tried maxed out.

My only issue with the Naked Eye is with the mount. Compared with the mounting method of the more expensive Crowley and Tripp microphones — for that matter, compared with most microphone clips — the mount supplied with the Naked Eye seemed to be unbalanced and unwieldy. Getting the mic from its box to the stand requires too many actions for my tastes; and the mic can't be put on a stand without setting parts of the stock mount (or the mic itself) down. Once attached to a stand, the Naked Eye doesn't hang straight. I generally ended up grabbing a generic shock mount from the mic closet to use with the Naked Eye — it was quicker and worked better than the mount supplied with the microphone. I realize that I seem to be complaining too greatly about this one thing, but a simpler mount

would be pretty easy for the company to do. Regardless, I'm very pleased and impressed with everything else about the Naked Eye.

## | SUMMARY

The Crowley and Tripp Naked Eye is an excellent microphone both for those inexperienced with ribbon mics and for those who have a closet full. The Naked Eye's asymmetric design — coupled with the (relatively) high output — means not only are a variety of tones available, but also the user isn't limited to a few high-gain preamps. I had the opportunity to try out a couple of other microphones in the C&T range a few months ago, and the Naked Eye has the flavors of the C&T Proscenium and the Studio Vocalist without the cost of buying two microphones — a very good thing.

*Dave Martin has been producing music for more than 20 years and has performed as a bassist for even longer. He has recorded bands too numerous to mention and thousands of “sound-alike” tracks and other production music for the karaoke and television industries. He is the owner of Java Live, a recording studio 20 minutes from Nashville. He can be reached via his website, [www.javajivestudio.com](http://www.javajivestudio.com).*

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by Stephen Murphy

# Don't Stand So Close to Me

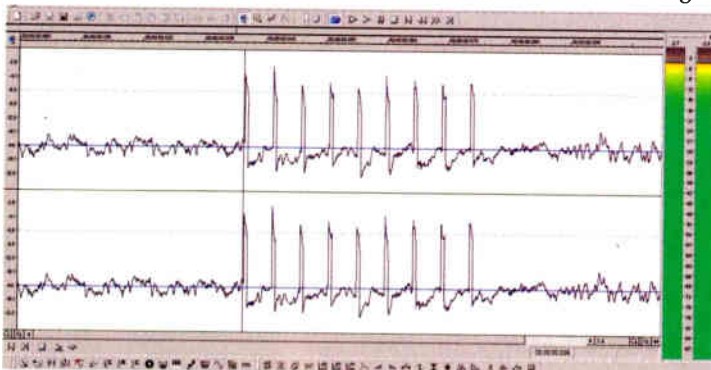


Hey you! Yeah YOU! Very carefully reach into your pocket and take out your weapon. Hold it over your head and take ten steps backwards...DO IT NOW!"

A snippet from the Fox TV show Cops? CSI Miami, perhaps? What about "24"? Getting warmer...

It does take place in Washington DC, but the weapon in question is not a nuclear device or a TASER or even a good old candlestick. It's a BlackBerry wireless device. The victims? My broadcast audio mix at the National Press Club, and every media outlet downstream.

"Sir! Step away from the console! This is your last warning!"



TDMA burst in graphic detail.

## WEAPONS OF MIX DESTRUCTION

Of course I am referring to that invasive, pulsing burst of RF interference that is repeatedly forced upon our innocent audio equipment by BlackBerry-bearing blackguards. Unfortunately, this tragedy is played out countless times each day with ever-increasing prevalence, affecting nearly every segment of the pro audio industry. Could you be its next victim? [Insert dramatic music here]

To be fair, we can't lay all the blame on BlackBerry-maker Research in Motion's doorstep (though perhaps they could do a bit more research and a little less motion). The true perpetrator of these crimes against audio quality is GSM (Global System for Mobile Communication née

Groupe Spécial Mobile) wireless communication technology in general and, more specifically, the electromagnetic interference caused by the time-division multiple-access (TDMA) multi-user channel access scheme they employ. Any GSM/TDMA wireless device has the potential to turn into an audio terrorist without warning or hesitation (wait – lemme do an imitation: "Terrorists hate our freedom, and our quality of audio.").

## TIME PASSAGES

To understand how we got in this quagmire, and if there is any hope for a way forward, we need to understand more about this region of technology.

A multiplex is essentially a multiple-input, single-output switch. Multiplexing is the umbrella term for the process of taking two or more discrete sources and combining them into a single, more complex stream. Mux is the shorthand term for a multi-

plex device (denoted by a trapezoid on schematics and logical flow charts), and "muxing" is shorthand for the process of multiplexing. These shorthand terms should be familiar to anyone who has used a single ADAT to record four tracks of 96-kHz audio (using S/Mux or sample multiplexing, in this case). Demultiplexing or demuxing is the reverse of the process, where the complex single stream is spread back out into multiple discrete streams.

The principal reason for multiplexing is to allow several streams to share a single, limited resource such as a transmission line, a broadcast frequency, an analog-to-digital converter chip or a digital signal processor (DSP). Multiplexing is usually applied for economic, limited-

bandwidth, design-streamlining, combined-processing and/or end-user convenience purposes. Multiplexing is used in real-world applications such as when two designated highways come together and share the same road for a certain

distance and eventually split back into two again.

There are many methods employed in multiplexing to divvy up a single transmission resource among multiple incoming data streams, including frequency division (such as in DSL where different frequencies are used on the same transmission line for upstream and downstream data) and its sibling wavelength division (used to send many streams down a single optical cable using different wavelengths of laser light).

TDMA is a form of time-division multiplexing, more widely known as TDM (OK kids, it's Quiz Time! Can you name an audio technology that uses TDM? I think you can.). Time-division multiplexing is the straightforward method of sharing a single resource among many users by allocating sequential time slots. So it is the time-divided mux/demuxing required of TDMA that causes the relatively high-wattage bursts associated with a single device (using its allocated time slot) sharing a limited resource (transmission cell) with other users.

## THREAT LEVEL: HYPERTENSION-RED

If you haven't noticed the mosquito-like annoyance caused by the proximity of a GSM/TDMA wireless device to your audio system components, you are either very lucky, not paying attention, or very dead. Even if you haven't had any of your own audio work interfered with in this manner, all you have to do is turn on your TV to a local or national news show and wait. Trust me, the time will come. I've heard it many times while watching CNN, CSPAN, and other live programming.

A recent example of how pervasive the problem is happened while I was watching a congressman address the House on CSPAN. A conspicuously loud TDMA interference burst came over the air. The person right behind the congressman reached into his pocket, pulled out and checked his GSM phone, and put it back

STUDIO SENSE continues on page 55 ►



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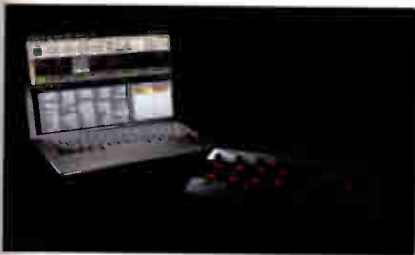
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by **Tony Ware**

## Up, Update, and Away!

The Natives are restless. Native Instruments, that is. They must be, the way the German company (connectable at [www.native-instruments.com](http://www.native-instruments.com)) is kickin' out the updates. First there is KORE 2, Native Instruments' multi-functional host system. This unified architecture offers powerful realtime parameter controls and accessibility functionality, and now adds a comprehensive sound library. This 500-item arsenal is culled from the engines of ABSYNTH, FM8, GUITAR RIG, KONTAKT, MASSIVE



Native Instruments KORE 2 plus controller

and REAKTOR. A new Sound Matrix channel grid allows easy access to Sound Variations and the routing of instruments, effects, KoreSound and MIDI files using drag-and-drop. Additionally there are step sequencer and arpeggiator modules, 30 effects algorithms, a streamlined hardware controller and more, all for \$499.

Next, NI has released Universal Binary support for its PRO-53 virtual analog synthesizer. Version 3.0.3 is either free or \$29 for users, depending on registration date. VOKATOR 1.2 — a vocoding synthesizer — has also received Universal Binary support. The license is \$29. Finally, NI-SPEKTRAL DELAY 1.6, the FFT-based effect processor, has that olde UB support as well, and its license is that same \$29.

### A WINDOW INTO SOFTWARE SOUL

Sweden's Propellerhead Software ([www.propellerheads.se](http://www.propellerheads.se)) has a compatibility announcement ... for Windows Vista (bet you thought I was gonna say "Universal Binary" again). That's right, all Propellerhead Software (Reason 3.0.4, ReCycle 2.1, Reload 1.0.1, ReWire and Rex) is Vista-compatible once updated and reauthorized.

### LOTS O' NEWS TO JAM IN, OR IS THAT JAMMIN'?

It's a common enough fantasy: What if Jimi Hendrix jammed with Bob Moog backed by, say, a philharmonic orchestra. Well, now your imagination is the only limit for this virtual virtuosity, as IK Multimedia ([www.ikmultimedia.com](http://www.ikmultimedia.com)) has a wealth of new software embodying all these signatures and more.

First, "Plug into the Experience" with AmpliTube Jimi Hendrix. In cooperation with Authentic Hendrix, IK has built upon its AmpliTube 2 Dynamic Saturation Modeling to create the first guitar amp and effects modeling software focusing on the authentic gear/tones of Hendrix. Included in this \$249/\$199 crossgrade package are



IK Multimedia AmpliTube Jimi Hendrix

nine vintage stomp effects (Vox Wah, Uni-Vibe, etc.), four vintage amp models and seven cabinets, plus five mics, that together represent Hendrix's vintage rig. Throw in four rack effects modules and a high-precision tuner in the chain and you have a searing, tone-saturated VST, RTAS and AU plug-in, as well as a standalone application for Mac OS X and Windows.

Next, IK announces Sonik Instruments, a range of synth-sampler engines powered by SampleTank 2.5 (recently enhanced with effects sections, a free user upgrade), featuring customized graphic interfaces, multiple timbral parts and insert/send effects per part (built upon T-RackS, now in Universal Beta 1.3 version, and AmpliTube algorithms). SampleMoog (\$249) offers 22 rare, collectible Moog synths from across the range, while SampleTron (\$249) features the wooly '60s "orchestral" sounds of the Mellotron, Optigan, Chamberlin, etc. On the opposite end of the spectrum, Studiophonik (\$349) — VST/AU/RTAS

and stand-alone availability — recreates over 30 realistic instruments.

But what about the orchestra, you say? That's where the Miroslav Philharmonik Classik Edition steps in, with 250 spatial orchestral sounds spanning instruments and ensembles. This \$249 VST/AU/RTAS plug-in (also a standalone application) offers 1.5 GB of samples, plus reverb, sample stretching and more.

### HARMONIOUS SOFTWARE/HARDWARE

Raise your voices in harmony and cheer ... actually, you only need your one voice, because TC-Helicon's Harmony4 (\$999) can do the rest of the work through your Pro Tools | HD console. Create one- to four-part harmonies (with individual gender, vibrato and more humanization features) and other memorable vocal special effects with Universal Binary and Windows XP versions. Harmony4 operates seamlessly in Digidesign's VENUE. Check it out at [www.tc-helicon.com](http://www.tc-helicon.com).

If you're pro-Pro Tools, we've got another one for you: Metric Halo's ChannelStrip TDM and RTAS are shipping in 2.2 Universal Binary, offering large format console-style channel processing, a 48-bit EQ to Pro Tools and "rock solid" support for all newer Digidesign technologies, making it



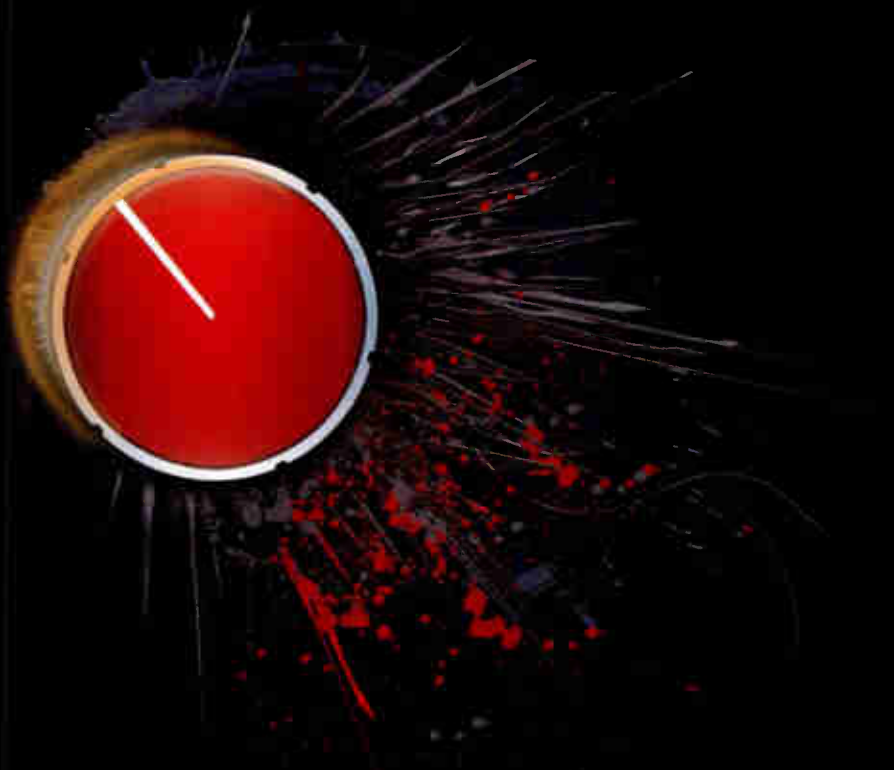
TC-Helicon Harmony4

an "accelebration" offering TDM licenses at \$499 and RTAS licenses at \$249. See more at [www.mhlabs.com](http://www.mhlabs.com).

### DRINK IT IN

Wanting to help make the most fluid mix possible, Focusrite just announced Liquid Mix version 2.0, delivering 32 channels of EQ and compression simultaneously. Offering an EQ side-chain on the compressor, as well as GUI functionality, stability and buffer optimization, the update is free to registered users (check [www.focusrite.com](http://www.focusrite.com)).





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VHD is already warming up audio in the Alpha Channel, the E Signature Channel and Duality, our latest large format analogue music console, so the chances are, you've already heard it.



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Putting Sound in the Picture

by Rich Rarey

# HHB FlashMic DRM85-C

HHB offers a FlashMic especially for cardioid-loving audio professionals.

The HHB DRM85 FlashMic crossed my home office desk some months ago, and I wrote then that the full-featured flash recorder in a microphone body was impressive, great for those professionals and serious amateurs who need to make quick monaural recordings. That is still true. But while the European market has clutched the DRM85 to its professional bosom, according to reports, the American market has chafed with the omnidirectional pickup pattern. The market chafed, that is, until the release of the DRM85-C Flash Mic with cardioid pickup pattern. This should satisfy the American need for tighter sonic pickup, while keeping the fine features of the original DRM85.

## FEATURES

The DRM85-C, like its predecessor, has 1 GB of flash memory, a USB port, headphone jack and is powered by two AA batteries. It's still a very light 13 ounces.

The Cardioid pattern drops 3 dB around 60 degrees off-axis, and 6 dB at 90 degrees, while the frequency response curve shows rising voltage levels above 1.1 kHz topping to a 5 dB relative peak around 5 kHz, descending 3 dB at 7.5 kHz and descending 25 dB to 20 kHz. A 12 dB/octave rolloff at 100 Hz can be engaged in boomy environments. Headphones are plugged into the standard 1/8-inch stereo mini jack on the microphone's base, and allow source monitoring and listening to track playback.

The DRM85-C controls are very simple: three buttons and a spring loaded jogswitch. The three buttons control "transport" functions of "Record," "Play" and "Stop/Menu." The jogswitch acts as a soft-selector for dialing

amongst menu items, and for adjusting the record and headphone levels. The status of the FlashMic is easily seen in a small backlit LCD display on the side of the body.

It is very simple. The FlashMic, in fact, has a pre-record buffer whereby the 10 seconds of audio prior to pressing the Record button are recorded to the audio file.

There are two audio formats: Linear 16-bit PCM (.wav) or ISO MPEG1 Layer2 ("MP2"), at a sampling frequency of 48 kHz, 44.1 kHz or 32 kHz. Both are popular audio formats, and the data-compressed MP2 format is robust and tolerant of multiple transcodings.

Recording times using the 1 GB internal memory range from three hours (.wav, 48 kHz sampling) to over 18 hours (.mp2, 32 kHz sampling, bitrate 128 kbps). There's no problem recording different formats on the memory card, so one can choose better quality for critical audio and later a lower quality format for more recording time, say for a stakeout.

Getting the audio out of the FlashMic is simple, too: plug the provided USB cable into the base of the FlashMic, and then the other end into your Mac or PC, and the computer recognizes the FlashMic as a USB Storage device and presents the file tree as just another removable drive. Simply click 'n' drag files from the "audio" directory and onto your computer's drive — it'll take about 10 minutes to transfer an hour's audio. The latest firmware revision even enables the FlashMic to operate from the computer's USB port power, so

even if the mic's AA batteries are removed, you can still access the audio.

The complicated part — audio encoding selections, file naming conventions, and the like — can be revealed or concealed to the user depending on how the nine user-defined presets are configured. The FlashMic is a CPU-like device, so HHB has included a CD-ROM for your PC or Mac containing a simple preset-configurator — the "FlashMic Manager" application — for creating and uploading the nine presets into the FlashMic. Each preset specifies over a dozen recording and audio properties for the mic. File format, sample rate, number of seconds to hold in the pre-record buffer (or none), manual record level or Automatic Gain Control (AGC), High-Pass filter, the LCD backlight

## FAST FACTS

### APPLICATIONS

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### KEY FEATURES

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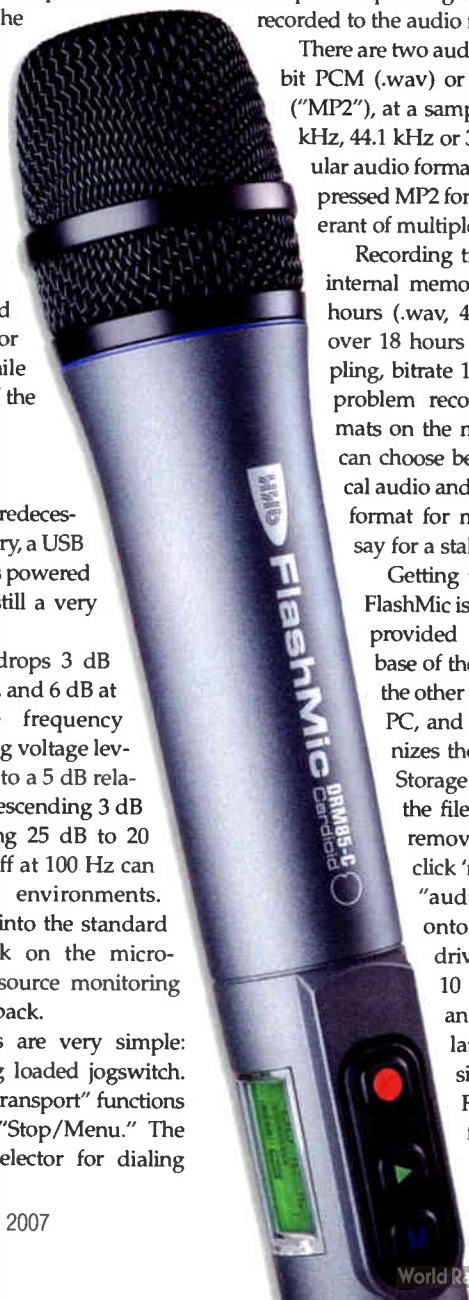
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options, and even a setting for properly displaying alkaline and rechargeable battery life.

Just as important, file naming conventions can be specified, plus the text of the "Reporter Name," "Company Name" and "Description" fields is embedded in the data chunks of the sound file. The configuration is stored in the FlashMic as a plain-text file in its own directory, but HHB cautions against editing it directly.

## IN USE

I took the DRM85-C FlashMic out of the box and stared blankly at it — it'd been months since I'd worked with the DRM85 — and then I realized, "Don't think; record." I pressed the jogswitch to power up the device and, after the short bootup process completed, pressed the Record button. The FlashMic obediently began



recording. Whew, no surprises. The LCD display counting down the remaining recording time. A small red LED in the microphone base also lights up when in record; the LED can be toggled on and off by pressing and holding the Play button while in Record — useful when you don't want others to see the red recording light.

The default setting uses AGC for level control. Pressing and Holding the Record button while in record "locks out" the transport and power buttons to prevent accidental stops. Pressing and holding Record unlocks the transport and power buttons. A brief press of the Record button places a marker — a "data flag" — in the file for easy locating later. The markers were visible at the waveform's top when I opened the file in Adobe Audition. After recording, press Stop, and the file is quickly closed. I listened to the file in headphones by pressing Play and shuttled around the file by turning the jogswitch "forward" or "backward." The headphone level was too loud, so I pressed and held the jogswitch until the display showed "Phon," and turned the level down by moving the jogswitch backward.

The DRM85-C's firmware is easily upgradeable; HHB provides a downloadable file and

instructions that involve nothing more than dragging a 268 kilobyte binary file onto the FlashMic's root directory and powercycling the mic. HHB is also provides a downloadable CD image for the FlashMic Manager software, making that installation even easier.

I checked the sonic qualities of the DRM85-C by first evaluating its cardioid pattern recording in a high-ceilinged living room, resting the mic on a candle holder — not on a stand and not even using the provided mic clip — and about 12 inches from my talkative subject. My subject, as with most people, quickly forgot about the mic, and I recorded over an hour in the room. I downloaded that large, linear file to my desktop computer and was pleased with the AGC's gentle touch, and impressed with the mic's sonic quality at one foot. Others talkers across the table from my subject were understandable, but well off-mic. Proximity is important, as with any standard cardioid mic; the user must remember the DRM85-C is not a "shotgun mic" or have parabolic characteristics.

A close friend who hosts and produces Podcasts listened to my playback, and complimented the DRM85-C in a salty fashion:

"Wow!" she said. "This mic recording

sounds really good...I gotta get me one of these; can you lend it to me for awhile?" I did, and she started voice-tracking her Podcast script on the spot.

"Wait," I begged. "Let's listen to the playback to get the right mic position for you."

We did, and found the DRM85-C is not pleasing against the lips; the unprotected capsule exhibited plosives and dull muffled sibilances — exactly opposite of what we'd heard in our initial recording. We backed the mic off to the "international broadcast standard" of 3 - 5 inches and the quality bloom returned. I mention this because the DRM85-C's light weight and lack of cabling will cause inattentive users to inadvertently swallow the mic, producing similarly stifled results.

#### | SUMMARY

My producer friend was thrilled, and left happy, vowing to do all of her Podcast voice-tracks on the DRM85-C — if I'd only let her. The FlashMic proves once again to be an easy to use, flexible, well-performing tool.

*Rich Rarey is master control supervisor for National Public Radio.*

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by **Mel Lambert**

# Producing Surround Sound at 49th Grammy Awards

Moving towards an all-digital broadcasting future.

The broadcast industry is bracing itself for the final furlong in the race towards all-digital transmission, including 5.1-channel sound. And there can be few opportunities more glorious for staging advanced production than The Recording Academy's annual Grammy Awards, broadcast this year in mid-February from its regular home at the Staples Center, Los Angeles. The show, co-produced for The Academy by Cossette Productions in association with Ken Ehrlich Productions, has encompassed HDTV and full 5.1-channel surround sound for several years.

"The 49th Annual Grammy Awards marks the fifth year that the telecast has been broadcast in HDTV/5.1 surround sound," stresses Neil Portnow, president of the recording academy. "Each year the bar is raised on the overall production, particularly on the audio quality. We are moving to total HDTV/5.1 broadcast in 2009, so this year's show once again extended the boundaries of broadcast television."

"This year the network wanted us to pay particular attention to the surround sound," offers Hank Neuberger of Third Wave Productions, who serves alongside Phil Ramone, Chairman Emeritus of the Recording Academy's Producers & Engineers Wing, as co-supervisor of broadcast audio for the Academy.

"CBS estimated that 35-percent of the viewing audience would be viewing high-def signals and listening to the digital audio playback," continues Neuberger. "So we decided to generate several completely separate mixes: one for the analog audience, in stereo, and also upmixed to Dolby Pro Logic II to provide surround playback for that audience; and a discrete 5.1-channel mix that would be carried with the HD signal [as Dolby Digital/AC-3-format audio data] but also downmixed [to stereo within TV receivers

and set-top boxes] for the SD audience."

That decision had important practical considerations, Neuberger stresses. "Aside from generating these complementary mixes, we needed to ensure that we could accurately monitor what people were listening to at home. So we needed four monitoring environments: discrete 5.1, a stereo downmix of that 5.1-channel mix, the main stereo broadcast mix for the analog audience, and the upmixed Pro Logic II in surround sound. Next year, or maybe the year after, when the analog audience is on a decline, we might be able to reduce the number of mixes we need to accurately monitor, but for this year that was our SOP."

## **XM/EFFANEL: ON THE JOB**

The awards ceremony's producers again turned to XM/Effanel Productions to facilitate multi-channel music mixing, while a separate mix area within the NEP Super Shooter ND3 video production mobile

handled the main broadcast balance. Live music mixing fell once again to John Harris and Eric Schilling in XM's Digidesign ICON-equipped L7 remote truck, while Tom Holmes manned the 86-fader/dual-layer Calrec Alpha Digital Broadcast Console in ND3. Sound reinforcement for the Staples Center audience was handled by ATK/AudioTek [For more details, see companion sidebar].

A supplemental mixing area was again provided to enable Harris and Schilling to refine their ICON D-Control settings. The Offline Remix Booth/ORB served as a temporary mix room featuring an identical work surface and monitors where music mixers could work on material recorded to Pro Tools | HD during rehearsals and refine static 5.1 surround-mix levels, EQ and dynamic settings. Updated ICON data was transferred via portable media and compiled into a Master Pro Tools Session prior to the air-date.

A major change this year was the provision of an Overflow Mix Area in the rear of the portable space housing ORB, where Joel Singer, XM/Effanel's Engineer-in-Charge, used a 32-input Yamaha DM2000

Digital Production Console to handle orchestral submixes during more complex awards productions, including contributions from Gnarl Barkley's Cee-Lo Green. "I patched into 32 mic lines coming off the stage from the orchestra, which I submixed into eight mono/stereo stems of

first violins, second violins, violas, celli and bass," recalls Singer. "These were passed back to two destinations: the front-of-house mixing console for the audience PA and L7, where John and Eric incorporated them into the main 5.1-channel balance. I could also monitor the final L7 mix to check how my stems were blending in the broadcast mixes."

"The primary feedback about the show



**Tom Holmes in the NEP Super Shooter ND3 video production area**



**From upper right, counter-clockwise: TV production area inside the L7; the XM's Digidesign ICON-equipped L7 remote truck exterior; John Harris and Eric Schilling in the L7's mixing area.**



was that [the sound] was very consistent," says music co-mixer Schilling. "As for Academy feedback, Phil [Ramone, broadcast sound co-supervisor] was very positive. He offered some constructive feedback during rehearsal and dress, which defiantly tightens up our mixes."

Singer also notes that the use of Genelec DSP 8200 Series loudspeakers in

8240 bi-amped active monitors, along with a Model 7260 subwoofer. "They sounded great, but with the added feature of adapting to the environment and fine tuning through DSP," enthuses Singer. "Musically these are the best sounding speakers we've used."

"Without our partnership with Tekserve in NYC the ORB room wouldn't exist," considers Singer. "They loaned us product and manpower." Tekserve supplied a quartet of seven-card Pro Tools | HD Accel Systems for the Mix Cores, each of which featured eight 192 I/Os — two with analog expansion and six with



Joel Singer in the Overflow Mix Area of the L7, with Genelec monitors

L7, ORB and his Overflow

digital cards. Recording systems consisted of a three-card Digidesign HD Accel rigs, with an identical I/O selection.

"Digidesign is the bomb!" says Singer. "We took a product two years ago and pio-

neered its use in live broadcast TV. It was a necessary step in our evolution, but also a big question mark in many people's minds. Now with other broadcast facilities turning to D-Control it again validates our choice and direction. It makes you feel good to say you were the first!"

According to Thomas Graham, Digidesign's West Coast Icon Product Specialist, "Joel Singer and the XM/Effanel team have developed an innovative workflow that essentially increases the amount of pre-mix time and allows the performing artists and their producers more opportunity to collaborate with the music mixers in order to achieve the best sounding mix. Hank Neuberger really summed it up best when he said: 'What we were doing was stretching the space-time continuum and creating rehearsal hours that we never had before.' The XM/Effanel team have been helpful in the development of Pro Tools features for ICON."

**BACKSTAGE ACTION**

Audio coordinator Michael Abbott again had the unenviable job of ensuring that **GRAMMY AWARDS** continues on page 48 ►



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Mix Area was a major advantage. The surround system comprised Model 8250 and

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GRAMMY AWARDS Continued From Page 47

everything backstage went smoothly for this landmark broadcast. "The Recording Academy wanted to make sure that we could produce great-sounding 5.1-channel mixes, and so we placed an emphasis on the ORB and being able to monitor in multiple formats. Quality control was a major emphasis this year with each area being provided with appropriate playback equipment, to ensure that we could listen in a variety of formats. And we had more mic and instrument sources than I can recall from previous years, which made changes between the A and B Stages all the more demanding."



Paul Sandweiss in the 5.1 Mix Room

A dedicated 5.1 Mix Room enabled engineer Paul Sandweiss to concentrate on blending together music mixes from L7

with a separate 5.1-channel audience mix from engineer Klaus Landsberg, plus live stage and pre-recorded "The nominees are ..." announcements and audio/video playback elements. This ensured that the HD listening audience could enjoy a truly enveloping surround mix.

"With so many performances and so many microphones and assorted inputs, this year's show was really about the on-stage engineers making the band setups work; it gets harder each year," concedes Sandweiss. "My job is to sit in a controlled environment and listen to all the hard

work that the other engineers have done! The 5.1 mix really centers around the music performances, which were nicely

handled by John Harris and Eric Schilling; all I do is adjust the other elements to follow along with the performances. Hank [Neuberger] expressed only positive feedback to this year's show both stereo/SD and 5.1/HD feeds.

"The Grammy Awards is the only live show I know of that supports two mixing formats — stereo and 5.1-channel — which I believe gives you the best stereo mix, as well as the best 5.1," continues the engineer. "Most networks only accept a 5.1 mix, and fold down a stereo mix [for analog audiences]. Unfortunately, it's similar to the old mono-from-stereo days, when the best sounding stereo mix did not make for the best sounding mono fold down."

Overall broadcast levels for both the SD Stereo and HD 5.1 broadcast were controlled via an array of TC Electronic processors. Effanel's L7 and NEP Video Production trucks were equipped with TC Electronic DB-8 Digital Television Processors, while ORB was outfitted with a TC System 6000 for additional effects. "The big change this year was the addition of Pro Tools plug-ins for the music mixes," points

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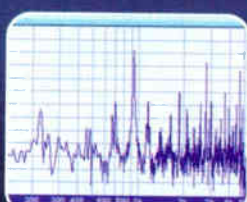
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out TC Electronic Chairman Ed Simeone. "L7 truck used the full suite of System 6000 plug-ins, including VSS-3, DVR-2 and NonLin reverbs." Audience mixer Landsberg used a TC DB-4's 5.1 compressor section for overall level control.

"Normally, DB-4 and DB-8 DTV processors are installed after TV network's master control," says Simeone, "but they have proven to be ideally suited for live television productions like the Grammy Awards. The units guarantee that audio leaving The Staples Center meets network level standards. The DB-8s in the stereo and 5.1 trucks are the final level-control processing before the stereo signal is relayed to CBS and, in the case of the 5.1 telecast, right before the Dolby Digital Encoder."

"We've used TC on-air processing on Grammy broadcasts for three years," says audio coordinator Abbott, "and the sound quality, reliability and ease of use are the reasons the engineers request it every year."

## DELIVERING THE MIXES, GATHERING THE SOUND

Technical staff from Dolby Laboratories was also on hand to supervise preparation of the various multichannel mixes delivered to CBS Television. Dolby DP563 Pro Logic II Encoders automatically created a Pro Logic II soundtrack for analog/SD stations.

"DP570 Multichannel Audio Tool units also enabled the mixers to emulate a wide range of home environments, thereby ensuring that the program sounded its best under any circumstance," says Rocky Graham, Dolby's Director of Broadcast Products. The DP570



Wireless Microphone bins

provides reference monitoring; it accepts up to eight audio channels with two independent sets of outputs. One set carries the main multichannel program signal, while the second — or Emulator Set — enables monitoring in different listening modes without affecting the main audio path.

On-stage microphones included models bearing Audio-Technica, Neumann, Sennheiser and Shure brands. Sting and Andy Summers from The Police used Sennheiser e865 vocal mics, while Stewart Copeland favored an HSP4-EW headset with SK500G2 body pack. Beyoncé used a Sennheiser SKM5200 wireless handheld

with a Neumann KK105-S capsule, while Shakira opted for a Sennheiser SKM3072 wireless handheld mic for her performance with Wyclef Jean.

Gnarls Barkley's stand-out rendition of "Crazy" featured Cee-Lo Green with his custom, gold-plated Sennheiser SKM5200 hand-held with Neumann KK105-S capsule. Philip Bailey used the same combination with Earth, Wind & Fire during a segment with Ludacris and Mary J. Blige. And Rascal Flatts utilized Sennheiser/Neumann wireless mics during their tribute to The Eagles. Most of the show's presenters were on Sennheiser SKM5200 wireless handhelds. The production made use of more than 40 Evolution Series e602II and e902 wired microphones; e935 and KMS105 microphones and 16 channels of EW300IEMG2 were also on hand for performers.

Audio-Technica provided close to 300 mics for the live broadcast, including hard-wired mics and Artist Elite 5000 Series UHF wireless systems. An Artist Elite 5000 with a custom AEW-T4100 handheld mic/transmitter handled vocals for Justin Timberlake, while contest winner Robyn Troup used a 5000 Series with AEW-T5400 transmitter during her "My GRAMMY Moment" duet with Timberlake. And 5000 Series/AEW-T5400 combinations were featured on vocals by Ludacris, and for Smokey Robinson's segment. 5000 Series wireless systems with AEW-T1000 UniPak and Artist Series ATM350 were used to cover for violins, violas and cellos.

Multiple channels of Aphex 1788A Remote Controlled Mic Preamps in the L7 remote truck were used for the recording and broadcast feeds. "First and foremost, they sound amazing," says XM/Effanel's Singer. "We tested other remote pre-amps [for use in L7] but none allowed for input-gain adjustment on the fly without popping and clicking." Another 32 channels for orchestral string overflows were rented from Audio Specialties, Burbank.

## MEANS OF MONITORING

Studio monitoring, in addition to systems from Genelec, included JBL LSR4300 Series cabinets in Paul Sandweiss' 5.1 Mix Room and five LSR6328P monitors and an

**GRAMMY AWARDS** continues on page 50 ►

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LSR6312SP subwoofer in the NARAS TV Committee room, plus a quintet of LSR4328P monitors in Klaus Landsberg's Audience Reaction mix room.

XM/Effanel's Singer summarized his experience at this year's event by saying he received some of the best response he



Klaus Landsberg in the Audience Reaction mix room

can remember. "The stand-out comment from people working the show was that, from the truck to air, it was the most consistent sounding show anyone could remember. Phil [Ramone] and Hank [Neuberger] thought it sounded very coherent from 5.1 to stereo. And that's what we strive for."

Singer also plans to expand the L7's recording system. "Right now we have 96-channel Pro Tools recorders — that's the max we can run with HD3, since it takes up all the card slots in the [Apple] G5. Since I want to use a Global SAN-type system for recording and movement of data, I would need to reclaim a PCI slot in the main computer. Hence the need to move to an expansion chassis [configuration]."

"The 49th Annual Grammy Awards was the most complex show to date and featured expanded staging employing literally hundreds of microphone inputs, a new level of multi-channel wireless complexity and elaborate set changes," concludes Recording Academy president Portnow. "Our production team, which included leading members of the P&E Wing, is a tour de force in the entertainment business."

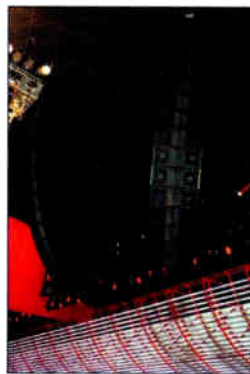
*Mel Lambert has been intimately involved with production and broadcast industries on both sides of the Atlantic for more years than he cares to remember. Now principal of Media&Marketing, a Los Angeles-based consulting service for the professional audio industry, he can be reached at mel.lambert@MEDIAandMARKETING.com.*

## | QSC-POWERED JBL VERTEC ARRAY PROVIDES SOUND REINFORCEMENT FOR GRAMMY LIVE AUDIENCE

Sound reinforcement design for the live audience in The Staples Arena posed its own unique problems. Valencia CA-based ATK AudioTek Corp again provided a flown JBL VerTec system powered by QSC amps via XTA Electronics DP226 processors. "There were four main clusters, each of which comprised 12 VT4889 line-array elements," recalls Jeff Peterson, ATK System Design Engineer. "Each cluster was also rigged with our custom digital angle sensor and laser aiming device, which indicates the aim point of the top of the array coverage. A new flown subwoofer array consisted of three individual clusters of four JBL VT4880a subwoofers. These separate clusters were time aligned to create a directional forward-steered array," and hence minimize low-end spill onto the stage.

QSC power amps comprised 30 PowerLight 9.0 models for LF cabinets and subs, plus QSC PowerLight 4.0 units for HF/MF cabinets. Delay loudspeakers included 33 VerTec 4889 Line Array Elements arrayed in three clusters, plus 12 ATK C-6 three-way cabinets.

"The single center-hung subwoofer cluster performed as we had hoped," offers Peterson. "Coverage on the floor was very smooth, and a considerable amount of LF energy still reached the farthest seats in the house. We used eight of our ATK CSW-218 subs on the floor to cover the closest seating areas."



All components were connected via a QSC RAVE 160/161 CobraNet-based network. "RAVE eliminated running 600 feet of multipair," says the design engineer, "and gave us networked amplifier control via a QSC CM16a in each amp rack. Other new devices we used this year were a pair of custom, four-way line-level 56-pair split systems, in addition to five three-way/56-pair splits. The line-level splits — designed by ATK's VP of Engineering Scott Harmala — let us distribute all of the high-level signals

on one 56-pair cable to each console, instead of running numerous 16-pair cables."

ATK again provided a pair of Yamaha PM1-D Digital Production Consoles for front-of-house, plus one each for Stage A and Stage B monitor mixes. Ron Reaves handled music mixes from the right-hand PM1-D FOH console — incorporating Joel Singer's orchestral mix stems — and deliv-



ered an L-R music submix, subwoofer/LFE mix and a vocals stem to ATK Vice President Mikael Stewart, who was manning the main PM1-D console. Here, Stewart added stage announcements, pre-recorded track and other elements.

"I used the PM1-D's snapshot function to set up mixes for each artist, and then recalled them between sets," says Reaves. "All signal processing was handled inside the PM1-D, including special effects and dynamics." The Recording Academy's Leslie Ann Jones also worked with Reaves to fine tune front-of-house audio.

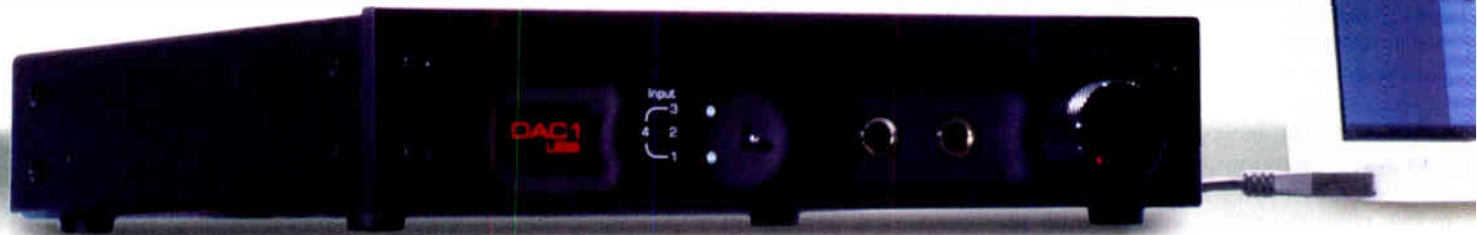
According to monitor mixer Michael Parker, who covered Stage B, "Three acts rehearsed off-site and brought in their PM1-D cards [containing I/O assignments and mix settings]. I coordinated with each monitor engineer so they could have a 'Grammy PM1-D Template.' They set up their personal mixes and incorporated my production I/Os. With a lot of emails and phone calls it was very successful.

"With a very polluted wireless spectrum at the Staples arena, hard-line IEMs [in-ear monitors] came into play in a big way!" Parker continues. "Bands, typically, had 10 to 14 stereo IEM mixes broken down into musicians that can be on hard-line Ears and those who absolutely needed to be on RF. I also had floor monitors under the downstage grate that handled any information for [bands] that wanted an IEM and floor-monitor combination."

— Mel Lambert

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by Strother Bullins

# A Studio to Go



Photo credit: Doug Rice/Alura

immy Daniel — a producer, engineer and all-around nice Southern gentleman — will be the first to tell you that he is, first and foremost, in the “content” business. He works primarily in the audio-for-broadcast industry, recording live performances and on-location sessions: audio content for television, radio and

internet/new media broadcasts. Music is his passion; providing audio content is his job. And Daniel, like many other young professionals in the pro audio industry, takes his passion and his job very seriously.

The results of Daniel’s seriousness is apparent in his latest business venture: JDM Mobile — a remote broadcast recording company featuring “Beluga,” a 40-foot, lead-lined mobile recording truck equipped with the most dependable and best-sounding recording gear he could find. Daniel bounces between offices in three Southern entertainment hubs — Atlanta, Nashville and Orlando — but will go almost anywhere in his posh white Freightliner to provide his clients with the highest quality audio possible.

“Live entertainment is an extremely large focus when it comes to content,” explains Daniel. “We have equipment just like any professional recording studio would, but we’re really honed in to capturing and posting audio for broadcast, as well as broadcasting live. We’re dealing with producers putting together content, and we’re producers ourselves

whether it’s for a televised award show or an AOL live performance. The differences between these clients can be tremendous; one client may want one setup because they’re broadcasting in one particular format. If you’re doing live audio-for-broadcast for a major network, it’s obviously going to be a different feel than audio capture for post or even a small live web portal.”

A recurring difference in the needs of JDM Mobile clients may be based on individual bandwidth requirements of their chosen broadcast medium (or the limitations thereof). “In some ways, it’s kind of silly, but some of the new media broadcast stuff can be captured at 24 bit/96 kHz, and then it’s dumbed down to fit the Internet,” offers Daniel. “However, we’re going out there to capture pristine audio, and there’s a lot of people who need to make things sound better in 8-bit. For the new media clients, that’s a huge dilemma and something we have to face. We capture things the best we can and make it sound great in the truck or in our post facility to provide our customer with the best audio that we possibly can.”

## THE TRUCK

Daniel has a history of equipping and building mobile recording trucks for others, but Beluga was the first truck he put together for his own use. He first called upon his friends at Frontline Communications — a company that builds no-compromise broadcast industry

trucks — for a heavy-duty chassis. “It was an amazing venture to jump into something with no compromises,” he recalls. “From the chassis to the box, to the frame, to the power, it was an extensive venture way

before the gear was considered, and we did it all to the nines. Building a mobile truck is just like building a great studio: it has to be built on a great foundation.”

Daniel, who professes to be an analog guy at heart, knew that the truck must be designed with an all-digital signal flow. “I love analog, copper and two-inch tape machines,” he insists. “But this truck was to be for broadcast and we needed fiber-optics for longer runs. We also wanted it to be very comfortable with enormous amounts of redundancy.”

## EQUIPPING BELUGA

Daniel spent about two full years on gear selection, trying and considering any and all pro-level equipment options. “I started out thinking of having Millennia Media preamps for onstage — doing some of the higher-end stuff,” he offers on beginning the process in microphone preamp mode. He eventually selected an all-Aphex 1788A Microphone Preamplifier System provided signal via a 96-input Radial Engineering Custom Shop isolated snake system.

“I put those up against other mic preamps and the Apogee converters I was currently using,” recalls Daniel. “In the end, the Aphex 1788A was where we hung our hats. We have been absolutely thrilled with their versatility and sonics. I’ve had clients ask, ‘Do those things sound great?’ and — after putting high-end preamps up against them — they’ve been chosen every single time. The 1788A is a tremendous mic preamp, as well as a tremendous converter. In choosing our own isolation splits, I went solely on my ears. I listened to a bunch of stuff and locked in on the Radial stuff with Jensen transformers. We have three total splits, which gives us a lot of flexibility.”

All signal after the Aphex converters travels MADI to Beluga courtesy of a RME ADI-648 64-channel multichannel digital audio interface. MADI hits another RME ADI-648 on the truck, then straight into a dual Yamaha DM2000 V2 console system linked to two Pro Tools |HD3 192-kHz DAWs running in redundancy. “I tested five different consoles,” explains Daniel on his choosing the DM2000, which allows him to go up to 128 inputs when required. “Because of the demands of the industry, there’s a lot of different input and output patching configurations that need to happen. Having two DM2000s in

Jimmy Daniel in front of the JDM Mobile Beluga recording truck



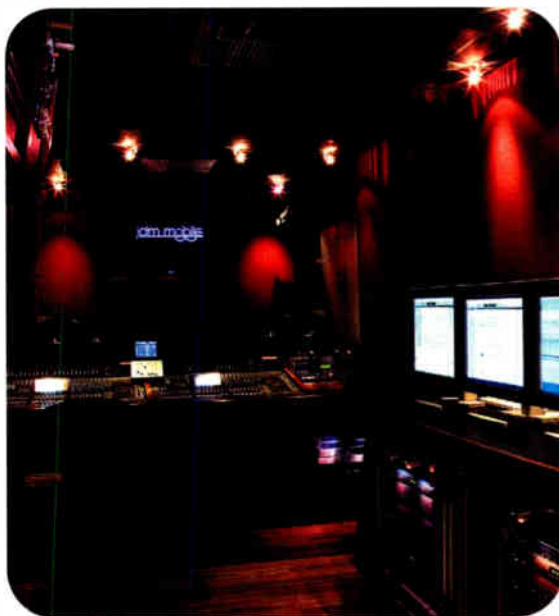
the truck gives us a very large spread for both 5.1 and stereo. And in my world where you have to worry about the 'one-pass catch,' this system is the most solid setup. Our audio quality — from the Aphex front-end to the two Yamaha consoles — sounds great and works incredibly well."

Daniel points out more crucial gear, and there's a lot of it (just check out his website). But most notable is his TC Electronics System 6000 and a surround configuration of five ADAM S3A powered midfield monitors. "We use the mastering section of the TC Electronic System 6000," he explains. "It's been a great piece for 5.1 broadcast. It's very flexible, and even when we're not in 5.1 we can use it for four banks of super effects processing. I originally heard the S3A monitors when Jazz at Lincoln Center's studio opened, and I was blown away by their flexibility. Sure, I may have a monitor preference that works just for me, but I didn't build this truck for me. I built it for all of our potential clientele. The ADAMs are great for the space, too; we didn't do any room monitors, so we needed something with a

bit of beef behind them — those do."

Daniel, asked what might be next on the Beluga's equipment list, can't hide his enthusiasm for the upcoming TASCAM X-48, a to-be-

released standalone 48-track DAW capable of 24 bit/96 kHz, 24 bit/192 kHz for 24-tracks. "We can't wait until they're available," he says. "They will be used primarily on stage as a safety. We really believe that it will reduce our worries as far as capture is concerned."



Inside the Beluga mobile recording studio

#### CAPTURING CRUCIAL CONTENT

It is in having a constant concern for never missing a beat where the equipment on Beluga primarily deviates from that of a traditional recording studio. Daniel needs great sounding audio as well as bulletproof, first-pass recording capabilities — and not necessarily in that order. "We're not in a bricks-and-mortar room, where we can get a second and third chance," he concludes. "We must have the best convergence of hardware and software that we can. It's absolutely crucial in audio-for-broadcast."

You can see what JDM Mobile is all about at their website, [www.jdmmobile.com](http://www.jdmmobile.com).

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

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by Ty Ford

# Lectrosonic SMQ Wireless Transmitter

The small, solid SMQ packs a punch

Wireless mic makers have been long been working on creating smaller, less obtrusive transmitters. The Lectrosonics SMQ transmitter is proof — although it is still slightly bigger than their SM model. Lectrosonics also has another challenge, however: the increasingly shrinking RF spectrum.

## FEATURES

The Lectrosonics SMQ (\$1,960) includes the transmitter, four AA batteries, a charger and a leather pouch/belt clip. The 2 3/8-inch x 2 3/8-inch body of the SMQ is machined out of solid aluminum billet and weighs just over four ounces with two AA batteries installed; it's heavy enough to sag the back of a lightweight bikini unless very snugly tied, but otherwise very serviceable. Two AA batteries power the SMQ for four to five hours. Some wireless systems come with a mic, even if you already have one or don't want theirs. Lectrosonics has mic option pricing: you can buy no mic or choose from several

they offer. There's a slight savings if you order the mic with the transmitter.

Lectrosonics has responded fairly quickly to the shrinking spectrum problem with frequency agile systems; the company's more recent receivers have built-in scanners to help find clear frequencies. The scanner in the UCR411a receiver supplied for this review was easy to use and very helpful.

The switches and lights on the surface of the SMQ are mounted under membranes with no exposed access to the guts of the body excepting the antenna and mic connections, and the lower screw plate holding the two AA batteries in place. This makes it water resistant, but not waterproof. The first Voice Technologies lav came with a rubber bootie at the connector to further ensure that moisture was kept from entering the mic connector.

The SMQ transmitter is compatible with earlier Lectrosonics 100, 200, 400 and IFB receivers. The SMQ uses patented digital hybrid technology that creates a +/- 75 kHz-wide digital audio signal transmitted over an analog carrier to the receiver. The transmitter's servo input incorporates a limiter with a dual-release envelope that protects from inputs as high as 30 dB above full modulation.

Audio is digitized by the SMQ's dual 88.2-kHz, 24-bit A/D converter, then pilot tone and limiter control data are added to the stream. Each transmitter has 256 available frequencies in 100-kHz steps over a total bandwidth of 25.5 MHz.

## IN USE

Recording the audio, I walk-tested how much

muscle 250 mW provides by pairing the SMQ with the Lectrosonics UCR411a receiver and my Sound Devices 442 mixer, and leaving the SMQ transmitter at my starting location. I found WMPB DTV's 14 kW 996 feet HAAT signal seven miles away with the scanner in the receiver. I tuned the SQM to the same frequency and set off for another walk around. I began picking up hits at about 50 yards while shading the receiver from the transmitter with my body. Getting my body out from between the transmitter and receiver gave me a few more yards, but the hits just kept on coming.

I continued to walk around the outdoor location, getting farther and farther away, with depreciating reception. The location I picked for testing posed some typical challenges users of wireless systems face: medium and large fixed objects and, in this case, several small buildings. I returned to my starting point and used the UCR411a's scanner to find a more friendly frequency, then retuned the transmitter. This time I got to 136 yards before the receiver started swishing. I got no direct hits, but at 136 yards the swishes were bad enough to make the audio unusable. Outside, even on a DTV frequency with the transmitter seven miles away, I was able to get 50 yards outside with just the rubber duck antennae on the receiver.

I did a battery swap-out and walk-tested the receiver in a 3/4 below-grade space with the SMQ transmitter hanging 20 feet away, or about 30 - 60 feet from the receiver depending on where I was in the basement. The system never took a hit.

I tried the three "Noise Reduction" modes. I settled on Lectrosonics' middle default of some, but not full. The hiss was more audible without Noise Reduction. The amount of hiss is tied to what mics are used; lavs with smaller diaphragms and higher self-noise figures obviously generated more hiss.

Changing input sensitivity and frequency on the SQM is very easy. The transmitter display is generally easy to read except that the digits "4" and "9" look very much alike because the



## FAST FACTS

### APPLICATIONS

Location sound, broadcast

### KEY FEATURES

Patented digital hybrid technology; dual 88.2-kHz, 24-bit A/D converter; small size, low weight, solid aluminum construction; membrane-protected switches and lights; built-in scanner; fixed HPF

### PRICE

\$1,960

### CONTACT

Lectrosonics, Inc. |  
☎ 508-892-4501  
➤ www.lectrosonics.com

top of the "9" so close to the upper edge of the window that it's difficult to make out, unless you're in the right kind of light.

Lectrosonics also sells the very handy RM remoter control for its SM Series transmit-

## PRODUCTPOINTS



- Small
- Powerful
- Versatile
- Efficient



- Premium price
- Transmits only at 250 mW

## SCORE

Solid wireless transmitter technology from a solid company

ters. It "plays" an audible tone from a small speaker, receiving it through the lav attached to each transmitter. Input sensitivity, sleep and lock modes can be changed — as can operating frequency by hex, block or MHz — at the touch of a button on the RM. These are very cool features that mean you don't have to go wardrobe, diving to make changes after the transmitter is properly positioned.



### | SUMMARY

The SQM is a very capable, well thought-out transmitter. My only suggestion would be to add a software switch to allow the unit to run at 100 mW and/or 50 mW when needed. That would provide more hours of battery life, which in some cases would be advantageous. Lectrosonics, however, does include not only four rechargeable 2200 mAh, nickel metal hydride AA batteries with each purchase, but also an Energizer 15-minute, four-AA battery charger. That's a nice thing to do.

*Ty Ford has been writing for Pro Audio Review since the first issue. He may be reached at [www.tyford.com](http://www.tyford.com).*

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## STUDIO SENSE

### | Column

STUDIO SENSE Continued From Page 38

into his pocket. Thirty seconds later – bzzt bzzt bzzt – once again he pulls out his phone.

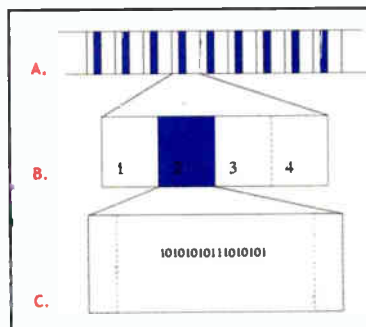
I suspect the problem is especially bad here in D.C. where everybody is "important" and therefore tethered to their power device(s). I routinely have to ask people to step away from the console, wireless receivers, etc. It can be quite touchy when you are asking a former U.S. president if he has a BlackBerry, and would he mind turning

it off before getting up to speak – no not to silent or buzz – all the way off.

The thing I'm worried about is that I've seen signs of acceptance and resignation to the intermittent problem, similar to the public acceptance to lower-quality music files.

In a future column, I'll report on any glimmers of hope in adopting a viable "way forward" on this battlefield. In the meantime feel free to send me any TDMA war stories to share with the troops.

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



TDMA Frame Structure: A.) Data stream is divided into frames. B.) Frames are divided into time slots. Each user is allocated one slot. C.) Time slots contain data plus guard-period brackets for synchronization. [Graphic credit: Wikipedia]

by Mike Rivers

# Two-Track Editors for the Windows PC

A plethora of new flash memory and minidisk recorders has hit the market, so there will be more stereo recordings coming in to your shop that never saw the inside of a DAW. And nearly

all of these can benefit from some editing. You'll find that editing a stereo mix is somewhat different if your editing experience has been primarily in working with individual tracks in a DAW. It's much easier to identify edit points when you're looking at or playing a single instrument than a full mix. Any multitrack DAW program can edit a stereo file. But this round-up focuses on programs dedicated to two-track recording, playback, editing and processing.

Examined are several programs, which span the range from basic editing with a little tweaking to a full suite of mastering tools. All work with files in the WAV format, and most can import and export audio data in various data-compressed formats. So here's an overview to whet your appetite and give you some idea of what's available.

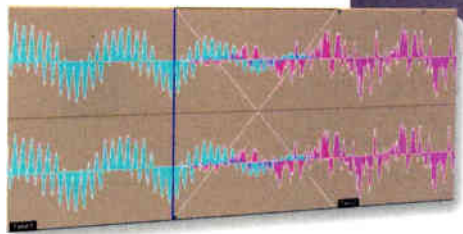
## SOUND FORGE 8 - THE ELDER STATESMAN

Sound Forge (\$299), originally developed by Sonic Foundry, started life as a sample editor back when Korg, Akai and Roland ruled the music sampler kingdom. It's now published by a division of Sony, and has grown from its modest beginning nearly 15 years ago into a very sophisticated program for working with stereo or mono recordings. Its sample editor heritage shows through in that it offers tools for generic loop editing, as well as creation and editing of loops for Sony's ACID music construction program.

Sound Forge uses the linear cut-copy-paste model for program editing. It crosses over to the source-destination model in that you can have multiple audio workspace windows open simultaneously, using one window as a destination in which to assem-

ble chunks of material copied from the source window(s). If I'm working on a project with a single long file as the source — cleaning up a concert recording for broadcast, for example — I'll work on it in a single window. Most of the editing for a job like this consists of cuts, and it's not likely that I'll want to rearrange the order of the songs. If I have several source files — several nights of shows to be compiled "best-of" fashion into a single program, for example — I'll open each source in its own window so I can pick out the best pieces for the compilation.

There are about as many ways of working a project as there are tools, and Sound Forge has a lot of tools. I found that it's possible to make very precise edits in Sound Forge. But an efficient way of compiling a project is to grab a bit more than you know you'll need from the source file, slap it in to the destination window (remembering to drop a marker to remind you that you have some trimming to do) and then trim up this rough compilation after you have all



the pieces in the proper order. Sound Forge allows you to preview the results of a cut before actually making it by playing a couple of seconds (configurable) before the cut, skipping over the cut and playing a couple of seconds beyond the cut. It's easy at this point to adjust the length and position of the cut if the pieces don't fit together smoothly, then

to give another listen.

Sound Forge has an extensive set of keyboard commands, far too many to remember all (I lost count at around 160). But it's worth taking the time to learn some. It uses the conventional Windows shortcuts for frequent commands including "Cut," "Copy," "Paste," "File Save" and "File Open." Most shortcut keys are pretty well thought out as to how they're placed on the keyboard, allowing you to work with the mouse in one hand the other hand on the keyboard. Right-clicking with the mouse brings up a short context-sensitive list of commands, which usually includes what you want to do.

A handy tool, which can be a default or selected when desired, automatically makes a cut at the waveform zero crossing nearest to the selection boundaries. You can further specify (another preference) whether you want to cut on a positive- or negative-going slope, or the nearest zero crossing. The cursor can similarly be made to snap to a zero crossing. Zero-crossing splicing seems like a good idea (it's a bit of 'Net wisdom that isn't really true). Unless you're splicing two waveforms of the same

frequency, however, there will be a change in slope at the splice point no matter where it occurs, and that can produce an audible click.

I consider it to be a serious omission that Sound Forge has no automatic or simple way to crossfade over a splice. It's possible that the developer believed the "zero crossing splice" theory and left it at that. There's a crossfade function, but this fades across the entire length of the selection, useful for fading music down when

narration comes up, but not for making "clickless" splices. It's possible to crossfade a splice using several steps, but it's not something that you'd want to do 100 times in a program.

The mouse wheel can either scroll through the length of the file or zoom in and out. Zooming with the mouse wheel makes this a breeze when trying to locate an edit point in a large file. It turns out in practice, however, that scroll wheel control is fussy about the mouse, or at least the mouse driver. Zooming didn't work on one of my computers that has an off-brand





mouse. The scrub tool is handy for "shutting the tape" and listening to the monkey chatter, however, it doesn't come near to emulating the feel and preciseness of rocking tape reels. You still have to locate tight edit points visually or guess-and-by-gosh. ("By gosh, that sounds OK to me.")

Sound Forge 8, being a Sony product, can import and export the latest version of AT-RAC compressed files used in Sony's new MZ-M200 Hi-MD recorder, as well as plain vanilla WAV files, MPEG-1 and -2, MP3, AIFF, OggVorbis and a bunch of formats I've never heard of. You can't edit video with Sound Forge, but it will import and export QuickTime, RealMedia and Windows Media files, displaying the video in frames above the audio track you're editing so you can see where to place cues. Audio — 8-, 16- and 24-bit — at sample rates from 2 - 192 kHz can be recorded, played, imported and exported. A resampling function allows you to convert sample rates up or down. There's a wealth of DirectX processing plug-ins bundled with the program, and it also supports third-party DirectX and VST plug-ins.

Sound Forge is fairly intuitive for basic operations, but it uses some unfamiliar terms for familiar items and functions weren't always on the menus where I expected to find them. The full manual, which I found difficult to use, is a PDF on disk. There's a printed Quick Start guide, which is a too-quick overview of the workspace and a rather sketchy reference to commands. I found myself getting befuddled quickly. There are plenty of worthwhile features here, but Sound Forge demands time to learn to use it effectively. I found myself looking through the printed

#### SONY SOUND FORGE 9

Since the completion of this round-up, Sony Sound Forge has undergone a major update. Sound Forge in its new Version 9.0 offers 16 substantial updates: multichannel audio recording, file editing and processing; drag-and-drop between channels; hardware meters with output gain control; phase and mono-compatibility meters; wet/dry mix and crossfade options for effects; Channel Converter tool for multichannel audio; multichannel spectrum analysis; enhanced user interface; multichannel WMF support; 5.1 Dolby Digital AC-3 export; Gracenote MusicID; Noise Reduction 2.0; iZotope mastering effects bundle; QuickTime and MPEG-1 and 2 import; and VST support.

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Keyboard Commands reference to find what I was looking to do.

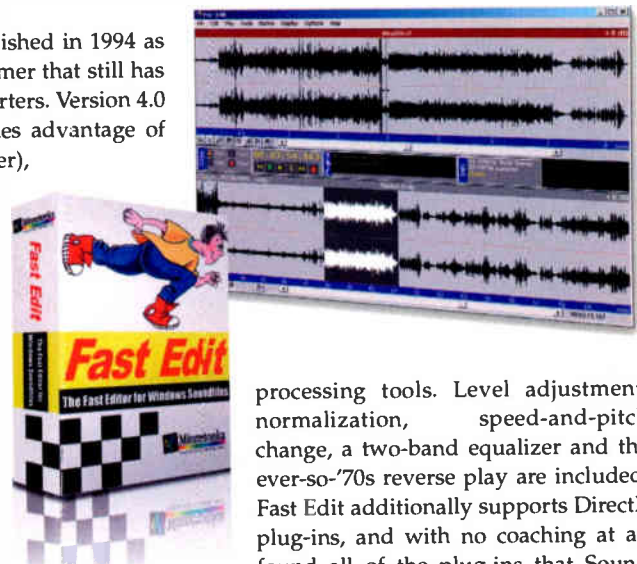
For more information visit [www.sony-creativesoftware.com/products/soundforge-family.asp](http://www.sony-creativesoftware.com/products/soundforge-family.asp).

#### FAST EDIT 4.0

Fast Edit, originally published in 1994 as Fast Eddie, is another old-timer that still has a large group of loyal supporters. Version 4.0 (\$199, released in 2001) takes advantage of 32-bit Windows (95 and later), but has changed little in terms of operation since the initial release. It now handles sample rates up to 96 kHz and word lengths up to 24 bit, storing high-resolution files in a 32-bit format. Minnetonka Software told me that a new release is in the works. The name describes the program well: It's fast, both in execution and operation, and it edits.

Fast Edit is a classic source-destination

editor. Open or record a file in the bottom read-only window, then paste segments (or the whole file) into the top window where clips can be trimmed, moved by cutting and pasting, and tweaked with fades and audio



processing tools. Level adjustment, normalization, speed-and-pitch change, a two-band equalizer and the ever-so-'70s reverse play are included. Fast Edit additionally supports DirectX plug-ins, and with no coaching at all found all of the plug-ins that Sound Forge installed on my computer.

**TWO-TRACK** continues on page 58 ►

Having only one source window might seem skimpy if you're assembling a project from several sources. But once a source file has been opened and its graphic data generated even an hour-long audio file opens instantly. It's no big deal to re-load a source. Most navigation can be done with just the mouse once you learn the "chords." Clicking the right mouse button while holding the left button, for example, switches between the play and selection cursors; right-clicking plays a selection. There are toolbar buttons for copy, paste, selection, loop playback of the selected area and zoom in/out, plus keyboard shortcuts for everything.

Tabbing moves between markers, and the < and > keys move to the next or previous splice (a very handy feature). You can also open a "History" list and navigate through an edited file by clicking on list entries.

Enable automatic crossfading and a crossfade is performed across each splice or the boundary of each cut is faded in or out. Default fade length is 10 ms, which seems to

work all the time; it can be easily adjusted, however. It's nearly impossible with cross-fading to make a splice that clicks. Fast Edit doesn't offer volume envelopes, but you can create complex fades or volume changes by setting markers at volume change points and performing a multi-segment fade between them. The starting and ending volume of each segment can be set, and you have a choice of a linear or logarithmic gain change between markers.

Normally the edited project is saved as a WAV file, but it can also be saved as an "Edit List" file. This is convenient if you'll be coming back to the project for another editing session. The Edit List saves the location of any markers you may have set in addition to conserving disk space.

The manual is concise and a bit more tutorial than Sound Forge's. It's not quite up to date with this software version — all the functions are there, but, for example, screen shots for the recording parameters stop at 48 kHz and 16 bits. The manual, like Sound Forge's,

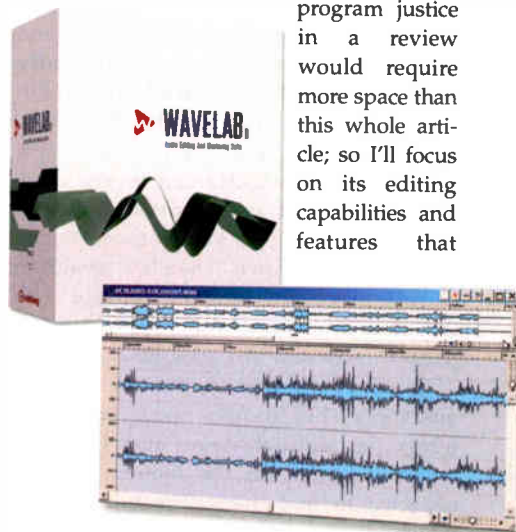
contains a long list of keyboard shortcuts that you'll learn when you use them often enough. The program is very straightforward. Once you learn the distinction between edit and selection cursors, everything else is pretty obvious. Fast Edit isn't loaded with features, but what it does it does very well.

For more information visit [www.minnetonkaaudio.com/fastedit/fastEdit.html](http://www.minnetonkaaudio.com/fastedit/fastEdit.html).

## WAVELAB 6

Calling Wavelab 6 (\$700) a two-track editing program is like calling a Ferrari a compact car. Yes, it is, but it's so much more. To do the

program justice in a review would require more space than this whole article; so I'll focus on its editing capabilities and features that



relate to editing.

There are two primary work spaces for audio editing. First is the Wave window, where drag-and-drop and cut-and-paste editing is performed. You can have as many of these open as you need for multiple source files, and freely drag selected areas from sources to one "master" destination Wave window.

The small upper panel provides an overview of the file, while the lower panel is where the work gets done. The zoom level and position of the two displays is independent, allowing you to use the overview to navigate through the file, bringing areas that need work into the main window. The mouse scroll wheel, like in Sound Forge, works as either a zoom or scroll, though they're opposites: Holding the Ctrl key in Sound Forge switches from zoom to scroll, while in Wavelab it switches from scroll to zoom. There's some intelligent interaction between the two windows. Selecting an area in the overview window sets the zoom level in the main window so that the selection just fills it.

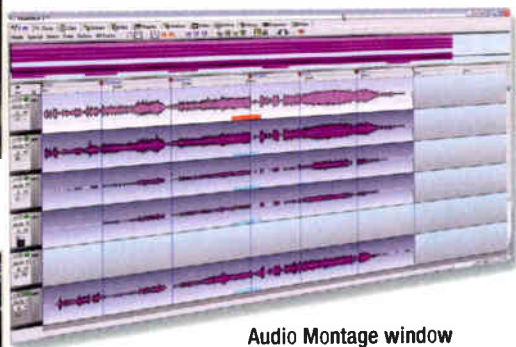
It took me a while to find it in the Wave window (it's called "Smooth Delete"), but you can do a smooth crossfade over the joint

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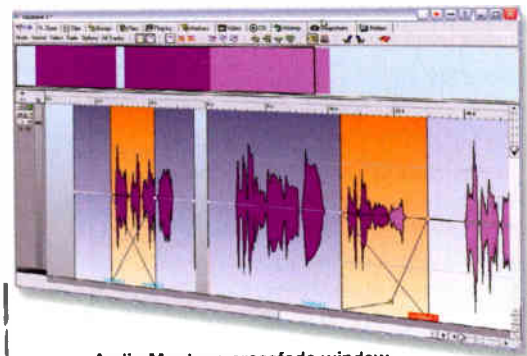


Audio Montage window

when deleting a segment. Except to paste with a crossfade (something that's automatic in Fast Edit) you need to work in the Audio Montage Window.

The Audio Montage window is Wavelab's object oriented editor. The overview window shows the arrangement of clips (objects) rather than waveforms, but the overall appearance is similar to the Wave window. Sections can be pasted from Wave windows, or a large file can be loaded into the Montage window and split into objects that can be individually manipulated.

It's usually easy to locate split points visually, and this is usually the quickest way to divide a concert recording into individual songs. Split points need not be terribly accurate since you can always resize clips just by dragging the edges. There's an auto-split tool that will split at markers, split at "silence" (defined by parameter you can set), or split at specific time intervals. Clients for CDs of con-



Audio Montage crossfade window

ference recordings often want them indexed at one- or five-minute intervals, even if that occurs right in the middle of a word.

You can perform all the standard editing functions in the working (lower) panel of the "Montage" window. Cuts and pastes can be made with crossfades, segments can be resized, crossfade times and overlaps can be adjusted, volumes can be matched, processing functions applied, and, when you're sat-

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ified with your edits the file can be rendered to create a new WAV file with all the edits and tweaks.

The cursor in Wavelab represents a rather large toolbox, and what it does is a function of its vertical position. This takes some concentration and a steady hand to be sure of doing what you want. There are a lot of functions and a lot of menus in this program — it's one where you'll find your repertoire expanding as you use it and discover a new tool.

Wavelab offers a slew of power tools under its "Analysis" menu. Most are oriented toward mastering tasks, but there are some that will help to make your edits better, or make your job as an editor easier. One example is the "Spectral Editor." This displays time (location in the audio file) along the horizontal axis and frequency along the vertical axis. Amplitude is represented by color. Opening this display and applying the Spectral Editor helps you locate a bothersome frequency and remove it from the audio file without cutting out everything else that occurs at the same time. It's an alternate solution to simply cutting out the segment with the offending noise.

The Spectrum Editor is similar to process-

ing a segment of the sound file with a narrow band pass filter (and this may indeed be what happens under the hood), but the value of the Spectrum Editor is the ease of visually locating trouble spot, and the analysis figures out what's required to remove or attenuate it.

It's particularly effective on two bugaboos of live recording: feedback and audience coughs. This illustration shows a burst of feedback marked for processing. Feedback is an easy-to-see example since it's usually, at least initially, a single frequency, that appears as a horizontal line in the spectral view. Note its color change from blue to yellow and back as the squeak builds and drops in intensity. A keystroke removed it without noticeably changing the underlying speech.

The "Average Loudness" analysis tool displays the apparent loudness over the selected area (or entire file). I found it to be a useful aid in matching levels when replacing a section with the equivalent piece of an alternate take. I could determine if a volume adjustment was required by analyzing the loudness of the original segment and its replacement, and, if so, how much of an

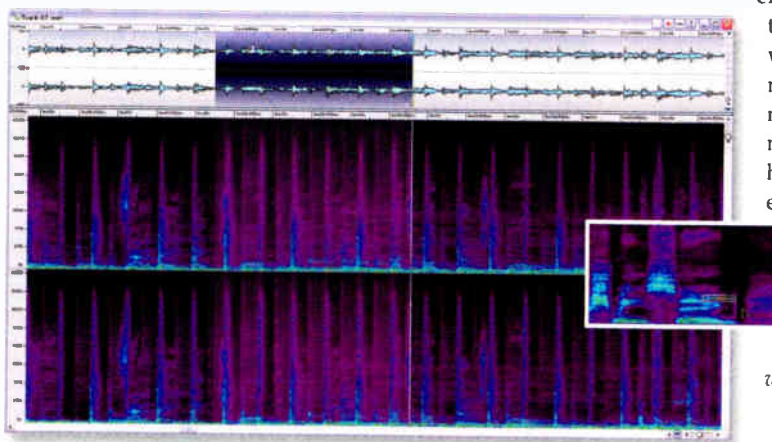
**TWO-TRACK** continues on page 60 ➤

**TWO-TRACK** Continued From Page 59

adjustment. It's a timesaver, and I've found that using the analyzer's numbers works the first time just about every time.

Normally I consider "normalization" to be a four-letter word. But Wavelab's "Loudness Normalizer" is interesting, though its effect can be quite comical if applied to the extreme. The Loudness Normalizer in better hands than mine when used by a skilled mastering engineer is effective for bringing levels into the ballpark when assembling a CD from various sources or evening out multiple speakers in a conference or forum recording.

Wavelab is a hugely powerful program, and this quick overview barely scratches the



Spectrum Editor Window and close-up.

surface. It can create and burn DVD-Audio discs as well as audio CDs, and it has a mul-

titrack mode for editing surround productions in up to 7.1 format. VST and DirectX effects are supported, and a tasty assortment is supplied with the program. It may be more than you need if you're not planning to go into the mastering business, and harder to learn for routine editing tasks than other programs. But it's a good choice if you want to have a wealth of tools at your fingertips.

For more information visit [www.steinberg.net](http://www.steinberg.net).

*Mike Rivers has a long list of credits with Rounder, Folkways Legacy and The Smithsonian. He has a degree in Electronic Engineering and is also the author of the last Mackie Hard Disk Recorder manual.*

## The Bargain Basement

### | SONY CD ARCHITECT 5

Sony's CD mastering program ("mastering" in the classic sense here, including writing and editing of subcode information) turns out to be a capable editing program in its own right. CD Architect 5 is bundled with Sound Forge 8 (or \$120 as a stand-alone download). And while it's not a true object-oriented editor it behaves like one in many ways. You can load the file of a whole concert recording into the Time Line window, split it into tracks at logical points (typically the beginning of each song) and then the fun begins.

"Splitting" doesn't actually modify the audio file, but it marks start and end points for each segment and develops a playlist for the project. Each segment begins and ends with a fade region (10 ms default), which you can modify by dragging a corner of the segment or selecting a different fade curve. Creating a real fadeout or fade-in is as simple as dragging the "fade" corner of a segment out to the length you need. Those wanting more control of transitions from one segment to another can open a second "layer" and create complex crossfade shapes between segments on each layer by using volume envelopes.

CD Architect is quite a capable editor once you get the hang of it. You can cut out extraneous material simply by dragging the edges of the boundaries, adjust spaces between the tracks, or crossfade one track into another with no silence. This is a cum-

bersome way to rearrange speech, but it's fine for shortening a rambling introduction and putting songs in their final order. You can burn a standard audio CD directly from the program, and you can also render the assembled and edited material into a single file, incorporating volume adjustments or any plug-ins that you apply. Those with the need and knowledge can enter ISRC codes and export the PQ code list. CD Architect could be all the editor you need for many projects.

### | SOUND FORGE AUDIO STUDIO 8

This is Sony's \$70 "light" version of Sound Forge. But it gives you similar functionality to the pro version with the exception of being limited to 16-bit audio files with sample rate up to 48 kHz and lacking the scrub function and volume envelopes. It lacks about half the processing tools and metering, so it's not as complete a "mastering" tool as the full version. But it will work fine to spiff up your field recordings. A plus, particularly if you're a beginner, is a good set of tutorials.

### | WAVELAB ESSENTIAL

This is Steinberg's \$150 scaled-down ver-

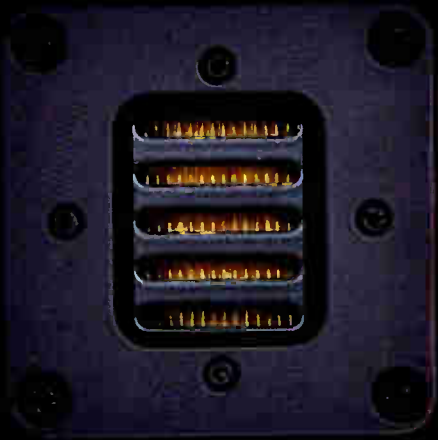
sion of Wavelab. It's based on Wavelab 4.0, offering both the Montage and Wave editing tools, CD-burning, ASIO and VST support, but sample rate is limited to 96 kHz. It's a good way to start out on the Steinberg path. If you like it and want the features of Wavelab 6 the upgrade is \$400.



### | AUDACITY

Audacity is a remarkably complete recording and editing program available free at [audacity.sourceforge.net](http://audacity.sourceforge.net). Audacity was developed by a group of enthusiastic volunteer programmers under Sourceforge's open source project, and includes many features of the commercial programs. It uses the cut-copy-paste model, and you can have as many windows open as needed to handle multiple source files. There are no automatic crossfades, similar to Sound Forge, but the program comes with a reasonable collection of tools and limited VST support. A nice bonus is that Audacity can play back and mix multiple files. This is handy in the production room when putting a music bed under narration, or even doing multitrack recording one track at a time.

— Mike Rivers



## The ADAM A.R.T. Tweeter Accelerated Ribbon Technology

ADAM's unique folded ribbon diaphragm moves air four times faster than any driver in any other professional monitor, resulting in incredible clarity, breathtaking detail and imaging like you've never heard before. You'll work faster, better, and more efficiency than ever.



### A7

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- 105dB SPL @ 1 meter
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- XLR balanced and RCA unbalanced connectors
- 2 year warranty

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## BBE® MaxCom Compressor - Nothing Beats a Good Squeeze

The BBE® MaxCom™ is a highly sophisticated Dual-Channel Compressor and Limiter with Gate, housed in a compact and easy-to-use package. It provides a live or recorded mix with all the clean, clear punch that only a truly professional dynamics controller can. However, the MaxCom is much more than a high-quality, superbly engineered compressor/limiter.



The MaxCom is the first signal processor of its kind to feature an on-board BBE Sonic Maximizer. The BBE High Definition Sound process employed by the Sonic Maximizer is world renown among music and sound professionals for remarkable improvements in total sound quality, bringing crystalline clarity and air to the source material as well as revealing rich harmonic subtleties. The MaxCom employs BBE's latest High Definition Sound technology, in the very same 4th Generation BBE Professional chipset which powers the 482i and 882i Sonic Maximizers.

The "over-engineering" in the MaxCom ensures that both its compression and gating provide versatility and excellent sonic performance in situations where other compressor/gates often produce undesirable processing artifacts. At the heart of the MaxCom is a THAT Corporation VCA known throughout the industry as the premier dynamics processing VCA. The same used in DBX's 166 compressor.

This single-rack-space dynamics processor includes two independent full-featured compressor/limiters, each channel having independent controls for Threshold, Compression Ratio, Attack Time and Release Time. Additionally, the MaxCom's "linking" function allows for phase-coherent tracking of stereo signals and adjustment from the Channel One controls. Comprehensive metering is achieved via dual bar-graph meters on each channel that indicate Input Level and Gain Reduction.

MaxCom's intuitive design makes it easy to operate, and its rigorous engineering and robust construction assure clean, distortion free audio and a predictable, repeatable range of control from undetectable to clearly identifiable. Whether the goal is to pull a "lost" instrument up in the mix, control vocal levels or dial in hard-limited speaker protection, the MaxCom can

accomplish the task with ease, reliability and sonic excellence. By offering a superb, full-featured compressor/limiter with an integrated BBE Sonic Maximizer system, the MaxCom has no equal at any price.

- BBE Sonic Maximizer adds clarity and definition
- "Auto" mode for easy setup
- Dual bar-graph meters for input level and gain reduction
- Stereo link for phase-coherent stereo compression
- Dynamically controlled, program adaptive gate for noiseless and seamless operation
- Full Function 4th Generation BBE Sonic Maximizer
- THAT Corp. VCA
- 2 channel compressor/expander unit with gate function
- Peak limiter to protect from clipping
- Inputs and outputs via balanced XLR or unbalanced 1/4" jacks
- Key in and out via 1/4" TRS

Frequency response: .....20-20,000Hz  
 T.H.D: .....Less than 0.05%  
 S/N ratio: .....More than 94dB  
 Threshold: .....-40dBu to +20dBu  
 Compression ratio: .....1:1-11:1  
 Attack time: .....0.1-200 msec  
 Gate threshold: .....off to +10dBu

Located in Huntington Beach, California, BBE Sound, Inc. has been providing music professionals and consumers with leading edge sound improvement technology since 1985.

In the professional music world, musicians perform with BBE Sonic Maximizer™ signal processors for the most dynamic and exciting audio quality possible. From small nightclubs to huge international tours, today's music professionals depend on BBE High Definition Sound technology to allow their full creativity and musical expression to reach the ears of their audiences. BBE High Definition Sound technology continues to evolve, taking the fidelity of both live and recorded sound to new heights of clarity, definition and sheer musicality.

BBE Sound continues to improve and refine BBE High Definition Sound technology for both the professional audio and consumer electronic markets, as well as developing next-generation audio improvement technologies. These next-generation technologies, such as BBE ViVa, BBE Mach3Bass and BBE MP, are already making their debuts in consumer electronics products from major manufacturers.

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## It's all Relative

Since introducing the C4 an amazing array of users have found both the compression and dynamic EQ features an invaluable feature for creating their mixes and letting them achieve results that have surprised long time industry veterans.



however it was soon discovered that this innovative threshold approach yielded great gains in a wide variety of situations. The "Relative Threshold" method varies from a conventional approach in that it compares the difference between the incoming signal and a user specified parametric filter placed in the circuit. Using a conventional threshold approach when the source signal varies in level, the amount of processing varies greatly. For example if used on a guitar amplifier and the volume is changed the user is forced to re-set the thresh-

old appropriately to compensate. By using the relative threshold approach Rane has created a dynamic EQ that lets the engineer focus on mixing instead of constantly adjusting thresholds and creates a more consistent sound. When used in normal compression mode the parametric filter is moved to the side-chain circuit allowing conventional broadband compression but lets the user make the compressor frequency selective a feature not normally found on compressors designed for the live engineer. The engineer has an amazing array of options available in one package to give them complete control of virtually any signal source.

In 2005 Serato Audio Research in partnership with Rane utilized these same DSP algorithms in TDM plug-in software form for Digidesign ProTools and specifically optimized for the VENUE live mixing platform.

Available in both a stand alone rack mount version from Rane and software plug-in from Serato Audio Research any engineer has access to this processing regardless of mixing platform, digital or analog.

Engineers such as Robert Scovill, Greg Nelson, Chris "Sully" Sullivan, Steve Emler, and others find the Rane processing a "must have" piece of gear whether in their racks or running in their digital consoles.

During pre-production rehearsals for the last Tom Petty tour Robert Scovill commented "I am currently using the Rane/Serato dynamic EQ on Tom's vocal — it is, by a thousand miles, the best thing I have ever used on his vocal. I am currently able to put his vocal in places within the mix that I have only ever dreamed of. It is truly startling and breathtaking. My system engineer and I are still trying to get our heads around it."

Robert introduced Greg Nelson, FOH engineer for Pearl Jam and Incubus to the technology and soon Greg was using it on a vast array of inputs and finding that the dynamic EQ solved many problems and allowed him to create a more consistent mix and was a must-have bit of processing.

Rane's approach to Product Development is to first understand the problem and then to provide appropriate solutions. This approach creates a problem solver and gives the end user the greatest benefit. We use the best technology for the best application.

This development cycle yields several innovative approaches for helping the live sound engineer by giving them the tools to process their audio in ways that have remarkable benefits in sculpting their mixes. Among the most revolutionary are the DSP algorithms introduced with the Rane C4, a 2U four channel Compressor/Limiter & Dynamic EQ. Of most interest to live sound engineers is the "Relative Threshold" approach to dynamic EQ. While a very small group of dynamic EQs have been available to live engineers in the past they all use a conventional threshold approach that limits their effectiveness for many applications.

Rane's DSP engineers originally approached the idea as an effective way to perform de-essing



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:: GREG NELSON, FOH: Pearl Jam and Incubus

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**IN THE CONSOLE OR IN THE RACK**

# CONTRACTING

The latest news and products

## NEW PRODUCTS

### LAB.GRUPPEN'S C Series Amplifier Line



With great power comes great responsibility, and Lab.gruppen's C Series Amplifiers are responsible for great sound from great power. The C Series employs a unique Class D implementation, more appropriate for lower power levels than Class TD. The amps power up with a universal switching power supply employing PFC (Power Factor Correction) to stabilize current draw and accept any mains voltage from 90 – 265 VAC (50 Hz or 60 Hz). Put the power in your hands through Lab.gruppen's proprietary NomadLink control and monitoring network, included. With the C Series, full power without clipping is assured, as overall input sensitivity on all models is adjustable in 3 dB increments from 23 – 44 dB.

**PRICES:** C 20:8X (8 x 250 watts) - \$3,095; C 10:8X (8 x 125 watts) - \$2,195; C 10:4X (4 x 250 watts) - \$1,795; C 5:4X (4 x 125 watts) - \$1,495.

**CONTACT:** Lab.gruppen | ☎ 818-665-4900 ↻ www.labgruppen.com.

### MARTIN AUDIO Ceiling Series Speakers



Let heavenly sounds rain down with these new two-way ceiling speakers sporting soft dome tweeters for exceptionally wide dispersion, making them ideal for low ceiling areas. Ceiling Series speakers comply with UL2043, UL1480, BS5839 pt8 and IEC60849/EN60849 standards, and feature extensive fireproofing. Models are available in 40 W 4-inch (100 mm), 75 W 6.5-inch (165 mm) and 125 W 8-inch (200 mm) versions, all at 16 ohms featuring switched 100/70 V line transformers with excellent low-frequency characteristics. There should

be no ceiling on quality sound, but there should be quality sound on the ceiling.

**PRICE:** \$129 - \$269.

**CONTACT:** Martin Audio | ☎ 519-747-5853 ↻ www.martin-audio.com.

### PRIMERA Bravo SE Blu



Ready to step up your game, developers? Or maybe that's: Ready to step up, game developers? Either way, this professional disc publishing system is ready for you. Whether a PlayStation3 developer or a studio professional exploring highest definition formats, this computer-automated Blu-Ray/DVD recorder (CD to come) is ready to pump out those senses-shattering discs. And while the Mac and Windows Vista-compatible USB 2.0 unit can't make the beautiful surround sound for you, it's 20-disc capacity and 4800 DPI Direct-to-Disc printing can churn out beautiful discs for whomever needs to hear them.

**PRICE:** \$2,995.

**CONTACT:** Primera | ☎ 763-475-6676 ↻ www.primera.com.

### AVIOM AV-M8 Mic Input Module



Sometimes setting up a conference room can be frustrating, boring. Well, take the bored out of boardroom with the Aviom AV-M8, offering eight analog audio channels/mic level inputs (via Euroblock connectors) transported across easy-to-install Cat-5e cable (thus sidestepping the need to drill conduit). Two channels include 30 dB switchable pad, allowing line-level signal, while all feature configuration DIP switches for gain, 48 V Phantom Power and low-cut filter. Utilizing plug-and-play and A-Net, the AV-M8 allows quicker timelines and is easily integrated into larger Aviom systems.

**PRICE:** \$1,499.95.

**CONTACT:** Aviom | ☎ 610-738-9005 ↻ aviom.com.

Ride 'em, cowboy! A Midas XL8 digital console ran the show at the Houston Livestock Show & Rodeo. Sound, via 108 **Electro-Voice** X-Line boxes, was provided by Houston's LD Systems. Guests included George Strait, ZZ Top and Houston's own Beyoncé.

Continuing the Texas rodeo theme, a new **Electro-Voice** speakers system (ZX5 speakers and QRx218S subwoofers) has been installed at Amarillo's National Center and Tri-State Fairgrounds. Work was performed by Little Orchard Sound of Newcastle, Okla.

A little edgier is the scene at The Venue, a club in Salt Lake City. New there is a **Soundcraft** Series FIVE console, **JBL VERTEC** speakers, **Crown** CDi 1000 and 2000 amps along with a smattering of **dbx** and **BSS** processors. Installation was courtesy of Sonoloux LLC.

Sparing no expense was the First Cathedral of Bloomfield, Conn. It recently installed a **Soundcraft** vi6 digital console – to go with a **Soundcraft** MH3 for monitors. Of course, **JBL** speakers, **Crown** amps and **BSS** processors followed the camel's nose.

The International Finance Corp., part of the World Bank, recently installed a **Beyerdynamic** MCW-D 50 wireless conference system at its headquarters in Washington, DC.

Columbia University recently installed **Turbosound** TA-500 speakers and TSB-118 subwoofers into Lerner Hall. Also in New York, the ever-so-trendy club, Duvet New York, installed TA-500 speakers and TSW-218 subwoofers.





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# Disc Makers Elite4 Automated DVD/CD Duplicator

An automated publishing unit for those with a burning desire to duplicate.

The Disc Makers Elite series — a CD/DVD publishing unit with robotic loading, four burners and an inkjet printer — has been around since the late '90s. The recent Elite4 (priced at \$4,290) can now produce over 175 copies from a physical master or disc image in a single session and supports CD, DVD plus double layer DVD formats. A pre-assembled industrial assembly houses the Autograph7 4800 dpi full-color printer, four NEC ND-3550D Dual Layer DVD±R burners, and robotics bolted to a sturdy metal base. And — once connecting two power cables, one for the printer and one for the robotics/burner, plus the USB or FireWire ports — the Elite4 now interfaces with PCs and Macs to burn, print, or both.

## FEATURES

Disc Makers suggests the following minimum Mac configuration: An 800 MHz G5 running OS X 10.3.7 or higher, 512 MB RAM, 40 GB, two available USB 2.0 slots,

7200 rpm hard drive with 70-percent available space (10,000 rpm drives SATA or SCSI drives are suggested). I loaded the included CD of software — Discrite 5.3.14 from CharisMac Engineering and Discus LE 3.15D, a limited version of Magic Mouse's Discus printing software (that only prints on the media, not on inserts) — and then read the small, easy to follow manual.

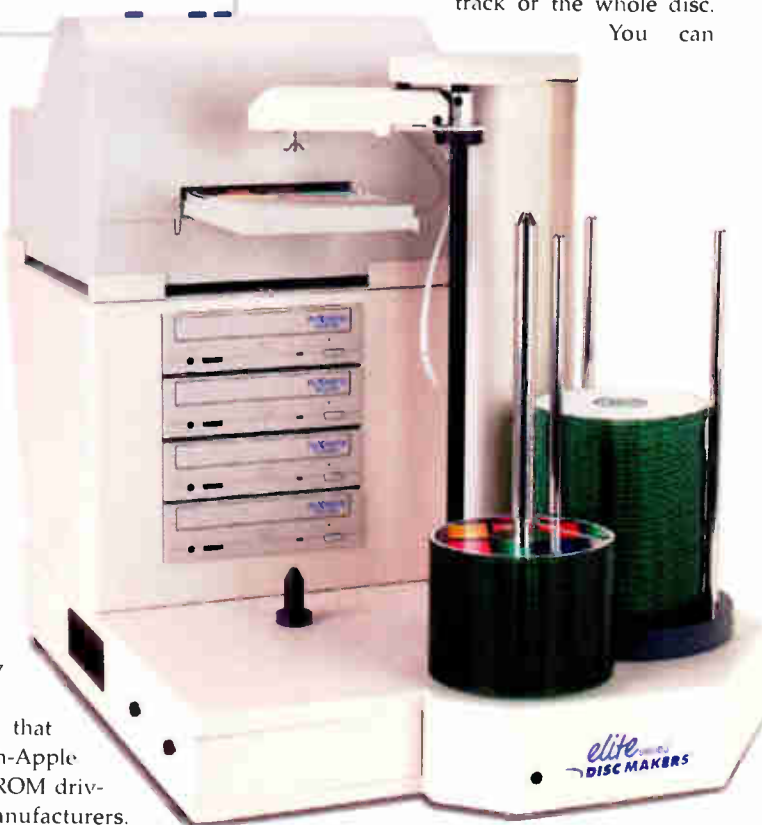
Discrite suggests that you remove any non-Apple and third party CD-ROM drivers from other manufacturers. According to the manual, Discrite's CD-ROM extension is capable of supporting virtually all SCSI, ATAPI, and USB CD-ROM and CD-R/RW drives on the market. After disabling any unnecessary extensions, the manual recommends you restart your Macintosh prior to installing Discrite to avoid installation interference and forcing a restart.

I hadn't read that part of the manual before I installed the software on my dual 2GHz processor G5 Mac and ran my first short CD batch. The only problem I ran into was iTunes wanting to start up every time I put a CD in any of the bays, but changing iTunes preferences fixed that. I broke the rules and tried running different apps,

checking for email, and doing a bit of sloppy FireWire connecting while burning single CDs. Sometimes I got away with it. Sometimes I didn't.

Disk Makers says stick with media of the same speed for each burn run, and although slower media is acceptable, the system is designed to work best with 16x DVD±R/48x CD-R. Special kits are available from Disc Makers to accommodate different disc sizes including business card-scale. While DVDs that have been produced with copy protection schemes cannot be duplicated, commercial audio CDs are quite easily ripped, whether one track or the whole disc.

You can



## FAST FACTS

### APPLICATIONS

Studio and House-Of-Worship; CD/DVD Burning and On Disc Printing

### KEY FEATURES

Mac OS X system for burning and printing 200 CDs or DVDs at a time

### PRICE

\$4,290

### CONTACT

Disc Makers | ☎ 807-707-0012  
➔ www.diskmakers.com

also use the system to erase CD-RW discs.

Starting Discrite brings up a Quick Start window that prompts you to choose "Data CD," "Audio CD," or "Copy a CD/DVD." You can stick with that or change the start-up window in Discrite's preferences. The standard window also provides a drop menu that allows you to select from Macintosh HFS, ISO 9660, ISO 9660 XA, Build a Mac HFS, Mac/ISO Hybrid, Disc Copy Image, CD/DVD Copy, CD Extra, Video CD, DVD VideoAudio, UDF, and Stream. Pick the one you want and click on START to begin copying.

You can record an audio CD from any combination of MP3/AIFF/SDII/WAV files

— just drag and drop them onto the Discribe window much the same way you might with Roxio's Jam. Curiously, Discribe creates new

## PRODUCTPOINTS



- Full CD-R and DVD-R copying with integrated inkjet printer
- Print, burn or both automatically, label design software included
- Integrates easily with Macs



- Four burners, a printer and a robot aren't cheap
- Unnecessarily makes AIFF files for CD burning
- Requires full attention of host computer for big runs

## SCORE

Four burners equal one easy solution for batch jobs

AIFF files from MP3 files as well as my WAV masters. Those conversions, while short, do take a few seconds each and, worse, require more disk space. Perhaps the team who wrote this Mac version was unaware that Macs handle WAV files these days. While you can adjust pre-gap times, title names, change the order of the files and play them, Discribe does not support the crossfades, normalization, text, and other metadata that Jam supports. You may have to do that fancy work with Jam and burn a disc to be your master.

### | IN USE

If you already have a CD or DVD master, simply load the disc into your computer or one of the Elite4 drives. Strangely, after I chose to put a finished CD in the Elite4's top drawer, Discribe told me there wasn't anything in the number one or number four drive. When I selected drive two, Discribe acknowledged the CD.

Burning with no printing takes less time, obviously. The system cranked out a dozen copies of a 25-minute audio CD in 14 minutes. According to the manual you can choose test then write, or write and then select write speed. Discribe supports "Verify After Writing" and "BURN-Proof/JustLink." Discribe can write CDs in the background with BURN-Proof/JustLink drives, leaving your Macintosh free for other tasks. Based on some occasional flakey results when I was running apps, however, I guess this unit's drives don't fully support that feature.

My first run was a dozen DVDs. During the first load, the progress window on the

number four drive did not move. After the other three bays had burned and the first DVD that was in printer number four opened, the robot arm extracted the DVD, the door closed, and the robot put the DVD on the deck not on the spindle. Ah, the reject depository! The next two DVDs in that burner went through without a hitch. At the end of the burn session, the system knew I wanted a dozen and my computer screen prompted me for another blank to complete the job.

I slipped one in the supply silo, and it was quickly picked up and delivered to the burner and printer. Done!

The ink-jet printer uses standard HP #56 and #57 ink cartridges. Coverage has a lot to do with how much ink you use, but Disc Makers says you can expect about 300 CDs per set of cartridges. Adjusting for how much ink, exactly how wide the blank's print area is, and how large the hole is takes some

**DISC MAKERS** continues on page 70 ►

**MORE THAN JUST A LOUDSPEAKER**

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tweaking. My advice is to find a blank you like and keep using it so you won't have to readjust the outside and inside diameters, or keep a log so you can make adjustments reliably and quickly. The test process that calibrates the exact center of the printed area isn't all that difficult, but you make some calibration "coasters" in the process. The Discus 3.15 manual suggests using 117 mm for the diameter, but that was about a millimeter short to cover the edges of the Ultra White Inkjet CDs they were thoughtful to send me. I evolved to 119.4 mm and 37 mm. That did the trick, though what you see in the print preview window is not quite what you get.

When you create a label with Discus 3.15, it makes a bmp file. Then you quit Discus and open Discribe to access the bmp file, print, and burn. If you only want to print discs, there's a handy utility in the Discribe menu that lets you choose the bmp file and do a print run.

You have to pay attention during print setups to make sure you choose the "disc printer" and "disc" paper from "Page Setup..." in the Robot Write window. When you click on "Print Settings" to actually print, you have to choose "disc printer" again. Hopefully, future versions will be more streamlined. Next you have to visit the "Copies & Pages" drop menu and "Paper Type/Quality" menu to choose the right paper. From the "Paper" menu I was coached to use "HP Premium Plus Photo Paper, matte."

The printed copies looked very nice. I had also heard about the new Taiyo Yuden inkjet printable glossy CD blanks with the glossy

paper settings. They cost more, but I like how the larger glossy surface (23.8 mm inside to 119.4 mm outside) looks very professional. Unless you know what to look for on the data side you might mistake the CD-R for a stamped CD. However, you do need to let them dry a bit, especially if you use a thick paint spray. But after they dry you can hold them under a spigot and rub them with your finger with no smearing — very, very nice.

I had problems with my first 30 CD burn/print run. The system stopped twice, I had to quit and restart the program, and then the computer. On that run, I had put the master in my computer's CD/DVD drive and hadn't clicked the "Cache Entire Disc Before Writing" button. Disc Makers forwarded a diagnostic kit to help find the problem, but before I used it I tried another 30 CD burn/print run with the cache setting selected. Bingo! All 30 made it through in about an hour-and-a-half. Several weeks later, I needed both a 75 and a 25 burn/print run and, with the cache feature enabled, had no problems.

### | SUMMARY

A lot of people don't remember the shift from black-and-white to color TV, or from AM to FM and FM Stereo. My point is once you experience on-disc printing, especially with the glossy blanks, it's very difficult to go back to flat, stick-on labels. Add to that the time savings created by having your own personal robot to take care of business and the argument becomes even more compelling. Sure, you have to make time for the robot because you can't be doing sessions, checking email or hacking around in general on the same computer you have connected to the Elite4. To get total freedom you need to have another computer with the right stuff, or Disc Makers also sells the ElitePro4 complete with its own computer. Having four burners and only one printer can create a production bottleneck, but I'm sure Disk Makers is working on a solution.

*Ty Ford has been reviewing pro audio gear for more than a decade and is a regular contributor to Pro Audio Review. His audio equipment reviews and V/O sound files can be accessed at tyford.com.*

### | BLU-RAY IN THE ROUND

Blu-ray Disc — named for the blue-violet read/write laser used on this emerging high-data-density format — will surely be an increasingly important storage medium for high-definition audio. After all, the Sony-developed Blu-ray Disc can store 25 GB per layer, and a few manufacturers have already introduced both single and dual layer (50 GB) recordable and rewritable offerings that support the format.

Media duplication manufacturers such as Disc Makers are taking note. The company introduced the ReflexBlu Line of Blu-ray Tower Duplicators as recently as March 2007. The first two models — ReflexBlu2 and ReflexBlu4 — are all-in-one creation, editing and duplication solutions for Blu-ray discs (BD-Rs), DVD-Rs and CD-Rs. ReflexBlu2 (\$2,999) has a duplication capacity of two BD-Rs per hour; ReflexBlu4 (\$4,999) can complete four BD-Rs per hour. Other features for both models include a complete disc creation software suite, USB connectivity for access to one drive for mastering and playback, and a 250 GB hard drive.

— Strother Bullins

**prismsoundrecording**

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# LIVE

The latest news and products

## NEW PRODUCTS

### CREST AUDIO CV-20 Live-Sound Mixing Console



No matter the house it's in, the Crest Audio CV-20 mixing console feels at home. The CV-20 is available in 24, 32, 40, 48 and 56 mono input configurations, with four-band sweepable EQ and fully parametric mid-frequency control on each. A wealth of audio

routing/panning is available through twelve aux sends, eight subgroup assignments and more. Full VCA control is on all inputs, groups and eight masters. There is LED bar graph metering on inputs and outputs, and an optional meter bridge offers VU. Assignment and mute scenes feature MIDI control, and a bus link module links two consoles.

**PRICE:** \$15,695 - \$30,125.

**CONTACT:** Crest Audio | ☎ 601-486-2000 ↪ www.crestaudio.com.

### L-ACOUSTICS KIVA System



When these speakers speak, you'll be happy to listen. The L-ACOUSTICS KIVA system is sleek, fully integrated and designed to meet the most strenuous demands. Components include the KIVA 80 Hz – 20 kHz full-range element (two 6.5-inch drivers in a bass-reflex tuned enclosure and a HF 1.5-inch diaphragm compression driver coupled to a DOSC waveguide for HG reproduction), a complimentary KILO 50 – 100 Hz low frequency extension (a single 12-inch neodymium driver in a tuned dual-chamber bass-reflex enclosure) and the LA4 amplified DSP controller (featuring 4 x 1,000 watts at 4 ohms, Network Manager software and Soundvision 3D simulation). With Wavefront Sculpture Technology, the system allows for even coverage among the largest of acoustic environments. The KIVA and KILO's modest sizes and low weights allow for almost any rigging situation.

**PRICES:** KIVA - \$2,350 each; KILO - \$2,650; LA4 - \$5,775.

**CONTACT:** L-ACOUSTICS US | ☎ 805-604-0577 ↪ www.l-acoustics.com.

### EVENTIDE TimeFactor



Eventide's legendary effects processors have certainly racked up the accolades over the past three decades-plus. Now Eventide stompboxes offer signature effects without the need for a rackmount. These rugged cast metal units are small enough for a pedalboard but pack that big studio-quality sound. The TimeFactor features 10 of Eventide's most popular delays, including digital, vintage DDL, tape echo, ducked, multitap and reverse. Users can customize

parameters for up to 27 native presets (or unlimited by MIDI), and access them through three footswitches. Instantaneous program change, true analog bypass, tap tempo, full parameter real-time editing via 10 onboard knobs and a plug-and-play expression pedal input for wet/dry mix control are just some of the convenient features. Additional functionality/upgrades are available through a USB port.

**PRICE:** \$499.

**CONTACT:** Eventide | ☎ 201-641-1200 ↪ www.eventide.com.

### QSC SC28 System Controller



For big sound look to this little box. This slim, two-input/eight-output DSP unit houses presets to optimize QSC loudspeakers (new tunings can be incorporated by USB), as

well as user-adjustable EQ and delay. The 48-kHz, 24-bit A/D and D/A conversion, balanced, line-level, analog XLR connectors, Infinite Impulse Response and Finite Impulse Response filtering, plus thermal and excursion protection all combine to make speaker arrays sing in acoustic correlation.

**PRICE:** \$1,331.66.

**CONTACT:** QSC | ☎ 800-854-4079 ↪ www.qscaudio.com.

Engineer Gene Gerhiser used a **True Systems** Precision 8 preamplifier when he recorded Yo-Yo Ma and the New World Symphony recently in Miami. Gerhiser used Schoeps MK4 capsules and **Neumann** 140 mics to feed the preamp.

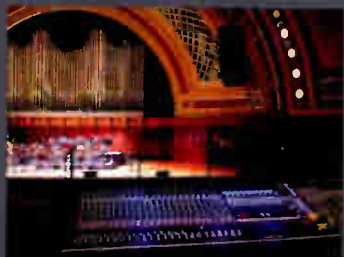
Gnarls Barkley lead Cee-Lo has been using a Neumann KMS 105 capsules and **Sennheiser** SKM 5200 handheld transmitter in recent live appearances.



Early users of **Sensaphonics** new 3D Active Ambient in-ear monitors include Ani DiFranco, Bela Fleck, and members of the Dave Matthews Band, Blue Man Group and the Rolling Stones.

Engineer and equipment designer extraordinaire Fred Forssell used **Lectrosonics** SM transmitters and Venue receivers to record several acts at the Wintergrass music festival in Tacoma, Wash. He used DPA microphones for many of the feeds.

Recent concerts such as Wynton Marsalis and Gilberto Gil at the Hill Auditorium in Ann Arbor, Mich. utilized **Electro-Voice** XLC speaker arrays, EV Dx38 processing, **Klark Teknik** DN EQs and a **Midas** Legend 1000 console. All provided by Aerial Enterprises or Whitmore Lake, Mich. An XLC array was also used at the Winter Thaw concert in Madison, Wisc. Sound there was provided by Intellasound Productions of Verona, Wisc.





# Contain Yourself

We're not necessarily referring to your enthusiasm for these four exciting new MG-Series mixer models. It's more about the unique single-knob compressors on their mono inputs, designed to keep loud from getting too loud and soft from getting lost in the mix. Combine this with MG's other performance-enhanced features and it's hard to resist a visit to your local Yamaha Live Sound dealer for closer examination.



# YAMAHA

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- Switchable Monitor Mix Routing
- Digital Multi-Effects (MG82CX/MG124CX)
- Mic Stand-Mountable (MG102C/MG82CX)
- \$149.00 - \$379.00 Suggested Retail Price

by Tom Young

# Analog vs. Digital Live Sound Mixing: Will We Ultimately Have a Choice?



which seemed to fix the problem. Anyone who has experienced this during a show will tell you it takes minutes, not seconds, to reboot a lot of these products, and in the live world that is an eternity.

Other discussions with some engineers have revealed the prominent opinion is that analog mixing days are numbered; and those numbers are ONES and ZEROS. I am personally concerned about this increasingly digital trend, and hope that is not the case, because I love analog. I started mixing on it many years ago, and I have affection for what it does to sound. But many of us might not have a choice as the analog market becomes smaller — driven by manufacturers discontinuing products and not providing support for old products.

I travel around from venue to venue mixing Tony Bennett, and the question I am most frequently asked is, "What digital console do you prefer?" The advantages to digital have become obvious, and when speaking with different engineers most agree that it is ultimately the wave of the future. Many agree during these conversations, however, that they still prefer analog mostly because it just sounds better. I read John Gatski's comments in the November 2006 *PAR* publisher's note, "Analog Ain't Dead," but I wonder with the marketing efforts of major manufacturers and potential profit margin in digital if analog's mixing future is in jeopardy.

The constant debate over which is better for mixing — analog or digital — still hasn't been decided. Analog sounds better in my opinion; digital puts many more features at your fingertips, however, and seems to provide value in its multitasking and processing capabilities. It appears fewer engineers have been desperately clinging onto their analog mixers because of the availability of digital consoles. I can say that in the last several years I have stored my show on every available high-end digital console except the new Midas product. Almost all engineers I have talked

to agree that on a simple one-off gig, analog is still faster to get a mix up and running.

Another issue that constantly comes up is the lack of comfort level of engineers to troubleshoot potential problems in digital opposed to analog. Most professional sound engineers understand analog console signal flow. When there is a problem the fix is often as simple as moving an XLR input or output cable to an unused input or output module. Most engineers, however, have experienced the frustration of when their personal computer locks up and usually their only alternative is to shut it down and restart it hoping for the best.

I recently did a show using a major manufacturer's digital console, and the console was randomly going back to previous settings or scenes and wiped out all my monitor gain settings. My system tech called the manufacturer and the comment was, "It shouldn't be doing that." Well, guess what: It did. The only solution again was to shut it down and reboot it,



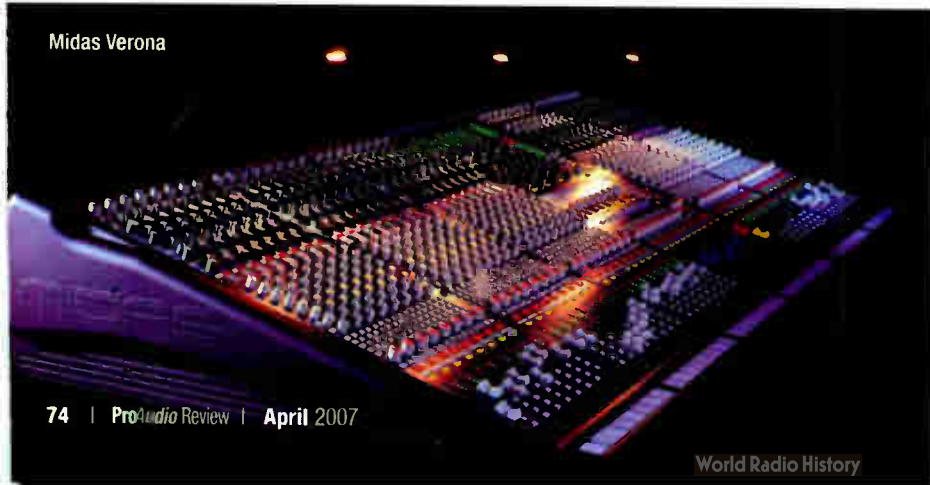
Yamaha PM5D

I am also concerned that the level of manufacturers' customer support for digital products is not at the level it should be. Company's like Yamaha provide 24/7 customer service support and hold digital console training classes for their line of digital consoles throughout the year and in various locations. But the training and staffing of specialists is not growing as fast as the products.

I have been doing live sound for a fairly long time, starting as a professional engineer when I was 20. I know a lot has changed being that was 30-plus years ago. There is a good chance if I started in this business five years from now and started to mix live shows I might never experience an analog console. Let's all hope that will not be the case, since there is still plenty of room left for both analog and digital mixing mediums.

*Tom Young is the live sound engineer for Tony Bennett.*

Midas Verona



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## API Vision Surround Mixing Console

**FEATURES:** 24 main, three stereo buses and a dedicated 5.1 bus; two 100 mm automated faders on each input, five-channel panning on each fader; comprehensive multi-format monitor facilities; re-settable switch assignment; API Legacy Series mic preamps,



equalizers and dynamics modules; custom built to specification;

five-year warranty on all parts.

**CONTACT:** API at 301-776-7879,

www.apiaudio.com.

## SOLID STATE LOGIC AWS 900+



**FEATURES:** 24-channel; SSL G series compressor; SSL SuperAnalogue mic preamps; SSL Twin Curve four-band parametric EQ; 5.1 surround sound; compatible with DAWs; TFT touchscreens; VU meters; includes legs, Total Recall and AWSomation.

**PRICE:** \$93,500.

**CONTACT:** Solid State Logic at 212-315-1111, www.solid-state-logic.com.

## HARRISON Trion

**FEATURES:** Post, live or broadcast architectures; 64 – 360 digital audio channels; 48 main buses, 16 aux. buses, eight control groups, four stereo program



outputs; 16 – 56 motor-driven faders; 40-bit internal signal path; XEngine Native DSP with 32- or 64-bit floating point audio at 44.1, 48 or 96 kHz sampling rates; 8-Band Parametric EQ, dynamics with compressor, limiter and gate; TFT displays; IKIS technology; XRouter.

**PRICE:** \$90,000 - \$200,000.

**CONTACT:** Harrison at 615-641-7200, www.harrisonconsoles.com.

## YAMAHA DM2000 VCM

**FEATURES:** 96-channel; 24-bit/96-kHz

processing; onboard DSP effects; Virtual Circuitry Modeling simulates '70s analog; Interactive Spatial Sound Processing (iSSP);



LCRS, 6.1 surround sound; joy stick; 100 mm motorized faders; LCD screen; SMPTE; word clock; linkable; Windows/Mac PC control software; I/O expansion slots.

**PRICE:** \$19,500.

**CONTACT:** Yamaha at 714-522-9011,

www.yamahaca.com.

## SOUNDCRAFT Ghost LE



**FEATURES:** Analog 8-bus with Mix B path; 10 auxes plus two stereo pair; 24/32-channel frames; 24-channel expander; four-band British EQ; ProMic preamps with 60 dB range; onboard computer; SMPTE timecode generator; 100 mm faders; machine control.

**PRICE:** Starts at \$6,295.

**CONTACT:** Soundcraft USA at 818-920-3212, www.soundcraft.com.

## STUDER Vista 5 Digital Mixing Console

**FEATURES:** Modular design; up to 240 channels; 24 bit/96 kHz; LCR, LCRS, 5.1 Studer Virtual Surround Panning; SCore Live DSP engine; Vistonic LCD screen control surfaces; 32 100 mm faders; talkback section;

rackmountable I/O.

**PRICE:** Starts at \$120,000.

**CONTACT:** Studer USA at 818-920-3212, www.studer.ch.

## AMS NEVE 88RS



**FEATURES:** 96 analog channels; > 24 bit/192 kHz; Encore Plus automation; two motorized faders per channel strip, with fader

starts and Overpress PFL; "Classic Neve" processing; enhanced spectral formant equalization; soft knee compression; 5.1 monitoring; 24-step precision volume; acoustically optimized frame.

**CONTACT:** AMS Neve at 425-454-9966, www.audio-agent.com.

## EUPHONIX S5 Fusion



**FEATURES:** 224 x 168 paths at 48 kHz; DF66 DSP SuperCore engine with 38 channels, scalable to over 100; expandable System 5 surface design with 24 multi-format faders, 8 touch-sensitive knobs per channel; eMix (with optional joystick); TFT multi-format metering; Euphonix converter, digital surround monitoring; EuCon Hybrid protocol for controlling up to five DAW workstations.

**PRICE:** Starts at \$150,000.

**CONTACT:** Euphonix at 212-889-6869,

www.euphonix.com.

## MACKIE VLZ3 Series

**FEATURES:** 12 – 16 channels; compact and ergonomic; XDR2 Extended Dynamic Range mic preamps; independent EQ controls; ultra-clean summing bus with extended headroom; built-in universal power supply; rugged steel chassis.

**PRICE:** \$389.99 - \$1,099.99.

**CONTACT:** Mackie at 800-258-6883, www.mackie.com.



## TRIDENT Series 8T

**FEATURES:** 16-, 24-, 32-channel frames; Trident S20/4T Celebration channel mic preamp; Series 80B replica EQ; 8-ax sends; monitor bus; mono selector; analogue VU meters; optional meterbridge; talkback section; two-year warranty.

**PRICE:** Starts at \$3,498.

**CONTACT:** Trident Audio at +44 1474-815-300, www.tridentaudio.co.uk.



## TL AUDIO M1 Tubetracker

**FEATURES:** 8, 12 input channel frames; tube preamps; three-band EQ with sweepable mid/bypass; two



aux sends per channel; effects return; alternate monitoring options; 100 mm K Series faders; stereo VU meters; optional 8-channel ADAT interface; 24-bit/96-kHz mix output option.

**PRICE:** Starts at \$4,320.

**CONTACT:** TL Audio at +44 1462-492-090, [www.tludio.co.uk](http://www.tludio.co.uk).

## ALTO Typhoon4800



**FEATURES:** 48 inputs with 40 mic preamps (48V Phantom Power); inserts on every channel; four band parametric EQ with sweepable center; eight aux. sends/returns; eight subgroups; talkback functions; 66 pounds; includes plywood flight case.

**PRICE:** \$4,718.

**CONTACT:** ALTO/The Yorkville Group at 716-297-2920 ext. 32, [www.altopa.com](http://www.altopa.com).

## BEHRINGER EURODESK MX9000

**FEATURES:** 24 channels; Balanced Mic/Line Input path with invisible mic preamp, pad, inserts, four-band EQ with semi-parametric mid-frequency bands, 15 dB boost/attenuation plus low-cut filter, pan pot, solo and mute; Mix B/Tape Return path with two-band shelving EQ, pan, level and mute controls; six aux. sends; 100 mm faders; ULN circuitry; LED meters; external rack-mountable PSU.



**PRICE:** \$1,629.99.

**CONTACT:** Behringer at 425-672-0816 or [www.behringer.com](http://www.behringer.com).

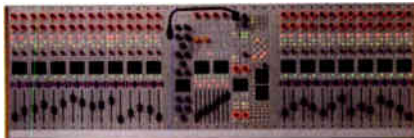
## CALREC Omega with Bluefin

**FEATURES:** 160 channels processing paths packaged as 48 stereo and 64 mono channels on one DSP card; up



to 64 dual layer faders; 19.6 minutes audio delay; 8 x mono, stereo of 5.1 groups; additional VCA grouping; 2 x main stereo or 5.1 outputs; 20 auxes; 48 multitrack outputs; **CONTACT:** Calrec at 917-825-3728 or [www.calrec.com](http://www.calrec.com).

## LOGITEK Artisan Console



**FEATURES:** Router-based for versatile operation and easy selection of sources to faders; modular design allowing 2 – 30 fader; two master mixes; eight submaster mixes; four aux. mixes; 24 mix-minus busses; one-button capture/recall of console layouts including EQ and dynamics settings.

**PRICE:** Under \$60,000.

**CONTACT:** Logitek Electronic Systems at 800-231-5870 or [www.logitekaudio.com](http://www.logitekaudio.com).

## ALLEN & HEATH

### MixWizard WZ3:16:2

**FEATURES:** 16 mic/line channels; balanced XLR/TRS jacks; direct output; four-band EQ; six aux. sends; two stereo returns on TRS jack; twin 12-segment bargraph monitors; 16 onboard FX programs with external editing software connected via MIDI; 100 mm faders; mono output fader; independent A-B output; Sys-Link 2 output option.



**PRICE:** \$1,199.

**CONTACT:** Allen & Heath at 866-474-7711 or [www.Allen-Heath.com/US](http://www.Allen-Heath.com/US).

## FAIRLIGHT DREAM

### Constellation-ANTHEM

**FEATURES:** Multiple configuration; up to 192 channels; six-band parametric EQ; Fairlight QDC processing engine; onboard 48 or 96-track recorder/DAW; 7.1 surround sound; onboard dynamics; OLED displays.



**PRICE:** starts at \$125,000.

**CONTACT:** Fairlight/MediaGear at 210-544-0540, [www.fairlightau.com](http://www.fairlightau.com).

# Mojave Audio

Brought to you by David Royer



### MA-200 large diaphragm tube condenser

### MA-100 small diaphragm tube condenser

*"How does such a small mic make such a big sound? This thing [MA-100] is fat!"*

Ross Hogarth (Grammy winning Producer/Engineer, Ziggy Marley, Jewel, Keb Mo, Black Crowes, REM)

*"Everyone at the session immediately noticed the presence and the natural sound the MA-200 delivered - particularly its clean, uncluttered high end."*

David Bryson - Counting Crows

*"The MA-200 did a fine job on a male vocal, imparting throaty warmth, a bit of transient crunch and an evenness that made the 'bars' in the room smile at the results."*

Mix - December 2006

Visit [mojaveaudio.com](http://mojaveaudio.com) for studio photos and audio samples.

818-847-0222  
Burbank, CA.

## LECTROSONICS Venue Receiver System

**FEATURES:** Dock for up to six UHF receiver modules; 258 UHF frequencies; antenna multicoupler; reception compatibility modes; Digital Hybrid Wireless technology; compander-free audio with 400 Series transmitters, compatibility modes for older analog; LecNet2 PC interface.

**PRICE:** starts at \$1,850.

**CONTACT:** Lectrosonics at 800-821-1121, [www.lectrosonics.com](http://www.lectrosonics.com).



## SABINE SWM7000 Series 2.4 GHz Wireless Microphone System

**FEATURES:** UHF system; True Diversity 2.4 GHz Smart Spectrum technology; 70 channels; FBX Feedback Extremator; parametric EQ; compressor/limiter; de-esser; Mic SuperModeling; ships with Audix OM3 capsule; handheld or lavalier/headset bodypack transmitters; rechargeable batteries; digital audio output.

**PRICE:** starts at \$1,259.99.

**CONTACT:** Sabine at 386-418-2000, [www.sabine.com](http://www.sabine.com).



## MIPRO ACT-707F Wireless Mainframe

**FEATURES:** 1RU rackspace accepting up to four ACT-707MC receiver modules; built-in four-channel antenna divider, mixer, switching power supply and monitoring headphone jack; front- or rear-mountable antennas; antenna extension accepted..

**PRICE:** \$1,155.

**CONTACT:** Mipro/Avlex at 877-447-9216, [www.avlex.com](http://www.avlex.com).



## ZAXCOM TRX900 Series

**FEATURES:** Digital Wireless system; time code stamped internal recording for back-up of transmitted audio; optional internal IFB receiver; graphic LCD display; stereo wireless transmission available.

**PRICE:** \$3,800 per channel.

**CONTACT:** Zaxcom at 973-835-5000, [www.zaxcom.com](http://www.zaxcom.com).



## PEAVEY Pro Comm PCX U-1002

**FEATURES:** 100-channel; true diversity; AutoScan technology resolving abrupt frequency changes; Channel Control System; LCD display; lightweight belt-pack transmitter; handheld, lavalier and headset configurations.

**PRICE:** \$799.99 - \$899.99.

**CONTACT:** Peavey Electronics at 866-443-2333, [www.peavey.com](http://www.peavey.com).



## AKG WMS 400 Pro Wireless Microphone System

**FEATURES:** Available in two 30-Mhz bands with 1,200 frequencies each; up to 12 simultaneous systems in each band; Integrated frequency management system with automatic scanning; infrared link for frequency/setup data uploads.

**PRICE:** Starts at \$479.

**CONTACT:** AKG Acoustics at 818-920-3212, [www.akgusa.com](http://www.akgusa.com).



## REVOLABS Solo Wireless Microphone Systems

**FEATURES:** UHF system; 1.9 GHz; 128-bit encryption; wideband audio frequency response; MaxFlex technology allows wearable, tabletop or XLR handheld adaptor-type Solo Microphones; up to 16 simultaneous microphones; single-, four- or eight-channel versions.

**PRICE:** Starts at \$249 per channel.

**CONTACT:** Revolabs at 978-897-5655 ext. 111, [www.revolabs.com](http://www.revolabs.com)



## SENNHEISER SKM 5200 Wireless Transmitter

**FEATURES:** For use with six Sennheiser and two Neumann interchangeable microphone heads; two channel banks, one fixed one variable; 20 pre-set frequencies and 20 programmable UHF frequencies in 5 kHz steps; sensitivity switchable in 1 dB steps; HiDyn plus noise reduction; backlit LC display for settings; automatic lock mode.

**PRICE:** \$2,115.



**CONTACT:** Sennheiser at 860-434-9190, [www.sennheiserusa.com](http://www.sennheiserusa.com).

## SONY UWP-C3 Wireless Microphone System

**FEATURES:** Plug-on transmitter converts wires microphone into wireless; operates in 24 MHz-wide band from 758 – 782 MHz; attenuator function; 50 mW RF power output; MIC/LINE switchable; space diversity reception tuner; RF squelch; headphone monitoring; backlit LCD; designed for Sony F-112 Dynamic mic.

**PRICE:** \$699.95.

**CONTACT:** Sony at 201-930-1000, [www.sony.com/proaudio](http://www.sony.com/proaudio).



## SHURE KSM9 Vocal Microphone

**FEATURES:** Dual gold-layered, low-mass Mylar diaphragm; switchable supercardioid/cardioid patterns; premium electronics; gold-plated connectors; champagne or charcoal grey; 50 Hz - 2- kHz response.

**PRICE:** \$850.

**CONTACT:** Shure Inc. at 847-866-2200, [www.shure.com](http://www.shure.com).



## AUDIO-TECHNICA Artist Elite 5000 Wireless Microphone System

**FEATURES:** UHF system; up to 200 channels; True Diversity reception; dual compander; Tone Lock squelch; Intelliscan; handheld, bodypack transmitter packages; dynamic and condenser handheld transmitters.

**PRICE:** Starts at \$3,119.

**CONTACT:** Audio-Technica at 330-686-2600, [www.audio-technica.com](http://www.audio-technica.com).



## CARVIN UX1000-MC

**FEATURES:** 960 user-selectable channels with diversity UHF receivers; four assignable groups; Diversity PLL Synthesized technology; two antennas; handheld microphone; low-battery indicator; mic on/off status indicator.

**PRICE:** \$429.95.

**CONTACT:** Carvin at 858-487-1600, [www.carvin.com](http://www.carvin.com).



# STUDIO | Review

AEA Continued From Page 30

makes it my preamp of choice for vocals. The TRP sounds nothing like either of those.

I'd place it instead in the "same sound universe" as the Millennia Media HV3C. The HV3C sounded a little bigger with some of my tube mics; the TRP sounded cleaner with others. But, in general, the two of them sounded similar. I'd say that TRP was more neutral, cleaner, airier if I had to put my finger on it, while the HV3C sounded a little bigger and bolder. And, since the Millennia Media preamp is such a standard around the world for classical recording, I'd consider that putting the TRP in the same category as the HV3C is giving it some pretty high-class company to keep.

For the ultimate vocal channel for my R84 —

cost being no object — I hooked the TRP into the line input in my Manley Voxbox, substituting for its own 40 dB gain tube preamp (not a good match for the R84).

This is now my dedicated voice system, as it's dead quiet, super clean, warm and — given the special EQ and dynamics adjustments of which the Manley is capable — that's a pretty hard to beat combination!

## | SUMMARY

AEA's "The Ribbon Pre" fills a much-needed niche in the world of mic preamps. Its sound fits right in with the best of the clean, neutral, uncolored solid state preamps and, with its 30 kohm input impedance and extremely high (and quiet) gain, is destined to become a first choice preamp for engineers who use ribbon

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

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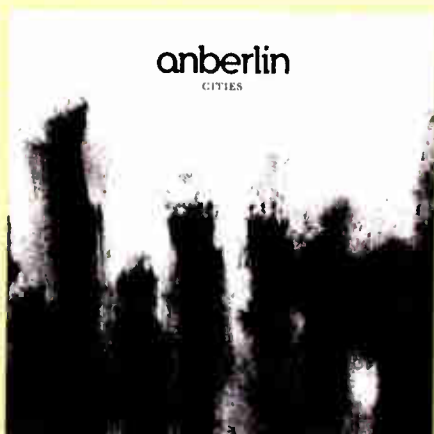
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## “Godspeed” | Anberlin



**SINGLE:** “Godspeed”

**ALBUM:** *Cities* (Tooth & Nail)

**DATES MIXED:** Fall 2006 at Paradise Sound near Princeville, Hawaii

**SINGLE PRODUCER:** Aaron Sprinkle

**SINGLE ENGINEERS:** Aaron Sprinkle, Randy Torres and Joseph Milligan

**MIX ENGINEER:** Mike Shipley

**ASSISTANT MIX ENGINEER:** Brian Wohlgemuth

**MASTERING:** Ted Jensen at Sterling Sound in New York City

**OTHER PROJECTS:** Shipley has worked alongside producers including Robert John “Mutt” Lange and Patrick Leonard, as well as artists the Cars, Foreigner, Def Leppard, Shania Twain, Green Day, the Black Crowes, Tom Petty, Devo and Cheap Trick.

**SINGLE SONGWRITERS:** Stephen Christian, Joseph Milligan, Deon Rexroat, Nathan Strayer and Nathan Young

**MIX CONSOLE:** Digidesign ICON

**MIX RECORDER:** Digidesign Pro Tools|HD

**MIX MONITORS:** KRK Expose E8

**SELECTED MIX OUTBOARD GEAR:** Crane Song STC8, Crane Song HEDD converter, Empirical Labs Distressor, Empirical Labs Fatso, Gates Sta-Level compressors, RCA BA-6A limiter, SPL Transient Designer, SSL XLogic G Series stereo compressor

**SELECTED MIX PLUG-INS:** Crane Song Phoenix, Massey CT4 compressor, URS Classic Console Strip

## ENGINEER'S DIARY

Last fall, supermixer Mike Shipley received a call from producer Aaron Sprinkle, who had just finished production on the latest from Anberlin, a quintet signed to Tooth & Nail Records. “I had worked previously with Aaron on half of a Jeremy Camp album,” recalls Shipley. “He asked me to mix *Cities* (Anberlin’s third full-length); I really like the band, so I jumped at the chance.”

Soon after, Shipley and mix assistant Brian Wohlgemuth packed up everything they needed from their Los Angeles digs for a flight to Kauai, Hawaii. Upon arrival, they would mix *Cities* and its first single, “Godspeed,” at a private studio Shipley helped build a few years ago, located on 10 acres of lush, unspoiled North Shore property. “We call it ‘Paradise Sound,’” Shipley explains. “I knew the control room very well and we needed a fresh breath of air. It broke the routine, put smiles on our faces, and as much as the project was hard work it was a lot of fun.”

Shipley’s Digidesign ICON, Pro Tools|HD rig and all necessary outboard processing was transported from the Kauai airport to Paradise in the back of a dusty minivan, a means of transportation that initially worried both Shipley and Wohlgemuth. “Here came the gear bouncing along this dirt road,” Shipley recalls. “A couple of local guys – who were as nice as pie – were driving all this tube gear on island time, bumbling along with the tailgate open. I was thinking the worst. But after we got the truck unloaded and Brian calmed me down – I was the one freaking out! – everything was okay. We were up and working the next day.”

“Godspeed,” a modern rock hit built to be larger-than-life, was mixed instinctually, explains Shipley: “It’s obviously a big song, so it needed subtle delays and flanging effects that didn’t pull away from the song itself. I tried to get a real depth of field, making it as non-linear-sounding as possible.”

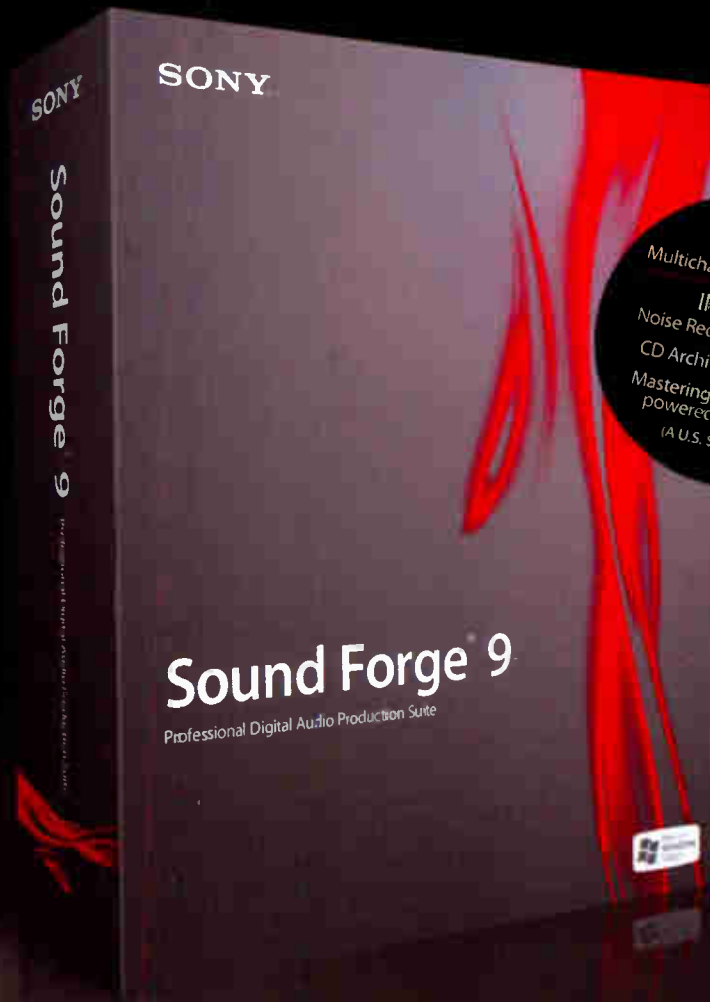
Shipley’s fly-to-Kauai gear consisted primarily of compressors. “I have effects in Pro Tools that I’m very fond of,” Shipley explains. “But nothing beats analog compression. I just love what the warmth of the Gates Sta-Level does for the bass and vocals.” The mixer also feels strongly about the quality of fave plug-ins such as the Massey CT4 compressor, URS Classic Console Strip, and the Crane Song Phoenix. “I believe that plug-ins like the Phoenix are indispensable in Pro Tools land.”



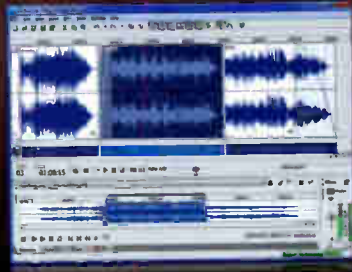
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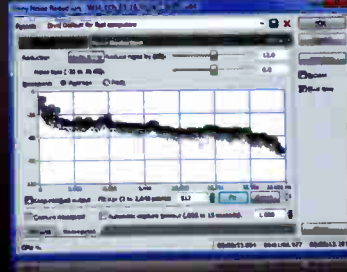
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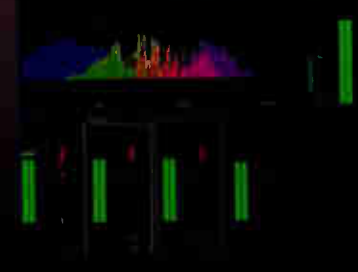
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