

ProAudio Review

DECEMBER 2008
VOLUME 14
ISSUE 12

The Review Resource for Sound Professionals

INSIDE:

- **UNIVERSAL AUDIO:** UAD-2 Solo DSP Engine
- **SINGLE SLICE:** Lucinda Williams' "Real Love"
- **IN USE:** Yamaha IM8 Series Mixing Console



"Perfect Solution"

BIAS Peak Pro XT 6 2-channel Mac Workhorse

A HOLIDAY TOUR
LIVE TELEVISION AUDIO
Tony Bennett and the Count Basie Orchestra

MASTERING MATTERS
UPGRADE ESCAPADES
Adapting To New Tools

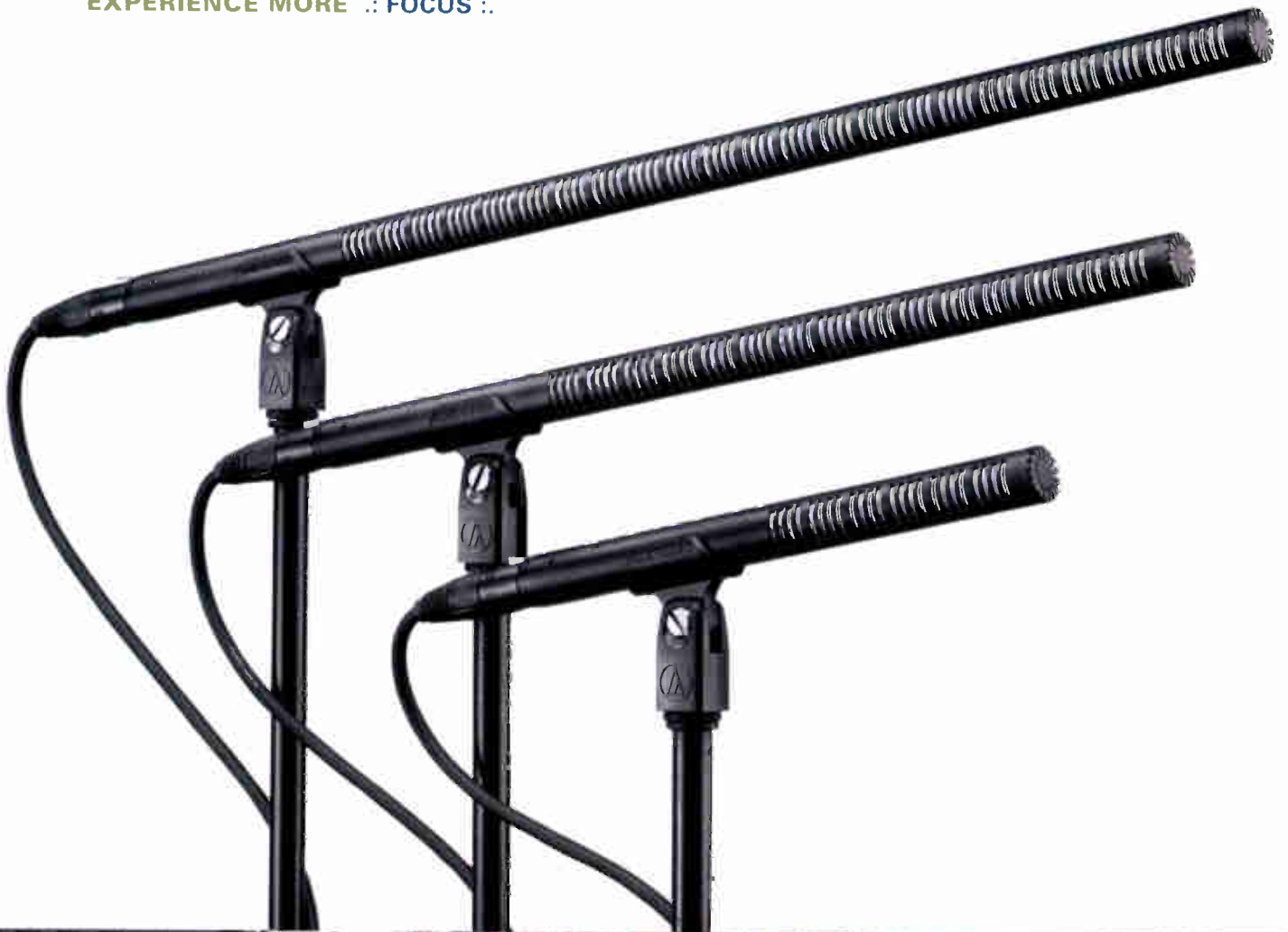
CONVERTERS & CLOCKS
WHAT DO YOU USE?
Pros' Preferences

reviews

NTI Digirator DR2 • RME ADI-8 QS • Sencore SoundPro SP495
Sound Devices 788T • Zaxcom Fusion

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Studio

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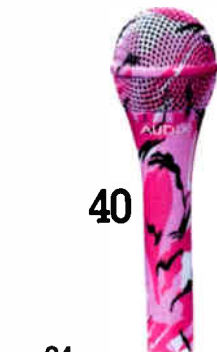
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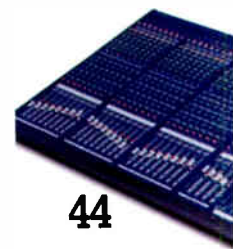
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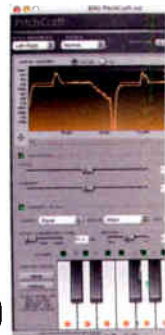
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Convenience And Performance Aren't Exclusionary Concepts



Portability is driving much of the technical innovation that shapes these early days of the 21st Century. On the consumer side of the electronics market, laptops are overtaking desktop computer sales. The iPod and its ilk have all but replaced the portable CD player and are taking the place of the portable DVD player for individual use. Car players that are iPod-friendly will further drive down the sale of wallet carriers for round shiny discs — it's just too convenient for the consumer to tote a single handheld device to replace a

bulky bundle of discs. That portable players and file-based delivery are hammering physical media sales in general is also noteworthy, if a discussion for another day.

Thus far, some sort of compromise has typically accompanied portability. This is not necessarily an operational compromise (the iPod again offering the example of a truly groundbreaking and intuitive human interface), though small devices with limited control and display elements can indeed present challenges for designers, not to mention limiting the physical I/O options. That these physical considerations exist should come as no surprise to anyone who, say, popped the top on a consumer component cassette or CD player of even a couple of decades ago — the insides being filled mainly with air and the case size determined by the need for front-panel metering, controls and the mechanical drive for the media and, to a lesser extent, by the rear-panel need for I/O. Increased reliance on remote controls accompanied a move to slim-line disc players, with even less real estate given over to front-panel control and display.

Performance compromises are those that concern us more as professionals. Consumers will not retreat from their desire for ever-more portable devices, despite the potential compromises that entails. Sure, you can put high-resolution audio and video on a portable device, but until we notch a step or two forward in portable data storage, the convenience factor of having more songs and videos at their fingertips will trump quality for most users, if they actually even know (or care) that they have such options.

Even if our definition of portable is a bit different than that of the consumer, portability in professional gear has benefited from the consumer revolution. Improved battery technology, high-performance/low-voltage and low-power consumption ICs, miniaturization of devices in mass storage while storage capacities climb, and innovations in display and control technologies are all tools that professional gear designers can use to develop portable devices with no performance compromises and truly staggering capabilities.

That truth became abundantly clear in the process of putting together this issue of *Pro Audio Review*, in this case being two devices each in the portable field recorder and portable test gear categories. Let the consumer make their compromises. While our definition of portability allows for devices significantly larger than an iPod, today's solutions can allow the audio pro to opt for convenience while actually raising their standards.

EDITORIAL

Frank Wells, Editorial Director
615-848-1769, fwells@nbmedia.com

Strother Bullins, Reviews And Features Editor
336-703-9932, strotherPAR@earthlink.net

Fred Goodman, Managing Editor
212-378-0423, fgoodman@nbmedia.com

Christopher Walsh, Associate Editor
212-378-0437, cwalsch@nbmedia.com

Lynn Fuston, Technical Editor

Rich Tozzoli, Software Editor

Bruce Bartlett, Dr. Frederick Bashour, Ty Ford, Carlos Garza, Will James, Tom Jung, Bascom H. King, Russ Long, Steve Murphy, Alex Dana, Randy Poole, Mike Rivers, David Rittenhouse, Richard Alan Salz, Alan Silverman, Rob Tavaglione, Dan Wothke, Tom Young
Contributors

ADVERTISING

Tara Preston, Associate Publisher
917-331-8904, tpreston@nbmedia.com
Karen Godgart, Sales Director, West Coast Office
323-868-5416, kgodgart@nbmedia.com
Matt Rubenstein, East Coast Sales
914-524-5045, mrubenstein@nbmedia.com
Zahra Majma, Specialty Sales Manager
212-378-0433, zmajma@nbmedia.com

ART & PRODUCTION

Annmarie LaScala, Art Director
Fred Vega, Production Manager
212-378-0445, fvega@nbmedia.com

CIRCULATION

Anne Drobish, Associate Circulation Director,
Audience Development
Michele Fonville, Circulation Coordinator
Subscriptions: Pro Audio Review, www.MyPARmag.com
P.O. Box 234, Lowell, Ma 01853
Tel: 888-266-5828
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E-Mail: Newbagg@Computerfulfillment.com

NEWBAY MEDIA AUDIO GROUP

John Pledger, Vice President/Group Publishing Director
Anthony Savona, Editorial/Creative Director
Robert Granger, Online Production Manager
David Leach, Web Analyst
Ashley Vermillion, Web Production Specialist
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Jack Liedke, Controller
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Administrative, Advertising, & Editorial Offices
B10 Seventh Ave., 27th floor
New York, NY 10019
TEL: {212} 378-0400
FAX: {212} 378-0435
E-MAIL: pro@nbmedia.com

A photograph of a musician performing on stage. The musician is in silhouette, wearing a dark shirt and a cap, looking down. A Shure microphone is mounted on a stand in the foreground, illuminated by a spotlight. The background is dark with some stage equipment and lights visible.

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Lexicon I-ONIX Series

Lexicon has just announced its new I-ONIX U22, U42S, and U82S USB desktop recording interfaces. Each I-ONIX Series model provides the following: USB 2.0 desktop recording interface, Steinberg Cubase LE4 multitrack PC and Mac recording software, Toontrack EZdrummer Lite virtual instrument software, and the new Lexicon Pantheon II VST/AU reverb plug-in. Each interface features newly designed dbx microphone preamps, 24-bit/96 kHz converters, and much more.

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Shadow Hills Mastering Compressor



Exclusively distributed by Vintage King Audio, the Shadow Hills Mastering Compressor is a truly high-end and vintage-style compressor/limiter for mastering-grade audio production, tracking, and mixing. Operational as stereo or dual mono, the Mastering Compressor offers two compression units in series—electro-optical and discrete Class A, respectively. Both channels feed switchable custom output transformers of nickel, iron, and steel. Features on the Shadow Hills Mastering Compressor abound; visit Vintage King Audio's website for more information.

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Left: The SP495 offers input via balanced Neutrik combo connectors. Below: On top, the SP495 offers digital I/O, analog output options, and a handy reference speaker.



consonant loss based on RT60 and level measurement), RaSTI and STI-PA analysis [these last two tests are based on the system resolution of bandwidth limited pink noise modulated by sine waves. The test signals are provided on CD]. Various standard noise curves can be plotted across a user's SPL vs. frequency test data with the Noise Curves firmware module. The Time Delay Analysis firmware graphs frequency response independent of room reflective energy [and includes "3D" waterfall graphs]. This function can be handy in helping determine whether poor room acoustics are negatively impacting a sound reproduction system that is, in and of itself, performing as conceived. RT60 measurements can be enhanced with the Multi-Band Decay Analysis module, computing decay time simultaneously across seven individual octave bands for further identifying problem areas. The final firmware module uses the mic input with the headphone outputs to form an Audio Stethoscope for gremlin chasing.

The SP495 can be interfaced with Mac and Windows computers via RS232. I used an USB to RS232 serial port adapter to successfully interface the SP495 with both a MacBook and a Windows XP laptop. For Windows, the TerraLink software module provides a "follow-me" type of functionality for test functions, and, with the SP495 in a PC-link mode, recall and display of stored test data. For Mac, there's a SoundPro module for similar computer display. There's also an optional Windows report software module for link to a customized Excel spreadsheet.

In addition to using the SP495 in the sound reinforcement environment in Ben's church install, I used the SP495 for home theater evaluation. While the SP495 is equipped with the measurement capability for level and delay settings, and can be informative about room reflections and decay with its extended

functionality, automated home theater testing for up to 6.1 systems [DTS and Dolby test signal generation] is facilitated by pairing the SP495 with Sencore's DAG5161 SurroundPro Digital Audio Generator [not tested]. The SD card interface can be used for transfer to computer of impulse response, and the SP495 can also function as a stereo recorder, with the SD card used to "sneaker net" such data [as well as RTA, FFT and other data stores].

The SP495 is not inexpensive. Many FOH engineers may opt for a laptop and one of the available software sound

analysis packages as a less expensive and more function specific choice for their needs. For the sound consultant set, needing an evaluation tool that provides data more traceable to relevant standards, and who is not simply dealing with the venue du jour but evaluating the space itself as well as sound system performance, the Sencore SoundPro SP495 Audio Consultant competently provides a comprehensive range of tools in a single portable package. For that class of user, simply reading through the feature set is likely to inspire gear lust.

RME

(continued from page 34)

channel. Meters can be set to peak hold or auto reset after three seconds. Red peak flashers illuminate at -2 dBFS.

To set up, since I use the ADI-8 DS for A/D conversion in front of my Digi 003R, all I had to do was move the TRS plugs and Lightpipe from DS inputs to QS inputs. All of my audio projects end up on either CD or DVD. As such, 44.1 kHz or 48 kHz/24-bit files work for my needs. I compared 44.1 kHz/24-bit recordings made with the same mic, preamp, and delicate finger-style acoustic guitar through both the DS and QS converters and found their sound was virtually identical.

The RME analog and digital gain-reduction circuits do noticeably squeeze the audio. I don't think I'd use them for tracking unless very lightly. I think

these features are better suited for sessions when ultimate density and loudness is the goal.

The translated-from-German manual is sometimes paradoxical. The vocabulary is frequently above that of standard American English, but sometimes the lack of understanding of our syntax and grammar are apparent and obscure the content. One more pass through an editor for whom English is not a second language would make the manual as spot-on as the device itself.

Summary

The RME ADI-8 QS is a hard-working, good-sounding, flexible box that can extend the capabilities of any studio at any level and even grow with it. Keeping the MADI card as an option is a very good idea; it keeps the unit's price down for those who will never need it.

Engineered Excellence Made in Austria

Martin Strayer

Sound engineer Joe Cocker, Dixie Chicks



"I've been doing vocals and acoustic guitar with it here at my studio and it's fantastic! I also love this mic on our electric guitar too! It is so true, clear, and beefy at the same time... it just reproduces the sound coming out of the speaker so well, so crystal clear and punchy. Truly a great addition to AKG's microphone line."

Sound & Recording _ 08/2008 (Germany)

"... utilizes many of the C 414's technologies... rugged construction... a budget-friendly alternative to the C 414."

AudioMIDI _ 07/2008 (USA)

"WOW! What a great microphone... shares the same DNA with the C 414... many qualities of the C 12 VR."

Media Biz _ 07/2008 (Austria)

"... clear, transparent sound... excellent transient response... an outstanding mic."

Keys _ 08/2008 (Germany)

"... very rugged... clear and open sound... an all-rounder which doesn't depreciate... good value for money."



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World Radio History

Upgrade Escapades

A few too many decades ago I became a recording addict. My first real gear purchase was a brass-capsule AKG C 414 EB pair, fresh from behind the counter of a mom-and-pop pro-audio store with the instruction, "Pay us when you can." (I did.)

I bee-lined to a friend's attic studio where we recorded a primo Ludwig drum kit with the new AKGs and his Neumann KM84 pair. We recorded to a Scully 4-track, spending hours experimenting with the mics, listening to playbacks and having a whole lot of fun learning what they could do.

Those days are some of my best memories, but a great thing about recording (especially the analog kind) is that the exploration and the fun continue for as long as you stay in it. There's always something that can re-spark the sense of discovery or bring a fresh perspective. Each acquisition promises new sounds, better results, and anticipated (but unobtainable) perfection.

Serious Upgrades

However, in a mastering context, upgrading — enticing as it may be — is deadly serious and even dangerous, especially when the monitor path is involved. Mastering consoles, converters, amplifiers, and speakers all have a major impact on mastered results and any substitution upsets the bedrock foundation for a mastering engineer: the knowledge and understanding of the environment. Of course, if you've been getting excellent results with your present rig, you don't really need anything new, right? My friends continue to remind me of how many times, after an acquisition, I've said, "I'm all set! I don't need anything else new at the studio."

The Great Normalizer

Upgrading has taught me something about how we hear. The ear is a great normalizer. A neurophysicist friend once explained that new connections are made in the brain over a period of time when we make a significant change in the daily music playback system. What sounds radically different at first soon becomes normal. Whenever I've made the decision to upgrade the monitor chain in my mastering room, I've allowed up to a week of downtime for the adjust-



Is the McIntosh MC252 Alan's last new piece of gear at his studio? [We doubt it. - Ed.]

ment to take hold and to get comfortable with the change. A lot of time needs to be spent listening to familiar reference material and to past work.

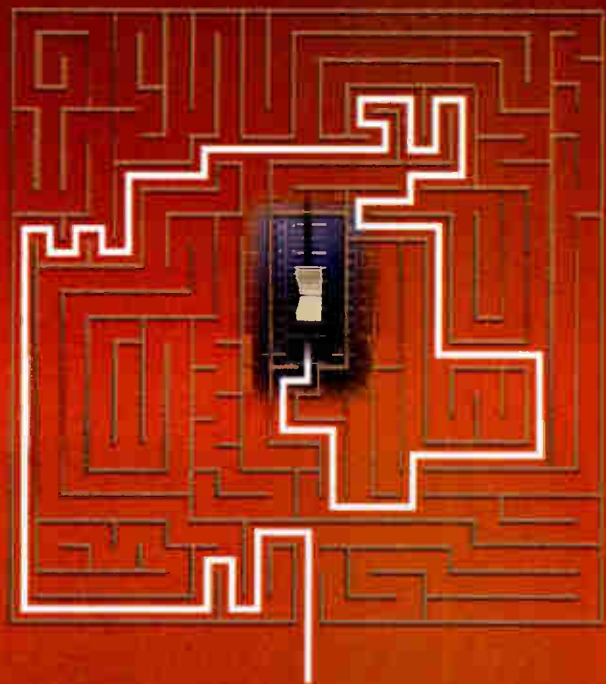
A recent upgrade involved bringing in a McIntosh power amplifier to replace the amp I had been working with for the last eight years. An opportunity presented itself to pick up one from an engineer friend at a good price. I'd always loved McIntosh amps, and we had scores of them at New York's A&R Recording Studios where I was on staff. They could handle any monitor well, from UREI 813 to Westlake BBSM to Yamaha NS10. I waited for the opportunity of a few clear days and finally did the installation.

On first listen, it seemed obvious it just would not work out. The sound was so completely unfamiliar it was scary. There seemed to be too much of everything. I did a few hours listening in the hopes that things would settle in, but no luck. I shut down the room for the night and took off.

The next morning I had high hopes for a fresh-eared listen, but the sound was still very disorienting. Fortunately, there was a project in for a friend who did not mind being part of the transition. After a difficult day, I took the results home—not bad, but not right. This sort of thing continued for two more days, each spent partly working on new material and the rest listening to references and old work.

The fourth day must have been the charm. At that point, for the first time, the system did not seem at all foreign at the morning's first listen. References and past work sounded good; in fact, they sounded better than I could recall, even though just the morning before things were off. As a reality check, I hooked up the old power amp; it seemed practically unusable, but this was the amp that had carried me through eight years of successful work! Evidently, the auditory transition was complete, and I was past the point of no return. This was an illustration of another madness that we've all experienced: It's all good 'til you hear the better device, and then there's no going back.

The amplifier upgrade is now a few months behind me, and I'm finding that I am working more smoothly. My job has become easier. The sound in the room is the best it has been. It's a great feeling, and even greater knowing that I'm all set! I don't need anything else new at the studio.



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BIAS Peak Pro XT 6

The latest version of the “perfect” Mac-based stereo editing and processing solution is an entirely different beast among standard industry DAWs.



The easiest way to define Peak Pro XT 6 is “an advanced sample editor that can edit individual audio files from a hard drive with single sample precision.” It is the perfect stereo editing and processing solution for engineers, musicians, and sound designers. While some of its features are the same as those found in all DAWs, it is an entirely different beast.

Peak Pro XT 6 is the latest flagship BIAS product. It includes Peak Pro 6, the SoundSoap 2 and SoundSoap Pro restoration plug-ins, the Master Perfection Suite, DDP 2.0 Export with CD TEXT addendum and the Peak Pro Production Pack.

Features

The broad BIAS Peak Pro XT 6 feature set includes digital signal processing (DSP) tools, dithering for distortion-free bit reduction, batch processing, support for

24-bit/96 kHz and 192 kHz, optimization for multiprocessor/multicore computers, and built-in support for Mac I/O and Core Audio hardware (third-party hardware may require compatible Core Audio drivers).

To operate effectively, Peak Pro XT 6 requires a G4, G5, or Intel-based Mac (with a 400 MHz or faster processor), Mac OS X v. 10.4.3 or higher (including Leopard), a screen with minimum 1024 x 768 resolution, 256 MB RAM (512 MB or

more is recommended), 330 MB available free disk space, 18 ms hard drive (average seek time) or faster, and QuickTime 6.0 or later.

The release of BIAS Peak Pro XT 6 adds a host of new features to this Macintosh-based, industry-standard, stereo audio editing, processing, and mastering application. Not only for mastering, the software suite is an obvious solution to everything from sound design for film, video, and multimedia, to music production and broadcast editing. The package touts a variety of new features like mastering and “classical editing” cross-fade modes, new controls like pre/post roll audition, the ability to insert new event at play-head, voiceover ducking DSP, cache in RAM, a new user interface and improvements to the delivery tools, and the inclusion of the

Russ Long, a Nashville-based producer/engineer, owns The Carport recording studio. He is a regular contributor to *Pro Audio Review*. Visit him online at www.russlong.ws.

“Not only for mastering, the BIAS Peak Pro XT 6 software suite is an obvious solution to everything from sound design for film, video, and multimedia, to music production and broadcast editing.”

Peak Pro Production Pack, a collection of analysis, equalization, audio cleaning tools, sound effects and more (separately valued at \$300).

Peak Pro XT 6 (\$1,199 list) offers a complete set of audio tools, which includes six powerful VST/AU plug-ins. There are also Peak 6 LE (\$129) and Peak 6 Pro (\$599), which have fewer features and components but may be adequate for some users. Peak Pro also includes the Peak Pro Production Pack DVD. Peak LE includes the Peak LE Production Pack DVD, which is identical to the Peak Pro Production Pack with three exceptions: It doesn't include SoundSoap LE or Reveal LE and the free Broadjam subscription is limited to six months.

Peak is designed to work with mono or stereo audio files. It serves as a complement to DAWs such as Pro Tools and Logic, and is the perfect tool to make a finished mix production-ready. Its robust and easy-to-use batch-processing feature is great for managing lots of audio files. When used with the RMS Normalization feature that matches average dynamic levels, the batch-processing feature is ideal for podcasters, as it allows large groups of audio files to be easily prepped for radio or podcast programs. Podcasts can be authored and uploaded directly from Peak as RSS feeds without requiring a separate Podcasting app.

Peak's new user interface includes global contrast control, scroll bars, and buttons. The new high-resolution logarithmic meters are user-configurable, and the program's windows are now "magnetic," automatically snapping together for optimal screen management. The new VBox effects and instruments matrix offers additional preset viewing and selection controls plus MIDI channel assignments. Additionally, the playlist offers an improved cross-fade control area along with a new stereo view and user-



BIAS Repli-Q, a EQ-matching application

regions, fades, and cross-fades. The custom default x-fades allows custom cross-fades to be created, saved and automatically applied to favorite x-fade shapes. Added controls include edit sync between source document and region. BIAS has also added several playlist waveform zooming tools (zoom to sample level, auto-zoom to cross-fade, zoom out all the way) to improve workflow. Ideal for editing complex musical takes, preparing tracks for commercial CD replication, or simply arranging sections of a remix, the new playlist in Peak Pro 6 is now even more powerful and optimized for the task.

adjustable waveform to list view ratio.

“Advanced Playlist Editing and Mastering” features in Version 6 add multiple new features to Version 5's feature set (which included PQ editing, CD-TEXT, DDP export option, and ISRC codes). Peak Pro 6 includes selectable Mastering or “Classical Editing” cross-fade modes. Volume envelopes allow volumes to be easily and precisely edited across entire

Peak Pro 6 has added several new tools to its Editing, Processing and Sound Design feature set. Perpetual Looper DSP ensures perfect “beat-free” sustained loops on monophonic sources. It is great for creating instrument sample libraries.

THE PERFECT MIX

hear back

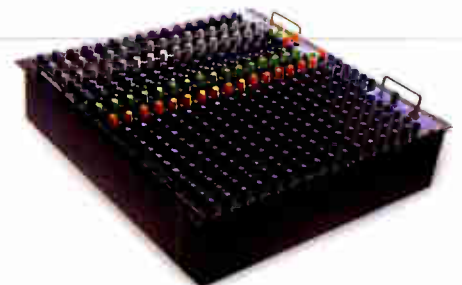
- Unlimited system size
- Local control of eight channels (stereo mix and six “more-me’s”)
- ADAT, analog, HearBus inputs are switch-selectable from front panel
- Built-in DSP Limiter with threshold adjustment



- Master Volume
- Stereo AUX In
- Balanced Line-Outputs
- Built-in Mic Stand Mount
- Standard CAT5e power

mix back

- Affordable, flexible 16 x 12 x 2 x 2 monitor mixer
- 16 Mic/Line inputs each with passive split and two stereo and twelve mono level controls
- Pin 1 lift and Mic power switches per channel
- Dual switch-selectable effects sends
- Stereo Aux inputs
- Channel Inserts
- Four-band EQ with dual sweepable mids
- Talkback with intercom and optional remote
- 16 master outputs each available as analog, ADAT, and HearBus



The Voiceover Ducking DSP analyzes the amplitude of a voiceover and automatically adjusts volume of the background audio. This feature is ideal for podcast authoring when a voiceover needs to be mixed with background music. The Change Pitch and Convolve Envelope feature applies dynamic changes over time. Pitch Envelope allows pitch changes to be controlled with a user-customizable envelope. It is perfect for creating special pitch-based effects over time. Convolve Envelope offers dynamic control over the amount of convolution being applied over time and is excellent for creating new sounds by mapping the harmonic spectrum of one file over another.

The Change Duration options allow algorithmic selection for optimizing results with vocals or musical content. The Vbox Link allows a cross-synthesis between virtual instruments, audio, effects, and live input with real-time mix, convolve, vocode, modulate and magnitude multiply modes. The proprietary Dither Cloning Audio Technology (DCAT) provides ultra-accurate emulation of the most popular dithering technologies as well as parameter controls to adjust, pre-

view, and apply frequency and attenuation skew. As in previous editions, POW-dithering is included. Cache in RAM provides a fast editing option that stores audio in RAM. This avoids disc-access holdups when working with large files and drastically increases processing speed for many DSP operations.

Peak Pro is frequently used for CD, web, video, and gaming audio file preparation. Version 6 adds new export capabilities while streamlining integration between other applications. Advanced PQ sheet/text export now includes custom playlist reports with durations, index points, CD-TEXT info, notes, etc. This data can be printed, saved to PDF, or exported as a tab-delimited file. Audio documents and playlists can be quickly exported directly to iTunes, and files can be dragged and dropped between iTunes and Peak playlists. The "Send To iTunes" option also allows files — or entire playlists — to be encoded using iTunes automatically, if desired.

The Default Save As feature stores frequently used file type, bit depth, and compression settings in the Save As dialog. Peak can read and write to all major

file types (AIFF, SDII, QuickTime, MP2, MP3, MP4, FLAC, WAVE, Broadcast WAVE, AU, Sonic AIFF, JAM Image, Raw, DDP, and more). It also supports the reading, writing, and even editing of metadata (audio information stored within the file such as artist, title, and SMPTE start time) for the most major of file types — MP3, FLAC, AIFF, WAVE, and Broadcast WAVE.

The Peak Pro Production Pack is a powerful assembly of audio plug-ins, utilities, and content. It includes BIAS Reveal LE, BIAS SoundSoap LE, WireTap Pro, a 1-year membership to Broadjam.com, AMG's "ONE" sample player (content-limited edition), and tons of content from PowerFX, Hollywood Edge, and Sound Ideas.

The LE editions of SoundSoap and Reveal integrate exclusively with Peak on Mac OS X, while the retail editions are cross-platform and compatible with third-party AU, RTAS/AS, and VST hosts. WireTap Pro from Ambrosia is an excellent application for capturing system/Internet audio with no loss of quality or additional cable connections. AMG's "ONE" sample player is a real-time sample playback instrument featuring a limited selection of quality loop content.

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BIAS SoundSoap Pro, a restoration plug-in for Peak Pro

SoundSoap 2 effectively removes broadband noise, 50 and 60 Hz hum, and rumble. I've found the plug-in to be intuitive and easy to use. It can be used as an Audio Unit, RTAS, or VST plug-in, or

as a standalone application with almost any digital audio or video file. SoundSoap Pro combines three reduction and restoration tools: Broadband, Click and Crackle, and Hum and Rumble, plus a sophisticated Noise Gate into a single plug-in. It includes an intelligent noise-learning algorithm that works miracles on noise-laden tracks. The parameter controls include Click and Crackle removal tools, a global spectrogram and several other visual tools that assist in the audio restoration process. The program works perfectly to remove unwanted room noise, hiss, electrical hum, rumble, and other background noise.

The BIAS Master Perfection Suite includes the PitchCraft, GateEx, Reveal, Sqweez, SuperFreq and Repli-Q plug-ins. PitchCraft is a natural-sounding pitch-correction plug-in that combines an X/Y Tuning History Graph, Keyboard Display and several sliders to control format and pitch changes. The GateEx gate/expander plug-in is a powerful gate in a comfortable and easy-to-use interface. The Reveal plug-in is a seven-tool audio signal analysis suite that provides engineers an intelligent way to analyze audio. It includes an oscilloscope, spectral analyzer, spectrogram, peak and RMS

Fast Facts

Applications

Studio, project studio, broadcast, post production

Key Features

The latest version of Peak Pro 6, SoundSoap 2 and SoundSoap Pro restoration plug-ins, Master Perfection Suite, DDP 2.0 Export with CD-TEXT addendum, and the Peak Pro Production Pack DVD.

Review Setup

Apple Macintosh 2 GHz
Dual Processor G5 w/2 GB RAM
Apple 2.33 GHz Intel Core 2
Duo Macbook Pro w/2 GB RAM
Lynx Aurora Converters
Lucid Gen-X-96 Clock
PMC AML-1 monitors
Focal Twin6 Be monitors
Audio-Technica ATH-M40fs Headphones

Price

\$1,199, \$599, and \$129 (Pro XT 6, Pro 6, and 6 LE, respectively)

Contact

Bias, Inc. | 707-782-1865 | www.bias-inc.com

Product Points



Relatively simple and very intuitive interface

Sonically better sample rate conversion compared to other DAWs



No support for surround sound, DVD-A, SACD, or Blu-Ray

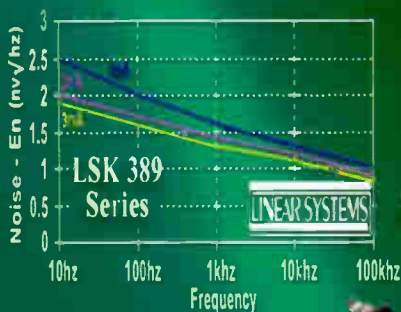
No PC support

Score

As far as working with stereo audio files goes, Peak Pro XT 6 is the best resource that I've encountered.

level history display, peak and RMS level meters, Lissajous phase scope and pan power history display. This plug-in is extremely useful in determining how certain processing is affecting the audio content while mixing or mastering. The Repli-Q plug-in is an EQ-matching application designed to analyze, edit, and compare the spectral content of audio recordings. Once analysis has taken place, the spectral characteristics of one recording may be applied to another. The Sqweez-3 and Sqweez-5 are multi-band compressor plug-ins. In both versions, each band of compression provides gain and threshold sliders as well as five band-specific pop-up controls [attack, release, knee, ratio, and max reduction]. The SuperFreq EQ plug-ins are 4-, 6-, 8- or 10-band equalizers.

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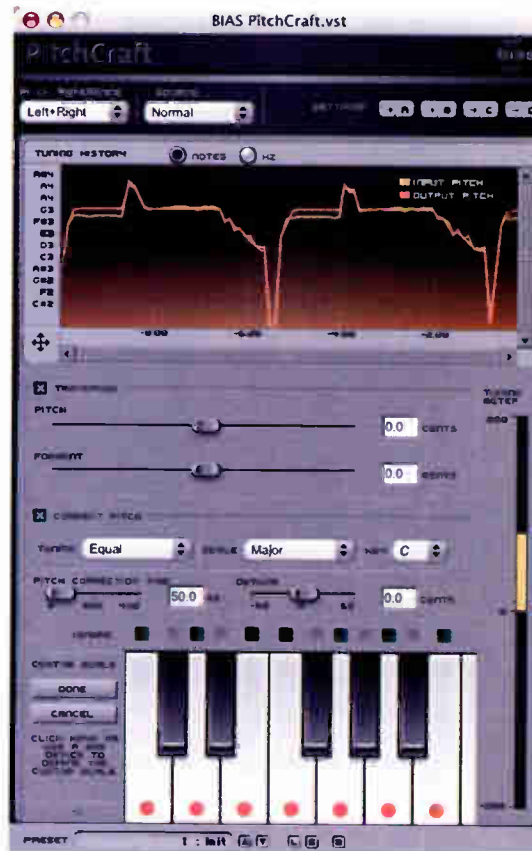
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Each band has adjustable bandwidth, gain, frequency, and an individual band bypass button.

In Use

Installing the Peak Pro XT 6 software suite was a breeze, and I'm happy to report that the package no longer requires a USB key to operate. BIAS has redesigned the entire look of the graphic interface, and it is visually better than ever. The drab gray look of the previous version has been replaced by a higher contrast, more invigorating GUI. The increased customization to be helpful too, though I'm surprised that there's still no support for storing custom window sets. As with previous versions, Peak's interface is relatively simple and intuitive. First-time users will be up and running in a minimal amount of time.

BIAS has redesigned the metering in Peak and the new high-resolution, logarithmic-scale meters offer user-definable configurations and the ability to meter audio before or after the master output slider. That said, I occasionally miss the old metering which was more accurate in some situa-



BIAS PitchCraft, a natural-sounding pitch correction plug-in

tions. An option for "Peak Legacy" metering would be nice.

Since sample rate conversion is still far from perfect in any software package, I think it is a good measure of the quality of the program and how it handles audio. Peak 6's USRC [Ultimate Sample Rate Conversion] sample rate conversion is quite impressive. I listened to several different files that I converted with Peak and found that in most cases the sonic difference was so small that it was almost undetectable and the results were always usable. It's sonically better than the sample rate conversion that I've ever used in any DAW.

The new cache in RAM feature makes for some mighty fast editing without the need to constantly load audio files from the drive. This is a great addition to the packages feature set. I also found the addition of a real-time bounce

feature to be extremely helpful when I was using analog gear to process an audio file or when I wanted to manipulate plug-in controls in real time as the bounce happened. As a user of Frontier Design's Tranzport wireless remote controller, I was happy to see that this device is now supported by BIAS with no additional configuration.

My overall impression with Peak Pro XT 6 is quite good. The package includes a substantial amount of content for a reasonable price. As expected with a package this extensive, there are several things that I found disappointing. There's no support for surround sound, DVD-A, SACD, or Blu-Ray. There's no PC support, either. [However, "the included plug-ins are cross-platform," offers the manufacturer. — Ed.] It would be helpful if it were possible to mix regions with different bit depths or combine mono and stereo documents within a playlist.

Summary

For working with stereo audio files, Peak Pro XT 6 is the best resource that I've encountered. New features include mastering and "classical editing" cross-fade modes, controls like pre/post roll audition, the ability to insert new event at play-head, voiceover ducking DSP, new user interface and improvements to the delivery tools. With the inclusion of the Peak Pro Production Pack, a collection of analysis, equalization, audio cleaning tools, and sound effects and BIAS Peak Pro XT 6 is a bargain.

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Zaxcom Fusion ENG Mixer and Multitrack Recorder

It's in the bag — Zaxcom's portable mixer, IFB, and multitrack machine



Technological innovation in the visual production circuit has been fast and furious in the last several years. As exemplified in the blurring — if not erased — line between film and HD video acquisition, the production scene has become a welcoming adopter of an across-the-board range of new technologies including a bevy of high-tech, on-the-set audio tools.

So ... has the ENG market kept pace with its production brethren? As Jon Stewart says, "Not so much." In a land oft ruled by fossilized dino-mixers that can be carbon-dated back to the early 1980s, evolution in audio tools (and attitudes) moves at a glacial pace. It is this general market — reality shows, news-gathering, independents, and other field-production work — that Zaxcom ambitiously targets with its Fusion (\$7,995) production mixer/multitrack flash recorder.

Features

The Fusion is the latest in Zaxcom's successful Deva multitrack hard-disk recorder line. As its name implies, the Fusion combines a number of mixing, recording, and cue functions into a fairly lightweight, highly portable package. While the other Deva units are more commonly found on audio carts in full productions, the Fusion's mirrored Compact Flash recording system eliminates the weight, moving parts, and power consumption of hard disks, posi-

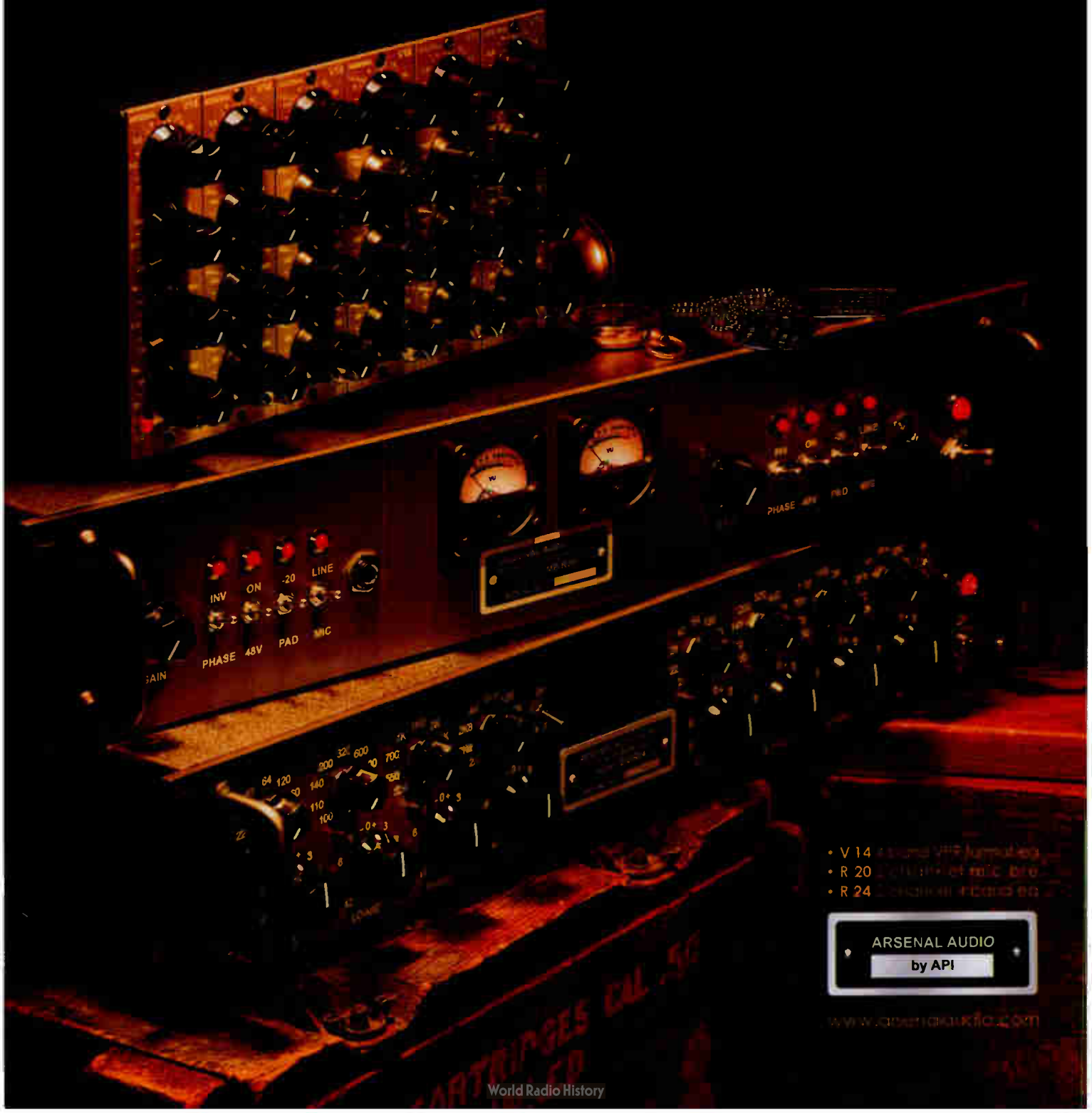
tioning it well for typical PortaBrace case run-and-gun work.

The Fusion measures 10.6 x 7.7 x 3.2 inches and weighs in at five pounds (without battery). All controls and indicators are found on its well laid-out front (or, when in a bag, top) panel, the only exception being its power switch, which is nicely tucked away on the left panel. Front-panel controls include generously large transport buttons, a number/cursor-control pad, and a set of eight dual-layer function buttons — all brightly back-lighted. Also on the front is a set of eight user-assignable rotary faders, a built-in mic for use with the dedicated slate button, and the all-important 3 x 2.25-inch full-color touch screen that serves as the main user interface for settings and metering.

Without venturing into overstatement, the Fusion has an immense

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Fast Facts

Application

Production audio, ENG, surround recording

Key Features

10 record tracks; 8 mic-/line-selectable analog inputs (on XLR) and 8 digital inputs configured in four AES pairs; 48V phantom power, limiter and HPF on each analog input; 6 balanced analog outputs, 8 digital outputs in AES pairs plus two outs to 10-pin Hirose; configurable Output, Disk, and HP mix matrices; 16-/24-bit record at 44100, 47952, 48000, 48048, 88200, 96000, 96096, and 192 kHz; 12 - 18VDC 4-pin XLR or NP-1 powered.

Price

\$7,995

Contact

Zaxcom | 973.835.5000 | www.zaxcom.com

amount of mixing and routing power under the hood, which we'll look at more in-depth in the In Use section below. To summarize here, the Fusion features 16 inputs, eight output busses, 16 outputs and up to 10 recording tracks.

Its 16 inputs are comprised of eight mic-/line-selectable analog inputs (on XLR) that utilize the same low-noise transformerless mic preamps found on the Deva 16, and eight dig-

Product Points



▶ Near-infinite mix, recorder, and cue matrix-based routing possibilities.



▶ Could be overkill for some ENG users, both in price and feature set

Score Although costly, the Fusion truly has no equal in the ENG products marketplace.

ital inputs configured in four AES pairs (on DB-15). Each of the analog inputs features selectable 48-volt phantom power, an analog limiter and an adjustable (30 to 240 Hz) high-pass filter.

The Fusion's 16 addressable outputs are comprised of six balanced analog outputs (on DB-25), eight digital outputs in AES pairs (DB-15), plus two analog sends to a camera via a 10-pin Hirose camera cable connector that also provides a mono-summed return from the camera's headphone amp. There is also a headphone amplifier with standard 1/4-inch stereo output derived from user-configurable presets.


The Fusion records 16- or 24-bit audio at sample rates of 44100, 47952, 48000, 48048, 88200, 96000, 96096, and 192 kHz, and is capable of on-the-fly sample rate conversion of incoming digital sources up to 192 kHz. Power is supplied to the unit via a standard 4-pin XLR that accepts between 12 and 18VDC (100-240VAC transformer included); alternatively, a standard rechargeable NP-1 battery can be inserted into the internal compartment to power the unit when on the go.

In Use

Let me state right off the bat that the Zaxcom Fusion is one fabulous, nearly all-powerful marvel of portable mix-and-record engineering. The wisdom and experience of 12 years of Deva development Zaxcom has imparted to the Fusion is undeniable, from its highly tactile hardware faders to its touch screen interface and near-infinite mix, recorder, and cue matrix-based routing possibilities.

It is a complex beast, to be sure, but that is because the Fusion leaves nary a stone unturned when it comes to the potential production demands it is capable of meeting. Despite the wealth of features, I found most of what I needed to be up and running mixing four sources (three Sony 88s and a Sennheiser boom) to two camera tracks while simultaneously routing the mics to four discrete internal tracks and dialing up a live-L, camera return-R phones mix without cracking the manual.

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“It is a complex beast, to be sure, but that is because the Fusion leaves nary a stone unturned when it comes to the potential production demands it is capable of meeting.”

Setting up the Fusion's user presets for my typical output mix, disk mix (record) and cue mix routing requirements was the obvious next step.

Truth be known, I already had a little experience on a Deva V years ago — and pretty good knowledge of pre/post crosspoint matrix routing and processing software (Lectrosonics DM, in

recorder in stereo and output two different cue mixes, in the context of the Fusion's total capabilities, I was still in meat-and-potatoes land.

To illustrate, in my work I had no need to use the Fusion's multiple digital input SRCs, configure custom metering screens, craft-embedded scene/take/notes metadata (beyond



my case) didn't hurt either — but I mostly “used the Force” and was quite happy to find the menu system, data entry, preset storage, and navigation to be thoroughly intuitive.

I also put the Fusion to work on a variety of ENG and production gigs including a live taping of the PBS television show, *The Kalb Report*, and a live forum called, “Battle for the Youth Vote 2008,” presented at George Washington University in partnership with CBS News. On both of the above, I used the Fusion's discrete tracks to later post CBS and XM radio versions. I also tracked all sources during a Billy Joel performance at the National Press Club just for the sport of it, since it was broadcast live.

The Zaxcom preamps are very high-quality, with very little noise and a good, if not forgiving, amount of headroom. Like most, I tend not to print with processing — especially given that I'm often the one to post mix — so I didn't much explore the EQ and compression available on the review unit, but I did make good use of the adjustable HP filters and limiters on inputs as well as the bus output limiter. I must mention the phenomenal job Zaxcom did with the Fusion's shielding and balanced circuitry. In one instance, a proximate Blackberry sent interference straight to my phones, yet not a blip was registered in the output mix feeds or on the recorded tracks.

Despite using the Fusion to simultaneously record multiple sources to discrete tracks (mirrored between its internal CF card and an external Avastor hard drive via the Fusion's built-in FireWire interface), mix to a real-time DVD

Fusion offers a well-labeled, comprehensive I/O.

simple naming and frame rates), or require its full-featured TC interface (beyond Record chase) or its user-configurable GPI triggering. In my spare time I did manage to check out some other features including M-S monitoring, assigning a quad-mic surround set to one fader, and even plugging in a USB keyboard for faster data entry.

Summary

One could mount an argument that its near \$10,000 price (when outfitted with optional cables, batteries, and effects) or its 300+ mixer cross points and 200+ user parameters (as highlighted in Zaxcom lit) make the Fusion overpriced and overkill for the target ENG market. In the opening, I mentioned the glacial pace of evolution in the adoption of new ENG audio tools. I specifically referenced attitudes, which are often correctly and/or necessarily based on “if it ain't broke, don't fix it” or “the need for speed” and, of course, tight budgets. You can connect those dots.

That said, there most definitely is a market for this stunning mixer/recorder; I'll just self-revise it to start at the rental-house and reality-show level and then move upwards into financed independents, 2nd and 3rd film units and any other full production units where mobility, flexibility, and centralized audio control are essential. In those applications, including those requiring high-resolution surround recording, the Fusion truly has no equal.

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Sound Devices 788T 8-Track Digital Audio Recorder

The 788T is a very good, totally pro-quality multitrack location recorder packed full of useful features.

The Sound Devices 788T is (over)simply described by Sound Devices as "a high-resolution digital audio recorder with time code." That's sort of like saying the Hummer H1 is an off-road vehicle.

This is a 4-pound, \$5,995, feature-rich, 8-track recorder with 160 GB SATA internal drive, rechargeable Li-Ion battery and external power supply/charger. In other words, this is not a prosumer hard-drive recorder; this is a very professional audio recorder specifically designed for film and video sound applications.

Features

The 788T is a second-generation device. Lessons learned from its predecessor, the 744T, have been applied. While viewed as a near-perfect recorder by some, the 744T reduced the range of wireless mic receivers at certain frequencies when packed next to those receivers. That problem has been remedied in the 788T. Also, the rechargeable battery is a lot easier to remove. Although fairly straightforward, I would suggest reading the manual before using it on your first shoot.

With the 788T, you can record up to eight tracks with very nice-sounding pre-

amps, line or AES/EBU digital sources to 16- or 24-bit WAV mono or WAV poly files (an interleaved multichannel file format. See <http://www.sounddevices.com/notes/recorders/mono-poly/> for details) to the internal HD (up to 2 TB), internal CF card and/or external DVD-RAM drive attached via either the 400 or 800 FireWire ports. Unlike the 744T, there are only seven sample rate choices between 32 kHz and 48.048 F.

The 788T generates highly accurate time code and recognizes composite NTSC, PAL, and Tri-level sync. The 788T supports 23.976, 24, 25, 29.97, 29.97 DF, 30, 30 DF, and 30+ frame rates via a five-pin LEMO connector on the right panel. It can be operated without time code or in free run, record run, free run jam once or 24-hour modes. The 788T also has its own sample clock with PLL circuitry and ignores external word clock, AES clock, and video sync during playback.

The left side of the 788T sports four female XLRs, four TA3M, a headphone

jack and rotary headphone control. The preamps in the 788T are new to Sound Devices. They are extremely quiet and very nice-sounding. The 788T came with firmware version 1.06 installed. I thought the input limiters sounded a bit grungy, but after downloading and updating the firmware to 1.07, they sounded a lot better.

The front panel is very straightforward: eight retractable input gain knobs separated by four toggle switches. Each switch instantly brings up the Input Settings Window, which allows access to each input's mic/line/digital, phantom power, input gain, high-pass filter (mic only), input limiter, track routing, and polarity settings. In this mode, the audio from the particular input selected will be soloed to the headphones. A separate input delay window allows each input to be digitally delayed in 10th-of-a-millisecond increments up to 30 milliseconds.

The rotary control on the right side of the chassis controls the selectable knee positions of the high-pass filter and through the main menus. At the bottom of the input window, a block of letters — LRABCDEF — are used to show track assignment for that particular input —



Ty Ford has been writing for *Pro Audio Review* since issue number one. You can find him at www.tyford.com.



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Any input can be routed to any track. Multiple inputs can be routed to a single track or any combination of tracks via four female XLR inputs and four TA3 inputs for balanced or unbalanced mic or line-level analog audio. The eight digital inputs enter through a single DE-15 (D-Sub) connector.

Each input gain pot is surrounded by an exceptionally cool and innovative LED light ring that glows green to indicate signal activity, red for the approach of clipping, yellow to indicate input limiter

activity, and flashing yellow to indicate an unlocked digital input. The green and yellow LEDs increase in intensity as the level increases.

The six analog and digital outputs are parallel, so whatever goes to analog output 1 also goes to digital output 1. On the right panel, there are four male TA3 active, balanced analog outputs, and a 3.5mm unbalanced stereo analog output for outputs 5 and 6. In addition, there are two male TA3 connectors for digital outputs 1-4. Digital outputs 5 and 6 exit through the DE-15 connector on the back panel.

The 13-segment green, orange, and red output meters, which also operate when recording, have nonlinear resolution and are small. The three red output LEDs fire between -3.0 dBFS to 0 dBFS. Personally, I think it would be better if red meant "over." The brightness of the LEDs is continuously adjustable for operation from full sun to total darkness.

Headphone amp gain is more than sufficient and also has a peak light of its own. Monitoring multiple inputs and outputs with one set of stereo headphones can be challenging. The 788T headphone output control supports the monitoring of each input and output, various selections or all, regardless of input and output routing assignments. In addition, the headphone circuit can also decode Mid-Side inputs to L/R and derives a stereo signal from the W, X, and Y channels when working with the 4-channel Soundfield B-Format.

Recording starts manually by pressing the REC button or from External TC-Auto Record or External TC/cont-Auto Record. The REC button stays red, so you can quickly see that you're still rolling. If you're wandering around the menus when the director calls "action," just hit the REC button and you're recording. One very nice touch, the 788T Pre-Record Buffer can be set to record a full 10 seconds before the Record button is pushed, except when in Record Run mode or in any of the four external time code operation modes.

In Use

During recording, I found that once in Record, subsequent presses of the Record button can be programmed to do nothing, or add a cue marker to the file being recorded, or to start a new file. The 788T records both Broadcast Wave and Poly-Wave files with additional metadata in the file's header, BEXT [Broadcast Audio Extension] and iXML data chunks. Poly Wave files are recognized by some nonlinear editing systems and for those that don't, Sound Devices offers free Wave Agent software, its file conversion utility for Windows-based computers, that will convert WAV files between

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World Radio History

Fast Facts

Applications

Location and mobile recording

Key Features

Eight mic/line/digital inputs; input matrix; records to internal HD; CF cards and external DVD-RAM drive; word clock; SMPTE; six analog and digital outputs; output matrix; AES42 connectivity

Price

\$5,995 list

Contact

Sound Devices | 608-524-0625 |
www.sounddevices.com

Product Points



- ▶ Supports up to 12-track recording
- ▶ Updatable firmware
- ▶ Interfaces with the new digital AES42 mics



- ▶ Inputs 4-8 require XLR to TA3 adapter cables if using XLR connectors

Score

A great effort for Sound Devices and a significant step up from the 744T.

monophonic and polyphonic formats [quoted from Sound Devices website].

A navigable file directory system allows files to be stored in different folders. Scene and take numbers can be changed with a USB keyboard via a separate USB jack on the back panel. False takes can be deleted.

Maybe you don't need eight tracks? How about six tracks of iso'd mics and two tracks for a stereo mix? If you get lucky, you might get a good stereo mix, but if you are really paying attention to your audio in post, pulling out unneeded sections can

Summary

Whether you're running as a highly mobile "bag" operator or have the 788T on a cart, this is a seriously good-sounding and stable recorder. In its design, Sound Devices has managed to reduce menu levels, increase comprehension and ease of operation. (They should get an award for just that.) Even when I purposely under-recorded and used normalization in an editing program to bring the levels up, the sound was excellent. \$6k for this 8-track recorder is a no-brainer. Nicely done, Sound Devices.

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MODEL U77

Universal Audio UAD-2 Solo

The latest card from UA offers a serious processing boost to DAW owners with PCIe systems.

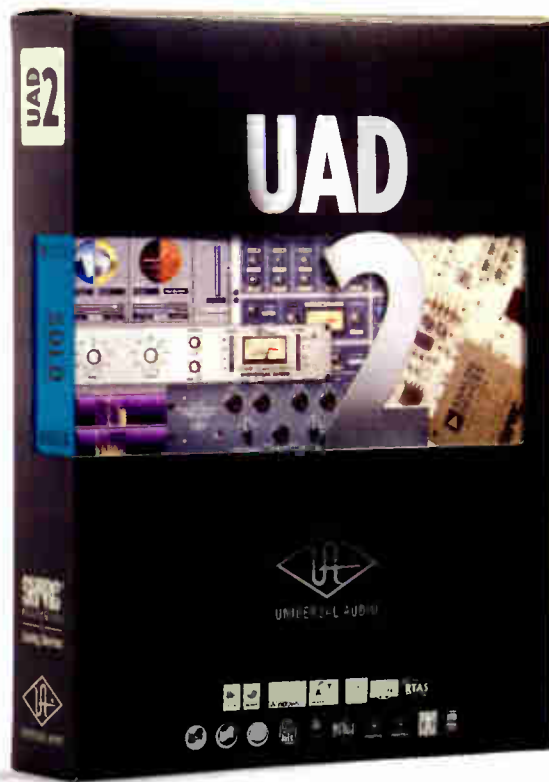
What if you're a DAW user and you've spent the last six hours working on a mix and your processor suddenly hits that dreaded brick wall? Do you remove a few plug-ins, maybe "print" a few tracks, or make sacrifices you don't want to make?

We've all been there. However, there are products available that — with a few minutes of installation — can not only get you through the power shortage, but also light up the entire city block. Universal Audio's UAD-2 Solo is just such a piece: a combination of software and hardware that can jump-start most any DAW.

Features

The UAD-2 Solo is a small PCIe card that slips inside your PCIe-ready computer and seamlessly runs a wide variety of Universal Audio software plug-ins. Compatible with VST and AU DAW setups (RTAS support is thankfully around the corner), it supports sample rates from 44.1 up to 192 kHz. It's ready to run on Mac OSX Tiger/Leopard and Windows XP/Vista operating systems. Fully compatible with the company's UAD-1 card systems, it can be up to 2.5 times more powerful than its predecessor and up to seven times more when using certain plug-ins.

As you would expect from the company with a 40-plus-year audio heritage, it



features a state-of-the-art, 32-bit, floating-point Analog Devices SHARC chip; these chips are monsters that can provide a serious amount of behind-the-scenes processing power. They are not only more efficient than the chips on the original UAD-1 card, but they run cooler (no heatsink is needed).

A core Mix Essentials II plug-in package is also provided with the system, which

includes the Pultec EQP-1A Program Equalizer, 1176SE Compressor/Limiter, RealVerb Pro Room Modeler, and CS-1 Channel Strip. Also provided is a \$50 Universal Audio voucher, which can be used towards purchasing any additional UA plug-ins.

Installation

The rather large package comes with the UAD-2 Short PCI-e card and a UAD Installation CD-ROM, Quickstart DVD and Catalog DVD. I first watched the helpful Quickstart DVD, then popped in the Install disc, installed the Version 5 software for my Intel Mac Pro, and then slipped the card into an available PCI-e slot; according to UA, it doesn't matter which one you use except for certain Mac G5 models. Within 20 minutes and with a quick restart, I was ready to go.

While the computer boots up, an LED light on the UAD-2 card will flash red and green. It glows solid green once the computer is fully booted. Since this was my first time installing a UAD card, it gave me peace of mind to see the light on. I knew I was going in the right direction.

In addition to the basic software plug-ins, an application called the UAD Meter is installed, a shortcut to which is installed on your Dock. When opened, a Control Panel opens with four tabs for System Info, Plug-Ins, Configuration, and Help & Support.

The next step is to authorize your product. Simply go to the Plug-Ins tab and hit the Authorize Plug-Ins button. You are immediately directed to the authorization page of UA's website. Then, within a few clicks, it's game on.

One caveat: I had to download the

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The **PRE420** is a 4-channel mic-preamp with a plethora of features, including built-in, independent stereo mix and solo busses. The sonic performance of the **PRE420** has been described as making the instrument "sound like it's being played right in front of me!" It delivers the audio with such clarity that no

textures are lost or obscured by distortion or noise. The remarkably low noise floor spans a wide range of gain setting, making the **PRE420** the perfect pre-amp for ribbon microphones. For room and ambient recordings, the ultra-low distortion performance puts the listener in the live-room. Also, the **PRE420** circumvents "Murphy's Law" with its bullet-proof "phantom-hot-plug" protection circuitry and incredible RF immunity.

The **ADC1 USB** is a reference-quality, 2-channel, 24-bit, 192-kHz A-to-D converter. The UltraLock™ clocking system delivers unvarying mastering-quality performance - regardless of clock source. The **ADC1 USB** offers variable input gain from -6 to +39 dB to interface directly with a wide range of devices. Precise levels are easily achieved with the 9-segment, dual-range LED meter.

The **DAC1 PRE** is a reference-quality, stereo monitor system controller with the DAC1's award-winning, 24-bit, 192-kHz D-to-A conversion system. The **DAC1 PRE** continues the legacy of the **DAC1**, which has become a staple of control rooms around the world. The analog inputs provide a simple and direct path to the monitors for mixing consoles, iPods, etc. The AdvancedUSB™ input supports native 96 kHz, 24-bit operation without cumbersome or invasive driver software. The built-in, 0-ohm HPA2™ headphone amplifier provides ultra-low distortion headphone monitoring.

Superior performance, reliability, and indispensable features have made **Benchmark** products absolute studio essentials.

latest version 5 software (5.1.0), which required a full re-authorization. It's only a few minutes of work. Note that aside from the Mix Essentials authorizations, there are also 14-day, fully functional demos of all of the plug-ins. So if you want to check out that Neve 1081, Helios 69, LA3A, or Fairchild, just hit the "Start Demo" button. If you have both a UAD-1 and a UAD-2 card installed, you can also choose which card the plug-ins will run on.

Booting up Logic Pro 8, it quickly ran through the "new" UAD plug-ins, and I was ready to go. Found under the Audio Units > Universal Audio menu, the full list of plug-ins appeared.

Simply insert the plugs into a session like any other plug-in and dial up your settings. Once opened, you can view an indicator on the plug-in toolbar that shows which type of card it's

running on, as well as a dropdown menu with .FXP Preset Management settings such as Load, Save, Copy, and Paste. This would allow you, for example, to save the settings of your 1176 LN and paste them directly into a "lighter" 1176 SE version. I tried this a few times, and it worked perfectly.

On the plug-in GUI is also a cool "Help" button, which can access info from either the UAD manual or info for that software from the company's website. Of course, there's also a "Buy" button, which lets you immediately purchase that plug-in online — a smart inclusion on Universal's part. Available only on the UAD-2, there is a "Low Latency" button that lowers latency to the point where you can use it live on an instrument while recording.

In Use

After using this card on a single session, I realized there was no going back. To be fair, I've been using UA plug-ins (especially the 1176) on my Pro Tools rigs since they first started developing them years ago. I still use them quite a bit on my HD setup, but now with this card, I have access to them for my Logic and Bias Peak work. To be able to put 1176s, Neve 1073s, Pultecs, Helios 69s, and DreamVerb on my tracks was a huge kick in the butt and far better than the standard Logic effects fare. There is literally no comparison to the before and after, especially with a Neve 33609 and UAD Precision Limiter on the Master fader.

This also applied to my Bias Peak Pro 6 work. I typically use Peak as a 2-track editor and master "assembler," but to be able to call up this bevy of plug-ins was sheer power at my fingertips. For RTAS users, UA expects to port all the plug-ins (except Nigel) to the Pro Tools platform before the end of the year. LE users with 003s and such will be able to have a huge power increase with the installation of just a single card.

Summary

The UAD-2 Solo is a compact and powerful system for those who need some additional plug-in help on their computers. Aside from the software sounding great, it's expandable; not only can you purchase additional plug-ins as needed, but up to four UAD-2 cards can also be used in one setup (along with four UAD-1 cards). Also, the UA family of Duo and Quad cards joins the Solo, each with additional processing power. If you're a DAW owner who's running on the edge and wants to make your mixes sound better, I highly recommend a UAD-2 Solo.

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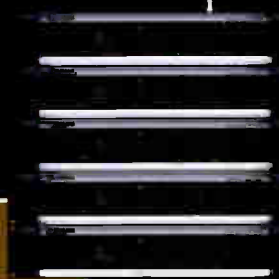
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Chandler Limited Germanium Compressor

For the mythical “desert island gig,” the Germanium’s extreme flexibility makes it a great choice in compression.



I often talk about compressors as “paintbrushes” because of the way they impart a particular character on our tracks. As engineers, we purchase a variety of them, each for their individual traits and colorations, never expecting any one processor to function ideally across the board (pun intended).

I now stand corrected. The Chandler Germanium Compressor has taken my own world of compressor assignments and methods and turned them inside out. This many features and this much versatility has made me rethink what I expect out of dynamics controllers.

Features

The \$1,680 (list) Germanium—named after a lustrous metalloid chemical element in the carbon group—is eye-catching with its yellow “chicken head” knobs and kidney-shaped VU against a dark blue chassis. It is an FET-based compressor with two XLRs for I/O (with transformers at input and output) and an outboard power supply (the PSU-1 at \$225 list) that can power two Germaniums (two compressors strapped together for stereo operation or another model in the Germanium line).

Germanium’s numerous controls include a hardwire bypass switch, a clean/dirty compression switch, a continuously variable input control that ranges from 1 to 11, a side-chain control,

a ratio control (again, 1 to 11), a compression curve selector, a wet/dry mix control, controls for attack and release times, a stereo link switch, the germanium output drive control and, finally, a feedback control.

Most notable is the Germanium’s choice of compression curves, which include six modes (achieved through a network of resistors, germanium diodes

“Why Germanium?”

by Wade Goeke of Chandler Limited

What originally got me started on the whole Germanium trip was that two of my all-time favorite units used them: the Neve 1057 [mic pre/EQ] and the EMI TG 12345 [EQ]. The sound of both intrigued me, but in some cases they were difficult to deal with—low headroom, oscillation, and noise. I wanted to make something with that type of sound and fix the negative aspects. My first experiments were very positive, and I knew I was on to something.

Historically, germanium transistors came between tubes and today’s standard silicon

and silicon diodes): R soft, Germanium soft and medium, Silicon medium and hard, and Zener hard. R soft is very neutral gain reduction, whereas the other modes offer the pleasantly warm, round, and distorted sounds that such transistor/diode materials are known for. Germanium, the element, in particular is often used for great guitar distortions, but is expensive and difficult to work with—noisiness being one its problems.

The side-chain control is also notable for its ability to HPF the control signal (without an external EQ) to prevent bass-induced pumping when used on final mixes and other full-bandwidth material. This control ranges from out (no HPF) to 300 Hz in six stepped settings.

transistors. Many users say that our Germanium pieces have a tube-like quality when set certain ways; it makes sense when you think of them as the transition from tube to transistors.

For me, [germanium transistors] offer a more organic, natural sound than silicon transistors. I do think there is more to the germanium thing than just throwing a couple of them together and expecting magic. A good deal of what I believe is special about our Germanium products is the implementation of the germanium transistors and the product design as a whole, not just the fact that we used germanium.”

The clean/dirty switch allows the unit to be neutral and accurate in the clean position (with THD between 0.2 and 0.5 percent) and somewhat distorted (THD from 2 to 5 percent) in the dirty mode, with only second- and third-order harmonics.

The wet/dry control allows phase-coherent blending of the original input signal with the processed signal, all summed at the drive/feedback controls. Such parallel compression is commonly achieved with mults and subgroups, but can be easily dialed in here.

Finally, the drive and feedback pots work together to affect the final output. The Feedback control changes how much inverted signal is routed back to the input and subtracted, which affects not only level, but also tonality and density. Turned fully to the right, feedback is minimized, with higher output, more distortion and a low bass bump. Drive affects how hard the germanium drive amp is hit, varying output level in conjunction with the feedback control.

In Use

In a nutshell, the Germanium is a fantastically versatile compressor that can do just about any dynamics squeezing you might require, albeit with patience and careful adjustment. Transparent dynamic control was unexpectedly quite achievable, even in the Germanium soft mode, with moderate input and ratios, clean mode, slower attack, a click or two of side-chain filtering, moderate germanium output drive, and 50/50 of feedback. To my ears, the Germanium is actually clean and hi-fi enough for mastering, although stepped controls for repeatability would help here.

The three Silicon and Zener modes aren't nearly as polite as the others, with more "grabbiness," a harder sound, and more sensitivity to both attack times and side-chain filtering. Whereas both Germanium soft and hard modes were typically flattering and tastefully "congealed," both Silicon modes were temperamental and sensitive, with hard being really hard. Zener was even harder sounding than Silicon hard, with a distinctive bark and transient grab that was distinctively aggressive.

So here's the fun part; with any input source/mode combo I found less than desirable, I could massage the attack, side-chain, ratio, clean/dirty, wet/dry blend, germanium output drive and feedback and get something I could actually use! This unit does indeed have a learning curve, but with experimentation it can do drums and whole mixes, it can do colorful and saturated. I couldn't make it do really dirty mangling, at least not all by itself. It can get all saturated, pumpy and grabby if you make it, but it's no distortion pedal.

Fast Facts

Applications

Studio, project studio, mastering, live/installed sound

Key Features

Features: FET-based compressor with hardwire bypass switch, clean/dirty compression switch, un-stepped input; side-chain control; a ratio control; a compression curve selector; a wet/dry mix control; attack and release controls; stereo link switch; germanium output drive control; feedback control; two XLRs for I/O (with transformers at input and output); PSU-1 outboard power supply is extra (\$225 list).

Price

\$1,680 list

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Product Points



- ▶ Allows parallel compression with mix control
- ▶ Multiple compression modes offer wide range of textures and knee behaviors



- ▶ No legend for ratios, attack or release times
- ▶ Limited metering

Score

I highly recommend the Germanium Compressor for those who mix in the box, like some analog texture and don't want to have only a single compression sound available.

Summary

Once I got used to having a plethora of compression options in one processor, the Germanium ultimately made my life easier, from tracking through mixing. I highly recommend the Germanium Compressor for those who, unlike myself, mix in the box, like some analog texture and don't want to have only a single, character-laden sound available. For the mythical "desert island gig," the Germanium's extreme flexibility makes it a great choice in compression.



RME ADI-8 QS A/D-D/A Converter

So far, my hot-rodged PT | LE system has brought lots of compliments and, most importantly, more than paid for itself. Naturally, I was eager to evaluate RME's latest 8-channel converter: the ADI-8 QS.

Features

The RME ADI-8 QS is a step up in functionality, and in price, from the DS; the QS replaces the DS's TDIF multi-pin with AES/EBU. In addition, for about \$1k more than the DS, the QS's basic features include eight channels of servo-balanced, isolated A/D and D/A conversion with less than 12 samples (.25 mS at 48K) of latency; up to four analog reference levels (-10 to + 20); 8-channel operation at single, double or quad speed

Aux ports for a 7.1 monitor mix. Input and output levels can be changed globally or individually by accessing the setup menu.

If you aren't a fan of mixing in the box and prefer to mix from your DAW through an analog console, multiple QS chassis can be linked for more channels, or the 64-channel MADI option may be employed.

The analog limiter can be pushed to achieve as much as 20 dB of gain reduction. The unit then behaves more like an effects box and A/D-D/A converter. I found no parameter adjustments for the analog limiter, but the digital limiter offers four ratio curves from 1.5:1 to 5:1 for denser audio.

Superclock is not supported, but RME's SteadyClock purports to surpass Superclock performance and selectable 75-0hm



up to 192 kHz; two ADAT optical I/O for eight channels up to 96 kHz; word clock I/O; MIDI I/O; digital input/output trim of 6 dB per I/O; analog and/or digital limiting; optional MADI I/O with delay compensation; hardware remote for store and recall of volume; and full remote control via either MIDI, MADI, or Windows PC. A Mac interface has been mentioned, but no firm release date has been set.

In Use

For your studio, this robust feature set could be the perfect fit, overkill, or may allow you to do things you hadn't thought of doing before. For example, in single-speed mode, you can output two identical 8-track digital streams via the ADAT main and

input termination is a handy feature. The clocking circuit also acts as a re-shaper, and RME suggests throughput of clock signals to devices in series rather than using T-connectors at the input.

Finally, signal level metering has been expanded in comparison to the DS; the QS shows eight LEDs per input and output

(continued on page 56)

Product Points



- ▶ Low-latency, low-jitter A/D and D/A converters
- ▶ Flexible I/O
- ▶ Solid clock



- ▶ No TDIF I/O

Score Almost \$3k for eight channels of really good A/D and D/A and lots of nice features to boot? What's not to like?

Fast Facts

Applications

Studio or mobile recording

Key Features

8-channel, low-latency A/D and D/A conversion up to 192 kHz, 24-bit, analog and digital gain reduction, highly accurate, re-shaping SteadyClock, four analog input and output levels, individual channel gain controls, optional 64-channel MADI interface.

Price

\$2,899 list (\$3,668 w/MADI option)

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Tim Hartfield (left, pictured with artist Rob Arthur) says using the Apogee Big Ben clock with Digidesign 192s makes "such a big difference, it's unbelievable."

In the workstation world, the point at which an analog signal is converted to digital is a critically important link in the signal chain. **by Christopher Walsh**

Needless to say, A-to-D and D-to-A converters run the gamut from budget to stratospherically expensive. Not surprisingly, the quality of a particular converter runs a corresponding path, though personal taste and other criteria are part of the equation as well.

Equally critical in any digital recording system is the master clock, or synchronizer. Stable and consistent sampling is essential to avoid jitter and allow an accurate A-to-D conversion, from which all subsequent work — processing, editing, mixing, and mastering — follows.

A sampling of audio professionals is likely to produce multiple references to the same handful of products; unlike, say, mixers or microphones, there aren't *too* many players in the converter and clock categories, and an even smaller number reside at the pinnacle, as measured by popular opinion.

Though the ears have it, many professionals also use the most-bang-for-the-buck yardstick. Converters can be very expensive; for the home studio owner, a DAW rig's onboard converters and internal clock may be adequate. But if the highest-possible sound quality is your goal, can you afford not to consider external converters and clocking?

"We've put half of our money into speakers and the other half into converters," says Scott Hull of New York-based mastering facility Masterdisk. "Nine or 10 years ago I came to find the top-of-the-line

Prism converters. When I moved to Classic Sound, I went from dCS to Prisms and just loved them. I can't master without them.

"If I had to work off the dCS's," Hull adds, "I would be able to. But the Prisms have a slightly more neutral, slightly more interesting bottom end. I'm really kind of stretching for differences, because it's somewhat program-dependent. I could find another set of speakers that might sound better, but I know these so well; that's why I use them. I feel the same way about the converter."

Familiarity, for Hull, is most important, neutrality a close second. The Prism AD-2 and DA-2, he explains, "tend to be a tiny bit more neutral, in that they don't feel hyped in the top end, they don't feel hyped in the bottom end, and that gives me a lot of latitude. I work on everything from extreme avant-garde to classical, and death metal in between. It needs to be very versatile, but unlike some of my contemporaries, I get very used to a particular sound of converters and don't change it very often. I don't try to match up the converter with the music as much as some people do. It's sort of an extension of my monitoring system and my speakers; I know how the other things that I put in the chain are going to interact with the converter. I feel like I've learned the converter so well that when I put something else in, I find myself just EQ'ing around it, EQ'ing to compensate for it."

Paul Antonell, owner and engineer at the Clubhouse Studio in

Christopher Walsh is the recording editor for *Pro Sound News* and the associate editor of *Pro Audio Review*.

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Paul Antonell of the Clubhouse in Rhinebeck, NY, finds the Digidesign 192's converters to be the most seamless and best bang for the buck. He uses an Apogee Big Ben clock; the studio also employs Lynx Aurora 16 converters.

Rhinebeck, New York says that, "after numerous listening tests and sessions using various converters — Apogee, Digidesign, Lynx, Lavry, Lucid — we decided, ultimately, that the Digidesign 192 is the most seamless, and best bang for the buck. We also decided to buy a Lynx Aurora 16 as an additional set of in and out. The Lynx sounds undoubtedly better than the Digi's, but when we purchased our Pro Tools system, the Auroras were not released yet.

"If we were to buy a new system now," Antonell feels, "we'd likely purchase the Lynx to outfit fully. They sound great, and while they may not be the *absolute* last word in converters, the price-to-performance ratio is amazing! That being said, we've been using our Digi 192 interfaces for four years now, running them off an Apogee Big Ben clock, and we couldn't be happier. They sound 'good enough,' easily, and the operation with Pro Tools is obviously seamless. With the Big Ben, the 192 becomes a very good performer, and good enough for us not to search out more expensive options for diminishing returns."

"I currently use Pro Tools |HD3 Accel with two Digidesign 192 converters, no external clock," says Los Angeles-based producer Brad Wood. "I run the outputs of the converters into a Tonelux console and back into Pro Tools for stereo mix. I usually track at 88.2k/24-bit and find that to sound great. I previously had Apogee AD-8000 converters for my Pro Tools MIX setup, but found the 192s and higher sample rate more pleasing to listen to."



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“The converter is sort of an extension of my monitoring system and my speakers.” —Scott Hull, Masterdisk

Dave McNair, a mastering engineer at Masterdisk who was previously a producer and recording engineer and continues to mix, feels that “when you get up in the region of the really high-end converters, they’re all pretty good; they just have minute differences. I’m really into the Lavry Gold AD122-96 MKIII, which is the A/D. Then I use the companion D/A, the DA924. I compared them to a lot of converters, and I just love the sound of them. I know that when I hear the Lavry stuff, everything is super effortless-sounding. That’s what I like about them. For the monitoring D/A,” McNair adds, “I use the current version of the Chris Muth D/A that is built into the Dangerous Monitor.”

“When I have needs for another, alternate converter in our production rooms or other circumstances,” says Hull, “the Benchmark DAC1 also comes up fairly high.”

An external clock can have a dramatic impact on a DAW-based recording system, audio professionals assert. “When a DAW is crunching numbers,” says New York-based engineer Tim Hatfield, “a good clock keeps them more ‘in time.’ The clock is just much more rigid, it shouldn’t waver at all. If you think about it in terms of music, it’s right on the grid, not stuff with swing. In this case, you don’t want the clock to swing — it’s not a drummer! When it comes to a clock, you want your drummer right on the grid.”

“I just use the converters that are in my [Digidesign] 192s,” Hatfield explains. “But I do use the Apogee Big Ben, and I think it makes a ton of difference. When I first got it, I had the MOTU 24I/O — 24 ins and outs— and also my 192s. At the time, I had kept the MOTU rig, because I still had projects going on Digital Performer. But the clock, with that, made such a big difference, it was unbelievable.”

In his home studio, Hatfield makes the A/D conversion in a Focusrite ISA43D MKII Producer Pack system “and go digitally into a Digi 002,” he says. “I use that digital input as the clock source. I don’t know if it makes that much difference — I never A/B’d it — though in my mind it does.

Plugging in the Apogee clock to the Digi 002 makes a huge difference, too. When you have a really great clock it’s amazing, and I don’t think there’s a better one than the Apogee Big Ben.”

“The Antelope Audio Isochrone 10M Atomic Clock is very expensive, but a really amazing box,” Hull offers. “It’s hard to not buy one once you’ve heard it. It really tightens everything up. It makes the converters generally sound a little bit better. It has its best, most profound effect — like any clock does — in a situation where

a clock’s kind of unstable to begin with. When I’m distributing my Prism AD-2 clock around the room, that’s pretty damn stable, so the differences there are noticeable, but just; it’s noticeable by a mastering engineer’s standards.

“It flies in the face of convention,” Hull adds, “but I would go as far as to say the Big Ben wasn’t better, [but] it was at least equivalent. Big Ben is good, and it’ll make a multitrack sound better instantly. But the Antelope atomic clock is really out at the extreme.”

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new live products



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To the benefit of American customers, Prism Sound has reduced the U.S. dollar price of its dScope Series III audio analyzer by 24 percent. The Series III system is a Windows PC (XP- and Vista-compatible) software application coupled with an external audio I/O processor for equipment-under-test (EUT) connection. In addition to a standard signal generation and analysis package for frequency response and distortion measurement, dScope Series III supports digital audio carrier testing, quasi-anechoic microphone/loudspeaker analysis and testing of Windows sound devices.

Price: \$9,850

Contact: Prism Media Products • 973-983-9577 • www.prismsound.com



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World Radio History



Audio for TV — Holiday Edition

As the song says, "'Tis the season to be jolly." This year, in addition to doing my normal touring gigs, Tony Bennett is promoting a new Christmas album we recorded this summer with the Count Basie Orchestra entitled, *A Swinging Christmas*. As part of a TV promo tour, I have visited some shows that we've done several times previously as well as a few new ones. Here is a roundup of the morning television circuit, which provides some interesting challenges — one is the time schedule [especially for someone who usually mixes in the evening].

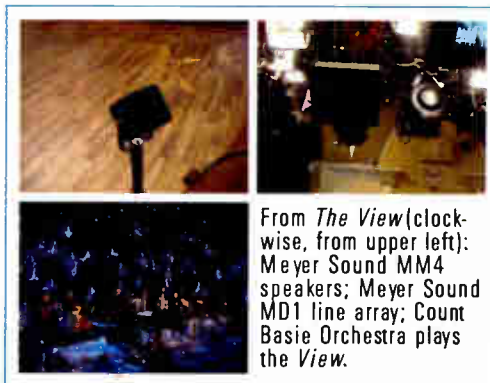
Today Show

To start off, the first show we performed on was the *Today Show*. I arrived at NBC studios at 2 a.m., their usual call time. A monitor system inside is supplied by PRG of New York and consists of Clair Bros. 12AM wedges and a Yamaha PM4000M console. Broadcast is handled in a room that was redone a couple of years ago when they updated the entire studio. A SSL console handles all production lavalier mics, music mixing, and playback. This is all mixed by one sound mixer — a pretty challenging feat — and is transmitted in 5.1 surround and stereo. I was amazed as I watched the show being mixed.

The View

My next stop was *The View*, which has a more civil load-in of 6 a.m. There is a live audience for this show, so a PA was installed by Sound Associates of New York. The system consists of Meyer Sound M1D line arrays and a CO speakers located throughout the studio for the music mixing. Meyer MM4 speakers are located around the seats for the dialogue. Two mixers handle the audi-

ence sound with a Yamaha PM5D for monitors and house L/R mix feed to a Yamaha DM1000 for the hosts' lavalier microphones. An SSL handles the on-air host mics in the broadcast booth and is fed the music mix from a Yamaha M7CL as well as all playback from the Yamaha DM1000. Three guys mix in the audio production room.



From *The View* (clockwise, from upper left): Meyer Sound MM4 speakers; Meyer Sound MD1 line array; Count Basie Orchestra plays the *View*.

Early Show

The *CBS Early Show* was next, which required an arrival time of 4 a.m. An older Crest console is used in the studio for monitors along with Clair 12AM wedges. The control room has an older Neve console and all production mics, while one engineer handles music. Playback and sound effects are fed by another Crest console and manned by another sound mixer.

Good Day NY

After I finished CBS, we ran over to Fox Studios for *Good Day NY*, a local morning program. We arrived at 10 a.m. to set up for taping. This studio provided quite a challenge, as I looked at the equipment it became obvious that they really aren't set up for music; with an assortment of only Shure SM57s, SM58s and one DI, I proceeded to mic up the band. They had two QSC self-powered wedges with no EQ for moni-

tors. A Yamaha MG console was all they had to mix the music, which feeds the control room. I must say it has been quite a while since I used SM57s on drum overheads and hi-hat. Mixing to tape through headphones in this console was not quite up to our standards, but after listening back it was acceptable, so we moved on.

Rachael Ray

The next day we taped the *Rachael Ray Show*, which is in a new studio and is taped to a live audience. The PA mixer uses a Yamaha DM2000, and the studio has JBL speakers flown for the music and control speakers throughout the seating area for the dialogue. The audience is on a moving turntable that rotates throughout the room to the various sets in the room. A Yamaha M7CL console is used for monitors with JBL wedges. One engineer in the control room handles on-air music, dialogue and playback and mans an SSL.

Conan and the Big Tree

We finished the album tour with the lighting of the Christmas tree at Rockefeller Center, followed by *Late Night with Conan O'Brien*, which we've done every year since he's been on the air.

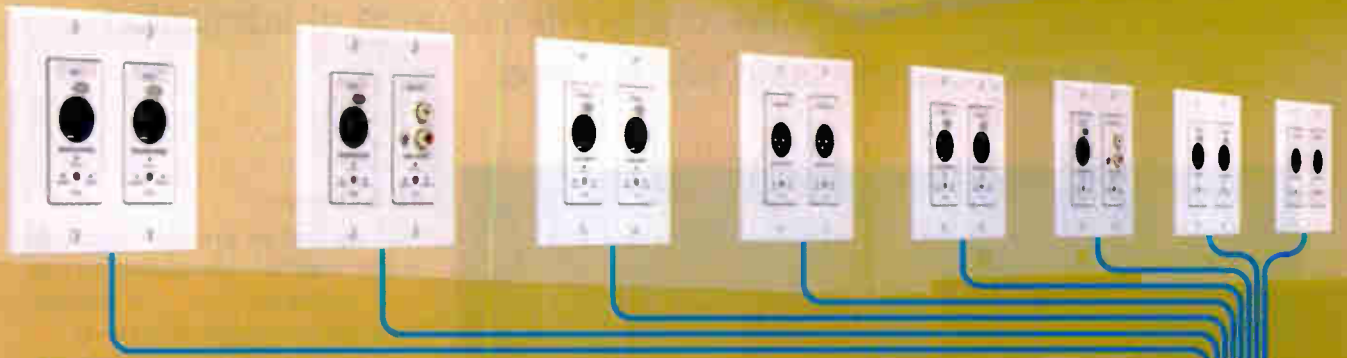
All About Results

It is interesting to spend some time working within the television audio world. What struck me were the differences between studios within the television production world; there is no de facto standard, by any means, on how audio is handled. Yet at the end of the day, it is all about results, and the engineers achieve them. My hat is off to the audio crews working in this field, as the hours they put in are long and the challenges they face are handled admirably.

Tom Young is the lead engineer for Tony Bennett and a regular contributor to *Pro Audio Review*.

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Yamaha IM8 Series Mixing Consoles

This midrange live mixer series from Yamaha offers expected standard features and one feature worth the price of admission alone: one-knob compression per channel.



In many ways, the Yamaha IM8 Series is quite similar to other large-format live analog consoles currently available in the pro audio marketplace.

Yet these three models offer two new and notable features to the world of big analog desks: a one-knob compressor on each channel and a USB interface for external playback and record.

Features

The Yamaha IM8 comes in 24-, 32-, or 40-channel frames. Additional inputs include four stereo channels on faders and four stereo auxiliary returns on knobs. They all come with eight pre/post selectable auxiliary sends, in sets of two or eight subgroups, a stereo and mono bus, and a 4-channel matrix section. The console also features four selectable mute groups. Each channel offers a 26 dB pad, phase

reverse, and a switched 12 dB/octave 80 Hz high-pass filter. The EQ section offers 4-band parametric EQ with low shelving at 100 Hz, a lo/mid band sweepable from 80 Hz to 1.6 kHz, high/mid band sweepable from 400 Hz to 8 kHz, and high shelving at 10 kHz. The channels are constructed in groups of eight with a separate center master section [for those concerned about servicing].

The console is powered by an external Yamaha PW8 power supply. The power supply takes up two rack spaces and weighs in at 24 lbs. Two power supplies can be redundantly linked via a PSL1010 link cable for zero down time in the case of failure.

At A Glance

I was eager to hear what this console had to offer, especially considering the one-knob compressor and USB interface. These features could truly be handy.

At first glance, I noticed the well-labeled layout and its sleek, modern look. On the rear panel, I liked how all the aux outputs were on XLRs, which is very convenient. The subgroup, matrix, and monitor outputs are on 1/4-inch TRS jacks. Personally, I prefer XLRs on these outputs; I used this console both for monitor mix and corporate use, both requiring a lot of zone outputs. However, this output arrangement is typical for this price range of consoles. I brought a lot of TRS to XLR adapters to remedy this.

All the inserts on both channels and outputs were single-point unbalanced, also typical in this price range. The tip is wired for send, ring for return, and ground on the sleeve. One problem with the layout of the back of this console is that the direct outs, 1/4-inch inputs, and inserts are located below the mono XLR inputs. It is awkward to plug in the inserts, and I had to unplug the XLR connector to make sure the 1/4-inch connector was going into the right hole. Since a snake fan-out almost always stays plugged into the XLRs, it was a cumbersome workaround; this presents a major problem when the need arises to hot-plug an insert mid show — there is no option of unplugging to make sure it's being plugged in correctly. I would have really liked to see XLRs on the bottom [like Yamaha's higher-priced consoles].

Fast Facts

Applications

Live music clubs/theaters, houses of worship (HOW), functions as a 2-track recording console

Key Features

24-, 32-, or 40-channel frame; eight aux sends; 2-track USB interface; one-knob compressor on each channel.

Price

MSRP 24-channel frame—\$4,799, 32-channel—\$5,799, 40-channel—\$6,499

Contact

Yamaha Live Sound | 714-522-9011 | www.yamaha.com/livesound

While labeling the console, it was difficult to lay down marking tape since the screws that hold the modules to the frame are in line with where the tape goes, both above and below the faders. Meanwhile, I like the aux masters being on faders, not knobs. This is not a new feature to Yamaha. I wish there was a mute button on the aux masters: a handy feature when a band goes on break and masters aren't set at unity or to set up different speaker zones for a show when you just want to listen to a particular zone. The mute group master buttons, in my opinion, are poorly placed. I like them within easy reach, which is usually closer to the faders. On the IM8, the mute masters are placed



IM8 inputs and outputs abound.

at the top right of the centrally located master section. On the other hand, the monitor and phone level knobs are well placed right above the master faders; they are nice and large enough for easy and accurate turning. The headphone jack is located under the armrest of the console in the center, preventing finger tripping as a cable crosses the faders.

Surprisingly Good EQ

The console itself sounded surprisingly good for its price range. Its EQ was pleasantly sensitive; little adjustments were noticeable and did exactly what was expected. Personally, I like a little bit of overlap with my EQ parameters, but there is a hole between 8 kHz and 10 kHz where the mid/high knobs and high knobs overlap and I wish the lo/mid band went even lower in frequency.

Product Points



- ▶ One-knob compressor creates simplicity
- ▶ Large Q of EQ makes it easier for those without a refined ear



- ▶ One-knob compressor limits control of compressor
- ▶ Two-track USB I/O has major routing limitations

Score The Yamaha IM8 is a good midrange analog console, ideal for installation in houses of worship and clubs featuring live performances.

The EQ sounds like it is a rather wide Q. I have no problems with a wide Q; sometimes it is handy, but sometimes it isn't. The preamps sounded very Yamaha—typical, in my opinion. They sound fine until they are hit hard; then they can become crunchy and thin-sounding.

That One-Knob Compressor

Undoubtedly, the IM8 console's most important feature is the compressor on each channel. The compressor worked as expected, smoothing out the dynamic range evenly as a compressor should. I must admit that, before using the console, I was a bit skeptical of the one-knob compressor. Yamaha took out adjustments that most engineers are accustomed to and think we need in compression: ratio, attack time, release, etc. A one-knob compressor is oversimplified and "dumbed down," if you will. Yet after using the one knob compressor, I liked its simplicity. As it compressed and reduced gain, it added gain so the output level always sounded about the same. To my ears, the

compressor sounded like the compression ratio was constant throughout the knob turn and, in using it, you are simply changing gain and threshold with one knob. The one thing I did not like about the compressor is that it exhibits a hard knee; I prefer to hear a smooth transition between when compression is present and not.

After my first show using this console, I wanted to assess the quality of the compressors. I put together a test rig in the shop consisting of a microphone split to three different channels. Channel one used the one knob compressor on board. Channel two used a channel of a dbx 1046. Channel three was a Behringer Composer pro compressor. I first went through each compressor and set the threshold and the output gain to the same as the setting I had for the Yamaha IM8 knob. Next I A-B'ed back and forth between the channel compressor and the 1046

until I hit a compression ratio setting closest to the console. I moved on and did the same with the Behringer. From this test, I observed that the compressor is set for a light 2:1 compression ratio. This seems to be consistent throughout the one-knob compressor settings, confirming my earlier observation.

Rating the three compressors in order of quality, highest would be the dbx 1046, then the IM8 onboard compressor, and lastly, the Behringer. From my experience with a variety of compressors, the IM8 compressors seem to be similar in quality to the original dbx 166. So, I would agree with the Yamaha ad that states you get "a free mixer after you pay for each compressor." This was not something that I expected in this console's price range, yet it was the case. As a matter of fact, I think its compressor blows away others of a similar price. The IM8 compressors came in very handy to offset its preamps; when the latter became crunchy, the former smoothed things out and the channel's audio became more pleasing to the ears.

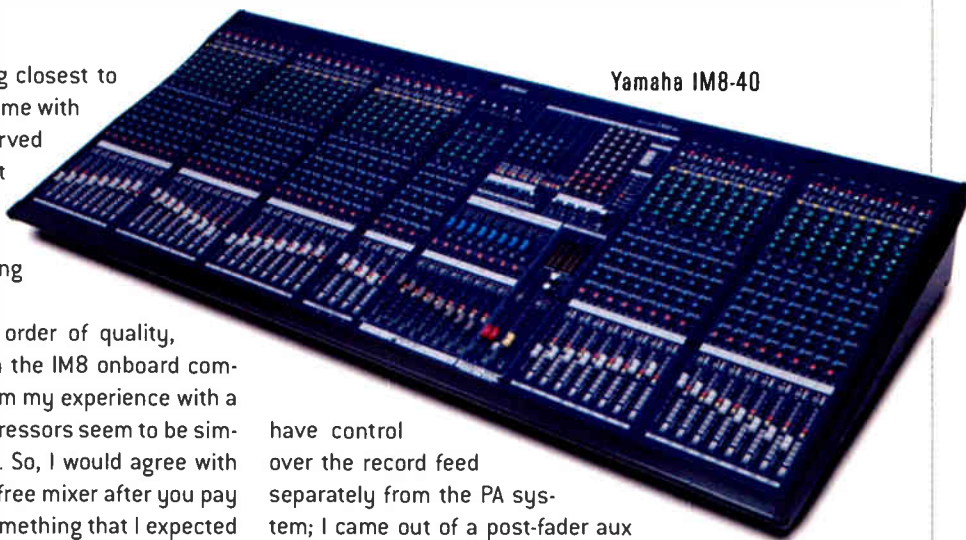
USB to DAW Connectivity

The second great feature of the IM8 — the USB interface — is a USB 1.1 B-type connector. In order to interface with the console, a driver from Yamaha can be downloaded from http://www.yamahaproaudio.com/downloads/firm_soft/audio_driver/usb_osx.html. The driver is also available on a software CD, included with the console. Mac requirements are an Intel processor, at least 10 MB of free space, and at least 512 MB of memory. PC requirements allow XP or Vista; XP requires a 750 MHz Intel processor, 10 MB of hard-disk space, and 128 MB of RAM, and XP must be upgraded to Service Pack 3, the 32-bit version. Vista users require a 2 GHz processor with 10 MB of hard-disk space and at least 1 GB of RAM. Vista must be upgraded to Service Pack 1.

The driver installed easily on my MacBook Pro. All I had to do was plug in the console and, from my preferences menu, choose the Yamaha USB for input and/or output on whatever application I was using for playback or record. Some applications actually offered a pop-up asking which input/output device I wanted to use. The USB worked flawlessly, and it was the easiest experience I've had with a Yamaha product and my computer.

However, I did find that the routing of the USB input through the console was very limited; it can only be routed to the stereo or mono bus. The downside is if you choose to run aux-fed subs or aux-fed sends to front fills or another zone, you can't send this input there. I got around this roadblock by routing the USB to the mono bus, which I wasn't using for the PA, and then routing it back into an unused channel. It used up a mono channel and it requires an XLR, but it fixed the problem.

For recording, I had to do something similar, as I like to



Yamaha IM8-40

have control over the record feed separately from the PA system; I came out of a post-fader aux subgroup (or any output of my choice) and came back into a mono channel, which was routed to the mono bus. In the end, it was easier to use my own USB interface and go into a stereo input and come out of the output I want for the record than to do the routing demonstrated above. I should point out that the console controls for the USB also controls a RCA input and a mini TRS input found on the front edge of the console next to the headphone output.

The IM8's software disk includes a copy of the Steinberg Cubase AI4 workstation, which will work with both Mac and Windows platforms; AI4 is a light version of Cubase 4 and includes features such as being able to record 48 audio tracks and the ability to plug and play with both Yamaha and Steinberg products. I didn't use AI4, but I personally own a version of Cubase LE and if AI4 is anything like it, I find it very easy to use with virtually no learning curve.


A Good Console That's Great For HOW and Clubs

The Yamaha IM8 is a good midrange analog console. It will be great for installation in houses of worship and clubs featuring live performances, where saving money on outboard gear and ease of use for novice engineers are a must. Thanks to having a compressor on each channel, the need to insert one has lessened, since there is one at your fingertips. The IM8's one-knob compressor takes the guesswork out of attack times, ratios, and makeup gain. Compression is simplified; just turn it until it sounds good. The wide Q of the EQ knobs offers an easy solution for users without well-trained ears.

If someone wants a stereo recording of a performance, or has a few songs that they want to play off of their laptop, the IM8 makes it easily possible. There are limitations in the IM8, such as not having the flexibility to route the USB input everywhere and a cumbersome input panel that is awkward for patching inserts or direct outs. The IM8 console is designed for the end-user without a lot of experience in getting a good-sounding mix, but who can figure out the basic routing of a console.

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NTI Digirator DR2

The DR2 is a handy digital testing gadget that is easy to use and highly portable.

The Digirator DR2 is the latest Minstrument from NTI, the audio test and measurement company headquartered in Liechtenstein. The Minstrument product line consists of a series of separate test generators and signal analyzers spanning a wide range of digital, analog, and acoustical tests.

These are lab-grade instruments, but are all compact, battery-powered (three AA cells), and rugged enough for field troubleshooting or system testing. The DR2 provides a comprehensive range of audio test signals as well as specialized test sequences for verification and adjustment of Dolby Digital, Dolby E, ProLogic, and DTS surround installations.

Features

The DR2 follows the form factor of NTI's second-generation Minstruments—a palm-sized unit with LCD, scroll wheel and Enter button, a set of quick-access buttons for Level, Frequency, and Waveform selections, a Power button that doubles as a display backlight switch, and a Mute button. Finally, there's a button to set the resolution of the scroll wheel: 0.1 or 1 dB steps for level and frequency steps of 1/3, 1/6, 1/12 octave or 1 least significant digit. The case fits comfortably in one hand, and a soft, shock-resistant jacket is provided, as well as a hand strap.

An RCA jack provides a transformer-isolated 75-ohm S/PDIF or AES3-id (with the supplied RCA-BNC adapter) output. A male XLR supplies 110-ohm balanced AES/EBU or unbalanced AES3 with the supplied XLR-BNC adapter. A TosLink optical output provides either stereo S/PDIF optical or an 8-channel ADAT Optical output. The normal output is stereo (two

channels) but either channel can be switched off, or the polarity of one channel can be inverted with respect to the other.

XLR and RCA outputs span the full sample rate range from 32 to 192 kHz. The S/PDIF optical output goes to 96 kHz, though the ADAT output operates only at 44.1 and 48 kHz. I was pleasantly surprised to discover that the ADAT output can be used to test four channels of a 2X sample rate S/mux input. The optical output can be switched off, providing a bit of additional battery life.

A female XLR serves as a sync input should it be necessary or desired to synchronize the Digirator's output with house sync. Normally the DR2's internal low-jitter clock provides the data clock reference, but when a suitable signal is connected to the Sync Input, it is automatically detected, the Impedance menu pops up allowing you to select 75-ohm or 110-ohm termination or bridging, and the sync signal can then be selected as the clock source for the output data stream. Sync can be word clock, PAL or NTSC video black burst, DARS (AES3 without the audio) or an AES/EBU or coax S/PDIF audio stream.

While the DR2 is primarily a signal generator, it is also capable of verifying a channel or device under test for bit-accurate transmission as well as throughput delay (latency). The sync input connec-



tor doubles as the input for Transparency and I/O Delay testing. Firmware updates and audio test file loading and management are via a USB port.

The Digirator generates a variety of test signals including sine, pink and white noise, stepped or continuous linear or logarithmic sweeps (called "chirp"), a burst for measuring throughput delay (latency) and a polarity test waveform which works in conjunction with the NTI analyzers. In addition, Dolby and DTS non-linear (encoded and compressed) PCM multichannel test signals in a variety of formats are available, offering channel identification, pink noise, tones, and a polarity test for surround systems.

The DL2 also functions as a WAV file player, allowing any waveform, test sequence, sound, or announcement that you can record to be output. Sound files, loaded via the USB port, are stored in folders in the DL2's nonvolatile memory. Selecting the FILE menu presents a list of available sound files. Because the memory size is limited and the non-linear test

Mike Rivers has a long list of engineering credits with the Smithsonian and is the author of the last Mackie HDR manual.

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files are quite large, only a basic set is pre-loaded. Additional test files and some instrument sounds are supplied on a DVD.

All test signals originate as 48 kHz WAV files, which get internally converted to the desired output sample rate. This limits the highest audio frequency to a tad below 24 kHz, a minor quibble, as the DR2 can't check the full-range frequency response of a 96 kHz channel.

The Digirator offers control over the Pro/Consumer format, Non-Linear, and Emphasis channel status bits. Changing the Emphasis and Non-Linear flags doesn't change the test signal, but it provides the ability to check the receiver's response to those flags.

Up to 10 specific test setups (waveform, level, frequency, sweep time, etc.) can be stored as Configurations, which can be named (limited to eight characters, so you gotta be creative) via the USB port.

In Use

The obvious generator functions are very straightforward. Choose the test signal, connect the Digirator to the device under test, and go. The display and menus are intuitive, and the quick access buttons speed adjusting the signal parameters. For the more complex tests, the LCD displays what you need to see including the clock sync source, sample rate and, when running with external sync, the incoming clock frequency. Selecting SWEEP or CHIRP from the Waveform menu pops up a submenu for setting the starting and ending frequency, dwell or sweep time, and for selecting a single or continuous sweep. Selecting FILE displays the current folder and file name. The cursor and scroll wheel are used to navigate folders and select the file with the desired test signal.

The Transparency test is a simple go/no-go verification that all the audio data bits pass through the channel unchanged. Passing the test gives confidence that the channel will successfully transmit non-linear (encoded and data-compressed) data such as Dolby and DTS multichannel surround audio formats as well as conventional linear PCM audio. If the test fails, it's time to get out a serious data analyzer such as NTI's DL1 Digilyzer to locate the specific problem.

The manual is sketchy about what, specifically, is tested in the transparency test, and how to tell whether the test passed or not since it neither says PASS or FAIL. A change in sample rate or changed data bits is indicated by the generic "DATA CHANGED" message, which is indeed suspicious.

As long as both ends of the channel are accessible, Transparency can be checked with a single instrument. Two Digirators can be used on opposite ends of the channel, if necessary. In addition to end-to-end testing, a recorder can be tested by recording the Transparency test signal and playing it back into the DR2's input for analysis.

I/D delay is another in-out test, which measures the time required for a pulse to pass through the AD-DA channel. The resolution for this test is 0.1 ms (also displayed as PAL or NTSC

Fast Facts

Applications

Installed systems, maintenance shop, any place where digital audio data needs to get from one place to another

Key Features

Generates a comprehensive set of test signals including stereo, 8-channel ADAT optical, and Dolby and DTS surround tests at all standard sample rates from 32 kHz to 192 kHz; plays WAV files for special test sequences, odd waveforms, and any audio material up to 20 kHz; outputs are XLR and TOSLink, sync and end-to-end test inputs are XLR and RCA phono; XLR-BNC and RCA-BNC adapters are included.

Price

\$1,850 list

Contact

NTI Americas | 503-684-7050 | www.nti-audio.com

Product Points



- ▶ Really simple to use
- ▶ Generates common wave forms as well as anything you can record



- ▶ Audio bandwidth is limited to 20 kHz for all sample rates

Score

A powerful tool for solving digital interfacing problems and proofing systems that is well worth the cost.

video frames]. For checking latency through a DAW chain, a display in samples would be convenient.

There's no official estimate of battery life, but NTI says that at trade shows, where it's continuously on, they get four to eight hours from a set of alkaline cells. Auto shutoff and intelligent control of the display backlight can increase battery life. NiMH or Li-Ion rechargeable cells work fine, though you can't recharge them in the unit. An optional AC power supply is available for bench use, however the manual recommends battery operation for the cleanest signals.

Summary

This is a handy gadget for testing digital inputs and transmission chains. It's easy to use, can go anywhere, and can perform a range of useful tests. It's a good tool for the maintenance shop or a broadcast facility, but probably overkill for the project studio. My only wish is that the audio frequency range extended to 40 kHz or above, but that would require a significant redesign.

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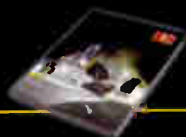
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The Professional's Source

Sencore SoundPro SP495

The SP495 provides a comprehensive range of tools for well-equipped sound consultants and select FOH engineers.



Sencore's SoundPro series, the successor to the TerraSonde Audio Toolbox family of test instruments, offers self-contained, multi-function test capability for audio measurement.

Sencore's SoundPro series, the successor to the TerraSonde Audio Toolbox family of test instruments, offers self-contained, multi-function test capability for audio measurement. The SoundPro SP495 Audio Consultant provides a full swath of core and optional test functionality focused on acoustic and audio analysis. The standards-compliant test capabilities of the SP495, while applicable to, say, an FOH engineer, the "Consultant" part of its name also suggests where the test set is most likely to find its broadest embrace — with audio consultants working in installed sound, including house of worship, corporate, home theater, theme park, public address, distributed sound, and industrial sound analysis.

The SP495 is a 5 1/2" x 12 1/2" x 2 1/4" device (a portion of the length

being an integral handle), weighing a bit over 2 1/4 lbs. It comes with a "line lump" DC power supply but can also operate from an internal (user replaceable) Li-Ion battery pack for as much as 5 1/2 hours (on a 6-hour full charge), depending on application. It also comes with a test microphone and padded soft case.

Core Functionality

I have dabbled with the SP495's predecessors, so it was easy to figure out how get basic functions running. The front panel of the metal case is uncluttered, with just a rotary dial encoder and a backlit color LCD screen. Spinning the dial and clicking it selects menu items and parameters for adjustment for all SP495 functions. You'll spin the dial a lot, based on the num-

ber of control elements on the test screens and the flexibility of the unit (for some functions, like the full-range generator frequency control, the interface would benefit from an increased parameter step size based on dialing speed). The bottom-panel sports combo 1/4-inch/XLR ins and an SD memory card slot, the top-panel analog and S/PDIF/TOSlink digital outputs and bidirectional USB I/O. I confess to reverting to the extensive manual (provided in PDF on disc) after a short time of familiarization — getting the most out of the SP495's capabilities and flexibility demands a more intimate knowledge of the device.

With the SP495 in tow, I met up with SR pro Ben Williams at a church where he recently finished a sound system overhaul. The SoundCore base software gives a range of, well, core functionality that we ran through first, beginning with simple SPL measurement (a range of 25-105 dBA SPL measurable with the supplied ANSI Type 2 microphone — an optional Type 1 microphone extends the range to 146 dBA). Though sound level is a simple test, the Sound Level Measurement screen serves to familiarize the user with a lot of SP495's flexible controls — speed of averaging/ response, automatic or manual mic pre gain, flat, A- or C-weighting, theater X-curve, relative measurement and peak hold, bargraph and direct numeric read out, speaker/headphone monitoring of input/output or USB I/O signals and generator level and signal controls (on/off, Pink/White noise, USB, sine

Frank Wells, formerly a radio broadcast and recording studio technician, is the editor of *Pro Sound News* and the editorial director of *Pro Audio Review*.

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and square waves — stepping by 1-2 Hz/octave/1/3 octave for tones). Settings like output level and mic gain are maintained from test to test while the unit is powered on.

For an install like a church or for testing environmental noise sources, it can be useful to know what levels are produced over time, facilitated in the SP495 by Leq averaged reading over a user-defined timeframe and by Dosimeter readings, measuring SPL over a defined time and correlating measurements to sound pressure exposure standards. There's also an automatable SPL chart recorder function that can log measurements over time, useful in part for evaluating environmental noise in a given locale, including elements like HVAC systems and external traffic noise sources.

RTA analysis of the room plotted the instantaneous frequency performance on a 1/3-octave scale, and with FFT measurement for up to 1/30 octave resolution (though with a slower refresh for the three-cascaded 1024-point FFTs). The plots could be frozen for deeper analysis, including cursor control to sweep to individual frequencies/peaks, and data can be stored for later use or use in a "difference" mode for before/after comparison. We also learned details on aspects of the room itself that Ben had already "intuited" during his system tweaks. RT60 measurements were facilitated with ease by the SP495, as well as a graphical display of energy decay vs. time, useful for mapping acoustic delays in a system and the incidence of reflection arrivals.

Additional core tests include speaker polarity (which can be performed up to 100 feet from the speaker...nice), actual speaker impedance measurement and a cable-tester function.

Giving You Options

The SP495 can be tricked out with a number of add-on firm-

ware options, and the review unit was fully loaded. The SoundPro TechBench option adds tests like electrical signal level, frequency count, sweeps of frequency response and impedance, distortion (THD+N and IMD) measurement, phase and crosstalk. The resolution on these tests is decent for a handheld device, and might just save a tech carrying an additional test instrument into the field. Included is an oscilloscope emulation for rudimentary waveform visualization.

A major concern for many in the install market is speech intelligibility, and the SP495 can be upgraded to include industry-standard ALCONS (a numerical prediction of

Fast Facts

Applications

Live Sound, Contracting/Install, Studio, Home Theater

Key Features

Standalone with a broad range of acoustics and sound system analysis test capability, extended through optional firmware. Comes ready with case and cabling.

Price

\$3,995 list

Firmware Options:

Speech Intelligibility \$395; Noise Curves \$195

Time Delay Analysis \$795 — comes with TerraLink software Audio Stethoscope \$95; Tech Bench \$295

Software Options:

TerraLink \$295; MacLink \$295;

Precision Type I Mic \$1,295

Contact

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Product Points

- ▶ Portable and self-contained
- ▶ Wide range of test capability and parameter control

- ▶ Price can add up quickly with add-on modules

Score

Recommended for audio consultants with a need for an extensive range of standards-compliant testing.

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“Real Love” — Lucinda Williams

Engineer’s Diary:

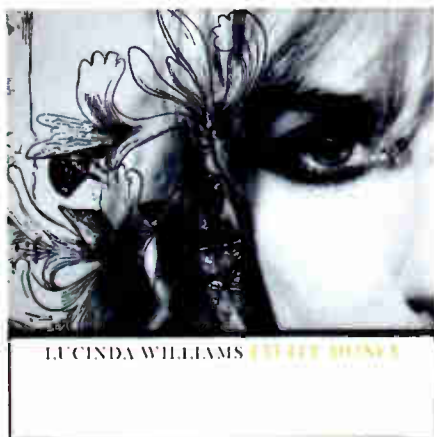


Eric Liljestrand

Since the breakthrough of 1998’s *Car Wheels on a Gravel Road*, Lucinda Williams has recorded a succession of equally essential albums at a pace far exceeding the years preceding it. Eric Liljestrand, who recorded her previous album, *West*, and subsequent tracks for various benefit releases, assumed the co-producer title for *Little Honey*.

“Real Love,” to this listener, recalls the Rolling Stones in mid-tempo, countrified bliss, electric guitars calling and answering as a solid rhythm section propels the group. “I think of this more as Faces than Stones,” Liljestrand answers, though he acknowledges that “several people have called this her *Exile on Main St.* record. It’s pretty fun.”

“Real Love,” was recorded in the Neve 88R-equipped Studio D at the Village in West Los Angeles, and mixed next door in Studio B, immediately following installation of the studio’s second 88R, this from the defunct Sony Music Studios in New York.



“Real Love” and the other 12 tracks on *Little Honey* were recorded to Pro Tools at 96k. “On *West*,” says Liljestrand, “I had wanted to go to 2-inch Dolby SR, and we never got the chance. This time I didn’t even bother and, although I love the sound of tape, it was the right decision, because it let us move more quickly. This record was so much about capturing the moment. Her vocal was cut live with the band

on all but two songs. I did so little to it—maybe steal a word from somewhere. She’s a dream come true: She’s got that rough, untrained-sounding voice, but you can understand every word she says, and she’s always in tune. What could be better?”

“Real Love” was mixed to a modified tube 1-inch Studer 2-track, which was brought into Lurssen Mastering along with a modified 1-inch Ampex ATR 102. “We ended up mastering off the ATR,” Liljestrand confides. “I was a little dubious at first, but it gave the midrange of the music, especially her voice, a real solidity. And, of course, the tube 1-inch just sounds enormous.”

Lurssen, who also mastered *West*, “has such a light touch,” says Liljestrand, “at least with me and Lucinda. The stuff comes out sounding just like I wanted it, only better.”

Song Facts

Single: “Real Love”

Album: *Little Honey*

Date Recorded: January/February 2008

Producers: Eric Liljestrand, Tom Overby

Engineer: Eric Liljestrand

Mixer: Eric Liljestrand

Mastering Engineer: Gavin Lurssen

Other Projects: Liljestrand has produced two albums by Diamanda Galás, served as engineer on soundtracks including, *Shortcuts* and *Kansas City*, and has worked with artists including Ringo Starr and Elvis Costello with the Brodsky Quartet. He won a Grammy for Bill Frisell’s *Unspeakable*, and edited and mastered the Grammy-nominated Lenny Bruce box set, *Let the Buyer Beware*.

Single Songwriter: Lucinda Williams

Mix Monitors: Genelec 1032

Tracking Signal Chain [Lead Vocal]:

Neumann U 47, Neve 1081 preamp, Universal Audio LA-2A

Recorder: Pro Tools | HD

Select Mix Plug-Ins: “I used zero plug-ins.”

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