Single Slice: Alabama's Hit "When it All Goes South"

PRO AUDIO REVIEW

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March 2001

Audio Processor Issue

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Vol. 7 Issue 03 • March 2001

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

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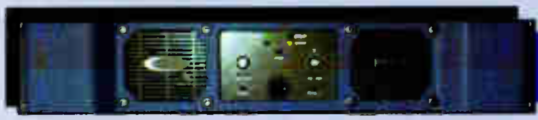

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SONY

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– **FRANK FILIPETTI, Grammy Award Winning Engineer/Producer**

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– **BOB LUDWIG, President of Gateway Mastering and DVD, Les Paul Award Recipient, and 9-time TEC Award Winner**

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PRO AUDIO REVIEW

Vol. 7 Issue 03 • March 2001

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

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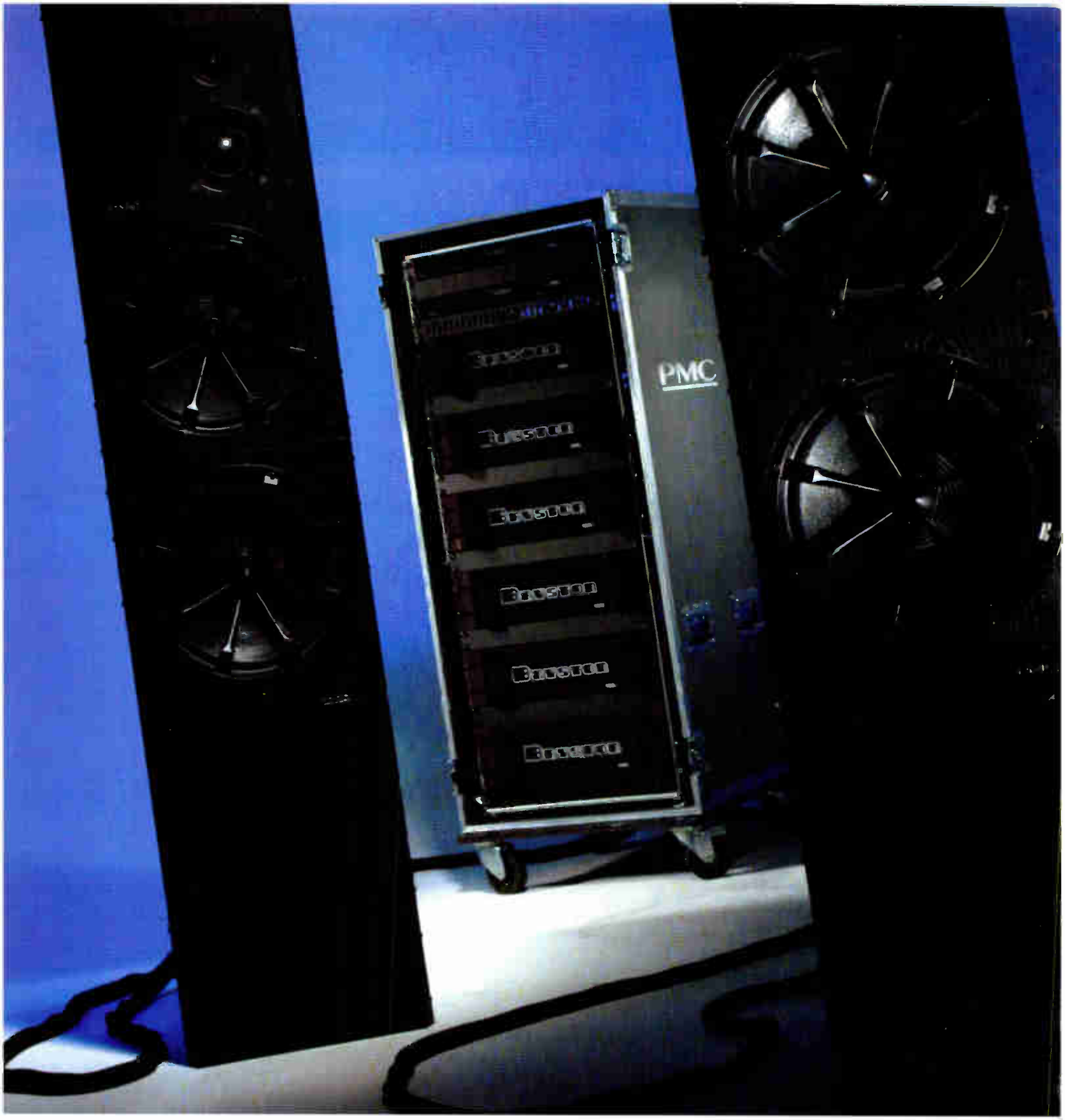
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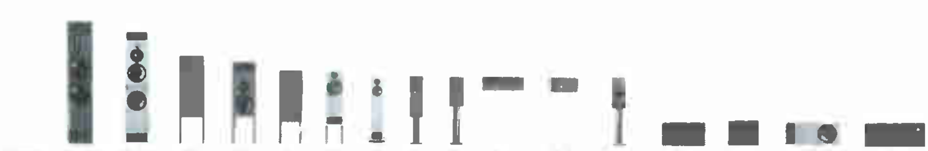

Power to the People

by Stephen Murphy

One of the big trends coming your way is the use of next-generation, application-independent digital signal processing (DSP) cards to enhance and empower otherwise-native recording platforms. This exciting and welcome development stands poised to open the door of reliable and powerful audio mixing to millions of budget-challenged recordists. While the pairing of dedicated DSP cards and hardware-independent native software applications seems oxymoronic, it is not as far-fetched (or far from the horizon) as you may think.

Certainly, native processing is the most available and least expensive entrance to the world of personal computer-based recording, but its limitations are reached quickly. Despite the increasing quality and input/output prowess of the latest multichannel interfaces, an important piece of the puzzle is missing. With the help of the dedicated converters and processing on the interface, we can now get many channels of high resolution audio to and from the computer. What we can do with the audio once it is in the computer is the weak link of native applications.

Native uprising

Try setting up a 24-track session of 24-bit/48 kHz audio (96 kHz is right out) on a native platform application. Add a little equalization on every channel, and a couple of modulation effects and reverbs on effects busses. It is not long before the internal processor maxes out and the session grinds to a halt. Don't even think about putting that multiband compressor across the stereo bus.

Other problems, even more insidious than session-ending crashes, exist with native processing. As CPU power approaches maximum use, other application system calls (especially from the operating system) can cause your audio to slow down or lurch ahead. Audio playback buffers cannot be kept filled, resulting in dropped samples, clicks and pops. The erratic performance can range from imperceptible to glaring and can seriously undermine your morale, or worse, your client's confidence.

Now, with several manufacturers vying for this potentially huge market, help is on the way. One of the early entrants to the native/DSP marriage is one you would never suspect: Universal Audio — the same Universal Audio who made news the last several years by reissuing exact reproductions of the vintage analog Teletronix LA-2A and UREI 1176LN (see my review on page 23 of this issue) compressors.

> The marriage of platform-independent DSP cards to native applications is sure to be a happy one.

Late last year, Universal Audio purchased TDM and MAS plug-in developer Kind of Loud Technologies. At the winter NAMM 2001 convention, Universal Audio debuted its new conceptual product called Powered Plug-Ins. The Powered Plug-Ins package consists of a DSP PCI card and a variety of card-enabled plug-ins including 5-band parametric EQ, 1176LN and LA-2A compressor emulations, mod/delay, room simulation and the acclaimed Kind of Loud RealVerb Pro.

The idea is simple: pop the dedicated DSP card into your computer (Mac or PC), launch your favorite audio application, and generously slather multitudes of plug-ins across your tracks — all without increasing the load on your CPU. The plug-ins appear in the same list as your native plug-ins, but are programmed to access the card, not your CPU. The beta demo I played with was a Cubase VST session that had 15 four-band parametric Eqs, 2 1176LN compressors and 4 RealVerbs running simultaneously, all without affecting the CPU. The UA card was only 48% in use, with room to double the above plug-in list.

Look for several other notable plug-in developers already on board to announce support for the UA Powered Plug-in card, with more likely to follow. There will almost certainly be card competition as well, with several other independent DSP cards already in development.

The people's tools

This month marks the debut of a new column called *Pro Tools User*. It is not really a column in the proper sense because it will take on various shapes: reviews, user profiles, expert tips and question and answer sessions. Third-party software releases and hardware peripherals will be regularly featured, as will software updates, issues and news.

To keep a fresh perspective, *Pro Tools User* will be written by a variety of authors, guest columnists and end users. It will spotlight the platform's use in the broad pro audio segments covered in ***Pro Audio Review***, including broadcast production, studio recording, multimedia production, location recording and audio-for-video and film.

Pro Tools User is designed to address numerous reader requests for increased coverage — and corral our existing regular coverage — of this near-ubiquitous platform into a home base. This month, regular ***PAR*** contributor and resident Pro Tools expert J. Arif Verner reviews the Access Virus TDM plug-in, and wraps up the Pro Tools scene at the 2001 Winter NAMM convention (see page 46).

IS THE ROAD TO YOUR MIX PAVED WITH GOOD INTENTIONS?

Choosing the right audio mixer can be mighty confusing. Especially with all the hype and exaggerated product claims, whizzing by at the speed of sound these days.

ELIMINATE THE CONFUSION.

Perfect for live sound, Peavey's new RQ 2300™ series mixers boast more XLRs than *any* product in their class. Plain and simple, that means you'll get the most inputs anywhere, for your hard-earned mixer dollar! There are three models to choose from. Each has a uniquely different channel configuration to meet any need, with great EQ and flexible signal routing.

QUALITY?

You bet! The full-featured RQ 2300's sound sparkling clean. Their high headroom bus structure uses each channel's full dynamic range and all of the fader's available gain, to yield pristine signal reproduction. And those high quality discrete mic preamps are amazing!

WHY TAKE OUR WORD FOR IT?

Well, for one thing, Peavey has been designing, building and supplying the world with complete sound systems for more than 30 years. Fact is, we've been satisfying customers for a heck of a lot longer than many of our "spin doctor" competitors have actually been in business!

So, don't let any silver tongued hypesters lead you down the wrong road. Head on down to your local Peavey dealer today and hear the new RQ 2300 series for yourself.

Available in:

RQ 2310

RQ 2314

RQ 2318 (shown)

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Rolls CL151 GLC Compressor/Limiter Gate

The CL151 GLC is a single-channel compressor/limiter, with an added noise gate and microphone preamp. It has balanced XLR and unbalanced 1/4-inch inputs, a TRS side chain and an unbalanced 1/4-inch output. The XLR input features jumper-switchable phantom power. The unit uses soft-knee compression circuitry with a threshold and ration control. Gate threshold and release time control sets gate parameters. Five LEDs indicate gain reduction level and a gate LED shows whether signal is gated or being passed. The GLC is the first of a modular system designed to be mounted side-by-side in a rack tray. Subsequent units from Rolls will include a mic preamp, equalizer, crossover, sonic exciter and low wattage power amp. Price: $120.

Contact: Rolls at 801-263-9053; *www.rolls.com*; or circle **Reader Service 44.**

Sound Enhancements AccuVerb Spring Reverb

Sound Enhancements' Accuverb is a studio-quality, tube-driven spring reverb housed in a double rack space, roadworthy enclosure. A warm sounding alternative to digital reverb units, AccuVerb uses custom, dual three-spring Accutronics reverbs. Equipped with 12AU7A tubes, the Accuverb can also be used as a tube preamplifier to add warmth to recordings. The unit is low noise with custom reverb shielding, low emission toroidal transformer and discrete plus and minus rectification. Monitored dual inputs and outputs allow for mono and/or stereo setups. Price: $1,399.

Contact: Sound Enhancements, Inc. at 800-284-5172; *www.accuverb.com*; or circle **Reader Service 45.**

Azden 200R VHF Receiver

The new 200R receiver from Azden is compatible with the company's numerous transmitters, including the 31LT bodypack/lavalier, the 31HT handheld, the 31XT plug-in and the 31IT instrument bodypack. The 200R is packaged in a 5.55 inch by 5.1 inch, by 0.95 inch, lightweight ABS case. It operates between 169 MHz and 198 MHz. The front panel of the 200R has an on/off switch and separate volume controls. The rear panel has a standard 1/4-inch line-level output jack, telescopic antenna and AC power input jack (using the standard BC-26U power supply). Price: $110 (packaged as a system with the 31IT instrument bodypack: $190).

Contact: Azden at 516-328-7500; or circle **Reader Service 46.**

MBHO Modular Microphone System

Eleven different exchangeable capsules are now available to use with MBHO's range of microphone preamps, such as the MBNM 603, MBNM 680 or MBNM 648. The capsules include omni, cardioid, wide cardioid, cardioid/speech optimized, hypercardioid, figure eight, large-diaphragm cardioid and large-diaphragm omni. With many capsules to choose from, the MBHO modular microphone system offers versatility for both live and studio applications. All electrical contacts of the capsules and the preamps are gold sputtered to be resistant to aging and corrosion. Price: Mic capsules: $237 to $728; preamps: $264-$492.

Contact: MBHO at 718-963-2777; *www.mbho.de*; or circle **Reader Service 47.**

continued on page 12

13,000 WATTS AT 85 LBS.

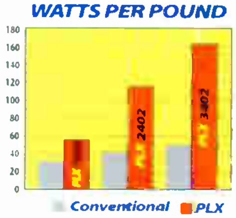
Four PLX 3402s produce over 13,000 watts while weighing less than 85 lbs.

You know QSC's 30-year reputation for quality. That's why QSC is the amplifier of choice for the world's leading touring companies. As an experienced pro, you also know exactly what you want in an amp—superb sonic quality, high power, and exceptional efficiency—all in a compact chassis without the weight. You're ready for the PLX Series.

Bone-Rattling Power Without the Bone-Jarring Weight

PLX amplifiers deliver up to 3400 watts—plenty of pure, clean power to drive your biggest subs. Power for bass-thumping lows and crystal-clear highs that will give your system new life. The best part is, all this power comes in a compact 2-rack space chassis only 13" deep and a nimble 21 pounds. Compare that to the 70 lb., 3-rack high, and 18" deep specs of conventional amps. With PLX, your energy goes into the performance, not setting up.

WATTS PER POUND



Applying State-Of-The-Art Technology to the Art of Music

How do PLX amps deliver so much power in such a small package? With our exclusive PowerWave™ power supply technology. It not only reduces weight and space, but eliminates hum and noise, delivering the full dynamic range needed for today's digital audio.

It Draws an Audience Without Drawing Much Current

The efficient Class H output circuit on the PLX 2402, 3002 and 3402 reduces AC current draw and cooling requirements by more than 40%. In fact, the PLX 3402 only draws 12 amps (120 VAC) during normal operation when loaded at 4 ohms per channel (compared to 20 amps for conventional Class AB output circuit). No heavy-duty circuit requirements. PLX runs on standard 15-amp circuit breakers.

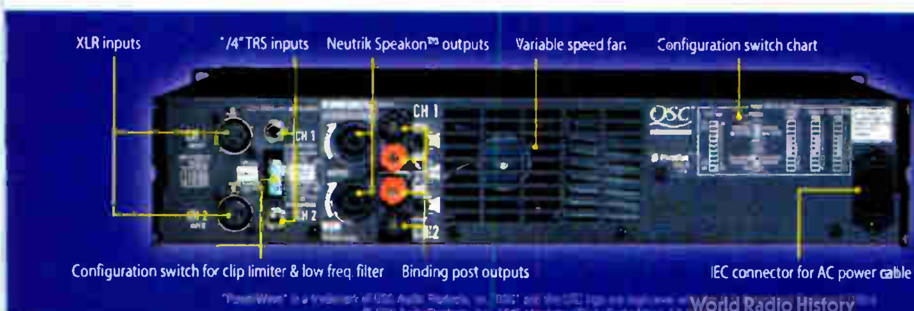
AC POWER DRAW

High power, quality, efficiency—PLX really racks up benefits without racking up weight. Visit a QSC dealer today or log onto www.qscaudio.com to learn more about the PLX Series.

PLX Series	8Ω	4Ω	2Ω	New Lower MSRP
PLX 1202	200	325	600	$838.00
PLX 1602	300	500	800	$978.00
PLX 2402	425	700	1200	$1,258.00
PLX 3002	550	900	1500	$1,398.00
PLX 3402	700	1100	1700	$1,638.00

8Ω 20 Hz–20 kHz 0.03% THD 4Ω 20 Hz–20 kHz, 0.05% THD 2Ω 1 KHz 1% THD

For more information about the PLX Series, call (800) 854-4079 or visit our website at www.qscaudio.com/plx/par.html



Rear view of the PLX 3402

POWERWAVE

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HEAR THE POWER OF TECHNOLOGY.

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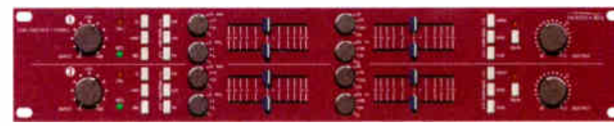
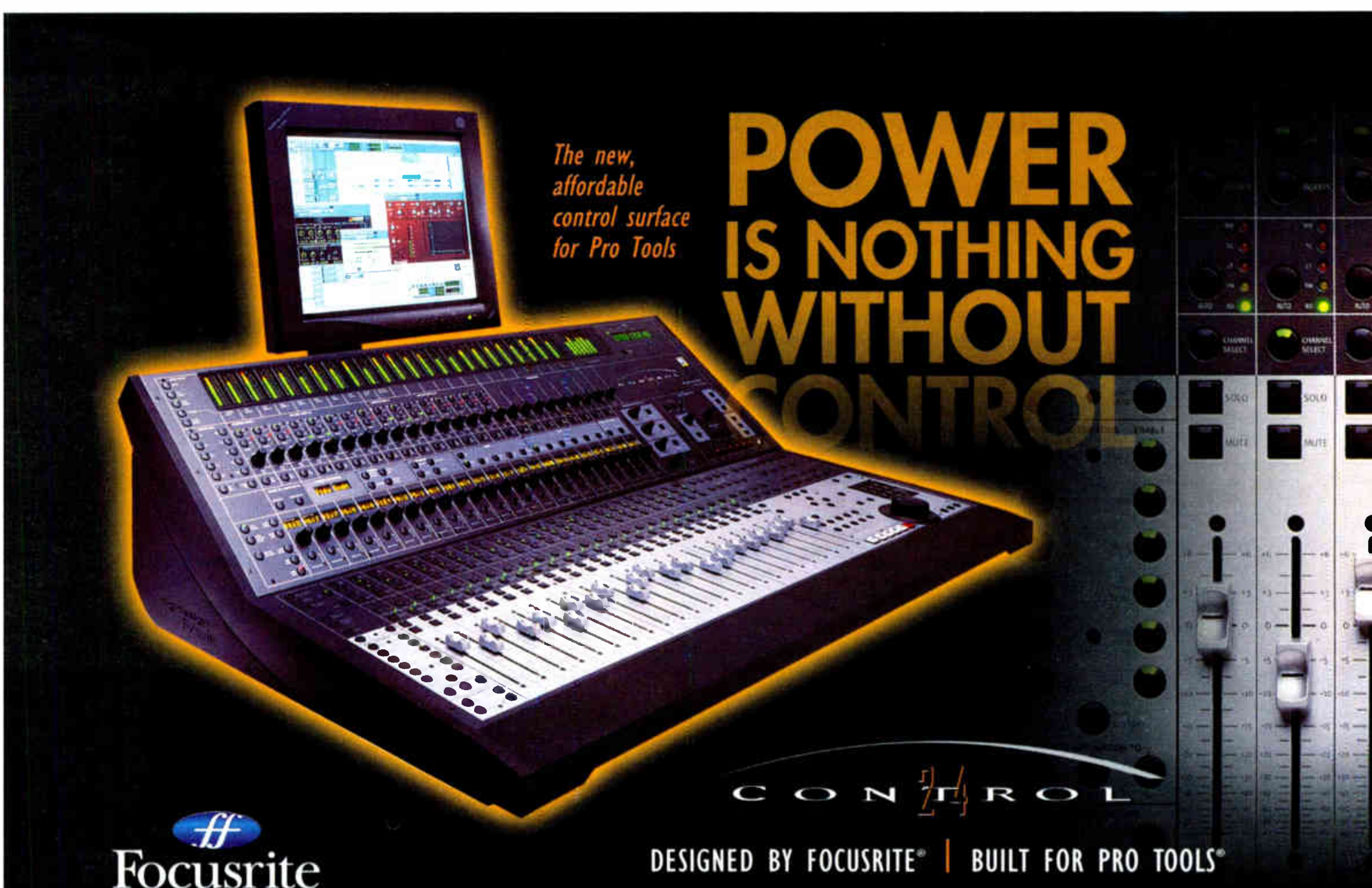
Sennheiser MKH 800 Microphone

Sennheiser's latest studio mic for the high end market, the MKH 800, is a multipattern microphone with numerous features. Patterns consist of omnidirectional, cardioid, wide cardioid, supercardioid and figure eight. Onboard are a 6/12 dB pad, a switchable (3/6 dB) low-cut filter (50 Hz) and a switchable (3/6 dB) presence boost (10 kHz). Price: $2,950.
Contact: Sennheiser at 860-434-9190; or **circle Reader Service 48**.

Trident-MTA A Series Channel Strip



Bringing the past back again, Malcolm Toft has resurrected his Trident A Series channel strip. Based closely upon the original Trident A Series console of the early 1970, the new A series features a preamp, EQ and low and high-pass filtering. The preamp includes 48V phantom power and phase reverse. The four-band EQ is inductive and can be bypassed. The low-pass filter is at 9, 12 or 15 kHz while the high-pass filter offers 25, 50 and 100 Hz as settings. Price: $3,699.
Contact: Trident-MTA/PMI Audio at 877-563-6335; or circle **Reader Service 49**.



Wisdom seeks wisdom

I enjoyed Stephen Murphy's piece "On Vocals: the KSM44" in ***Pro Audio Review*** (01/01; p. 14). He obviously has had vast experience working with a variety of mics and studio gear. Which leads me to a question I'm sure Murphy has been asked many times — what is a great (condenser) mic and mic preamp combination to use strictly for voiceover production? I'm searching for clean, full, intimate sounds with loads of depth and natural highs. Mainly male voice, but the occasional female voice as well. I would like to keep the cost around $2,000 for the package.

Wisdom Media would be very grateful for any wisdom Mr. Murphy can share.

Ken Dietz
Wisdom Media Group/Wisdom Radio

Editor Stephen Murphy responds:

Thanks for the complement on my wisdom, although the phrase "time served" might be more accurate. As you could tell, I really liked the Shure KSM-44 and have no trouble recommending it for your application.

I enlisted the help of PAR contributor Ty Ford for additional suggestions:

Check out the Neumann TLM103 ($995), AKG's new C4500B ($665), A-T 4047 ($695), A-T 4050 ($995) and the Sennheiser MD441U ($895). For preamp/vocal processor combos, try the DBX 376 ($599), Drawmer MX60 ($699), Focusrite MH400 ($675), the new Symetrix 628 ($1,249), Joemeek VC1Q ($799) and the Rane VP 12 ($599). If any other readers have more suggestions for Ken, send them on in!

Embraceable u

A comment made in Edward Foster's fine December piece on the decibel ("Audio Tech 101," 12/00, p. 38) prompts me to ask the following: Does the "u" in dBu stand for unterminated or unreferenced?

Oliver Berliner; *via e-mail*

Edward Foster responds:

Neither. In fact, it shouldn't really be a u at all but the lower-case Greek mu, which looks like a u if you take off the leading tail. Many books and magazines substitute u for the proper symbol because some printing fonts don't support the Greek alphabet.

Why mu? Because voltage-wise, 0 dBu is equivalent to 0 dBm, assuming dBm is used properly, which often it is not. dBm is a power reference; dBu a voltage reference. dBm indicates 'dB referenced to 1 milliwatt on a 600-ohm line' and was concocted by the phone company, which used 600-ohm lines. On a 600-ohm line, 0 dBm is equivalent to 0.7746 V and that voltage serves as the 0 reference for dBu. Some people miss the subtlety and use dBm for voltage but it is only cricket to speak of dBm in terms of power. dBu is the proper term for voltage and the two are equivalent only if the impedance is 600 ohms.

Care to opine?

Send letters to:
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Falls Church, VA 22041
Or e-mail par@imaspub.com

D.W. FEARN VT-4 LC Equalizer

by Dr. Frederick J. Bashour

The equalization curves obtainable through the classic inductor/capacitor circuitry used in the old passive Pultec and Lang units have a special sound quality all their own, primarily because they are not "ideal" curves. This distinction has not been lost on modern manufacturers of pro audio equipment. Manley Labs and Tube-Tech have each introduced neo-Pultec equalizers.

Manley Labs owns the rights to the Pultec name and has produced several different models that faithfully copy the classic units' passive circuitry, while adding extra EQ points and additional bandwidth choices as well as Manley's own makeup gain tube amplifiers.

Doug Fearn, on the other hand, runs a company that manufactures just a few specialized tube products. Completely handmade, Fearn's creations roll off the assembly line at the rate of about six per month, and are gobbled up as fast as they are produced. Before introducing his equalizer, it seems as though Doug Fearn waited patiently to see what every other manufacturer would do before concocting a unit that combines the best features of all the others.

Fearn also added his own special sonic circuitry and such custom touches as the same type of 1/4-inch aluminum front panel as used in the VT-1 and VT-2 mic preamps, finished with the same red DuPont IMRON paint found on classic motorcycles.

Completely handmade, Doug Fearn's creations roll off the assembly line at a rate of six per month.

Features

The VT-4 ($3,900) is a single-channel equalizer spaciously built into a three-rack-space chassis that uses passive LC circuitry with Class A triode vacuum tube stages for its input and output amplifiers. Although employing four Svetlana 6N1P ultralow-noise triodes, the tube circuitry itself is largely derived from that which is used in the VT-1 and VT-2 mic preamps, ensuring a similar sonic quality. The sound is deep, resonant, present, smooth and creamy.

The input is line level, transformer balanced-bridging, but the tube input section has a stepped adjustable gain control that accommodates input signals as low as -10 dBm. The output stage uses the same custom Jensen transformer that is used in the mic preamps. The power transformer, and the inductors, are also by Jensen.

All controls are stepped for precise repeatability and uniform matching between units. The same high-caliber parts employed in Fearn's mic preamps are used here: silver contact rotary switches, one percent metal film resistors, and polystyrene and polypropylene capacitors.

Here's a listing of all the VT-4's EQ points and their adjustable parameters:

Low-cut frequency adjustments are at 30, 40, 100 or 400 Hz: 0 to -18 dB shelving in 2 dB steps. Low-boost frequencies are 20, 40, 60 or 140 Hz: 0 to +12 dB shelving in 2 dB steps. Mid-cut frequency choices are 200, 300, 400, 500, 600 or 700 Hz: 0 to -18 dB in 2 dB steps. High frequencies can be boosted at 2, 3, 4, 5, 8, 10, 12 or 16 kHz: 0 to +14 dB in 2 dB steps, while highs can be cut at 1.7, 4, 10 or 28 kHz: 0 to -14 dB shelving in 2 dB steps.

There are five settings for high-frequency bandwidth (Q): 0.6, 0.8, 1.0, 1.4 or 1.7. Gain is adjustable in 3 dB steps from -9 to +9 dB. The balanced input and output XLRs and I.E.C. power connector and its

At a Glance

Applications:
Studio and remote recording; mixing and mastering

Key Features:
Pultec-type LC passive equalization with an extended range of adjustment parameters; Class A triode vacuum tube electronics; stepped rotary switches

Price:
$3,900

Contact:
D.W. Fearn at 610-793-2526; *www.dwfearn.com*; or circle Reader Service 82.

continued on page 16

READER SERVICE NUMBER 167

DW Fearn continued from page 14

associated on-off toggle switch are on the rear panel.

In use

So, what does it sound like? Remarkably similar to Fearn's own VT-2! Throughout my listening tests, I auditioned solo instrument tracks (including piano, voice, violin, cello, oboe, bass clarinet and flute) and full mixes of my own classical productions as well as representative pop and jazz CDs.

The VT-4 is a special and unique equalizer. In most instances, its stepped switches made subtle, but noticeable changes which were extremely helpful as problem solvers and sonic enhancers. I very much enjoyed the ease of having a finite set of adjustable parameters again.

There's definitely something to be said for the ease and repeatability of stepped EQ switches — and not just for mastering purposes either. The controls all seem to be set up such that, as they are advanced further away from flat, the relative action between switch positions becomes greater in subjective effect — either sharper (Q-wise) or stronger, depending on the particular control.

It was helpful to have the subtle changes occurring in the +2 and +4 positions; sometimes all I ever want is just a smidgen of EQ modification. The VT-4 makes such adjustments easy.

I won't go into the wonderful EQ curves one can make by using the cut and boost controls simultaneously; if you use a Pultec-type equalizer, you'll know what I mean. I will single out, however, the 28 kHz cut-shelving control. I've never experienced this feature before, but immediately understood exactly what Doug Fearn meant when he wrote in the manual: "The highest frequency setting is particularly useful for digital recording. Attenuation in this range may help avoid effects common with anti-aliasing filters in A/D converters." Indeed, judicious use of this control makes very little apparent difference to the overall high end sound of the program material (and remember, I can still hear the 15.75 kHz local oscillator in beat-up old TV sets), but using it certainly makes a difference in the end result when one records using, say, the built-in converters in a DAT recorder.

I discovered this phenomenon about 15 years ago, when I first began using tube microphones on classical recording sessions. I learned that the typical ADCs of that era made mincemeat out of close-miked, spiky loud brass or high soprano vocal sounds, as picked up by typical solid-state microphones.

Darned if those old tube mics, on the other hand, didn't slow down the signal just enough so as not to completely freak out the converters. I concluded it had something to do with the converters' inability to sample spiky transient distortion products.

Now we have an equalizer with the ability to prepare bright, transient-rich program material for safe passage through the analog-to-digital conversion process.

I then made a subjective noise measurement — by cranking my monitor gain up all the way — and then switching back and forth between the Fearn box and various other level-matched pieces of tube and solid state line level gear in my studio. The VT-4 employs four 6N1Ps, an amazingly quiet new vacuum tube, with just a smidgen of extremely low level hiss, and no buzz at all. I've never encountered this tube before, but the bottom line is the Fearn unit was substantially quieter than any other tube equalizer I own; it might even be the quietest piece of tube gear in my control room!

In my reviews, I usually compare the sounds of competing units whenever I can. In this case, the closest equivalents to the D.W. Fearn VT-4 were the Manley mastering versions of its EQP-1A and Mid-Frequency Equalizers or, to a certain extent, the EQ section of the VoxBox. In fact, the VT-4 could be accurately characterized as competitive with all the Manley units combined; in other words, its circuit topology is based upon features taken from both standard and mid-frequency Pultec models.

First I listened to full mixes and compared the sound of the Manley and Fearn units as line amps with their EQ sections bypassed, to see if the sonic differences I'd noticed several years ago between the two companies' mic preamps were also present in their equalizers. The answer is yes. The largest difference between the Manley "sound" and the Fearn VT-4's sound was in the midrange and high-end quality, although it was a bit difficult to qualify. The VT-4 sounded a bit more "real," while the Manley sounded more "high fidelity" on both mixes and individual instrument tracks. Fearn's lows sounded "plummier" and more resonant, while the Manleys were tighter and a little deeper.

I could usually make the Fearn and my Manley boxes equalize my various sound sources in a similar manner (at least to the extent to which their respective adjustable parameters were similar), but the overall character of their sound quality remained. I could always tell which unit I was using. On some sources I preferred the EQ effect from the Manley, while on others I preferred the Fearn's.

Summary

I highly recommend this unit. If you want an easy-to-use equalizer in the classic style, which does its thing smoothly and sounds luscious and creamy, the VT-4 is for you. Its price may seem to be a little on the high side at first glance, but once you realize that this handmade box does the work of both standard and mid-frequency equalizers — and as a whole lot more — the cost makes perfect sense.

*Dr. Fred Bashour is a jazz pianist, church organist, classical music producer/engineer, intermittent college professor, consultant to university music libraries on the digital storage of course listening materials and a **Pro Audio Review** contributor.*

Product Points

D.W. Fearn VT-4 LC Equalizer

Plus
- Luscious sound
- Quiet and dependable operation
- EQ bands from both standard and midrange classic passive models

Minus
- High price
- Single-channel only

The Score
One of the best modern adaptations of passive circuitry.

READER SERVICE NUMBER 168

Sabine Graphi-Q 3102 Digital EQ/Procesor

by Andrew Roberts

On the Bench, Page 20

Recent technological advances have introduced a group of processors that can perform a wide range of functions, preserve user presets and consume only a fraction of the space of their combined, single-function predecessors. The Sabine Graphi-Q series of processors offers multiple functions, takes up minimal space and, at $1,299.95 (full front panel controls), lowers the price threshold for this expanding class of processors.

Features

The unit I received for evaluation, the GRQ-3102, is one of four models in the Graphi-Q line. The GRQ-3102 is a dual-channel unit with faceplate controls; the GRQ-3102S ($799.95; blank front panel) is the related slave version without front panel controls. The 3101 ($1,099.95; full front panel) and 3101S ($699.95; blank front panel) are the comparable single-channel versions.

All units in the series come with GRQ remote software for Windows. With the exception of the two-rackspace-high 3102, which needs the space to accommodate its dual 31-band graphic EQs, the Graphi-Q units are all one rackspace tall. The 3102 weighs nine pounds and is nine inches deep.

The GRQ-3102 has a full arsenal of loudspeaker-related processing. It has two 31-band graphic EQs, a full-function dual-channel compressor/limiter, two digital delays, high and low shelving filters and two channels of Sabine's FBX Feedback Exterminator. Each channel of FBX contains 12 separate, automatically placed, narrowly attenuated parametric filters that hunt down and reduce feedback when it occurs.

Analog aficionados will find great comfort in the 3102's traditional-style controls. The two 31-band EQs have sliders that offer a choice of either 6 or 12 dB of cut/boost. The front panel also has high- and low-cut filters (3 kHz to 20 kHz and 20 Hz to 1 kHz), compressor function controls (ratio, threshold and gain), a compressor LED display (level and gain reduction), delay time adjustment, bypass buttons (delay, FBX and EQ), and a group of control switches for the FBX section with LEDs that indicate filter status.

At a Glance

Applications:
Live sound; contracting

Key Features:
Dual graphic EQs; dual compressor/limiters; digital delays; 24 FBX feedback filters

Price:
$1,249

Contact:
Contact: Sabine at 904-418-2000; *www.sabine.com*; or circle Reader Service 89.

The 3102 also has two small LCD-type displays that Sabine calls "Tweak-n-Peek" windows. In default setting, these displays show the current delay time setting. When any front panel control is moved, however, that channel's window displays the parameter value of the control being tweaked. For example, if you start to push up a graphic EQ slider from its 0 dB detent, you will immediately see that gain increase show up in the display window as a +dB setting.

The rear panel on the 3102 is home to balanced XLR inputs and outputs, as well as TRS 1/4-inch balanced in and outs. There are also two RS-232 terminals, one to connect to a PC or laptop controller and the other to chain additional units together. The 3102 also has a Phoenix block connector for contact closure switching.

The FBX section features 12 parametric filters that can be divided into two categories: fixed and dynamic. The fixed filters will not change the frequency of a particular filter notch once they have latched onto a feedback cycle. They will deepen the filter notch if additional feedback is detected at that frequency, unless you depress the Lock-Fixed button in the FBX section. The dynamic filters hunt for new feedback frequencies as they occur. The factory default is nine fixed and three dynamic; this ratio can be adjusted to your liking.

Setting the FBX filters can be accomplished by performing a short process once your sound system is set up and ready to go. With master sends down, bypass all gates and press and hold the reset button to erase any previous filters. Program the ratio of fixed vs. dynamic filters by pressing and holding the Set-Fixed button until you have the desired number of fixed filters. Set the filter width (default is a constant Q of 1/10 octave) and then slowly raise the system gain.

As feedback rings occur, the 3102 detects these frequencies and pulls them down with fixed filters. Once the fixed filters are assigned, the dynamic filters begin to engage as more feedback rings occur. As the system settings and room acoustics change, the dynamic filters latch onto new frequencies as they occur. Additionally, the Graphi-Q has a Manual Turbo Mode and an Auto Turbo Mode, which allow for fast and quiet FBX Filter setup.

The Graphi-Q Remote Software allows you to remotely control all of the 3102's front panel controls and much more. Using the software controller embellishes most of the front panel control parameters. For

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Sabine continued from page 18

instance, not only can you set the FBX filters, you can adjust their depth and width. You can also trade FBX filters for fully parametric EQs. It is also possible to globally adjust the filter width of each band of the graphic EQ and view and edit the response curve. You can also extend the power of the 3102's digital delays by using the GRQ software. From the digital delay page, adjust the delay increments by milliseconds, feet or meters — very handy for quickly setting up delay speakers in a large venue.

The 3102 can also incorporate ambient air temperature into the delay equation. Since the speed of sound is affected by temperature, this can be a valuable parameter adjustment — especially when working outdoors. Compressor/limiter adjustments are expanded. Using the software allows for adjustment of attack, release, knee and limiter threshold in addition to the ratio, threshold and gain parameters that appear on the front panel. The GRQ software allows the user to store up to 69 presets, control and link up to 16 channels of Graphi-Q, and set passwords for security purposes.

In use

My evaluation of the GRQ-3102 consisted of about a half-dozen sound reinforcement jobs, most of which included a full band and challenging acoustic environments. I used the 3102 to process both front-of-house (FOH) speakers and monitor mixes.

Using the 3102 sans software was initially easy, thanks to its analog-style controls. It quickly became clear, however, that the 3102 is quite complicated: when I was not using the GRQ software I had to rely on the manual quite a lot. Many functions the unit performs require pressing buttons that seemingly have no connection to a given task. For example, to change the graphic EQ range from 6 to 12 dB, you simultaneously hold down the delay up/down buttons.

Setting up the FBX filters requires holding down certain buttons for varying amounts of time. Due to the time that passed between shows, it was hard to remember the procedure for these adjustments. Consequently, I found myself reaching for the manual frequently, despite the presence of the analog-style controls. Fortunately, the 3102's manual is very concise and it proved helpful when using the Graphi-Q in variety of situations. Despite a few idiosyncrasies, I found the 3102 to be quite effective.

As long as there was adequate time for setup during sound check, the 3102 helped me wring a little more volume out of the monitors — a big plus when working with loud bands on an ambient stage. I particularly liked the 3102's graphic EQ. It was great having two ranges from which to choose. In one venue, the stage was acting

> **Despite a few idiosyncrasies, I found the 3102 to be quite effective.**

as a bass resonator and there was a lot of 200-Hz rumble in the monitors. Pulling the 200-Hz slider down at -6 dB on the monitor EQ helped somewhat, but when I switched the range to -12 dB, I was able to remove most of the unsavory boominess.

The dynamics processor in the 3102 worked well when used in tandem with my laptop running the GRQ software. It seemed a tad less transparent than my analog compressor, however, when used to lightly compress the FOH mix. It worked great as a limiter on the monitors.

I found the dynamics LEDs on the 3102's faceplate difficult to read. The LED lamps have a lot of bleed from one light to the next, so it was hard to properly gauge gain reduction and compressor level.

It was nice having analog-style controls to adjust parameters on the 3102. It expedited the setup process since there were no menu pages to scroll through. Since many of the controls were multitasking, however, it made things a bit trickier in the beginning. It was sometimes difficult to get a precise adjustment out of the miniature pots used on the compressor. When setting the ratio, the display would often jump past my intended destination with the slightest turn. For this reason, I began using the GRQ software, which offered very precise adjustments.

I encountered one problem with the Graphi-Q that I think Sabine should address. At one show, I had to power off the system for a couple of minutes. When I powered the system back up, feedback ensued.

Initially, I thought my Graphi-Q settings had been dumped, but what was happening was the FBX filters were not being applied while the unit was booting up. If Sabine could add an output mute during the power-up process this problem would be eliminated. If not, there should be some indication that the unit is booting up. This would have alerted me to the problem and allowed me to preemptively stop it, thus keeping my blood pressure down a bit.

Summary

The Sabine Graphi-Q 3102 is a very powerful processing tool that is a remarkably good deal. It is well designed and constructed despite a few minor irritants, some of which may be mitigated with future software upgrades (which, by the way, are free). With powerful EQ, good dynamics processing, a unique feedback reduction system and the power of comprehensive external control, the 3102 should be a welcome addition to most professional sound systems.

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

Product Points

Sabine Graphi-Q 3102 Digital EQ

Plus
- Powerful and compact
- Increases system volume without feedback
- Analog-style controls

Minus
- Hard to read display
- Confusing control procedures
- No power-up mute

The Score
A powerful tool to fight feedback and improve system sound.

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On The Bench

Sabine Graphi-Q 3102 Digital EQ Bench Measurement

Sabine's GRQ-3102 may be small, but it is loaded with technical goodies that warrant extensive bench testing. It took more than the usual number of graphs to do this product justice.

I concentrated on what I took to be the GRQ-3102's main features: 31-band graphic equalization spaced every one-third-octave on the ISO-standard frequencies, high-and low-cut filters adjustable over a wide range of frequencies, and adjustable-threshold/adjustable-ratio compression.

Since the GRQ-3102 works entirely in the digital domain, I ran an extensive set of tests on its converters, including THD+N vs. frequency and level, quantization noise, dynamic range, linearity error, etc. To say the results were impressive would be an understatement!

Take linearity error, shown in **Figure 1** (see p. 100 for graphs). It is a mere 0.13 dB to -90 dB FS and barely worse than 0.2 dB at -100 dB FS! A word about what "dB FS" means in this case — usually I'd define 0 dB FS for a device like this at one-half or one decibel below input clipping, but input clipping occurred at so high a level on this device (+29.2 dBu) that I defined 0 dB FS as +24 dBu since that had the virtue of being a professional reference point. Should you have a source that approached +29 dBu in level, the GRQ-3102's performance would actually exceed my tabulated figures! By the way, the clip indicator comes on at +28 dBu so you are given adequate warning of incipient overload.

THD+N at +24 dBu (my 0 dB FS) is plotted in **Figure 2** while THD+N vs. level (at 1 kHz) is shown in Figure 3. (In the first three figures, the solid line corresponds to Channel A and the broken line to Channel B.) As you can see in Figure 2, THD+N is practically uniform across the spectrum, which suggests it is probably more noise than distortion.

The absence of peaks in the high-frequency region is particularly impressive because it indicates that the GRQ-3102's digital filtering is adequate to prevent cross-modulation with the carrier. Between -40 and -50 dB FS (-16 to -26 dBu), THD+N drops sharply by 12 to 13 dB. Typically, such changes occur at a higher level than this, but that is of no significance in this case because the THD+N is so minuscule even at +24 dBu (my 0 dB FS). The scale in Figure 3 has been shifted down 10 dB to show the superb performance of this unit.

The dynamic range data — almost 109 dB, A-weighted; better than 106 dB, unweighted; and nearly 100 dB, CCIR-weighted — go right along with the distortion data of **Figure 3**. Quantization noise is an impressively low -91.6 dB FS and residual noise figures are almost identical with the dynamic range data, which implies that the GRQ-3102 generates essentially no distortion and introduces negligible amounts of noise. **Figure 4** shows the noise spectrum of Channel A as the solid line and the spectrum analysis of the dynamic range measurement signal (1 kHz at -60 dB FS) as the broken curve.

Okay, now that we've demonstrated that Sabine's A/D and D/A converters are pretty impressive, what does the GRQ-3102 manage to do with them? Well, it can make some pretty perfect-looking equalization curves as you can see in **Figure 5**, which shows the response of the 1 kHz equalizer set for 3, 6, 9 and 12 dB of boost and cut.

I used an expanded frequency scale (300 to 3 kHz) so you can see how classic the curves are and how well they peak to exactly the amount of boost or cut that the display indicates.

Figure 6 shows the response of four of the other sections (20 Hz, 125 Hz, 5 kHz and 20 kHz) with maximum boost and cut. These curves don't look quite so pretty, but that's because of the measurement technique, not because of any imperfection in the GRQ-3102. Audio Precision test equipment takes measurements at discrete frequencies and although I take many more data points than most researchers do, there's no guarantee that a data point will lie exactly on the center frequency of each equalizer section. By the way, Figures 5 and 6 were taken using the +/-12 dB range mode; the GRQ-3102 also offers a less aggressive mode, which affects a +/-6 dB range at the extreme settings. I verified that this worked as claimed but didn't take a full set of response curves for publication.

Figures 7 and **8** show the response characteristics of the low- and high-cut filters respectively. You get many more choices of cutoff frequency than are shown; I set each control at its lowest and highest frequencies and then took data at two intermediate points. As you can see, the -3 dB points of each filter are extremely close to the indicated frequencies.

Figure 9 shows the response of the two channels with the equalizer bypassed. Note the sensitive vertical scale (0.2 dB per division), how flat the response is (down less than 0.3 dB at 20 kHz) and how well balanced the channels are (within +/-0.06 dB).

Response with the equalizer active and all controls set at their respective detents was precisely the same, provided the front-panel controls were properly calibrated. The sliders on the GRQ-3102 look like those of a conventional analog equalizer, but they don't control the signal directly as with an analog equalizer. The GRQ-3102 controls send instructions to a DSP that does the equalization in the digital domain and it is necessary to calibrate the controls so response is flat when at the detent.

Part way through the test sequence, the 800 Hz equalizer on my sample acted up and produced a slight boost in response. Recalibrating fixed the problem and it never recurred. While I chalk this up as a fluke, I have an obligation to report any problem that occurs. The last two curves show the compressor characteristics under quasi-steady-state conditions. **Figure 10** shows the effect of the ratio control with the threshold control set to zero. The solid (topmost) curve corresponds to a 1:1 setting, i.e., no compression, while the bottom corresponds to a 5:1 compression ratio. **Figure 11** shows the effect of the threshold control with the ratio control set for infinite (hard) compression. The top curve corresponds to a threshold setting of +20 dB, i.e. linear operation up to quite high input levels, while the lowest curve corresponds to a -10 dB setting. At that setting the GRQ-3102 is in hard limiting by -24 dB on the scale. For these curves, 0 dB corresponds to +24 dBu so -24 dB, is 0 dBu.

Everything seems fine as far as interfacing the GRQ-3102 with other equipment is concerned. As stated earlier, the input is virtually overload-proof and the clip indicator warns of potential problems. Output impedance is a low 52 ohms; input impedance a high 25 k-ohms. The device can supply gobs and gobs of signal (in excess of +30 dBu) and its gain control is very well calibrated.

As far as I'm concerned, the Sabine GRQ-3102 is a technological masterpiece!

— Edward J. Foster

Universal Audio
1176LN Solid State Limiter

by Stephen Murphy

When I was selling the studio I owned and operated for nearly 10 years, I was determined not to take much equipment with me. This was for two reasons: I wanted to leave the studio's microphone and outboard collection intact; I was also looking forward to wiping the slate clean and starting over with my personal production studio.

There were only two pieces of gear on my "items that do not convey with sale" list: one quality mic (I even let the new owner choose from the Neumann, AKG and Sony microphones) and one UREI 1176LN Limiter. He gave me the mic (Neumann U87), but would not budge on the 1176.

Needless to say, when I saw Universal Audio — now resurrected by founder/designer Bill Putnam's sons, William and Jim — was back in business cranking out near-exact reissues of the 1176LN, I jumped at the chance to review one.

Many revered engineers find the 1176's unique and expressive character indispensable, and I could not agree more. It should be noted there is the occasional engineer who thinks the 1176LN is overrated and overvalued. There must have been something wrong with their unit.

At a Glance

Applications:
Studio; live sound; post production

Key Features:
Mono FET compressor/limiter; XLR and barrier strip balanced I/O; 4:1 to 20:1 ratios; variable attack and release; 115- and 230-Volt operation.

Price:
$2,295

Contact:
Contact: Universal Audio at 831-454-0630; *www.uaudio.com;* or circle **Reader Service 83.**

Features

The Universal Audio 1176LN ($2,295) is an accurate reproduction of the original 1176 D/E revision, with the exception of new XLR I/O added to compliment the original barrier strip connectors. The 1176LN is a mono FET compressor/limiter with a fixed threshold. This means you turn up the input knob until you get the desired amount of gain reduction and adjust the output knob to compensate — couldn't be simpler.

The 1176LN also features an attack knob — which doubles as the off (bypass) switch when turned fully counter clockwise — a release knob and a large, illuminated VU meter.

Attack time ranges from an ultra-quick 20 microseconds up to 800 microseconds, and release times range from 50 milliseconds to 1.1 seconds. With the attack knob in the off position, the FET compression stage is taken out of the path, but the input and output amplifier stages remain in line. The 1176LN essentially becomes a direct line amplifier—many engineers use the unit in this mode solely for the unique character the amplifiers add to the signal.

Speaking of signal path, the 1176LN employs mic-level transformer and input amp stages, a FET gain reduction circuit, a UA 1108 preamplifier output stage and a custom Putnam-designed output transformer. The VU meter can display output level

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Universal Audio continued from page 23

referenced to either +4 dBm or +8 dBm, or it can be set to display gain reduction (RMS).

Four push buttons set the compression ratio; available settings are 4:1, 8:1, 12:1, and 20:1. Of course, you could push all four buttons in at the same time and see what you get (kind of an 1176 rite of passage — everyone must discover it on their own). A small hole for the recessed meter calibration pot and a power button complete the front panel feature set.

The back panel of the 1176LN has XLR I/O as well as the original balanced Jones barrier strip connections. As with the original 1176LN, connections are provided for an external VU meter and an 1176 SA stereo adapter ($99), for running two units linked for stereo. A removable IEC A/C cable brings power to the unit, which can be configured for 115- and 230-Volt operation.

In use

When I used to explain to people at my studio why I liked the 1176LN, one of the things I told them was that it is the best tracking compressor/limiter. Not "tracking" as in "recording tracks." What I meant was, with proper attack and release settings, the 1176 tracks the incoming signal incredibly well. You can be listening to a bass or vocal track, for instance, that sounds smooth and effortless with nary a hint of compression in use. Then you look over at the 1176LN's VU meter dancing away — 1/2 dB here, 4 dB there and so on — and know who is putting in the real effort.

I am pleased to report that the new model performs as well as my old favorite. For this review, I used the UA 1176LN on a number of different instruments and vocalists over several months with predictable success.

My favorite use for the 1176LN is for vocals, electric and upright basses, and other "single line" instruments. I usually stick to the 4:1 ratio, with medium attack and reasonably quick release — one of my pet peeve sounds is that of a compressor coming back up with a sluggish release. This was never an issue with the 1176LN.

For most recordings, I set the compressor so it does not engage until the louder bits (i.e. no constant compression) with a maximum gain reduction of -4 dB. This is the area where the operation of the 1176LN is most transparent, imperceptibly reigning in the audio while adding its characteristic tone.

To give you an idea of "the 1176 sound," various engineers have referred to it as edge, growl, present and urgent. Generally speaking, the more level you send through the unit, the more those descriptive terms come into play.

The unit can also be overdriven to great effect. The best way to do this is to take the compressor out of line (attack knob fully CCW), crank up the input and lower the output — actually, you had better lower the output first and then crank up the input! Of course, you can also leave the FET in, set the ratio to 20:1 and do some hardcore limiting for a different kind of overdrive. This sounds great on drum and percussion room microphones.

Summary

Over the last 30-plus years, the original 1176LN earned legions of dedicated, almost cult-like fans. Why? It goes beyond its simple operation and unique sound. It is one of those tools of our trade that you can make a shape with; it becomes an extension of your creativity. Like its predecessors, the Universal Audio 1176LN reissue is easy to use and sounds great. I am proud to count myself as a loyal, dues-paying member of the cult of the 1176.

*Stephen Murphy is the editor of **Pro Audio Review**. As a recording engineer/producer, Murphy has worked on many successful audio and video productions, including Platinum and Grammy-award winning recordings.*

Product Points

Universal Audio
1176LN Solid State Amplifier

Plus
- It's back!
- Ease of use
- Classic 1176LN character

Minus
- Pricey for a single channel

The Score
The welcome return of an old favorite — the great sounding and easy to use 1176LN Limiting Amplifier.

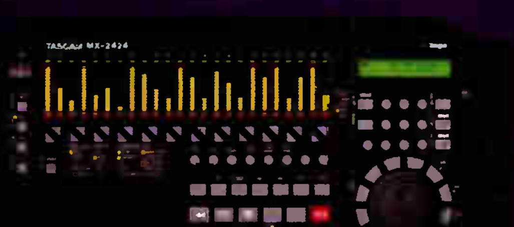
MX-2424 Profile:
Rudi Ekstein of Foxfire Recording

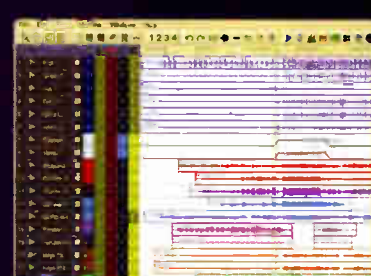
Rudi Ekstein may not be a household name. But his studio, Foxfire Recording, has been thriving for over ten years, with over 40 hours of bookings every week. And the new cornerstone of Foxfire is the TASCAM **MX-2424 24-Track 24-Bit Hard Disk Recorder.**

When you can have any recording system you want, why pick the MX-2424? "After looking at other hard disk multitracks, I chose the MX-2424 based upon its incredible versatility," says Rudi. "First and foremost, the MX has fantastic sound quality that is comparable to anything I've ever heard. The ability to use 24 channels of analog and digital I/O simultaneously was another big reason for my decision. Plus, the ability to edit from the front panel, to easily set locate points and to use the auto-punch and scrub features have helped make sessions run smoother and quicker."

With audio file format and disk drive compatibility with your favorite DAW systems on Mac® and PC, easy interfacing with popular analog and digital gear and all the advantages of our world-class hard disk engine, it's easy to see why thousands of musicians, project studios and professional facilities like Foxfire have chosen the MX-2424 for their main recording system. For the complete MX-2424 story, see www.tascam.com or visit your TASCAM retailer. You never know...the next MX-2424 profile could be yours.

MX-2424 24-TRACK 24-BIT HARD DISK RECORDER/EDITOR





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READER SERVICE NUMBER 176

Panasonic
AD96/AD96M Analog-to-Digital Converters

by Stephen Murphy

On the Bench, page 28

For the past year, Panasonic's professional audio division conspicuously disappeared, leaving many in the industry scratching their heads. Panasonic discontinued its line of venerable DAT recorders, had little presence at the major audio shows and, for a time, its Web site disappeared — all under the guise of internal restructuring.

While all this was going on, however, Panasonic's Professional Audio Group was definitely not in hibernation. At the winter NAMM 2001 show, Panasonic made a strong and welcome return to the pro audio world, debuting the next generation of the popular Panasonic DA7 digital console, the DA7 mk II.

Also getting notice at the show were Panasonic's new multichannel, high-resolution analog-to-digital converters — the AD96 and AD96M. A digital-to-analog model, the DA96, will be released late spring 2001.

Features

The AD96 ($2,195) and AD96M ($2,495) are eight channel 24-bit/96 kHz Delta-Sigma converters aimed squarely at the growing high-bit/high-sampling rate recording market. The two models are identical, save the addition of eight low-noise microphone preamps on the AD96M. The front panel real estate used by the additional preamp input knobs and phantom power switches on the AD96M resulted in a more basic metering system than the comprehensive one found on the AD96.

Each channel on the AD96 features a 10-point bar graph meter with two view levels: Normal and Zoom. The Normal (coarse) view ranges from Peak to -54 dB in large steps, and the Zoom view ranges from Peak to -30 dB in small increments. A rear panel DIP switch sets the dB level for the Peak level indicator (0, -0.5, -1, -2, -3, -4, -5 or -6 dB FS) and the user-definable Reference level indicator.

The Reference indicator is the third point from the top of the 10-point bar meter and can be set to light at –20, -18, -16 or -14 dB FS. In addition, the AD96 features a Peak Hold function with three settings: off, 2-second hold and infinite hold. Line input level adjustments are made on front-panel trimpots using a mini screwdriver.

The channel metering on the AD96M consists of a single multicolor LED with three level stages indicated by color. Green indicates signal present (above -38 dB FS); amber indicates signal at the user-definable reference level (same settings as the AD96 above); red indicates signal at the user-definable peak level (same settings as the AD96 above).

The AD96M has no peak hold or zoom functions. Mic/line input level adjustments are made on the eight front panel knobs, continuously adjustable from +4 to -60 dB.

From here on out, features described pertain to both the AD96 and the AD96M. Analog input is via eight rear panel Neutrik XLR connectors. Digital audio is output to four male XLR connectors and one ADAT optical connector. The XLR outputs can be configured to eight channels of standard/high-speed single-wire AES/EBU audio (two channels per XLR) or four channels of the newer dual-wire AES/EBU specification (one channel per XLR).

An optional WZ-AESAD card ($399) can be added to access all eight channels of dual-wire AES/EBU data via a D-sub connector. An optional WZ-TDIAD TDIF digital output card ($399) is also available. Use of either optional card requires removing the factory-standard XLR AES/EBU output module. The ADAT optical output remains available

continued on page 28

At a Glance

Applications:
Studio; project studio; location recording; mastering

Key Features:
Eight-channel analog-to-digital converters; eight-channels of mic preamps on the AD96M; 128x over-sampling Delta-Sigma 24-bit/96 kHz converters; four AES/EBU XLR connectors and an ADAT optical output standard; single-wire or dual-wire AES/EBU operation; optional D-sub AES/EBU and TDIF output cards available.

Price:
AD96: $2,195; AD96M: $2,495

Contact:
Panasonic at 714-373-7200; or circle **Reader Service 60**.

Panasonic continued from page 26

and independent regardless of optional output card choice.

The units are capable of outputting eight channels of 16-, 20- or 24-bit digital audio at 44.1, 48, 88.2 or 96 kHz through AES/EBU connectors. The ADAT optical connector (or optional TDIF card) outputs either eight channels at 44.1 or 48 kHz or four channels at 88.2 or 96 kHz (each channel uses two tracks on the recorder at the higher rates).

BNC word clock in and thru/out (user-selectable) connections are found on the rear panel. The units can lock to incoming clock signals at frequencies ranging from 44.1 kHz to 96 kHz +/- 6 percent.

In use

With our bench tester, Bascom King, handling the objective testing (see bench test on p. 28), I put the two AD models through their subjective evaluation.

I used the two converters on a variety of projects for the last several months with excellent results. Although I had a few complaints, these units provide high-quality audio and flexibility at a reasonable price.

First I used the AD96 to transfer tracks from a two-inch analog deck into Pro Tools. This was achieved easily and without reference to the manual. Tracks came straight off the analog deck and into the XLR inputs on the converter, and the resulting digital audio was outputted at 24-bit 48 kHz on the four AES/EBU outputs in single-wire mode.

I also dumped analog tracks straight into an ADAT XT-20 using the AD96's ADAT optical port sending 20-bit 48 kHz audio. Again, this operation was easily negotiated and the resulting tracks sounded noticeably clearer and more detailed than corresponding tracks recorded through the ADAT's internal A/D converters. I love it when you can actually hear a difference!

I did use the high-sample rate ADAT optical mode to send four channels of 96-kHz audio to eight tracks on the XT-20 with apparent success. Unfortunately, without the corresponding DA96 to test, I had no way of playing back these tracks. I also recorded two channels of 24-bit/96 kHz audio to my computer using an AES/EBU output on the AD96 running into an RME digital audio card, again without any hitches or glitches.

I recorded a variety of sessions using the AD96M and its eight internal mic preamps. These sessions included tracking acoustic and electric guitars, bass, vocals, keyboards and percussion. After careful level adjustment, all tracks went to tape (and hard disk) flawlessly.

While not quite as sonically pleasing as other dedicated (and expensive) mic preamps available for comparison, the AD96M held its own, beating out the internal preamps on a Spirit digital and

continued on page 29

On The Bench

Panasonic AD96/AD96M Analog-to-Digital Converters Bench Measurement

Panasonic's A/D converters come in two flavors, the AD96 with line level inputs measured here, and the AD96M with microphone level inputs. The unit behaved and measured very well on the bench.

Frequency response at the four sampling frequencies available is shown plotted in **Figure** 1 (see p. 90 for graphs). The horizontal scale in the figure is linear and only goes down to 10 kHz to best illustrate the high frequency response characteristics. Not shown is the low frequency behavior which is essentially flat down to 20 Hz. Square wave response shape is that of typical FIR digital filtering having symmetrical ringing about a vertical center line through the middle of each half cycle.

Distortion behavior of the AD96 is such that there is some real amount of distortion near full scale, and in fact, the unit cannot get up to 0 dB FS without starting to clip on one half cycle. This phenomenon, however, is gone with input levels down only 0.1 dB from full scale.

This is illustrated in **Figure** 2 in a plot of distortion of a 1 kHz test tone vs. input level relative to 0 dB FS for Channels 1 and 2. **Figure** 3 is a plot of total harmonic distortion vs. frequency and level for the 44.1 kHz sampling frequency. Results are similar for the higher frequencies but with a higher noise floor due to the wider bandwidth. The levels shown are -0.1, -1, -5 and -10 dB FS. The dominant harmonic was the third.

Deviation from linearity was quite good for this converter and was very similar between the channels and at all sampling frequencies. An example of this is plotted in **Figure 4** for channels one and two done at 96.0 kHz f_S.

Channel separation was very good for the AD96. Results were essentially independent of sampling frequency except that the Channel 2 to Channel 1 direction at the 96.0 kHz sampling frequency, for some reason, was worse. This is shown plotted in **Figure 5** for both directions at 96.0 kHz f_S. All other channel separation measurements were more or less like the Channel 1 to Channel 2 direction in the figure.

— Bascom H. King

Bench test continued on page 90

Panasonic continued from page 27

MCI analog mixer. Relatively speaking, at roughly $300/channel including high resolution A/D converter, these mic preamps sound good indeed!

While the two units performed well—producing high-sample rate/high-resolution digital audio with a great degree of options and flexibility — a few omissions in features and ergonomics should be mentioned. First, the lack of external inserts between the mic preamps and converters limits the usefulness of the M model. There were certain times when some compression on the way in to the converter would have resulted in a better signal to tape (and corresponding higher resolution).

This, of course, was especially true of dynamic instruments where initial transients shoot out some 20 dB over the average signal (a timbale and trumpet track come to mind). In these cases, an external mic pre was run into a limiter and then into the AD96M. At this price point, inserts, or defeatable peak limiters on each channel, are more wish list items than complaints, as they would add significant cost to the units.

Also, the addition of a peak hold function on the M model would be extremely useful. After using and relying on the peak hold function of the AD96, its omission on the AD96M model was all the more obvious. The single, tricolor LED for metering does not allow this option, but there is sufficient space for the addition of another dedicated peak indicator (hint, hint).

Summary

Overall, the Panasonic AD96 and AD96M are excellent values and incredibly easy to use. The high-resolution audio they produce is attributable to solid construction, quality parts and innovative circuit design; the resulting performance of these units is hard to fault. One may want to opt for the AD96 and use external microphone preamps if the lack of inserts is an issue — otherwise the low-noise mic preamps on the M model are a sonic bargain.

Product Points

Panasonic
AD96/AD96M
Analog-to-Digital Converters

Plus
- Sound quality
- Excellent value per channel
- Flexible output modes

Minus
- No insert after mic pre (AD96M)
- No peak hold function (AD96M)

The Score
An excellent value in multi-channel high-sample rate/high-resolution A/D converters.

C-Audio
Pulse Series Power Amplifier

by Will James

Founded in Cambridge, England in 1980, C-Audio (now owned by Harman Int'l. Inc.) is a leading power amplifier distributor throughout Europe and Asia. In addition to the Pulse series power amps reviewed here, the C-Audio family includes the SRX series & the GB series.

Features

The Pulse series power amps are unusual and attractive looking. The units I checked out were the Pulse 4 x 300 and the Pulse 2 x 1100 (C-Audio also makes a 2 x 650 Watt model). The most striking feature is the attractive slate-blue faceplate, which also forms its interesting handles.

At a Glance

Applications:
Live sound; contractor; installation

Key Features:
4-channel amplifier (4 x 300 Watts); 2-channel amplifier (2 x 1100 Watts); switching power supply; XLR and 1/4 inch TRS inputs; Neutrik NL4 and binding post outputs

Price:
2 x 650: $1,495; 4 x 300: $1,650; 2 x 1100: $1,850

Contact:
BSS Audio at 615-360-0277; *www.caudiousa@harman.com*; or circle **Reader Service 84.**

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PROFESSIONAL RECORDING EQUIPMENT

MANLEY

Langevin

COMPANY PROFILE: Manley Laboratories, Inc. in recent years has expanded and thrived under EveAnna Manley's leadership. Our 11,000 sq. ft. building houses our own machine-shop, printed circuit board manufacture, audio transformer winding, engraving, and silk-screening facilities. All custom design, R&D, assembly, testing, and quality control processes are performed with precision and pride at the Manley factory, located just 35 miles east of Los Angeles.

NEO-CLASSIC: MANLEY ALL-TUBE GEAR
We take a purist approach to everything we build; refining, executing, and expanding upon Manley's legacy of vacuum tube design philosophies proven over years of real-world experience, using high quality modern components, many of which are fabricated in-house. This attention to detail delivers the rich, present, and natural sound our vacuum tube designs are renowned for. Never small, sterile, or boring.

Beyond this, Manley means reliability, real technical support, and a company attitude that professionals depend on.

BIG BANG FOR THE BUCK: LANGEVIN
LANGEVIN is a legendary marque of premium electronics whose lineage goes back to World War II. **MANLEY** acquired the **LANGEVIN** brand name several years ago. With these products we offer you the different sonic flavor that **ALL-DISCRETE CLASS A CIRCUITRY** brings using fresh, original designs built alongside and to the same exacting standards as the Manley equipment.

We believe that good music and those who create it deserve the finest gear.

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C-Audio continued from page 30

The front panels of the two-rackspace power amps are concise and to the point. There are two rotary level controls on the two-channel version and four rotary level controls on the four-channel unit. Both versions have a rocker style on/off switch, as well as four LED function readouts that display main power on, level activity, protect and remote operation. Flanking the controls and LEDs are two 4-inch cooling fans, one on each side.

The rear panel features the balanced input connections, which are XLR and TRS 1/4-inch in parallel on the four-channel version, and a TRS coupled to both a male and female XLR connector on the two-channel version. For output, there are both Neutrik NL4s and Neutrik binding post connectors on the two channel models, and your choice of one or the other aforementioned connectors on the four-channel model. The outputs of the power amps may be bridged via a recessed switch (bridge in pairs on the four-channel).

The internal workings feature variable speed cooling fans (blowing rear-to-front), controlled by an onboard microprocessor that constantly monitors for possible faults in the switched mode power supply and/or power amp output sections.

The switching-style power supply provides more consistent power for longer periods. BSS provides the following specs: Model 4 x 300: each of four channels driven-170 W RMS at 8 ohms, 300W RMS at 4 ohms, 330 W RMS at 2 ohms; Model 2 x 650: both channels driven 400 W RMS at 8 ohms, 650 W RMS at 4 ohms, 850 W RMS at 2 ohms; Model 2 x 1100: both channels driven 700 W RMS at 8 ohms, 1100 W RMS at 4 ohms, 1,500 W RMS at 2 ohms. Frequency response is 20 Hz to 20 kHz, and the slew rate is 50 Volts/microsecond.

Each of the models weighs about 24 lbs. Available accessories for the C-Audio amps include fan filters, internal crossovers, rear-rack ears, and input transformers. Also available are two accessory cards. The CP212 Anti-Clip Limiter Card provides limiting for two or four-channel amps. The CP211 contractor card adds a variety of contracting features to a two-channel amp.

Product Points

C-Audio Pulse Series Power Amplifiers

Plus
- Lightweight, easy-to-handle
- Quick response for accurate sound

Minus
- Internal crossover is optional

The Score
Clean audio from these lightweight, high-powered touring amplifiers.

In use

At Sedona, Arizona's Jazz on the Rocks festival, which featured nouveau Latin-African Jazz ensemble Los Hombres Calientes, I used the C-Audio 2 x 1100 Model with a pair of Yorkville TX8 double 15-inch, full-range, three-way speaker cabinets, and the 4 x 300 Model biamping a pair of JBL 4704 monitors on two different mixes.

The full-range mains responded with excellent clarity and dexterity. The 2 x 1100 model stayed cool throughout the entire show. The four-channel amp running the monitors handled the mixes containing heavy amounts of full-range percussion, upright bass and grand piano.

For the second performance, I used the four-channel amp to power the 8-inch cone mids and the 2-inch compression tweeters in the Yorkville TX8s. The results were equally good; the amp had plenty of headroom.

Summary

These C-Audio power amplifiers are high quality professional amplifiers. They are easy to use and stay cool. Thanks to their lightweight switching power supply, several of them can be racked without being too cumbersome.

Their internal components are good quality and overall construction is excellent.

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

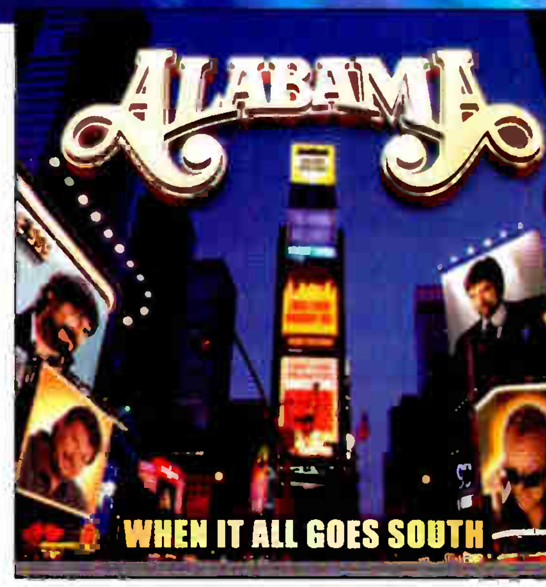
World Radio History

READER SERVICE NUMBER 178

Single Slice

by Chuck Taylor

Alabama's "When it All Goes South"



Album: When It All Goes South

Date Recorded: April 2000

Engineer: Mike Bradley

Previous Projects: Brooks & Dunn, Dolly Parton, Kentucky Headhunters, Olivia Newton-John, Wade Hayes, The Mavericks and Alabama

Single Producers: Don Cook, Alabama

Single Songwriters: J. Jarvis, R. Carnes, J. Carnes

Studio: Soundshop Studios, Nashville

Mastering Engineer: Hank Williams; Mastermix Studios, Nashville

Instruments: The track "When It All Goes South" was cut with a basic rhythm section consisting of bass drums, piano, acoustic and electric guitars. Considerable overdubbing transpired over several months, with sampled effects, a banjo, additional keyboards and guitars, and a drum loop

Console: Trident Vector

Microphones: An AKG C12 tube mic was utilized for the lead vocal; AKG 421s and Shure SM57s for electric guitars and much of the percussion; Neumann Tube 67s, Electro-Voice CS15s and Audio-Technica 4053s were used in a variety of capacities; a B&K 4051 was used as a center room mic.

Microphone Preamplifiers: Demeter on a variety of tracks, Avalon on the bass, Summit tube preamps for vocals, and Trident console preamps for electric guitars.

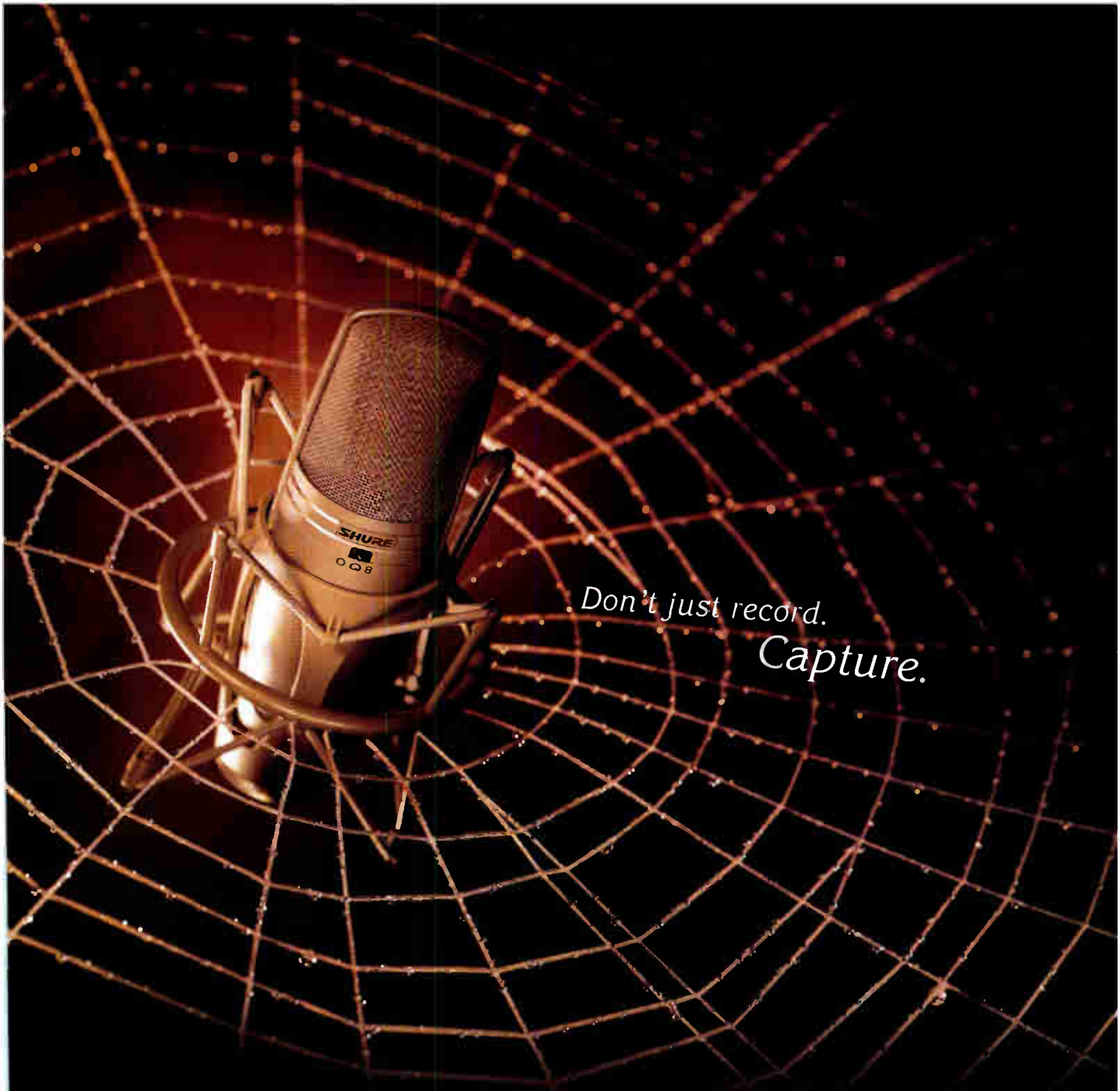
Processors: EQ and compression were about the only processing applied to the recording including a GML EQ, a TC Finalizer and a Tube Tech compressor.

Engineer Mike Bradley

Engineer's Diary

Alabama's "When It All Goes South" may have started out as a straightforward little country song, but by the time the enduring quartet — 20 years together now with more than 60 million units sold — was finished, "I can't tell you how many mixes we'd been through," says the record's engineer Mike Bradley. "When we started out, we kept things pretty simple, but then Randy Owens, Alabama lead vocalist, wanted to make it a little more dance-oriented." As a result, Bradley considerably beefed up the lower end of the song and juiced it up with a loop. Then three electric guitar players and a pianist came into the studio and added more layers, along with a harp solo. Next, the guys took the song on the road, resulting in more changes, including a trio of female background singers added at the song's end. "It just kept growing," Bradley says with a laugh. "There are a ton of different versions of that song; the actual album version is about five minutes long. We'd edit it down and then it would get longer again."

Chuck Taylor, a regular contributor to ***Pro Audio Review***, *is senior writer for* ***Billboard*** *in New York.*



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READER SERVICE NUMBER 179

Reinforcing Orchestras and Symphonies in Large Venues

by Tom Young

It used to be that orchestras were reinforced without using mics or perhaps by using a stereo pair of mics in acoustic concert halls. Orchestra members, being purists, are skeptical of a sound engineer's ability to properly amplify their performance.

Orchestras often perform in unfamiliar surroundings, sometimes with popular singers, to reach a larger audience and provide additional revenue for their programs. I recently engineered shows with the Portland Symphony in an arena, and last summer I engineered an outdoor show with the Buffalo Symphony in a baseball stadium. Enhancing orchestras in these types of venues requires a closer look at microphone choices and placement, close attention to the design of a proper sound reinforcement system, and even the effective use of monitors to help the musicians achieve a quality performance.

Natural reinforcement

The key to providing sound for a symphony is to offer a natural reinforcement of the performance that, in a large setting, still sounds acoustic in nature and not overly amplified. The main speaker system requires as much isolation as possible to the stage to eliminate 80 or 90 open microphones from unwanted feedback with the PA.

At a recent show in an arena, the PA was hung first, followed by stage installation. Upon inspecting the hang, I found the PA to be five feet on-stage. The sound system had to be dropped and re-rigged to provide the proper isolation. My preference when hanging an arena system is for the speaker clusters to be at least three to five feet in front of the stage. Ground-fill speakers can then be used to provide coverage for the seats closest to the stage.

When possible, I like to supplement the PA with delay speakers in order to distribute the sound around a large venue. The perception is not overly amplified coming off the stage from the main left and right clusters.

Using monitors effectively enhances the experience for the orchestra performers in arenas and outdoor venues. Hanging a pair of side fills in the far upstage corners of the stage at a fairly high trim of 12 to 15 feet, at a 15 to 20 degree down-angle, provides light reinforcement of the strings throughout the stage. With symphonies accustomed to working within an acoustic-enhancing orchestra shell, this type of monitoring provides life onstage and helps with the musicians' intonation when playing. Also, small monitors for certain sections — like percussion and brass — positioned very far upstage, help tighten up the sound.

Violinist using a Countryman Isomax microphone

Individual mics for tighter sound

When used effectively, microphones for individual musicians can provide a tighter sound so the orchestra does not sound too distant. Using contact microphones like the Countryman Isomax on the strings can provide nice natural sound and stable control when used in the monitors.

The most difficult obstacle to overcome is convincing the musicians that these types of microphones are necessary and that when placed will not damage prized wooden instruments. Both of these manufacturers have designed excellent mounting clips for placing on the strings above the bridge in order to position the microphone about three to four inches above an instrument.

Supplementing these microphones with area microphones in the string section provides natural life to the house mix. Condenser microphones, such as the Neumann KM-184, Sennheiser MKH-80 or Audio-Technica 4051, are my preferences for this type of application.

Most symphonies are not accustomed to playing into close microphones. When miking the brass and woodwinds overhead, section mics work well. Placement for an individual instrument at a distance of 12 to 18 inches from the instrument provides individual control and fairly good isolation. I have found that good condensers like the AKG 414, Neumann KM-184, Shure KSM series and Audio-Technica 4033 work well in these situations.

A challenging experience

Providing sound reinforcement for an orchestra or symphony in an arena or large outdoor venue is a challenging experience that comes with great musical rewards for the sound engineer. Nothing compares to listening to an 80-piece orchestra and the dynamics created in its musical performance — from the most quiet of passages, to the roar of a crescendo.

When building the mix for this type of show, I prefer to start from scratch, listening to the orchestra acoustically first. It is important when providing live sound in these situations to properly provide sound reinforcement for the performance and not alter the dynamics played by over-amplifying the show. With proper system design, monitoring, microphone selection and placement, the experience can enjoyable for all.

*Tom Young, a regular contributor to **Pro Audio Review**, is currently the live sound engineer for Tony Bennett.*

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JBL EON15 G2 Biamplified Speaker System

by David O'Brien

JBL Professional has been an innovator in the field of studio monitor and loudspeaker design for decades. Since its release in 1995, the company's EON series has become one of the most popular powered sound reinforcement speaker lines in the world, with more than 300,000 systems sold.

Features

The original EON15P, a biamplified, powered speaker system, featured a 15-inch speaker driven by a 130 W amp and a high-frequency horn driven by a 60 W amp in a lightweight polycarbonate enclosure. JBL has now released the upgraded version of this highly successful speaker, the EON15 G2 ($849).

The new EON improves on the input section by adding two balanced 1/4-inch TRS inputs with rotary trim pots to the existing mic/line switchable XLR input. There is a line level XLR output jack that is switchable between Mix (a mix of all three inputs) and Loop (only input one). A two-band equalizer section, providing 15 dB of boost or cut at 120 Hz and 5 kHz, is also included.

JBL has more than doubled the old power rating, bringing it to 300 W for the low-frequency driver and 100 W for the high-frequency driver. The new system comes in a compact, easy-to-carry package made of durable copolymer material. It weighs in at a lean 46 pounds — actually one pound lighter than the older model.

In use

The configuration of these speakers makes them exceptionally versatile. I used them on several jobs in a variety of capacities.

As keyboard monitors they were great. I was able to mix the keyboards in the EON15 G2 and send a single output to the main board. They sounded very full and the EQ provided ample control over the tone.

As vocal monitors they performed well, and they had no trouble providing enough volume to cover a particularly loud stage volume. The in-line graphic

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Community Professional Loudspeakers R.5-66 Speakers

by Will James

Traditionally, outdoor speakers have a rather thin sound and they are fairly unattractive. They sometimes attempt to reproduce a full-range program such as music, but they usually sound like an AM radio, representing a 1 to 3 kHz spectrum, at best. Community Sound has come up with an interesting solution to this problem in its new product — the R.5 speaker series.

Features

The R.5-66 ($599) is a full-range, two-way speaker system capable of handling 200 W RMS at 8 ohms. The R.5 has full-range capability presented in a rounded polyethylene enclosure — ensuring survival of the components in even the most extreme weather conditions.

The R.5 comes with a swivel-mount bracket for installation in any location or angle, in addition to a four-foot, two-conductor speaker lead. The R.5 contains a 12-inch woofer for the lower frequencies. The woofer has a voice coil immersed in a ferrofluid to assist in weatherproofing, heat dissipation and corrosion resistance.

Community recommends a 70-Hz high-pass filter — at least 12 dB per octave — to reduce lower frequencies that cause overexcursion of the woofer. To handle the high frequency duties, there is a one-inch weatherproof driver, mounted coaxially to the woofer and both mounted to the perforated metal grille. The grille is actually a three-layer affair, minimizing sun and water penetration through the use of a

continued on page 40

At a Glance

Applications:
Outdoor PA, contracting and fixed installation

Key Features:
Weatherproof; full-range coaxial speaker system; 12-inch woofer; linen HF driver

Price:
$599

Contact:
Community Sound at 610-876-3400; *www.loudspeakers.net*; or circle **Reader Service 86.**

JBL continued from page 38

EQ came in handy for combating feedback problems.

I also used them successfully for front-of-house at a live band event catering to 300 attendees. The EONs provided plenty of power, and setup and breakdown was a breeze. In this case, difficult room acoustics made use of additional equalization a must.

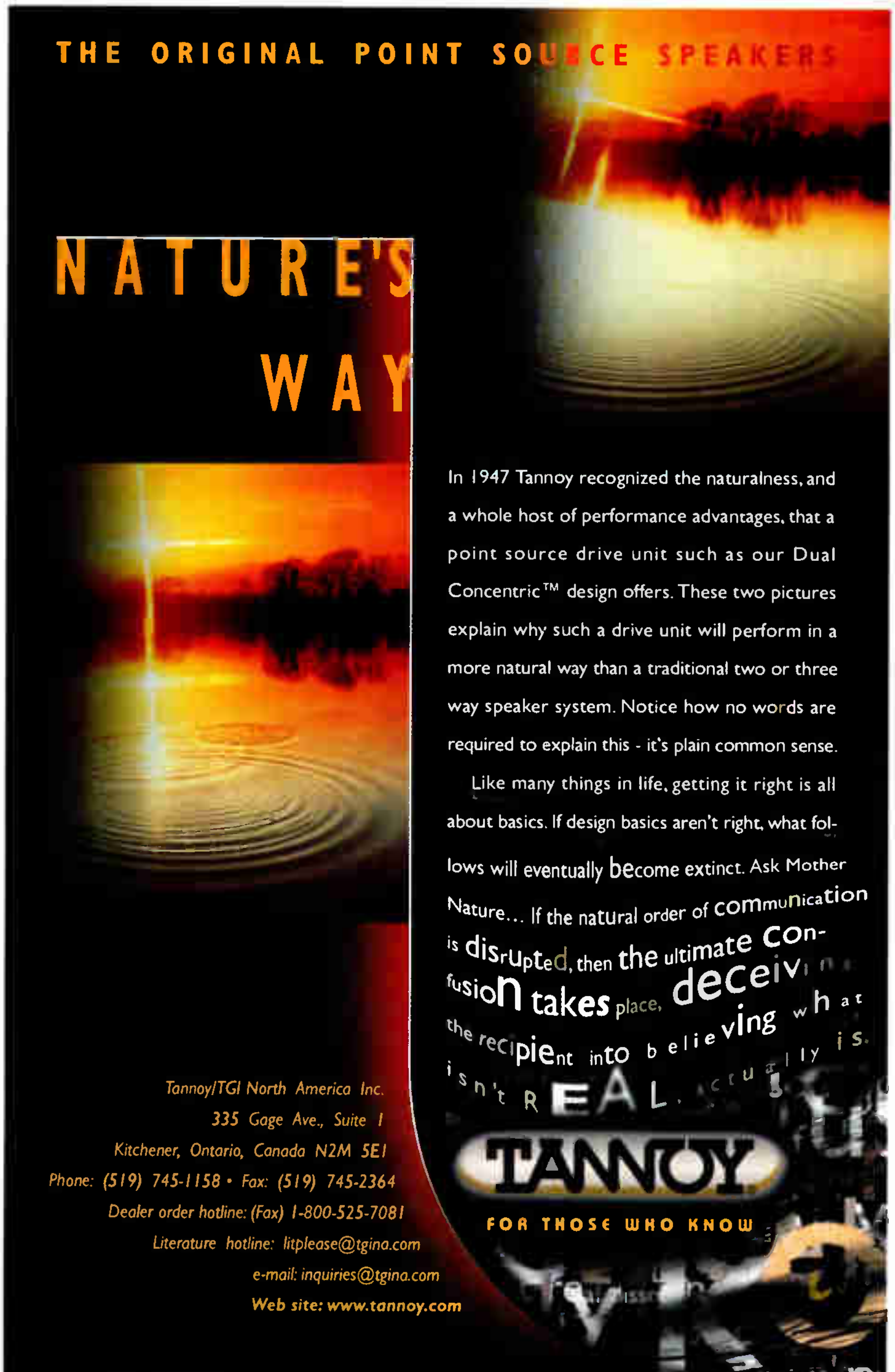
I was especially impressed by the method JBL has devised for cooling the amplifiers. The cooling fins are located in the speaker ports so that the speaker itself pushes air across them. The harder you drive the system, the more you cool the amp. This eliminates the need for any cooling fans and increases power efficiency.

Summary

Having the speaker and the power amp all in one unit is a valuable feature. Combine that with an easy-to-understand user interface, lightweight package, plenty of power and versatility of use and you've got a system that is sure to be every bit as popular as its predecessor.

Contact JBL Professional at 800-852-5776; 818-894-8850; *www.jblpro.com*; or circle **Reader Service 85**.

Dave O'Brien works on both sides of the microphone as a live sound engineer and bass player/vocalist. He is also a location sound engineer for radio, television and film productions.



Community continued from page 38

double powder-coated steel layer and fine stainless steel mesh, coupled with an acoustically transparent open cell insert.

In use

I wired the R.5-66 to a C-Audio 400 x 4 power amplifier, plugged a portable Sony CD player into a Spirit Notepad mixer, and wired the whole thing together at my warehouse. I did as Community suggested, and employed a high-pass filter (the Spirit mixer has a 100 Hz, 12dB/octave HPF).

I was impressed with the full-range response of the R.5. Considering the filter rolling out the lows, it had a very nice low end, and remarkably crisp highs. I ran the volume to a pretty good level and the R.5 responded quite well, with no audible clipping or flattening.

I tried the mounting mechanism — C-clamping the speaker to an overhead beam. The R.5 mounted and locked down nicely with no difficulty.

Summary

I found the Community R.5 a solid speaker system that will hold up well to extreme weather situations. In addition, the R.5 did have a sound far superior to the sound of traditional outdoor horns, which are generally nasal and grating to listen to.

The R.5 is the perfect speaker for theme parks, marine applications and wide spectrum weather situations. The enclosure appears to handle rain and direct sun extremely well, and the components are of good quality. Community earns my recommendation for its new R.5-66.

Electrix
Mo-FX Time Synchronized FX Effect Processor

by Alan R. Peterson

Electrix audio processors all share a very distinctive look, like a cross between a heat sink and a Scandinavian washing machine panel. It is what the engineers at Electrix put inside these cases, however, that makes them stand out.

Electrix has broken some interesting ground with the Mo-FX ($449), a time-synchronized multi-effect processor consisting of four effect stages — or blocks, as the manual calls them. There is a distortion stage, a flanger, a tremolo generator and a delay line, clocked via MIDI or a handy Tap Tempo button.

The addition of MIDI control frees the Mo-FX from being only a real-time performance device for remixing and deejaying. All parameters can be sent and received as MIDI Control Change data, allowing the device to function as part of a MIDI music production system. The Mo-FX is an 18-bit/44.1 kHz digital device, but it features classic analog-like effects.

Features

The rear panel contains 1/4-inch TRS and RCA inputs and outputs, a set of TRS insert jacks, MIDI in/out/thru jacks with a 16-position channel selector switch, a footswitch jack and a fused IEC power socket with on-off switch. A button on the back panel selects line or phono levels on the RCA inputs.

Features common to blocks on the front panel include three-band filtering on the Flange, Tremolo and Delay blocks, and large Momentary buttons on all four blocks. Each block can be set to respond to one or more frequency ranges by stepping through the Band buttons. By tapping the button, you may choose high, mid or low bands, or any combination of the three.

Pressing a Momentary button drops the selected effect into the line for as long as it is held down. This is a classic club deejay effect when used to accent a musical phrase or drum break. When a block is hard-switched in line (via the Engage buttons), the Momentary button works in reverse and bypasses the effect.

The Mo-FX excels when clocked internally or via MIDI clock signals. For the moment, I will concentrate on its live performance aspects.

Working from left to right, the Distortion/Insert stage comes first. Two dials control the overall level and amount of drive on this effect. If desired, the distorted signal can be used to feed the successive blocks; otherwise, the effects work in parallel. Here, too, is where an external device can be patched in on the insert jacks.

For all it has going for it, the Distortion effect on my demo unit had one problem: the level control did not affect any change until about the 9 o'clock position, or one-fifth through its travel.

The Flange block is thick and meaty, offering plenty of control over positive and negative depth, regeneration and flange speed. The Mix knob goes up to 100 percent level, then just a little beyond. According to the manual, an extra 9 dB of boost is available to lift the flanged material above the main mix.

When the Flange Speed knob is fully set counterclockwise, the flanger can be manually swept by the Depth control — a pretty hip feature to beat-match flange accents during live performance. The Band button lets you pick the frequency range you want flanged.

The Sync button ties the Flange block to the Tap Tempo button or to MIDI clock messages — more on synchronization shortly.

The Tremolo block modulates the amplitude (volume) for a nifty old-time, surf-guitar, Silvertone amp tremelo sound. Again, the Mix control tacks on an extra 9 dB if needed, and the Band button selects the frequency range you want modulated. The Wave control dials in one of seven tremolo contours, from an unadorned sine wave to a 12 percent duty-cycle square wave for a choppy effect.

Don't neglect the Auto-Pan button on the Tremolo block. This creates a wide stereo panoramic effect that sounds great in monitor speakers or on the club floor.

Expect some fun with the Delay block. When engaged, it delays (or stores) up to 3.3 seconds of audio. Using the Regeneration knob, the Delay block can play back a single delay hit, a repetitive loop or anything in between. With the P-Pong button locked in, the stored audio ping-pongs from left to right.

Giving the Speed dial a spin during looped playback causes "time smear," warping and bending the playback pitch in a manner similar to working the delay lever on the old Echoplex box. A slice of music or even a spoken-word recording can be bent into a wild new effect by grabbing and holding a sample in looped playback.

By the way, speeds are marked as ratios compared to the sync clock rather than as arbitrary numbers or measurements in

continued on page 43

At a Glance

Applications:
Music remixing; club DJ; live performance; studio and post production

Key Features:
Hands-on adjustment of all parameters; MIDI sync and control

Price:
$449

Contact:
Contact: Electrix/IVL Technologies at 250-544-4091; *www.electrixpro.com*; or circle **Reader Service 87.**

Electrix continued from page 42

milliseconds. A setting of 1:1 locks an effect to each beat, while a setting of 2:1 causes an effect to occur rhythmically twice in a beat, and so on. Flange and Tremolo settings range from 1:8 up to 8:1, while the Delay tops out at 4:1. The settings are at best an approximation, and the controls must be nudged to score an exact ratio on the fly.

To the far right can be found the Tap Tempo button (for rhythmic synchronization), the Bypass button and the Dry button, which provides a constant or interrupted feed-through of the original signal.

In use

The Tap Tempo button lays down the sync clock on the Mo-FX. Tapping the button at least twice to the beat of the music syncs up the flanger, tremolo and delay duration. Tapping more than twice refines the accuracy of the timing.

With an old Roland drum machine as the audio input, I got busy with the Tap Tempo button. True to form, all blocks synced up closely, but it was still necessary to tweak the speed dials to lock precisely onto the beat.

Said drum machine was then MIDI'd to the Mo-FX. The device latches onto MIDI clock messages after the Tap Tempo button is held down for 600 ms. The trouble is, the Mo-FX does not know where the downbeat is; no problem, though, as a single tap puts everything in step.

The Mo-FX shines over its fellow Electrix products when you tie its MIDI Out port to a sequencer. Each knob and button sends Control Change data that can be recorded into a sequence and will unfailingly repeat the same performance every time.

When running a sequenced mix from a computer, one or more tracks can be tapped from the multichannel audio interface and patched to the Mo-FX. The sequence data will "work the panel" on playback, modifying parameters exactly as they were entered.

The Delay block does not function as a triggered sampler, and sending Note Numbers or Velocity data does nothing for the Mo-FX. Your pitchbend lever, however, does affect the flange depth. And remember to alter the MIDI Channel rotary switch in the back if you reassign the Mo-FX's data to a new channel in the sequencer.

Summary

If you are a born button-pusher and derive pleasure in creating or remixing audio with your fingertips rather than a mouse, check out the Electrix Mo-FX. The real-time experience is fun and gratifying, and the ability to record all performance data is a creative lifesaver.

Some might wish for an onboard sample playback feature in the Delay block, but it is presumed the Mo-FX will be used as part of a greater system, and some means of sample playback already exists. I would have enjoyed a sample reversal feature to flop the playback of the Delay block's contents; there is nothing quite like playing instant satanic messages for the dance floor!

In a world where flashy digital effects reign supreme, it is refreshing to hear some big fat distortion and classic flanging.

*Alan Peterson contributes to **Pro Audio Review** and sister publication **Radio World**.*

Good Hunting In NAMM

by Brett Moss

Back from NAMM! While NAMM did not have as much software as, say, the Mac World show, I still saw enough software to write half a dozen columns.

So get ready for the whirlwind Cook's Tour ... and playing off the nautical theme, we shall start at Waves...

New from Waves (*www.waves.com*) is the latest member of the popular Renaissance plug-in processor family, Renaissance VOX. Featuring a compressor and a gate, VOX is crossplatform (i.e. TDM Mac [$400] or Native Mac/PC [$200] — AudioSuite, RTAS, VST, MAS and DirectX). And bowing to demand, Waves has produced a software version of the L2 Ultramaximizer for TDM Mac ($800). Perhaps a native version is in the offing.

Above: Emagic Waveburner Pro V.2; right: Waves L2 Ultra-Maximizer

Brand spanking new is Cakewalk's (*www.cakewalk.com*) SONAR. A new-from-the-ground-up multitrack recorder/editor/mixer. Windows 98SE/2000 are the intended platforms. SONAR starts at $479 and the full version (packed with software synths and other goodies) goes for $739.

Emagic (*www.emagic.de*) has upped the Mac-based Logic Audio to 4.7. New is 7.1surround sound for TDM systems, console-style inserts, support for the latest Emagic plug-ins and software instrument programs and VST 2.0 support too!

Oh, the upgrade is free for registered users!

Did I mention that Emagic is providing a custom version of Logic for M Audio's Delta I/O cards? Well, they are (and M Audio is bundling Sonic Foundry's ACID Xpress, Siren Xpress, Qdesign's MP3 player and MixMeister DJ with the card).

Further from Emagic, the WaveBurner Pro ($299) CD mastering utility sports new features such as support for VST plug-ins, ASIO support, more onboard mastering tools and support for more file formats (WAV and MP3 added) and bit depths.

And lastly for Emagic is the codevelopment deal with Tonos (*www.tonos.com*). Tonos is developing a Web site for artists to collaborate (in the Collaboratory!) and record tunes. The heart of the portal is the TC8 eight-track — Internet server-based recorder. Free for 30 days then pay $29.95 per year.

Ultrasupercool is Universal Audio's (*www.uaudio.com*) announcement of the Vintage Compressors plug-in line to model the company's retro hardware boxes such as the 1176LN and Teletronix LA-2A. First out will be Windows VST versions.

Just a smidge of noncomputer software news.

TASCAM (*www.tascam.com*) has announced that it has signed a deal with Be (*www.be.com*) to utilize the BeOS-based BeIA (Be Internet Application) as the base OS for upcoming products. And TASCAM has also created www.mx2424.com, a forum for MX-2424 multitrack recorder users to get answers, discuss things and download updates and manuals.

Mackie (*www.mackie.com*) is providing an upgrade to the software for the Digital 8 Bus console. Version 3.0 improves plug-in support, networking and surround sound mixing functions along with many unspecified "user requests." As traditional, a free download is available at the site.

Version 2 of Korg's (*www.korg.com*) OASYS PCI's operating system now includes PCM support. Am I wrong or is that a "D'oh!"? The download is free from the site.

The Gigafolks at NemeSYS (*www.nemesismusic.com*) recently inked two deals for support of its Giga software libraries. Ilio Entertainments (*www.ilio.com*) and Spectrasonics (www.spectrasonics.net) will produce new sample libraries in the Giga format.

And on that note, next month I promise to do a little roundup of the software instruments and related thingies that I saw at NAMM.

Send your software update news to Brett Moss, Equipment Editor, Pro Audio Review, P.O. Box 1214, Falls Church, VA 22041

Cover all the bases with Portable One Dual Domain

Today's audio testing requires a comprehensive audio analyzer. A hand-held jitter meter or audio monitor just won't do. Whether you're facing a new high performance a/d converter, multimedia audio, a transition to digital broadcasting, or just another routine week of maintaining a mixture of digital and analog equipment, you'll need to make three types of measurements: digital audio, digital interface, and analog audio. The Portable One Dual Domain does all three.

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READER SERVICE NUMBER 182

Access Virus TDM Plug-In

by J. Arif Verner

Digidesign, in collaboration with German synthesizer developer Access Music GmbH, has created the Access Virus TDM Plug-In ($795). Virtual synthesis has come to the Pro Tools environment with awesome results.

Sonically, the software and hardware synths are identical. In fact, patches can be transferred via MIDI between the two. Although it is great to tweak real knobs on the hardware unit, the plug-in has the advantage of additional polyphony. How about a maximum of 128 voices with 64 multitimbral parts? One DSP chip supports a 16-voice Virus plug-in with up to eight DSPs per system. Your Mac or NT computer needs Pro Tools 24 MIX or MIXplus running v5.0 (or higher).

Visually, the plug-in is similar to the synthesizer. Rows of black knobs are set against a dark red background. Rotate the onscreen knobs or type in a number to change the values. There are six pages to navigate around the program. To edit the tone generator, for example, go to the Oscillator page. For routing sources, destinations and modulation, go to the ModMatrix page. Simple enough.

Analog-modeling subtractive synthesis

Access Virus is based upon analog-modeling subtractive synthesis. Each voice has three oscillators (two main oscillators plus a suboscillator an octave below the first one), two resonant filters, two envelope generators and three LFOs. The LFO waveforms can also be used to modulate other synth parameters.

Another interesting feature is Saturation, which offers five settings that create distorted/enhanced overtones. To fatten up a sound, Unison mode stacks two or more voices for each note played.

Virus has three modes of operation. The most obvious is to use a MIDI keyboard to trigger the software's sounds. Even with my humble G3, I could not perceive any latency. Response was fast and immediate. MIDI notes can also be recorded into a Pro Tools sequence for Virus to play back. And finally, there is a small onscreen keyboard triggered by the mouse.

Access Virus effects include chorus, delay, vocoder and an arpeggiator. The vocoder is great for simulating robotic vocal sounds. Using the vocoder reduces the polyphony and limits use of the filters, but the tradeoff is well worth it. The arpeggiator has all the standard modes of operation including up, down, random, chords and notes as played. Also cool is the ability to sync the arpeggiator to the Pro Tools MIDI clock. Tempos from 63 to 193 BPM are supported.

To take it a step further, Virus can be used as a signal processor. Try sending an external audio signal through its filters, effects, saturation and modulation modes. If that is not enough, add the vocoder.

The fun factor of Access Virus is extremely high. It is easy to set up, sounds awesome and there are tons of great presets. For a list price of $795, the plug-in is a steal. Compare this price to the keyboard at $2,295 and the rack module at $1,795. This is one Virus you will be happy to have!

New at NAMM 2001

by J. Arif Verner

Digidesign is now shipping its flagship application, Pro Tools 5.1. This conveniently named upgrade opens the doors to the 5.1 surround sound environment. Also new in this version is a cool feature called Beat Detective, which performs automatic groove detection, region conforming and edit smoothing.

Other additions include MultiShell II for improving DSP efficiency, a MIDI event list editor and integration with Avid Picture Workstations. Look for an in-depth review of Pro Tools 5.1 in an upcoming issue of ***PAR***.

Digidesign also reaffirmed its continued support of the PC platform. A PC version of PT 5.1 is currently in development, but a release date has yet to be determined.

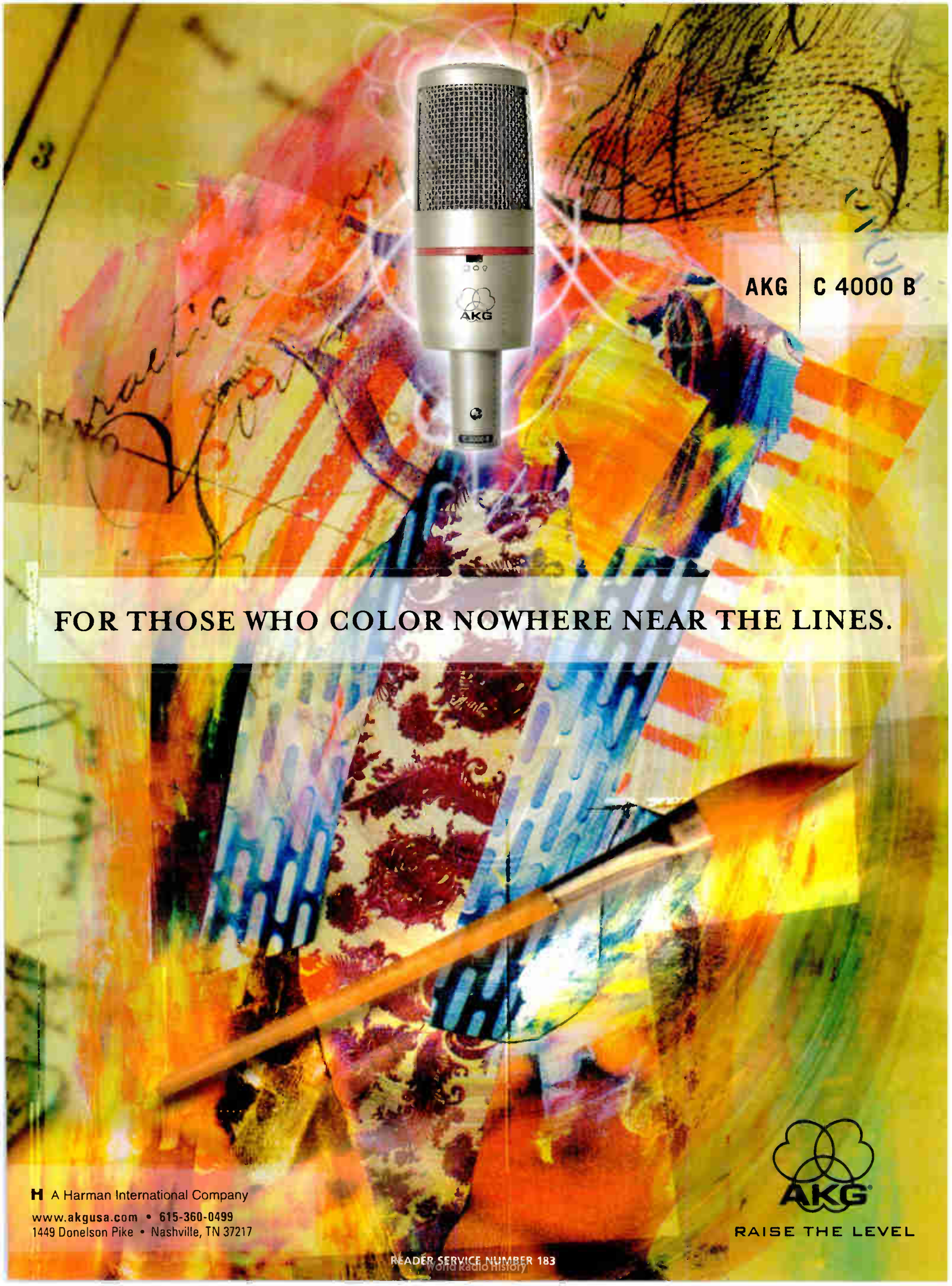
The big news on the hardware front is Control 24 ($7,995). Designed in collaboration with Focusrite, Control 24 is a hands-on control surface with an analog front end for TDM-equipped Pro Tools systems. Connecting to a computer via Ethernet, Control 24 provides 24 touch-sensitive motorized faders, 16 Focusrite mic preamps, EQ and dynamics, touch-sensitive automation and 5.1 surround monitoring.

Edit Pack is the newest add-on option for ProControl. Designed for 5.1 surround applications, Edit Pack uses two motorized, touch-sensitive panning joysticks along with a two-button trackball and a color-coded QWERTY keyboard. Edit Pack also features 20 dual-function edit switches for controlling Pro Tools software. Edit Pack brings high-resolution surround metering to ProControl, while featuring a comprehensive Machine Control section.

Soft SampleCell is a standalone software sample player; a host-based version of Digidesign's Sample Cell II Plus PCI card. The program supports 16- and 24-bit audio files. Also included is onscreen editing with up to 64 dynamically allocated voices.

Digidesign also announced software distribution agreements with DUY, Bomb Factory and Massenburg DesignWorks.

*J. Arif Verner, the owner Infinite Sound, a project studio specializing in music, film and audio production, is a regular contributor to **Pro Audio Review**.*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RAISE THE LEVEL
READER SERVICE NUMBER 183

MindPrint
DI-Port Digital Audio Interface

by Loren Alldrin

German audio manufacturer MindPrint is a relative newcomer to the domestic recording scene, picked up by Steinberg North America (makers of Cubase and WaveLab software) for distribution in 1999. I tested MindPrint's all-in-one EnVoice mic processor/preamp last summer. Recently I got to check out its new DI-Port digital audio interface.

Features

DI-Port ($449) is part of a new class of interfaces designed to meet the needs of the modern computer-based studio. It offers microphone or line input preamplification, A/D and D/A conversion and monitoring in one compact, desktop unit.

DI-Port has a pair of Class A preamps fed by three different physical inputs. Front-mounted jacks are Neutrik combos, allowing connection of XLR or 1/4-inch plugs; on the rear of the interface sits a pair of RCA jacks. A switch on the unit's front toggles between the combo jacks and the rear-mounted RCA jacks.

Each analog input has a gain knob, green signal LED and red peak LED. DI-Port offers no pad switches, but its broad 75 dB gain range dips low enough to handle hot microphones and loud sound sources. A phantom power switch routes DC to both mic inputs for condenser microphones. There is no phase invert switch on either input.

DI-Port offers 24-bit D/A and A/D conversion on S/PDIF inputs and outputs. All digital I/O is on the back panel and consists of RCA and TOSlink optical jacks. Rear-panel switches select between DI-Port's 44.1 kHz and 48 kHz sampling rates, as well as digital clock mode.

In master clock mode, DI-Port uses its own internal clock regardless of what's present at its digital input. In auto mode the interface detects a valid input clock source and slaves to it automatically. A red LED located between the two digital clock switches lights when a valid clock source is detected at the unit's digital input.

Because of the delay that plagues software-only recording systems, many hardware interfaces offer their own zero-latency monitoring features. DI-Port has a simple but effective monitoring system that lets you set the blend of the digital signal returning from the computer (your playback tracks) and the analog inputs. The interface's mix knob goes from analog input only to digital input only; midway on the knob, the two signals are combined in equal proportion.

A front-mounted headphone output lets you plug phones into the DI-Port for hassle-free monitoring. The back panel has RCA monitor outputs suitable for driving a pair of monitors, as well as a combo stereo output on a 1/4-inch TRS jack. With a standard insert-style Y-cable, this latter jack gives you stereo 1/4-inch outputs. A single volume knob controls the level to the headphone and monitor outputs.

The last set of jacks on the DI-Port consists of the D/A out jacks, located on the unit's rear panel. This pair of RCA jacks carries a stereo analog output converted directly from the unit's digital input. For a straight analog output from your computer, these are the jacks to use.

At a Glance

Applications:
Project studio; remote recording; multimedia

Key Features:
Two Class A mic preamps; 24-bit A/D and D/A conversion; built-in monitoring; headphone output; RCA and optical S/PDIF I/O

Price:
$449

Contact:
MindPrint/Steinberg North America at 818-678-5100; or circle Reader Service 88.

In use

With MindPrint touting the quality of the DI-Port's Class A preamps and 24-bit converters, I was eager to give the box a listen. I recorded several instruments and voices (with a handful of different microphones) at 24-bit resolution into my computer. I used the computer as the digital clock source, with the DI-Port in master mode.

On playback, the DI-Port proved capable of recording some excellent-sounding tracks. The unit's mic preamps have a balanced, full sound. Their top end has ample

MindPrint continued from page 48

detail but doesn't sound hyped or unnaturally extended. Low-mids and bass were full yet defined.

In an informal mic preamp shootout, the DI-Port's preamp section came in a close second to a unit costing three times as much; it totally outclassed the mic preamps in a $2,000 digital mixer.

Which brings up an additional application the DI-Port can be used for — basic microphone preamplification. Though it doesn't have analog outputs dedicated to the preamps, you can get a pure preamp signal from the monitor jacks if you crank the mix control fully clockwise. The mic preamps sound good enough to justify their use in other analog-only applications.

> **DI-Port may be all the PC-based recordist needs to get great sounding tracks into and out of their computer.**

The converters on the DI-Port are very natural and transparent, and folks may want to consider using this box as an outboard converter rig for a DAT deck or other digital recording system (especially an older one). As I've noted time and again, we've crossed a threshold where sound quality is concerned — even low-priced 24-bit products like the DI-Port commonly deliver excellent clarity and accuracy.

The DI-Port's quality of construction is first rate both inside and out. Knobs and switches feel solid and smooth, and I've always been a fan of Neutrik XLR connectors. Knobs and jacks are clearly labeled and easy to access. I appreciate the fact that the DI-Port has a rear-mounted power switch and front-mounted power LED.

I didn't appreciate the unit's large, ungainly wallwart power supply that takes up about three AC outlets. Although it would add a bit to the cost, I'd prefer AC wires on both sides of the transformer to free up some power-strip real estate!

DI-Port's headphone output could use a little more juice. Rated at just 300 mW into low-impedance phones, the little box was unable to push my high-impedance AKG K240 phones past about medium-loud. If you want real loud phones (to hear yourself over a drummer, for example), make sure you're using low-impedance, very efficient headphones with the DI-Port.

Buyers should know that this interface makes a few assumptions about how you record. First, it assumes you don't need inserts to patch in analog processing before hitting the converters. This may not be the case for many folks, myself included. I almost always run mic signals through a tube compressor on their way to the converters, and would have gladly traded the DI-Port's odd 1/4-inch stereo monitor output for some insert jacks.

Second, DI-Port assumes your computer is already equipped with an S/PDIF digital I/O card. Thankfully, S/PDIF I/O is becoming pretty common even on multitrack PCI audio cards (such as Soundscape Mixtreme, Frontier Designs Dakota and others). I couldn't help think, though, about how cool DI-Port would be with USB serial support. It wouldn't need a card installed at all and would work like a charm with laptop computers. Maybe someday.

Summary

One key to a product's success is how well it handles the twists and turns of today's music production process. DI-Port offers good flexibility where I/O is concerned, allowing you to record a variety of different input sources without re-patching cables. You can use the DI-Port in several different roles in addition to simple computer interface.

Sonically, DI-Port won't disappoint. It mates excellent-sounding preamps with crystal-clear converters, resulting in sound that's good enough for the toughest master session. Home recordists that own the DI-Port can rest assured that their interface is capturing every nuance of their instruments and microphones.

DI-Port sets out to be the only box today's computer-based recordist needs to get great-sounding tracks in and out of their computer. Though I really missed analog inserts for additional outboard processing, I think DI-Port comes extremely close to achieving that goal. It's easy to see how the DI-Port, a good mic, some headphones or active monitors and a S/PDIF-equipped computer would be all you need to capture excellent recordings virtually anywhere.

*Loren Alldrin is **Pro Audio Review's** project studio editor and author of The Home Studio Guide to Microphones.*

Product Points

Mindprint DI-Port

Plus
- Excellent sound quality
- Onboard monitoring
- Versatile analog input scheme

Minus
- No analog inserts
- Weak headphone amp
- Real estate hogging wallwart

The Score

A great-sounding, one-box solution for folks who already have S/PDIF digital I/O on their computer.

A professional sound system for under $500
• A 6 channel 200 watt RMS professional system
• Compact, easy to use with lots of flexibility
• Lightweight design with aluminum chassis and rugged plywood enclosures
• 24 bit DSP digital processor with over 256 effects
• Crystal clear highs & deep bass right out of the box
• The best pro sound value on the market today
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REVERB CHORUS
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24 BIT EFFECTS PROCESSING
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CARVIN PA620
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LEVEL 3
LEVEL 4
LEVEL 5
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As shown:
620-805 complete system with 1 PA620 mixer • 2 805 heavy-duty 10" 200 watt speakers • 2 PH50 50' cables • 1 CM50 pro mic & XLR25 cable
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READER SERVICE NUMBER 185

The HDR24/96 versus recording on

A fully-equipped LESS than three

With an HDR24/96 it's so easy to record, edit, and manipulate tracks, so easy to be creative…whether you're recording for yourself, your band or for a Fussy Client.

With all due respect, recording onto linear media (a.k.a. tape) has some pretty severe limitations: Access time to cue points is slow. Punch-ins erase stuff you previously recorded. And the tracks just sit there side-by-side on the tape with no chance to easily slip, slide, cut or paste them in new ways.

Hard disk recording and workstation editing for less than the price of linear recording.

It's no secret that non-linear hard disk recording is the way to go. But until now, 24- track/24-bit recording and playback required serious investment in a digital audio workstation. — And a heckuva lot of mousing and clicking.

Only the HDR24/96 combines the intuitive, analog-like convenience of a tape deck with the editing versatility of a computer-based workstation.

As easy to use as an analog recorder.

All basic functions are right there on the HDR24/96 front panel including transport buttons and individual Record Enable buttons for each track. Just hit Record and Play

* based on current U.S. list and pro audio dealer "street" prices at the time of ad production and on the assumption you will buy a CRT-type SVGA monitor and not an ultra-pricey flat panel model like our art director insisted on using for this ad.

** based on average of length of current pop songs using 24 tracks @48kHz/24-bits and a liberal number of extra regions and virtual takes. Does not apply to extended trance remixes.

long thin strips of rusty plastic.

Mackie 24-track HDR24/96 Hard Disk Recorder/Editor costs tape-based, 8-track digital recorders*...and does much more.

without even cracking the manual. But if you plug in an SVGA computer monitor, things get even better.

The graphic interface that tape recorders always should have had.

Even if you immediably don't use the HDR24/96's editing functions, you'll love the graphic interface for recording.

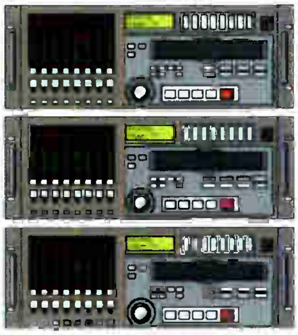
It gives you one-click access to all deck functions without a lot of annoying pull-down/fly-out menus.

Choose from 2x, 4x, 8x, 12x or 24-track views and then watch them scroll smoothly past a centerline.

Mark hundreds of cue points and four locate points for looping and auto-punch-in modes. Cue points are visible on screen and are accessible from a side list.

Use the mouse to "scrub" individual tracks, Cue, Punch and Loop points with continuously variable velocity.

Each track also supports eight "virtual tracks," so you can do multiple takes and comp them together easily.



Think of MackieMedia™ as "tape in a brick."

Right out of the box, the HDR24/96's internal drive will record 90 minutes of 24 tracks at 48kHz. Your backup choices are simple—

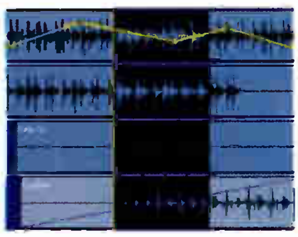
1. Record directly to a MackieMedia™ M90 external drive. They're considerably less expensive than the SCSI drives some HD recorders require — $10 a song** — and they're in stock at your Mackie dealer.
2. For a quick back-up of just a song or two, we also offer an optional 2.2GB ORB™ drive that uses *really* inexpensive media.
3. For real economy use the HDR24/96's 100BastT Ethernet port to back up to your computer and its media.

Even with three OPT•8 I/O cards, a MackieMedia removable disk, SVGA monitor, keyboard and mouse, the HDR24/96 costs less than three digital tape recorders...which don't offer loads of workstation-style editing features, super-fast access and true 24-bit recording.*

Serious editing tools built in... with 999 levels of un-do.

Once you've experienced non-destructive editing of tracks, you'll never go back to linear recording.

You can mark a segment (or multiple non-adjacent segments) as a *region* and then cut, copy and paste it anywhere — onto a blank track or right in the middle of an existing track without erasing anything (the part of the track after the insert just "slides down").



Zoom in to the waveform level. Drag, cut, paste, and slip tracks and segments of tracks just like on super-expensive workstations. Adjust track levels. Add editable crossfades. All with 999 levels of undo.

You can audition regions or modify their start/end points instantly, capture them as "sound elements" for later use or quantize them to user-defined time grids. And all regions are easy accessible from a side menu.

Create fade-ins, fade- outs and crossfades just by dragging and dropping them ...and then set their length by dragging the mouse.

Add volume envelopes for simple level automation of regions or whole tracks.

Then use Track Render to combine all or selected regions of a track just as you hear it — complete with crossfades, volume envelopes, mutes, etc. — into a single region.

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This ad only scratches the surface of the HDR24/96's features, options and capabilities. Visit our web site...or get your hands on an HDR24/96 and experience (pun intended) *unparalleled* creativity.

*Twenty-four track songs for under ten bucks each!! Divide the cost of a MackieMedia M90 into the 20+ pop tunes you can record on it and you're looking at under a ten-spot for each 24-track master. —Much less if you do a little disk drive house-keeping. Remember, hard drives can choose only to store audio data, not silence. Tape just rolls merrily along...costing money... whether you're using a track or not.***

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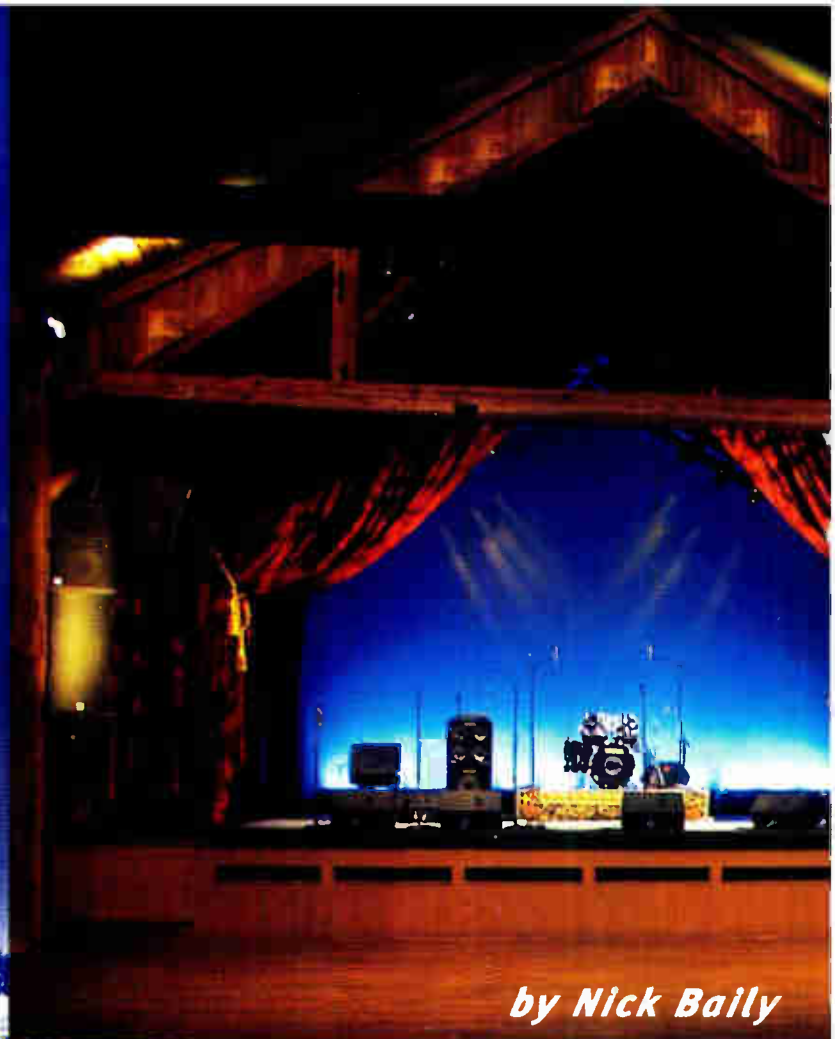
READER SERVICE NUMBER 186

Barn Raising: Upgrading the Sound at the Historic Barns of Wolf Trap



by Nick Baily

Top: Interior of the Barns, looking toward the stage. Bottom, from left to right: neo-folk act, SGG & L on stage, Soundcraft SM12 32x12x2 used for monitor mixing, the Barns exterior, Barns Production Manager Bob Grimes

Most techs would say installing a permanent system for a live room is never easy, but you would think Greg Lukens might disagree. After all, he has spent dozens of years as a systems designer, equipment salesperson, engineer and general audio guru at Washington Professional Systems, just outside of Washington, D.C.

When Bob Grimes, production manager at The Barns of Wolf Trap, approached him about updating the antiquated sound system at this legendary venue, however, there were some special issues to deal with. Together, they came up with a unique plan that took advantage of recent gear innovations while navigating the complexities inherent in the historic landmark and nonprofit purchasing process.

Land grant

Wolf Trap, in Vienna, Virginia, is the only U.S. National Park devoted to the performing arts. Founded in 1966 with a grant of 100 acres of land and funds for a large outdoor amphitheater by Catherine Filene Shouse, the Filene Center venue seats 6,000 people and is well known for the top-level performers that grace its stage in the summer.

Some years later, Shouse attended an informal concert at an 18th century barn in Maine and was so impressed with its acoustics that she made arrangements to identify and relocate two historic barns to a site at Wolf Trap. Reconstruction was completed in 1981,

and the venue now hosts an average of 120 shows throughout the winter months.

"Originally we did all the performances without a sound system," explains Grimes, "and the acoustics were stellar."

Most of the shows at the time were classical chamber, roots- and folk-based artists, but the total lack of sound reinforcement was difficult — for obvious reasons. It was simply impossible to make up for any natural level differences between performers and the room's 352-person capacity was large enough to make delicate passages hard to hear at times.

"But the sound of the room was so good, we didn't want it to sound 'like a PA,' so to speak," he continues. What they ended up with over the years was a pair of Meyer 804s, better known as studio monitors than live mains, a Soundcraft 200B console at the FOH position and a hand-built Interface Electronics monitor console inherited from the Filene Center. The time was ripe for a change.

In with the new

Grimes approached Lukens at WPS with a tall order: the new system had to be clean sounding and transparent, it had to respect the aesthetics of this historic landmark and it had to be done on a budget. Complicating matters was the fact that the cost had to be spread over a three-year period, with approximately $35,000 available in each year.

"In this business, long range committed funds budgeting is just not the way things get done," says Lukens. "Most of the time when people want something, they have put it off for too long, and need it all for 'next week!' In this case, we had the challenge — but also the luxury — of planning things out carefully."

The centerpiece of the new system is the JBL DSC 260 ($1,999), a digital systems controller that handles crossover and EQ for the entire system, as well as time-delay and limiting. The unit replaces a whole stack of equipment, and its attraction is more than just compact size — it is the promise of a much cleaner and more elegant way of handling signal path.

The DSC 260 has a discrete channel for each band (high left and right, mid left and right, low left and right) of the system. Each driver component in turn is driven by its own discrete amplifier channel. Each channel can be individually addressed and adjusted for frequency response, dynamics and timing/delay. Instead of the standard one-third-octave equalizer used to flatten out house response, the controller can add one parametric filter at a time.

"You see a lot less phase shifting as a result," says Lukens. "Whereas usually you are dealing with 30 filters in front of each channel, now it is rarely more than four or five. And we also have the ability to manipulate individually when the highs reach the listener, and so on, because of the delay control features. It makes for much more coherent imaging."

The first installment focused on the mains and subs, amplification and controller. Rounding out the package were two JBL 120 degree AS2215s, tilted down for coverage of the first rows, and two of the 90 degree models handling the balcony and back of the room. As mentioned, discrete amplification is integral to this system, with JBL MPA 600 and 1100 amplifiers handling this duty. To balance out the system, the SIA SMAART Software system was employed, which uses a computer and calibrated microphones to enter settings into the DSC.

An early decision was made to keep everything in the JBL family, both for the ease and compatibility of the equipment and the price. "If a venue has money to burn, sure, there are other options. What is nice about this system is that you keep everything in the same family and, in my opinion, you get the best bang for your buck," says Lukens, who points out that he used similar components at the Birchmere, a Virginia club with a storied history of quality acoustics and live recordings.

"Don't forget — we are nonprofit," says Grimes, "so the stuff can't break — we have no money for backups. We also had to get everything installed while still keeping to a full schedule of performances."

The second year was devoted to the consoles. At the FOH position, the old Soundcraft 200B was upgraded to a 40x8x2 Soundcraft K3 Theater with a four-channel matrix, enabling separate feeds to the bar, DAT, and hearing impaired fans. For monitor mixing, a Soundcraft SM12 32x12x2 was brought in. In the third year the installation was completed with a full monitor system, using JBL SP MS112s and SP212-As, also controlled by an additional set of four DSC260s and existing QSC components for power.

Planning pays off

"The really fun part in the third year was that life in FOH got so much easier since the monitor wash went away nearly completely," says Grimes. "That is on account of the DSC as well. There is an amazing difference from pre to post digital control, and you can see it with the touring engineers. It used to be that at the end of the show you'd see all sorts of things done to the house graphic EQs, really major changes. Now, more often than not, they are completely flat. It is pretty amazing really."

An additional consideration was that the system needed to be completely removed in the summer months — for the opera season at the Barns — and both consoles are often drafted for duty at large multiple-band shows at the Filene center.

With careful planning and some new tools, Greg Lukens, Bob Grimes, their staffs and supporters of the arts made it possible for The Barns at Wolf Trap to have sound reinforcement that does justice to the room's stunning aesthetics. The next time you are in the Washington, D.C. area, stop by and hear for yourself. And now you know it only looks easy.

Nick Baily has spent 10 years in windowless rooms as a recording engineer, producer and studio manager. He currently lives and works in New York City as a manager of an online digital music company.

Digital Audio Distribution:

eMusic.com

by Carl Lindemann

Will digital distribution kill the CD? Some say our musical appetites will soon be satisfied at the cost of a fixed monthly fee, like cable TV. Even after having lots of fun with the eMusic Unlimited subscription service, however, I cannot buy into the notion that this is going to put Tower Records out of business anytime soon.

I see a new market growing up alongside CDs. It is a convenient, less expensive means to gather tunes for on-the-go listening. It takes the place that prerecorded cassettes had in my in-dash stereo. Even when I already had the CD, it was worth paying a few bucks for the convenience of not having to make my own copy. Digital downloads will do the same.

eMusic.com

eMusic.com offers a growing collection of MP3s with more than 150,000 songs on 12,500 albums and 8,000 artists represented. For those who buy songs one at a time, single downloads are a reasonable 99 cents. Albums go for $8.99. But the real deal is eMusic Unlimited for $9.95 a month. As the name suggests, this provides unlimited access.

The money is split 50/50 between eMusic and whoever owns the copyrights to the music. With the albums and singles, that is easy to figure. With the Unlimited service, the pot gets divided up depending on a breakdown of the percentage of downloads. All songs are watermarked, so it is easy to spot unauthorized trading.

Some 15,000 Napster users have already been notified that they are in violation of copyright laws. According to Brett Thomas, eMusic's vice president of technology, the majority of those contacted have cooperated. "Some were not even aware that they had the option to turn off file-sharing. It is more of an educational thing than anything else," he said.

Simple signup

As an end-user, signing up was simple. After enrolling, I scrolled through the offerings in the various musical categories and started to download at will. If I had tried this at my home outside of the reach of high-speed access I would have been disappointed. Fortunately, I conducted my tests while house sitting at a friend's place with DSL.

I sampled widely from the offerings including tracks from John Hiatt's latest to cuts off of the Bill Evan Trio classic *Sunday at the Village Vanguard*. Downloading a dozen or so songs at a time on DSL made this as fast as moving through the checkout line. Admittedly, the catalog is more eclectic than encyclopedic. This had a curious effect on the way I used the service. I tried a few things I would never have listened to otherwise, including Sue Foley's excellent *Love Comin' Down*. My gut feeling is that subscription services will be a boon for such deserving, lesser-known talents.

eMusic MP3s are encoded at 128 kbps with the Xing 2.0 encoder. Thomas sees this bit-rate as a happy medium between efficient size and quality. I downloaded several files to compare against CD copies. Some sounded better than others. This was reminiscent of earlier experiences with prerecorded tapes. Here, quality depended on how the encoder reacted with the material.

Occasional shifts in the stereo image and other weird processing artifacts were obvious when played through my studio monitors. The real-life test came after using Adaptec's Easy CD Creator to convert and load the files onto a CD-R for a road test. These copies were not quite as clean as the original store-bought CDs, but having a disposable copy for the car has its advantages.

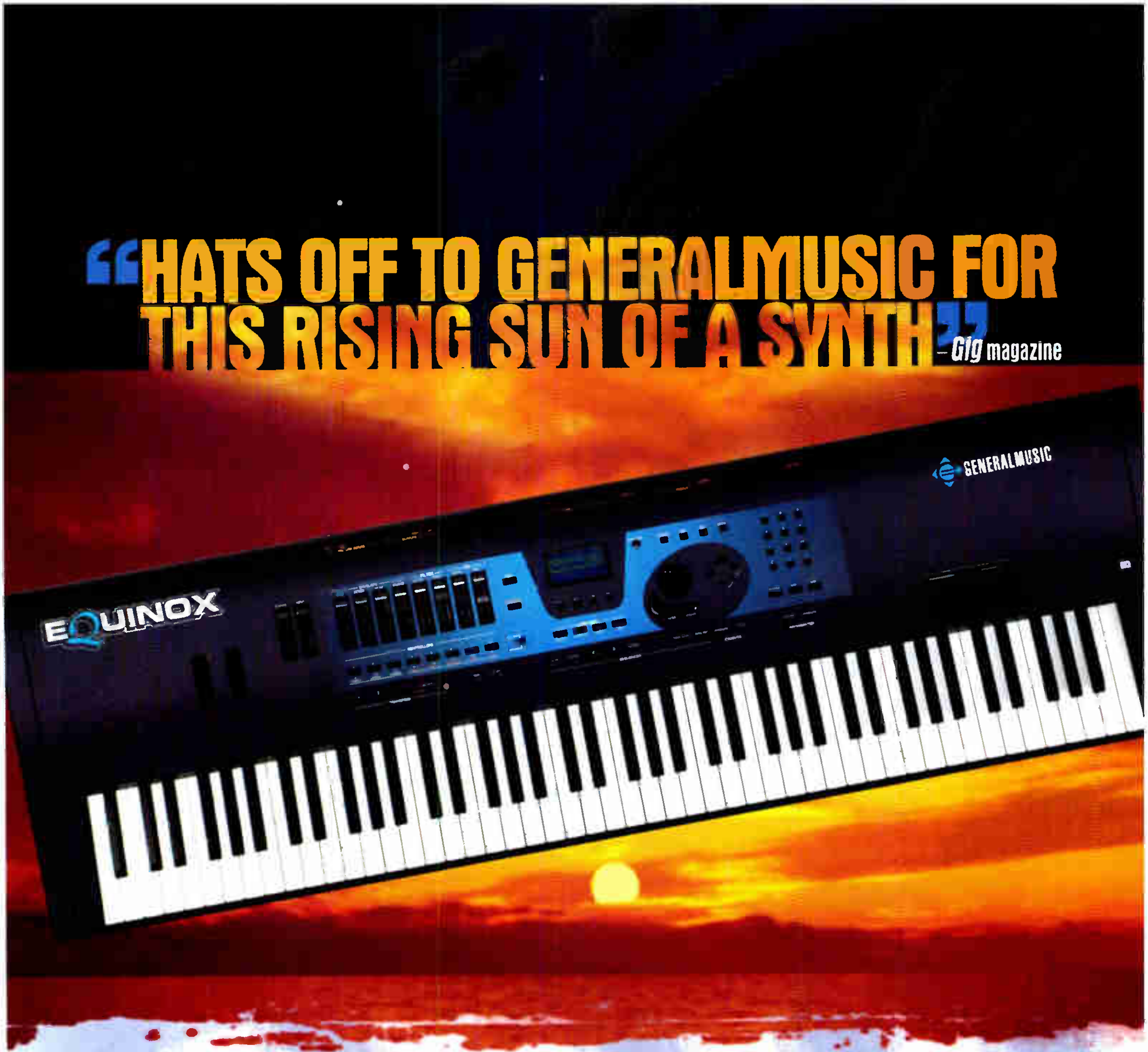
The trial mix I put together was disposable — good for a few listens before I got tired of it. Driving around town with this cranking out of the car stereo sounded perfectly alright. But for critical home or studio listening, I'll take the CD originals.

Behind the scenes, eMusic gets the songs online by ripping uncompressed PCM audio off of CDs onto a 7 TB back-end server. Once in the system, the arduous encoding process begins. Currently, Thomas is reviewing other encoders to see which delivers the best quality. If there is a change, the entire collection will be redone. Because the uncompressed audio is already on the system, re-encoding is more manageable. Files can be accessed without touching the CDs again, but still, it is time consuming. The task will take several weeks with its automated fleet of 20 Linux-based dual processor Pentium III 400 machines running around the clock. The methodology is sound; eMusic.com is ready for whatever turns codec technology takes.

Summary

The upshot here is that the pattern the movie biz saw with videocassettes seems likely to repeat itself — home viewing did not kill off the cineplex. The convenience encouraged audiences to watch more and spend more. Digital distribution offers a similar compromise in quality for increased convenience. It addresses different needs than existing channels. Is there any better way to build a business than finding new ways to service customers?

*Carl Lindemann is a frequent contributor to **Pro Audio Review**. He is an independent public radio producer, and consults professionally on new media issues.*


"HATS OFF TO GENERALMUSIC FOR THIS RISING SUN OF A SYNTH"
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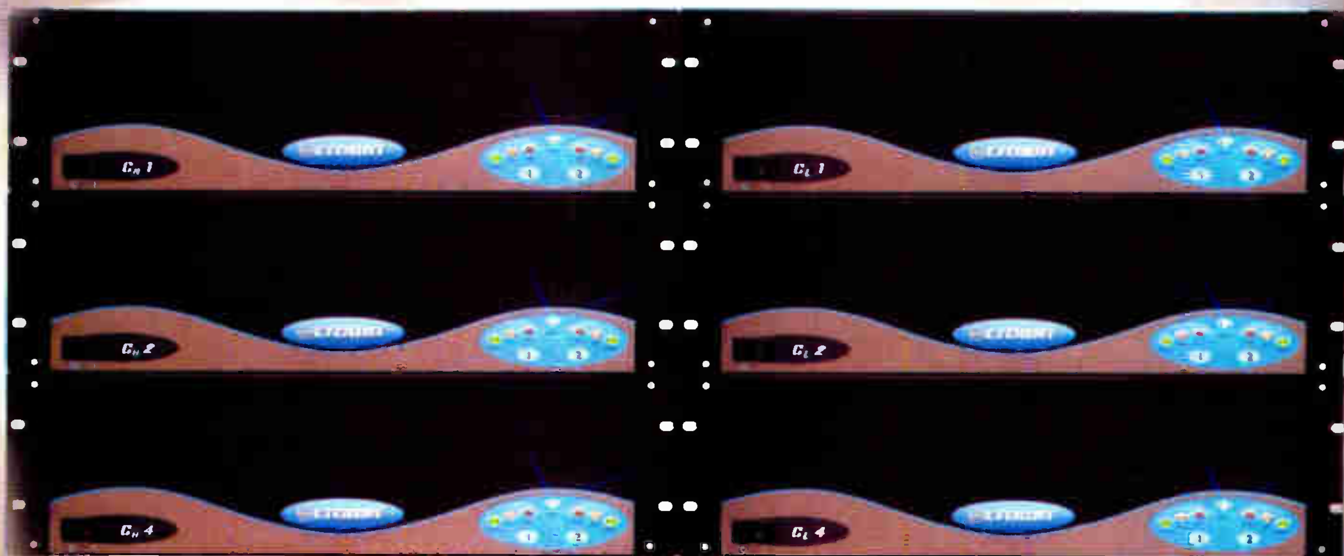
A Hot Time in Orlando

by Brett Moss



Above: Crown contractor CH AND CL series amps; below: Electro-Voice NDYM SCU wireless mic family

Audio tradeshow nomads will be forgiven if they seem a little disoriented at NSCA. Not too many weeks ago they were at the Orange County Convention Center in a warm locale with palm trees near a large Disney park and now they find themselves at the Orange County Convention Center in a warm locale with palm trees near a large Disney park — it is *déjà vu* all over again.

But here we are at the National Systems Contractors Association convention (March 8 - 10, 2001) in the OCCC in Orlando — on the other side of the country. This year promises several hundred seminars on technical and business issues, product demos and, of course, over 500 booths with lots of products.

The recent NAMM show saw a distinct lack of new amplifiers for the pro audio audience, but if NSCA holds to tradition that should be rectified.

Show and tell

Hafler's latest amps, the GX series are equipped with specs and features aimed at the cinema market such as specialized delay times and crossovers. Power-wise one amp is 300W per channel and the other is 600W.

Crown is showing amps from the CH and the CL lines. Power ranges go from a couple of hundred Watts to over 3,500W, bridged. Versions are also available in 70/100V.

Electro-Voice is building upon its Precision line of amps with the Px series. The Px series offers plug-in slots for processor modules for customization of EV speaker systems.

Neutrik is one company that is clearly appreciated by NSCA crowds. New for them is the Patchlink SPL, a modular patch bay for 1/4" plugs. Also from Neutrik are a module for that bay — a split return module and the PatchBox — a standalone patch box for 1/4" plugs.

Venice is the name of the new small mixer/console line from Midas. Available in rackmountable eight mono, 16 mono or 24 mono input versions (all with an additional four stereo inputs), the Venice incorporates features from the Heritage and XL series.

Innova Son is also downsizing its consoles. The Sensory Compact Live is a smaller version of the Sensory line of digital consoles. The Sensory line features include parametric EQs, comp/lims and

gate/expanders per input channel.

Soundcraft is bringing to NSCA the M series of small — mid sized mixers that debuted at NAMM. Available in four mono/four stereo, eight mono/four stereo, 12 mono/four/stereo input channels and can operate as tabletop or rackmount. Naturally Soundcraft is showing its larger FOH desks — the Series TWO and Series FOUR.

For looking at new multiprocessors Klark Teknik may be the place. The DN7453 and DN7454 are time alignment units. The 7453 offers one input and three outputs while the 7454 has two inputs and four outputs. Also at the KT booth is the DN9848 speaker system processor (and its little sister, the DN9824). The goodies list includes delay, several crossover filters, memory and factory presets.

For speakers, new items can be seen from ServoDrive/Sound Physics Labs - the Contractor Cube and a to-be-named monitor cabinet.

The big speaker news for Electro-Voice is the QRx series. These speakers consist of 12", 15" and dual 12" full-range speakers along with 18" and dual 15" and 18" woofers. Also new for EV is the Sx325 speaker, the latest in the Sx contractor line. The full-range 325 features a 12" woofer along with 90 x 90 degree dispersion pattern.

Powered speakers are the center of D.A.S.'s NSCA efforts. The ACtive series include 12", 15" and 18" models.

MacPherson is debuting new accessories for its AXIA speaker lineup. For wheeling those guys around there is a dolly, and for covering them up a form-fitted nylon loudspeaker covering is available.

For speaker processing, Electro-Voice's Dx38 now sports new software. The RACE program expands and fine-tunes speaker system product measurement and performance.

White Instruments is showing a new member of the ParaMedic family of processors. The ParaMedic 66 is full-featured system processor with six inputs/six outputs. Also new is the upgraded 4700 EQ — now with a range up to 100 kHz for work with DSD/SACD projects.

Cobalt used to be part of Telex's mic line but it has been ported to Electro-Voice and expanded. The new members are the Co4 instrument mic, Co5 vocal mic, Co7 vocal mic and the Co9 vocal mic.

Following on its NAMM debut, Electro-Voice is introducing a new UHF wireless receiver, the SCU. The SCU is part of the N/DYM line and will be the center of the usual choice of bodypack or handheld wireless mic system.

In the hearing assistance arena, Sennheiser is bringing the SP 230 infrared system to town. The 230 operates on two separate channels with different frequencies (2.3 and 2.8 MHz) for each channel. Sennheiser is also showing its in-ear monitors — now with the option of Future Sonics earbuds.

And lastly, RCI Custom Products will show a new portable 16-output mult box housed in a heavy duty Haliburton case.

Neutrik Patchbox

Klark Teknik DN9848

Digital Crossover System

by Will James

For years, the name Klark Teknik has been synonymous with high-end signal processors and analyzers. The DN300 and DN360 are considered by many to be the very best graphic equalizers. Klark Teknik has now introduced the model DN9848 digital crossover.

Features

The Klark Teknik DN9848 ($5,539) is a digital crossover/low-level signal speaker management system. As with all crossovers, the intention is to optimize any speaker system's operation by properly routing the various frequency bands of the composite audio spectrum to the various individual components. This is achieved digitally with both internal operations through the use of a third-party software program called StarDraw, in this case a version created specifically to provide remote control for Klark Teknik system controllers.

The DN9848 is a single-rack-space unit with a multitude of displays and parameter controls on its front panel. On the left side are four LED stacks that show the audio levels present for each of the four inputs labeled A, B, C and D.

Alongside each LED stack is a menu button that illuminates when depressed to signify menu activity on that particular input channel. When the menu is engaged, the options are accessed by three rotary data entry knobs located just below the LED stacks. A green LED ring indicates that a particular knob is active.

On both input and output sections, the yellow menu keys access the respective control menus. One press enters the menu structure, at which point the green rings around the three encoders light to indicate which are active within that menu page. Another button press steps to the next menu page and so on back to the top. Any menu can be exited by pressing the Home key once.

Next on the face of the DN9848 is the parameter display, flanked by a recall button, a store button, a setup button and an 8-pin round PC serial port providing computer access via RS-232. The unit will also perform RS-232 to RS-485 conversion if necessary.

Just to the right of the parameter display is the output section, which is comprised of eight LED stacks, identical in appearance to the input section LEDs, and accompanied by a menu access button on each. In this case, when you engage the menu, the three data entry knobs correspond to output phase angle, high- and low-pass filter frequencies and signal polarity inversion, each ringed by the green LED when active.

Just below each output LED stack is a knob that controls the output level of each frequency group, ringed with a red LED to indicate activity. This rotary is also a push/push switch that activates the output channel mute. The red ring around the control lights to indicate that the output is muted.

The rear panel has a series of 3-pin XLR sockets, four females for input, eight males for output and a parallel pair — one of each flavor — to enable communication between the unit and other DN9848s, or a PC via RS-485.

The internal workings of the Klark Teknik unit are both useful and plentiful. There are a variety of security programs that lock out unauthorized users from adjusting any or all parameters. You can name your settings and place them into a user registry. There is a temperature/delay application that lets you delay the inputs relative to the varying speeds of sound at different temperatures, input parametric EQ and input compression (including threshold, ratio, attack and release times and hard or soft knee).

The output options are equally wide in scope, offering filter cutoff frequencies, filter type (Butterworth, Bessel, Linkwitz-Reilly), slope (6, 12, 18, 24, 36, and 48 dB per octave) relative to the type of filter, delays for time alignment of different frequency groups, and choice of continuous rolloff or shelf-type EQs.

At a Glance

Applications:
Live sound, contracting

Key Features:
Digital crossover/low-level signal speaker management system; parameters controllable onboard or with computer; multiple filter and slope configurations

Price:
$5,539

Contact:
Klark Teknik at 616-695-4750; *www.klarkteknik.com*; or circle **Reader Service 73.**

continued on page 62

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Klark Teknik continued from page 60

In use

I tried out the DN9848 on two different speaker systems. The first setup was a JBL four-way system, powered with QSC PowerLight amps.

I adjusted the DN9848 outputs to accommodate the appropriate crossover frequencies — setting the filters to 24 dB per octave in both high- and low-pass modes. As I applied signal to the system, I noticed the exactness with which the DN9848 completed each of its tasks. As I adjusted the phase angle from component to component-crossover frequency, I sensed the speakers shifting into proper alignment and noted the quality of the composite sound increase substantially. (Proper phase alignment, either physical or electronic, is still a property of time. Consequently two signals intended to carry the same program material — regardless of frequency — reproduced out of sync of each other will result in misalignment, or chronological mismatch which, in the end, equals bad sound.)

> I was able to dial in both speaker systems with ease.

The second test was to process the audio in a Yorkville TX series speaker system that uses Yorkville power amps. It uses a proprietary processor/crossover that monitors output voltage and current from the power amps, which is actually the only thing the Klark Teknik does not do. In both trials of the DN9848, I was very pleased with the results, and I was able to dial in both speaker systems with ease.

The owner's manual is well written, clearly explaining all the operations. I initially used the device without referring to the manual, as I often do to determine the user friendliness of a given piece of gear. The DN9848 was most friendly.

Summary

I found the Klark Teknik DN9848 to be a product of superior quality. The noise was nonexistent, the audio quality was superb and interconnection was achieved with ease.

If you have a good working knowledge of crossovers and their relative parameters, the Klark Teknik digital crossover system is an outstanding piece of equipment. If you do not have a working knowledge of crossovers, I recommend starting with something that has fewer parameters and controls, and working your way up to this fine crossover. I seldom hand out 10s on the "1 to 10" scale, but this one is deserving of a 10.

Product Points

Klark Teknik DN9848
Digital Crossover System

Plus
- Exacting control of speaker systems
- Adaptable to virtually any speaker system
- Digital, very quiet

Minus
- None

The Score
An in-depth and well-designed digital crossover system with excellent fidelity.

NAMM: Bigger, Louder, Better

by Brett Moss

Okay, so the hot ticket at the Winter NAMM show was not a product but the Spinal Tap concert (so bass-heavy that it shook almost a dozen recessed lights out of their sockets in the Hilton ballroom) — courtesy of Shure.

But you don't want to hear about what you missed. You want to hear about the new products you will be seeing in the future.

Hard disk competition

The biggest product splash may have been made by Alesis, with its announcement of a 24-track hard disk recorder — in the $2,000 price range. The ADAT HD24 is, at this time, sort of using the ADAT moniker (the 'T' actually stands for 'technology' and not 'tape'). The two hard disks will be removable/hot swappable and standard (and inexpensive) EIDE.

Spinal Tap rocking NAMM

In the news for currently shipping hard disk recorders, the MX-2424, from TASCAM, was shown with upgraded software (for 96 kHz recording) and a new GUI.

Akai has pumped its line of DPS personal studio hard disk mixer/editor/recorders up to 24-track with the DPS24. The unit offers 24-bit/96-kHz performance and motorized faders, along with all of the earlier features of the DPS family.

Korg brought out two new digital personal studios of its own. The big boy is the D1600, a 16-track mixer/hard disk recorder with a 20GB hard drive onboard. A little smaller is the D12, a 12-track mixer/recorder with a 6-GB hard drive. Both offer the option of a CD-RW drive.

For Korg's Triton keyboard/workstation combo units there is a new rack-mounted expansion module.

Every show has one or two gadgets that always seem to attract a lot of attention. Zoom had one of those with its handheld PS-02 Palmtop Studio. This little mixer/recorder/FX box records to SmartMedia solid-state media

continued on page 64

NAMM **Wrapup** continued from page 63

cards. Up to 16 minutes can be had on a 32 MB card — 32 minutes at lower quality.

Denon took no chances this year at NAMM, releasing products in numerous recording and playback flavors. Looking for a new MiniDisc recorder? The DMD-100P is tapping the higher end of MD performance. Need a dual-tray CD-R/RW recorder with an onboard HDCD decoder — the CDR-W1500P is there. DVD/CD player? The DVD-1000P is new or if you want a five-disc carousel the DVM-1800P might be your ticket. Or if you just need a dual-well DJ-style CD player, the DN-2100F is stuffed with features. All of this can be sent through the new AVR-2801P A/V Receiver. The 2801 offers onboard Dolby Digital, Dolby Pro Logic and DTS decoding along with a Surround Simulator. If 5.1 is not enough, then go for the 7.1 version, A/V3801P.

Alesis ADAT HD24 hard disk recorder

Denon DM1000 MiniDisc recorder

Software competition

Though dedicated multitrack hard disk recorders are cool, becoming more DAW-like everyday, the growth in the audio industry seems to be in the software realm.

Winter NAMM showed no less than half-a-dozen impressive PC or Mac-based DAW systems. And the number of full-featured computer-based digital music production systems was almost as impressive. Squeeze in the plug-in makers and one begins to suspect that there is enough music software out there for its own trade show.

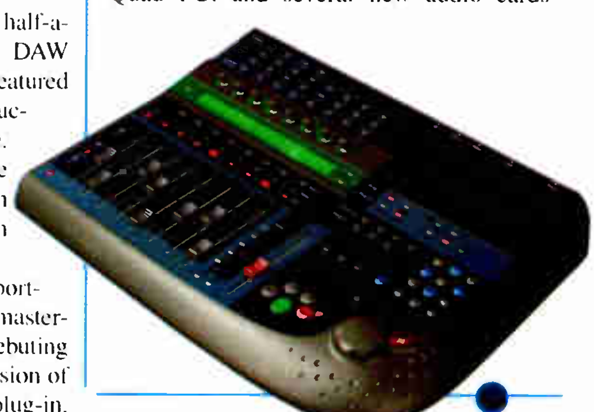
For NAMM audiences, Waves has ported its L2 Ultramaximizer hardware mastering processor to Pro Tools TDM. Debuting was a TDM and a native PC/Mac version of the Renaissance Voice processor plug-in. Included in the plug-in is a compressor, gate and limiter.

One more announcement from Waves concerned a development deal with Yamaha to provide DSP plug-in cards (YGDAI slot) for the AW4416 workstation.

The big news from Emagic and Mackie is that the ever-expansive Mackie will be building a control surface for Emagic's Logic Audio system. Mackie also announced the acquisition of Belgium's SYDEC NV. Although the name may not be familiar, you know them as the designers (and actual owners) of the Soundscape DAW properties.

Emagic showed the diminutive EMI 2/6, a crossplatform USB port box with two ins and six outs. On the software side, Emagic announced upgrades to the Logic platform along with a large number of software instrument packages and plug-ins. Coolest — the EVP88, a vintage electric piano synth (think Fender Rhodes and Wurlitzer). Also announced was increased module support for the latest version of SoundDiver. The WaveBurner CD mastering program is now upgraded to WaveBurner Pro with new features.

Continuing this thread, MIDIman announced a strategic partnership with Emagic for software development of a version of Logic for the Delta sound cards. MIDIman's bigger news, however, was the launching of the SurfaceOne, a futuristic-looking control surface. SurfaceOne utilizes touch/pressure-sensitive Tactex surfaces, fiber-optic technology and MIDI specs. Adding to its traditional line of I/O boxes MIDIman/M Audio showed a USB four-in/four-out, the USB AudioSport Quad PC; and several new audio cards



Steinberg Houston control surface

including a TDIF flavored Delta PCI card and a PCMCIA card.

Digidesign's big news concerned control surfaces and other hardware. Really big and neat is the Control 24, a dedicated 24-channel control surface that looks and acts more like a traditional mixer. Top of the features list are 16 Focusrite microphone preamps and surround sound monitoring. A bit smaller was a new surround sound mixing add-on control surface for the ProControl control surface, Edit Pack. The unit features two joysticks and a built-in QWERTY keyboard. More surround news from Digidesign came in the form of SurroundScope, a metering plug-in with range from stereo all the way up to 7.1.

Steinberg demonstrated a version of Nuendo for Mac. It also showed HALion, a VST-based software sampler and the TC Works Surroundverb 5.1 surround sound reverb for Nuendo systems. On the hardware side, Steinberg showed the Houston control surface for Nuendo and Cubase VST systems; Nuendo 8 I/O, an eight-channel 96 kHz A/D-D/A converter and Midex 8, a crossplatform USB/MIDI interface for Cubase VST.

Cakewalk brought out its brand-spanking new, Sonar system. Sonar is a Windows 98/2000-based multitrack recording/editing package. Fully DirectX-compatible it works with any DirectX plug-in now available.

CreamWare, yet another of the big German hardware/software companies, showed the Luna II, a PCI card sporting the Analog Devices SHARC chips. The Luna II matches up with a FireWire-based rackmounted processor box for a full DAW system. Luna is fully compatible with CreamWare's Pulsar and Scope systems.

And speaking of Scope, CreamWare also showed a couple of the Mongo-big full-length (most PCI audio cards are actually half-length) PCI processor cards - including one with 16 SHARC chips on it.

Not to be left out, TC Works showed the upgraded Spark 2.0, a Mac-based DAW system with some new goodies and an improved engine. But the real treat from TC Works was the PowerCore, yet another Mongo-big full-length PCI card. This one boasts four Motorola DSP chips and, get this, its own Apple Power PC (Motorola) CPU onboard.

PowerCore can power almost any Mac-based DAW system.

Winter NAMM gave many their first look at the resurrected PARIS DAW system, now from E-Mu. Called PARIS Pro, it features a rackmounted expansion/processor box, expansion modules and a separate control

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Digidesign Control24 control surface

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surface with faders. A computer (PC or Mac) is used as a controller.

Native Instruments, makers of the Reaktor program, showed a high-powered frequency delay, NI Spektral. The idea is that individual tightly defined frequencies can be specifically delayed to create unusual effects or process feedback or noise.

Lynx Studio Technology showed its follow up to the original Lynx card. The crossplatform LynxTWO promises 24-bit/192 kHz specs and onboard SMPTE time code.

Universal Audio, showing a distinct departure from its retro line, showed a prototype of its UAD-1 PCI card with some serious processing horsepower. UA was cagey about the identity of the processing chip but it demonstrated the ability to stack processing plug-ins on top of each other across dozens of channels — all without choking the system. UA also showed plug-in versions of its retro processors such as the 1176LN.

Universal Audio's software division, Kind of Loud Technologies, showed several Pro Tools 5.1-compatible surround sound plug-ins including versions for Dolby Digital and DTS-DVD and DTS-CD.

ART D/I/O preamp system

In something of a departure from its usual business, live sound products, Crest Audio debuted a FireWire I/O box for computer hard disk recording and editing. The FB-88 was codeveloped with Digital Harmony and is crossplatform, and sports eight channels of I/O (ADAT or analog).

NAMM is definitely a show for software/computer music fanatics. As hinted above, many of the large DAW programs offer some sort of software musical instrument support or programming. Besides those there are numerous companies that specialize in such products.

Top of that list would be NemeSYS Music Technology and its Giga products. New at the show were several sample libraries including woodwinds, bass and even tympani. Each library contained hundreds of megabytes. New loop/music production tools could also be seen at Cycling 74 (radiaL) and Propellerheads showed the latest incarnation of its Recycle! production tool and Reason software instrument program.

With so many programs and so much processing horsepower, storage is increasingly becoming a concern. StorCase Technology was one of the companies to make the trip to the Orange County Convention Center. Its product line includes outboard hard disk drives and SCSI and ATA/EIDE RAID arrays in various sizes — from one-disk desktop models to nine-bay towers and rackmounteds.

Processing the hardware way

No doubt that software processing was big at NAMM but traditional hardware makers had new products to offer.

While ELOP II sounds like the name of an Egyptian pharaoh, it is really the name of the latest two-channel electro-optical limiter from Manley Labs.

Universal Audio showed its eagerly awaited 2-610 mic pre. The 2-610 is modeled on Bill Putnam's original mic pre for the 610 console.

Making a return to the processor industry, Alesis rolled out the first pair of processors for the upcoming 400 series. The CLX-400 is a compressor/limiter/expander and the PEQ-450 is a parametric EQ. The line is aimed at the budget market.

But two new processors proved to not be enough for Alesis. Following up on the AES release of the airFX is the airSynth. Utilizing the same motion detector to control the processing as the airFX, the airSynth generates sounds rather than manipulating them. Alesis also debuted Ineko, a grid-based processor aimed at the DJ market.

Apex, distributed by Yorkville, showed a two-channel 15-band graphic EQ with VU meters, the AQ1.

Long-time coming for Drawmer was the DC2496 High Resolution A/D Converter. The DC2496, as the name implies, is a 24/96 converter with redither and format conversion functions.

Blue was the color for the new S Class line of processors shown by Samson. New at the show was the S-Vox, a four-channel mic-pre/I/O box aimed at the computer hard disk recording market. The S-Gate 4 is a four-channel expander/gate/ducker.

The hot item for PreSonus was the DEQ624 graphic EQ/digital processor. It is a two-channel, 31-band graphic EQ with a 24-bit digital processor (compressor, limiter, dynamics) section.

Fans of the EL8 Distressor from Empirical Labs were happy to see the EL7 FATSO (Full Analog Tape Simulator and Optimizer).

TC Electronic's new division, Helicon, showed the VoicePrism Plus, a digital voice processor with powerful DSP effects such as chorus, reverb, harmony and EQ.

From Australia, Lake showed its TheaterPhone HSM6240 — a surround sound headphone processor. The format is Dolby Headphone (which utilizes Lake's own technology).

It's a small mixer world

Small mixers were in for Yorkville. First there was the its-bitsy MicroMix M8, an eight-channel powered mixer. And Yorkville's Apex line showed a DJ-style mixer, the AX2.

Midas is expanding downward with the Venice family. The Venice series offers eight mono or 16 mono or 24 mono inputs along with four stereo inputs on each board and include features from the Heritage and XL boards.

The DN-X800 was Denon's offering to the NAMM gods. Besides the usual DJ-

Panasonic DA7mkII digital mixer

style mixer features the DN-X800 offers digital S/PDIF outs.

Covering all the angles, Studiomaster debuted a traditional tabletop mixer, a rackmount mixer and a DJ-style mixer. The Trilogy 406 is a large multi-application mixer with 32 XLR inputs. The 16-2 BP is a rackmountable mixer with 16 XLR inputs. Perhaps stealing the thunder from these two is the Fusion, a DJ mixer with a

Studiomaster Fusion DJ-style mixer

lot of features onboard including DSP, video inputs and three-band EQ on almost every input.

The prolific Samson produced a new family of small mixers, the MDR line. The line is available is 6, 8, 10, 16 input models.

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feature

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Soundcraft debuted its new Spirit M Series. The Spirit Ms come in four mono/four stereo, eight mono/four stereo, 12 mono/four stereo input channels and can operate as tabletop or rackmount. Crest showed the XR-20, a rackmounted 20-input mixer that is to be the first of a family.

As if small wasn't enough, Behringer debuted the MXB1002, a tiny five mono/four stereo input mixerette that can run on batteries.

Yamaha showed mixers small and consoles big. The MX12/6 (eight mono/two stereo inputs) and MX20/6 (16 mono/ two stereo inputs) are smallish multipurpose mixers. The EMX3000 is a powered box mixer with a 300W amp and eight mono/two stereo input channels.

On the bigger end, Yamaha debuted the MC series (24, 32-channel models) of lightweight, ergonomically improved live sound consoles aimed at the cost-conscious market.

Yamaha MC3212 console

Shure did not release a new mixer but it did debut a console aux send expander called, appropriately enough, the AuxPander. The AuxPander is 8 x 8 and linkable to create a 16 x 8 matrix.

Shure AuxPander

TASCAM debuted a DJ-style mixer, the X-9 Performance DJ Mixer. The X-9 features a number of effects such as reverb, delay and echo along with a parametric EQ. Rane, too showed a new DJ mixer. The MP44 has output limiters, a remote master volume control (for controlling out of control DJs) and emergency page ducking.

And in a new twist (actually the second go 'round) Furman Sound showed its own DJ-style mixer, the DJM-80. It's a four-channel model with three-band EQ.

On the heftier side, Panasonic's audio arm (rumors of its demise were exaggerated apparently) showed the upgraded DA7mkII. The mkII offers improved software, operation and cosmetic features. Panasonic also showed the last piece of the converter puzzle with the debut of the WZ-DA96 D/A converter. The DA96 joins two A/D — all three are 24/96.

Lucid GENx6 sync generator

Lucid debuted four new digital audio distribution amplifiers. The AESx4 and the CLKx6 offer four AES and six word clock or Superclock signals respectively. The SPDIFx7 offers SPDIF, AES and TOSlink outputs; the GENx6 distributes and generates sync.

Land of powered & plastic speakers

Making a big move into the industry, Soundbridge Acoustic Labs showed a lot of speakers. Newest for them are a group of powered live sound speakers, the 4000 series. Including 12-inch and 15-inch full-range speakers and stage monitors, the line also adds an 18-inch subwoofer.

Making a move into the powered live sound speaker market, EAW showed the E-Powered series. The EP starts with a two-way speaker, a three-way speaker and a dual 15-inch compact subwoofer. The family offers pole-mounting arrangements.

Electro-Voice too showed a new line of live sound speakers with an emphasis on subwoofers. The Rx line consists of 12 inch, 15 inch and dual 12 inch full-range speakers and 18 inch and dual 15 inch and dual 18 inch subwoofers.

Yorkville hit the trifecta by showing live sound speakers, contracting speakers and studio monitors. Big on the live sound side were the elite series and the NX series. Both series are powered. Shown were the EF500P, a 15-inch two-way speaker with almost a 1,000 W in power and the NX520P, a 12 inch two-way speaker in one of those molded cabinets that seems to be sweeping the industry.

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GREAT MINDS THINK ALIKE.

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1-Computer Music, January 2001; 2 Sound on Sound, January 2000
** - Dependent upon CPU resources. Multicard drivers for Mac coming soon.*

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Tannoy CMS 50 speaker

NAMM Wrapup continued from page 68

For the contractors in the audience, Yorkville's C110 and C120 mini speakers from the Coliseum line were on tap. Most interesting was the rebirth of Yorkville's YSM1 studio monitors. Now available in a powered version (YSM1P) and a slightly re-design of passive version (YSM1i). Each is two-way with a 6.5-inch woofer.

Hafler TRM8 now in vinyl

Live sound speakers were clear- lythe emphasisfor Behringer. New for the show were the Eurolive series — consisting of two standard full-range speakers (15 inch and 12 inch), two stage monitors (15 inch and 12 inch) and a sub-woofer (18 inch). Also new was the Ultrawave B300, a powered PA speaker with an onboard preamp and EQ. For the studio, Behringer brought out the Truth B2031 powered monitor (8.75-inch woofer).

Much of JBL's emphasis was on the EON G2 powered speaker line. Added to the 15 inch is a 10 inch full range and a subwoofer along with packaging options (including mixer, cables and microphones). JBL also brought along samples of the new VerTec line array series.

Yamaha brought out the MS400, a plastic-cased powered speaker (300W amp, 15 inch woofer). Crest's live sound speaker line is the LQ series — available 10 inch, 12 inch and 15 inch full range sizes. Add those to Mackie's SRM450 and one begins to wonder how large the market for these speakers can be? It is true that Mackie's Greg Mackie and Yamaha's Wayne Hrabak were seen at the Yamaha speaker display discussing their respective speakers.

Some of the big news at the Akai booth did not concern Akai products at all but the HK Audio line of powered live sound speakers, which Akai will now distribute in the US. The German company makes mid and large speakers for the live sound and DJ markets.

For processing those big live sound systems, BSS showed the latest in the "Drive" line — in this case the FDS-336 and FDS-334 Minidrives. Both have two inputs while the 336 has six outputs and the 334 has four. Like other drive units they feature crossover controls, parametric EQs, limiters, delay and boodles of filters and memory functions.

Samson's Expedition series of all-in-one powered PA systems with wheels saw new members and new package/features.

Showing its expanded line, this time into contracting, FBT showed contractor market speakers and a line of processors.

Over on the studio side another entrant in the Alesis new product bonanza was the ProLinear 820, a biamplified monitor with onboard DSP (four-band parametric EQ, high-pass filter, gain control ...).

Hafler seemingly dipped its TRM powered speaker family into the vinyl vat. Renamed TRM6.1, TRM8.1, TRM10.1 and TRM12.1, all retain their respective previous construction but add a vinyl-coated exterior.

Amps looked to be the odd man out this year.

One of the few companies showing a new power amp was Yamaha. The CP2000 is capable of cranking out 1,000 W at two ohms or 2,000 W at four ohms, bridged.

BGW displayed a new group of amps in the GT family. Designed for live sound/touring applications, the line is Class H and runs up to 1,400 W per channel.

If you want your equipment cool and quiet, Noren Products demonstrated equipment enclosures that might be right up your alley. AcoustiLock enclosures utilize "Heat-Pipe" technology to wick away heat and gaskets to kill noise and vibration.

Those wacky microphones

We are living in an era of creative microphone design. Unusual shapes proliferate and ribbons and tubes are back. Audix gave us a quick peek at its new lollipop-shaped studio microphone, the SCX25.

Reworking a successful product was Earthworks's approach. The new SR68 is a hypercardioid version of the SR69 cardioid microphone.

Shure offered the big and the small. First there was the NAMM debut of the KSM44 multipattern studio mic. Also debuting was the Beta 53, a lightweight headworn microphone aimed at the theater and presentation markets.

Marshall Electronics brought out a low-cost condenser for the instrument mic market, the 603S. Sennheiser gave

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"I thought I owned the best preamp... ...until I heard the Aphex 1100."

Stephen Krause, award winning recording engineer and producer with over 60 films, 10 TV series and 20 records to his credit, is always in search of better tools. He compared just about every preamp that came on the market to his favorite. Nothing impressed him– until he tried the Model 1100 tube preamp from Aphex Thermionics.

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Manley Labs Langevin
Dual Vocal Combo Mic Pre/EQ/Limiter

by Russ Long

Manley Labs, one of the most respected names in high-end audio, purchased the rights to the Langevin name and circuit designs in 1992. Manley reserves the Langevin name for all its solid state products.

The new Langevin Dual Vocal Combo is a combination of two successful products — the Langevin Dual Mono Mic Preamplifier with shelving EQ and the Langevin Dual-Channel Electro-Optical Limiter.

Features

The 20-pound Dual Vocal Combo's signal path is 100 percent discrete. There are no op-amps in the signal path. In fact, the only op-amps in the unit are those used to drive the LEDs in the opto circuit and the meter circuit. The box's microphone input transformer is custom wound by Manley Labs in its in-house magnetics department.

Each channel includes a front panel accessible high-impedance (150 k-ohms) 1/4-inch jack for the direct input of bass and electric guitars, synths, samplers, drum machines, etc. The rear panel XLR input is interrupted when a cable is inserted into this jack.

The microphone preamplifier and equalizer are roughly based on the vintage Langevin AM-4 channel strip. This section has a frequency response of +/-0 dB from 10 Hz to 20 kHz and <0.05 percent THD at 1 kHz. The limiter is the same as the Manley Electro-Optical Limiter (also similar to the limiters found in the acclaimed Manley VoxBox) except with a discrete transistor gain make-up stage rather than a vacuum tube gain make-up stage.

The rear panel of the Dual Vocal Combo includes a female XLR connector for each channel's microphone input, two male XLR connectors for the balanced line outputs and two 1/4-inch jacks for unbalanced line outputs. There are also two TRS 1/4-inch jacks to directly patch into the limiter and bypass the microphone preamplifier and EQ. This allows the limiter to be used in conjunction with another mic pre, or in a mix situation.

The problem is, as soon as you insert a plug into the Limiter In jack, it breaks the normal from the EQ into the limiter. This is fine if you always have easy access to the rear panel of your Dual Vocal Combo. I don't like having to climb behind the rack every time I want to switch from one input to another. I wish Manley had added an additional front panel switch for each channel that selected either microphone or line input. Having female XLR connectors for line input instead of 1/4-inch TRS would be more convenient as well.

A standard IEC connector on the rear panel provides power to the box. Two ground terminals are also included on the rear panel. These posts are intended to assist in installations where a special grounding scheme is being used. The top post is the audio circuit ground and bottom is the chassis/AC ground.

The front panel is beautifully finished in brushed red aluminum and is factory-guaranteed to get plenty of second looks. The knobs and switches are, in typical Manley fashion, all of the highest quality.

Input Attenuate is the input gain control for the microphone preamp. Its range goes from off to 45 dB of gain. The phantom power switch activates 48V phantom power for each respective channel. This is a special locking switch that must be pulled out before it can be turned on or off to prevent accidental switching. Each phantom power switch has an LED that glows when phantom power is on.

Each EQ section has a bypass switch and two frequency select switches. One switch sets the corner frequency for the low frequency shelf as either 80 Hz or 40 Hz shelf and another sets the corner frequency for the high frequency shelf as either 12 kHz shelf or 8 kHz. LF and HF knobs allow the EQ to be adjusted from -10 dB to +10 dB.

In LA-2A fashion, the limiter controls are minimal, allowing for speedy setup and ease of use. The In/Bypass switch takes the limiter in and out of the signal path. The Reduction knob adjusts the limiting threshold. The optical limiter's attack and release times are fixed at 20 milliseconds and 500 milliseconds, respectively. The Gain knob controls the make-up gain after the limiter.

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At a Glance

Applications:
Studio; project studio; broadcast; live sound

Key Features:
Two-channel discrete microphone preamp/EQ/electro-optical limiter; XLR I/O and TRS limiter line input; switchable phantom power

Price:
$2,000

Contact:
Manley Labs at 909-627-4256; *www.manleylabs.com*; or circle **Reader Service 72.**

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Manley continued from page 72

In the center of the box's front panel are two VU meters. The meters are smaller versions of standard Sifam VU meters. Each meter has a corresponding switch that determines whether the VU meter monitors final channel output (if Meter Output is selected) or how much limiting is occurring (if Reduction is selected). The Sep/Link switch lets the two limiters act independently (if Sep is selected) or collectively (if Link is selected).

The manual is well written and easy to understand. It includes a helpful section on cable wiring as well as a recall sheet that can be copied and used to document settings. All Manley and Langevin owner's manuals can be found online at *www.manleylabs.com*.

In use

Initially I found it a bit confusing that the left and right channels are exact mirror images of each other, with the exception of the EQ, which follows the same footprint (LF controls on the left and HF on the right).

The left EQ knob adjusts low-frequency EQ for both channels but the left limiter knob adjusts Gain on the first channel and Reduction on the second.

It would make better sense to me if the left limiter knob on both channels adjusted the Reduction and the right adjusted the Gain. Once I became used to this oddity, however, I stopped cranking the gain of the first channel every time I wanted to adjust the threshold. I also found the lack of a phase reverse switch disappointing.

After extensive use, I concluded that, despite these minor flaws, the Dual Vocal Combo is an extraordinary machine. It sounds great and is easy to use and extremely fast to achieve the end result.

I've been putting the box to the test over the last several weeks, using it in a wide variety of circumstances. I have yet to be disappointed. The DVC sounds wonderful on electric guitars. The preamp is punchy and full with a tight bottom end and the EQ allowed me to add top-end sparkle without getting harsh.

The Langevin Dual Vocal Combo is a fantastic bargain, offering a superb performance-to-price ratio.

It worked equally well with bass guitar. I especially like the low-end definition I was able to attain. When boosting the low frequency, the bottom got bigger without getting muddy or undefined.

I used the Dual Vocal Combo along with a pair of Earthworks SR77 microphones and a Steinway Grand piano to achieve an amazing piano sound for a jazz album I recently completed. The limiter worked especially well with the piano giving the precise amount of control without sucking the life out of the performance.

I used the Brauner VM1 Klaus Heyne Edition microphone (*PAR*, 10/00, p. 22) through the Dual Vocal Combo, yielding what is perhaps one of the finest vocal sounds I have ever achieved. I also had exceptional results using the Sony C-800G to record vocals through the Combo.

With the limited flexibility of the Dual Vocal Combo's equalizer there were times when I found it necessary to add my GML equalizer between the Combo and tape (or hard disk in some instances) to allow me to attain the desired sound. The lack of any midrange equalization and the lack of ability to adjust the equalization bandwidth is the only real sonic limitation I found in the box. As long as the original sound source is of reasonably good quality and the microphone is decent, the box will work great. If you are looking for the kind of EQ that will really let you tweak and twist a sound into something that it isn't, however, this is not the machine to use.

The mic preamp and EQ also sound superb on drums and percussion, although I found the limiter was typically too slow to be effective on kick, snare and toms. The limiter is very "LA-2A-ish" in its compression characteristics — it tends to work best on vocals (thus the name Vocal Combo), bass, guitars and keyboards.

Summary

At $2,000, the Manley Dual Vocal Combo is a fantastic bargain offering a superb performance-to-price ratio. Two channels of vintage Langevin AM-4 modules, without any limiting whatsoever would likely cost well over $2,000 if the price included rack-mounting and fitting the box with a power supply.

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

Product Points

Manley Langevin

Plus
- Classy Manley electro-optical limiting
- Great-sounding preamp
- Reasonable price

Minus
- No mic/line input switch — requires manual patching
- Confusing front panel layout
- No phase reverse switch

The Score
Overall, a superb value at $1,000 per channel of excellent preamplification, EQ and limiting.

Stewart Audio
World 1.2 Power Amplifier

by Roger Williams III

Founded by Wade Stewart in 1982, Stewart Electronics develops innovative niche products aimed at the pro sound contractor and MI market. Early on, the company produced a variety of half-rack-space amplifiers, preamps and mixers that were later consolidated into a line of power amplifiers designed to deliver high power from a small, lightweight package.

Stewart Electronics (now Stewart Audio) was one of the first companies to adapt the high-speed switching power supply to audio power amplifier applications, reducing conventional power supply transformer sizes (and weight) significantly. The Stewart World 1.2 power supply — by virtue of its design — recharges 1,000 times faster than conventional Class AB types, thus requiring less space for storage (capacitors).

Although other manufacturers have introduced advanced designs — such as the Carver PT1250 or the QSC PowerLight — that are called "Pulse Width Modulated" or "Class H," the aim is the same: providing higher power at lower mass. Any road dog (and his chiropractor) will vouch for these virtues.

Features

The Stewart World 1.2 amplifier is a two-channel single rack-space unit that weighs 11 pounds, delivering 420 Watts at 4 ohms per channel and 1,200 Watts at 4 ohms in bridged mono operation. Frequency response is claimed from 20 Hz to 20 kHz with an A-weighted S/N ratio of 100 dB and a quick slew rate of 30 Volts per microsecond.

A high input impedance of 20 kilohms helps impedance matching with most available input sources. Current sensing circuitry detects when load impedance drops below 0.5 ohms and disconnects the power supply, protecting the unit and connected speakers until the perceived "short" is removed. At that point the amplifier will slowly ramp up to full power.

An inrush current limiting circuit minimizes power draw at initial turn-on, which may help to preclude the need for power sequencing devices.

The heavy, black-powder-coated aluminum faceplate features a centrally located power on/off switch with a red LED power indicator and detented level controls with a green LED –20 dB signal indicator and red LED clip lights for each channel. Massive heatsinks along the sides lead to the back panel. Each channel has a balanced XLR and TRS 1/4-inch input, connected in parallel to facilitate daisy chaining with other amplifiers if desired.

A bridge/stereo mode switch and 15-Amp breaker are to the left of the power cord strain relief. A pair of five-way binding posts for speaker connections and 1/4-inch phone output jacks are wired in parallel. The nearby binding post sets are conveniently to facilitate bridging with dual banana plugs.

The unit is solid and has a good fit and finish; indeed, prolific use of Torx-style screws prevented me from popping off the lid and checking under the hood, which chassis connection-straddling stickers warned against.

In use

I was anxious to test the World 1.2 with a pair of what I consider to be top-shelf compact speakers — Audio Composite Engineering 1160s. I was in a small club setting with an ensemble acoustic act. I had a Mackie VLZ mixer and Shure Beta 58 microphones for vocals. The Stewart exhibited exceptional control — punchy midrange and crisp, high-mid consonants for speech. An airy high-frequency quality nicely translated high E and B strings on an Ovation acoustic guitar via a Shure SM 81. In this application, I liked what I heard from the World 1.2.

I also had favorable results in my project studio, although the Stewart could not match the transparent sublime openness of my Hafler Pro 3000 through Tannoys — but this may be a case of apples and oranges.

And while I did not have the chance to try it as such, the World 1.2 would probably be a great fit for a foreground distributed system in a restaurant, club or multiple room residence — the sonic quality and ability to drive low impedance loads is certainly well suited for these uses.

Summary

At a list price of $999, and with all the advantages of power versus weight and size, the Stewart World 1.2 offers good value on the investment. With several application options, great sonics, solid construction and a three-year warranty, this amplifier is a quality option in a crowded marketplace.

*Roger Williams III, a systems designer for MAS Audio, longtime NSCA, ICIA member and Syn-Aud-Con grad, is a regular contributor to **Pro Audio Review**.*

At a Glance

Applications:
Live sound; contracting and installation

Key Features:
High power output; compact, light weight package; balanced XLR and TRS 1/4-inch inputs; 1,200-Watt output at 4 ohms bridged; solid construction; good sonics

Price:
$999

Contact:
Stewart Audio at 209-588-8111; or circle **Reader Service 74**.

3D Audio Pre CD Volume 1

by Russ Long

Choosing the right microphone preamplifier is one of the most important decisions an engineer can make. Today the options are more diverse than ever with radical differences in price, performance and features from one preamp to the next. Ultimately, the choice boils down to personal opinion — making the decision even more difficult. Nashville's 3D Audio has a solution: a CD featuring recordings of identical sound sources using 33 different preamps.

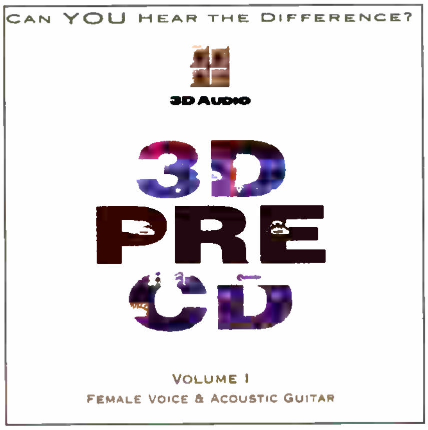
It is obvious that 3D Audio's Lynn Fuston put much time, research and preparation into this project. The order of the preamps is completely random and the disc is designed so the listener can have a blind listen with no preconceptions. The CD's liner notes offer complete technical details about the project's creation, including a description of how the preamps were adjusted within .02 dB of each other. The CD includes recordings of both female voice and acoustic guitar. The recently released Volume 2 includes male vocal and snare drum.

I spent about an hour with engineer Micajah Ryan (whose credits include Bob Dylan, Megadeth and John Prine) listening through the female vocal recordings. We each kept notes and refrained from any discussion until the listening was complete. At that point we compared notes and I must say it was a phenomenal experience. We were both surprised many times over as our preconceived notions were often confirmed but sometimes contradicted.

There were some very expensive preamps that we both agreed just didn't sound very good, and there were some lower priced pres that we both quite liked. Although we agreed more often than not, there were some preamps that I liked a lot that Ryan was not fond of at all, and there were some that he thought sounded fantastic that I didn't find to my liking.

More than anything, this CD allows listeners to compare, as accurately as possible, an extremely wide range of microphone preamps.

Although combining 33 different preamps in one location is a mighty feat, I was a bit disappointed that a few of the industry's mainstay preamplifiers were not included in the line-up. I would have liked to hear the



Trident A Range, Neve 8108 and the Neve 8078. It also would have been nice to hear some of the common console preamplifiers, such as those found in the SSL and Neve VR desks. Maybe future editions of 3D Audio's Mic Pre CDs will include 50 or more preamp varieties.

Needless to say, the 3D Pre CD is quite possibly the best $29.95 that any engineer — from a student to a seasoned pro — can spend. I anticipate it will be an eye-opener to anyone brave enough to listen. This, plus other listening CDs, can be ordered from the 3D Audio Web site at *www.3daudioinc.com*.

Audix
VX-10 Condenser Microphone

by Tom Young

The latest addition to Audix's microphone line is the VX-10, designed as a handheld vocal condenser microphone that can also be used for a wide variety of applications in live, studio and broadcast situations. My previous experience with Audix is with the CX-111 large-diaphragm mics, which I use as drum overheads. Since I am pleased with those, I was looking forward to trying the VX-10 mic. At a suggested retail price of $599, it is worthy of consideration by anyone desiring a quality handheld condenser at an affordable price.

Features

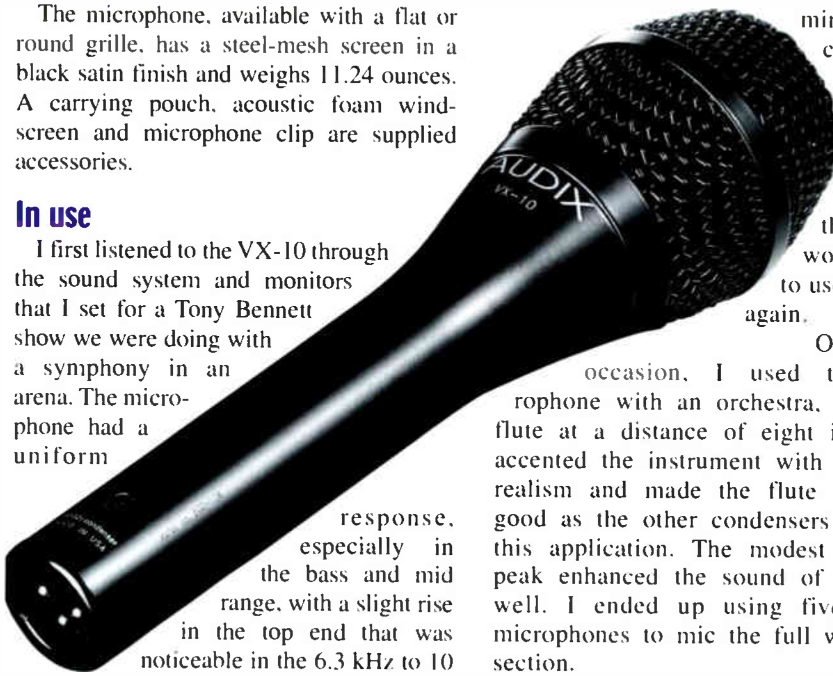
The VX-10 is a cardioid-pattern condenser microphone requiring 48 to 52 V of phantom power. The microphone features a 16mm gold vaporized diaphragm and a frequency response of 40 Hz to 20 kHz. With a

> **Both singers commented that they would like to use this microphone again.**

maximum SPL rating of 138 dB without distortion, a signal-to-noise ratio of 73 dB, and a floor noise of 19 dB SPL (A-weighted), it is a quality product.

The Audix VX-10 is designed with a field-replaceable threaded capsule, enabling the user to carry spares to service the microphone in unexpected situations — a well-thought-out feature. Simply unscrew the capsule and replace it. The male XLR connector is gold plated.

The microphone, available with a flat or round grille, has a steel-mesh screen in a black satin finish and weighs 11.24 ounces. A carrying pouch, acoustic foam windscreen and microphone clip are supplied accessories.



In use

I first listened to the VX-10 through the sound system and monitors that I set for a Tony Bennett show we were doing with a symphony in an arena. The microphone had a uniform response, especially in the bass and mid range, with a slight rise in the top end that was noticeable in the 6.3 kHz to 10 kHz range.

In a room that had the tendency to accent low frequencies, this microphone was impressive in its clarity. I liked its minimal proximity effect and its frequency response. The VX-10 also performed well off axis. And, when I pulled it away to a distance of 18 inches, the microphone exhibited a very uniform response.

I also used the VX-10 on a corporate show that featured a dance band with male and female vocalists. They used the VX-10 as their main vocal microphone. The main PA for the show was a Turbosound Floodlight system with EAW monitors. The PA and monitors were tuned using pink noise and a CD to a relatively flat curve.

Both singers sounded natural and full with no required EQ on the console input. The contoured rise in the top end provided a nice intelligibility to the sound without any sibilance problems. This also provided a nice cut for the singers in the monitors. The VX-10 provided a hot level in the monitors that was very stable with minimal EQ contouring required. Both singers commented that they would like to use this mic again.

On another occasion, I used the microphone with an orchestra, miking a flute at a distance of eight inches. It accented the instrument with a natural realism and made the flute sound as good as the other condensers I use in this application. The modest presence peak enhanced the sound of the flute well. I ended up using five VX-10 microphones to mic the full woodwind section.

I also tried the microphone on a guitar amplifier. Placed close to the speaker, it provided a nice clear sound as a result of its minimal proximity effect. The dynamic mics that I usually use on guitar have much more low-frequency emphasis at close range, requiring more EQ.

Summary

The Audix VX-10 is a good-sounding microphone for vocals that is also well suited for a variety of instruments. It is a microphone that provides all the right characteristics and sturdiness to survive the rigors of live performance; the field replaceable capsule is a bonus. Whether used in a live performance, recording or broadcast, this microphone at this price point will not disappoint.

Contact: Audix at 800-966-8261; 503-682-6933; *www.audixusa.com*; or circle **Reader Service 75**.

*Tom Young, a regular contributor to **Pro Audio Review**, is currently the live sound engineer for Tony Bennett.*

Back Off, Mic

by Loren Alldrin

The microphone you use and where you put it are two crucial factors in capturing a great recording. I would like to discuss a mistake I see (and hear) committed frequently by aspiring engineers — miking too closely, too often.

Close miking is when you stick a microphone within a foot or two of an instrument, although some purists would claim that 6 feet is mighty close. The common practice of putting a microphone 2 or 3 inches from the noisemaker certainly qualifies.

Why do people place the microphone so close? There are many reasons, some of which are more valid than "because that is how they do it in the music videos." For starters, close miking effectively quiets the hum of the nearby computer fan, the clamor of traffic outside the bedroom window and the babble of a roommate's television. It also minimizes room ambience, which sounds awful in the small, untreated rooms many people record in.

Poking the mic in close also results in a thick, full sound, courtesy of the bass-enhancing proximity effect of directional microphones. Top-end detail also improves, as higher frequencies tend to be strongest right at the sound source. Finally, close miking gets maximum signal strength out of the mic, which means less preamp gain (and usually less noise).

Too close for comfort

With all the benefits of close miking, it is easy to lose sight of the drawbacks. And there are several.

First, proximity effect can make mud out of individual tracks and full mixes alike. Close-miked sounds usually have enhanced low-mid frequencies, which can cause a serious buildup as tracks are combined. This buildup is hard to overcome, even with generous use of EQ. Unless your arrangement is spot-on, maintaining clarity in a mix is a challenge, even without proximity effect problems.

Second, close miking can result in an incomplete sonic picture of an instrument. Imagine a directional microphone placed a few inches from an upright bass. The mic emphasizes the range of frequencies coming from directly in front of it, at the expense of those off to the sides. Place it in front of an f-hole, and all you get is body resonance. Point it at the fingerboard, and you get mostly the clack and buzz of the strings. When was the last time you listened to something from 3 inches away?

A close microphone position gives the perception of an upclose, urgent sound because it downplays reflections and ambience from the room. Mic all your tracks this way, and suddenly every instrument in the song is clamoring for front-and-center position in the mix. When things get crowded, the depth of your music suffers. Can artificial ambience (reverb) help the crowding? Yes, to some degree. Do some types of music sound good with every part jumping out of the speakers? Certainly, but many do not.

> The mic you use, and where you put it, are two crucial factors in capturing a great recording.

Finally, close miking accentuates a sound's dynamics because it eliminates the smoothing effect of room ambience. Loud sections seem louder as a result, and you may need to apply more compression to keep things on the level. Dynamics are hard enough to control, especially with unseasoned musicians.

Don't stand so close to me

Although you may have to soundproof your room a bit, or put your computer in a wooden box, pulling your mics back can dramatically improve your music.

You can say goodbye to proximity effect. Sounds will have a more balanced frequency range, which means less EQ. You may find yourself doing far less low-frequency cutting and high-frequency boosting to maintain clarity in your mixes. Sounds will be thinner than their close-miked counterparts, but that is often just what the mix ordered.

Miking from several feet away restores early reflections and reverb to the recording. This makes for more perceived space around the sound, as well as a greater perceived distance from the listener. Mixing the resulting tracks with others that were close-miked lets you build front-to-back depth into your mixes. Ever marveled at a mix that was just wonderfully open and spacious? Odds are, not all the instruments were recorded with close mics.

Although there are other benefits, the last major one is a more balanced, holistic sonic picture of an instrument. Moving the microphone back allows it to catch all the sound radiating from an instrument, not just the small area directly in front of it. Instruments recorded from several feet away sound more like they do to the naked ear.

Backing off

The next time you record an acoustic guitar, banjo or mandolin, try miking it from 3 or 4 feet away. Use just two mics on a drum kit, and position them about 6 feet away for starters. Push background vocals back in the mix by miking them from several feet. Take your usual miking distance and quadruple it by default, at least until you get a feel for how greater miking distances sound.

Distant miking is not the answer for every recording situation, but it is well worth experimenting with. Now back off!

*Loren Alldrin is **Pro Audio Review's** project studio editor and author of* The Home Studio Guide to Microphones.

Audio Processors

Peavey CEL 2 Compressor

Features: Two-channel; threshold, ratio, attack release controls for compressor; threshold control for limiter; threshold control for expander; side chain; bypass. **Price:** $279.
Contact: Peavey at 601-483-5365; or circle **Reader Service 91.**

HHB Fat Man 2 Tube Preamp Compressor

Features: Single-channel; preamp/compressor; 48V phantom power; low-cut filter (90 Hz); gain input, output, makeup controls; threshold, ratio control for compressor; 15 factory presets; 12AX7A tube; VU meter. **Price:** $469.
Contact: HHB Communications at 310-319-1111; or circle **Reader Service 92.**

Manley Laboratories ELOPII Limiter

Features: Two-channel electro-optical/FET limiter; input, attack release, output controls; onboard mic preamps with 48V and phase reverse; linkable;VU meters. **Price:** $3,250.
Contact: Manley Laboratories at 909-627-4256; or circle **Reader Service 93.**

Joemeek VC1Q Studio Channel

Features: Single-channel; mic preamp/photo optical compressor/EQ/enhancer; three-band EQ; parameter controls per stage; 48V phantom power; VU meter; optional digital output card ($199). **Price:** $799.

Contact: Joemeek/PMI at 310-373-9129; or circle **Reader Service 94.**

Alesis airFX

Features: Multi-effects processor; motion sensor parameter control; 50 factory presets.
Price: $249.
Contact: Alesis at 800-525-3747; or circle **Reader Service 9S.**

Lectrosonic DSP4/4 Audio Processor

Features: Four inputs, four outputs; 12 filters per channel; notch filters; EQ; feedback extermination; delay; compressor/limiter; memory settings; RS232 port.
Price: $1,650.
Contact: Lectrosonic at 800-821-1121; or circle **Reader Service 96.**

Smart Research C2 Stereo Compressor

Features: Two-channel; threshold, ratio, attack, release, makeup controls per channel; crush frequency enhancement mode; link; side chain. **Price:** $2,995.
Contact: Smart Research/Sunset Sound at 323-469-1186; or circle **Reader Service 97.**

Rane DC 24 Dynamic Controller

Features: Two-channel compressor/limiter/gate expander; threshold, ratio controls for gate expander and compressor, threshold control for limiter, master output control per channel; controllable crossover; side chain; bypass. **Price:** $599.
Contact: Rane at 425-355-6000; or circle **Reader Service 98.**

Audio Processors

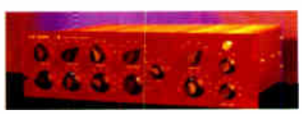
Pendulum Audio ES-8 Variable Mu Vacuum Tube Limiter

Features: Two-channel; input, threshold, attack, release, output controls per channel; six presets; Class A; side chain; bypass; linkable; switchable VU meters. **Price:** $3,495.
Contact: Pendulum Audio at 908-665-9333; or circle **Reader Service 99.**

Symetrix 565E Dual Compressor/Limiter/Expander

Features: Two-channel; threshold, release expander controls; threshold, release, ratio compressor controls; threshold limiter control; output gain controls; stereo link; bypass; side chain. **Price:** $399.
Contact: Symetrix at 800-288-8855; or circle **Reader Service 100.**

D.W. Fearn VT-4 Vacuum Tube LC Equalizer



Features: Single-channel; five-band; frequency, boost, cut, HF Q, input level controls; Class A; bypass. **Price:** $3,900.
Contact: D.W. Fearn at 610-793-2526; or circle **Reader Service 101.**

Yamaha SREV1 Sampling Reverberator

Features: Four-channel; programmable; CD-ROM drive; parameters editable through PC; sample library; optional separate controller with faders; optional expansion board. **Price:** $5,499.

Contact: Yamaha at 714-522-9011; or circle **Reader Service 102.**

Ashly Audio Protea System II 4.24G Multiprocessor

Features: Four-channel; compressor/limiter/EQ; 24-bit A/D-D/A; 24-bit processing; 28-band; delay/time alignment function; high, low-pass

filter; 128 presets; RS232 port. **Price:** $2,599.
Contact: Ashly Audio at 716-872-0010; or circle **Reader Service 103.**

Focusrite ISA 430

Features: Single-channel; preamp/compressor/gate/expander/limiter/opto-de-esser/parametric EQ; split-mode independent section control; Q control; boost control; Class A; optional 24/96 A/D. **Price:** $3,495.

Contact: Focusrite/Digidesign at 650-842-7900; or circle **Reader Service 104.**

Millennia Media STT-1 Origin

Features: Single-channel; one solid state mic preamp; one tube mic preamp; 48V phantom power; phase reverse; one solid state parametric EQ; one tube parametric EQ; one discrete opto compressor path; one tube opto compressor path; VU meter; front panel 1/4" input. **Price:** $2,895.
Contact: Millennia Media at 530-647-0750; or circle **Reader Service 105.**

Audio Processors

TL Audio VP-1 Tube Processor

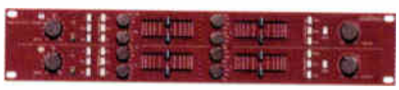
Features: Preamplifier, compressor/limiter, parametric EQ, expander/gate, de-esser; 30 dB pad; variable high-pass filter; controls for each section; EF86, 12AX7A tubes; LED meters; VU meter.

Price: $2,699.

Contact: TL Audio/ HHB Communications at 310-319-1111; or circle **Reader Service 106.**

Trident - MTA A Series Dual Discrete Channel

Features: Mic preamp/EQ; two-channel; input, output controls; four-band EQ; high-pass filter; low-pass filter; 48V phantom power; phase reverse; bypass. **Price:** $3,699.



Contact: MTA/PMI at 310-373-9129; or circle **Reader Service 107.**

Drawmer DC2476 Masterflow Pro Digital Audio Processor

Features: Parametric EQ/expander/compressor/stereo width/tube modeler; 32, 44.1, 48, 88.2, 96 kHz sample rates; 24-bit A/D-D/A;

word clock; MIDI I/O/thru; PCMCIA drive; frequency, bandwidth, boost/cut, threshold, attack, release controls EQ controls.

Price: $3,100.

Contact: Drawmer/TransAmerica Audio Group at 805-241-4443; or circle **Reader Service 108.**

Behringer Ultra-Dyne Pro DSP9024

Features: Dual processing engines; six-band compressor; limiter; de-esser; exciter; noise gate; delay; tube emulator; 24-bit A/D-D/A; Virtuoso mastering process; LCD screen; PC-based remote control software. **Price:** $599.

Contact: Behringer at 425-672-0816; or circle **Reader Service 109.**

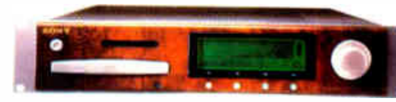
DACS FREQue II Dual Ring Modulator

Features: Twin ring modulators; weight, edge spectral controls; fine, tune, range controls; onboard twin oscillators; frequency shifting; cascadable effects routing; bypass. **Price:** $1,490.

Contact: DACS/Independent Audio at 207-773-2424; or circle **Reader Service 110.**

Sony DRE-S777 Sampling Digital Reverberator

Features: Two-channel; 24-bit; 44.1, 48 kHz sample rates; seven standard reverb programs; reverb time control; word clock; RS232 port; CD-ROM drive; optional sample libraries. **Price:** $5,000.



Contact: Sony at 800-686-7669; or circle **Reader Service 111.**

Shure DP11EQ Dynamics Processor

Features: Gate/expander/compressor/limiter/nine-band parametric EQ; no overshoot-style peak limiter; high, low-cut/shelving filters; Windows software; lock-out function; nonvolatile memory; linkable via ShureLink. **Price:** $800.

Contact: Shure Brothers at 800-257-4873; or circle **Reader Service 112.**

dbx 480 Drive Rack Loudspeaker Management System

Features: Four input channels; eight output channels; crossover controls; pre and post EQ; notch filters; driver alignment; delay; real time analyzer; limiter; LCD screen; LED meters; optional Windows PC remote software. **Price:** $2,495.

Contact: dbx at 801-568-7660; or circle **Reader Service 113.**

Aphex Systems Model 720 Dominator II Peak Limiter

Features: Two-channel; three-band; input gain control; EQ controls; release, density controls; peak limiter controls; Automatic Threshold Limit circuit; bypass; LED meter.

Price: $1,350.

Contact: Aphex Systems at 818-767-2929; or circle **Reader Service 114.**

Eventide Orville Harmonizer Effects Processor

Features: Up to eight-channel; 24-bit A/D; 44.1, 48, 88.2, 96 kHz sample rates; parametric EQ/compressor/limiter/delay/reverb/DSP; over 800 presets; word clock; MIDI I/O/thru; PCMCIA drive. **Price:** $5,695.

Contact: Eventide at 201-641-1200; or circle **Reader Service 115.**

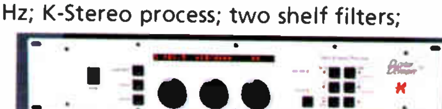
BBE Sound 862 Sonic Maximizer

Features: Two-channel; contour, process controls; bypass. **Price:** $599.

Contact: BBE Sound at 714-897-6766; or circle **Reader Service 116.**

Digital Domain Model DD-2 K-Stereo Processor

Features: 24-bit; 96 kHz; K-Stereo process; two shelf filters; POW-r dither; high-pass filter; low-pass



THE MUSIC OBSERVER

CARTERET, NJ

FINAL EDITION
50 CENTS

SUNDAY, NOVEMBER 15

EXTRA!! EXTRA!!

XPB AMPS "Racks Up To The Competition"

Stable down to 2 Ohms!

Carteret NJ-

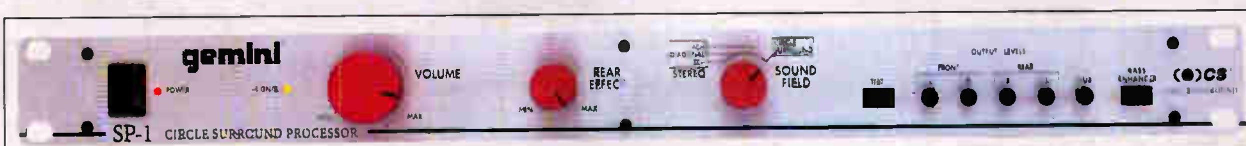
In a stunning upset, Gemini's new XPB series power amps have surpassed the competition in a head to head battle for the mobile DJ and install markets. Sound contractors and DJ's alike have been left scratching their heads wondering how it is possible for this low cost series of power amps to go head to head with the "big three" amplifier manufacturers and emerge victorious. Gemini credits it's over-sized toroidial transformer, filter and heat sinks and 5-way protection circuitry that allows this 2U workhorse to out perform the competition. When working with difficult impedance loads down to 2 ohms, most amps heat up and shut down, but the XPB handles the load with no problems. The crystal clear low end remains intact even after hours of use.

Reports are coming in from all over the world on how both XPB amplifiers offer a wide range of features to meet the most demanding audio professional's needs. Three modes of operation--stereo, parallel mono and mono bridge with front panel mode indicators as well as the ability to daisy chain the active balanced inputs together, make these amps versatile enough to handle any set-up configuration. Other features such as True clip LED's, front to rear airflow with 2-speed fan, turn-on in-rush current limiting circuitry and dual aluminum extrusion heat sinks provide the thermal stability and reliability that are necessary in today's competitive amplifier market, all backed by Gemini's rock solid 5 year parts and labor warranty. Contractors, live sound engineers and mobile DJ's around the world agree –
The XPB series "rack up to the competition".

MAKE	MODEL	2 Ohms	4 Ohms	8 Ohms	4 Ohms	8 Ohms
		Stereo	Stereo	Stereo	Bridged	Bridged
GEMINI	XPB-750	360 W	275 W	175 W	750 W	550 W
CROWN	Power Tech 1	N/A	305 W	220 W	N/A	600 W
CREST	V450	325 W	225 W	150 W	650 W	460 W
QSC	USA 400	250 W	200 W	125 W	N/A	400 W

MAKE	MODEL	2 Ohms	4 Ohms	8 Ohms	4 Ohms	8 Ohms
		Stereo	Stereo	Stereo	Bridged	Bridged
GEMINI	XPB-1600	800 W	500 W	300 W	1600 W	1000 W
CROWN	K1	750 W	550 W	350 W	1500 W	1100 W
CREST	V1100	700 W	550 W	300 W	1400 W	1100 W
QSC	RMX 1450	700 W	450 W	280 W	1400 W	900 W

SP-1 Circle Surround Processor



Carteret, NJ- Gemini discovers a new way to give DJs more control of their mix with the SP-1 Circle Surround(tm) Processor. With the help of SRS™ labs, Gemini has merged function with technology in creating this 2 input, 5 output pre-amp/processor for the DJ market. The product Development team at Gemini reports that Circle Surround™ is a spatial 4 speaker plus subwoofer effect that requires no encoding or decoding. The SP-1 works well with any audio source. The 5 individual level controls and onboard white noise generator provide precise control to help the DJ achieve the exact balance required for the venue. Sid Vanderpool, editor of DJ Zone Magazine, was quoted as saying: "I heard many of my old discs in a completely new way when played in Circle Surround™ through the Gemini SP-1."

The SP-1 has three modes of operation in addition to the Circle Surround(tm) mode (1) Two channel stereo, (2) two channel, four speaker stereo and (3) 2 channel, 4 speaker diagonal stereo.

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READER SERVICE NUMBER 193

2094/1100

Audio Processors

filter; presets; bypass. **Price:** $3,500.
Contact: Digital Domain at 407-831-0233; or circle **Reader Service 117.**

Klark Teknik DN9848 Loudspeaker Multiprocessor

Features: Crossover/system processor; four inputs, eight outputs; parametric EQ; delay; compressor; limiter; Butterworth,

Linkwitz-Riley, Bessel filters; memory; factory presets; security lockout. **Price:** $5,539.
Contact: Klark Teknik at 800-392-3497; or circle **Reader Service 118.**

Sabine Adaptive Audio GRQ3102 Graphi-Q Processor

Features: 31-band graphic EQ/feedback controller/compressor/limiter/delay; two-channel; 24-bit A/D-D/A; 12 feedback notch filters; ratio, threshold, gain controls on compressor/limiter; high, low-cut filters; bypass; ClipGuard; slave version available. **Price:** $1,299.

Contact: Sabine Adaptive Audio 904-418-2000; or circle **Reader Service 119.**

Quantec 2402 Yardstick Room Simulator/Reverberator

Features: 24-bit; 44.1, 48 kHz sample rates; Quantec Room Simulator; MIDI compatible; RS-232 port. **Price:** $2,695.

Contact: Quantec/HHB Communications at 310-319-1111; or circle **Reader Service 120.**

Miles Technology MTI-3 TriSonic Imager

Features: SpreadSound enhancement; surround processing bandwidth; TriSonic bandwidth, surround level, gain, balance controls. **Price:** $599.
Contact: Miles Technology at 616-683-4400; or circle **Reader Service 121.**

ART HQ15 Graphic Equalizer

Features: Two-channel; 15-band; Feedback Detection Circuitry; 6 or 12 dB cut/boost; high-pass filter; low-pass filter; VU meter. **Price:** $299.

Contact: Applied Research and Technology at 716-436-2720; or circle **Reader Service 122.**

Avalon Design VT-747SP

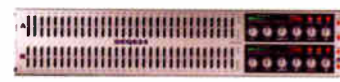
Features: Compressor/six-band parametric EQ; input, threshold, release, compression, makeup gain, output controls; tube and solid state signal paths; Class A; side chain; VU meter. **Price:** $2,495.
Contact: Avalon Design at 949-492-2000; or circle **Reader Service 123.**

Mercury Recording Equipment EQ-P Program Equalizer

Features: Pultec-style; two-band; separate boost, attenuation controls per band; sharp/broad switch; attenuation selector; bypass. **Price:** $2,599.

Contact: Mercury Recording Equipment/Marquette Audio Labs at 510-581-3817; or circle **Reader Service 124.**

PreSonus DEQ624 Stereo Graphic Equalizer Plus Dual Dynamics Processor



Features: Two-channel; 31-band; high-cut filter; low-cut filter; compression, ratio, soft gate, gain controls; 6, 12 dB cut/boost, 24 dB cut range; security lockout; bypass. **Price:** $599.
Contact: PreSonus at 800-750-0323; or circle **Reader Service 125.**

White Instruments ParaMedic 48 System Processor

Features: Four inputs; eight outputs; 24-bit A/D-D/A; delay; crossover controls; parametric, graphic EQs; high, low-pass filters; limiter; matrix router;

RS232/485 ports; networkable; security functions. **Price:** $3,669.
Contact: White Instruments at 512-389-3800; or circle **Reader Service 126.**

Antares Audio Technologies ATR-1a Auto-Tune Intonation Processor

Features: 20-bit A/D-D/A; 56-bit internal processing; pitch correction, scale, speed controls; factory presets; memory; bass mode; fully programmable; MIDI I/O; SysEx; bypass. **Price:** $849.
Contact: Antares Audio Technologies at 888-332-2636; or circle **Reader Service 127.**

API 2500 Stereo Bus Compressor

Features: Threshold, attack, release, ratio controls; soft, medium, hard knee compres-

sion; compression knee type switch; makeup gain, output controls; link; VU meter. **Price:** $2,995.
Contact: API/Transamerica Audio at 702-365-5155; or circle **Reader Service 128.**

Audio Toys, Inc. (ATI) Pro6 Multi Mode Audio Signal Processor

Features: Mic preamp/compressor/parametric EQ/gate/duck; high, low-pass filters; four-band parametric EQ; parameter controls for each stage;

48V phantom power; bypasses; stereo link. **Price:** $1,995.
Contact: Audio Toys, Inc./ Transamerica Audio at 702-365-5155; or circle **Reader Service 129.**

Audio Processors

ImageLink stereo link; bypass; switchable VU meters. **Price:** $3,170.
Contact: Prism Media at 973-983-9577; or circle **Reader Service 143.**

Universal Audio 1176LN Limiting Amplifier
Features: Single-channel; input, output controls; attack, release controls; switchable ratios; switchable VU meter; reproduction of original Universal Audio 1176LN.

Price: $2,295.
Contact: Universal Audio at 831-454-0630; or circle **Reader Service 144.**

Crane Song Spider
Features: Preamp/mixer/A/D converter; eight-channel; 48V phantom power; phase switch; pan control; 15 - 24-bit dither; 44.1, 48, 88.2, 96 kHz sample rates; tape emulator; word clock; bypass; LED meters. **Price:** $5,000 - $6,000.

Contact: Crane Song at 715-398-3627; or circle **Reader Service 145.**

Empirical Labs Model EL8 Distressor Compressor
Features: Two-channel; input, output, attack, release controls; switchable compressor ratio; distortion switches; side chain; bypass. **Price:** $2,799.

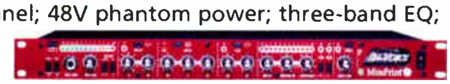

Contact: Empirical Labs/Wave Distribution at 973-728-2425; or circle **Reader Service 146.**

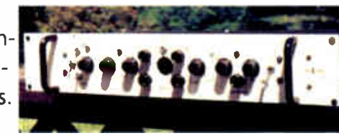
Purple Audio MC76 Limiting Amplifier
Features: Single-channel; Class A; attack, release, input, output controls; selectable 4:1, 8:1, 12:1, 20:1 compression ratios; switchable VU meter. **Price:** $1,995.

Contact: Purple Audio/Wave Distribution at 973-728-2425; or circle **Reader Service 147.**

Amek System 9098 Dual Compressor/Limiter
Features: Two-channel; threshold, attack, release, ratio compressor controls; level, release controls for limiter; output gain controls per channel; hard knee switch; fast attack mode; stereo link; side chain; ambience switch; bypass; Rupert Neve design. **Price:** $2,946.

Contact: Amek at 818-973-1618; or circle **Reader Service 148.**

Mindprint En-Voice Preamp/Processor
Features: Single-channel; 48V phantom power; three-band EQ; frequency, cut/boost controls per band;

tube saturation; compressor; threshold, compression controls; master output level; LED meters. **Price:** $749.
Contact: Mindprint/Steinberg NA at 818-678-5100; or circle **Reader Service 149.**

DaviSound TB-3 Inner Tube Compressor Preamp
Features: Two-channel tube compressor/preamp, drive, compress, recover, release, mic controls; 12AX7 tube; activity LEDs. **Price:** $995.

Contact: DaviSound at 803-276-0639; or circle **Reader Service 150.**

DigiTech Vocalist VR Harmony Processor
Features: Harmony pitch control; reverb;

input control; chordal, scalic processing; chord matching control; MIDI compatible; bypass; LCD screen. **Price:** $400.
Contact: DigiTech at 800-777-3637; or **Reader Service 151.**

DOD Electronics Dimension 3 Digital Audio Processor
Features: Two-channel; input level, mix, output

level, dual parameter controls; reverb, delay, chorus, flange, phase shift, rotary speaker, pitch/detune, tremolo/pan. **Price:** $160.
Contact: DOD Electronics at 801-566-8800; or circle **Reader Service 152.**

Stage Accompany SA 2310 Graphic Equalizer
Features: Two-channel; 31-band; switchable ±6/12 dB; constant Q filter; high-pass filter; level controls; ARMoR chassis; LED meters. **Price:** $2,397.

Contact: Stage Accompany at 800-955-7474; or circle **Reader Service 153.**

Gentner AP800 Conference Processor
Features: Eight inputs; 24V phantom power; Echo Cancellation; three-band EQ; high-pass filter; gate controls; automatic gain control; mic priority controls; decay rate controls; RS232 port; remote control PC-based software available.

Contact: Gentner at 800-945-7730; or circle **Reader Service 154.**

FMR Audio Really Nice Compressor (RNC)
Features: Single-channel; threshold, ratio, attack, release, gain controls; bypass; side chain. **Price:** $200.

Contact: FMR Audio at 800-343-9976; or circle **Reader Service 155.**

On The Bench

Panasonic AD96/AD96M Analog-to-Digital Converters Bench Measurement

continued from page 90

Input Sensitivity
Average of all eight channels level trims set for -6/+6 dB FS,
Reference level set for:

-20 dB FS	24.8 V, 30.1 dBu/5.7 V, 17.4 dBu
-18 dB FS	20.0 V, 28.2 dBu/4.6 V, 15.5 dBu
-16 dB FS	15.7 V, 16.1 dBu/3.6 V, 13.3 dBu
-14 dB FS	12.5 V, 24.2 dBu/2.9 V, 11.5 dBu

Input Impedance
All conditions of input level trim and reference level — 20.6 kilohm

Input Overload
Unit reaches digital full scale before input overload occurs at — N/A

Output Polarity
Digital output audio signal polarity relative to audio signal input — noninverting

Frequency Response

44.1 kHz f_s	+0.0, -0.20 dB 20 Hz - 20.6 kHz -3.0 dB @ 21.5 kHz
48.0 kHz f_s	+0.0, -0.20 dB 20 Hz - 22.3 kHz -3.0 dB @ 23.4 kHz
96.0 kHz f_s	+0.0, -0.60 dB 20 Hz - 44.7 kHz -3.0 dB @ 46.6 kHz

Total Harmonic Distortion

At -0.1 dB FS, measurement bandwidth = $F_s/2$

44.1 kHz f_s	< 0.0017% 20 Hz - 20.7 kHz
48.0 kHz f_s	< 0.0016% 20 Hz - 22.5 kHz
96.0 kHz f_s	< 0.0020% 20 Hz - 45.0 kHz

At -10 dB FS, measurement bandwidth = $f_s/2$

44.1 kHz f_s	< 0.00025% 20 Hz - 20.7 kHz
48.0 kHz f_s	< 0.00025% 20 Hz - 22.5 kHz
96.0 kHz f_s	< 0.001% 20 Hz - 45.5 kHz

Linearity Error
44.1/48.0/96.0 kHz f_s — < +/- 1.0 dB 0 to -130 dB FS; +6 dB @ -140 dB FS

Signal to Noise Ratio
Input termination 600 ohm, noise relative to full modulation,44.1/48.0/96.0 kHz F_s

Wideband ($f_s/2$)	113.5/113.5/100.1 dB FS
A-weighted	116.2/116.3/113.7 dB FS

Dynamic Range
44.1/48.0/96.0 kHz f_s
A-weighted, <10 Hz - Fs/2 BW — 116.5/116.9/116.4 dB

Quantization Noise
44.1/48.0/96.0 kHz f_s
20 Hz @ -1.0 DB FS, THD+N in 400 Hz - $F_s/2$ BW
-114.1/-114.5/-113.9 dB FS

Channel Separation
44.1/48.0/96.0 kHz f_s
Ch1 > Ch2 & Ch2 > Ch1 — > 120 dB 20 Hz - 2 kHz; > 100 dB @ 20 kHz

Note: Unless otherwise noted, all measurements are for the Channels 1 and 2, all I/O in balanced mode, word length 24 bits, reference level set to -20 dB FS, high impedance instrument load (200K II 270 pf)

PORTADISC

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Introducing the new HHB PORTADISC™, a professional portable MiniDisc recorder that's always ready to take the hard knocks of audio acquisition on the move.

The PORTADISC uses a tough, precision MD drive for maximum durability - not a flimsy, consumer mechanism. And with a frequency response of 10Hz - 20kHz and a dynamic range greater than 96dB, you'll be making recordings of exceptional quality.

Despite its surprisingly affordable price, the PORTADISC is packed with the features you need like ultra low noise balanced mic inputs, coaxial and optical digital I/Os, flexible powering options and a comprehensive range of accessories supplied as standard.

And the PORTADISC is the only MD portable to feature a USB interface so you can stream audio to and from your workstation.

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PRO AUDIO REVIEW PAR excellence 2000

HHB FIRST WE LISTEN

READER SERVICE NUMBER 197

NAMM Wrapup continued from page 70

NAMMers their first look at the MKH 800 multipattern studio microphone.

Yorkville's Apex line showed a new full-featured large diaphragm studio condenser mic for the upper mid-level market, the Apex450.

Drum mics were the ticket for Electro-Voice with the introduction of the N/DYM Drum Mic Kit. The kit is a collection of two- N/D468s, one N/D478 and an N/D868 mic in an aluminum flight case.

Audio-Technica did something a little different — it unpackaged a pair of drum microphones from the KP-Drums KitPak for individual sale.

Soundelux brought to NAMM one of the highlights of the AES show, the ELUX 251. The 251 uses a capsule based on the Telefunken ELAM 251 microphone. ADK continued to expand its line of mics — now offering everything from traditional condensers to hefty tube mics such as the Area 51.

Marshall Electronics MXL 603 microphone

One of the few new preamps shown at the show was the P2 digital from True Systems (distributed by Neumann).

The P2 is a two-channel mic pre with M/S decoding and 24/96 A/D converters. ART showed the two-channel DI/O Preamp System which sports a 12AX7A tube in the preamp section and digital outs. ART also showed a version without the digital outs.

In the ether of wireless, several companies, such as Sony and Audio-Technica, announced that its systems were now compatible with new areas of the UHF wireless spectrum.

Azden is continuing to mine the VHF spectrum. The 32BT is a body-pack transmitter and the 200R is receiver aimed at the budget market.

Samson had on hand several teeny-tiny UHF wireless transmitters, including one that was not much more than a 1/4 inch plug for guitar use. Another one for lavalier use is small enough to be unobtrusively clipped to a shirt or a wind instrument.

The big news from MediaFORM is the general upgrade of its shipping duplication drives from 12X up to 16X. All retain MediaFORM's SmartDrive technology.

Microboards's newest products are the Orbit line of inexpensive semiautomated dupers (acquired in the Champion acquisition). Microboards personnel hinted strongly that they would be jumping into DVD mastering and duping with both feet very soon with its upcoming MASS 5.1 system and the DVR-400 DVD-R tower duplicator. Not to be left behind, Logic brought its U Master duplication tower to its first NAMM.

Azden's 32BT VHF Wirelsss transmitter

And finally, Yamaha and Roland announced a future MIDI standard allowing communication between the various current flavors, GM2, GS and XG.

On The Bench

Sabine Graphi-Q 3102 Digital EQ Bench Measurement

continued from page 20

Gain to Balanced Out:		0.0 dB with Gain at 0.0
		+12.2 dB with Gain at Max (+12)
Output Impedance:	Balanced Output	52 ohms
Maximum Output:	Balanced Output	+30.2 dBu (25.0 V)
	Unbalanced Output	+28.0 dBV (25.0 V)
Input Impedance		25k-ohms
Input Overload:		+29.2 dBu (22.3 V)
Frequency Response:		+0.00 dB, -0.27 dB, 11 Hz - 20 kHz
		3 dB at < 10 Hz & 24.05 kHz
THD + N:	at +24 dBu output	< 0.00371%, 20 Hz - 20 kHz
Noise:	A-weighted	-108.5 dBu
	CCIR-weighted	-99.7 dBu

Dynamic Range:	Unweighted	106.4 dB
	CCIR-weighted	99.7 dB
	A-weighted	108.9 dB
Quantization Noise:		-91.6 dB FS
THD + N vs Level at 1 kHz:		< -90.7 dB FS, 0 dB FS to -90 dB FS
		< -103.1 dB FS, -50 dB FS to -90 dB FS
Linearity Error:		< 0.13 dB, 0 dB FS to -90 dB FS
Channel Separation:		> 96.3 dB, 20 Hz - 20 kHz

Notes:
Unless otherwise noted or implied, all measurements were made from the balanced input to the balanced output with the Gain at marked 0 and with the Delay and FBX defeated. Measurements made on one channel are made on Channel A. Unless otherwise noted or implied, THD+N measurements were made using a 22 kHz analysis filter.

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PRO CD RECORDERS

ALESIS Masterlink ML-9600
High-Resolution Master Disk Recorder

The MasterLink ML-9600 combines stereo hard disk recording, CD burning, DSP, and mastering functions to deliver compact discs in the standard 'Red Book' 16-bit/44.1kHz format, or high resolution 24-bit/96kHz CDs that utilize Alesis' revolutionary CD24 technology. The ML-9600's amazing sonic quality and powerful built-in tools offers a uniquely versatile and affordable solution for everyone from large commercial facilities to project studios and recording musicians.

FEATURES–
- Reads/writes 16-bit 44.1kHz Red Book Audio CDs as well as files in Alesis' CD24 24-bit/96kHz high-resolution mastering— an AIFF compatible file format that can be read by MacOS, Windows and Unix computer platforms
- 24-bit 128x oversampling A/D/A converters
- Built-in 3.2GB IDE hard drive
- 4x CD burning using standard CD-Rs
- Up to 20-40k Hz frequency response
- 113dB S/N ratio (A-weighted)
- Supports 16-, 20- and 24-bit wordlengths and 44.1, 48, 88.2, 96 kHz sample rates
- Built-in sample rate conversion & noise shaping
- Create and store up to 16 playlists containing as many as 99 tracks

Inputs and Outputs
- **Analog**- XLR-balanced and unbalanced RCA connectors
- **Digital**- AES/EBU (XLR) and coaxial S/PDIF (RCA) I/O
- 1/4" headphone out w/ level control

Editing
- Gain control
- Cropping allows adjusting start and end points
- Join and Split for combining and separating song sections.

DSP Finishing Tools
- Equalization, Compression Normalizing and Peak Limiting

Includes
- Infra red remote control and rackmount brackets

HHB CDR850 Professional CD Recorder

The new HHB CDR850 is one of the most comprehensive CD-R, CD-RW recorders available today. It delivers the outstanding sound quality that HHB is known at a lower price than previous models. Equipped with a complete range of analog and digital I/O and easy to use one touch recording modes make the CDR850 suitable for any audio environment no matter how sophisticated or demanding.

FEATURES–
- CD-R, CD-RW compatible
- Balanced XLR and unbalanced RCA analog I/O, AES/EBU digital in, coaxial & optical S/PDIF digital I/O
- Sample rate converter accepts any digital signal from 32 to 48 kHz including varispeed (pitched) signals
- All functions are front panel accessible
- Automatic digital source synchro recording- all or 1 track. Ideal for compilation recording
- Manual digital source recording recording mode is used for recording from digital sources when manual control of the recording is required. Recording is started and stopped manually.
- Copies all CD, DAT, MD, DVD and DCC track starts
- SCMS management can create discs that cannot be copied, copied once or are not copy prohibited.
- Includes full function IR remote control
- ***CDR850 PLUS*** adds word clock I/O (BNC)

TASCAM® CD-RW2000
CD Recorder

The CD-RW2000 is the Tascam's most advanced stand-alone CD Recorder. Building on the success of the Tascam CD-RW5000, the CD-RW2000 introduces a host of features demanded by professional users including comprehensive analog and digital I/O, word sync input, input monitoring and digital gain adjustment which allows level adjustments without going back to the analog source. A RAM buffer is included to ensure ID markers won't clip the start of a track and the adjustable auto cue function lets you locate right to the start of actual audio, rather than the track ID.

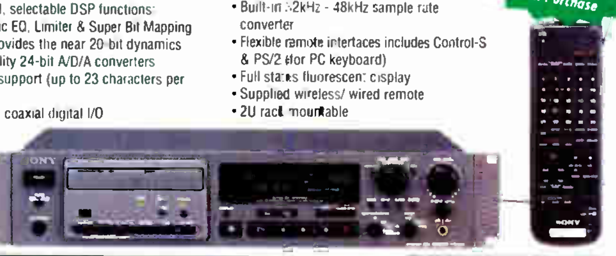
FEATURES–
- CD-R, CD-RW, CD-R-DA & CD-RW-DA compatible
- XLR balanced and RCA unbalanced analog I/O
- AES/EBU, coaxial and optical S/POIF digital I/O
- Records and plays 74min and 80min CDs
- 24-bit A/D/A converters
- 32 - 48 kHz sample rate conversion to 44.1kHz
- Word Sync input and 15-pin parallel port
- Automatic and manual track numbering
- Selectable SCMS copy protection
- Adjustable gain on digital I/O
- One touch digital fade in/out w/ user specified fade time
- Built-in RAM buffer for precise ID markers
- Wired remote included

SONY CDR-W33 CD Recorder/Player

The Sony CDR-W33 is a rack mountable CD recorder that supports audio format CD R and CD-RW media. Features include built-in DSP functions, unbalanced analog inputs and outputs with 24-bit converters, digital I/O (coaxial and optical) and CD -Text support.

FEATURES–
- Supports CD-R-DA, CD-RW-DA discs
- On-board, selectable DSP functions: Parametric EQ, Limiter & Super Bit Mapping which provides the near 20-bit dynamics
- High quality 24-bit A/D/A converters
- CD-Text support (up to 23 characters per name)
- Optical & coaxial digital I/O
- Unbalanced analog RCA jacks
- 2x finalize
- Built-in 32kHz - 48kHz sample rate converter
- Flexible remote interfaces includes Control-S & PS/2 (for PC keyboard)
- Full status fluorescent display
- Supplied wireless/ wired remote
- 2U rack mountable

Free CD Walkman With Purchase



PORTABLE DAT-RECORDERS

TASCAM DA-P1 Portable DAT

Built for the engineer on the move, the DAP-1 is Tascam's professional, portable DAT that sounds as good as its studio-cuffed big brothers.

Features–
- Rotary 2 head design, 2 direct drive motors
- XLR mic/ line inputs (w/phantom power)
- Analog and S/PDIF (RCA) digital I/O
- 32/44.1/48kHz sample rates and SCMS-free recording
- Built-in MIC limiter and 20dB pad
- TRS jack w/ level control for monitoring
- Includes shoulder belt, AC adapter, and battery
- **BP-D1** rechargeable battery pack
- **CS-D1** carrying case
- **Porta Brace ARDAP1** field case

HHB PORTADAT PDR-1000TC+

Designed with the help of leading location sound recordists, the PORTADAT series redefines the standard for sonic quality, performance and reliability in a portable field recorder. The PDR-1000TC Plus with master sync timecode module and headphone matrix is guaranteed to provide world-class performance. Features include 4-head, 4-motor direct drive transport, XLR mic/line inputs, S/PDIF and AES/EBU digital I/O, +48v phantom power, and are supported by a wide range of optional accessories.

Buy a PDR-1000TC Plus For Our Special Low Price and Get (100) 125 Minutes DAT Tapes, 4-Bay Fast Charger and Aluminum Carrying Case Free!

FEATURES–
- The most accurate timecode DAT recorder available, it includes the MS-1000 Master Sync module to ensure that drift doesn't exceed 1 frame every 10 hours
- Supports 16-bit quantization at 44.1 and 48 kHz and 12-bit 32khz (Long Play) recording
- 4-head design allows off tape confidence monitoring while recording.
- Highly reliable transport uses 4 direct-drive motors
- 2-XLR balanced switchable mic/line inputs
- XLR-balanced AES/EBU and coaxial (RCA) S/PDIF I/O
- Switchable 6dB/octave High Pass Filter at 100Hz, 30dB pad, peak limiter and 48v phantom power
- Back-lit LCD display shows clock and counter, battery status, peak level metering, margin indicator, source/tape monitor, transport status, and selected sample rate.
- Built-in monitor speaker and 1/4" stereo headphone output with monitor level control.
- Lockable record level feature
- Supplied NiMH battery, with no memory effect, lasts for 2 hrs and 1.5 hours with timecode operation.
- Built-in clock - time and date stamps every recording.
- Record, generate and reference to SMPTE/EBU timecode in all existing standards, jam sync and conversion of absolute time into linear timecode.
- Selectable frame rates- 24, 25 (PAL), 29.97 DF/NDF and 30 fps.
- Timecode I/O via XLR connectors. Word sync out, video/word sync input and thru, via BNC connectors.
- Timecode can be recorded as time of day, preset for free /record run, and can be set to internal or external timecode generation. Built-in chase synchronizer allows it to slave to an incoming timecode source.
- Lemo socket for compatibility with Aaton film cameras.
- Facilitates pull up from 29.97DF fr/sec. to 30 fps drop
- HM-1000 headphone matrix offers 5 selectable monitoring configurations: stereo, mono left, mono right, mono sum and mid-side stereo.

Standard Included Accessories-
- RB-110 AC adapter/ 2-bay charger
- MHB220 NiMH battery • HHB-15 DAT tape
- Carrying case

PROFESSIONAL CD PLAYER & MINIDISC

SONY MDS-E11 MiniDisc Recorder

A professional space saving MiniDisc player/recorder with XLR balanced analog I/O and comprehensive remote control capabilities. Ideal for live theatre, video production, installation, broadcast and studio.

FEATURES–
- Compact single rack space design
- Balnced XLR line inputs and outputs swithchable from +4/-10 dB, unbalanced RCA (phono) I/O
- Coaxial S/PDIF digital I/O
- Automatic sample rate converter to 44.1kHz with 18-bit processing offers low jitter and superior signal to noise
- Automatic pause feature at the end of each track ideal for live sound effects or narrations
- RS232, Parallel fader start, control-S and relay record/play remote capabilities
- Relay remote function allows you to cascade several MDS-E11's together for uninterrupted sequential play and record
- Comprehensive LCD display
- Headphone out with level control
- **INCLUDES:** Infra red remote control

CDP-D11 1RU Professional CD Player

The CDP-D11 is a single rack space CD player, with a durable chassis/transport ideal for the road and installation use. The elegant 1U rackmount design provides economy of space and easy transport while the heavy-duty transport is designed to withstand the rigors of production and mobile use. Features include Balanced and unbalanced as well as digital outputs and a 4MB RAM buffer.

FEATURES –
- Compact 1U rack height
- RCA unbalanced and XLR balanced outputs
- Optical and coaxial digital outputs (S/PDIF)
- ±12.5% Vari-speed
- Instant start
- Flexible control interface (parallel & RS-232)
- Auto Cue and Auto Pause
- Advanced shock protection w 4Mb D-Ram memory
- Relay play control bus
- Dual mode remote.
- Mark locate feature
- Dual mode remote functions as wired or wireless IR remote and includes direct numeric entry of track numbers

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FEATURES-
- 24 input channels, including 16 built-in analog inputs (w/+48V phantom powering on 12 channels), *plus* 8 optional digital inputs (ADAT, Tascam, AES/EBU formats available)
- 4 freely assignable analog outputs, *plus* expansion options for 4 additional analog outputs, or 8 assignable digital outputs
- Digital I/O expansion slot for optional multi-track recorder interface card (ADAT, Tascam, AES/EBU format options) *plus* 8 channels analog input card options
- Coaxial digital stereo I/O and balanced XLR stereo outputs
- Total 120 bands of 44-bit EQ including 4-band parametric EQ on all main input & output mixing channels & 2-band EQ on sec. input channels *plus* a library with 40 preset & 40 user programs
- Dedicated central panel controls for the 4-bands of EQ - like the 02R - & dedicated pan control
- 2 internal stereo multi-effects processors with the same DSP as the Yamaha ProR3 and REV500, and an effects library with 42 preset programs and 57 user programs
- Input delays of up to 250ms and output delays of up to 300ms
- 22 dynamics processors for individual processing, and a dynamics library with 40 preset programs and 40 user programs

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MX-602A & MX-802A
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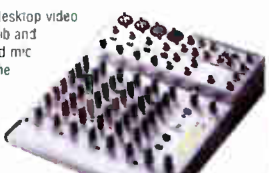
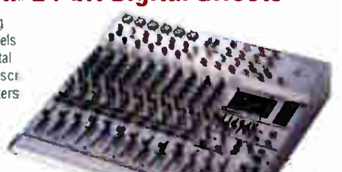


The professional MX-602A and MX-802A are ideal for project studios, desktop video production, small club venues and much more. They for MIDI suites, sub and monitor mixer and as handy tools for level and impedance matching and mic preamp. Any broadcast studio or AV/film application can benefit from the transparent "big console" audio performance and features.

FEATURES-
- Total of 6 inputs- 2 mic/line and 2 stereo (**MX-602A**)
- Total of 8 inputs- 4 mic/line and 2 stereo (**MX-802A**)
- 2 aux sends and 3-band EQ per channel
- 1/4" balanced Master and Monitor outputs
- 1/4" balanced Mute 3/4 Bus Outs
- Stereo RCA tape inputs and outputs
- 1/4" Headphone out with level control
- 2x stereo aux returns

MX-1804X
14 x 2 x 2 Rack Mixer With 24-bit Digital Effects



Part of a new generation of high-performance compact analog mixers, the MX-1804X offers six mono and four stereo channels and provides functions that are quite commonly found in digital consoles- namely a built-in 24-bit stereo multi-effects processor. The multi-effects processor offers 20-bit A/D and D/A converters and 32 presets of reverb, delay and modulation effects algorithms based on Behringer's Virtualizer rackmount effects unit. Whether for stage or studio, the 7-band graphic EQ assigned to the main outputs of the MX1804X allows you to adapt the mixer to various room acoustics and during mixdown gives your overall sound the finishing touch and sparkle it deserves.

RANE
SM82 Stereo Rackmount Mixer

The Rane SM 82 is a sixteen channel line mixer divided into 8 stereo channels and housed in a single rack space. It is ideally suited for mixing keyboards, synthesizer expander modules and drum machines, as well as effects mixing in guitar racks. It also functions as an excellent submixer for stereo line inputs and effects to a larger console.

FEATURES-
- 8 stereo (two channel) 1/4" input channels with rotary input level control, a stereo aux send level control slider and a left to right balance control
- Will accept mono inputs, converting the balance control to a pan pot
- Aux return with a rotary return level control and a slide control for left to right balance
- 1/4" T.R.S. output section with a rotary output level control and a slide control for left to right balance
- Fully expandable using 1/4" TRS main expand In and Out jacks
- Short internal signal paths for clean sound
- Only 5.25 inches deep

SIGNAL PROCESSING

WAVES
L2 ULTRAMAXIMIZER
Brick Wall Peak Limiter

The L2 is a proprietary brick wall look-ahead peak limiter with IDR (Increased Digital Resolution) dithering technology based on the award-winning L1 software. Featuring 48-bit internal processing and support for 96kHz sampling rates as well as digital and analog I/O with 24-bit A/D and D/A converters means the L2 is ideal for the maximum number of audio applications, from mixing to mastering to concert sound. The L2 Ultramaximizer performs high quality re-quantization to 24, 22, 20, 18, and 16-bit outputs, plus the Waves ARC (Auto Release Control) technology continuously controls the optimal release time for maximizing levels and minimizing audible distortions.

FEATURES-
- 2U rackmount limiter with 48-bit processing significantly increases the average signal level of typical audio signals without introducing audible side effects
- 44.1, 48 kHz, and x2 88.2 and 96 kHz sample rates
- Linked stereo and dual mono operation
- Dedicated rotary controls and numeric displays for threshold (-30.0 to 0.0dB in 0.1dB steps), output ceiling (-30.0 to 0.0dB in 0.1dB steps.) and release (0.01 to 1000ms)
- The left and right rotary threshold controls transparently reduce transient peaks by several dB allowing the overall signal level to be raised several dB
- The use of look ahead technology allows the L2 to anticipate peaks before they happen, thereby minimizing the possibility of artifacts.
- **ARC (Auto Release Control)** technology bypasses the variable release control to dynamically control release times to fit the human ear's expectations. This allows a greater amount of limiting and therefore maximum level without distracting artifacts
- **IDR (Increased Digital Resolution)** is Waves' proprietary wordlength-reduction (quantization), dither and noise shaping technology which preserves and even increases the resolution of digital signals.

Three push button controls allow simple control over IDR parameters-
- Quantization- The wordlength of digital signals can be quantized and output to 24, 22, 20, 18 or 16 bit resolution. For example power wordlength reduction can be used to re-quantize 24 bit input signals to 16 bit— ideal for DAT and CDR applications
- Dither- There are two types of dither. Type 1 provides no nonlinear distortion. Type 2 exhibits lower dither (hiss) level. L2's Dither function can also be completely switched out (off).
- Noise shaping- choices include Moderate, Normal, Ultra, or non (off).
- Dedicated bargraph meters for input, output and attenuation with infinite peak hold and peak meter reset
- 24-bit Balanced XLR, and unbalanced RCA analog I/O
- AES/EBU (XLR) and S/PDIF (coaxial) digital I/O
- Internal, S/PDIF (RCA), AES (XLR), external word sync (BNC) sync capabilities

Applications include Analog to Digital and Digital to Digital mastering, Digital recording 16 bit dithered/24bit, Analog limiter insert, Digital limiter insert, mixdown through the L2 to DAT and A/D/A conversion

lexicon
MPX-500 24-Bit Dual Channel Effects Processor

The MPX 500 is a true stereo 24 bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display

FEATURES-
- 240 presets with classic, true stereo reverb programs as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo
- Balanced analog and S/PDIF digital I/O
- 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls.
- Tempo-controlled delays lock to Tap or MIDI clock

SIGNAL ROUTER

FURMAN
SRM-80 Signal Router

The new SRM-80 Signal Router/Monitor makes mixdown and dubbing simple, professional and affordable. It facilitates mixdown and dubbing between different media types without tying up additional console channels. For added convenience, a speaker selector switch that accommodates up to three sets of speakers and a high power headphone amp lets you monitor any selected source.

FEATURES-
- Signal routing and monitoring for four stereo devices such as DAT machines, cassette decks, CD in addition to main mix from board
- Balanced TRS input and output jacks for Source "A" input and amplifier sends. RCA jacks for "B", "C", and "D" inputs and outputs
- Speaker select function switches between two high powered pairs of speakers, and one self-powered pair
- Gain adjust for each speaker pair allows for equal loudness when switching between them
- High power headphone amp
- 40-segment high resolution metering for L/R monitor outputs with peak or average response switch
- DIM button reduces level by -15dB during interruptions
- MONO sum button to speakers and headphones only allows easy identification of mix problems
- Line Level may be set to -10 or +4 dBu sensitivity
- Ground lift switch
- Optional SRM-RU Remote Control Unit: Duplicates the MONO, DIM, and SPEAKER SELECT push button controls of the SRM-80. Uses a standard MIDI cable, and is supplied with 10' length cable.

MICROPHONES

AKG C2000B Condenser Mic

For A Limited Time Only Get A Free H-100 Shockmount

The C 2000 B is an all-purpose cardioid condenser microphone perfectly suited for both recording and live sound situations. The newly developed small-diaphragm transducer capsule is made using a unique manufacturing process that ensures high sensitivity, lowself noise, and excellent bass response.

FEATURES-
- Cardioid polar pattern
- Switchable bass rolloff filter (6 dB/octave @ 500 Hz) and -10dB pad
- Built-in pop screen reduces unwanted noise
- Rugged construction, elegantly styled die-cast metal housing, and silver-gray finish
- **30 Hz to 20 kHz frequency response**

SHURE KSM44/SL Multipattern Condenser Mic

The KSM44/SL is a multiple pattern dual large diaphragm condenser microphone built without compromise using premium electronic components and gold-plated internal and external connectors. The KSM44/SL is a premium vocal mic and is equally adept for close miking a wide range of acoustic instruments, amplifiers and for ambient room miking.

FEATURES-
- Dual 1-inch,gold-layered Mylar diaphragms
- Class A, discrete, transformerless preamplifier
- Cardioid, omni and bidirectional polar patterns
- Subsonic filter eliminates rumble from mechanical vibration below 17 Hz
- Integrated 3-stage pop grille and shock mount
- 15 dB pad and 3-position switchable low-frequency filter virtually eliminates unwanted background noise and controls proximity effect
- Includes ShureLock elastic-suspension shock mount and swivel mount, protective pouch and locking aluminum carrying case
- **20 Hz - 20 kHz frequency response**

VP88 MS Stereo Mic

A single point mid-side stereo microphone made to withstand rigorous production environments. Phase accurate and natural sounding for any stereo miking situation such as drum overheads, sound effects gathering and live music recording.

FEATURES-
- Time-coherent Mid-side stereo condenser
- Adjustable stereo spread controls the amount of stereo effect
- Mono compatible ideal for broadcast applications
- 105 dB dynamic range
- 129 dB Max. SPL
- Switch selectable phantom power or 6 V battery operation
- 80Hz Low-cut filter (12dB/oct)
- LED power indicator
- Internal shock mount for reduced stand vibration and handling noise
- Built-in "pop" screen
- Brass and nickel-plated aluminum construction with stainless steel grille. Finished in satin black vinyl enamel
- Includes Windscreen, swivel adapter and 30 inch Y cable
- **Frequency Response 40 to 20,000 Hz**

OPTIONAL ACCESSORIES-
- **A88SM** isolation mount with 6 mounting options including camera shoe 152.50
- **C110** 25 ft. 5 conductor, long life extension cable 64.50

SENNHEISER MKH-416 P48U3

A high-end shotgun microphone offering highly accurate sound reproduction and a rugged weather-proof design. The pressure gradient and short interference tube principals allows a wide frequency response while retaining a hypercardioid polar pattern in the low and mid frequencies while an even tighter lobar pattern is achieved above 2kHz. Ideal for long distance miking for film and video or as an ENG or lecture microphone.

FEATURES-
- Pressure gradient/ short tube interference shotgun mic
- Super-cardioid/Lobar polar pattern
- High sensitivity, Low self-noise
- 128dB @ 1kHz maximum SPL
- 48 V phantom power required
- **Frequency Response 40Hz - 20kHZ**

AUDIX OM-2 Dynamic Handheld Mic

The OM2 is a dynamic handeheld microphone designed to provide high quality sound for a wide variety of vocal styles. The exceptional feedback rejection of the OM-2 is considered a benchmark by which other handheld mics are judged.

FEATURES-
- Hypercardioid dynamic
- Proprietary VLM (very low mass) capsule technology
- 50Hz-16kHz frequency response
- 140 db SPL handling
- Durable zinc alloy construction
- Electronically cut blast filter with black E-coat finish
- Includes cordura carrying case and nylon mic clip

audio-technica AT-841a Boundary Mic

This rugged omnidirectional boundary mic when placed on an unobstructed flat surface such as a conference room table or floor offers increased sensitivity and signal to noise characteristics over standard microphones. This is because the microphones half space design rejects out of phase reflections coming from the flat surface. A perfect solution for theatrical stage, television production and conference recording.

Features:
- Micro omnidirectional condenser boundary microphone
- 30Hz - 20kHz frequency response
- Battery or phantom power (9-52v) operation
- 25' detachable cable terminates
- 80hz Roll-off switch (18dB Octave)
- Includes AT8531 power module, battery and protective soft pouch
- Also available as **AT841Wa** white version of above

WIRELESS MICROPHONES

SONY UHF Wireless Microphone system

Consisting of multiple handheld and bodypack transmitters and receivers, Sony's UHF Synthesized Wireless Microphone system is widely recognized as the outstanding wireless mic system for professional applications. Operating in the 800 MHz UHF band range and equipped with a PLL (Phase Locked Loop), they provide up to 94 channels of interference-free operation. Additional features, like space diversity reception, LCD indicators, reliable and sophisticated circuit technology assure low noise, wide dynamic range, and extremely stable signal transmission and reception. Ideal for broadcasting stations, film production facilities, and ENG/EFP work.

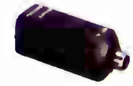
MB-806A Multi-Channel UHF Synthesized Diversity Receiver

- 19-inch rack mount tuner frame accommodates up to 6 WRU-806A tuner modules (sold seperately)
- When used with a WD-820A Antenna Divider up to 3 systems providing 18 mics can operate simultaneously
- Six XLR-balanced output connectors plus an XLR balanced mix output
- Auto channel assignment for extra tuner modules, detects and skips unusable channels
- Built-in antenna divider with 9-volt DC power
- Mic/line output level switch
- Rear mounted antennas

WRT-807A

- High sound quality for vocals - powerful, crisp and clean sound as well as presence in the low end and mid frequency range
- Dynamic microphone capsule, same as used in the Sony F-780 professional vocal microphone
- LCD for display of channels, attenuator and accumulated hours
- Up to 5 hours of continuous use with a single AA battery
- Battery alarm transmitted to compatible receivers
- Newly-developed lockable power switch to prevent accidental operation

WRT-808A Plug-In Transmitter



- Plug-In transmitter works with any dynamic microphone with an XLR connector
- 94 channel frequency agile
- Runs on two AA batteries
- Battery level indicator
- Level control
- AF/Peak indicator shows condition of audio input level
- Low-Battery alarm works with compatible receivers

WRT-805A

- Smoothly tapered body with compact lightweight design
- 6 hours of continuous operation with a single AA battery
- LCD display of channel, attenuator and accumulated hours
- Accepts ECM-122BMP and ECM-44BMP and ECM-77BMP lav mics
- Low-Battery alarm works with compatible receivers

WRT-822A

- Accepts all Sony BC Series lavalier mics including the ECM-44SC/BC, 55 SC/BC, 66 SC/BC and 77 SC/BC. Use for wireless guitar applications, with optional K-1161 cable
- Backlit LCD information display
- 8 hrs. operation with 2 AA batt.

AZDEN 411DRH System

The 411DRH is a half rackmount, crystal controlled PLL synthesized UHF receiver with 53 user-selectable channels in the 794-806MHz band. Up to 9 systems may be used simultaneously. It has both 1/4-inch and XLR output jacks as well as volume adjustment.

41HT Handheld Mic Transmitter
- Newly-designed handheld with supercardioid uni-directional mic element or Audix OM3 head and 63 user selectable channels. Uses 2 AA alkaline batteries or Azden ni-cads with the AMC-2A Charging Station

41BT Bodypack Transmitter
- 63 user-selectable channels, input level control, standby switch, Hiroshi 4-pin connector and metal clip. Available w/ EX503H or Sony ECM44H lavalier microphones

41XT Plug-In Transmitter
- 63 user-selectable UHF channels Allows you to use your favorite dynamic microphone with an XLR output. Adjustable output volume control, power on/off and audio mute switches. LED AF Peak and Power indicators

SHURE UC Wireless Series

The Shure UC Wireless Series is a new, easy-to-use UHF wireless system featuring over 100 fully selectable frequencies for flexibility in a variety of applications. UC systems are available with hand-held, bodypack, lavalier, headset and cable transmitters and up to 16 can be used simultaneously.

UC4 Receiver
- Compact, 1/2 rack design and exclusive MARCAD technology
- Choice of over 100 user selectable frequencies
- Two 5 segment RF LED meters and 7 segment audio LED meter
- 2-band adjustable EQ
- Balanced XLR and unbalanced 1/4" outputs

UC2 Handheld Transmitters
- Available with SM58, Beta 58A, SM87 and Beta 87 capsules
- Combined power on,off and mute switch
- Adjustable audio gain control
- Internal antenna
- 8 hours battery life 9V battery (included)

Beta 87 Shown

UC1 Bodypack Transmitter
- Tini Q-G or optional Lemo connector
- Remote audio mute connector allows external audio and/or RF muting capability
- Two-position attenuator (0 and -20dB) to accommodate different input sources

On The Bench

Sabine Graphi-Q 3102 Digital EQ Bench Measurement

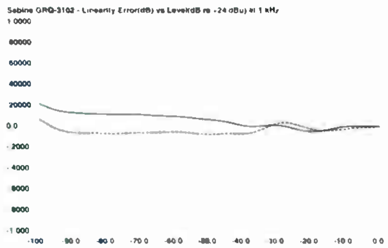


Figure 1: *Linearity error (dB) vs. level (dB re +24 dBu) at 1 kHz*

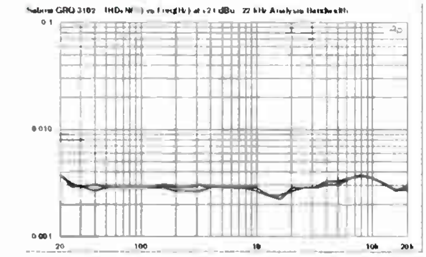


Figure 2: *THD+N (%) vs. Frequency (Hz) at +24 dBu — 22-kHz analysis bandwidth*

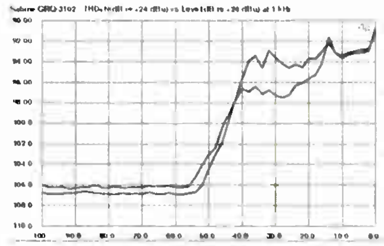


Figure 3: *THD+N (dB re +24 dBu) vs. Level (dB re +24 dBu) at 1 kHz*

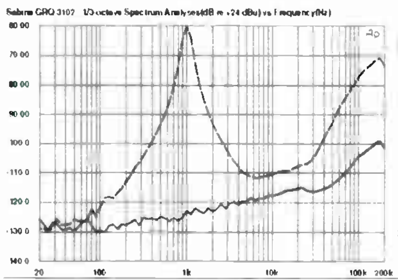


Figure 4: One-third-octave spectrum analyses (dB re +24 dBu) vs. Frequency (Hz)

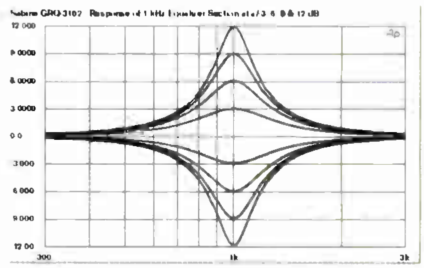


Figure 5: *Response of 1 kHz equalizer section at +/-3, 6, 9 and 12 dB*

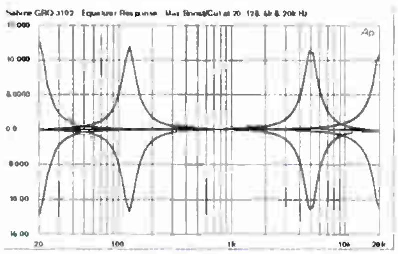


Figure 6: *Equalizer response — maximum boost/cut at 20 Hz, 125 Hz, 5 kHz and 20 kHz*

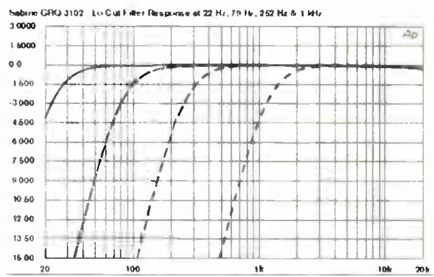


Figure 7: *Low-cut filter response at 22 Hz, 79 Hz, 252 Hz and 1 kHz*

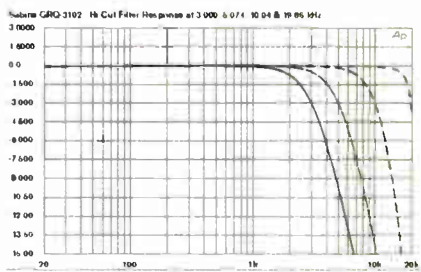


Figure 8: *High-cut filter response at 3.000, 5.074, 10.04 and 19.86 kHz*

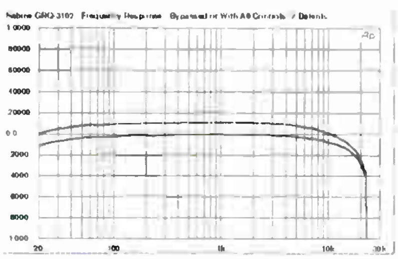


Figure 9: *Frequency response — bypassed with all controls at detents*

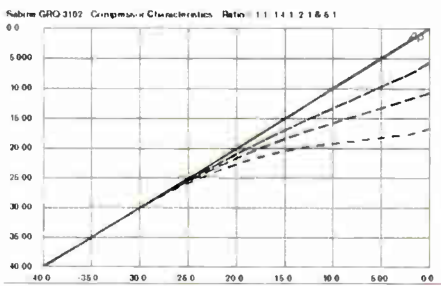


Figure 10: *Compressor characteristics — Ratio = 1:1, 4:1, 2:1 and 5:1*

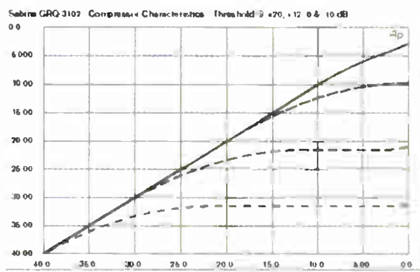


Figure 11: *Compressor characteristics — threshold at +20, +12, 0 and −10 dB*

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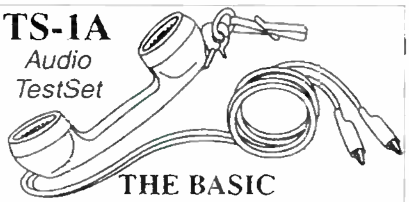

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B&H Pro Audio SuperStore Demo Room Features Cutting Edge Audio Gear

by Michele Kramer

B&H SupersStore in New York City now carries **Digidesign's** Pro Tools TDM editing system. The system is being showcased in the B&H's interactive pro audio demo suite. The suite features a studio monitor demo wall, more than 25 large diaphragm condenser studio microphones, mixers and MIDI equipment. Manufacturers represented in the room include **JBL** and **TASCAM**; the room is furnished with **Raxxess** studio furniture.

Al Spinelli, pro audio department manager, at B&H SuperStore's Pro Tools TDM demo station

Linda Ronstadt's Christmas 2000 album, *A Merry Little Christmas*, was recorded on **Alesis** M20 digital multitrack recorders. An M20 was used by Rich Tozzoli when he recorded the jazz album *Romero, Live at Trinity Church*. Tozzoli also used an M20 to record Joni Mitchell's Central Park tribute.

KRK Systems provided a 5.1 setup for a recent surround seminar that toured Texas and was produced by **Fits & Starts Productions**. The tour made stops in Austin, San Marcos, San Antonio, Houston, Arlington and Dallas over seven days. Other participants in the event included **Digital Theater Systems, Fairlight** and **Studio Technologies**. The seminar provided an in-depth look at 5.1 surround for music: how to do it and what tools you need.

The Mastering Lab recently closed the door on one era of recording technology to make room for the new. The independent disk-mastering house retired its cutting lathes in December 2000 to concentrate on digital audio technology. Mastering Labs' legendary albums included *Chicago*, *Hot August Night*, *Sticky Fingers*, *Frampton Comes Alive*, *The Wall* and *Who's Next*.

Sony Music Studios in Japan recently purchased eight additional **DK-Audio** MSD600M/SA master stereo displays. This brings its complement to 23.

National Events, a Newington, Va.-based special-events production company, used a **Soundcraft** Series FIVE console in tandem with a Soundcraft SM16 monitor desk for two Washington, D.C. events — the Million Family March and the Taste of DC — during October 2000. The Taste of DC wound through 11 blocks, from the White House to the Capitol, and the consoles were used at one of the event's larger stages near Pennsylvania Ave. and 12th Street to mix Joan Jett and the Blackhearts, Los Hombres Calientes and other national and local artists.

The Tsunami Beach Club in Louisville is a 20,000 square foot bar/nightclub with a **QSC**-powered sound system. Three PowerLight 6.0PFC amps provide power to 12 single 18" **Cerwin Vega** SL-36 subs, which are stacked around the perimeter of the dance floor. Two PowerLight 4.0s drive eight **JBL** mid-high cabinets above the dance floor.

Yahoo Broadcast (a division of Yahoo!) used **Neutrik** connectors when they built Yahoo Studios, a facility to incorporate video broadcasting into the Yahoo! Web site. Yahoo Studios uses Neutrik NC3MEZY-B and NC3FEZY-B Easycon connectors and NPPTB patchbays and patch cords. The deadline was tight for the studio build and the Neutrik connectors were reliable and easy to use.

Sam's Town, a Las Vegas casino and country music venue, is building a new 1,200-seat music performance and special events venue adjacent to its existing casino building. They will be using a **Crest** V-12 console for FOH. Also in Las Vegas, the Excalibur hotel is using a 52-input Crest X-VCA console, installed during its recent sound system upgrade. The X-VCA is used for the "Tournament of Kings" production, which runs seven nights a week, two shows each night.

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