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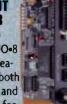
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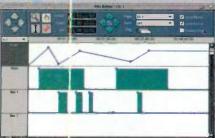
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FEATURES

44 LIFE IN THE SLOW LANE

Who couldn't use a little more time? Thanks to advances in time-compression technology, a new breed of hardware and software recorders makes transcribing and learning tunes easier than ever. By David Rubin

58 COVER STORY: EQUAL TIME

Make your mixes shine with these 14 EQ tips from the pros. We'll discuss the various types of equalizers; then we'll delve into drum applications, get a grip on guitar timbres, and add a fine veneer to vocals.

By Jeff Casey

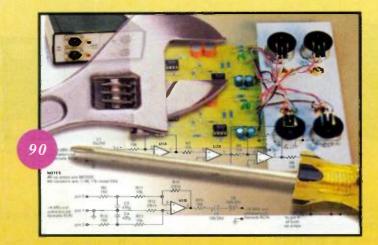
78 MASTERING CONTINUITY

The beauty of a compilation album lies in the array of artists contributing to the project. The headaches start when you need to create a seamless master from multiple sources. We show you how to give any master tape a consistent, unified sound. By Myles Boisen

90 DIY: BUILD THE EM -10/+4 LEVEL CONVERTER

Pro, semipro, and consumer audio gear can coexist happily with this handy, dandy, easy-to-build converter box. No more cheap adapters and signal loss due to mismatches! Plug +4 dBu "pro" gear into the converter's XLR connectors and -10 dBV "semipro/consumer" gear into its RCA jacks, and presto! Instant compatibility.

By Peter Mosher

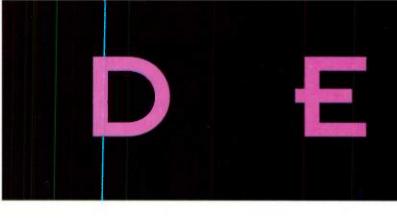


DEPARTMENTS

6	FRONT PAGE
12	LETTERS
20	WHAT'S NEW
194	AD INDEX/SALES INFO
195	CONTACT SHEET
208	CLASSIFIEDS

5

58

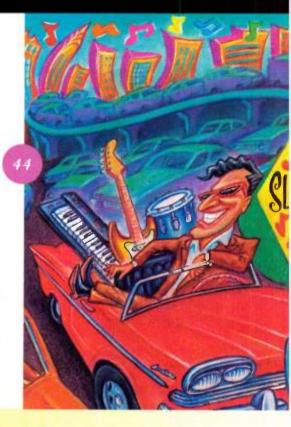


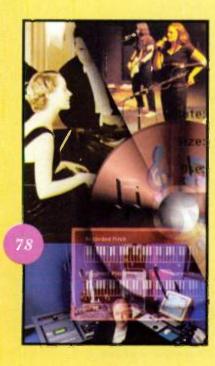
Electronic Musician

OCTOBER 1999 VOL. 15, NO. 10

COLUMNS

- **4-2 TECH PAGE:** Nonvolatile RAM Powerless RAM holds powerful possibilities.
- **100 DESKTOP MUSICIAN: Interactive Web Music** Liven up your site with music that follows your mouse.
- **108** SQUARE ONE: Fractals and Music Music and math make interesting bedfellows.
- **114 RECORDING MUSICIAN:** Recording Electric Guitar Guitar-recording secrets your mother never told you.
- **122 WORKING MUSICIAN: Ocean of Promotion** Music-promotion tips for those with big dreams and small budgets.
- **132 PERFORMING MUSICIAN: Music for Airports on Stage** Bang on a Can takes Brian Eno's ambient masterwork on the road.
- **226 FINAL MIX: Pass the Salt, Please** Separating hyperbole from truth is no mean feat these days.





REVIEWS

- 136 ALESIS QS8.1 keyboard synthesizer
- 146 TRACER DC-ART 32 3.06 (Win) audio restoration software
- 152 BLUE Blueberry large-diaphragm condenser microphone
- 158 CYCLING '74 Pluggo 1.04 (Mac) VST plug-ins/patch converter
- 166 GENERALMUSIC Equinox 61 synthesizer workstation
- 174 HHB Radius 20 and Radius 30 tube parametric EQ and tube compressor
- 182 AARDVARK Aark 20/20+ (Win) hard-disk recording system
- 188 ETEK NoteMix MA 400 portable powered mixer
- **198** QUICK PICKS: USB *Raricussions* (Mac/Win); PreSonus MP20 preamp; MicroBoards AudioWrite Pro CD-R (Mac/Win); Beatboy *Ramon Yslas Contemporary Percussion and Drums* (Mac/Win)

How Sweet It Is

On the eve of the final Audio Engineering Society convention of the 1990s, it's interesting to reflect on the state of the pro-audio industry at the time of the first AES convention of this decade. We're going to see a very different show this year, one that reflects a decade of progress.

At the 1990 AES convention in Los Angeles, the distinctions between the pro-audio and homerecording worlds were clear, though harbingers of change were starting to appear. The personal-



studio movement was still aborning, and it got little respect. MIDI was a dirty word in the pro community. The pre-ADAT Alesis displayed only semipro products, and Mackie Designs had not yet appeared at an AES show.

Some two dozen companies showed computer products at that 89th AES convention, notably Digidesign, Digital Audio Labs, Hybrid Arts, Mark of the Unicorn, Opcode, Passport, Spectral Synthesis, and Turtle Beach. You've never heard of most of the others. Even for the established leaders, acceptance in the pro community came slowly.

In 1990, quality condenser mics were prohibitively expensive by personalstudio standards, as were pro-quality powered monitors. Digital Audio Workstations were expensive, closed systems consisting of dedicated hardware and software. For the most part, pro-audio products were intended for commercial studios; few home recordists could afford them.

At this year's 107th AES convention in New York, relatively few manufacturers will pitch products exclusively to owners of commercial facilities. Today, we personal-studio owners have become an extremely significant market force, and the pro-audio manufacturers are well aware of it. Of course, some products will still be targeted at high-end commercial studios where corporate clients drop big bucks. But the owner of your local commercial studio will probably check out many of the same products that interest us. Products that have become personal-studio hallmarks, such as computer hardware and software and low-cost "traditional" audio hardware, will be taken for granted as mainstream professional tools.

This sea change has occurred because the personal studio and the professional studio have converged. A large and increasing amount of professional work is produced in personal facilities, supported by ever more efficient, affordable, and high-quality tools. Furthermore, the average income (and therefore, buying power) of semipro and nonprofessional studio owners has significantly increased. In short, whether you operate your studio as a business or not, you can afford tools that meet professional standards.

Ten years ago, we personal-studio owners mostly produced demos and other nonprofessional projects because the state of the technology and the industry defined the personal facility as a hobbyist's playground. A few creative souls made professional recordings at home, but they were exceptions. Not anymore. Today, we are more likely to be limited by our skills and knowledge than by the type of tools we can afford. And that, my friends, is the sweet fruit of the personalstudio revolution at the end of the 1990s.



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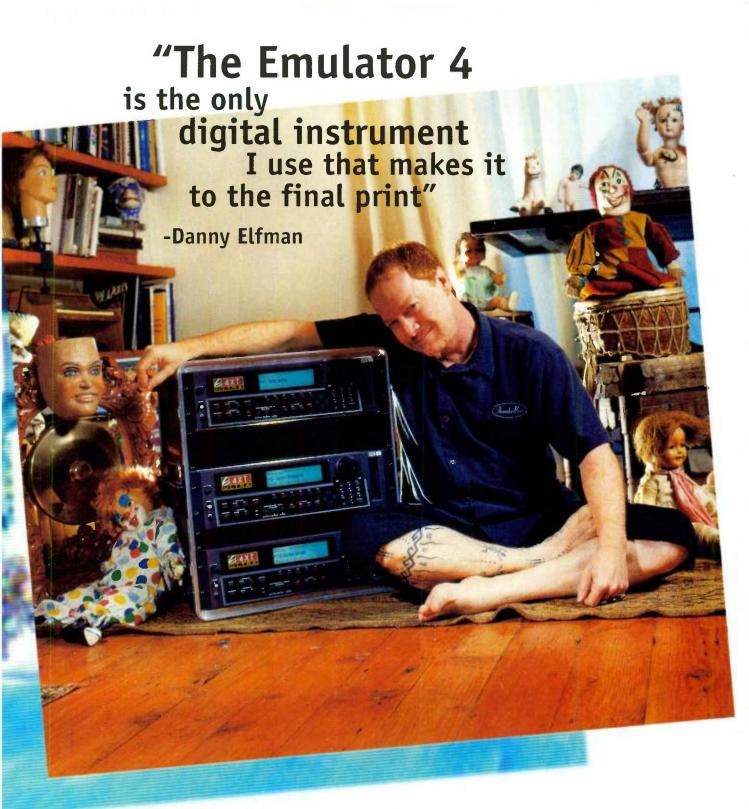
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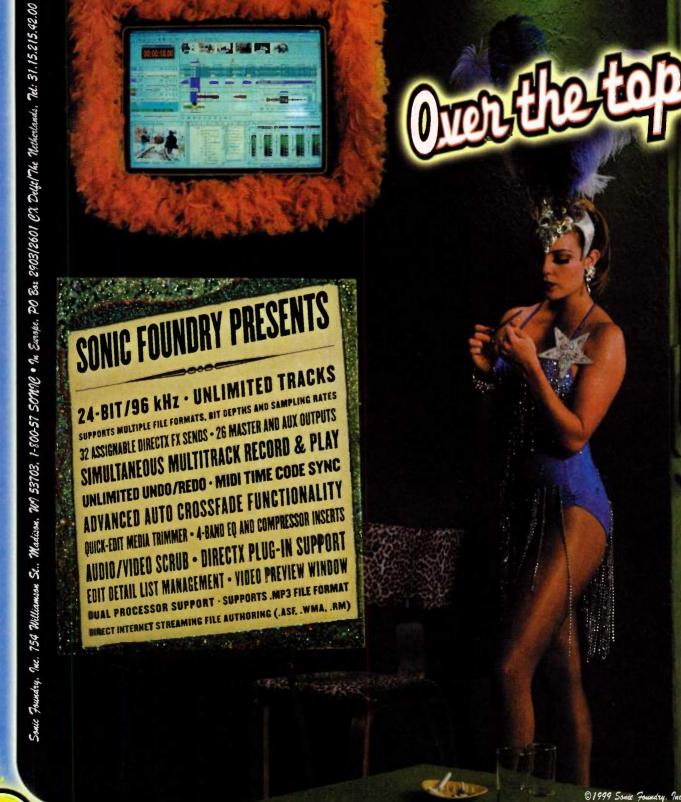
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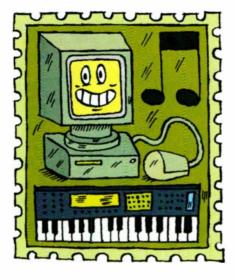


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LETTERS



THE VIBE IS THE THING

hank you for what was, flat out, one of the best-spirited yet fundamentally important interviews EM has published in years: "A Room with a Vibe" with Mark Hudson (August 1999). Hudson simultaneously moves forward while retaining what was right about the past. In an exciting era of true technological revolution, it cannot too often be restated that, in the creative arts at least, what is essential is feel. How refreshing to hear Hudson's embrace of such supposedly anathematic concepts as signal leakage and just doing what works, regardless of the eyebrows it might raise at an AES conference. His observation that "the technology is too accessible" is clearly not the rant of the Luddite but the insight of a practitioner who suggests we not be beguiled into thinking all those ones and zeroes are somehow smarter

than we are. It is hard to imagine concluding a reading of this interview without experiencing a newly rekindled enthusiasm to get into the studio. David Flitner

Methuen, MA

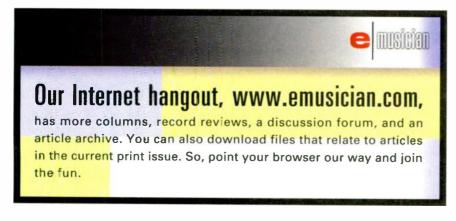
BOOKMARK THIS SITE

would like to alert fellow readers of *Electronic Musician* that "Desktop Musician: Music Sites for Web Surfers" (May 1999) failed to mention the Sonic Spot (www.sonicspot.com). I've found it to have one of the most indepth listings of music shareware and freeware for the PC. It also has the most comprehensive list of software synthesizers I've found anywhere, each with a good description and download available directly from the site. The Sonic Spot has become my home base for electronic-music software.

> L. Hess lh456@yahoo.com

PARTITIONING DILEMMAS

I'm using a Pentium 233 and Windows 98 in my home studio, and I just bought a second hard drive for recording my audio tracks. I partitioned my hard drives in 2 GB sections, thinking that this would be more efficient for running my audio tracks. Is this the correct approach, or should I partition the two hard drives as two 8 GB drives? The Windows 98 manual says that partitioning your hard drives as one large partition will run more



efficiently using the FAT32 system. But which method works best for hard-disk recording? Please help me understand the difference.

> Bill Rodgers rodgeed@aol.com

Bill—Using FAT32 to partition your hard drives offers many advantages, such as smaller cluster sizes, which is a more efficient use of space. But other issues can adversely affect digital audio performance if you use FAT32. Among these is the fact that the drive heads must access data far more often when FAT32 is used than if you had partitioned your drives with FAT16, and this could slow down drive I/O.

In reality, however, I don't think that you're going to notice much of a difference between the two, and I doubt that you'll get worse than one or two fewer tracks if you use the more modern partitioning system. FAT32 does allow you to use much larger partitions, as you noted. But remember that once you make the switch to FAT32 (if you're converting existing FAT16 drives), there's no going back. Also keep in mind that Windows NT cannot recognize FAT32 drives, so if you happen to be running a dual-boot system, forget about it!—Dennis M.

STOMPING AT THE MACKIE

From your complex answer to the question of using Boss pedals with Mackie 8-Bus mixer ("Letters: Pedals and Mixers," July 1999), it seems that you don't truly understand impedance and interfacing. I am an electronics engineer and own a small studio.

I have used the Boss pedals on numerous mixing consoles and have had good results without DI boxes or matching transformers, which are not required. Boss pedals have input and output impedance very similar to rackmount devices, except for the 1 M Ω input impedance, which makes little difference. Of all the impedance mismatches, this is the easiest to fix: simply put a resistor (say, 10 k Ω) across the input to the pedal. You don't need (continued on p. 16)



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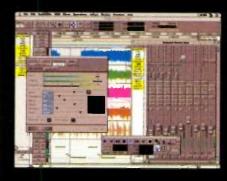
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LETTERS

(continued from p. 12)

anything else as long as you are driving the input with a low-impedance output such as a Mackie send. Having said that, I have never actually needed to fit a resistor, as the 1 M Ω input works fine as long as you watch the levels. The pedals have a relatively low output impedance, so interfacing into the high-impedance Mackie return is not a problem either.

This brings me to my second point: the pedal should work okay as is. I would suggest that the reader has a gain-structure problem (input too high, output too low) or a problem with the power supply to the pedal (a flat battery or low supply voltage). I've had better results in recording by just running off the battery, as there are then no hum problems.

> Stuart Smith captured_audio@one.net.au

THE DOPE SHOW

hank you for making such a dope magazine. I'm a newcomer to the recording world and involved with hip-hop, so there's a lot of gear that could be used for our particular art form. Your magazine breaks down all the extraneous talk and gives straightforward reviews of products. I really appreciate this, because I am trying to build a home studio and don't want to invest money on a product that is just media hype.

Secondly, I would like to thank you for providing actual recording techniques in your pages. It seems that every other magazine assumes that you are an engineer who knows the do's and don'ts of recording, and they choose to skip over these vital things. Your tips on compression and mic placement have helped out my recordings greatly.

> Lazarus Essential Media tolbert@etcrier.net

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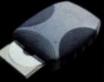
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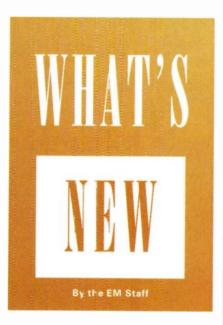


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EIVALUE





🔺 MARSHALL MXL 2001

Arshall Electronics has made a serious pitch to budget recordists with its MXL 2001 fixed-cardioid, largediaphragm condenser mic (\$199). The MXL 2001 has a 1-inch gold-sputtered Mylar diaphragm and a machined brass body.

According to Marshall, the mic's overall frequency-response range is 30 Hz to 20 kHz, with a fairly even on-axis response and only a slight boost above 5 kHz. It can handle sound pressure levels up to 130 dB.

The MXL 2001 ships with a mic-stand adapter. Options include a high-isolation shock-mount (\$49.95) and a 48V phantompower supply (\$69.95). Marshall Electronics, Inc.; tel. (310) 390-6608; fax (310) 391-8926; e-mail sales@mars-cam.com; Web www.mars-cam.com.

Circle #401 on Reader Service Card

BUCHLA MARIMBA LUMINA

A new mallet-instrument MIDI controller has arrived: the Marimba Lumina from Don Buchla. The marimba-style instrument has

52 electronic bars, above which are two ribbon controllers. The limited Gold Edition (\$8,500; pictured at right) has gold-plated bars; a lower-priced version will be available in several months.

The Marimba Lumina offers lots of expressive control: it responds to several new variables, such as note density and the mallet's position along the length of the bar. Four color-coded mallets are in-

cluded, and the instrument is sensitive to which mallet has struck a bar, allowing a different musical response for each.

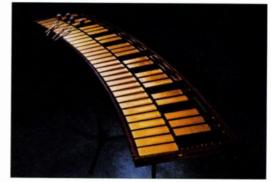
The Marimba Lumina provides 48-note polyphony, comes with 50 editable presets, and allows the range of bars to be divided into zones for splits and layers. You can edit programs with the aid of an 80-character LCD.

The controller has a row of jacks along

SPL CHANNEL ONE

The Channel One (\$999) from SPL is a channel strip that comprises a tube preamp, de-esser, compressor/limiter with noise gate, and EQ. The processors in the new unit are based on designs previously implemented in separate SPL products. The unit also has a headphonemonitoring section designed so that computer-based recordists can hear the processed signal directly, avoiding latency and phase problems in the monitor signal.

The Channel One's mic/line preamp features 48V phantom power, a highpass filter, and phase reverse. An unbalanced ½-inch input with separate gain control lets you use the channel strip as an instrumentlevel preamp. The de-esser section has an automatic threshold and a knob for attenuating sibilant frequencies centered around 8 kHz. The compressor/limiter uses its right side. These include MIDI In, Out, and Thru ports; ¼-inch TRS inputs for footpedals, footswitches, and the like; ¼-inch stereo outputs; and an AC power



connector. Also on the right panel is a small memory-card slot for storage and retrieval of up to 50 programs per card.

The Marimba Lumina folds in half for easy transport. It ships with a soft case, a stand, and two memory cards. Buchla and Associates; tel. (510) 528-4446; e-mail mlumina@buchla.com; Web www .buchla.com.

Circle #402 on Reader Service Card

a dual-VCA design for increased transparency and lower distortion.

Finally, the Channel One's equalizer features two semiparametric EQs: one operates between 30 and 700 Hz, and the second ranges from 600 Hz to 10 kHz. The unit also has an "air" band at 15 kHz and a distortion knob for adding tube coloration.

The rear-panel mic input and output are on balanced XLR jacks, and the line input and insert are on ¼-inch TRS connectors. The Channel One can be equipped with an optional 24-bit, 96 kHz A/D converter (\$360) and Lundahl transformers (\$255). SPL rates the unit's frequency response at 20 Hz to 100 kHz and dynamic range at >112 dB. Sound Performance Labs USA/ MTC (distributor); tel. (718) 963 2777; e-mail info@spl-electronics.com; Web www.spl-electronics.com.

Circle #403 on Reader Service Card



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act is, if you purchase your gear from one of the major retailers, you're going to get a great, low price. The big stores all carry the top brands, receive volume discounts from the manufacturers, and then "price-shop" each other to make sure they're not undersold.

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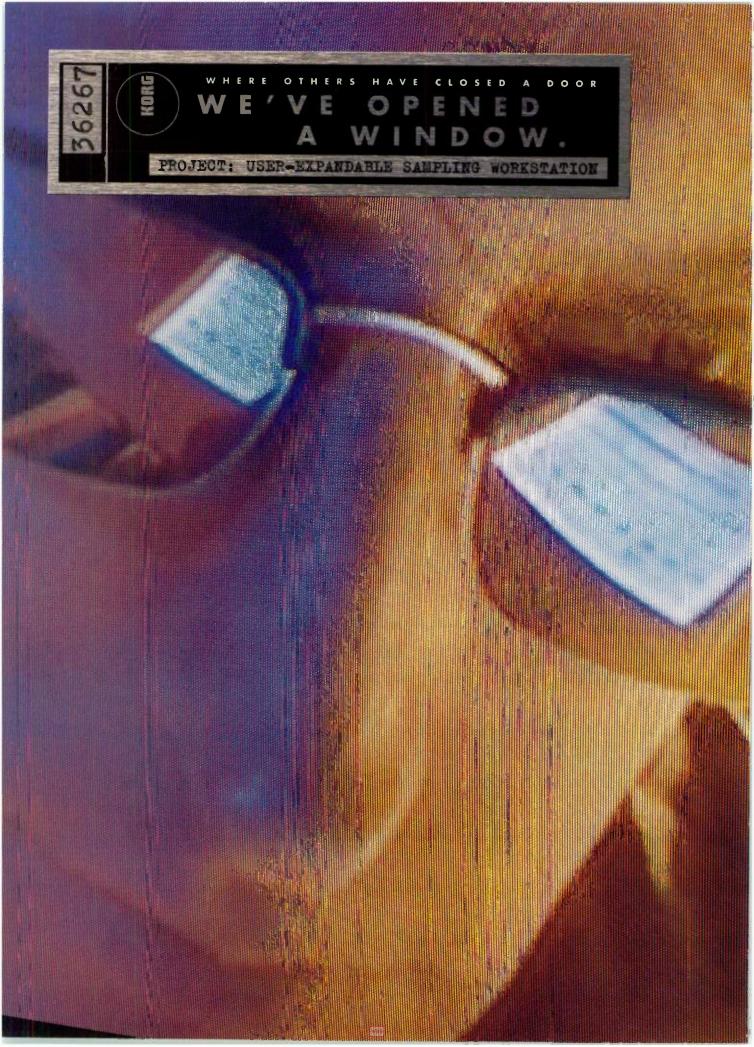
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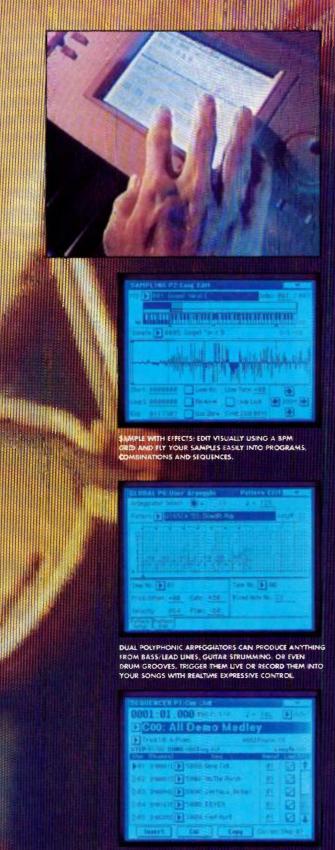
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SOUND CONSTRUCTION ISO BOX

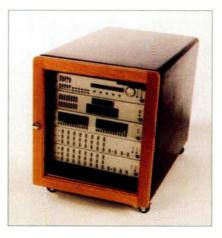
Sound Construction and Supply has introduced the Iso Box as a solution to the problem of digitally produced noise in sound-sensitive environments, such as recording studios. By eliminating noise from hard drives and other gear, the Iso Box allows you to keep important yet noisy equipment close at hand in your work space.

Constructed of medium-density fiberboard to contain hard-disk "seek" noise, the Iso Box is lined with 1-inch Auralex acoustical foam to minimize spindle (audio drive) noise and white noise. The ¼-inch laminated glass door reduces noise even further. The rear of the box opens for easy access to the back of

your gear and is equipped with an HVLPfan cooling system.

The unit's electrostatic intake filter keeps equipment dust free, while parabolic diffusers made from Owens Corning 703 1-inch insulation provide wideband pressure absorption. Red fan lights and a digital thermometer on the front door allow you to confirm that equipment is running at an optimum temperature.

The Iso Box comes in 12-, 16-, and 20rackspace sizes (\$750, \$950, and \$1,100, respectively), all of which are 34 inches deep. The unit comes stock in a black lacquer finish; you can also order it with oak or maple doors and a clear finish. Sound Construction & Supply; tel. (615) 313-7164; fax (615) 313-7799; e-mail



tbeeten@custom-consoles.com; Web www.custom-consoles.com. Circle #404 on Reader Service Card

🔻 YOWZA MUSEDIT

wza Software presents MusEdit (\$79), an inexpensive new musicnotation program for Windows. The software is as simple to use as a word-



processing program; you can do note entry using a mouse, computer keyboard, or MIDI keyboard and build a score of any length, with as many as 20 instruments. The program also tran-

scribes Standard MIDI Files. You can play *MusEdit* files back on your computer or save them as Standard MIDI Files, and you can play off of your *MusEdit* scores while they automatically scroll on your screen.

MusEdit offers several types of notation, including standard treble and bass clefs, rhythm notation, guitar tablature, and chord diagrams. The program easily translates between standard notation and tablature; it can also interpret "text-tab" notation—the format you'll often find available on the Web and rewrite it in regular tablature. You can add lyrics in any font.

In tablature notation, you have the option of showing stems to indicate note value. The program comes with a dictionary of more than 8,700 chords, and the Chord Designer lets you define your own chords with tablature and diagrams for four-, five-, or six-string instruments, using up to seven frets.

Yowza includes a 170-page manual along with *MusEdit*. You can run the program on a 80486 or Pentium PC with Windows 95, 98, or NT and a minimum of 8 MB of RAM. Yowza Software; tel. (800) 234-0427 or (510) 908-0027; fax (510) 528-7475; e-mail info@musedit.com; Web www.musedit.com.

Circle #405 on Reader Service Card

INTERVAL TAB TRANSCRIBER 3.0

Transcriber 3.0 (Win; \$35), from Interval Software, is a notation program that transcribes Standard MIDI Files into tablature for guitar or any other stringed instrument. You can combine more than one MIDI track into a single *Tab Transcriber* file, and the program allows you to work on multiple files separately. Once you have put your music in notation form, you can edit it with the mouse by clicking on the fretboard position to change fingerings. *Tab Transcriber* also can transpose to any key.

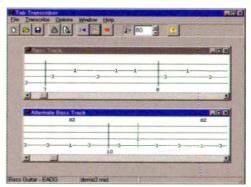
If you select the Open String Bias op-

tion, the transcription will use open-string positions wherever possible. You can also choose from more than 25 predefined tun-

ings, and the Tuning Library allows you to add your own custom tunings. To create a custom tuning, you simply indicate the number of strings and frets on the instrument and name the open strings.

Tab Transcriber supports MIDI playback, and your scores can be exported as ASCII files for printing. The program will run under Windows 95, 98, or NT, on a 75 MHz Pentium with at least 16 MB of RAM. Interval Software; e-mail mibra@cam.org; Web www.cam.org/~mibra.

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IN AN EFFORT TO AVOID ANY MULTI-FUNCTION COMPARISON TO THAT OF A SWISS ARMY KNIFE, WE HAVE ELIMINATED THE TOOTHPICK.





M-OO The comparison would have been flattering, however, the M-OO is worthy of high praise on its own merit. Not only is the M-OO extremely versatile, articulate and accurate, it delivers tremendous output, along with surprising bass and clarity for a monitor of any size. And while the M-OO is built to the construction standards of a polar ice breaker, its compact nature makes simple duty out of schlepping it from one session to the next. And because the M-OO is magnetically

shielded, it is ideal for use with PC based workstations. Further, it is sold separately so you can easily gang together 5.1 systems or daisy chain up to 10 M-OO's per channel for fixed installations. As for the toothpick, all considered, we simply figured it was something you could learn to live without. www.nhtpro.com



(4.weighted @ 1M) THD @ 90 dB SPL: <1.0% (100Hz - 10kHz @ 1M), Response: ±2 dB (1/3 oct. swept noise): 98Hz - 20kHz @ 1M, 93Hz - 20kHz @ 2M. -6dB LF cutoff: 80Hz SPECIFICATIONS: Amplifier power: 75W continuous ims/ch, 150W (100ms peak). Peak acoustic output: 111d8 SPL (100 ms pink noise @ 1M). Residual hum/noise: <20 d8 SPL

(in-room response). Monitor Dimensions/Wgt: 9"h x 5.7"w x 7.3" d, 14 lbs. Monitor Enclosure Materials: Cast aluminum/zinc alloy body, mica-filled polypropelene bafile.

Professional Audio by Vergence Technology NHT is a registered trademark of the Recoton Corporation



EVENT PROJECT STUDIO

In the second se

Each monitor has an amp that supplies 70W of continuous power to the woofer and 30W to the tweeter. The PS5 and PS6 feature components designed especially for the Project Studio series, including transducers, amplifiers, and active crossover circuitry. The PS8 has the same woofer, tweeter, and crossover used on the 20/20bas but has a newly designed amplifier.

Fourth-order crossovers are used on the PS6 and PS8; the PS5 uses a secondorder crossover. The crossovers on the PS5 and PS6 are set at 2.6 kHz; those for the PS8 are set at 2.2 kHz. The crossovers have customized filter shapes designed for better amplitude and phase blend from the woofer to the tweeter.

Event rates the frequency response of the PS5 at 52 Hz to 19 kHz, the PS6 at 45 Hz to 20 kHz, and the PS8 at 35 Hz to 20 kHz (±3 dB). Each set of monitors features balanced XLR and ¼-inch input connectors and front-panel clip LED indicators. Event Electronics; tel. (805) 566-7777; fax (805) 566-7781; e-mail info@ event1.com; Web www.event1.com. *Circle #407 on Reader Service Card*

ALESIS ADAT/EDIT

White the action of the action

The system consists of the ADAT/PCR PCI card and the software applications

ADAT/Edit and ADAT/Connect. The ADAT/PCR card has ADAT Optical I/O, as well as ADAT 9-pin synchronization input and output. Eightchannel synchronous transfers can be accomplished at 16-, 20-, or 24-bit resolutions.

The ADAT/Connect software recognizes ADATs connected to your computer and lets you transfer audio tracks between the MDMs and the computer. You can designate start and end points and name each track within the program. Tracks are transferred with singlesample accuracy.

Codeveloped by Alesis and Emagic, ADAT/Edit is a multichannel audio-editing program that features standard cut, copy, and paste functions, as well as DSP effects such as reverb, time expansion/ compression, pitch shifting, and EQ.

AUDIX ADX-50

udix has released the ADX-50 (\$299), an electret-condenser microphone designed for use in studio, broadcast, and sound-reinforcement applications. The mic features a cardioid polar pattern and can handle sound pressure levels of up to 132 dB. The body of the ADX-50 is machined from solid brass and has a black finish. Drivers included with the package give you the option of working with other popular multichannel editing applications, including Emagic *Logic Audio*, Cakewalk *Pro Audio*, MOTU *Digital Performer*, Steinberg *Cubase*, and Opcode *Studio Vision*.

Also included in the package are two 3-meter ADAT Optical cables and a 3-meter ADAT Sync cable. Alesis Studio Electronics; tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www .alesis.com.

Circle #408 on Reader Service Card



The ADX-50 includes a mic-stand adapter and windscreen. Optional accessories include the APS-2 2-channel phantom-power supply (\$119). Audix rates the ADX-50's frequency range at 40 Hz to 18 kHz (±3 dB) and self-noise at 28 dB. Audix USA; tel. (800) 966-8261 or (503) 682-6933; fax (503) 682-7114; e-mail info@ audixusa.com; Web www.audixusa.com. *Circle #409 on Reader Service Card*



🕨 KURZWEIL K2600

urzweil's K2600 series is the latest in the company's line of sampling synthesizer workstations. The K2600 is available in three forms: an 88-key weighted-action keyboard (\$6,820), a 76-key semiweighted action keyboard (\$6,256), and a 3U rack-mount module (\$5,175).

The keyboard versions feature eight sliders, a 600 mm and 20 mm ribbon controller, pitch and mod wheels, a floppy-disk drive, and an LCD. The sliders, pitch wheel, and mod wheel are all user assignable. Each unit has eight outputs, all on ¼-inch balanced connectors. The D/A conversion for the outputs is 20 bit. You can purchase a factory-installed sampler upgrade for \$795 that includes a ¼-inch audio input with a 20-bit A/D converter. The signal from this input can be processed in real time through the K2600's entire architecture. Also on the upgrade are AES/EBU



and S/PDIF optical digital I/O and stereo analog input on balanced XLR and unbalanced %-inch connectors. There is additional 8-channel Kurzweil KDS digital I/O.

The K2600 has more than 450 preset programs and a variety of acoustic, analog-synth-style, and techno sounds. There is also a stereo grand piano and modeled tonewheel organs. The K2600 is compatible with all K2500-format sample CD-ROMs, as well as the K2500's optional Orchestral and Contemporary sound ROM boards. You can expand the synth's ROM to 44 MB and sample RAM to 128 MB. The K2600 ships with the company's KDFX, which gives you DSP effects on four stereo insert buses and a stereo master-effects bus. A 32-track sequencer is included.

The K2600 has MIDI In, Out, and Thru connectors, in addition to inputs for four footswitches and two continuous controllers, one of which can be assigned as a breath controller input. Kurzweil Music Systems, Inc./Young Chang; tel. (253) 589-3200; fax (253) 983-8206; Web www.youngchang.com/kurzweil. *Circle #410 on Reader Service Card*

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🕨 EBTECH SWIZZ ARMY

When you're rushing around five minutes before showtime, trying to troubleshoot and replace the bad cable connection that revealed itself only moments ago, a cool head will get you only so far. Ebtech's Swizz Army 6-in-1 cable tester (\$149.95) is designed to help you finish the job, giving you the ability to test the continuity of XLR, balanced and unbalanced ¼-inch, RCA, Tiny Telephone (TT), MIDI, and balanced and unbalanced ¼-inch connectors.

An LED shows the pin-to-pin wiring for any cable. The display also features an intermittent-signal detect function that shows which wire has a failure. You can also check nstalled cables by testing from one end of the cable with a shorting cap placed on, then taken off, the other end.

The Swizz Army can generate test tones of 440 Hz and 1 kHz, selectable between +4 dBu, -10 dBV, and

-50 dB (mic) levels. In addition, you can test for phantom power on pins 2 or 3, and you can use the unit to confirm that



an XLR connector's shield is connected to pin 1 so that it is grounded. When you hold the Reset button while entering Test Tone mode, the Swizz Army becomes a metronome. Each press of the Reset button increases the metronome's rate from 40 to 220 bpm in 1 bpm intervals. Ebtech; tel. (858) 271-9001; fax (858)

271-9079; e-mail sales@cymation.com; Web www.cymation.com.

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📕 M AUDIO CO3

A new digital signal converter has been released by M Audio, a division of Midiman. The CO3 (\$249.95) converts between AES/EBU, S/PDIF coaxial, and



S/PDIF optical (Toslink) formats. The back panel of this ½-rackspace unit provides an input and an output for each type.

The front panel of the CO3 has an inputsource button; selecting a source format illuminates the appropriate LED and disables the other two inputs. Another LED lights up when a valid audio stream is detected. The SCMS button gives you four copy-protection options to pick from: no copy protection, pass-through, force "original," and force "first generation."

Whichever source you use, the CO3 can send it to all three outputs simultaneously, putting the signal through unmodified to the corresponding, same-format output while performing the necessary conversions for the other two. Alternatively, the CO3 can send a SCMS-encoded S/PDIF signal to the coax and optical outputs while converting the unencoded signal to AES/EBU.

The CO3 has built-in jitter correction to ensure the integrity of the converted signal. It accommodates sample rates up to 100 kHz and resolution up to 24 bits. Midiman; tel. (800) 969-6434 or (626) 445-2842; fax (626) 445-7564; e-mail info@midiman .net; Web www.midiman.net.

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CARD GAMES 🔺 🔺 🔺

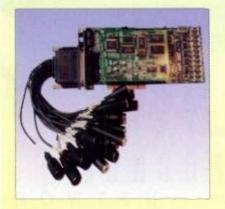
V SONORUS

New from Sonorus is the MedI/O (\$599), a multimedia sound card designed to provide all the connections that you need to work with your DAW. You get a grand total of 12 simultaneous I/O channels—8 ADAT digital, 2 S/PDIF digital, and 2 analog—and the MedI/O can sync with a StudI/O card to add even more channels.

The card has a 24-bit optical port that you can switch between ADAT Optical and S/PDIF (cable included), and a massive breakout cable attaches the rest of the connectors. These include stereo XLR mic inputs equipped with phantom power; two sets of stereo ¼-inch balanced line inputs; a ¼-inch guitar input; a CD-ROM audio input; stereo S/PDIF I/O on RCA jacks; stereo ¼-inch balanced line outputs; and a ¼-inch headphone output. The breakout cable also has word-clock I/O and two MIDI In, two MIDI Thru, and six MIDI Out ports.

Onboard compression, limiting, and gating are available for the MedI/O's vocal and guitar inputs. The guitar circuit also offers speaker-cabinet simulation. The card has a 64-voice DirectX synth/sampler engine and provides an interface for connecting a WaveBlastercompatible daughterboard.

The card comes with drivers for Windows 95, 98, and NT; WDM; DirectX; and ASIO. EASI drivers and drivers for Mac OS, BeOS, and Linux are in the works. Thinkware (distributor); tel. (800)



369-6191 or (415) 777-9876; fax (415) 777-2972; e-mail info@sonorus.com; Web www.sonorus.com.

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EMAGIC

B magic's Audiowerk2 Production Kit (\$299) includes the Mac- and PCcompatible Audiowerk2 PCI audio card and a CD-ROM that includes several customized Emagic applications. The Audiowerk2 uses the same 18-bit A/D/A converters as the company's Audiowerk8 card. Two channels of unbalanced analog input and output and S/PDIF digital I/O are supplied on RCA connectors. You can also use the analog and digital outputs independently, giving you a total of four output channels. Sound Manager, MME, and ASIO drivers are included.

Software bundled with the Production Kit includes Emagic's cross-platform ZAP data-compression software and the company's *MicroLogic AW2* (Mac) and *Logic Audio Composer* (Win) digital audio sequencers.

You also get Emagic's WaveBurner AW2 (Mac), a program for mastering and burning CDs. WaveBurner AW2 enables you to master Red Bookcompatible audio CDs. The program can read mono, split-stereo, or interleaved-stereo files in AIFF or SDII formats. Resolutions of 16 and 24 bits are supported, as are 32, 44.1, and 48 kHz sample rates. You can also convert 32 and 48 kHz signals to 44.1 kHz in real time. WaveBurner lets you perform nondestructive edits with automatic and custom crossfades.

The program also supports all PQ editing functions, so you can insert track and index numbers, ISRC identification code numbers, SCMS copyprotection code, and more. Emagic USA; tel. (530) 477-1051; fax (530) 477-1052; e-mail emagic@emagicusa.com; Web www.emagic.de.

Circle #414 on Reader Service Card

VOYETRA TURTLE BEACH

he Montego II Home Studio (Win; \$299) from Voyetra Turtle Beach is a package that includes Voyetra's *Digital Orchestrator Pro* and other software and the Turtle Beach



Montego II sound card. Two wavetable synths ship with the system. One uses a Roland GS-compatible chip set that provides 64 sounds. The second has 64 hardware-synth-based sounds and 256 software-synth-based sounds that load into 4 MB of PC system RAM. In addition, you get a DLS sampler.

The card features 18-bit A/D and D/A converters and samples at up to 48 kHz. Montego II has an unbalanced %-inch mic input; analog, stereo line inputs on a %-inch TRS connector; and analog stereo output on two %-inch TRS jacks. Stereo S/PDIF digital I/O is provided on Toslink optical and coaxial RCA connectors. The card's joystick port connects to a supplied MIDI interface.

The system requires at least a Pentium 75 MHz running Windows 95, 98, or NT 4.0. Turtle Beach rates the Montego II's frequency response at 20 Hz to 20 kHz (±1 dB), total harmonic distortion at 0.005% (A weighted), and signal-to-noise ratio at >97 dB. Voyetra Turtle Beach, Inc.; tel. (800) 233-9377 or (914) 966-0600; fax (914) 966-1102; e-mail sales@tbeach.com; Web www .voyetra-turtle-beach.com.

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WRH

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Ready to experience a whole new level of creative inspiration? Take giant steps forward with the world's favorite synthesizer family. The **QS6.1**th, **QS7.1**th and **QS8.1**th keyboards and **QSR**th module...available today at your Alesis Dealer.



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Version 8 for PC & Mac is here—Automatic Accompaniment has arrived!

The award-winning Band-in-a-Box is so easy to use! Just type in the chords for any song using standard chord symbols (like C, Fm7 or C13b9), choose the style you'd like, and Band-in-a-Box does the rest... automatically generating a complete professional quality five instrument arrangement of piano, bass, drums, guitar and strings in a wide variety of popular styles.



- 100 STYLES INCLUDED WITH PRO VERSION: Jazz Swing Bossa Country Ethnic Blues Shuffle Blues Straight Waltz • Pop Ballad • Reggae • Sbuffle Rock • Light Rock • Medium Rock • Heavy Rock • Miami Sound • Milly Pop Funk • Jazz Waltz • Rhumba • Cha Cha • Bouncy 12/8 • Irish • Pop Ballad 12/8 • Country (triplet) • and 75 more! BUILT-IN SEQUENCER ALLOWS YOU TO RECORD OR EDIT MELODIES.
- BUILT-IN STYLEMAKERTM. You can create your own 5 instrument styles using the StyleMaker section of the program. SUPPORT FOR OVER 70 SYNTHS BUILT-IN. Drum & patch maps included for over 70 popular synths. General MIDI, Roland GS & SoundBlaster soundcard support included.
- STANDARD MUSIC NOTATION and leadsbeet printout of chords, melody and lyrics. Enter your songs in standard notation & print out a standard lead sheet of chords, melody and lyrics.
- UTOMATIC HARMONIZATION. You can select from over 100 barmonies to barmonize the melody track, or barmonize what you play along in real time. Play along in "SuperSax" barmony, or barmonize the melody with "Sbearing Quintet "Create your own barmonies or edit our barmonies.

AUTOMATIC SOLOING. Simply select the soloist you'd like to bear and play with (from over 100 available) and Band-in-a-Box 8.0 will create & play a solo in that style, along to any song! This is bot! These solos are of the highest professional quality, rivaling ones played by great musicians, and best of all, they are different every time!

NEW! ADDITIONAL FEATURES IN VERSION 8.0 Band-in-a-Box 8.0 for PC & Macintosh breaks new ground with over 80 additional features!

AND-IN-A-BOX 8.0 IS HERE! This major new upgrade to Band-in-a-Box includes over eighty new features! Among them, the most amazing new feature is called "Automatic Dsongs". Simply select the style of song you'd like to create, and Band-in-a-Box 8.0 will automatically generate a complete song in that style, in the key and tempo that you want, complete with intro, chords, melody, arrangement and solo improvisations. It will even help you out by auto-generating an original title for your newly created song! This is HOT! The songs created using Band-in-a-Box are of professional quality, and best of all they're different every time! And there's much more in version 8.0... on-screen guitar fretboard, animated drum kit display, long filename support, "undo" option and much more!

OUR CUSTOMERS LOVE VERSION 8.0! "I'm in awe... it truly writes great songs!... Band-in-a-Box is better than ever, it's just what I was boding for The Drum Screen is fun!... Hey, you guys actually read my Wisblist!... You've done it again, the Melodist is unreal!"

GREAT NEW STYLES FOR BAND-IN-A-BOX 8.0! Styles Disk #15 - "Nashville"! Styles Disk #16 - "All Blues"! 2 MEW disks of our most requested styles.

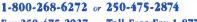
- Styles Dick #11: 22 Great NEW "Nashville" Country Styles ... \$29 As usual, we paid a lot of attention to your requests for Country styles as posted in the PG Music Forum - Styles wish list. Included are: Contemporary and Classic Country styles such as Honkytonk, Cajun, Rock, Hillbilly, Ballads, Blues, Shuffle, Rockabilly, and more. A few of the styles
- (eg. Shania.sty) cross over into the Pop/Rock genre. Styles Disk #16: 22 Great NEW "All Blues" Styles... \$29
- This collection has a good mixture of shuffle (swing 8ths) and straight 8ths blues styles which emulate the sound and feel of such groups as: Dr. J, B.B., Chuck B, James B, SRV, Curtis M, Eric C, Elmore J, Howlin' W, John-Lee H, and more.

BONUS BLUES JAMS! 40 Hot jammin' tracks in the key of C, and 40 more in the key of F! You'll never run out of the blues.

Soloist Disk Set 8: 3 NEW Soloist KnowledgeBases ("Killer" Jazz Waltz, Older Waltz, Jazz Fusion). An exciting aspect of the Soloist feature in Band-in-a-Box is that the program is able to increase its musical intelligence with new Soloist Disk Sets - it learns by "ear" and constantly gets better and better! This stunning new Soloist Disk Set includes new KnowledgeBase files as well as new Soloist definitions to extend and improve Band-in-a-Box Version 7.0 or higher. When it is installed in your bb directory you'll see new Soloists available and the existing Soloists will be automatically enhanced with dramatic results!

SPECIAL! SoloistPAK - all Soloist Disk Sets 2-8 PLUS Bluegrass MIDI Fakebook ... \$99 Available on CD-ROM or floppy disk

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BAND-IN-A-BOX PRICES

- FIRST-TIME PURCHASE (Windows® or Macintosh)
- Band-in-a-Box Pro Version 8... \$88 Ver. 8, Styles Disks 1-3, Harmonies Disk 1, Soloist Disk Set 1 + Melodist Disk Set 1.
- Band-in-a-Box MegaPAK Version 8... \$249 The MegaPAK contains "the works"—Version 8 -Version 8 PLUS Styles Disks 1-14, Soloist Disks 1-8, Melodist (1), Fakebook and Video add-ons.

GREAT

DEAL!

- BAND-IN-A-BOX VERSION & UPGRADES (Windows® or Macintosh)
- Regular Upgrade to Version 8 from Version 7 (requires Version 7) ... \$49 Ver. 8, Styles Disk 12 + Melodist Disk Set 1. Available on floppy disks or CD-ROM. Regular Upgrade to Version 8 from Version 6 or earlier or crossgrade ... \$59
- Includes regular Version 8 update above and Soloists Disk Set 1 VERSION 8 MegaPAK UPGRADES (Windows® or Macintosh)

Contains "the works"-Version 8, ALL add-on Styles Disks, ALL add on Soloists Disk Sets.

- The MIDI Fakebook, & PowerGuide GD-ROM video instruction
- MegaPAK upgrade from Version 7 (requires Version 7) ... \$149 MegaPAK upgrade from Version 6 or earlier or crossgrade... \$159
- ADD-ONS FOR BAND-IN-A BOX:
- NEW! Styles Disk #16 "All Blues" ... \$29
- NEW! Styles Disk #15 "Nashville" Country Styles... \$29 Styles Disk #14 Jazz/Fusion jazz rock fusion styles... \$29
- Styles Disk #13 Euro-Tek dance/pop/Techno styles ... \$29
- Styles Disk #12 (included with Version 8 upgrade) ... \$29
- Styles Disks #4-11... each \$29
- NEW! Soloist Disk Set #8 Killer Jazz Waltz, Older Waltz, Jazz Fusion... \$29
- Soloist Disk Sets #1-7... each \$29 SPECIAL! Soloist PAK - all Soloist Disks 2-8 + Bluegrass MIDI Fakebook ... \$99
- The MIDI Fakebook for Band-in-a-Box... \$29
 - Includes 300 songs in a variety of styles: Traditional/Original Jazz & Pop 50 songs; Classical (Mozart, Beetbowen, etc.) - 200 songs: Bluegrass - 50 songs

COMPREHENSIVE VIDEO INSTRUCTION FOR BAND-IN-A-BOX

Band-in-a-Box PowerGuide CD-ROM Video... \$49 Includes Volume 1 (Basics) and Volume 2 (Advanced) of "Inside Band-in-a-Box"

SYSTEM REQUIREMENTS: Windows® 98, 95, NT, 5.1; 8 MB available RAM: fast 486 or better; 15 MB available disk space (Proversion); any sound card (e.g. Sound Blaster) or MIDI module (e.g. Rohard Sound Canvas). Macintosh: OS 7.5 or later; 68020 or better, including any PowerPC (601, 603, 604, G3 or iMac).

HELP! I forgot to send in the Registration Card, but I want to upgrade now. No problem. Since the pgrade checks for any previous version of Band-in-a-Box, you can order the upgrade even if you forgot to register.



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THE ROCK GUITARIST

Multimedia Guitar Program

ability to mute or solo any audio track

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Listen to hot session players perform great

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"chops" - nearly an hour of bot jazz plus tins

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HCHORALES

THE BLUES GUITARIST Multimedia Guitar Program

Professional fully featured music program containing studio-recordings of great electric blues guitar music. Listen to hot session players perform great sounding blues music, while you learn the riffs, licks and tricks! This interactive program has great "chops"--- nearly an hour of hot blues plus tips and techniques.

THE BACH CHORALES Volumes 1 & 2 Multimedia Vocal Program

Inspiring performances of J.S. Bach's famous four-part Chorales by a professional choral ensemble, complete with a detailed multimedia history of the composer's life and times. On-screen notation, lyrics and chord progressions in perfect time with the singers

THE ROCK SAXOPHONIST Multimedia Instrumental Program

Fully featured professional music program containing studio-recordings of great rock n' roll saxophone music. Hot session players perform great sounding rock music, while you learn the riffs and tricks! Seamlessly integrated MultiTrack audio, MIDI, chord symbols, and music notation and chord progressions

owerTracks



here are over 20 new features in PowerTracks Pro Audio 5.0... Stereo recording, VU meters fo recording/playback levels. Leadsheet Notation window, Drum window with animated display of drum instruments for playback and recording, long file names and more (over 20 new features in all). PowerTracks Pro Audio is a professional, hully featured digital audio & MIDI workstation, packed with features for musicians, students & songwriters. With seamlessly integrated digital audio/MIDI recording, and built-in music notation, PowerTracks turns a typical soundcard equipped PC into a music production powerhouse!

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each volume only \$49 ormance Pri



THE MODERN JAZZ PIANIST mo Performance Program

The Modern Jazz Pianist is the software that makes it "too easy" to learn how to be a great jazz planist. Top studio musicians Renee Rosnes Miles Black. Ron Johnston, and Brad Turner perform over 50 tunes in a wide variety of nodern jazz styles, such as those by Herbie Hancock, Fred Hersch, Cedar Walton, Mulgrew Miller and many others. PLLS Song memos. biographies, and information on important modern jazz nianists



THE GOSPEL PIANIST

Plano Performance Program The Gospel Pianist is a powerful program for playing and studying a piano style that is both universally appealing and which underlies much of the blues, jazz and popular music played today. Over 50 "Gospel Style" piano standards played on MIDI keyboard by top Gospel pianists. Includes Music Trivia questions, Guess the Song game, program notes, pianist biographies (all on disk) and much more. Powerful gospel piano performances with that "old-time" feeling!

THE BLUEGRASS BAND Instrumental Performance Program

Our most "feel good all over" program so far, with more than 50 virtuoso performances of Bluegrass standards played live on MIDI equipped bluegrass instruments (banjo, fiddle, hass, guitar and mandolin). We've recorded top Bluegrass musicians, these MIDI files are hot! PLUS Lots of Bluegrass pictures, biographies, and trivia (all on disk) and much more. Dazzling performances to make you "feel good all over

CONDITIONAL

THE PIANIST The Pianist



Plano Performance Program The Pianist is a music program containing an amazingly comprehensive collection of nearly 900 of the world's greatest Classical Piano sterpieces, performed by world-class concert pianists! PLUS Music Trivia questions, Guess the Song game, program notes, bios (all on disk) & much more! Vol.1: 215 selections; Vol. 2: 200 selections: Vol. 3: 170 selections (incl. arrangements & duets); Vol.4: 200 selections; Vol. 5: Complete Beethoven Sonatas

THE LATIN PLANIST

Piano Performance Program The Latin Pianist features popular Latin pianist Rebeca Mauleón-Santana (editor of Sher Music's Latin Real Book) playing over 50 tunes in a wide variety of Latin piano styles. Includes authentic Latin and Salsa plano songs and styles such as Conga, Cumbia, Merengue, Son, Mambo, Cha-cha-cha, Guaracha, Samba, Partido Alto, and much more. This program is hot, hot, hot!

THE BLUES PIANIST

Piano Performance Program The Blues Planist comes in two volumes, each with over 50 great down-home blues plano stylings by top professionals playing a wide variety of blues piano styles - Boogie Woogie, slow & fast boogies, Jazz blues, New Orleans style, Chicago blues and more. These are the styles made famous by Pete Johnson, Albert Ammons, Jelly Roll Morton, Meade Lux Lewis, etc. Full of info and trivia on the great piano blues masters



THE BARBERSHOP OUARTET Volumes 1 & 2

Multimedia Vocal Program

All-time favorite Barbershop songs and an interactive multimedia history of barbershop singing in America. Made with the assistance of SPEBSQSA (Society for the Preservation and Encouragement of Barbershop Quartet Singing in America)

ALL MULTIMEDIA/MIDI PERFORMANCE SOFTWARE TITLES FEATURE...

✓ Separate audio tracks for each part ✓ Solo, mute, combine and mix the tracks independently & Transpose the music to the key of your choice 🖌 Focus on any section with the versatile loop feature 🧹 Slow parts down for further study with the 1/2 time feature 🖌 Choose audio and/or MIDI playback & Print the parts & Control audio playback with the mini-mixer window 🖌 Transpose or change tempo 'onthe-fly' 🖌 Jump to any position in the song 🖌 Jukebox mode for continuous play 🖌 Mark and play your favorite songs Adjust volume, panning settings for individual parts ✓ Split the piano into right and left hand parts automatically Play along with the performance in real-time on any instrument V Much more!

THE NEW ORLEANS PIANIST Plano Performance Program

Over 50 "New Orleans Style" piano music standards, played on MIDI keyboard by top New Orleans pianists Henry Butler, Jon Cleary, Doc Fingers, Tom McDermott, Joel Simpson and David Torkanowsky. This is the wonderful 'rolling', 'bluesy' New Orleans piano style made famous by Professor Longhair and Dr. John. This program makes it "too easy" to be a great New Orleans planist!

THE JAZZ SOLOIST

Instrumental Performance Program The Jazz Soloist is a music program with professional jazz quartet arrangements of over 50 songs (per volume). Each song features a great jazz solo played by top jazz musicians, as well as piano comping, bass and drums. Includes a standalone "Jazz Soloist" program with MIDI files (files also included in Band-in-a-Box format). Vol. 1: Swing (50 pieces); Vol.2: Swing (50 pieces); Vol.3: Latin/blues/ waltzes (60 pieces)

THE CHILDREN'S PIANIST Plano Performance Program

The Children's Pianist includes over 70 great piano performances of the worlds best-loved children's songs - ideal for listening or singalong! The words are displayed in a large 'Karaoke" style display while the song plays so you can sing along! (Windows® only) These pieces are presented with the care, artistry, and craftsmanship that will spark the interest of young and old alike. Includes piano arrangement tutorials

THE NEW AGE PIANIST

Plano Performance Program Over 70 "New Age" and "New Age-Jazz" style piano pieces, performed by top New Age artists. This is a beautiful collection of solo piano compositions inspired by the natural world. Full range of piano techniques, from the style of George Winston to Chick Corea and Keith Jarret. Song memos, biographies and information on important New Age musicians, Includes photo album of stirring nature scenes and real time

THE CHRISTMAS PIANIST

Plano Performance Program The Christmas Pianist contains great piano performances of over 50 all-time favorite Christmas songs and carols - ideal for listening or singalong. The words are displayed in a large "Karaoke" style display while the song plays so you can sing along (Windows® version only)! The onscreen piano keyboard lets you see the music as it's played. Fill your home with wonderful plano music this Christmas!

IN DI



-M D B









New Version. Cool Features.

Same Low Price!

SOUND ADVICE 🔺 🔺 🔺

V SWEETWATER SOUNDWARE

he online store at Sweetwater Sound's Web site carries the company's full lineup of sample CDs. But now, if you're not sure you want to the Proteus 1, 2, and 3 (\$249). Each set comes on a SIMM card that the user installs into one of the Proteus 2000 four internal ROM slots.

Q Up Arts developed the 16 MB Holy

Grail Piano and creat-

ed the sounds using

Virtual-Model Sam-

pling, a technique

Q Up Arts developed

that is designed to

provide the most real-

istic simulation pos-

sible of the sounds

and mechanics of

an acoustic piano.

Aspects such as sym-

pathetic string reso-

nance, the kick and recoil of the hammers



buy a full collection, you can purchase and download individual sample sets formatted for Kurzweil K2000 and K2500. Sweetwater offers more than 200 sounds, which represent several of its acclaimed sample CD-ROMs including Total Stereo Session Drums, Ultimate Guitars, Exotic Instruments, and Finger Juice.

The SoundWare online store (www .sweetwater.com/soundware.tpl) allows you to browse by CD or by instrument type, or to see the complete list of available samples. You can preview sounds, and the documentation for each of the CD-ROMs is available online. Should you later wish to purchase one of Sweetwater's full CDs, the price you paid for any individual samples from that CD will be deducted from the cost of the disc. Sweetwater Sound: tel. (800) 222-4700 or (219) 432-8176; fax (219) 432-1758; e-mail sales@sweetwater .com; Web www.sweetwater.com. Circle #416 on Reader Service Card

🕨 E-MU

P-mu has released two new ROM sound sets designed for its Proteus 2000 sound module: *Holy Grail Piano* (\$295) and *Protozoa—Sounds of* as they strike the strings, and the resonance displayed when damping a string have been captured in stereo at various velocities and programmed into the sound set. The *Holy Grail Piano* sound set features all the same sounds as the Q Up Arts CD-ROM of the same name.

Protozoa is, essentially, the original Proteus 1, 2, and 3 synths in SIMM form. It provides all of the samples and presets from the original units, in addition to a bank with new, layered sounds taken from all three modules, giving you the ability to access the original sounds and use them with the Proteus 2000's greater polyphony, in addition



to its filters and effects. E-mu-Ensoniq; tel. (831) 438-1921; fax (831) 438-8612; e-mail info@emu.com; Web www .emu.com.

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SYNTHFOOD

Synthfood, a new company based in the United Kingdom, has released its first product, a \$20 bank of 128 Patches and 32 Performances that can be loaded into Roland JVand XP-series synths. Included are pads that evolve over a long period of time, a wide range of basses, drones, ambient beds, and a variety of ethnic sounds. To download a free demo of the patches, go to the Synthfood Web site at www.btinternet .com/~synthfood.

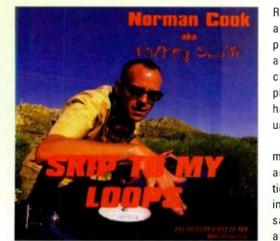
Circle #418 on Reader Service Card

🕨 AMG

MG has released a new series of CD-ROMs (\$49 each), with audio files created in Steinberg *Re-Cycle*'s REX format. These audio loops, sliced and diced within *ReCycle* version 1.7, have been encoded so that they are able to instantaneously follow tempo changes within both Mac and Windows versions of Steinberg *Cubase*, without the need for time stretching or compression. Each audio file is also provided in WAV, AIFF, and audio formats.

Eight discs are currently available.

160 DB: The Ultimate Drum 'n' Bass Collection includes drum breaks, upright bass licks, deep synth bass, bizarre effects, Rhodes licks, and more. Underfire 1 is a collection of drum loops and more from British artists Terminalhead. Beatz Diffusion features drum loops, many of which are 16 bars long, and fills from Tony Mason, Keith



LeBlanc, and others. The more technominded Dance Diffusion features beats from such artists as ColdCut and Black II Black. Pop Diffusion broadens the palette with synths, guitar, and percussion, some of which have been culled from previous AMG releases such as Guitarras Atomicas.

Erasure's Vince Clarke (Lucky Bastard) and Fatboy Slim (Skip to My Loops) are each given their own REX CD-ROM. Both of these discs feature a collection of drum and synthesizer loops. Dave Ruffy's Ruff Cutz rounds out the collection, with drum loops that have been extensively edited, equalized, and ReCycle'd by Ruffy. AMG/Big Fish Audio (distributor); tel. (800) 717-FISH or (818) 768-6115; fax (818) 768-

4117; e-mail info@bigfish.com; Web www.bigfishaudio.com. Also available from Steinberg North America; tel. (818) 678-5100; fax (818) 678-5199; e-mail matt@amguk.demon.co uk; Web www.amguk.co.uk.

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ILIO

lio Entertainments has released a new, comprehensive, string-orchestra sample library. *The Virtuoso Series String Orchestra* (\$995) multiple-CD- ROM set, created by violinist and composer Kirk Hunter, provides an assortment of articulations for realistic orchestral creations. The samples were recorded in a large hall in Los Angeles for its natural ambience.

The violin samples offer the most variety in section sizes and articulations. Violin sections of 24, 8, and 2 players are included, along with solo-violin samples. Trills, tremolos, up and down bow strokes, marcato, pizzicato, and

sordini are just a few of the articulations sampled for violin.

The celli also come in three section sizes (10, 6, and 2 players) and as a solo instrument, while the violas are offered only as a 16-player section. For these two instruments you get espressivo, section slides, hard and soft bow attacks, pizzicato, trills, and tremolo. The double basses were recorded in a 5-player section; bass samples include espressivo, with hard and soft attacks, and pizzicato.

The Virtuoso Series String Orchestra is available in Akai, Roland, Kurzweil, SampleCell, and E-mu EOS sampler formats. Ilio Entertainments; tel. (800) 747-4546 or (818) 707-7222; fax (818)



707-8552; e-mail ilioinfo@ilio.com; Web www.ilio.com. Circle #420 on Reader Service Card

V KORG

File format. Two disks contain only Trinity atterns. Wave Sequences holds 200 patterns that re-create the effect of wave sequencing, originally featured on Korg Wavestation synths. Trendy Loops



is a collection of 200 drum, pad, and bass sequences created for use in hip-hop, trip-hop, and techno.

The hybrid Trinity and SMF disks are designed and named for specific styles of music. *Trance* and *Techno* each feature 150 patterns, both incorporating drum loops, fills, bass lines, and synth leads. *Garage, Jungle*, and *Hip-Hop* each give you "full-band" sequences and allow you to bring up each of the instruments individually, as well.

Each disk in the collection ships with *Click and Play*, a cross-platform tutorial CD-ROM that features more than 400 MB of information. Korg USA, Inc.; tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

Circle #421 on Reader Service Card

There's No Reason To Wait Fo First Royalty Check

odeling amps shouldn't be just for the rich, a light that the way we feel. Introducing the JT50 Minage, the newest member to the Johnson family. Keeping in the tradition of designing amps for the real working guitarist, the JT50 focuses on giving you complete flexibility over your tone and effects, something the competition assumes you don't need.

Let's start with the 12 modern and vintage amp models. The JT50's exclusive V•Tube digital technology delivers the warmth and dynamic response you would think only possible from a real tube amplifier. Accurate EQ points closely emulate some of the most sought after amps in the industry. To compliment your tone, a comprehensive effects section gives you 3 effects at a time. Select from one of 6 Mod/Pitch effects, Delay, and 2 Reverb types, each with individual controls that allow you to tailor your sound to your needs (not just some canned settings like those other modeling amps provide). The JT50's speaker compensated headphone output provides the means for practicing when running a speaker may not be desired or can be used as a stereo direct out for recording or gigging, or if needed, you can use both the speaker and direct out simultaneously. A stereo FX loop keeps your favorite effects processors from gathering dust. The 21 user preset locations give you plenty of room for your own creations. And with a respectible 50 Watts of power, you are assured of having enough juice for any live situation. How about all of this for \$650?

Now that price is NOT the ssue, we think you owe it to yourself to audition the JT50 Mirage at your nearest Johnson dealer. You will be glad you did.

Your









Johnson JT50

12 Different Amp Models

Authentic EQ Points

6 Different Modulation Effects

Fully Controllable Delay

Programmable Hall and Spring Reverb

Speaker Compensated Direct Out

Headphone Out

Effects Loop

21 User Presets

24 bit A/D-D/A Converters

50 Watt RMS Power Amp

Johnson Premium Loud Speaker

Optional J8 Foot Controller

Optional JT3 Foot Switch



A Harman International Company

KEY CHANGES

Be, Inc., has announced a development partnership with Echo Corporation. The companies have joined to produce drivers that will allow Event's Layla, Darla, and Gina multichannel audio interfaces (which are Echo products) to be used on the BeOS platform...A new version of Sabine Inc.'s Positive Feedback booklet is available by mail from the company, from the Sabine Web site at www.SabineUSA .com, or as a free download in Adobe PDF format. The 20-page booklet explains how feedback is caused and offers several solutions for combatting it, as well as information on the company's feedback-attenuation and wireless-microphone product lines... Drum machines galore can be found at the Drum Machine Museum Web site at www.drummachine.com. The online museum was founded with the aim of "documenting, preserving, and archiving every drum machine ever made." You can find specs and photos of more than 60 drum machines, and you can download audio files from these units, as well. The museum also has become the exclusive distributor for Technosaurus analog synthesizers...Sweetwater Sound has posted the Music Industry Career Center, an online bulletin board that lets musicindustry employers and workers post job listings, résumés, and other related messages. You can visit the Career Center at www.music-careers .com...Antares Audio Technologies and Mackie Designs have teamed up to bring Antares's Auto-Tune (price to be announced) to Mackie Digital 8-Bus consoles. The pitch-correction software will run on a Mackie UFX effects-processing card ... Keyfax Software has added a page to its Web site, www.keyfax.com, that offers free samples of the company's Twiddly Bits Standard MIDI Files.

-Rick Weldon

🗸 YAMAHA S80

The Yamaha S80 (\$1,995) combines an 88-note, Aftertouch-sensitive, weighted-action keyboard with expandable synthesis capabilities. Weighing just under 54 pounds, the S80 features 24 MB of ROM preset sounds, which include many of the sounds from Yamaha's EX-series synths.

The unit's synthesis engine provides 64 notes of polyphony, 16

multitimbral parts, and 64 multimode resonant filters. Also included are onboard effects, pitch and mod wheels, a breath controller input, and an unbalanced ¼-inch microphone input that allows the S80 to process an external signal. The keyboard has more than 100 onboard effects, including reverb, chorus, delay, a 4-band equalizer, and all of the effect algorithms from the EX series.

Among the S80's 256 preset sounds are new stereo-sampled pianos, strings, brass, and choir voices. In addition, it has 128 user-programmable memory locations and 128 memory locations for storing Performance setups.

The S80 has two plug-in expansion slots, which allow the unit to hold any two boards from Yamaha's PLG-150 line of Modular Synthesis Plug-in Expansion

V Z-SYSTEMS Z-3SRC

A nupdate of Z-Systems' z-2src digital audio sample-rate and format converter, the new z-3src (\$1,500) adds support for 24-bit, 96 kHz audio. The z-3src is designed to provide quick conversion between 16-, 20-, and 24-bit files with any sample rate between 32 and 96 kHz.

Designed for multiple applications, from post-production and mastering to broadcast, the 1U rack-mount device boards. Six boards are currently available—VL virtual acoustic, AN, DX, XG, VH vocal harmony, and PF—many of which are complete synthesizers on cards, including independent effects processors. Thus, the polyphony and effects of the expansion cards supplements the S80's stock 64-note polyphony and effects. The cards' programming parameters can be accessed from the S80's front panel.

The S80 features a Toshiba Smart-Media storage slot to provide extra storage for user Voice and Performance banks and direct playback of type 0 Standard MIDI Files. The unit

> has MIDI In, Out, and Thru connectors, and a mini-

DIN jack for connection to a Mac or PC. The S80 comes bundled with a CD-ROM that contains Yamaha's *XGWorks* MIDI sequencing and editing software and several software utilities. Yamaha Corporation of America; tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web www.yamaha.com.

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features AES/EBU I/O on XLR connectors and S/PDIF I/O on RCA connectors. It also has Toslink optical input and output, and AES11 synchronization is offered on an XLR connector.

According to Z-Systems, the z-3src has a signal-to-noise ratio of 120 dB and THD+N of <-120 dB. Z-Systems Audio Engineering; tel. (352) 371-0990; fax (352) 371-0093; e-mail z-sys@z-sys.com; Web www.z-sys.com.

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STOP THE PRESSES

D igidesign announced the release of *Pro Tools* 5.0 software and introduced the Digi 001 (\$995) digital audio workstation (DAW), which includes *Pro Tools LE* version 5.0.

Unlike previous versions of *Pro Tools*, version 5.0 includes a fullfeatured MIDI sequencer; Digi-Translator for converting *Pro Tools* sessions to Media Composer and Symphony-based OMF files; and Avoption, which allows users to capture, import, and play back Avidcompatible video within *Pro Tools*.

The Pro Tools graphic user interface has been updated to include new scrolling preferences, grabber tools, and trim-tool options. Digidesign has also added keystroke combinations that will be familiar to Avid-style video postproduction users. Other interesting new features include the ability to timecompress/expand a region to picture and scrub while trimming a region. A new set of predefined keyboard macros provides quick access to commonly used editing commands.

Digidesign also has established a new division known as Digi, which will focus on products for the personal stu-



dio. The first of these products is the Digi 001, a DAW that includes a PCI card, a multichannel breakout box, and *Pro Tools LE*, which is a light version of *Pro Tools* 5.0 for Mac OS. (A version for Windows is planned for release sometime next year.)

The Digi 001 PCI card uses a Motorola 56301 DSP chip and includes a pair of optical digital I/O connectors that can carry either stereo S/PDIF or 8-channel, 24-bit ADAT Optical data.

However, the breakout box is where the action is. Designed to handle the chores usually relegated to a smallformat mixer in the personal studio, it can be rack-mounted and is small enough to sit on a desktop.

The breakout box offers eight analog inputs, configured as a pair of frontpanel XLRs with mic preamps and six back-panel unbalanced ½-inch jacks. The inputs to channel 1 and 2 use Neutrik XLR connectors that can accept ¼-inch TRS inputs at +4 dBu. Each of these inputs includes a gain pot, pad switch, a software-controlled highpass filter (fixed at 80 Hz), and 48V phantom power for the mic preamps. The gain for each of the six unbalanced inputs is software controlled.

Analog outputs include two ¼-inch TRS jacks operating at +4 dBu, six unbalanced ¼-inch jacks at -10 dBV, a pair of ¼-inch TRS monitor outputs at +4 dBu, and a ¼-inch stereo headphone jack. The box also provides stereo S/PDIF digital I/O on coaxial connectors.

Other back-panel connections include MIDI In and Out and a ¼-inch footswitch jack for record punch-in. The monitor and headphone outputs have front-panel volume controls.

Pro Tools LE is a light version of Pro Tools 5.0 that can handle up to 24 tracks of 16- or 24-bit audio and has extensive MIDI features. Pro Tools sessions can be opened and used in either the LE or TDM version of the program, allowing for greater flexibility when sharing work with other studios.

An interesting new development in Pro Tools LE is the Real-Time Audio-Suite (RTAS) plug-in format. According to Digidesign, RTAS plug-ins will operate in real time and will be light versions of their TDM siblings. For example, the most important parameters of a TDM plug-in will also be available in the RTAS version. However, RTAS plug-ins will load into system RAM rather than card-based RAM. Digi 001 comes with a number of RTAS plugins, and more are reportedly on the way from many of Digidesign's development partners.

-Gino Robair



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Nonvolatile RAM

A new type of RAM could enhance your gear's memory.

By Scott Wilkinson

redu work Relation in the power is turned off. Readonly memory (ROM) retains data without power, but that data can't be changed. (The data in so-called flash ROM can be changed, but this type of memory works too slowly to be an nand

effective replacement for RAM.) The concept of nonvolatile RAM that requires no power to hold its data seems like the Holy Grail of computer technology, and a number of companies are on the verge of finding it. Conventional RAM chips use tiny capacitors, which depend on a constant supply of power to retain the charges that represent 0s and 1s, but many new designs use magnetic techniques to store data.

There are several approaches being taken in the effort to create powerless, nonvolatile RAM. Among the magnetic possibilities is a phenomenon discovered only ten years ago called *giant magnetoresistance* (GMR), in which a magnetic field changes the electrical resistance of a thin conductive film. Honeywell has developed experimental chips based on this concept with capacities of more than 1 megabit. However, GMR devices consume a lot of current, which tends to burn them out when the memory elements are reduced to submicron size. Motorola is working on a way around this problem by doubling the strength of the GMR effect (which reduces the power requirements) in a device called a *pseudospin valve*.

IBM and Motorola are working on another technique, which depends on the quantum-mechanical ability of electrons to tunnel through a thin insulator sandwiched between two magnets. The current through the insulator depends on the orientation of the two magnetic fields. A team of IBM engineers has demonstrated such tunnel junctions in which individual bits are as small as 200 nanometers (nm) wide and can be switched in 5 nanoseconds (ns) or less. However, this effect is very sensitive to the depth of its thinnest layer, which is typically a film of aluminum just 0.7 nm (approximately four atoms) thick, making large-scale manufacturing difficult.

Yet another approach exploits a phenomenon discovered by Edwin Hall 120 years ago, in which a current moving in a thin film can be deflected to one side by a magnetic field. The Hall Effect is used in MAGRAM (magnetic RAM) devices being developed at Honeywell and the University of Utah, which have built 1-micron devices that can write bits in just 8 ns.

A different, nonmagnetic technique utilizes ferro*electric* devices, in which the position of an atom within a crystal lattice represents the value of a bit. A company called Ramtron uses ferroelectric crystals in conjunction with conventional CMOS semiconductor technology to build what they call FRAM chips. Each crystal consists of only a few atoms; when an electric field is applied, the central titanium or zirconium atom moves in the direction of the field (see Fig. 1). To change the state of the crystal, you simply reverse the electric field. If the electric field is removed, the atom stays in place, preserving the data.

Clearly, nonvolatile RAM would be a great boon to electronic musicians of all persuasions. The ability to retain data in RAM without power would provide enhanced security against power failures and propel portable music devices light-years ahead of their current limited applications. It will be very exciting to see this technology emerge from the lab and onto the commercial market for all to use and appreciate.

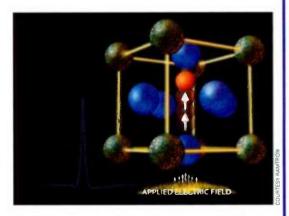


FIG. 1: In a ferroelectric crystal lattice, an electric field moves the central atom one way or the other, representing a digital 0 or 1.

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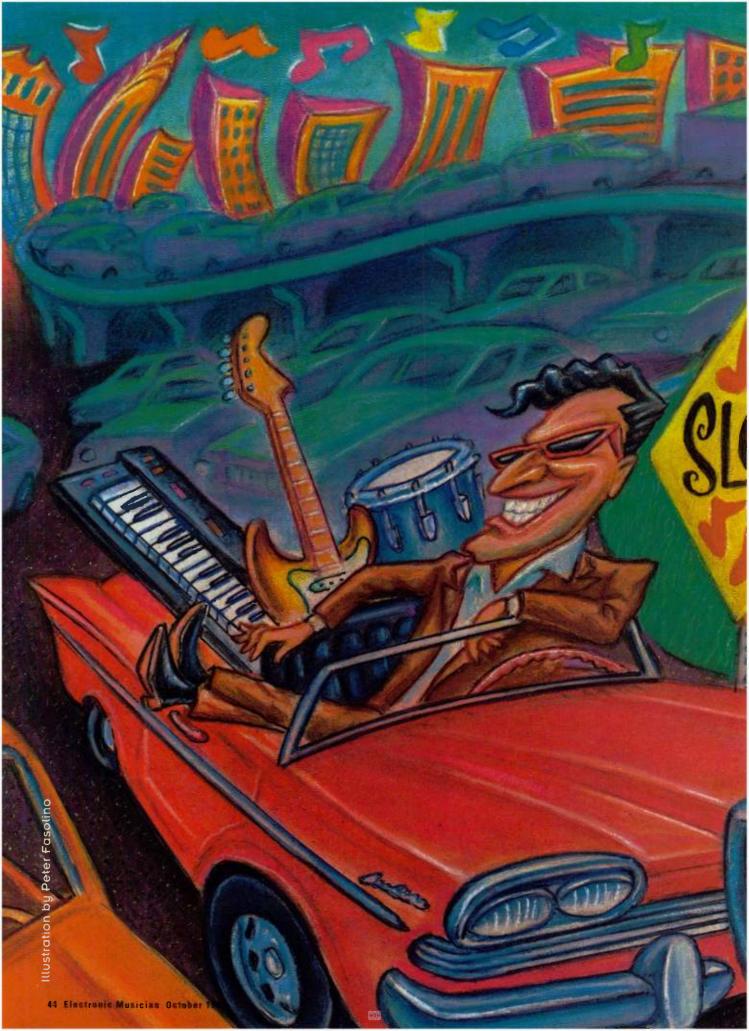


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How I learned to slow down and take notes one at a time.

in the

e live in a fast world: fast cars, fast food, *Fast Times at Ridgemont High*, and perhaps most important, fast music. At least that's how it seems when you're trying to learn a new riff or instrumental solo. Grabbing a hot lick off your favorite John Coltrane CD is invariably much more work than it should be. As any good music teacher will tell you, the key to learning a new piece is to start out by playing it slowly. If only we could get the musicians on those tapes and CDs to do the same. Fortunately, modern technology has come to our rescue in the form of several new digital recording tools.

A number of reasonably priced, easy-to-use hardware and software recorders are currently available that enable you to select a recorded section of music, loop it, and slow it down to the point at which the original blizzard of notes becomes a more stately procession of tones. And unlike the variable-speed tape recorders that many of us struggled with in days of yore, these new gizmos, and their software counterparts, maintain the pitch as they change the playback speed. That's because they use audio timecompression and -expansion algorithms similar to those found in high-end digital audioediting software. Some of these recorders have pitch-changing algorithms as well, so you can play in any key you want or match your tuning with a particular recording.

Because these recorders enable you to carefully scrutinize a musical phrase—note by note, if necessary—and play along in different tempos, they can be invaluable as instrument study aids. Teachers can even prepare lessons on tape with the expectation that the student will practice at home with one of these devices. Furthermore, because these products make it possible to transcribe even the most difficult musical passages, they enable you to take challenging instrumental parts that

are unavailable in print and put them down on paper. In fact, a number of music magazines accept transcriptions for publication if they are carefully notated and, above all, accurate.

By David Rubin



As you can see, these specialized recorders can be many things to many people. To give an overview of the market, I'll explore several popular hardware recorders, including Reed Kotler Music's TR-1000 (\$249.95), Akai's U400 Riff-O-Matic (\$249), and Sabine's BT-316M BackTrak (\$149.95). On the software side, I'll look at Reed Kotler's Transkriber 1.2 (Mac/Win; \$49.95) and RePlay Technologies' CD Looper Pro 2.0 (Win; \$99.95).

Reed Kotler calls its TR-1000 a "digitalmusic study recorder," Akai describes its device as a "variable-tempo phrase sampler," and Sabine calls its recorder a "riff decoder and digital sampler." To keep things simple, I'll refer to them throughout this article as "loop recorders." Let's begin by looking at some general features to consider when shopping for a hardware loop recorder.

SO, SO SLOW

Although each of these devices specializes in recording and slowing down music, it's important to recognize that you pay a penalty in terms of sound quality when you use them. Slowing a musical phrase to one-half (or less) its original speed while maintaining the origi-



FIG. 1: The Akai U400 Riff-O-Matic packs several great features into a compact, die-cast aluminum box.

nal pitch takes some heavy processing. As you might expect, audio fidelity pretty much goes out the window. Even the most sophisticated time-expansion algorithms produce noticeable and often intrusive artifacts at these slowed-down levels.

Depending on the source material and the playback speed, these artifacts typically appear as a kind of helicopterlike stuttering or chattering effect. Sometimes the music takes on an "underwater" quality. Reverb tails and other background ambience in the original material may produce tremololike effects, and artifacts that resemble chorusing or flanging may also turn up. The quality of the slowed-down audio depends primarily on the nature of the algorithms that are being ap-



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plied to the source. Remember, these products are designed specifically for identifying pitches and not for transforming the "Flight of the Bumble Bee" into a new-age meditation classic.

These recorders have no internal moving parts; they record everything into RAM, so they must balance the need for audio fidelity against the limitations of available memory and recording time. None of the units give you the option of increasing the RAM. Reed Kotler's TR-1000, however, does offer standard and extended-record modes, allowing you to trade audio quality for more recording time, much like the standard and long-play recording options on a VCR.

The TR-1000 records a generous 95 seconds at 22 kHz or a whopping 190 seconds (more than 3 minutes) at 11 kHz, both with 8-bit resolution. That's enough to record an entire song—a handy feature. The Akai U400 Riff-O-Matic offers 16-bit resolution with a 29.4 kHz sampling rate, which yields a recording time of 35.7 seconds. Sabine's BT-316M BackTrak combines 16-bit audio with a 14 kHz sampling rate for 30 seconds of recording time.

The BackTrak is unique in an important way: it saves your recording in memory, even when you turn the power off. (The other two units lose the recorded material when you turn off the power.) So you can practice a difficult riff over several days without rerecording it each time.

And speaking of power, the BackTrak and TR-1000 provide a power on/off switch,



FIG. 2: The Sabine BT-316M BackTrak is the only loop recorder with "flash memory," which allows the unit to store recordings even after you turn it off.

but you have to unplug the Riff-O-Matic to turn it off. That makes an otherwise well-designed unit unnecessarily awkward to use. These recorders all rely on detachable wall-wart power cords as their only power source.

PLAYING WITH JACKS

All of the hardware loop recorders offer an assortment of I/O options for

connecting to the outside world. The Riff-O-Matic and the BackTrak provide ¼-inch stereo jacks for headphones and line-in connections, and ¼-inch phone jacks for output. The TR-1000, on the other hand, provides a more solid ¼-inch stereo headphone jack and separate left and right RCA jacks for lineout and line-in connections.

All three devices include a volume





control for the headphones, and they also have separate ¼-inch instrument-in jacks so you can play along on an electric guitar or keyboard synth and hear your playing through the headphones. The TR-1000 actually provides separate Guitar (mono) and Keyboard (mono/stereo) jacks. The BackTrak includes a Split/Mixed switch that allows you to hear the recorded material on one channel and yourself on the other.

VALUABLE INPUT

The first step in using a loop recorder is to capture a phrase or riff. In theory, this seems simple enough; in practice, however, it can be tricky. All of the recorders provide an input-gain control but offer little in the way of metering. Optimizing recording levels is important with these devices, because when you slow things way down, recorded noise and distortion just add to the overall noise produced by the slow-down algorithms.

The Riff-O-Matic's rear-panel level control is small and awkward to use, but unlike the other devices, the Riff-O-Matic does provide a three-segment record level meter, which makes up for the inconvenience. The other two recorders have top-panel input controls, but the TR-1000 has only a single Overload LED, and the BackTrak lacks any indication of record level.

After you adjust the input levels, the next step is to record a section of music to loop. Grabbing just the phrase that you want (while limiting the material before and after) is especially critical on the BackTrak, because it gives you no way to trim or adjust the start and end points of the recording; in other words, what you record is what you loop. With the BackTrak, you begin recording by pressing the Play and Stop buttons together. Recording ends when you press the Stop button or when the memory is used up. The approach is similar with the TR-1000: press the FF/Rec button to start recording; press the button again to stop.

The Riff-O-Matic uses the same procedure as the BackTrak except that the Riff-O-Matic also offers a unique Continuous Record option. Pressing the Record button alone puts the unit in record standby mode. After you hear the desired riff, you press the Play/Stop button. The Riff-O-Matic then captures the previous 35.7 seconds of source material (or less if you haven't been in standby mode that long). This is a very handy feature, because it allows you grab a riff after you hear it go by, instead of having to anticipate when it's coming.

In most cases, it's quite difficult to start and stop recording at precisely the points where you want looping to begin and end. For that reason, the TR-1000 and Riff-O-Matic enable you to set and reset the loop points (also known as the playback region) within the recording. This feature is very important, because it allows you to focus on different notes and phrases in a recording without needing to rerecord the material each time. And you can be more lax in capturing the material, because you can always fine-tune your playback region later. Moreover, looping controls also let you repeat an extremely short segment, such as a single chord or note, and have it repeat continuously for extraclose scrutiny. The Riff-O-Matic even has a dedicated Freeze button for isolating individual notes. It captures a very short segment, which you can then shift forward or backward through the music to zero in on a particular note.

Loop points are set on the fly, much like punch-in recording on a tape deck.

The TR-1000 is especially powerful in its loop-editing capabilities, allowing you to shift the entire playback region forward or backward as a whole (in 0.1-second increments or continuously). Furthermore, it allows you to expand or contract the playback region by moving the boundaries incrementally or continuously. These functions also work well for closing in on short segments or chords. In fact, you can grab a tiny loop and then move it left or right, much like the Freeze function on the Riff-O-Matic. And unlike the other recorders, the TR-1000 lets you play a phrase through once without repeating, instead of automatically looping every time.

SLOW GOIN'

After you record a phrase and set the loops, all three recorders let you play back the music at several slower-thannormal speeds. The Riff-O-Matic offers only two slow-down speeds: 2/3 and 1/2. The BackTrak goes a bit further by adding a 1/3 speed. The TR-1000, however, is the undisputed champion in this category: it offers a tremendous 27 slow-down speeds ranging from 3/4 to an amazing 1/26 of normal.

What's more, the TR-1000 lets you choose from seven slow-down algorithms, which may provide better results when used for different types of instruments or ensembles. For example, one algorithm is for very fast notes, designed to more effectively break guitar strums into individual notes. I tried several of the algorithms on various recordings but always returned to the default setting, which seemed to give the best overall results. (In some cases, I couldn't tell the difference between similar algorithms.) The other algorithms might still be useful under certain circumstances; keep in mind that the algorithms are independent of the playback speeds, though some seemed to work better at particular speeds than at others.

So, how slow is slow enough? Well, that depends on the type of music that you work with. Blazing flamenco guitar

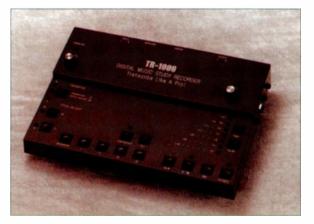


FIG. 3: The Reed-Kotler TR-1000 offers far more looping options and playback speeds than does any other loop recorder. Its seven algorithms and 28 playback speeds make it the most versatile unit available.

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solos, bluegrass banjo breakdowns, and bebop piano riffs may be hard to follow even at 1/2 speed, although most of the time I find that 1/2 speed works well for learning riffs. Once you feel comfortable with a phrase, the 2/3 speed is a nice, medium pace for practice. Being able to drop down to 1/3 speed can be useful every so often, especially for learning tricky passages, and 3/4 speed (offered only on the TR-1000) is sufficient for a little slowdown while still maintaining much of the audio quality.

Slower speeds (below 1/3) are primarily useful when transcribing music into standard notation or tablature. Reed Kotler, himself a professional transcriber, pointed out that superslow



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speeds can reveal subtle playing techniques that might otherwise go unnoticed. For example, guitarist Eric Clapton is known to play a note a half step or whole step below the target pitch and quickly bend the guitar string upward to the proper note. The technique happens so fast that it may go unnoticed unless you slow the recording way down. For proper notation, these details are important in capturing a particular style. And keyboard players working on Oscar Peterson or Art Tatum solos may appreciate speeds of 1/10 or even occasionally 1/26. Of course, when you slow down a piece of music to 1/26 of its normal speed, the sound quality is strictly from outer space. Nonetheless, it might come in handy now and then.



TUNING IN

The TR-1000 is the only unit in this group that allows you to change the pitch of a recording in small steps. It lets you fine-tune the pitch, in case the original source material was not recorded at the usual concert pitch (A = 440Hz), and it allows you to transpose into other keys. With the TR-1000 you can adjust the pitch up or down 100 cents (one semitone) in 10-cent increments for fine-tuning, or you can transpose up or down one step at a time through 12 semitones.

Transposing could make your life a little easier if, for example, you're a guitarist who is trying to learn an alto saxophone part. Just push a button, and a sax part in E_b comes out in E. In many instances, simply transposing a phrase can make it easier to transcribe. Keep in mind, however, that transposing a

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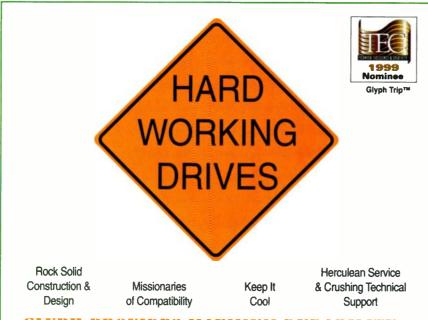
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phrase changes its playback speed.

The Riff-O-Matic is unique in offering a dedicated Vari Pitch function. When this function is activated, the pitch increases or decreases in proportion to the playback speed. In other words, the Vari Pitch button disables the pitch-maintenance aspect of the time-stretching algorithm and causes the unit to function much like an old variable-speed tape deck operating at full, 2/3, or 1/2 speed. That reduces the amount of processing needed for slowing down the audio and therefore yields better sound quality with fewer artifacts. But it also makes you change keys as you step through the speeds, so it's not terribly convenient for practicing riffs or



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transcribing music. The 1/2-speed Vari Pitch setting is the most useful, because it drops the pitch one octave. However, it can also make mid- to lowrange instruments sound quite muddy, so it's best reserved for high-pitched instruments.

The TR-1000 has a similar variablepitch capability: if you record at 22 kHz and play back at 11 kHz, the pitch drops an octave—giving you the same result as the Vari Pitch function without the usual artifacts. Alternatively, you can simply transpose a recording down by 12 semitones to achieve a similar result.

SEPARATION ANXIETY

Transcribing or learning a solo part from an ensemble recording can be particularly frustrating if you're unable to focus on the one instrument that you're interested in. Most loop recorders offer only a little help in this area. For example, to optimize results, the TR-1000 and the BackTrak allow you to select the right or left channel alone during recording. That can help you isolate an instrument if it is panned off-center in the mix.

The TR-1000, however, goes further by offering a Bass Isolator function. In essence, it destructively filters out the upper frequencies to emphasize the bass part. You can apply the Bass Isolator more than once to get "increasing bass isolation," but the sound quality quickly deteriorates, so it's best used in moderation. You can, however, improve the sound quality of the "isolated" bass part by transposing the recording up an octave.

The Riff-O-Matic offers a nondestructive midrange bandpass filter (called Solo) that attenuates the high and low frequencies to help you focus better on some guitar parts. You can turn this on and off with a single button, which makes it easy to compare the filtered and unfiltered versions as you practice.

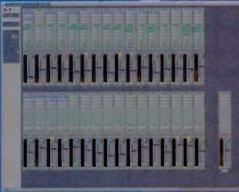
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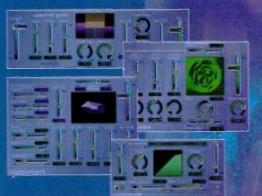
When it comes to assessing audio quality, it's difficult to draw firm conclusions about different loop recorders. That's because the slow-down algorithms themselves have the greatest impact on the sound, and each device takes a slightly different approach to the problem of slowing music while maintaining pitch. In other words,

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certain devices may sound better than others for playing a particular type of music at a particular speed. To compare the three recorders in this group, I recorded several selections of fast guitar and piano music and played the music at full and 1/2 speeds (as well as miscellaneous other speeds) while monitoring through a pair of Sony MDR-V6 headphones. All three units performed comparably, although there were definitely audible differences.

The Riff-O-Matic had the brightest sound with the most presence. Its clarity was helpful when listening to chords and arpeggios. The TR-1000 and the BackTrak sounded a bit duller, but they both did a better job of focusing on the main instrument while subduing the background instruments. That could be an advantage with some ensemble recordings. The TR-1000 produced an overall smooth sound that made it easy to follow notes at 1/2 speed.

In spite of its 16-bit processing, the BackTrak delivered no noticeably superior audio quality compared with its competition; the lower sampling rate apparently offset the higher resolution. In general, these devices all sound fine at full speed, and they perform competently at moderate to slow speeds. You just have to remember that the audio quality declines considerably the slower you get.

In the final analysis, each loop recorder's feature set and design characteristics will probably be the determining factor when it comes time to buy. To that end, here's a brief summary of the good news and bad news for each product.

Akai U400 Riff-O-Matic. The Riff-O-Matic (see Fig. 1) packs several good features into its compact, die-cast aluminum box, although the rear-panel level controls are a pain, and the lack of a power switch is inexcusable. This unit also loses points for offering only two slow-down speeds: 2/3 and 1/2 (although 1/2 speed will get you through most situations).

On the plus side, however, the Riff-O-Matic scores big for offering several features that other loop recorders lack, including a three-segment level meter, a Solo filter, and a handy Continuous Record mode. The front-panel layout is also easy to get used to.

The Riff-O-Matic's unique Note Grabber function captures and loops a very small audio segment, which you can slide forward or backward to home in on a particular note or chord. The unit's overall audio quality is excellent, and the 35.7-second recording time is usually adequate, although I often found it to be a few seconds short of what I would have liked. Setting and clearing loops is a breeze, and the Vari Pitch feature may appeal to those who yearn for the good old days of variablespeed tape recorders.

Sabine BT-316M BackTrak. Sabine took a decidedly minimalist approach to designing its highly affordable Back-Trak recorder (see Fig. 2). The palmsized unit lacks adjustable loop points, a level indicator, a fine-tuning function, filters, and transposition controls. In fact, it does little beyond recording and playing back loops of up to 30 seconds in duration.

The BackTrak offers three slow-down speeds, including a 1/3 speed for those extradifficult phrases, and the unit's front-panel level controls are easily accessible. The BackTrak offers 16-bit audio, although I found the audio quality to be no better than that of the other recorders.

The BackTrak's biggest claim to fame—which really is significant—is its ability to retain a recording in memory

even after you turn the power off. That makes it possible to work on a riff over a period of time without leaving the unit on or rerecording the phrase. This feature, along with the modest price tag, makes the BackTrak worth serious consideration, although the lack of other features is a definite drawback.

Sabine also continues to offer its less expensive BT-300 Back-Trak (\$119.95), which lacks the 16-bit audio and the power-off memory feature but is otherwise similar to the BT-316M.

Reed Kotler TR-1000. Measuring just over 7×10 inches, the TR-1000 is by far the largest and most substantial loop recorder in this group (see **Fig. 3**). It also offers the greatest level of control over your recorded material.

The TR-1000 can record for up to 190 seconds, which means that you can loop a whole song (or any part of it). That feature alone makes this unit a real standout, but the TR-1000 doesn't stop there. It also offers pitch adjustment, transposition, and an unrivaled 28 speeds (from full to 1/26) along with 7 algorithms to optimize performance. Loops can easily be established and then moved forward or backward in time. In addition, the playback region can be expanded or contracted.

The TR-1000's user interface is a bit arcane in areas, and its multifunction buttons take some getting used to, but the recorder is not difficult to learn. What's more, the TR-1000 is the only hardware loop recorder to offer a full set of tape-style transport controls (including rewind, play/stop, pause, and fast-forward). The ¼-inch stereo headphone jack accommodates pro-level headphones for serious transcription work, and the RCA output jacks make it easy to connect the TR-1000 to your stereo system. A Bass Isolator function reduces the high-frequency content in a recording.

Reed Kotler also markets the simpler, smaller, and less expensive TR-400 recorder (\$149.95). It records up to 150 seconds of music, plays back with a choice of ten speeds (from full to 1/8),

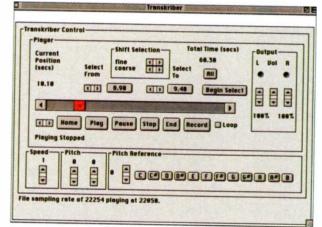


FIG. 4: Reed Kotler's *Transkriber* 1.2 offers most of the same features as the TR-1000 hardware recorder, along with a few additional features of its own.

and offers four slow-down algorithms. It also gives you control over the playback region and includes the Bass Isolator feature (along with a function to transpose up an octave). Overall, the TR-400 is a very good buy and worth considering.

THE SOFT SIDE

The hardware loop recorders in this group are all compact, portable, and easy to hook up. Most of the functions that these recorders perform, however, can easily be handled by a PC. Hooking things up to a desktop computer is not as convenient as with a stand-alone box, and you can't toss a computer into a guitar case or gig bag when you leave the house. Nevertheless, computers do offer some significant advantages.

For starters, they typically have much more RAM than the hardware boxes, which allows you to record longer sections of music for analysis. Furthermore, with a built-in CD-ROM drive, you can capture audio material directly from a CD, and with a hard drive you can save multiple recordings and settings for later recall. Those are both major conveniences, but there's more. Because most computers now support 16-bit, 44.1 kHz audio, recording and playback fidelity (even when slowed down) is noticeably better. Finally, software loop recorders cost much less than their hardware counterparts and provide a big-screen, point-and-click user interface for making adjustments and setting parameters.

Now let's take a brief look at two interesting loop-recording programs.

Reed Kotler *Transkriber* **1.2.** Reed Kotler's *Transkriber* is essentially a software version of the TR-1000 hardware recorder, available for both Macintosh and Windows. It offers most of the same functions and controls, although the onscreen user interface differs somewhat from the hardware front panel (see Fig. 4).

Nevertheless, you can still define and adjust the playback region in the same ways, and you can still choose from more than two dozen slow-down speeds (although only one alternate algorithm is provided). Aside from its pitchadjustment and transposition functions, *Transkriber* has a Pitch Reference feature. This lets you play a note through your sound card or computer when you click on 1 of 12 chromatic buttons. The buttons provide a handy reference if you don't have an instrument nearby. *Transkriber* also includes an Equalizer command, which offers several filters optimized for various instruments. A vocal/lead eliminator function facilitates transcribing rhythm-section parts from stereo pop recordings.

RePlay Technologies *CD Looper Pro* 2.0. RePlay Technologies would like to make *CD Looper Pro*

your default Windows CD-player utility. The company has created a handy control-panel interface with several powerful features beyond the usual transport buttons (see Fig. 5). For example, CD Looper Pro supports the CDDB Internet Database (www.cddb.com), so if you're online when you play a CD, song information such as artist, title, track name, and comments can appear onscreen. (Once you download the CD's information, it's stored on your hard disk, so you don't always have to go online.) The program also supports customized playlists with continuous or random-play options.

More importantly, however, *CD Loop*er Pro lets you create an unlimited number of loops per track by using the Punch In and Punch Out buttons. (You can set loop points as the CD plays or while it's stopped or paused.) If you click the Punch In button instead of the Punch Out button at the end of a phrase, *CD Looper Pro* automatically begins a new loop at that point. This is a great feature for quickly creating a succession of loops.

The loops appear in a list with Begin and End times and a useful Description field for adding helpful comments. After a loop is created, you can adjust its boundaries independently in oneand ten-frame increments. What's more, you can set the loops to play continuously or for a specific number of times, and you can add a pause between repetitions. You can even group loops together for learning a piece in sections, so as you learn each loop, you can combine it with others until you have learned an entire part or solo.

The current version of *CD Looper Pro* incorporates a new slow-down engine that offers speeds of 1/2, 1/3, and 1/4 in any of three algorithms. The audio quality with the default algorithm

apPoints Begin 00.20.25 00.36.02	End 00:33:72	Description
Begin 00:20:25	00.33.72	Description
00.20.25	00.33.72	Description
00 36 02		
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01:13:14	01:25:49	
02:53:07	02:58:07	
03:50:14	04:00:45	
P	unch in	Paranga
	02:53:07 03:34:49 03:50:14	025307 025807 033449 034569 035014 040045 Punch In

FIG. 5: RePlay Technologies' *CD Looper Pro* 2.0 offers a range of features for capturing and playing audio from CDs, including the ability to create and save multiple loops.

sounds quite good, even at the slowest speed. After you slow down a loop, you can speed it back up to full speed in 10 percent increments. Unlike *Transkriber* (which can play back at different speeds directly from RAM), *CD Looper Pro* converts a recording or CD track into a WAV file before it is processed. The program then overwrites the file with a new file each time you make changes. Because of that, expect to wait a bit (for processing) each time you change speeds or alter a file.

CD Looper Pro includes three proprietary plug-ins, which the non-Pro version of the program (\$69.99) lacks: *NoteGrabber* for importing and exporting WAV files, *OverDubber* for mixing your playing with the output of an audio CD, and *PitchChanger* for transposing and adjusting pitch. The extensive Help files are the only documentation.

FINAL BIT

There is clearly more to altering musical speed and pitch than one might first suspect, and ultimately a loop recorder's usefulness depends on whether it captures a suitable balance of power, flexibility, and price.

Because loop recorders are such versatile tools, they've found a home in a wide variety of applications, from beginning music lessons to professional transcription gigs. Different users, therefore, have different needs, and choosing the right product may hinge on the presence or absence of a single crucial feature. Whichever loop recorder you choose, though, it's great to know that digital audio technology can help us all slow down and become better instrumentalists.

Associate editor **David Rubin** is always looking for new ways to slow down while staying in tune with the universe.

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he fact that every professional mixing console comes with an equalizer on each channel speaks volumes about the importance of EQ in the recording and mixing process. Mixing a record without EQ would be pretty difficult: we'd be forced to rely on capturing a pristine performance from a perfectly tuned instrument, in an acoustically correct room, using expensive and accurate microphones that are properly placed in relation to the sound source. How often does that happen?

[7.M

So we have an arsenal of tools to help compensate, and EQ is arguably the most important one. But although every console has EQ, and most studios offer some sophisticated outboard units, not every engineer knows how to properly equalize a sound. Let's correct that problem by discussing the basics of EQ and the different types of equalizers and when they're traditionally employed. I'll also offer some specific applications that you may find useful when tracking or mixing your project.

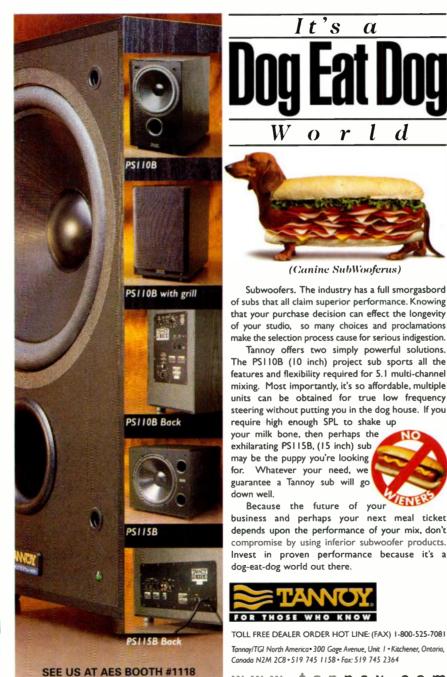
Nail your mixes and cover your tracks with these tried-and-true EQ techniques.

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THE GOAL OF EQ

To get some outside input, I called on two professional audio engineers from New York City. Derek Martin (who hastens to point out that he is no relation to Sir George) is a freelance music engineer who floats around the Big Apple, hitting studios such as Quad Recording, Waterfront Sound, and Apex Recording. His résumé is impressive, with credits ranging from Kool and the Gang to Nine Inch Nails. Greg Petricelli works at Midtown Digital and specializes in audio for video. He recently completed work on commercials for Nike and IBM. These two engineers provided valuable insights into how equalization is approached in different applications.



www.tannoy.com circle #528 on reader service card Before we can talk about specific applications, though, we have to ask the basic question: what's your ultimate goal when using the EQ process? Petricelli notes, "The purpose for using an equalizer is to change the frequency content of a signal. However, the reasons for changing the content vary. Usually it's to coax the very best sound out of an instrument."

In most cases, though, an instrument that sounds great by itself also needs to fit into the context of a mix. "Getting a track to blend in a mix so that its frequency response doesn't interfere with any other tracks is the very goal of the mixing process," says Petricelli. "Usually you want to eliminate frequencies that aren't needed by that particular instrument." But, Martin is quick to add, "oftentimes you'll use EQ to get a track to stand *out* in a mix, above the rest of the tracks—like a vocal, for example."

You can also use EQ to fix minor flaws present in a signal. For example, a 60 Hz AC hum can be notched out of an electric-guitar track, or you can use a de-esser to filter out unwanted sibilance in a vocal track. Although many sound designers use EQ creatively to sculpt sounds into new and different ones, for the purposes of this article we'll focus on how to use EQ within the context of a mix, be it for a music CD or a television commercial.

ALL SHAPES AND SIZES

Equalizers are essentially filters, although they can often boost as well as attenuate signals. Both active and passive EQs are available, and some products use a combination of the two designs (for example, passive high and low shelving bands and active mid bands). Active EQs traditionally employ op amps to boost and cut frequencies. Most console EQs are active, as are most outboard EQs except for a handful of high-end processors.

Passive filters cost much more than active ones and are, therefore, not commonly found in personal studios. They traditionally use coils or capacitors to attenuate frequencies; strictly speaking, they never boost. The tone controls on most guitars, for example, are passive filters. However, a passive EQ may have an amplification stage before or after the filter (or both). That means you can effectively boost a frequency band by cutting everything on both sides of it and amplifying the

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overall signal before or after cutting.

The most common, and the most useful, studio equalizer is the parametric EQ. A true parametric EQ affords you control over three parameters: gain, frequency, and bandwidth or Q (see **Fig. 1**). Until recently, most parametric EQs were outboard units, but many digital mixers now use them, as well.

Quasi-parametric EQs have gain and frequency controls but only a limited number of preset bandwidths. Just as common is the semiparametric, or midsweep, EQ. Semiparametric EQs have gain and frequency controls but no bandwidth control.

Shelving EQs work at either end of the frequency spectrum, attenuating or boosting all frequencies above or below a specific cutoff, or *knee*, fre-

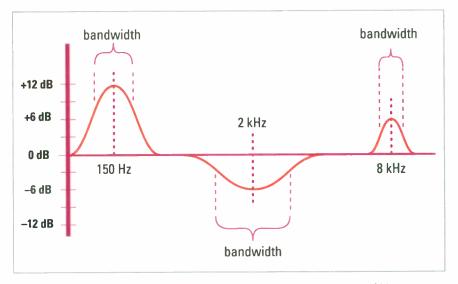
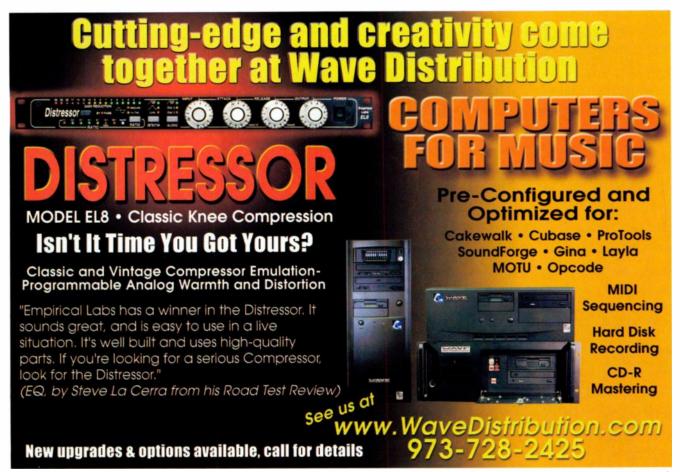


FIG. 1: A true parametric EQ affords you control over frequency, gain, and bandwidth.

quency. These equalizers are so named because the graph of frequency and amplitude looks like a shelf (see Fig. 2). Shelving EQs typically offer gain and frequency controls. There is usually a preset rolloff that gradually slopes from the knee frequency to the shelf; however, some of the more elaborate units available allow you to adjust this ratio.

The channel strips on many analog consoles have high and low shelving filters and a midrange bandpass filter with sweepable center frequency. Lowercost mixers often have shelving filters for the highs and lows and one fixedfrequency bandpass filter for the mids.



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Some consoles also offer a "rumble filter," which is a low-cut shelving filter with a preset cutoff frequency.

Graphic EQs, although commonly associated with home audio equipment and P.A. systems, are also useful in the recording studio. Graphic equalizers divide the frequency spectrum into a number of evenly spaced bands with fixed bandwidths and with gain controls for each band. The result is a number of fixed-bandwidth bell curves that can be boosted or attenuated as needed (see Fig. 3). Graphic EQs are usually used to "tune" the monitoring system. There are times, however, when you might want to patch in a graphic EQ on a track so that you can play with several frequency bands simultaneously (more on this later).

EQ ETIQUETTE

Although most people prefer to EQ during mixdown, some insist on equalizing during tracking. Martin, Petricelli, and I agree that dramatically changing an instrument's frequency content prior to recording is usually not the best approach. "You want the instrument to sound natural and good

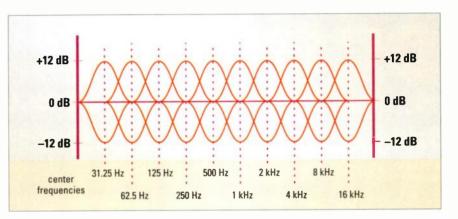


FIG. 3: A 10-band graphic EQ. Notice how each frequency is preset, as is its bandwidth.

going to tape," Martin explains, "and this involves a whole number of things. For example, you have to make sure that the instrument is a quality instrument that's tuned properly; as they say, 'garbage in, garbage out.' You also have to make sure you've miked the instrument properly, with the right capsule and from the right angle. But aside from that, I see no benefit in making sonic changes at this point; you're not going to know what you need to tweak until you hear the track in context with the other tracks."

However, Petricelli also points out that there may be certain instances when you want to address a problem before the signal gets printed. "If I had a high-frequency buzz coming through an amp," he says, "I would certainly want to get rid of it before recording—and if that meant using a notch filter, then so be it. It's less to worry about later."

That brings up the issue of adjusting the EQ on instrument amps. Both engineers agree that this falls into the category of tuning an instrument. "You wouldn't leave an out-of-tune tom-tom to be fixed in the mix with EQ," Petricelli points out, "so I see no reason why that doesn't apply to a poorly equalized guitar amp." Martin says, "Obviously you'll also be tweaking the amp track's EQ in the mix, but you still want that amp to sound great in the room, with the mic on it."

Another debate that inevitably arises is whether additive or subtractive EQ benefits a mix more. In certain instances only one approach would be appropriate; for example, using a highpass filter to roll off everything below 40 Hz is definitely a necessary application of subtractive EQ-there's no other way to go. But some stalwarts (especially jazz and classical engineers) take a more extreme position, insisting that nothing should be added if it isn't there already. These engineers use only subtractive filters. On the other hand, some rock and pop engineers simply turn up the knobs until they like the sound.

Most engineers, however, combine both philosophies and use whatever approach works best for the situation at hand. Petricelli explains: "I usually start equalizing using a subtractive approach, turning down the gain of a parametric EQ and sweeping the frequency knob until I hear something that needs to come out. As soon as I do that, though, I may realize that I need to add a few decibels at a higher frequency to compensate. I'll then repeat the process, with the gain of another

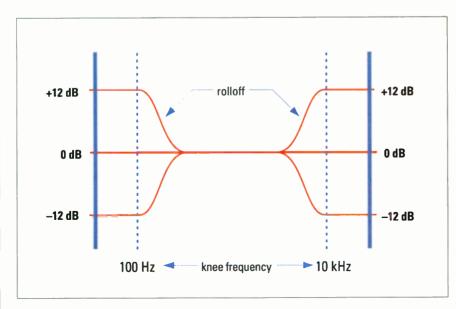


FIG. 2: Shelving EQs work at opposite ends of the frequency spectrums. Notice how the graph represents a shelf, hence the name.

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band turned all the way up, sweeping the frequency knob."

FOURTEEN EQ TIPS

Now that I've discussed some basic principals of EQ and the various types of equalizers, let's look at some specific EQ applications and the ways in which you can approach each. Most of these tips require at least a 3-band parametric equalizer.

Because drums are usually the hardest instruments to equalize, I've devoted some space to them. We'll start with the percussion instruments; work our way through string, wind, and horn instruments; and finish with the human voice and special applications. Keep in mind that you can apply these tips to digital samples, as well as to acoustic instruments.

1. KICK DRUM

A kick drum is one of the most important components of any mix, because it drives the beat (assuming you're not working with a chamber orchestra). But you must determine the type of character you want that beat to have.

First, you need to listen to the sound of the drum as captured by the microphone. This approach is imperative for any EQ application-you should always listen before you tweak. If the kick has a naturally hollow tone and is tuned correctly, you have a bit of room to maneuver. If the drum is padded and produces more of a "thud," your options are more limited, although you can still dial in a number of "kicking" sounds. Remember, if you're lucky enough to be involved in the entire recording and mixing process, you have the liberty to go back in the chain and make desired changes (like removing a muffler from a kick drum); if you're simply mixing a project, however, you must work with what you have.

Here we'll discuss three specific kickdrum sounds, all of which you can dial in if you have a track with an unpadded kick drum. Although these tips give you good starting points, you'll need to make tweaks because each drum is different. The first sound I call the '80s Big Hair Kick Drum. You know the sound: punchy, with lots of midrange and a bit of a thump.

Petricelli describes how he typically obtains this somewhat dated sound. "If you have a 4-band parametric EQ," he relates, "you start by rolling off everything below 60 Hz. This will eliminate rumble. Moving up one band, boost between 78 and 84 Hz by 3 to 6 dB, with about a 1.0 Q factor. Again, you may need more or less, depending on your track. This will provide the kick that hits you right in the chest. Next, dial in the punchiness between 1.5 and 2.5 kHz, boosting by about 6 dB. Here, a bandwidth of 1.5 to 2.5 should work nicely. Finally, notch out 120 Hz by about 4 dB with a 1.0 Q setting. Play with the parameters until your kick drum sounds like a White Lion record. If you have only a midsweep EQ with high and low shelving bands, try boosting the mids around 2 kHz by about 6 dB, turn the high shelf up between 4 and 6 dB, and boost the low shelf by about 2 dB."

Although this sound was popular in the 1980s, today we've returned to our roots, and engineers usually strive for a classic rock kick-drum sound, à la John Bonham. "The Bonham kick drum is the quintessential rock drum sound," Martin explains. "I usually obtain it by boosting the frequencies between 120 and 240 Hz by about 4 dB or more. You'll also need to roll off everything above 1.5 kHz. Sometimes, depending on the drum, you also might want to notch out 80 Hz a bit-not too much, just by 1 or 2 dB. Then add a little bit of 60 Hz, but again, just by about 2 or 3 dB."

A lot of alternative records have a hollower, more gritty-sounding kick drum. You can dial in this sound by rolling off everything below 100 Hz yes, you heard me right. Boost 125 Hz by about 3 dB, and add about 4 dB between 250 and 350 Hz. Then roll off everything above 2 kHz.

2. SNARE DRUM

Two of the most widely used snare-drum sounds in popular music are a tight, punchy snare and a loose, full-sounding snare (usually used on ballads). Naturally, the way the drum is tuned will influence how you decide to EQ it.

A snare never needs frequencies below 150 Hz, so roll them off. The center frequency of a snare is usually around 1 kHz, give or take a few hundred hertz, so it usually benefits any snare to boost around that point by approximately 3 to 6 dB.

For a tight-sounding drum, Martin explains, "you'll want to work with the upper mids, around 5 kHz, and some higher frequencies, like at 8 or 9 kHz. Try boosting in each area, starting with 3 dB and working your way up. Also, you'll need to sweep the frequency parameters until you nail it. Attenuating low frequencies is a good idea, so get



FIG. 4: The EQ for an overhead pair of drum mics, displayed in Ensoniq's Paris hard-disk recorder. The only other drum mics were on the kick drum and snare. For each EQ band, the top parameter is gain, the middle parameter is frequency, the lower parameter is bandwidth, and the bottom icon indicates the filter type (bandpass or shelf). Note that the Paris defaults to a bandwidth of 1.5 for shelving filters.

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rid of anything below 250 Hz, and if you still have an EQ band left, roll off everything above 11 kHz. A tight gate usually sounds good on this type of snare."

"A ballad snare," Petricelli notes, "needs a bit more bottom end, so try boosting around 250 Hz to give it a fuller tone. I'd start with an increase of 6 dB. You won't need to boost as much high end as you would with a tight snare—though adding a little around 7 kHz might be helpful, and you can roll off everything above that. The key element here is your center frequency: you don't want too much ring to the drum, but you want it to have *some* ring. Find the resonant frequency of the snare by turning the gain all the way up and sweeping between tween 10 and 14 Hz by about 3 dB."

These days, though, a lot of engineers record drum tracks with only four mics: a kick, a snare, and two overheads. This gives the drums a more natural, almost raw sound, but it makes equalizing them much more difficult. In this case, you need to keep the overheads' low end in place to sufficiently capture the toms, but not so much as to add boominess to the kick or snare drums.

Let's look at an example from a project I recently completed. The drum overheads were recorded pretty flatnot bad, but not all that exciting. After playing around with the console EQ, it became clear that we needed some outside assistance, so I patched in a CLM Dynamics Expounder. To liven up these two tracks, I started by adding about 10 dB at 150 Hz to bring out the toms, which were overshadowed by the cymbals. I then notched out an offending ring at 250 Hz produced by the middle tom. In addition, because the snare had a prominent ring at 1 kHz and we already had a dedicated



800 Hz and 2 kHz. When you find it, narrow in on it with the Q control. Now you can adjust the gain accordingly. A loose gate and heavy reverb complement this type of snare sound."

3. DRUM KIT

Drum overheads can be a bit tricky, and how you EQ them depends on how the entire drum kit was miked. "If you have individual mics on the tom-toms, snare, and kick drum," says Martin, "you only need the overheads to capture the cymbals and to blend the kit. In this case, you could probably just use a low shelving filter to reduce the amplitude of everything below 4 kHz. This way, you have individual control over the snare. toms, and kick, and the overheads take care of the rest. Depending on what mics were used, you might have to boost some high frequencies to get a decent cymbal blend-probably besnare track, I pulled out 1 kHz by about 6 dB with a 1.0 bandwidth setting, which also helped to round out the toms. Finally, although we had plenty of cymbals in our mix, I added about 3 dB at 9.5 kHz, just to smooth things over.

A similar example can be seen in Figure 4. Here everything above 1.5 kHz sounded great, but again the toms were being washed out. In this instance, I used the console EQ, boosting the low shelf by 8 dB at 191 Hz to add some low end. This also augmented the kick drum. I boosted 3 dB at 152 Hz, and I notched out a tom ring at 666 Hz and a snare ring at 1,017 Hz.

4. HAND PERCUSSION

Percussion comes in many forms, but here we'll focus on the more common instruments, such as shakers, tambourines, congas, and hand claps. Shakers and tambourines fall into roughly the same category with regard to EQ: both need to be bright and should cut through the mix on the high-end side. Martin discusses his philosophy on hand percussion: "For shakers, I usually just roll off everything below 2 kHz and add a bit of high end, like 6 dB at 9 kHz. That usually works fine. Tambourine requires a bit more 'clanginess,' so for it I roll off below 800 Hz, boost 1.5 or 2 kHz by about 4 dB, and add a little around 7 kHz."

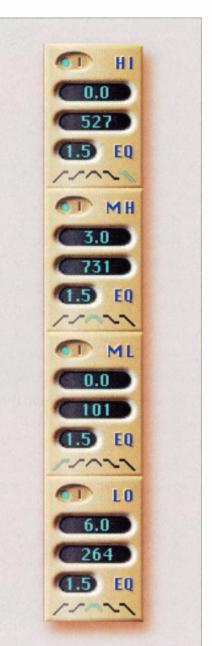
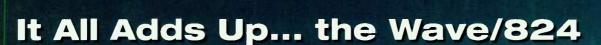


FIG. 5: A great setting for a clear and full-bodied electric bass guitar sound, this EQ can be modified slightly to produce a funkier, punchier sound.



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Petricelli often records congas for his commercial work. "If recorded properly," he notes, "congas don't really need that much tweaking with EQ. I usually find the resonant frequency of the



FIG. 6: This shows how I equalized an acoustic guitar for incorporation into a large mix. Notice the abundance of high-end content. drum by doing an EQ sweep, and either add or subtract a little depending on the situation. A boost generally helps to bring out the natural timbre of the drum. But you need to be careful not to add too much, especially if the resonant frequency is low, as it will clash with the other drums and the bass. To bring out the attack of the drum, I usually boost a bit in the midrange—say, around 5 kHz, maybe by 6 dB. You can roll off everything above and below the range of the drum."

Finally, hand claps are generally fattened up by adding some low mids, usually around 250 Hz by about 2 dB, with a 1.5 Q factor. Also, to get more of an attack, boost some of the midrange (around 1.5 kHz by about 4 dB) and add some highs (around 8 kHz by 2 or 3 dB should suffice).

5. ACOUSTIC PIANO

Piano EQ can be approached in two ways, depending on how much accompaniment the piano will have. Will it be the main instrument in the mix, perhaps with a vocal and bass? Or will it be in the back of a full-band mix with seven or eight other instruments?

Martin explains, "If you have a solo piano, with little or no accompaniment, and it's been recorded properly, you really shouldn't have to do much to it. Often, I like to boost a little low end, around 140 Hz, but only if there's no other bass instrument in the mix. Also, try adding a bit of high end, around 8.5 kHz—but not too much, just about 3 dB.

"If, however, you have to fit the instrument into a tightly packed mix," he continues, "you'll need to do some subtractive EQ. A piano naturally has a 'honky' sound because much of the playing is done in the middle of the keyboard. This is what you want to watch out for. Yes, you want the midrange to be out in the mix, but you don't want it to sound obnoxious, so you may have to do some attenuating around 3 or 4 kHz. Also try rolling off the frequencies below 140 Hz, because they will definitely clash with your bass guitar and kick drum. A slight boost around 8 kHz can bring out the upper keys. All in all, how much you boost or how much you cut will depend on the nature of the mix and its components. Try to coax the midrange and upper mids from the piano, but walk a fine line so that it doesn't sound honky."

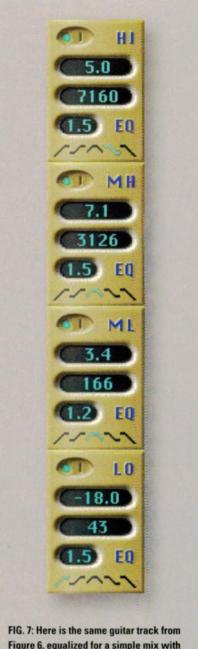


Figure 6, equalized for a simple mix with only a vocal and some percussion.

6. BASS GUITAR

With electric bass guitar, you have a lot of options, especially if the track was recorded with a direct injection (DI) box. One of my favorite bass sounds is illustrated in **Figure 5**. This track was recorded with a DI box, and we wanted a sound that brought out each note being played, without "slappiness." We started by rolling off everything above 520 Hz and below 100 Hz. We used a boost of 6 dB at 260 Hz to fatten up the tone, and a boost of 3 dB at 730 Hz to add some finger noise. This EQ, along



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For a player who slaps and punches and needs a funkier sound, you could use the same EQ settings, but instead of rolling everything off at 520 Hz, boost some mids around 2 kHz by about 4 to 6 dB. This provides the slap. You might want to decrease the lowend rolloff to about 50 Hz to give the track more rumble.

Often a graphic EQ can be used to zero in on a bass sound. "I usually use a graphic EQ on bass tracks," says Petricelli. "If the track has been done direct, this is almost a must, because you don't have the liberty of equalizing the player's amp, moving the mic, and so on. I use a 20-band EQ that can dial in just about any bass sound I want. The nice thing is that with a graphic EQ, you can see the curve you're drawing by looking at the unit. The tips for getting a sound are the same as with a parametric EQ; you just have more frequency options."

7. ELECTRIC GUITAR

As with the piano, how you treat the electric guitar depends on the other elements of the mix. For example, if you have only one electric guitar in a mix with just drums and bass, you can make the track sound *large*. However, if you have three other electric guitars, a piano, keyboards, percussion—the works—then you have to fit it nicely into a spot on the soundstage. Let's look at a couple of such scenarios.

Let's say that you can do whatever you want to this guitar to make it sound as big as possible, as long as it doesn't conflict with the bass guitar. Martin notes that "if I'm working with a simple rock band that has only one guitar player—you know, The Who *Live at Leeds*—I'll try to fatten up the low end as much as possible. Usually, a boost of 3 dB around 160 Hz is a good start, as long as it sits well with your bass. Also, try adding a little around 700 or 800 Hz. Depending on the sound of the amp, you might need to pull out some midrange; if you do, it will generally be around 3 kHz. How much high end you add depends on the sound you're going for: a boost of about 6 dB at 7 kHz will give you a crunchier sound." Again, it's always good practice to roll off anything above or below the frequencies that are not needed; but listen carefully before you cut, lest you accidentally eliminate desirable, but subtle, low-level subharmonics and high harmonics.

For our second scenario, let's assume you have to fit this guitar into a mix with a full band, including two other electric guitars. My advice is to focus on the midrange. Roll off everything below 200 Hz and above 9 kHz. Then do some sweeps in the middle to see what needs to go in and what needs to come out. I find that a boost around 4 kHz and a cut around 6 kHz generally works, although sometimes the opposite is true. You have to listen to the track and decide according to the mix at hand. If you have more than one electric guitar, make sure that they sound a little different from each other. This could mean using different guitars, amps, mics, miking techniques, or processing. There's nothing more grating than three guitars playing the same thing, all with the same sound.

8. ACOUSTIC GUITAR

To illustrate how dramatically the EQ of an acoustic guitar can change from one mix to another, let's look at Figures 6 and 7. Both EQs were applied to the same track, but in different mixes.

Figure 6 shows an acoustic guitar track that has been worked into a mix with drums, bass, piano, electric guitars, percussion, and lots of vocals. This track was recorded with a very nice Martin acoustic guitar miked with an Audio-Technica AT 4033. Here, I rolled off everything below 90 Hz and added a bit of body around 360 Hz. I really boosted the middle and upper frequencies, though: 10 dB at 2 kHz and 9 dB at 7.1 kHz. With everything else happening in this mix, the acoustic guitar had to sound bright—and it did.

Now look at **Figure 7**, which shows the same guitar track, only within an alternate mix of acoustic guitar, vocals, and percussion. This time, I decreased the low-end rolloff to 43 Hz to give the guitar a fuller sound, and I made a 3.4 dB boost at 166 Hz to achieve the same goal. The midrange boost

has been moved to 3.1 kHz and is at only 7 dB, and instead of increasing 7.1 kHz with a 9 dB bell curve, I've used a smoother shelf to gradually raise the high end.

Martin notes that strings or string patches follow a similar approach. "A string section has many of the same tonal characteristics as an acoustic guitar," he explains. "I usually roll off most of the low-end content, assuming that the strings will be placed in a full mix. I then boost the upper mids a bit around 4 dB at 7 kHz, and maybe give the sample some 'airiness' by boosting with a shelving EQ around 10 kHz."

9. BRASS AND WINDS

Some horns are naturally strong in the midrange, so bringing out those particular frequencies is important. For instruments such as the trumpet, roll off the low end completely, around



FIG. 8: The EQ of two vocal tracks for the same song. The EQ on the left is processing a male's voice, while the one on the right is working on a female singer's vocal.

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200 Hz or even higher. Depending on the recording or sample, you might have to tame some midrange to avoid a honky sound.

Conversely, with brass instruments that rely on the low end, you might want to roll off the upper frequencies—say, above 9 kHz. Usually a boost in the low mids helps bring out these instruments in a mix, probably around 1.5 kHz. When working with tuba or with tuba samples, be sure to use a filter to roll off frequencies below 40 Hz; this way you ensure that no speakers get blown, especially during low notes.

Most wind instruments need to be airy, which commonly involves frequencies above 9 kHz. Usually a shelving EQ works well to augment this characteristic. A bassoon can produce extremely low notes, so don't filter out its low frequencies; but for many wind instruments you should use a low-frequency filter.

10. LEAD VOCALS

A vocal track must have body, presence, and air, but not so much as to interfere with the rest of the mix. Vocals also need to be out front in the mix, and this is usually accomplished by boosting the midrange. Petricelli elaborates: "Vocals are what sell the product, whether it's a song or a sung commercial, so they need to be audible but not annoying. It's a fine line to walk." Understand that vocal tracks are the most sensitive you will have to EQ, and because so many variables are involved, making categorical statements about vocal EQ is difficult. You'll have to listen and decide for yourself.

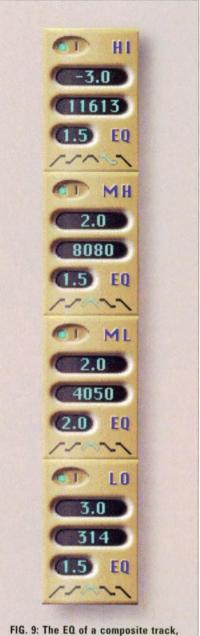
It's important to note, however, that there is a big difference between recording male and female voices. "I just finished a session in which we had two singers, a man and a woman, who were doing a duet," Martin explains. "They were recorded in the same room, with the same mic, roughly ten minutes apart. When we went to mix, the tonal differences were amazing."

Based on what Martin told me, I recreated the two EQ settings shown in Figure 8. The tracks were recorded with a Neumann U67 mic and run through a UREI LA-2A limiter. This illustration should give you an idea of how vocals are typically equalized. The male singer's track appears on the left EQ, and the female's track is on the right.

Immediately, you'll notice that the male vocal gets an upper-frequency boost of 1 dB with a shelving EQ, while the female vocal requires a 3 dB shelving cut at 8.8 kHz. The male vocal also needs a 2 dB boost at 7.5 kHz and a 5 dB cut at 5.1 kHz. Martin told me that this was because the singer had a cold and sounded a bit nasal. Conversely, the female singer's track was treated with two 4 dB low-end boosts-one at 733 Hz and another at 283 Hz-to make its low-end content complement that of the male singer's track. Interestingly, both vocal tracks got a similar boost around 2.5 kHz, illustrating how important it is to bring out the midrange in a vocal track.

11. BACKGROUND VOCALS

"There are two kinds of background vocals," Martin points out. "The first is your standard harmony that sits an octave above the lead and comes in on the chorus. This is almost like another lead vocal, and I generally EQ it using the same approach. The other type of background vocal is the choral effect, in which you may have three or four backgrounds, at all varying octaves, panned across the soundstage. For this I use



consisting of lead guitar, vocals, and hand claps.

a different approach: these vocals typically sound better when they have a bit of 'air' to them and sound almost unearthly.

"To get this effect, I usually roll off low frequencies on high harmonies, usually around 400 Hz, and attenuate extremely low frequencies on low harmonies, around 100 Hz. Then I take out as much midrange as I can without losing the clarity of the voice—try filtering out anywhere between 1 and 4 kHz. Finally, I add a lot of upper mids and high frequencies. I usually put a shelving

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EQUATION

EQ on around 8 or 9 kHz, sometimes higher, and boost it until I have that angelic effect."

12. NARRATION

Obviously, if you're working with a narrative that doesn't need to blend into a music mix, you can fatten up the sound of the announcer's voice. Petricelli commonly does this.

"In this case," he explains, "you can leave a lot more low end in the voice than you could if you were doing a music mix. Announcers are supposed to have big, godlike voices, so you need to make them sound that way. Start by boosting a bit between 60 and 120 Hz, depending on the tonality of the voice you're working with. Then go all the way up top and add some high end, probably around 7 kHz. You might need to pull the mids back a little, but make sure you don't lose any clarity. It's somewhat similar to the way you'd set your home stereo."

13. REMOVING ARTIFACTS

As mentioned earlier, filters and EQs are great tools for removing sonic garbage. The most obvious example of this is using a filter to notch out a 60 Hz AC hum. However, you can use a notch filter to get rid of almost anything, and you can use a parametric EQ to create a notch filter. Martin relates a story.

"We were doing a session in which we had a percussionist playing the congas" he says. "When we listened to the playback-months after he had originally cut the track—we noticed an odd clicking sound that came up every so often. We couldn't figure out what it was. It turned out that the guy was wearing a ring on his right index finger that occasionally hit the lug on the side of the drum. We couldn't redo the track, so I thought I'd try to EQ it out with a parametric. By turning the gain all the way down and gradually sweeping the frequency knob and zeroing in with the Q control, I was able to isolate not one, but three different frequencies that were causing this

click. We eliminated the noise without affecting the sonic characteristic of the drum."

14. ONE EQ FOR TWO

Often, especially in the world of digital audio workstations, sound designers wind up with more than one instrument on a particular track. If your automation package allows you to automate EQ changes dynamically, there's no problem: you simply switch from one EQ setting to another at the appropriate time in the mix. If your software doesn't offer this feature, however, you'll have to get creative, or at least reach a compromise.

Let's look at another example from a project I recently did (see Fig. 9). One track contains a lead guitar, background vocal, and hand claps. The vocal and the hand claps were recorded using the same mic (an AT 4033) and had a similar sonic content-but, as you can imagine, the guitar sounded quite different. So we needed to make some compromises. The first thing I realized was that none of the instruments needed anything above 15 kHz, so I gradually rolled it off at 11 kHz. I then boosted the midrange a little at 4 kHz to bring out the vocal; that, unfortunately, also made the guitar sound clangy. So to compensate, I added some low end for the guitar, around 300 Hz, which didn't really affect the vocal in a bad way. I then boosted about 2 dB at 8 kHz for some clarity to the guitar and vocal. Lo and behold, with all these changes, the hand claps sounded fine!

The moral of the story is that if you have to comp tracks together, make sure the instruments on them have similar frequency responses.

TWEAK IT, BABY!

An EQ can either ruin a recording or make it shine; it's a very powerful tool. That power is harnessed by the ear, experience, and knowledge of the person operating it. The first two things can't be taught in an article or a classroom, but the tips offered here should be helpful to anyone, whether you're a novice or a pro. These solid approaches to EQ have been used for many years by seasoned engineers. But that's not to say they're set in stone; like anything else in this business, creativity is the key.

EM associate editor **Jeff Casey** has a 10-band fully parametric EQ on his car stereo.

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and editing a recording project is analogous to assembling an art exhibit. Every piece may be great on its own; however, the pieces will have a much greater impact as a collection once they are cleaned up, properly framed, and displayed in a way that shows each in the best possible light. This kind of attention to detail makes the whole artistic experience greater than the sum of its parts.

sterina

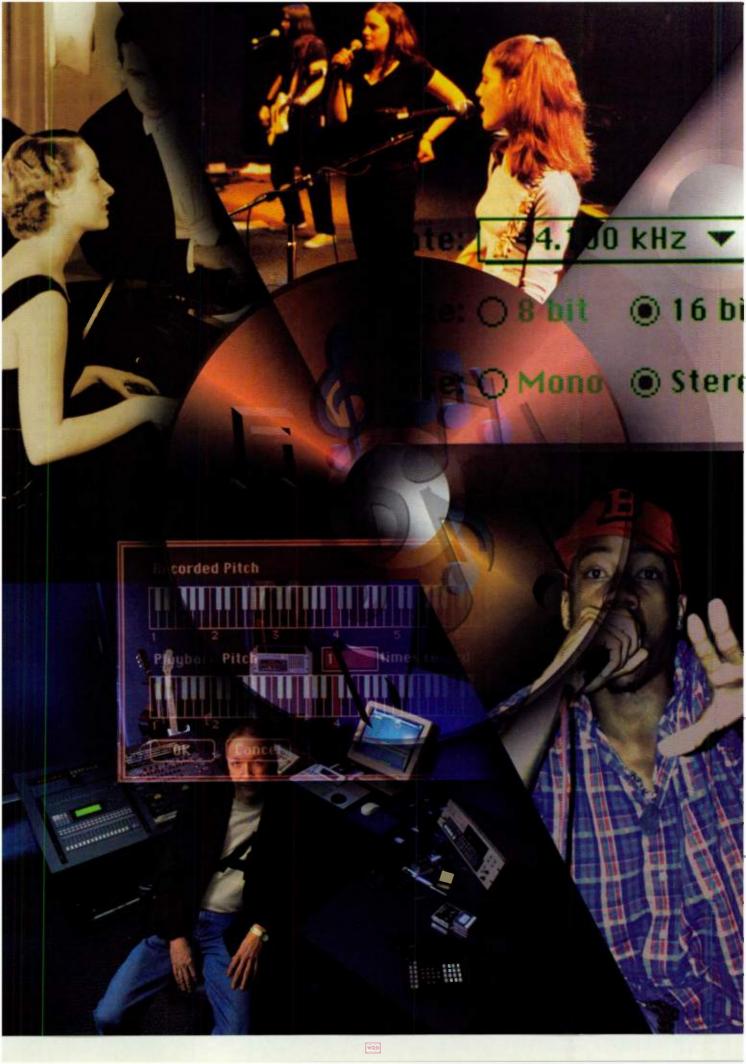
The goal of mastering is to make the music sound good on a variety of systems and delivery media and to unify the program from beginning to end through careful quality control. Mastering a project from one studio and one set of sessions is straightforward because the masters will be on one format and the sound will be consistent from one piece to the next. Assembling compilations, on the other hand, makes interesting demands on the mastering engineer.

Compilations run the gamut, from tribute albums to holiday records to ethnic music collections, and they range in size from a single CD to multiple-disc box sets (see Fig. 1). Because each song usually comes from a different artist and a different set of recording circumstances (including location and studio recordings), it is likely that the engineer will receive a wide variety of masters on various analog and digital media.

The mastering engineer's challenge is to achieve a consistent, unified sound throughout the compilation. This is the primary focus as the engineer masters each piece and makes improvements to the mixes by using EQ and compression, changing the stereo image, and adding reverb.

PREPARATION

As with any phase of studio recording, the amount of preparation you do can make or break a session. This is especially true with multiple-source mastering jobs. Media compatibility is a common problem, and dealing with it can be particularly time-consuming. I have received masters on almost every imaginable format, from old reel-to-reels and cassettes to assorted digital media, and I've learned the hard way that it is essential to know ahead of time the format of each mix I will receive. That way I can have the proper machines on hand.



MASTERING CONTINUITY

comparison to make sure that you aren't dulling the high frequencies or losing important detail and definition.

Broadband EQ boosting is another effective mastering tool. When applied skillfully, it can enhance the fullness and overall impact of a recording. Typical scenarios for this type of EQ processing include high-shelving boosts above 10 or 11 kHz to add shimmer and upper harmonics; low-shelving boosts below 80 to 100 Hz to enhance the power of the low drums and bass fundamentals; upper-bass boosts between 200 and 400 Hz for more punch and roundness (especially on digital multitrack recordings); and upper mid or treble boosts between 1 and 8 kHz to increase overall brightness and to aid the projection of vocals, guitars, drums, and solo instruments. EQ increases in the 400 Hz to 1 kHz range are rarely needed due to the fact that most microphones and monitors exhibit fairly even response in this region. However, increases in this range, when used cautiously, can bring out chordal instruments, warm up vocals,

and make thin-sounding mixes bigger.

You may choose to do corrective EQ in the computer rather than during the transfer. Certainly the precision, affordability, and high quality of many equalization plug-ins (not to mention the allure of multiple undo levels) make this an attractive option. If you plan to EQ in the computer, keep your input levels a little lower during the transfer to allow for potential signal boosting and the subsequent increase in gain it may produce. Any overall gain changes, nor-

malization, or compression should be done after all EQ decisions have been made. If you need to equalize a normalized region, avoid digital overloads by dropping the overall gain of the track by 2 or 3 dB before adding EQ.

FURTHER ENHANCEMENTS

During the transfer stage there are additional enhancement options, plus a few critical quality-control checks, that you need to make whenever you work on master tapes from various sources.

Occasionally I add digital reverb to an entire mix, either at the client's request or on my own. I generally choose

a short room or ambience setting with a considerable low-end cut and a decay time of one second or less. Then I listen closely to how the added reverb affects the drums, bass, vocals, and lead instruments. A small amount of reverb (7 to 15 percent) is usually sufficient to lend some extra air, sustain, and spaciousness to the mix without making it sound artificial.

Some mastering engineers apply compression during the transfer stage. Since most mixes I receive have some degree of compression or limiting already, I usually address compression later in the mastering process. But when it is warranted, as on live recordings and un-

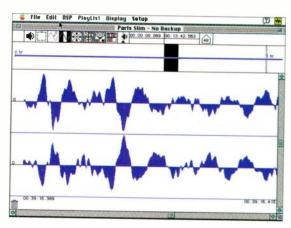


FIG. 3: Tracks that are 180 degrees out of phase are easy to spot. This problem can be fixed with the Invert function in the editing program, or by using a phase-reversing connector on the out-of-phase channel during the transfer to digital.

compressed mixes with infrequent amplitude peaks, I use a high-quality compressor in the analog domain or employ the digital compressor in the M2000.

Most semiprofessional, analog stereo compressors add significant coloration to the signal chain, which is one reason I don't recommend them for mastering. If your compressor has a hard-wired bypass, click it in and out to see how the circuitry affects a stereo mix before you use it on a mastering session. For units without a true bypass function, feed a signal into your computer with and without the compressor in the chain and compare the differences.

MAXIMIZING LEVELS

Whether or not you decide to compress at this stage, level control is an essential part of the transferring process. If you have analog sources or analog processing, be sure to maintain proper gain staging throughout the signal chain to avoid adding noise with signals that are too low or introducing clipping distortion from signals that are too hot.

For digital transfers, level control is generally not available on DAT machines and CD players. Digital multieffects units provide level control for digital transfers, ranging from +6 to -16 dB, with 16-bit dithering on the output. Note that every time you chain digital units together in this fashion, it's vital that the digital source (DAT, CD, and so on) is functioning as the master clock for all downstream processors (including the computer) to avoid jitter or resampling errors.



For the compilation *Wavelength Infinity: A Sun Ra Tribute*, engineers Ed Herrmann and Myles Boisen received master tapes on DAT, analog cassette, and CD-R. The challenge was to give the music and spoken-word pieces uniformity over the course of two CDs.

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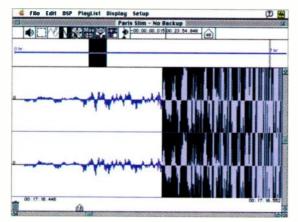
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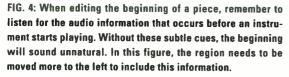
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MASTERING CONTINUITY

Make sure that each piece is transferred into the computer as hot as possible so that amplitude peak values are within 1 or 2 dB of the maximum digital ceiling. It's also important to check the left/right balance, particularly on analog sources, to ascertain that the channel gains are matched and that any center-imaged information (usually the kick drum, snare drum, lead vocals, and solos on studio recordings) actually appears midway between the speakers. Don't rely solely on the level meters; they are not always calibrated perfectly. Use your ears, and monitor constantly for distortion or imbalance in the stereo signal before it goes into the computer.

Vague center imaging may indicate out-of-phase information in the stereo spectrum, ranging from slight phase cancellation to complete 180-degree phase reversal of a single channel. The easiest way to test for this is to sum the stereo pair to mono by panning the left and right channels to the center. If some instruments or frequency ranges change timbre or disappear completely, there is a phasecoherency problem with the track (see **Fig. 2**). Fixing it is not as easy as finding it, and unless one side is truly 180 degrees out of phase, this is a job





for a professional. A track that is completely out of phase can be fixed in the analog domain by using a phase-reversing, balanced XLR or TRS connector during the transfer to digital.

A pair of tracks that are 180 degrees out of phase can also be spotted visually and remedied in computer software (see Fig. 3); in this case the left and right channels will appear as mirror images, and the side that displays its transients as a negative

voltage can be corrected using the DSP Invert function.

DROPOUTS

It is also important that you be able to recognize digital dropouts and problems that can occur because of high block-error rates (BLERs) during the transfer to the computer. You may already be familiar with the unmistakably "fritzy" sound of DAT dropouts, which produce brief periods of unpleasant fuzz or stuttering silence within a mix.

High error rates also can cause minor dropouts and slight "CD skipping" artifacts or subtle graininess that is most audible in the highs. Although BLERs can be monitored on some

> DAT machines, there is no way to fix dropout problems short of editing around damaged sections, mastering from a safety copy or an alternate mix, or playing the tape in different machines. (Occasionally, DATs recorded in one type of equipment will have compatibility problems when played in other machines.)

DIGITAL EDITING

Once everything is loaded into the computer, the next step is to edit the tracks and assemble them in the proper order. When editing the beginning of a

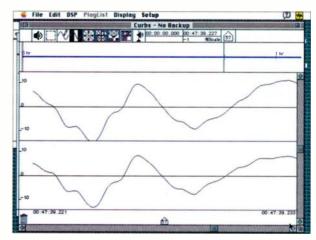


FIG. 5: To reduce the potential for clicks, pops, and waveform distortions, always begin and end your regions on zero crossings—the point where the waveform crosses the horizontal "zero" line. You can view the exact zero crossing by zooming into high magnification of the waveform.

piece, start the region as close as possible to the first significant sound, without trimming it so tightly that an important aspect of the sound is removed. The start of the piece may sound artificial without a vocalist's breath, a guitarist's initial pick stroke, or a drummer's hi-hat tick. Begin your region a little to the left of such sounds (see Fig. 4).

Likewise, be careful not to cut off any significant room sounds or reverb at the tail of the piece. If you're drawing a fade-out over the end, try a longer fade region first, then shorten it by small increments if necessary. It's easy to make a fade-out that sounds too abrupt, so err on the conservative side, and check your fades on speakers and headphones. Sources with significant background noise or tape hiss may require longer fade-outs. Quick fade-ins of 500 milliseconds or less will reduce the shock of a noisy beginning coming out of digital silence.

For tops, tails, edits, and DSP functions, always start and end your regions on zero crossings, where the waveform crosses the horizontal "zero" line on both the left and right channel displays (see Fig. 5). This will greatly reduce those annoying little clicks, pops, and waveform distortions that plague even the best editors from time to time. If you're using a playlist-based editing program with crossfade capabilities, learn what the various crossfade options are—and don't be afraid to use a crossfade of 10 to 100 ms to get rid of a clicky edit.



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MASTERING

ON THE SPOT

You can use EQ and gain "spot" changes-whether on an intro, a single note or phrase, or an entire section of a piece-to correct mixing or performance flaws, remove clicks and hum, clean up quiet sections, and even smooth over a troublesome edit. There is often a lot of this detail work involved in cleaning up semiprofessional and live recordings; how much you want to do depends on your time and workload, processing power, skill, and, most of all, patience. At this stage, you should also do the DSP Invert phase correction mentioned earlier, along with any other specialized operations such as pitch shifting, reverb, and time compression/expansion.

Before moving on to the final mastering stages, you should have each of the songs trimmed, edited, tweaked, and sounding the way you want, at a reasonable gain level. Note that it is not necessary to normalize every piece prior to the final consistency check; in fact, doing so may be a duplication of effort, especially if you're planning to use mastering-compression software.

ASSEMBLY

The final sequencing of the project is done by selecting the individual items from a menu and placing them in their proper order. For pop CDs, I start by inserting two to three seconds of silence between pieces. Punk and alternative bands tend to favor shorter gaps than this, while jazz discs and projects with lengthy pieces often work better with four to eight seconds of silence between cuts.

Audition these silences and judge the impact of the gaps between the pieces. This will also give you the opportunity to double-check the tops and tails on each piece and get a sense of their relative levels and EQ. To accurately gauge how much of a gap is needed between pieces, it's important to audition at least the final 45 seconds of each track and listen through the gap into the beginning of the next cut. This may seem like a long time, but a shorter listening period will often fool you into thinking that a smaller gap is needed, resulting in a program that seems to rush the listener along from one selection to the next.

One more crucial task awaits you: the final consistency check, where your collection of masterpieces is touched up and finally put on display. If you've done your EQ correctly, each piece will sound full on its own and compare favorably with every other cut on the CD. The disc as a whole should match the tonal characteristics and bass-to-treble balance of other CDs in that genre.

Occasionally, I add digital reverb to an entire mix.

Chance are, however, that one or more pieces won't make the grade, even with their gain bumped up to match their neighbors'. As you listen, make detailed notes of these deficiencies before returning to your editing program to make corrections. If you're feeling particularly ambitious, you can return to the source material, access your detailed notes, and transfer the piece again with an improved EQ curve.

MASTERING COMPRESSION

Most mixes have a few volume peaks (usually on vocals, solos, or drums) that are at least 2 to 3 dB above the average

level. Normalization brings these occasional peaks up to 0 dB on the digital scale, which should make a piece loud enough for your purposes. But if normalizing doesn't do the job, only mastering compression can effectively tame these peaks while bringing the overall level up by 2 to 3 dB.

For continuity of levels, it's important to compare the beginning, middle, and ending of every cut to determine an average volume or *core level*. During this process, a single track (or group of tracks) with a hotter-than-average core level serves as a benchmark for the CD as a whole.

Load the tracks, in sequence, into a CD-R burning application (I use Digidesign's *MasterList CD*) to determine the amount of processing you will use in the mastering compression program. Raise the gain of the quieter pieces in 1 dB increments until the core level of all pieces is consistent. Be prepared to lower the gain on pieces that seem unnaturally loud in the context of the compilation; these are often mixes without drums, or quiet and drony pieces whose lack of dynamic range allowed them to be transferred into the computer at a core level close to 0 dB.

These gain changes in the CD-R program are not final; they are merely a handy mock-up. By using masteringcompression software to implement the gain boosts (typically 1 to 4 dB on normalized material) rather than boosting in the CD-burning program, I avoid getting digital overloads when I burn my CD-R. Return to the digital editor to use the mastering compressor and make any last-minute changes based on your findings during the continuity check.

READY TO BURN

At this point there is one more important decision to make regarding the overall compression level for your CD. If the level of the "hot" pieces is satisfactory, you can just implement the calculated gain changes for the quieter pieces using your mastering compressor, and then you're ready to burn your CD master.

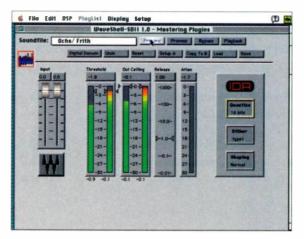


FIG. 6: Once you have determined the gain changes you want, you can use a mastering compressor such as Waves' *L1 Ultramaximizer* to implement gain boosts without causing digital overs when you burn a CD.



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MASTERING CONTINUITY

But if the core-level continuity you've achieved for this project is still noticeably below that of comparable CDs, you may decide to raise the gain of all the hottest pieces by a set amount (for example, 3 dB) to bring it up to a more competitive level. If you choose to go this route, it means that you have to add 3 dB of gain to all of the changes calculated in the playlist mock-up, as well. The resulting cumulative boost of 4 to 6 dB that some of these pieces will now receive should prompt you to check again for overcompression before going back to the CD-burning software.

In most instances, I use a mastering compressor (I prefer Waves' *L1 Ultramaximizer*) on all of the pieces (see **Fig. 6**). Even when pieces are at unity gain and

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It may not be the most exciting reading, but it's worth your while to read up on the technical uses of dithering and explore the options that your software or converter provides.

FINAL PASS

Now sit back and listen through the entire project. If you're exhausted (as I often am at the end of a complex multisource job), burn a reference copy on CD or DAT and check your work after you've recuperated.

If you're dealing with a client or outside producer, make sure he or she gets a reference copy (clearly marked as "not for production"), and be prepared to back up the project or leave it on your hard drive for a few days until you get approval from all interested parties. When everyone's happy with the final version, burn a master CD to send to the replicator, and make a duplicate disc in case the master gets lost or damaged. If there is any doubt in your mind that you haven't heard the last of this project, do a complete data backup; save all playlist and editing information, if possible.

Remember that mastering is a highly skilled vocation, and there are no software shortcuts or hardware substitutes for the finishing touches that an experienced mastering engineer can add to a CD compilation. But there's also no harm in practicing the tricks of the trade in your own studio. Take your time, listen closely to every change you make, and learn all you can from your mistakes. The time and effort you spend will pay off every time your finished project is heard, whether it's on a boom box or over the ritziest audiophile stereo system.

Myles Boisen is a guitarist, producer, composer, and head engineer/instructor at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California. He can be reached via e-mail at mylesboise@aol.com.



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n the audio world, "professional" studio gear generally features line-level audio inputs and outputs on balanced, three-conductor XLR connectors, operating at a signal level of +4 dBu. In contrast, "consumer" gear uses unbalanced, two-conductor RCA or "-inch connectors, and it sends and receives audio signals at -10 dBV. "Semipro" gear (including most synthesizers and low-cost signal processors) adheres to the consumer standard, except that unbalanced, "-inch connectors are more prevalent than RCAs. (For more on the various types of line-level signals, see "Square One: The Wizard of dBs" in the January 1996 **EM**.)

Mechanically and electrically, the +4 and -10 audio standards bear little resemblance to _____ each other and rarely coexist without a fight. With some creative wiring, you can often get around some of the major gremlins that crop up when you mingle them in your studio, but curing one problem sometimes creates another. These problems can include hum or buzz, RF noise, impedance mismatches, phase cancellation, and overloaded or underdriven inputs. Such glitches often appear at the most inconvenient times. and they can cause you to pull your hair out in bafflement. You end up scrambling for an extra ground wire, a preamp to boost a signal, an attenuator to reduce another, or one more weirdly wired cable that makes little sense electrically, but hey, it clears up the problem-more or less.

Of course, the simple solution is not to mix different audio standards in the first place. But we must live with the facts that we need both types of gear, that interconnection differences exist, and that we need a reliable method of interfacing different types of equipment.

By Peter Mosher



CIRCUIT OPERATION

In order to go back and forth between the two audio standards, you need two separate circuits: one that translates an unbalanced -10 dBV signal into balanced +4 dBu, and one that does the same thing in the opposite direction. For this project, both circuits have been kept quite simple.

The top half of the schematic that is shown in **Figure 1** is the consumer-toprofessional level converter. An unbalanced signal enters the circuit at capacitors C1 and C2 and is passed to the inverting input of op amp U1A through resistor R1, which sets the input impedance of the circuit to 10 $k\Omega$. U1A amplifies the signal slightly and feeds it to the noninverting input of U2A. The signal that appears at the right side of R5 is a boosted and invert-

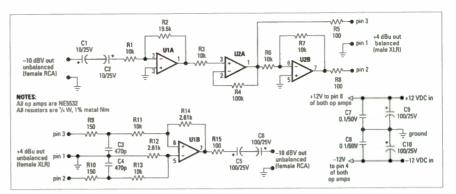


FIG. 1: The top portion of this schematic converts -10 dBV to +4 dBu, and the bottom portion converts +4 dBu to -10 dBV.

ed version of the original input signal.

U2B inverts the signal from U2A again and feeds it to the output through R8. As a result, the output appearing at pin 2 is an amplified, inphase version of the input. The signal at pin 3 is a phase-inverted version of this output. Together, they make up the balanced signal output.

The bottom half of the circuit is the pro-to-consumer converter. A balanced input signal is fed through an RF filter made up of R9, C3, R10, and C4. The signal then passes through resistors R11 and R13 to the inputs of U1B, which is wired in differential mode. As a result, this op amp passes the *difference* between the signals that appear at its inputs.

The strength of balanced circuitry is that any induced signal common to both signal wires is canceled out when it passes through the op amp. The engineering term for this is *common mode rejection ratio* (CMRR), which is a measurement of a differential circuit's ability to reject





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a smile on my face."

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PT 60



unwanted common-mode electromagnetic interference (that is, signals that are in phase in both signal wires). This brings us to the importance of using precision resistors in balanced circuitry.

You'll notice that all resistors specified in the circuits are 1 percent tolerance, metal-film type (see the sidebar "Parts List"). If you're used to building projects with lower-tolerance, carbon-film resistors, you might wonder why precision types are necessary. Strictly speaking, they're not, but if you don't use them at least for critical components (R1, R2, R11, R12, R13, and R14), you're going to have level problems and poor CMRR. The nature of balanced circuitry demands the use of tight-tolerance resistors; if you have access to a 4¹/₄-digit ohmmeter, matching the critical ones to within 0.01 percent

or better would not be going too far.

Both circuits were designed using the lowest possible number of parts to minimize signal coloration. In the unlikely event that you can't find the necessary parts at your local electronics shop, you can easily get them all by mail order. I've found that Digi-Key usually has everything I need in stock, and it delivers



The circuit itself is very simple.

promptly (tel. 800/344-4539 or 218/ 681-6674; Web www.digi-key.com).

BREADBOARD OR CIRCUIT BOARD?

If you prefer, you can breadboard the circuit, but if you're planning to make more than two converter channels, I recommend using a printed circuit board. PCBs are extremely reliable and rugged, and they are the most timeefficient way to go, even in small quantities. In addition, you'd really have to work hard to make wiring errors, and it's much easier to troubleshoot a welldesigned circuit board than to find a problem in a rat's nest of wires and parts sticking out at all angles.

If you're still not convinced, let me tell you about a dead-easy method of making PCBs at home. It's called the Toner Transfer System (TTS), which is made by DynaArt (tel. 813/524-1500; e-mail mail@dynaart.com; Web www .dvnaart.com). To do this, you'll need a blank circuit board (copper-clad on one



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side only), the TTS kit, a suitable etchant (ferric chloride or ammonium/sodium persulphate), a clothing iron, and a hobby drill with a tiny drill bit. You'll also need some way of printing the image; you could use a photocopier, but I prefer to print a scanned image of the circuit board on a laser printer.

The procedure is simple: just cut out the actual-size PCB template shown in **Figure 2** and scan it into your computer. Using a graphics program, copy and paste the template as many times as you need for the number of circuits you plan to build and line them up squarely. Print the templates onto a sheet of TTS paper using a laser printer. Cut the templates into pairs, side by side, so they measure 4 by 4 inches.

Clean the copper side of a 4-by-4-inch circuit board with a powder-type sink cleanser, rinse it, and dry it with a clean

PARTS LIST

Integrated Circuits	
U1, U2	.NE5532

Resistors (¼W, 1%, metal film)

R1, R3, R6, R7, R11, R1310 kΩ
R219.6 kΩ
R4100 kΩ
R5, R8, R15
R9, R10150Ω
R122.61 kΩ

Capacitors

C1, C2	.10 µF, 25V electrolytic
C3, C4	470 pF, 25V ceramic
C5, C6, C9, C10	100 µF, 25V electrolytic
C7, C8	0.1 µF, 50V ceramic

Connectors

XLR-F	female 3-pin XLR connector
XLR-M	male 3-pin XLR connector
RCA	female RCA connector

Miscellaneous

AD1	.±12V bipolar adapter
PC board	.homemade, see text
Case	Jameco or equivalent

cloth. Place the template face up on a hard, flat surface that you won't mind scorching, and orient the circuit board, copper side down, on top of the template. Heat the iron to the "cotton" setting, and place it on the back of the circuit board, applying light but steady and evenly distributed pressure for about three minutes. Lift the iron, let the board cool a little, and place it face up in a bath of room-temperature water. Let it sit there until the paper literally slips off on its own; don't "help" it off, or you'll risk damaging the imprint.

Rinse the board in room-temperature water, place it in the etchant, and leave it there until all the unwanted copper is gone. Using acetone or nailpolish remover, remove the printer toner from the copper traces. You're left with a nice pattern of copper lines that look exactly like the template. Drill the board and mount the parts on the opposite side of the board from the copper lines, following the placement diagram (see Fig. 3).

This might sound complicated, and it does take a few tries to learn, but it's easy once you've made a couple of

> PCBs. If you've ever made circuit boards using the old photographic method, you will especially welcome this new technique. Give it a try. Be very careful with the hot iron and circuit board, though, and use caution with the etchant, which is toxic and very corrosive. If these things make you nervous, get someone with experience to help you.

CONSTRUCTION HINTS

There are a few golden rules regarding the construction of the level converter that will save you a lot of time and frustration when you get to the point of actually turning the thing on. For example, there are many reasons not to build your own power supply these days. As you've probably noticed, I didn't even include a schematic for one in this project. Ready-made, regulated 12-volt bipolar supplies are widely available, which means that you don't have to mess around with potentially lethal line voltages. (I've used up

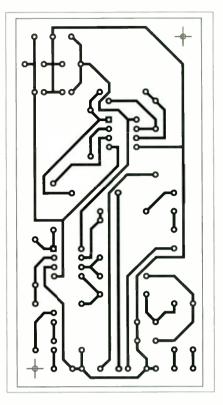


FIG. 2: Scan this full-size PCB layout into your computer, and use a TTS kit to make your own printed circuit board.

most of my nine lives on this one; don't tempt fate.)

These power supplies can often be found at your local thrift store for a couple bucks. Look for ones that were originally used for old home PCs, modems, and game computers; they not only have the voltages you need for analog projects, but they have a regulated 5-volt supply, as well, which you will need should you branch out into digital projects. Take me seriously on this; once you've salvaged a used commercial power supply, you'll never consider building another one from scratch for small projects.

Be careful to get the capacitor polarities correct, especially C9 and C10, which will turn into firecrackers if you wire them backward. Make sure the op amps are oriented correctly, too.

Never use bargain jacks; get the best ones you can afford. Gold-plated jacks are ideal if you can mate them with gold-plated connectors.

TESTING

This must be one of the all-time easiest projects to test, because there are no adjustments to be made. First, establish a reference point: put a test CD with



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internet

LISTEN TO WHAT



HAVE TO SAY



Igor Khoroshev, keyboardist, currently recording and touring with Yes, feels that the OX-7 Virtual Tone Wheel Organ Module is "an incredible machine. I strongly believe that will be the number one B-3 sounding machine. because I've never heard simulated organ sounds that were so great."



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a 1 kHz, 0 dB test tone into a consumergrade CD player. Using a recording device with -10 dBV inputs and a highresolution bar-graph level meter, adjust the CD player's output level and/or the recorder's input level so that the meter reads exactly 0 dB.

Next, connect the line outputs of the CD player to the -10 inputs on the converter, route the converter's +4 outputs directly to its +4 inputs, and connect the converter's -10 outputs into the

recorder's inputs. If you read 0 dB on both channels, everything is working as it should. If the levels are too high, too low, not there, or they sound like a broken analog synth, go back and check for connector-wiring errors, poor solder joints, or hairline cracks in the circuit board. Make sure you have properly

noise. You can build a 2-channel converter for each one and run a highlevel, balanced signal to the mixer inputs instead. This will increase noise immunity dramatically. You can use this trick with any -10 dBV equipment that's located any distance from the mixer.

Say you have a bunch of gear, both balanced and unbalanced, mounted in a rack. The patch bay includes modules that handle -10 and +4 signals, which means you're constantly faced with level problems. Building two channels of two-way level conversion and connecting them to the patch bay will let you deal with many of the problems you encounter on a regular basis.

For the ultimate luxury, build enough converters for all of your semipro and consumer equipment and

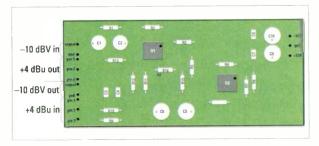


FIG. 3: The parts are mounted on the opposite side of the PCB from the copper traces.

identified all the resistor values, as well; the color coding on precision resistors can be tricky to decipher.

APPLICATIONS

In my line of work (forensic audio enhancement), you never know the format of a tape that someone is going to submit for examination. Most of the processing and playback equipment I use is -10 dBV unbalanced (cassette, microcassette, or VHS). When someone came to me one day with a Betacam tape and machine, I had a problem: +4 dBu balanced outputs. I had no way to quickly and properly connect this device to my setup. So I built a level converter with two channels each of -10 to +4 and +4 to -10 conversion, and it deals with most of my interconnection problems very nicely.

Of course, many other applications are possible, especially in a music studio. Suppose you have keyboards strewn around your studio, and the lowlevel, unbalanced cables running from their outputs to your mixer pick up

make *all* the signals coming to the patch bay balanced. Only one standard signal level and format will appear at all the jacks of the patch bay, and many of your interconnection hassles will disappear entirely. In most personal studios, however, it is more practical to convert a few pro-standard items to the consumer standard. Although this is not as elegant as running +4 dBu balanced lines exclusively, at least everything will be interfaced properly.

If you've ever run into any of the headaches I've mentioned, you owe it to yourself to take the time to build a level converter. Once you've discovered how many problems it can solve, you'll wonder how you ever got along without it. They're so cheap and easy to build, you'll be heating up your soldering iron to make more of these useful gizmos before you know it.

Forensic work can be downright weird at times. To unwind, Peter Mosher likes to create his own "individualistic" electronic music using homemade analog synthesis equipment.

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Electronic Musician magazine

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Of course, words are cheap (well, actually, when printed in a magazine ad like this they're fairly expensive). But hearing is believing. Try out the ATR-1 at your local Antares dealer or call us for a free demo CD. Either way, we're confident you'll be convinced. Really. Here's what some ATR-1 users have to say:



"With the ATR-1, vocal sessions can focus on attitude, not intonation." ~MADAME MARIE CURIE*



"Nothing helps your peace of mind on tour like an ATR-1 in the rack." ~FRANZ KAFKA*

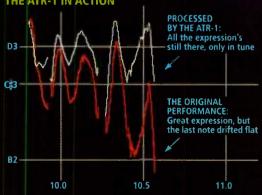
*not their real names



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THE ATR-1 IN ACTION



OK,)let's be honest. For most of you, "Perfect

three-day weightloss. Unless, of course, you

happen to be one of the thousands of audio

Pitch in a Box" is right up there on the

credibility scale with Elvis sightings and miracle

professionals who already depend on Antares's

amazing Auto-Tune[™] pitch-correcting software



Interactive Web Music

Music on the Web isn't just plain MIDI or streaming audio anymore.

By Scott R. Garrigus

Ithough the Internet's uses for business and entertainment are expanding daily, the essential format remains largely unchanged: text and graphics continue to make up most of the available content. And though most Web sites are still silent, a growing number of companies are recognizing the benefits that sound and music add as an enhancement to the browsing experience.



With SSEYO's *Koan X*, you can drag and drop templates to create a KoanMix file for driving the generative KoanMusic player engine.

This development opens the door to new opportunities for the modern musician. If you can create bandwidthfriendly MIDI and audio files, you may be able to supplement your income with online gigs. However, before you get too excited about the prospect of making big bucks by composing for the Web, you should know that Internet music production is no longer simply a matter of creating General MIDI files or streaming audio for background playback. These days the buzz is all about interactive music.

WHAT'S INTERACTIVE MUSIC?

Interactive music combines computer programming and musical composition to create a virtual environment in which the music content is influenced by external stimuli. So instead of being "set in stone" like, say, a movie soundtrack, interactive music changes based on input from the listener. In its simplest form, interactive music can be thought of as the triggering of sounds or musical passages by a user's actions. In a video game, for example, the player might click on an object to make his or her character stronger; this action could then trigger a distinctive "powerup" sound or musical sequence to let the player know that the character has indeed gained strength. Clicking on a different type of object could trigger a different sound to identify that object, and so on. At a Web site, for example, a

Inspiration...

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DESKTOP MUSICIAN



FIG. 1: Beatnik Converter enables you to create RMF files that hold the MIDI and audio files used in your composition.

(For an example, check out the interactive "Fame" demonstration at www .davidbowie.com.)

The Beatnik player works on both the Macintosh and PC platforms with Netscape Navigator, Microsoft Internet Explorer, and the America Online browser. Beatnik also has its own custom file format, called Rich Music Format (RMF). An RMF file is like a "container" for Standard MIDI Files and custom audio data. RMF files can include MIDI data and up to 128 custom instruments, each consisting of up to 128 custom sounds (similar to Downloadable Samples), along with a tamper-proof copyright notice and total file protection for secure delivery over the Internet.

SSEYO's KoanMusic works a bit differently. Although it outputs MIDI data, its input doesn't come from Standard MIDI Files. Instead, it gets instructions from its own custom file formats and uses those instructions to generate music "on the fly." The process is very similar to algorithmic composition (in which you set certain parameters, and the generative engine creates music based on your settings). The KoanMusic player plug-in works on the Mac and PC with Navigator and Internet Explorer.

HOW THE TWO WORK

To create interactive music with either of these interactive music technologies, you must learn some programming in either Java-Script or VBScript. These scripting languages allow you to develop computer programs that run inside an Internet browser. Because both KoanMusic and Beatnik support JavaScript as well as VBScript, and because VBScript works only with Internet Explorer, you're better off focusing on learning JavaScript. That way, you can be sure that whatever you create will run on both computer platforms and with either browser.

With Beatnik, you first create an RMF file to hold whatever MIDI and audio files you want to use in your composition. You can do this by downloading *Beat*nik Editor for the Mac or

Beatnik Converter for the PC (see Fig. 1). At the time of this writing, Beatnik Editor was still in beta, so it's a free download. Beatnik Converter costs \$34.95, although you can download a demo to try it out for 21 days. After creating your RMF file, you embed the Beatnik player into a Web page for basic playback using the following code:

<EMBED SRC="myfile.rmf TYPE="audio/rmf" WIDTH=144 HEIGHT=60 DISPLAY=SONG AUTOSTART=FALSE LOOP=FALSE PLUGINSPAGE="http://www .headspace.com/to/?player">

To add interactivity to the playback, you must use Beatnik's special library of JavaScript "methods." The music-Object library lets you control MIDI track mute and solo, volume, program changes, and much more. And because Beatnik has a built-in software synthesizer, you can even play individual notes through JavaScript. This excerpt of code, for instance, displays a link labeled Click Me. When a visitor clicks on the link, Beatnik plays the note C2:

Click Me

You'll also find support for Beatnik scripting in Macromedia *Dreamweaver* and NetObjects *Fusion*. With these Webauthoring programs, you can create interactive music pages without learning HTML coding or JavaScript. Further,

INTERACTIVE MUSIC RESOURCES

You can find all the tools and information you need to work with the Beatnik and KoanMusic players at the following Web sites:

Beatnik Home Page

www.beatnik.com

Download the free Beatnik player, along with demo versions of *Beatnik Converter* and *Beatnik Editor*. The full Beatnik authoring documentation is also available here.

SSEYO

www.sseyo.com

Download the free SSEYO KoanMusic plug-in, as well as demos of *Koan Pro* and *Koan X Platinum*. You'll also find information on using the KoanMusic technology.

Netscape's Online JavaScript Documentation

developer.netscape.com/docs/manuals/communicator/jsguide4 This online book describes the core JavaScript language and its extensions for use with a Web browser.

Microsoft's VBScript Home Page

msdn.microsoft.com/scripting/default.htm?/scripting/vbscript The official source of information for the VBScript language includes sample code and tutorials.

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DESKTOP MUSICIAN

Beatnik files can be encapsulated in a Macromedia *Shockwave* 7.0 movie or controlled with Macromedia *Flash* animations for more elaborate user interfaces.

The KoanMusic player uses what SSEYO calls a KoanMix file, a collection of templates consisting of special sets of "instructions" similar to precomposed MIDI sequence loops and one-shots. The KoanMusic engine intelligently combines the templates in a KoanMix file to produce ever-evolving musical pieces that are never played exactly the same way twice. Although KoanMix files can (but do not have to) contain sequenced information, they also include rules that can allow free real-time composition as well. It is this truly generative aspect that makes KoanMusic so bandwidth friendly.

To create your own KoanMix files, you must purchase *Koan Pro* (\$199.95) or *Koan X* (\$29.95; see opening art). The difference between the two products is that *Koan Pro* lets you create your own templates; *Koan X* doesn't. Koan-Music has its own set of JavaScript and VBScript "methods," so you can con-



TOLL

FREE

trol precisely how the music is generated. In fact, the latest KoanMusic player engine provides an even more sophisticated set of functions because it allows you to extract information from the musical process. This means that every time the music starts a new bar, for example, or every time middle C is played on MIDI channel 8, you could have a message pop up on a Web page. You can even drive Macromedia Flash and DHTML animations from a Koan-Music piece. Also, KoanMusic now supports MP3 samples on the Windows platform. (For an example, check out SSEYO's Flash-based generative Vul-KoanMP3 mixer on its Web site. MP3 mixer is freely distributable and comes with a dance-groove sample set.)

Of course, I've only scratched the surface of what these versatile interactive music technologies can do. A detailed discussion of how to use Beatnik

There is no standard format for applying interactive music to a Web site.

and KoanMusic to their fullest is beyond the scope of this article. But don't worry: further information abounds on the Internet, along with sample code and tutorials for each technology (see the sidebar "Interactive Music Resources").

BE PREPARED

Most Web-site developers are just beginning to appreciate the potential for adding interactive music and sound to their pages. You can get a jump on the rest of the crowd by studying the technology *now*. If you're well prepared, you may be able to launch a new career as a Web maestro. And even if that doesn't happen, you may still be able to use your newfound skills in the videogaming industry. However you look at it, interactive music is here to stay, and it's destined to grow in popularity.

Scott R. Garrigus is an author, musician, and multimedia expert. He plans to add interactive music to his Web site soon. Contact him at www.garrigus.com.

www.ReliableMusic.com

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Fractals and Music

Patterns in nature can be a musical inspiration.

By Gustavo Díaz-Jerez

ost musicians have seen fractal images—those hauntingly beautiful pictures generated through mathematical equations. But how many people are aware that the same equations can be used to create hauntingly beautiful music? This article will explore fractals and discuss how they can be employed in musical composition. The technique is so versatile that you can use it to create music in any style, and it has great potential for adding an exciting new element to your music.



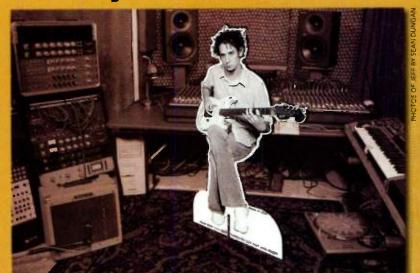
FIG. 1: Fractal images are generated by fairly simple mathematical formulas, yet they are very rich in detail. This is partly due to their use of *iterative*, or looping, patterns.

Studying music from a mathematical point of view is nothing new; it dates back to ancient Greece. Around the fifth century B.C., Pythagoras uncovered mathematical relationships in music, in which musical intervals are represented as ratios of whole numbers. For example, the interval of an octave would be represented in the Pythagorean system as a 2:1 ratio because the frequency of the higher pitch is twice that of the lower. By the same system, the interval of a fifth would be 3:2, a fourth 4:3, and so forth. Other systems relating music to math have been developed, mainly in the 20th century. Among the many examples are Joseph Schillinger's System of Musical Composition and a method of composition developed by Olivier Messiaen in the 1940s. The more recent discovery of fractals and particularly their link to music has opened a door to composers, behind which lies immense creative possibility.

WHAT ARE FRACTALS?

Fractals are visual representations of certain mathematical functions, which show increasing detail upon magnification. A very important phenomenon of fractals is that they manifest *selfsimilarity* at all scales. Benoit Mandelbrot, one of the fathers of fractal geometry (and the man who coined the term *fractal*), loosely defines fractals as "shapes that are equally complex in

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FIG. 2: One way to employ fractals in music is to map the numeric output of algorithms to note sequences. Figure 2a is the melody that results from a common algorithm called the Morse-Thue sequence; this melody is self-similar. Figure 2b shows the resulting melody if every other note of 2a is removed. Figure 2c shows the result of removing every other note from 2b.

their details as in their overall form. That is, if a piece of a fractal is suitably magnified to become of the same size as the whole, it should look like the whole, either exactly, or perhaps only slightly deformed."

One would expect that the construction of such complex shapes would require complex rules, but in reality, the algorithms (equations) that generate fractals are typically extraordinarily simple. Their visual results, however, show great richness. The seeming paradox is easily demystified: these algorithms involve "loops."

The key to the richness of detail that fractals exhibit is something that mathematicians call iteration. Most equations that we learned in school are linear-that is, the input is proportional to the output. For example, the equation $x^2 - 1 = 0$ is a linear equation. The equations that generate fractals, however, are nonlinear. Nonlinear equations involve iteration, which means that the solution of the equation is repeatedly fed back into itself. It is an arresting thought that something produced from a purely mathematical procedure can be so aesthetically pleasing (see Fig. 1).

FROM FRACTALS TO MUSIC

The secret behind fractal music involves something that mathematicians call *mapping*. Mapping means creating a direct relationship between the numerical output of an equation and certain parameters that can employ that output. Fractal images, for instance, are produced by mapping the output of equations to colored screen pixels. The parameters in fractal music are sonic rather than visual and can include pitch, rhythmic values, and dynamics. This mapping process is the link between the worlds of numbers and sound.

The mapping of nonmusical material to pitches is another ancient idea. A medieval technique known as soggetto cavato (a theme "carved" from words) maps the individual letters of words to music. For example, Hercules Dux Ferrarie, a mass by Josquin des Prez dedicated to Hercules, Duke of Ferrara, uses a theme based on the vowels in the Duke's name. The sequence of vowels e-u-e-u-e-a-i-e is first mapped to the six solmization syllables (ut, re, mi, fa, sol, la), which generates the following syllables: re, ut, re, ut, re, fa, mi, re. When converted to traditional note names, one gets the pitches, D-C-D-C-D-F-E-D. This technique has been used extensively by composers throughout history, including J. S. Bach.

There is no single way in which to map numerical output to music; in fact, different mappings can be applied to the same numerical output. For instance, the output of fractal equations can be mapped to different note sets, and the resulting melodies will differ accordingly.

To illustrate how this mapping process works, let's analyze one of the most fertile algorithms: the Morse-Thue sequence. This number sequence, although relatively simple to construct from a mathematical standpoint, shows many interesting properties, including selfsimilarity. It is generated by the nonnegative integers 0, 1, 2, 3,

4, 5, 6, and so on, expressed in binary notation (base 2):

0 1 10 11 100 101 110 111 1000 1001

Now, add the digits of each term in the sequence, yielding

 $\underline{0} 1 \underline{1} 2 \underline{1} 2 \underline{2} 3 \underline{1} 2 \dots$

Notice that if you take out every other number (keeping the underlined ones), you get back the same sequence:

01121....

So how do you turn all this into music? That is where mapping comes into play. One way is to map the numbers to a scale. If C major is to be our scale, then the note C would be mapped to the number 0, D to 1, E to 2, F to 3, G to 4, and so on. This mapping results in the melody shown in **Figure 2a**. Notice that, if you remove every other note in the melody, what remains is the same melody as the



FIG. 3: Using a different mapping process can create an entirely new melody from the Morse-Thue sequence. Multiplying every term in the sequence before converting them to notes produces the melody seen here.

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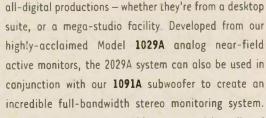


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original (see Fig. 2b). If you repeat this procedure with the already truncated melody, the result is no different (see Fig. 2c); the melody is *self-similar*.

You are not restricted to base 2; using a different base yields different melodies. You can also map the numeric values to other scales or groups of notes to get different results. To make things even more interesting, you can multiply every term in the sequence by a constant value. This can produce curious results, depending upon which combination of base/multiplier you have chosen. A very fruitful combination, for example, is base 2 and multiplier 33. Mapped to a C major scale, this combination produces the melody shown in **Figure 3**.

The characteristics of these "fractal melodies" are mind-boggling. If you analyze them in detail, you will find all types of self-similarities and interrelationships among the pitches, and because there is an infinite range of numbers to choose from, you have an inexhaustible source of melodic subject matter.

Other algorithms can be mapped to musical parameters in a similar manner. Every algorithm generates a unique set of melodies, but each has its own unique "fingerprint," even when different mappings are applied. Many of the programs mentioned later in this article automatically compute the entire mapping process, making it transparent to the user. This allows musicians to focus their creativity exclusively on musical questions. Little or no math background is needed to learn most fractal software.

Of course, as with any other "system of composition," it is ultimately the composer's musical talent that determines the quality of a piece of music; plugging in some parameters and pushing a few buttons is not going to produce a masterpiece. In the hands of a skilled composer, however, fractal music can serve as a source of inspiration and as a tool for raw musical material, which can later be developed, incorporated, and refined.

NOW IT'S YOUR TURN

Many fractal music programs are available today, and you'll be surprised how powerful some of them are. Many programs are freeware or shareware that will get you started quickly in fractal music composition. What follows is a short description of some of the

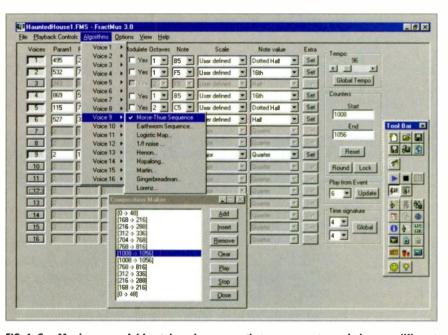


FIG. 4: *FracMus* is a powerful fractal music program that can generate music in many different styles. The program, written by author Gustavo Díaz-Jerez, is available for Windows computers as freeware.

most interesting fractal music programs on the Net. Some of the Windows programs listed require the Visual Basic Runtime Libraries, which you can obtain from www.softseek .com/Utilities/VBRUN_Files.

FractMus (Windows 95/98/NT). My own program, FractMus 2.5, is very easy to use and allows you to choose from ten well-known note-generating algorithms, including the aforementioned Morse-Thue sequence (see Fig. 4). You can assign different algorithms to all 16 MIDI channels independently. The program has 15 predefined scales and allows you to create your own. Other features include predefined rhythmic values (along with user-defined ones), inversion of melodies, modulation, tempo change, time signature, stereo effects (panning), and full percussion support. It also includes a Composition Maker and a Composition Randomizer, two tools that help you create your compositions almost effortlessly.

With *FractMus* you get immediate results: the music is computed and played in real time. The program writes Standard MIDI Files from your compositions, allowing you to load them into your favorite sequencer or music editor for further manipulation. In addition, *FractMus* translates your compositions into fractal images. It comes with complete documentation and dozens of example files created by me and other users. The program is very flexible and will let you create music in any style you can think of, from minimalist to stochastic. It's great for commercial music, too. Last but not least, the program is free. You can download it from my home page at www.geocities.com/ SiliconValley/Haven/4386.

A Musical Generator (Windows 95/98/NT). A Musical Generator 1.2 by MuSoft Builders is another worthwhile program. It draws musical material from many sources, including complex maps, mathematical constants (such as π), images, text, and user-provided data. The program is very well documented and has an easy-to-use interface. It also includes several sample files and tutorials. The program writes Standard MIDI Files and is distributed as shareware. You can download it from MuSoft Builders' home page at www.musoftbuilders.com.

Art Song and MusicLab I: Music from Chaos (Windows 95/98). These shareware programs by David Strohbeen use iterated function systems and quadratic functions to create strange attractors from which the music is derived. They can also transform graphic images into music. A large number of musical parameters are user-adjustable, allowing you to create different musical styles. In addition, you can change these parameters at any time during the compositional process. Each of these programs generate standard MIDI files, and you will find full online documentation. You can download Art Song and MusicLab I from members.aol.com/ strohbeen/fmlsw.html. You'll also find links to other fractal music software at that site.

MusiNum (Windows 3.1). Lars Kindermann's free program generates MIDI files using the Morse-Thue sequence exclusively. The user controls musical parameters such as instrument, scale, and duration. The program is easy to use and well documented. You can download *MusiNum* from www.forwiss .uni-erlangen.de/~kinderma/musinum.

LoShuMusic and FibonacciBlues (Macintosh). These two Mac shareware applications use the Fibonacci sequence and aleatoric procedures as a basis for music generation. You can download them from ftp://mirror .apple.com/mirrors/Info-Mac .Archive/art/fibonacci-blues-02.hqx and ftp://mirror.apple.com/mirrors/ Info-Mac.Archive/art/loshu-music-02 .hqx respectively.

Symbolic Composer (Macintosh). Symbolic Composer is a commercial program that uses a proprietary language of 700 building blocks to compose. It lets users apply a wide range of algorithms to most musical parameters. This program is not particularly easy to use or learn, but in the hands of a skillful

The algorithms that generate fractals are typically extraordinarily simple.

composer, it is one of the most powerful tools for experimental music. You can download a working demo of the program from symcom.hypermart.net.

Csound (all platforms). Along with its many other talents, the Csound programming language can be a good vehicle for trying out experimental mappings of data to sound parameters,

such as those used in fractal music. In fact. Csound wizard Hans Mikelson has just published some fascinating Csound files that use different elements of fractal graphics as parameter values for Csound instruments. For example, one of Mikelson's files uses the number of loops in the image to determine the spectrum of his sounds, and another maps RGB values of a color image to the frequency domain (did someone say "MetaSynth"?). You can read about the process at www.werewolf.net/ ~hljmm/Ezine/synthesis. Note EM associate editor Dennis Miller's fractal image on the cover of the summer Csound 'zine.

I hope that this survey will give you an incentive to explore the vast realm of fractal music. Regardless of the musical style in which you work, fractals can be a great way to generate new and interesting material.

Gustavo Díaz-Jerez is a concert pianist, composer, and computer programmer. He is a graduate of the Manhattan School of Music in New York, where he is currently pursuing his doctorate in musical arts.





Recording Electric Guitar

Expand your sonic palette with these not-so-ordinary techniques.

By Myles Boisen

he electric guitar ("El Gtr" in engineer shorthand) is one of the easiest instruments to record. Even a modest rig—a good guitar coupled with a decent amplifier makes the engineer's job a cinch, offering plenty of level, a variety of easily adjustable tones, and—with most modern amps, at least—an assortment of "flavor enhancers" such as tube saturation, overdrive, and compression.



You can add an airy, percussive edge to electric guitar tracks by miking the bridge area and blending the sound of the pick on the strings with the amp tracks. Here, composer/arranger Steve Kirk's 1965 Gibson ES-175 is miked with a large-diaphragm Manley Reference Cardioid tube condenser.

In addition, the limited bandwidth of a typical electric-guitar track is ideally suited to the frequency response of affordable dynamic microphones. But that doesn't mean that using the ageold standard of miking guitar amps—a Shure SM57 shoved up against the grille cloth—is the best way to get El Gtr to stand out in a mix.

Like many engineers, I learned the basics of recording guitars by doing live sound and occasional session work. But my "higher education" began when I was hired by a blues/R&B-oriented mail-order record company, and I "had" to listen all day long to recordings from the '40s, '50s, and '60s. No matter how primitive or poor the recording quality on those old discs, I was constantly amazed by the array of exciting sounds produced by electric guitar. Later, when I started recording blues sessions in my own studio, I learned firsthand about the key elements that contributed to the great tones that I'd heard on those classic recordings.

TUBES ON 10

Nothing sounds as good as a tube amp turned up to 10. You can do this with some old amps, and they will sound fairly clean; others will explode. Use caution and keep an eye out for plumes of smoke. Newer tube amps generally have separate preamp and master-gain controls that can duplicate the gritty anarchy of yore, minus the lease-breaking



designing our new workstation we had to carefully consider many things, but above all, the most important was what you might actually want. So if you're about to settle for the usual mix of synth and sequencer with a few extra frills thrown in; wait ! Wouldn't you like to have a top of the line pro piano too? What about a drawbar organ? Or how about a universal sample player? Or a groove machine? The Equinox Pro is all of these things and much, much more at an unbelievable price that makes sweet dreams come true. Take a look...

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ERFORMANCE-Combine up to 16 ounds together in any kind of mult

UINOX

DRUMS KITS - 43 on board drum kits from a brand new sample library ranging from hip-hap, dance and techno to fusion, rock, jazz and 70s. Editing is quick and easy and you can save as many user drum kits as you like. (up to 2048 kits)

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SINGLE INSTRUMENT GROOVES Any musician will find our collection of "single instrument grooves" indispensable. These superb phrases, offering everything from drum loops and bass lines to clavinet riffs. organ licks, guitar prooves and a whole lot more, have been created by some of the world's top session musicians. And they re yours to use as you please in any sequence or song creation of your own. We ve already taken care of the royalty payments !

CONTROLLER/6 MIDI PORTS - With no less than 6 independent MIDI ports, the Equinox PRO offers full control over 32 channels of MIDI data. The keyboard can be used as a 16 zone controller with each zone having its own programmable upper and lower plit points and velocity limits (which may overlap with other zones of course !) 3 PROGRAMMABLE PEDAL JACKS

D.T. GROOVES

Having 1024 on-board grooves purs the Equinox PPO head and should ins above any of those stand alone Groove Stations" All grooves can be edited filtered, remixed and even scratched DJ style" in real time. lus you can create your own grooves from any midi file or sequencer song

HARD DISK - The Equinox PRO comes as standard with a 2Gigabyte internal hard disk. Anything can be stored on the hard drive - Sounds, Performances, Drum Kits, Imported Samples, Grooves, Midi Files etc. The hand disk can also be accessed "invisibly" so that new songs can be loaded while the sequencer is already busy playing another song SCSI INTERFACE

GENERALMUSIC

lit or layer configuration. The sliders d buttons can instantly be used as nutes and volume controls for your 16

DRAWBARS - One innocent looking little outton transforms the Equinox PRO into a versatile drowbar organ. While the sliders become a set of traditional organ drowbars, the front panel offers instant access to Key Click, Percussion and Rotary speaker speed (all user editable). There's even a choice of Drawbar voicings (Smooth Hord Jazz and Rock) and the added flexibility to edit each drawbar's pitch, and pan position dividually

REAL TIME CONTROLLERS Whether your operations single sounds performances, samples, grooves or sequencer songs, grabbing a handful of sliders will instantly transport you to analog beaven, (and without the need for WD40) 51 ders are pre-assigned to control envelope attack/ decoy/ release filter cut-off / resonance and LFO death and speed but any slider can ensity be re-programmed to control the parameter of your choice. In sequencer no de the sliders can also be set to function as a 16 track mixer with the press of a single button.

POWERFUL SEQUENCER The versatile to track sequencer offers a staggering 250,000 events of storage and allows you to store 16 songs in memory at once. Add to this the power of 1/192 resolution. Groove quantize, Event list editing and the ability to record ALL slider move ments (either real-time editing of filters, envelopes etc or multiple track mixdowns) plus the life saving UNDO function and you'll start to see why the Equinox PRO is really a composer's dream machine. The sequencer even has it's own 16 tracks independent from the rest of the synthesizer so while a 16 track song is playing, you can freely select any complex multi split or layered performance to play along with it Simply put it's just like having two separate keyboards EFFECTS- Up to four independently assignable DSPs are available offering 85 crystal clear digital effects. Ranging from simple reverbs and choruses to complex

composite algorithms like GUITAR FX, (Distortion, Gate, Delay and 4 Band EQ using only one DSP section 1), 3D ENHANCER, AUDIO EXCITER, RING MODULATOR, 10 BAND EQ and 4 PART PITCH SHIFTER.

ARPEGGIATOR - With 16 factory presents and to us a programmable patterns the Equinax PRO's Arprogrator will so no you searing back to the 70's (with a tow little t choole ical mice les thrown in for good measure) As well as the usual UP and DOWN directions there s also RANDOM and INPUT (the order in which the rotes wer play: d). Velocity can be pre-set, disable d or controlled by your playing style and there s also a CRESCENDO feature to reduce or increase velocities as the pattern cycles around There's even a HOLD button which lets your pattern do it's thing while giving you both hands free to tweak the arpeggiating sound with the real time controllers Breakthrough Price: \$3,695.00

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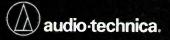
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• RECORDING MUSICIAN

SPLs. For jazz and other clean guitar styles, it's okay to turn the volume down a bit, as long as you don't "underdo" it.

But having hot tubes is only half the recipe for getting great tone. Room sound is the other ingredient necessary for obtaining a full-bodied guitar track. It didn't take me long to figure out that the guitarists on my formative blues sessions were slyly contributing to my "education" by nudging the mics away from their amps as soon as I left the room. Thanks to their clandestine efforts, my ears opened up to an entire new world of electric-guitar sounds.

I've since developed several recording techniques that are a sure cure for the El Gtr blahs. Try the following four tricks in their order of appearance, as they are progressively more complex.

SIMPLE DOES IT

Once you have the essential elements in place—a great amp, guitar, and guitarist—you almost can't help but get a great guitar tone. Crank the amp up to the appropriate level and begin with some mic comparisons. It's especially telling to audition different types of mics: for example, dynamics, ribbons, and large-diaphragm condensers. (I rarely use small-diaphragm condensers for miking guitar amps; on the other hand, I've found that almost any microphone will strike gold once you find the right spot for it.)

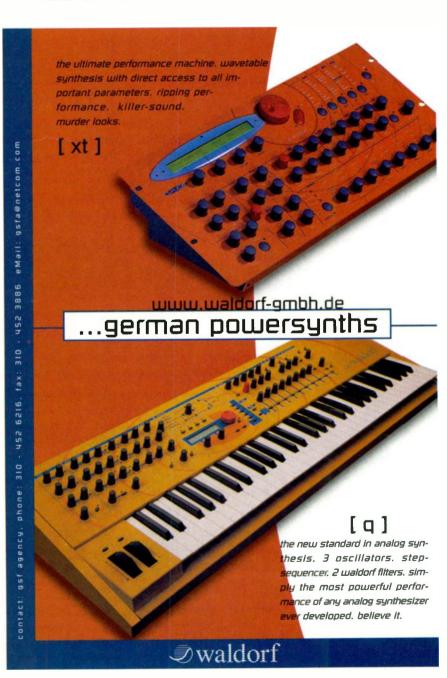
My favorite dynamic mics for this application are the Sennheiser MD 421 and 441 and the Shure Beta 58 (which has a fuller sound than the SM57). My favorite ribbons are the Royer R-121 and Coles 4038. For large-diaphragm condensers, I lean toward tube models, in particular the Lawson L47MP and Neumann M147.

Start with all of the mics clustered together three to six inches from the grille cloth, pointed at the center of the speaker. On a multiple-speaker cabinet, don't assume that all the speakers sound the same. Rather, listen to each of them at a sensible volume, and then mic the one that sounds best. If the speakers sound alike, a miking position close to the floor will generally provide a little more low end.

Back in the control room, audition each mic, preferably as the guitarist plays along with the other instruments. Listen carefully to how each microphone sounds on its own and, more importantly, to how it works in the mix. Usually, one microphone will come up a winner on the first pass. Don't stop there, however. Instead, leave the "winning" microphone where it is and experiment with the placement of the other two mics. Time--and mic selection---permitting, you may also wish to do a second round of testing with other microphones.

The key elements of mic positioning are distance from the source and orientation to it. Moving the mic closer to the amp provides more definition, increased highs and lows, and less room sound. As you pull the mic back, the sound becomes less detailed, more "midrangey," and more blended with the ambience. Depending on the room you're in, a distant-miked amp may gain a natural presence and unique character in the mix, despite an apparent decrease in definition. On the other hand, placing the mic too far back will result in a washed-out, murky, or hard-to-control tone.

Mic orientation, or the angle of the mic in relation to the speaker, becomes more critical as the mic is moved closer



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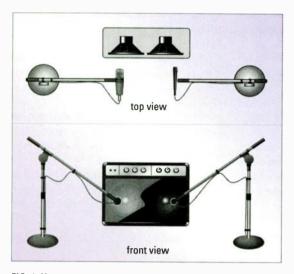


FIG. 1: You can create a monstrous sound in the mix by using two different mic models on a twin-speaker amp. Make sure that the two microphones are the same distance from the amp, positioned at the same angle relative to the speakers.

to the amp. Pointing the mic at the center of the cone will yield more active highs and better transient detail but fewer lows. As you move the mic toward the outer rim of the speaker, maintaining a 90-degree angle with the grille cloth, the low frequencies gradually increase because of proximity effect and other factors, resulting in a sound that may be warmer, softer, or more powerful. Many engineers like to blend these complex characteristics by angling the mic between 30 and 60 degrees off-axis from the center of the speaker.

Of course, it is vital that you experiment and let your ears be your guide with all the techniques mentioned in this article. Don't hesitate to try a crazy placement or an unusual mic such as a PZM (pressure-zone mic). With persistence and a bit of luck, you will likely discover some tricks of your own.

TAKE TWO

Once you've mastered the intricacies of single-transducer miking, it's fun to start working in stereo. For true stereo recording, you need a matched mic pair as well as a twin-speaker amplifier, preferably one with built-in stereo chorus and vibrato (such as a vintage Magnatone or a Roland Jazz Chorus). Two separate amplifiers fed by the same stereo delay or multi-effects unit will also work.

Mic each speaker or amp, pan the two channels apart, and let the effects work their magic. Hard-panning to the extreme left and right produces the most dramatic results; should this prove too dizzying, try panning one microphone toward the center, or move the tracks toward a more centered symmetrical position.

You can use similar twomic techniques, minus the effects, on a single amp to capture a variety of largerthan-life guitar sounds. One trick that I stumbled upon involves miking a twin-speaker amp with two mics that are close in response, but not matched (see Fig. 1). The first time I tried this, on a session with guitarist Paris Slim, I used an Electro-Voice RE20 and a Sennheiser 441. Place one mic on each

speaker at the same distance and orientation, and check the pair for phase cancellation by panning them to the same spot and listening in mono. The minute differences between the speakers, mics, and mic positions, combined with double-tracking, creates a monstrous presence when the tracks are hard-panned in the mix, and opens up a world of possibilities for separate EQ and effects processing. If you don't need the guitar to dominate the mix, you also can sum these mono-compatible tracks together to a single pan position for a noticeably bigger sound.

To capture aggressive, distorted guitar sounds, my studio partner Bart Thurber likes to use two mics in an XY configuration on a single speaker: a Shure SM57 aimed at the middle of the speaker and a Sennheiser 441 (with the high-end boost switch engaged) pointed at the edge of the cone. The SM57's signal is sent to a compressor, and the two mic signals are then mixed together and recorded to one track. This technique provides some compression for the harshest high frequencies and strong, midrange volume peaks picked up by the SM57, while simultaneously delivering full highs and lows through the 441.

Another variation on the two-mic technique involves miking the front and back of an open-backed cabinet.

SELECTED DISCOGRAPHY

The following CDs, engineered by Myles Boisen, are recommended listening for the electric-guitar miking techniques described in this article.

The Dynatones, Shake That Mess (Blue Suit Records, 1999)

"Bulldog" (tubes on 10)

"Ace of Spades" (stereo vibrato)

"Memphis Women" (air guitar)

Available from www.dynatones.com

Steve Kirk, Steve Kirk Pop (SKP Records, 1999)

"River of the White Lake" (tubes on 10) all tracks (air guitar, multisourcing) Available from skirk@Imi.net

Ben Marcato and His Mondo Combo Party Mix (Urgent Records, 1999) "Too Lazy to Work" and "Little Joe from Chicago" (air guitar)

"Smack Dab in the Middle" (tubes on 10, air guitar) Available from www.benmarcato.com

Paris Slim, Bleedin' Heart (Globe Records, 1996)

"The Day I Met the Boogie Man" and "The Candle's Burnin' Low" (air guitar) Available from www.westcoastblues.com

Ronald Thompson (avant-garde seven-string guitar), Spiritpark, vol. 3, Music for Percussion, Saxophone and Guitar (Spiritpark Records, 1999) and Spiritpark, vol. 1, Music for Solo Guitar (Spiritpark Records, 1999) all tracks (air guitar, multisourcing) Available from Spiritpark@aol.com

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For this application, be sure to place the mics at an equal distance from the speaker itself and reverse the phase of the rear mic.

TO AIR IS HUMAN

Forget Risky Business (remember the famous scene of Tom Cruise rockin' out in his boxers?); this technique, which I consider real air guitar, is serious business. It entails capturing the airy, percussive sound of the plectrum strumming or picking the electric guitar's strings-either in acoustic isolation or combined with the ambient sound from the amp-and then mixing this sound with the recorded amplifier sound. The addition of just a little percussive plucking can enhance the presence wonderfully for any style of guitar playing. In my opinion, it's the greatest studio-recording innovation since John Bonham's distinctive drum sound.

The blues was my inspirationspecifically, the late-'40s solo recordings of John Lee Hooker. "Hobo Blues" is an excellent example of early Hooker on which his violent string slapping-clearly audible in a blend of amp sound, haunting vocals, and trademark foot stomping-creates an indelible realism and engaging intimacy. My recording of Paris Slim's "The Day I Met The Boogie Man" (see the sidebar "Selected Discography") was one of my early experiments with this technique. Since that time, I have used a discrete "air-guitar" mic whenever I have had an available track for it. Guitarists may initially be skeptical of such unusual miking, but it's always a treat to watch their faces light up as they listen to the monitors deliver the bright, transient sounds that they have been accustomed to hearing during their years of practicing their instruments.

I have achieved my best results with this technique when miking resonant hollow-body guitars, getting the mic in as close as possible to the guitarist's picking hand. Large-diaphragm condensers, especially the Neumann U 87 and Manley Cardioid Reference tube mic, have proven superlative performers on big-box guitars such as the Gibson ES-175 (see photo on p. 114). The small-diaphragm Oktava MC 012 and medium-diaphragm Shure KSM32 have worked wonders on solid-body instruments, most notably on improvi-

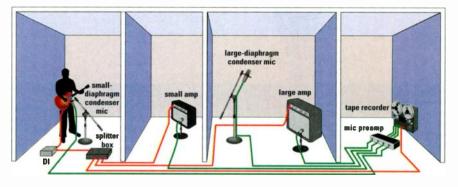


FIG. 2: For huge guitar sounds and any number of processing options, use an air mic on the guitar itself and split the guitar output, sending one signal direct to tape and the others to amps located in separate rooms, each miked differently.

sational-guitarist Ron Thompson's seven-string custom axe.

Mix magazine columnist Stephen St. Croix recently documented his own variation of this technique. He uses two "air" mics—one each on the low and high strings—with the resulting tracks panned hard left and right in the stereo spectrum.

An overdubbing session is ideal for air-guitar miking because there is no leakage from other instruments. I usually prefer to maintain total isolation between the two sources, placing the guitarist and amp in separate rooms. But for some production styles, the acoustic air mic can also do double duty as a distant room mic for the amp, with the ratio of pick sound to ambience determined by mic placement and amp volume. I've recorded some very hefty-sounding rock 'n' roll power chording this way, as well as a variety of vintage-style solos and rhythm parts. At the board, a low shelving or low-midrange EQ cut, combined with a subtle high-end boost around 4 to 6 kHz, will usually help these tracks jump out of the mix.

The key to capturing any kind of ambient tracks is a good reverberant space, although a narrow or dead room can also work, as long as there is sufficient distance between the guitarist and the amp. I usually put the air mic at least ten feet from the amp, positioned offaxis, or in an omnidirectional pattern to pick up as much reflected sound as possible. Placing a baffle between the guitarist and the amp will increase the apparent room size, as will making the amp sound pass through a doorway or turn a corner into another room.

Should space restrictions or volume levels make these methods impracti-

cal, try adding an air-guitar part as an overdub to a conventionally miked guitar track. The principle is similar to vocal doubling, for which the same part is performed twice; you may not be able to do this for an improvised solo, but for rhythm parts or composed lines, it's a snap. In addition, double tracking with a bright acoustic guitar or a smooth-sounding hollow body will add extra richness and some slick, bigbudget zing to your mixes.

MULTIPLICATION ROCK

After you have the hang of mono and stereo miking, room miking, and air guitar, you may be ready for the final frontier of El Gtr exploration. The time-consuming technique that I call "multisourcing" combines all the aforementioned methods, multiplied by the infinite possibilities created by splitting the guitar output and sending it simultaneously to different amps (using, for example, a Whirlwind Selector splitter box).

When I first tried multisourcing, on a solo project by Club Foot Orchestra guitarist Steve Kirk, I used an air mic, a direct source (Manley tube DI box or speaker emulator output from Kirk's Marshall JMP-1 tube preamp), a close mic on a clean-sounding Fender Princeton amp, and close and distant mics on a cranked-up Marshall cabinet (see Fig. 2). And that was just for the first rhythm track! As you may imagine, mixing was a lot of fun, and after that day there's been no going back to the old SM57 shoved up against the grille cloth. If you dare, you can take it from there. The only limitations are your time, the guitarist's patience, and available tracks. Oh yes-and lots and lots of mics.

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Ocean of Promotion

Make a splash with creative and well-planned marketing.

By Lygia Ferra and Erik Hawkins

n the September 1999 "Working Musician" column ("Expose Yourself"), we gave you several tips to help you kick your CD-promotion campaign into high gear. This month we offer another installation of marketing ideas that you can add to that arsenal. Remember, although these tips are intended for the independent musician on a limited budget, you must have a sum of money large enough to realistically execute them. We suggest a budget of \$10,000—an extremely modest amount in the world of record promotions. Again, your campaign should be



regional in scope so you can use your limited budget most effectively; covering one area well is better than launching a poorly funded national campaign.

Refer to the table in the September issue ("The Budget Laid Bare," p. 108) for an example of how you could allocate a \$10,000 budget, including some of the marketing ideas that are presented this month. You can also find that table online at the **EM** Web site, www.emusician.com.

GONE SURFING

A Web site is a great means of letting people know about your project. You can post sound bites, pictures, and information about upcoming shows and retail locations that sell your CD (see Fig. 1). People will bookmark your site and visit more often if you update it regularly. Keep it interesting by periodically offering contests, free song downloads, and new images. Search engines that use spider programs to find and compile Web sites are also more apt to list your site and keep it in the top 20 if it's updated frequently. (For more on promoting your Web site, check out the April and May 1998 installments of "The Biz" at www.emusician.com.)

Although many Internet providers offer free Web sites as part of their service, using a dedicated Web host server is the most professional way to go. This allows you to have your own domain name (such as www.myband.com),

RELIVE THE PAST REGULARLY.

DELAVU

ith the Yamaha OIV, you can create the exact same mix every time, right down to the EQ settings on input 12. That's because the OIV digital mixer lets you save every detail of its 32 parametric EQs, 22 limiter/compressor/gates, two 32-bit

effects processors and 15 motorized faders. Then, when the time is right, you hit one button and recall the mix precisely and instantly. And, with an external sequencer, you can even let the 01V perform the entire mix. So last night's performance sounds exactly like tomorrow's. The OIV comes with 24 inputs, 6 busses, 6 aux sends and 12 mic preamps. If that's not enough for you, link OIVs together to create a much larger digital console without paying the price. In fact, the OIV comes in at a paltry \$1,999, far less than the cost

of all the "external" digital gear it includes free.

With its advanced components and its ability to save past mixes, the Yamaha OIV should make your mixing future quite clear. Get one today at an authorized Yamaha Pro Audio dealer.

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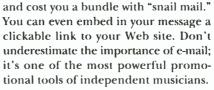
whereas a standard Internet service such as America Online, or a thirdparty music distributor such as IUMA, relegates you to an unwieldy Web address (for example, www.internetservice .com/members/music/myband). A good dedicated Web host, such as Best Internet, will help you register your domain name and get you started uploading your first pages (for contact information, see the sidebar "Promotional Resources, Part II"). It costs \$70 to register your domain name for the first two years, and there's usually about a \$50 setup fee with your server and a monthly service charge of about \$25.

You will save hundreds of dollars in programmer fees if you learn HTML. It's easy to comprehend, and plenty of free HTML software is available for download. AOL is a great resource for HTML tutorials and software.

YOU'VE GOT MAIL

The mailing list is an old standby, and for good reason: it works. If you haven't already started one, get one rolling. Sign up people at shows and people who like your sample tape, or get names from other local bands in your musical genre. Collect names ambitiously, but try to avoid including people who won't be interested in what you are doing. Including too many errant names on your list will simply waste your money on postage.

The Internet version, the e-mail list, is equally effective and much easier to work with. Trade e-mail lists with your friends, pick up "cc'd" addresses...the sources for obtaining e-mail addresses are boundless. You don't have to worry about whether all your addressees are part of your target audience, because sending e-mail is free. Armed with a list of 1,000 e-mail addresses, you can accomplish in an instant what would otherwise take days



POSTCARDS FROM PARADISE

Postcards are inexpensive and have a variety of promotional uses. You can use them for mailings and give them away as gifts. Music stores usually provide an area where bands leave postcards and

PROMOTIONAL RESOURCES, PART II

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GoCard

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Modern Postcard

1675 Faraday Ave. Carlsbad, CA 92008 tel. (800) 959-8365 fax (760) 431-1939 e-mail modern.cs@irisgroup.com Web www.modernpostcard.com

Network Solutions

tel. (703) 742-4777 Web www.networksolutions.com or www.internic.net

TSA Books

Tim Sweeney & Associates 21213-B Hawthorne Blvd., Ste. 5255 Torrence, CA 90503 tel. (310) 542-1322 fax (310) 542-1300 e-mail koti@pacbell.net Web www.tsamusic.com

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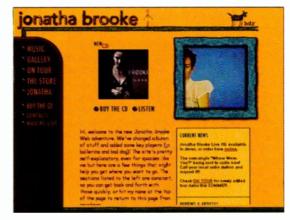


FIG. 1: A good Web site must be informative and fun to visit. Make a point of updating your site frequently with current performance dates, new graphics, and goodies such as free song downloads or contests.

flyers. Make your postcard a work of art—use a cool picture, eye-catching graphics, a profound phrase—so folks are more apt to hang on to them, put them on their bulletin boards, or mail them to their friends. You can generate photo-ready artwork on your personal computer with a good graphics program and a scanner. Leave the back of the card blank (except for your name and contact information); use this space to adhere stickers, made on your computer, with current information about upcoming shows and special events.

Count on spending about \$150 for 1,000 full-color postcards. (Check out Modern Postcard; they do great work.) Printing is inexpensive enough that you can afford to make several different postcards for variety. If you'd like to have your postcards distributed, talk to Go-Card, the company that places those postcard racks that you see in coffeehouses, gyms, and record stores. The company will print and distribute 30,000 cards for around \$2,400. Describe your target market to them, and they'll help you choose the area best suited to your music from their more than 1,500 locations.

GIG A GOOD LOT

You've heard it countless times before, but this tip bears repeating: get out there and gig. Performing remains one of the best ways to promote yourself. The key to making live shows productive and lucrative is planning your strategies well in advance. Getting people to attend your show is *your* job, so promote your gigs at least two weeks before you play to ensure a respectable draw. Club owners hate empty clubs

[nord modular]

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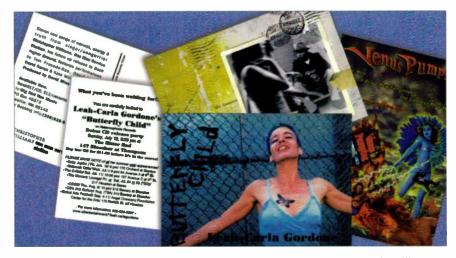


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Distributi di By Armadiilo Enterprises 1\$251 Rooseveit 31vd Silite 206 Clearwater, FL 33760 727-519-9869 and will quickly put you on the "not worthy" list.

Set up a booth at the front door for collecting mailing-list names and selling your merchandise. Ask a friend to work the booth and offer him or her a commission as extra incentive. Your stage show should be tight but not overrehearsed; you want good audience interaction, minimal dead time between songs, and a confident performance. Finally, choose your gigs wisely. Don't play so many shows that you spread your limited promotional resources too thinly—one killer show with a big audience is more effective then a ton of half-baked shows with low attendance.

If you're concerned that you won't be able to rustle up an adequate draw to satisfy the club owners, open up for more popular bands. Choose acts whose style is similar to your own, and piggyback your promotional efforts on theirs. In order to find the acts you'd like to open for, you'll need to meet the right people—in other words, schmooze. For more in-depth advice, we highly recommend Tim Sweeney's audio-cassette book, *Guide to Successfully Playing Live.*



Postcards are inexpensive to make and can be both artistic and useful. People will use an attractive postcard as decoration and to mail to their friends.

DON'T JUST STAND THERE

Authors make guest appearances where their books are being sold to help sell units. Musicians can do the same thing. Playing live in stores is a great way to meet the public and support the retail establishment where your album is being sold. Most stores are more than happy to have you come and perform. because it's an opportunity to generate sales for the store. Large chain stores usually have somebody in charge of booking in stores; in fact, some major chains, such as Borders, actively pursue in-store performances. Try asking for the entertainment coordinator. At smaller mom-and-pop stores, this will be handled by the owner or manager.



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circle #573 on reader service card



FIG. 2: To make your CD-release party alluring and memorable, be creative in choosing a venue. The Crucible Steel Gallery in San Francisco provided a relaxed but festive atmosphere for singer/ songwriter Erica Ballinger to celebrate the release of her EP *Life*.

Treat in-store performances just as you do a regular club gig. Promote the show a few weeks ahead of time. Make sure your product is prominently displayed near the area where you'll be playing (banners work well and cost only \$50 to \$100, depending on size and number of colors). Set up a small table to sell additional merchandise such as T-shirts, stickers, and posters (you'll probably need to give the store a percentage of these sales); and collect names for your mailing list. Often, stores are even willing to split the cost of print ads if your draw is big enoughmore people in the store means more potential store sales.

YOU OUGHTTA BE IN PICTURES

Where better to find a captive audience than at a movie theater? Think of all those people in their seats waiting to see the movie, staring attentively at the ads and trailers before the film starts. Most theaters run ads for about 10 minutes, beginning approximately 20 minutes before the movie, followed by major movie previews. You can buy these ad spots through the National Cinema Network.

Prices can be as high as \$300 per week or as little as \$92 per week, depending on theater and location. You get three ten-second spots at every screening in the theater that week. National Cinema Network takes your images and text and creates a slide that goes out to the theaters of your choice. (Note: it's just a slide; no sound or video is available.) In a five-screen theater you would get 315 exposures in a single week. The key to having this promotion pay off is to make sure that your slides get into theaters where your potential audience goes. Pick neighborhoods carefully and know each region's musical preferences. Your National Cinema Network sales rep can help you target your market.

PARTY ON

Have a record-release party. It may sound pretty routine, but you can make it fun and profitable (see Fig. 2). Certainly, after finishing your album, you'll want to have a party and invite all your friends and family to celebrate your accomplishment. Go one step further, however, and let the press and A&R reps know about the event. These are the people who can make things happen for you, so concentrate your efforts on luring them to your gathering.

Everybody loves free food and entertainment, so use these enticements to get VIPs to show up. Have your party catered, provide some free beverages, and have a few local acts perform with your band headlining. Make sure your invitees know about these perks—they'll probably be hungry after work and come just for the food. With full bellies they'll be more receptive to your music.

Obviously, you could spend a ton of money on this sort of event, but don't. Be creative and take advantage of your resources, and you should be able to keep it under \$1,000. If you have a friend with a mansion, use that; if not, find a club owner who is willing to strike a deal and let you have the club on an off night. Reserve the early part of the evening for VIPs and then charge admission later. Hand out complimentary drink tickets only to VIPs. Offer to promote the catering company along with your event in exchange for a discount on their services. Start planning and promoting your party several months in advance: book a location, contact a caterer, get the names of VIPs, send out invitations, and so on. If you plan this event well, it could open many doors for you.

PRIZE PATROL

People love freebies; it's human nature, so take advantage of it. But don't give stuff away too easily; make people do something-even just visit your Web site-to get it. Hand out postcards that prompt people to go online and register to win goodies such as T-shirts, CDs, or backstage passes. Give people extra incentive to sign your mailing list by holding a drawing. Make it public knowledge that you give away some sort of prize at every one of your shows. If you get more people to your shows that way, it's worth springing for prizes. We know of one band that landed an endorsement deal with a well-known percussion company and now hands out inexpensive rhythm instruments at their shows. Not only do audience members get to play along with the band, but they get a souvenir as well.

Advertising your contests creatively will expand your audience. Try coupons (Val Pak will include your coupon in 10,000 mailers for about \$600) or flyer inserts in the Sunday newspaper (about \$30 for 1,000 inserts). Just check the demographics and make sure these promotions will reach your target market. However, the bottom line is, if it says "free" on it, somebody will look at it. If they sign up on your Web site for a free CD and they like it, you've won a new fan.

CALL MY AGENT

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seriously. The two usual assumptions are that artists are "flakes" and that they must not be successful enough to do business with if they are calling contacts themselves rather than having an agent do it. It's not up to you to reform their opinions, though. A better tactic is to work within the narrow framework of what is acceptable to them: have a friend act as your agent, or pretend to be somebody else and act as your own agent.

It's best to have some trusty friends function as your mouthpiece. Coach them on how you'd like to be presented, give them a goal, and let them go at it. Even better, team up with a fellow musician who's at the same stage in his or her career, and trade off business negotiations. This gives you both incentive to do a good job for each other, because each of you is counting on the other; if you slack off, you can bet your friend will become resentful and slack off, too. This kind of relationship requires a high level of trust, maturity, and good communication, so choose your partner wisely. With some effort and prudence you'll discover why the adage "Two heads are better than one" rings true.

If you're daring, create a fictitious agent and play the part yourself. Of course, this is very risky: if you're found out, major repercussions could result (such as being blacklisted), because nobody likes to feel they've been played for a fool. However, many artists pull this off successfully. Never underestimate your ability to accomplish things on your own, but do bear in mind the consequences if things should go awry.

SINK OR SWIM

Working with a tight budget is a challenge, but it can be rewarding when, through your own ingenuity and persistence, you're able to launch a campaign that gets the attention of major labels, the press, and fans. Just keep in mind these three important points: (1) keep it regional; (2) know your target market; and (3) have a plan. Choose your promotions carefully; nothing is worse than spending your money on something that yields no response. If you're unsure about your strategy, poll your fans. If you're still unsure, don't do it—spend your money on a campaign that you *know* will get some attention. Finally, get your marketing plan together at least six months in advance. It takes time to organize things, and it takes a lot of foresight to execute several promotional tricks concurrently.

The promotional ideas presented here, while tried and true, represent just the tip of the iceberg. Be creative in your marketing approaches, and there's no telling what you can accomplish. Don't lose sight of your goals to get your music out to the public, to sell units, and to have fun doing it. And above all, be realistic and don't take on too much at once, or you may find yourself broke with a stack of products that you can't move. Good luck!

Lygia Ferra is a songwriter/producer whose first independent release as a solo artist, Strange Peculiar, is slated to hit retail stores by the end of the year. For more information, visit www.lygiaferra.com. Erik Hawkins is a musician/producer working in Los Angeles County and the San Francisco Bay Area. He recently started his own indie label, MuziCali Intertainment (www .muzicali.com).

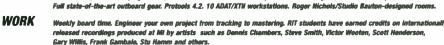


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Musicians—ask about our pro-audio CD recorders. Software publishers—ask about our high-volume multi-drive duplication systems. Dupe-It is sold and intended for backup and in-house design purposes only. Copyright laws must be observed. Dual Drive Model is pictured above.

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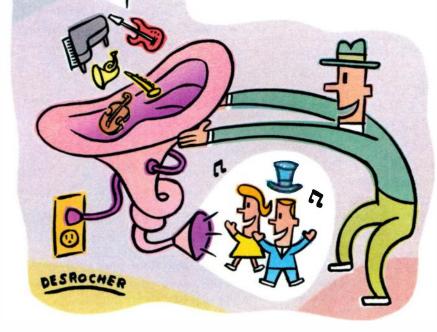


Music for Airports on Stage

Blending Bang on a Can's acoustic and electronic instruments live.

By Gino Robair

hen Brian Eno's tape-loop masterpiece Ambient 1: Music for Airports was released in 1978, it spawned an entire genre of music. But one thing Eno never imagined was that a group of musicians would have the audacity to transcribe and perform the four-movement work in such a way as to make it their own. In 1998, one of New York City's top new-music ensembles, the Bang on a Can All-Stars, did just that. Bang on a Can's artistic directors-Michael Gordon, David Lang, Julia Wolfe, and composer/performer Evan



Ziporyn-each transcribed and arranged one of the Airports movements, all of which would be brought to the stage by the six-piece All-Stars ensemble.

To keep the inherent austerity of Music for Airports from becoming boring, the group infuses its performances with subtle dynamic nuances. Transmitting this subtlety from the stage to the audience requires a high degree of sensitivity from the sound engineer. For the past four years, the Bang on a Can All-Stars have relied on the front-of-house talents of Andrew Cotton to make this happen.

"Even though they're playing through a sound system, the Bang on a Can All-Stars want to sound like an acoustic group," explains Cotton. "My job is to blend the instrumental textures together-starting with the acoustic instruments-and lift the mix from there to the volume it needs to be."

OUT IN THE HOUSE

Cotton prefers the adrenaline rush of engineering a live performance over doing a studio session. "With concerts," he explains, "you have five minutes to spend on a kick-drum sound rather than the five hours you might spend in the studio."

Beginning his music career as a trumpet player, Cotton gravitated toward recording studios and, finally, live sound reinforcement. His background as a concert musician keeps him in sync with the demands of amplifying acoustic ensembles, and his expertise has made $\frac{Q}{q}$ One day, there will be a monitor whose performance profile is based on science, not opinion.

A monitor whose compact size and power output will stun you; and whose low end performance and dynamic range will beat competitors costing three times as much.

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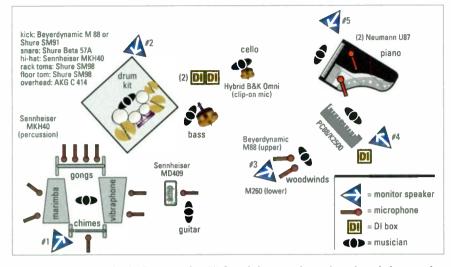


FIG. 1: The stage setup for the Bang on a Can All-Stars is kept consistent throughout their repertoire. This allows for stress-free transitions between pieces.

him a popular live-sound engineer with groups in the United States and Europe (including the Philip Glass Ensemble).

Andrew Cotton is also a member of Richard Nowell Sound in London, a U.K.-based sound-reinforcement company that specializes in contemporary classical and jazz concerts, as well as high-profile one-off events. "Most of the core work is for the Royal Festival Hall, the Barbican Center, and Contemporary Music Network tours," says Cotton. "Our company fills the gap between the larger rock-act sound companies and the little guys, but with engineers that are sympathetic to the needs of classical and contemporary music."

STAGE TRANSLATION

The particular challenges that Cotton faces with the Bang on a Can All-Stars result from their blend of classical chamber-ensemble timbres (such as clarinet, cello, piano, and acoustic bass) and contemporary instrumentation, including electric guitar, sampler, and drum set. The group actively commissions works that straddle the line between rock and classical performance practices. Although the All-Stars can play with great volume and abandon, they are in complete control of every aspect of their performance. This allows Cotton the option of avoiding the traditional tricks of the trade.

"What makes this group sound more like a classical ensemble is their control over dynamics," notes Cotton. "Therefore, I rarely use compression or gating on the instruments. My job is to retain the dynamics of the performers; if you compress them, you lose this. The only instrument I ever compress is the snare drum, and only if the hall requires it."

Any equalization he performs on the instruments is merely corrective, and the amount varies from venue to venue. "Basically, I add EQ to the instruments only to make them sound natural, since much of the music is written with an acoustic-ensemble sound in mind," he notes. "Some pieces, on the other hand, get more rock-style sound reinforcement than the classical-sounding pieces.

"The EQ settings depend on a number of variables, such as the sound system and the type of mixing desk I get. The biggest variable is the size of the venue, which determines whether it's merely sound reinforcement (as in a small hall) or full amplification that we're using."

To keep the amplified ensemble sounding as realistic as possible, Cotton adjusts the time alignment of the main P.A. system. "Using a stereo delay unit, such as a BSS 804, I delay the main stacks by 10 to 15 milliseconds, depending on the venue," he explains. "I do this to keep the image centered on the stage. So, even though it's loud, the results sound acoustic; the musicians sound like they're coming from the stage, not from the speakers."

AIRPORTS LIVE

Cotton began preparing for the live performance of *Airports* by attending the mixing sessions for the CD. "I wanted to see how the piece felt. During the sessions, I made mental notes and let the piece sink into my subconscious."

However, the stage setup proved to be a challenge when it came time to premiere the work in New York City's Alice Tulley Hall. To approximate the richness of Bang on a Can's studio version of *Airports*, the group brought in 12 extra instrumentalists and 11 singers for the work's unveiling.

But unlike the musicians, Cotton didn't get a rehearsal with the new pieces. "I got my hands on *Music for Airports* for the first time at sound check," recalls Cotton. "Before we began, we had to figure out the placement of the additional instruments,



The high sound quality of the Bang on a Can All-Stars' performances can be attributed to Andrew Cotton's skill at maintaining the musical nuances of the performers through the sound system.

with sight lines and visibility a major concern. The biggest trick was fitting the added players on stage so that the core group didn't have to be moved."

Scaling the large-ensemble setup of the premiere performance down to the core All-Star sextet, while maintaining the richness of the orchestration, was the next challenge the group faced. "Because such an elaborate setup is not practical for touring, we sampled tracks from the multitrack studio masters. The transcribers arranged keyboard parts for members of the group to play in each movement. We mainly sampled vocals, clarinet, and violin, and we chose some stock samples as well."

CLOSE QUARTERS

The input requirements for Music for Airports consist mainly of mics and miked amps (see Fig. 1). Although Cotton rents much of the equipment needed for the tour, he brings his favorite mics with him. These include Sennheiser MKH 40s for the percussion setup; a pair of Neumann U 87s for the grand piano; a Beyerdynamic M 88 for the clarinet and saxophone; a Beyerdynamic M 260 for the low end of the bass clarinet; a hybrid B&K mic-a small omni capsule mounted on a clip-on extension-for the cello; and a Sennheiser MD 409 for the guitar amp. In addition, the acoustic bass uses two DIs (one for each pickup), and the sampler (a rackmount Kurzweil K2500R controlled by a Kurzweil PC88) uses one.

Despite the amount of volume that the band generates on stage, the Bang on a Can All-Stars eschew any kind of baffling to separate the amps from the miked acoustic instruments. "Baffling doesn't work with the way the band tries to interact on stage," explains Cotton.

For maximum continuity during performances, the instrumentalists stay in the same places on stage from piece to piece—a benefit for the sound engineer. "We have a setup that works for everything," Cotton explains. "This makes for smoother transitions between pieces during a concert."

ROAD PREPARATION

Cotton estimates that he spends as much time preparing for a tour as the tour itself takes. He spends most of this preparation figuring out what items he will need for a specific set of compositions and the logistics of getting them to every show. Because of transportation issues, the group rents many of the larger items in each city that it visits.

"Normally, we rent the backline equipment, such as amps and percussion," Cotton explains. "I also have to make sure we get a big enough mixing board. We need at least a 32-input board-sometimes I'll request a 40-input board depending on the pieces we're doing. The group uses lots of percussion-a drum kit, gongs, vibes-and that gobbles up 14 or 15 channels right there. The mixing desks I prefer are the Yamaha PM4000 or PM3500, the Midas XL3 or XL250, and the Crest VX. We also need good monitors and a monitor engineer for each show. Fortunately, the sound world tends to speak the same language. For example, the word sampler is the same in most languages."

However, Cotton says that the Bang on a Can All-Stars prefer to carry their own sampler on the road rather than risk trying to find one in every city. "We use 112 MB of memory in the K2500, so we bring our own. It's too difficult to find a reliable one on the road." The K2500 programming is done by Cotton and Ziporyn.

Outboard effects are also key to the road show. "I rent at least two decentquality reverbs, such as a Lexicon PCM-70, -80, or -90, as well as a stereo delay, like the BSS 804."

GLASS ON THE ROAD

The operational strategies are quite different when Cotton tours with the Philip Glass Ensemble. "The Philip Glass Ensemble is more electronic based, with keyboards. Live, they have more of a 'created' sound than the Bang on a Can All-Stars, like on a CD."

On a recent tour to Europe with the Philip Glass Ensemble, Cotton was in charge of monitors. "Handling the monitor mix is as critical and complex as doing front-of-house. To begin with, the ensemble, which includes four singers and three woodwind players, uses 12 monitor mixes. In addition, during the performance, I look after the keyboards and computer. The group has three keyboardists using five MIDI controllers, and all the keyboards are sending MIDI information to offstage computers and synths. Everything is connected to a 7100-series Macintosh computer that is running MOTU's Performer and Digidesign's SampleCell. They use one main computer and have another for backup."



The Bang on a Can All-Stars relied on Andrew Cotton to bring across the subtlety of the group's recordings of Brian Eno's *Music for Airports* in a live setting.

REAL-TIME SCHEDULE

The Bang on a Can All-Stars' busy concert schedule keeps Andrew Cotton on his toes. A typical performance includes five compositions, many of which are premiere performances. Having works by a diverse set of composers, such as Arnold Dreyblatt, Glenn Branca, and Gavin Bryars, on one program stretches the ensemble, the audience, and the sound engineer to their limit.

As always, it's up to Cotton to make sure the defining details of each composition reach the ears of the listener. With the premieres, sound check becomes a critical time for the engineer. "That's usually the first time I get to hear the new pieces, so to make sure things work within those, we usually spend more time on them during the sound check."

One of the newer pieces performed by the All-Stars requires a more conspicuous use of signal processing than usual. In this case, Cotton's job is more performance oriented, something he obviously enjoys.

"In Michael Gordon's piece *I Buried Paul*, everything except the drums and Mellotron samples is heavily processed—mostly chorused and flanged," explains Cotton. "For example, the score requires that the clarinet be put through a phase shifter. Much of the exact nature of this processing was worked out during rehearsals. The composition is derived from the end of the Beatles' "Strawberry Fields Forever." The whole piece sounds like it's being played backwards."

Gino Robair is an associate editor at EM.

REVIEWS

Q\$8.1

The cost-effectiveness bar for keyboard synths gets raised a notch.

By Geary Yelton

ince its introduction in 1995, the QS8 has been Alesis's flagship synthesizer. Its noteworthy success in this era of modest keyboard sales has spawned a revised version, the QS8.1, which features a slightly larger display, an improved General MIDI bank, greater dynamic range, and dedicated buttons for selecting sequences and transposing the keyboard. The 64-voice QS8.1 combines an 88note, hammer-action, weighted keyboard with a fairly realistic acoustic

piano sound, making it a serious choice for pianists in the world of MIDI.

Like the original QS8, this year's model features 16 MB of samples in ROM; you can double the memory through the unit's two PC Card expansion slots. The QS8.1's sounds range from pianos and synth timbres to winds and strings, making it a wellrounded, general-purpose synthesizer. With a flash RAM card and the included *Sound-Bridge* software, it's even possible to load samples from your computer to create a customized sound set. Although there's no internal sequencer per se, you can store as many as 100 sequences on a card.

GETTING IN AND OUT

Along with stereo main audio outs, the QS8.1 comes with a pair of auxiliary outputs and a rear-panel headphone jack (see Fig. 1). It's apparently assumed that the aux outputs will be used for routing sounds to an external effects

136	Alesis QS8.1
146	
	Tracer <i>DC-ART 32</i> 3.06 (Win)
152	BLUE Blueberry
158	Cycling '74 <i>Pluggo</i> 1.04 (Mac)
166	Generalmusic Equinox 61
174	HHB Radius 20 and Radius 30
182	Aardvark Aark 20/20+ (Win)
188	
	Etek NoteMix MA 400
198	Quick Picks: USB <i>Raricussions</i> (Mac/Win); PreSonus MP20;
	MicroBoards AudioWrite Pro
	(Mac/Win); Beatboy Ramon Yslas Contemporary Percussion and Drums (Mac/Win)



Like the hot-selling QS8, Alesis's QS8.1 synthesizer is chock full of features, such as 16 MB of waveform ROM highlighted by excellent piano sounds, 64-note polyphony, a very nice 88-key weighted action, two expansion card slots, and good effects. Enhancements include a slightly larger display, an improved GM bank, greater dynamic range, and Seq Select and Transpose buttons.

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processor, because they bypass the internal effects. The MIDI In, Out, and Thru jacks are supplemented by a serial data port for direct connection to a Macintosh or PC. A sustain-pedal input is supplied, along with two jacks that accept assignable footswitches or continuous-controller pedals. The QS8.1 ships with a sustain pedal, a standard AC power adapter, two CD-ROMs, and documentation.

This synth offers a built-in ADAT Optical digital output that can send all four QS audio outputs directly to any Lightpipe-equipped device without leaving the digital domain. That's a big advantage if you have, for example, an ADAT or Lightpipe-equipped audio card or digital mixer. A 48 kHz In port allows you to synchronize the QS8.1's digital clock with an Alesis BRC remote controller. You won't need this input if you only use one ADAT; the Alesis tape deck will automatically sync to the QS8.1's digital output.

ON THE SURFACE

The keyboard, which is manufactured for Alesis by Fatar, features real hammers to emulate the feel of an acoustic piano keyboard. I prefer a lighter weighted action-and, arguably, the QS8.1 does this as realistically as any of its competitors. Speaking of weight, even though it has 88 weighted keys with hammers, the entire synth weighs 60 pounds, which is less than most weighted-keyboard instruments. The keyboard generates both Velocity and Release Velocity data, providing a level of flexibility that most synth keyboards lack. Its Aftertouch is less flexible-it generates Channel Pressure but not Key Pressure; the synth engine, however, can respond to Key Pressure from an external source.

The pitch-bend and modulation wheels are located just above the left side of the keyboard (rather than all the way to its left), making the entire keyboard shorter than most 88-key units. This arrangement is quite comfortable, requiring less of a stretch when playing higher notes. The wheels themselves are a bit smaller and have a shorter throw than most-less than 60 degrees. Although this shorter throw makes it easier to go full tilt, the truncated range provides less subtle control of pitch and modulation depth than I'd like. Of course, any modulation destination can be controlled with either wheel.

To the right of the wheels, the volume slider is flanked by four assignable sliders, one of which lets you change parameter values. Moving a slider that has a control function assigned to it brings up a small bar graph in the display that reflects the slider's position. These sliders are useful not only for editing QS8.1 sounds on the fly, but also for controlling external MIDI instruments.

Although not blessed with the overwhelming selection of knobs and sliders you see on some recent synths, the QS8.1's user interface is clear and straightforward. A six-button cluster to the left of the display consists of the Edit Select button, which determines whether you're editing Programs (single sounds), Mixes (multitimbral combinations of Programs), or effects; a pair of Page buttons that scroll through pages and parameters and set the MIDI channel; two Value buttons; and the Save button.

To the right of the display, a couple of Play mode buttons let you select Programs or Mixes. Below these are a pair of dual-function buttons that you can use to step forward and back through Banks, compare an edited Mix or Program to the original, and access the global parameters.



PROS: Well-designed weighted keyboard with hammers. Very good piano sound. Serial port for direct computer connection. ADAT Optical output. Expandable via card slots. Doesn't weigh a ton. A lot of synth for the money.

CONS: No floppy drive. No arpeggiator. Nonresonant, lowpass-only filters. Boostonly 2-band EQ. Short-throw pitch and modulation wheels. Some "acoustic" Programs sound unrealistically electronic in multitimbral Mixes.

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Farther to the right are two rows of buttons whose functions depend on whether the unit is in Play, Mix Edit, Program Edit, Effects, Edit, or Drum mode. The top row contains 13 buttons numbered in increments of ten, from 00 to 120; below it is a row of ten buttons numbered from 0 to 9. With these two rows of buttons, you can select the Sound Group, which organizes Programs by type (pianos, brass, and so on); choose individual Programs and multitimbral Mixes; assign MIDI channels for Mixes; access individual parameters and the Function Groups into which parameters are organized; edit the Sound Layers for Programs; route sounds to the effects buses; and the like. (I'll explain the details of this architecture shortly.)

Finally, on the far right of the display, along the bottom row of buttons, are the Transpose button and the Seq



Select button, which selects a PC Card– based sequence for playback.

The QS8.1's user interface is reasonably intuitive, and I had few problems finding my way around the unit. Learning was made even easier by the clear, concise, and well-organized reference manual, which includes a handy foldout diagram that labels and briefly explains everything on the synth's front and back panels. The manual's only glaring fault is that its index needs to be more comprehensive. You also get a Quick Reference Guide that contains enough information to get you up and running if you're averse to opening the manual, and two charts listing the factory-set Programs and Mixes.

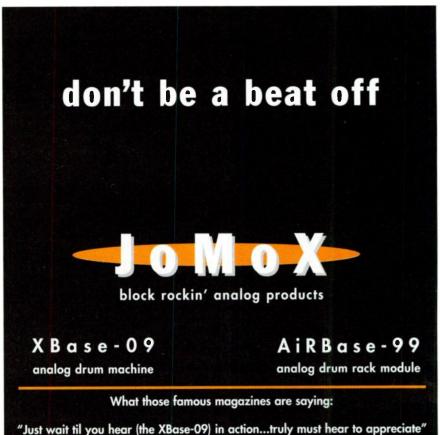
BASIC ARCHITECTURE

Like most synthesizers, the QS8.1 plays individual Programs as well as Mixes. Each Program contains up to four individual Sounds that may be layered or split. Each Sound consumes one Voice of polyphony, so fewer notes are available in Programs that stack Sounds. You can play as many as 64 or as few as 16 simultaneous notes.

In Mix mode, the QS8.1 can receive on 16 MIDI channels simultaneously. Programs within Mixes are easy to edit individually and are even more easily replaced. A Mix may contain up to 16 Programs from any Bank, which can be split or layered across the keyboard. In fact, you can split the keys into as many as 16 different zones—a rare feat among MIDI keyboards—and the zones can overlap. The keyboard can therefore send on 16 MIDI channels at the same time, with separate volume, panning, and transposition for each part. You can, of course, trigger selected Programs from an external sequencer while other Programs are triggered from the keyboard.

Each Bank contains 128 Programs and 100 Mixes. There are three Preset banks, a User bank, and a GM bank, totaling 640 Programs and 500 Mixes. Filling both PC Card slots can provide two additional banks of 128 Programs and 100 Mixes each, for a possible total of 896 Programs and 700 Mixes.

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to another, the QS8.1 carries the sound of the previous Program over without interruption, which is the way it *should* work (but doesn't with some synths). It's even possible to hold a key, change Programs, and sustain the note from the first Program. When you change Mixes, however, the sustained note stops sounding, with approximately a half-second delay before new notes sound. Considering the numbercrunching that goes on in that halfsecond, this delay is acceptable.

Drum mode lets you assemble custom drum kits. You can assign up to ten samples to each Sound—so with four Sounds per voice, you have up to 40 samples in a drum Program. The pitch, volume, panning, and effects routing of each sample can be determined individually, as can the range of keys that trigger a sample. You can assign a drum sample to as many as four keys without transposing the sample, allowing flams and fast retriggering. LFOs and envelopes are disabled in Drum mode except for amplitude decay.

UNDER THE HOOD

Q S 8.1

The QS8.1 is a 16-bit, 48 kHz sampleplayback, subtractive synthesizer whose lineage can be traced back to the original Alesis QuadraSynth (reviewed in the July 1994 EM). Up to four Sounds can be shaped with envelope generators and nonresonant lowpass filters and then combined into a single Voice (see Fig. 2). Although it's no different from the way most synths combine sounds, this process of adding complex timbres and then filtering them is what Alesis calls "Composite Synthesis."

Filter, amplitude, and pitch each have a dedicated envelope generator and dedicated LFO per layer. The envelopes are a bit more complex than the customary ADSR: in addition to

and the second se	
Keyboard Action	88-key, weighted hammer
Controllers	(1) pitch wheel; (1) mod wheel; (1)
	volume slider; (4) programmable sliders
Synthesis Type	subtractive sample playback
Polyphony	64 notes
Multitimbral Parts	16
ROM Programs/	512/400 (three preset banks and
Multitimbral Mixes	one General MIDI bank)
User RAM Programs/	128/100 (one bank)
Multitimbral Mixes	
Sample Resolution/	16-bit linear/48 kHz
Playback Rate	
Effects Processors/Sends	1 (stereo)/4
Effects Types	Pitch: mono and stereo chorus, mono
	and stereo flanger, resonator, pitch detune
	Delay: mono, stereo, ping-pong
	Reverb: plates 1 and 2, room, hall, large, gate, revers
Internal Sound Memory	16 MB
Expansion	(2) PC Card (PCMCIA) slots; accepts 8 MB flash
	RAM cards (samples, Programs, Mixes,
	sequences), 8 MB QCards (samples, Programs,
	Mixes), or 512 KB SRAM cards (sequences only)
Audio Outputs	(2) ¼" unbalanced master L/R; (2) ¼"
	unbalanced aux; (1) ADAT Optical (carries
	same signal as four analog outs);
	(1) ¼" stereo headphone
Other Ports	(2) ¼" footpedal jacks; (1) ¼" sustain-pedal
	jack; MIDI In, Out, and Thru; Mac/PC-compatible
	serial interface; 48 kHz BRC sync input
Power Supply	120 VAC internal
Dimensions	19" (W) x 8.5" (D) x 1.75" (H)
Weight	60.9 lbs.

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FIG. 1: The QS8.1's rear panel is identical to that of the original QS8. Besides the usual MIDI, foot controller, and analog audio ports, it features an ADAT Optical digital output and a 48 kHz sync input for use with an Alesis BRC remote controller.

attack, decay, sustain level, and release time, you can control the initial delay time and the sustain decay time.

The unit's Amplitude Range function lets you select from 13 Velocity curves to customize the feel of the keyboard. Velocity curves can be specified for each Sound in a Program, or for each sample when in Drum mode. (You set up Velocity switching and crossfading in the same way.)

Alesis provides a wealth of modulation routings. Most of the QS8.1's modulation sources have dedicated routings to logical destinations, such as Aftertouch to pitch-LFO depth for controlling vibrato, and Velocity to filter frequency for controlling brightness. You can turn these routings on and off. Half the modulators can be quantized so that they are stepped rather than smooth.

When you want to go a bit further than hardwired modulation, matrix modulation allows custom modulation routings. Alesis provides six generalpurpose modulators for controlling as many as six parameters of any number of sources, and a single modulator can be routed to many destinations. For example, Release Velocity can change LFO speed, LFO depth, and release times for pitch, filter, and amplitudeall at the same time. There are 18 possible modulation sources and 32 modulation destinations. Sources range from pedal inputs and controller sliders to a trigger-rate follower, which responds to how quickly notes follow one another. Destinations include effects send, envelope times, and portamento rate.

SUGAR AND SPICE

The QS8.1's stereo effects processor, which uses the same chip as the Alesis QuadraVerb 2 (reviewed in the June 1995 EM), offers five basic algorithms called Configurations. An Effects Configuration is stored as part of each Program. Because only one Configuration is possible at a time, all the Programs in a Mix share a common effects setup. This means that if four Programs in a Mix each use a separate Configuration, you must decide on one Configuration for the entire Mix. However, each Sound within a Program can be assigned its own Effects Send, allowing each Sound its own effects depth. Of course, the same holds true of Programs in a Mix: you can process each to any depth desired or bypass the effects completely. Although it isn't obvious, you can also use buses 2, 3, and 4 as inserts if you use Configuration 5, but not with other Configurations.

Each Configuration is an arrangement of effects types and effects routings. Within each Configuration are more effects types than the Configuration name indicates. For example, 1 Reverb actually provides three Pitch effects, four Delay effects, and one Reverb, all available at the same time (see Fig. 3). Within these categories are resonator, detune, two types of chorus, two types of flange, three types of delay, and seven types of reverb. Another Configuration-2 Reverbs-provides one Delay, two Pitch effects, and, of course, two Reverbs, each with a choice of seven Reverb types. The three remaining Configurations are Lezlie+ Reverb, Reverb+EQ, and Overdrive +Lezlie.

Two additional Configurations offer equalization for the effects sends as well as the synthesizer's entire output.

The equalization on the QS8.1 is less powerful than that found on many other contemporary synthesizers: there's no cut, only boost, and you get just four EQ parameters—low- and high-frequency range, and low- and high-frequency gain.

THE PALETTE

QS8.1 Programs are organized into 11 different categories (Sound Groups) ranging from Pianos and Chromatic to Rhythm/FX and Drums/Percussion. The selection covers the bases for most styles of popular music. I like the general character of many of the unit's sounds, including quite a few acoustic instruments. Nevertheless, the acoustic and electric instrument sounds all come across with a slight but noticeable electronic feel. In this respect, the QS8.1 is not a naturalsounding synth.

Whenever I'm involved in a discussion about which weighted-action synthesizer combines the best in piano sound and playability, someone always mentions the QS8. Alesis knows it's got a good thing going: the QS8.1 features an abundance of excellent acoustic and electric piano sounds, including the same acoustic piano samples as the original QS8's.

The Chromatic category includes some very good tuned percussion, harpsichords, and clavinets. There are enough Hammond and pipe organ samples—including Keith Emerson's Hammond C3—to satisfy almost anyone, from rockers to church music directors. The acoustic and electric guitars are also quite good, but you may want to tweak the sustain on most of them for added realism. Some of the QS8.1's synth basses are acceptable, but the bass guitars are unimpressive.

The string ensembles sound pretty good, but to my ear most of the solo

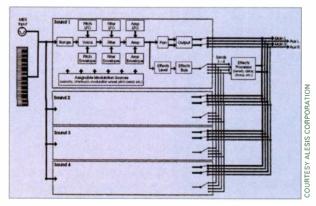


FIG. 2: The Composite Synthesis voice architecture is straightforward. Up to four Sounds can be processed with nonresonant lowpass filters; modulated by dedicated pitch, filter, and amplitude envelopes; panned; and combined into one Voice.

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strings fall flat. With one or two exceptions, I also didn't care for the brass. Among the winds, the saxophones are unusually good, and there's a huge selection of panpipes.

The Programs in the Synth classification are, for the most part, excellent and quite useful. The best are slowly evolving pads and Keith Emerson-style leads. In fact, many were sampled from Emerson's modular Moog and then shaped to sound like his signature timbres, with names like AquaTarkus, FmDBgining, and Fanfare GX. Several of the QS8.1's fine synth sounds are replications of popular timbres from other manufacturers' synthesizers.

Many of the acoustic sounds suffer from too many samples being squeezed into 16 MB of memory. A lot of loops happen earlier than they should, and many loops are too short. All synthesizers have this shortcoming to some extent, but interestingly, I found these problems to be most obvious when I played the QS8.1 multitimbrally, as an ensemble. In that context, it just doesn't sound as convincing as some of its competitors. I was surprised by this limitation: usually a sound's weaknesses become more apparent when it is used as a solo instrument.

ROOM TO GROW

Up to 16 MB of storage can be added to the QS8.1 by inserting PC Cards into the two expansion slots. You can choose from three kinds of PC Cards: flash RAM, SRAM, and Alesis QCards.

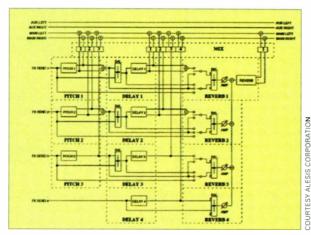


FIG. 3: This typical effects Configuration, 1 Reverb, gives you a lot more than reverb. It actually contains three pitch effects, four delay effects, and a reverb. Furthermore, the pitch effects can be stereo or mono chorus or flanger, a pitch detuner, or a resonator; the delay can be mono, stereo, or ping-pong; and you get a choice of several reverbs, including gated and reverse.

Flash RAM cards store sounds sampled from external sources. You can organize sample files on your computer with Alesis's *SoundBridge* software and then port them over to the synth for storage in flash RAM. The card can also store a bank of Programs and Mixes and 50 sequences; its maximum total storage is 8 MB. (You can use flash RAM cards with more memory, but the synth will recognize only 8 MB.)

QCards are 8 MB ROM cards containing samples, 128 Programs, and 100 Mixes, allowing expansion of the QS sound library. Several different QCards are available, including *Classical Instruments Plus, EuroDance, HipHop Grooves & Instruments*, and *Vintage Keyboards*. In addition, you can use 512 KB SRAM cards, which hold eight extra Banks of Programs and Mixes but no samples.

A BUNDLE OF GOODIES

The QS8.1 comes with two cross-platform CD-ROMs, which contain software for both the Mac OS and Windows. The *Alesis Synthesizers Software Pack* disc offers MIDI files and samples from a variety of third-party companies, as well as Banks from several Alesis synthesizers; it also includes Emagic's *MicroLogic AV* digital audio sequencer. The *QCard Audio Demo* features Program Banks for all the available QCards and the factory-preset User Bank. Audio tracks contain music recorded with all the various QCards.

Both CD-ROMs also contain *Sound-Bridge*, an application for Mac and Win-

dows platforms that converts audio files, Program Banks, and MIDI sequences into a format that can be downloaded to SRAM and flash RAM cards in the QS synth's expansion slots. Supported file formats include WAV, AIFF, Sound Designer, and Mac OS system sounds. It's even possible to transfer complete SampleCell instruments, including sample data, keymaps, and loops. Because you can store both sample and sequence data in flash RAM, you can load a sequence's audio tracks as sample data

(assuming that the files can fit onto an 8 MB card).

Both of the CD-ROMs also contain *Freeloader*, an application for downloading MIDI files from your computer to expansion cards in the QS8.1. In addition, the discs offer assorted demo and support software from Emagic, Mark of the Unicorn, and Steinberg, among others.

MORE FOR THE MONEY

The QS8.1, like its predecessor, is destined to be a favorite among keyboard players who want a one-stop solution for less than two grand. The new synth has a lot to offer for the money, including a good keyboard feel, an intuitive user interface, and the ability to load your own samples. In fact, you may want to buy one just for its hammer action and piano sounds. The QS8.1's other onboard sounds are well rounded enough to cover most styles of popular music. If those sounds aren't enough for your needs, you can expand the range by adding PC Cards.

The QS8.1's designers clearly cut some corners to save money, but for the most part they cut the right corners. The synth has lowpass-only, nonresonant filters, but you'll probably never notice the difference. A big chunk of waveform ROM is filled with piano samples, at the expense of some of the other acoustic timbres. The 2-band EQs have just boost and no cut; a more sophisticated EQ would be nice for tweaking custom programs, especially when you want to even out the response across the keyboard. I also wish the QS8.1 came with a floppy drive for reading sequences, because then floppies could be freely swapped between the synth and a computer.

As a master MIDI controller, the QS8.1 has much to recommend it. The throw of the pitch and modulation wheels could be longer, but that aside, the synth is easy to work with, you get lots of zones, and the sliders are handy.

There's no real breakthrough technology here, but all the functions are implemented in a dependable, triedand-true fashion. The QS8.1 is a synthesizer for the people—and just as with the original QS8, Alesis is likely to sell a ton of them.

Geary Yelton has barely enough time to play with his toys, let alone write about other people's. But that doesn't stop him.

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DC-ART 32 3.06 (WIN) Professional audio restoration tools for a reasonable price.

By Scott R. Garrigus

hether you do your recording in a professional environment or a home-based studio, noise can be a problem. The source of the trouble can be tape hiss, clicks and pops from bad cables, low-frequency room rumble, ground-loop buzz, or other common culprits. By taking some precautions, you can prevent most noises, but when some still make it into your masterpieces, *Diamond Cut Audio Restoration Tools 32 (DC-ART 32*) may be the cure you need.

Originally designed for restoring old phonograph recordings, *DC-ART 32* (created by Diamond Cut Productions and published by Tracer Technologies) has evolved into a professional application that can remove most types of noise from just about any audio material. The program provides a variety of sophisticated filtering algorithms that include Impulse Noise, Continuous Noise, and Dynamic Noise filters. There are also Harmonic Reject, Median, and Average filters, along with a wide range of equalization and dynamics-processing tools.

I installed *DC-ART 32* on a 300 MHz Pentium II and tested the program with a number of audio recordings. I successfully removed hiss from some of my old analog tapes and cleaned up the clicks and crackles from various albums and 45s. But even though the results I got were excellent, actually working with the program was a bit of a chore.

HOW IT HANDLES

When you open a file in DC-ART 32, it is displayed in a window with two panes. The source waveform is shown on top and the "destination" on the bottom (see Fig. 1). The panes can be synchronized so that if you scroll in one, the other follows. (Synchronization doesn't apply to zooming, though.) Despite appearances, you can have more than one source/destination file set open simultaneously, each with its own window. Unfortunately, when you close DC-ART 32, it doesn't remember which files you had open or the various settings (such as window position) that you applied to them.

When you process a file, *DC-ART 32* reads the data from the source waveform, applies the filter, and then displays the result as the destination waveform. At that point, you can apply

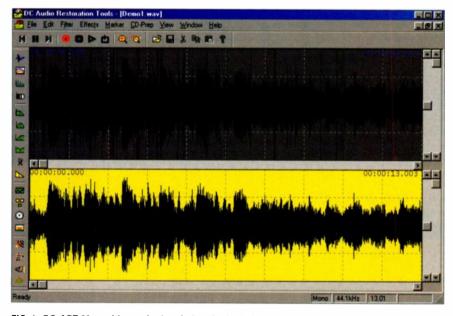


FIG. 1: *DC-ART 32* provides a single window for both the source and destination waveforms. The source is displayed in the top pane and the destination in the bottom.

DC-ART 32 Minimum System Requirements 80486 CPU; 16 MB RAM; Windows 95 or NT; sound card

additional filtering to the destination waveform in case you have more than one type of noise to deal with. Initially, this seems like a great way to work. Your source file stays untouched, and you can see, hear, and compare the results easily. The problem is that any subsequent processing to the destination waveform can't be undone. The Undo command in *DC-ART 32* works only with basic functions such as cut, copy, and paste. And if you select and process the source waveform again, your previous destination waveform results are overwritten.

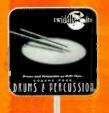
To manipulate the results of a previous filter process and still have the option to undo your actions, you have to use the Make Destination the Source function. This simply copies the destination waveform from your current window to the source waveform of a new window. After running a few filter processes, your workspace becomes cluttered with a lot of windows, which is especially cumbersome because you can't name the windows unless you save them to disk.

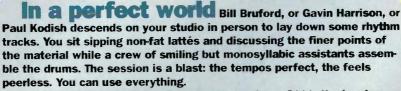
DC-ART 32 does provide some features that make its operation a bit easier. "Nonmodal" dialog boxes let you keep one filter's dialog box open while you access another filter dialog. You can also apply any of the program's other functions while any number of dialog boxes are open. This can be a real time-saver, but you have to use this feature judiciously: too many open dialog boxes just add to the aforementioned clutter.

To help streamline processing tasks, DC-ART 32 lets you store presets for any of its filter functions, including all the noise-removal and equalization filters. The program also has a wide variety of default presets that cover typical analog-tape and phonograph noise prints, European and American buzz filters, vintage radio equalization curves, and more. One addition I'd like to see is the ability to string several presets together for quick batch processing.

DC-ART 32's absence of keyboard shortcuts will likely displease most







But then Bill/Gavin/Paul slips you an envelope. S***, the invoice: You open it with trembling fingers to read the words 'No Charge,' and you look up into Bill/Gavin/Paul's smiling eyes and waving finger as he says "But next time, man, you're gonna play on my record, OK? Deal?"

You both laugh, knowingly, as only fellow and equal artists can do. And then sushi arrives.

In an imperfect world you've just spent another lousy hundred bucks on another sample CD. It's got some loops you can use but, heck, you've heard a lot of these a zillion times on the radio. When did this come out? 1997. You curse the clerk at BigBucks Megastore.

But by lunchtime you've got a groove loaded and found a workaround to incompatibility issues between your sequencer and digital audio package. The tempos are kind of limp, which you can fudge. Still you wish you could change the sound of the snare. You wish you could program decent drums yourself. At the end of a hard day you fill in some sample clearance forms and toss a pizza into the microwave.

a smart world you reach up for the Jewel-Case marked Twiddly•Bits Vol 8 MIDI BreakBeats and flip open the lid. You insert the disk into your PC (or Mac), open Cubase (or Cakewalk, or Performer..), and load a file. You Solo one of a dozen tracks containing exuberant 2-bar drum 'n' bass loops, and hit play. Geez! Are those sounds coming out of my gear? Still in shock, you loop the groove over 16 bars and snip out the final bar, substituting it for a death-defying fill currently residing on Track 14. You switch to an ambient drum kit. Yea! the groove becomes even more intoxicating, especially when you hit the gas and effortlessly take the tempo up to 155. The track already has life, energy, and your own slant on things. You wonder how many grooves are on this disk. You look. About 700. For how much? \$39 bucks? Why haven't I bought any of these before? Food comes in and you don't even notice.



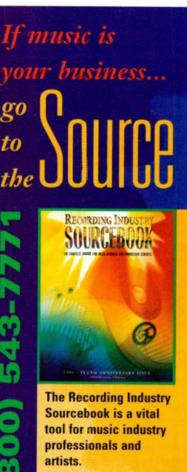


Drummers we use include Gavin Harrison (Level 42, Incognito), Bill Bruford (Yes, King Crimson), Shane Meehan (Us3), Alfredo Dias Gomes, Paul Kodish (Chemical Brothers, Appollo 440), Chronic Music's The Beat Professor, Ron E. Beck, Hugo Degenhardt (Womack & Womack, Steve Hackett), Dave Spiers, Andrew Small, and T.E.T (Anthill Mob, Doug Wimblsh), and Al Eaton (Queen Latifah, Ice-T). All disks are sold license-free

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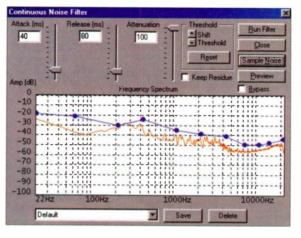


FIG. 2: The Continuous Noise filter shows the frequency spectrum of sampled noise along with a threshold graph with individually adjustable inflection points.

power users. The program provides only a few shortcuts covering basic functions such as opening a file and copying, cutting, and pasting data. There's no way to add or customize your own shortcuts. Most functions must be accessed with the mouse or with multiple key combinations (such as pressing the Alt key, then the first letter of the menu you want to access, and then the first letter of the function you need).

FILTERS FOR ALL

DC-ART 32 provides a nice set of filters for dealing with many types of audio disturbances. Impulsive noises, such as pops, ticks, and clicks, are handled by the Impulse Noise filter. It scans an audio file for sharp spikes in the waveform and replaces them with an approximation of what the signal might have contained during that brief occurrence. You can set the Threshold (the signal level that a spike has to surpass in order to be identified and removed), the Size (the minimum number of audio samples a spike has to occupy before it's removed), and Tracking (a parameter that determines how well the program distinguishes between spikes and percussive musical content). During my tests, I was able to remove all of the impulse noises from my LPs without any audible side effects. That was very impressive.

Continuous noises such as tape hiss, low-frequency rumble, hum and buzz, and other sustained background disturbances are dealt with by the Continuous Noise filter, the Dynamic Noise filter, or the Harmonic Reject filter. Each has its strengths and weaknesses. The Continuous Noise filter is the most effective of the three, but it's also the most difficult to use. You begin by highlighting a part of the audio waveform that contains only the noise you're trying to remove. Then you click on the Sample Noise button, and DC-ART 32 displays the frequency spectrum of the sample (see Fig. 2). The red line in the graph represents the frequency spectrum of the noise, and the blue line represents the threshold.

You can adjust the en-

tire threshold using the Shift and Threshold buttons, or change the threshold for only part of the spectrum by dragging the dots up or down along the blue line. *DC-ART 32* breaks up the spectrum into 1,000 separate bands. Each dot represents 100 of those bands, and you can change the center frequency of each dot by dragging it left or right. You can't add more points along the threshold for more accurate adjustments, but even so, the filter performs quite nicely. As you fine-tune the threshold, you can have the program



PROS: Filters for all kinds of audio disturbances. Wide range of equalization functions. Professional-quality output. Nonmodal dialog boxes. Helpful demo file and excellent documentation. CONS: No DirectX support. Nonstandard Undo function. Very little workspace customization. No definable keyboard shortcuts. Supports only WAV file format. CIRCLE #438 ON READER SERVICE CARD play a continuous preview of the file. You can also listen to the part of the signal that's being removed by selecting the Keep Residue function.

It takes quite a bit of trial and error to get the Continuous Noise filter just right so that it doesn't cut out an important part of the signal or introduce unwanted artifacts. Most of the time, the filter does its job very well, but if all you need to do is remove some highfrequency hiss, the Dynamic Noise filter might be a better choice. This filter works on the same principle as basic analog single-ended noise reduction. It provides a movable lowpass filter that attenuates high frequencies only when there is no high-frequency music content present in the signal.

The Harmonic Reject filter (more commonly known as a "comb" or "multiple notch" filter) lets you eliminate noises that are centered around a certain frequency, such as 60 Hz groundloop hum or harmonically rich buzzing caused by radio frequency interference. You can adjust the fundamental filter frequency from 5 Hz to 5 kHz and the attenuation from 0 to 100 dB. For buzzing noises, you can choose to filter odd or even harmonics and set the number of harmonics from 1 to 500.

DC-ART 32 also provides Median and Average filters. According to the manual, these filters have no analog equivalent. You simply set a number of samples (3 to 20 for Median and 2 to 100 for Average), and then let the filter do its thing. These

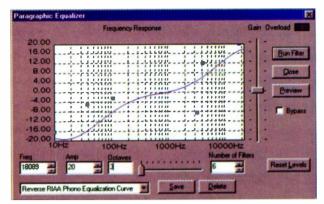


FIG. 3: The Paragraphic Equalizer is extremely powerful because it lets you define up to ten filters, each with separate frequency, amplitude, and bandwidth settings.

filters work by taking the median or average values from the source waveform, within the sample window you defined, and passing those values to the destination waveform. I found these filters to be particularly effective in reducing the crackles on my LPs.

EQ AND DYNAMICS

In addition to its special noise filters, *DC-ART 32* provides a wide range of

equalization and dynamics-processing tools. Its arsenal includes lowpass, bandpass, highpass, and notch filters. Each provides the usual frequency parameter along with a filter slope that can be set to 6, 12, or 18 dB/octave. Of course, instead of filter slope, the notch filter has a bandwidth parameter that can be set from 0.01 to 1.99 octaves. A basic 10-band graphic EQ with a ± 12 dB range for each band is also

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real: +1-603-654-6427 (fax: 6107) virtual: www.earthwks.com postal: box 517 Wilton NH 03086 provided, along with a paragraphic equalizer (see Fig. 3).

I found these to be quite powerful tools—more powerful than the same functions found in other professional software, because you can define up to ten filters, each with separate frequency, amplitude, and bandwidth settings. What's more, you can adjust each filter by typing in values or simply dragging points along a graph. This allows you to create some extremely impressive equalization curves.

The Dynamics Processor is not quite as useful. It provides basic compression, expander/gate, and de-esser functions. You can set the threshold, ratio, attack, and release for the compressor and set the threshold and ratio for the expander/gate. The de-esser can only be activated or deactivated; it has no frequency or bandwidth controls.

OTHER TOOLS

To correct speed variations in analogtape and vinyl recordings, *DC-ART 32* has a Change Speed function that lets you vary the pitch/speed of audio over time. What's especially cool about this feature is that you can define the changes graphically by dragging points to create a curve (see Fig. 4). The Gain Change function operates in a similar manner and allows you to define linear, logarithmic, or curved gain slopes over the course of a recording.

DC-ART 32 also has reverb and Virtual Valve Amplifier effects. The reverb provides only basic room-size, reflections, decay, and early-level settings. It sounds quite good, but it can't compare to dedicated reverb plug-ins.

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FIG. 4: With the Change Speed function, you can vary the pitch/ speed of audio over time by simply dragging points on a graph to define a scaling curve.

The Virtual Valve Amplifier, on the other hand, is very cool. It provides drive, operating-point, detail, and mix parameters along with a Spectrum setting. You can even define the tube type/configuration for Triode (12AX7, 12AT7, or 12AU7), Pentode (6EJ7), 2-Stage Class A, 2-Stage Class AB, Exciter, and Transformer Class AB. My only gripe with *DC-ART 32* as an effects tool is that there is no DirectX support. The internal effects are a nice addition, but I'd much rather be able to use my own plug-ins with the program.

Finally, DC-ART 32 has some other useful basic functions like Reverse File, Markers, Crossfade, Make Waves (a testsignal generator), and Gain Normalize. There's also a CD-Prep menu with functions that help you find and mark silent passages automatically and break files into smaller chunks in preparation for CD burning. It provides no actual CD-burning functions, though.

FINAL PROCESS

DC-ART 32 can be cumbersome to use. to say the least. Among the major annoyances: the program supports only the WAV file format, Undo doesn't function in the standard manner, and you have to open a new waveform window for each filter process. Luckily, the program comes with a very helpful demonstration file and excellent documentation; the manual is one of the best I've seen. Not only does it include information about all the features of the program, but there's also a tutorial for every feature. If you follow these instructions, DC-ART 32's esoteric working procedures won't get in

the way as much.

Even though DirectX support is lacking, DC-ART 32 is an excellent set of tools for dealing with a wide variety of audio blemishes. Most of the tools provide a good range of parameters, and the output quality is very professional. Furthermore, at \$199 the program is one of the least expensive in its category. For all the power it provides, I have to say that DC-ART 32 would be a valuable asset to any audio engineer. @

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L U BLUEBERRY A high-quality condenser mic for top-dollar vocals.

By Myles Boisen

fter years of working behind the scenes restoring vintage mics, Baltic Latvian Universal Electronics (BLUE) has stepped into the spotlight as a microphone manufacturer. Although best known for its deluxe—and pricey tube Bottle mic (a design inspired by the Leipzig 7151 and Neumann CMV 3, but with completely redesigned circuitry), the company offers three other large-diaphragm condenser mics: the Cactus, Mouse, and Blueberry. Each combines exquisite, hand-built quality with unforgettable, eye-catching style.

At \$1,295, the Blueberry is BLUE's most affordable mic to date and the model most within reach of personalstudio budgets. A solid-state microphone, it offers only a cardioid polar pattern and provides neither an attenuation pad nor a highpass filter.

GOT THE LOOK

You can tell that the Blueberry is something special even before you open the box. The mic's hefty wooden case is built to old-world quality standards; except for the logo on the lid, it looks like an antique jewelry box.

Inside, the Blueberry rests in regal dignity on a contoured bed of crushed blue velvet, sewn by hand onto foam padding. The microphone body is a rectangular block of powder coatfinish aluminum (you can guess the color), with a unique oval grille mounted on top inside a platinumcolored yoke. A raised brass "Blue" logo and classy engraved script ("The Blueberry") decorate the front of the mic.

ACCESSORIES

The bottom of the Blueberry has a recessed XLR connector alongside a standard threaded socket for attaching the mic to a stand. I found it easiest to mount by screwing a boom arm into the mic with one hand, rather than struggling to twist the mic onto a stand with both hands while having to worry about dropping it. The recessed mount contributes to the Blueberry's sleek lines, but it is inconvenient at best: the lack of a joint or swivel makes angling the mic relative to the boom arm impossible—which in turn makes the Blueberry difficult to position on some sound sources. (BLUE has announced that a swivel mount will ship with future Blueberries.)

Fortunately, the optional S1 (\$259) and S2 (\$159) shock-mounts allow angling of the mic in any direction. The S2 was designed as a more affordable version of the S1. It attaches directly to four knurled nuts on the mic body via elastic, tension-mounted suspension loops. The S1 is a two-piece affair—essentially, it's an S2 that attaches to a rectangular section that slips around and clamps onto the mic.

Both shock-mounts are well constructed and hold the mic securely. However, the suspension loops are a bit bouncy, and although the assembly does make it possible to angle the mic in any direction, its size presents additional difficulties for getting the mic into tight places (for example, on a snare drum). But despite these reservations, I recommend that you get one or the other shock-mount, due to the limitations of the Blueberry's recessed mount and the very real possibility that it can become cross-threaded and stuck on a stand. This happened to me on the last day of testing, and it took a nerve-racking 30 minutes of careful manipulation to free the mic-a process I would never want to repeat.

Also available is the W1 dual-mesh pop filter, which is made of metal and attaches to the S1 shock-mount. The mic's heavy-duty, dual-mesh grille proved sufficient, though, so I never needed to use the pop filter. Also optional is the BB Blueberry high-definition mic cable. The S1, W1, and BB are available as a package for \$318. You can also buy the BB cable separately for \$34.95.

GUTS AND GLORY

As with all BLUE transducers, the construction, look, and feel of the Blueberry is superlative down to the smallest detail. Of course, it's the unseen parts that are the crucial ones, and these, too, are first-rate. The preamplifier in the mic body uses Class A discrete circuitry (no integrated circuits) and a custom transformer output.

The microphone capsule exhibits tonal characteristics similar to the BLUE Bottle mic's B0 capsule—eight interchangeable capsules can be used with the Bottle—and employs a singlemembrane, 6-micron diaphragm. Critical diaphragm tensioning and a unique metal formula (mostly gold, but partially aluminum) are said to be responsible for the high sensitivity and superb transient response of the capsules, which are manufactured and individually tested by BLUE.

PURPOSE IN LIFE

I got off to a bit of a false start with the Blueberry because of some misleading



Designed primarily as a vocal mic, the Blueberry delivers a bright, detailed, intimate sound.

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the impres-

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audio

A transducer and a port can't equal the LF output of the HR824's two transducers.

bass output, other monitors resort to using ducted ports that can convert cone movement into extra low frequency air movement. But for optimal output, a ducted port needs to have the same area as the low frequency transducer an 8-inch near field monitor would need an 8-inch vent. Needless to say, you haven't seen any vents this big on other near field monitors. When vent size is reduced, bass output is

compromised. And, forcing a lot of energy out of small ports can

create audible wheezing and whooshing. Instead, the HR824 adds a large passive transducer with the cone area of another 8-inch woofer. This ultra-rigid, honeycomb laminate piston tightly couples with the HR824's active bass transducer. With a combined cone area greater than a single 12-inch woofer, you get exceptionally extended bass without port noise complaint.

MASSIVE POWER THAT WOULD **PROBABLY POP A PASSIVE MONITOR.**

Punching out crisp bass requires a lotta watts. The FR Series high-current bass amplifier module inside the HR824 delivers a solid 150 watts of power with peak output in excess of 250 watts (plus another 100 watts for mid and treble). That's significantly more than any other 8-inch active monitor. Moreover, the HR824's servo coupling and ultra-short signal path put that power to work far more effectively than a passive monitor and a 250-watt stereo amp could.

PART OF A TIGHTLY-INTEGRATED

SYSTEM. Our servo bass system is only

one contributing factor to the HR824's

amazing accuracy.

Internal power amplifiers are "fed" by

phase-accurate, low

distortion electronic

circuitry instead of a

crude coil-and-capacitor

passive crossover. The HR824's proprietary logarithmic wave guide not only widens

trebie dispersion but also smooths the midrange transition between high and low-frequency transducers. Thanks to the wave guide's flaring design, the HF transducer's output is acoustically the same diameter as the LF transducer's at the critical 3500Hz crossover point.

The HR824's LF transducer even contributes to midrange accuracy. In many monitors, woofer cone harmonic vibrations bounce around inside the enclosure and then exit through the thin woofer cone. The result: smeared imaging and muddled details. Instead of a chintzy chunk of fluff, the HR824's enclosure is utterly packed with high- density absorbent foam. Cone vibrations go in, but they don't come back out.

Rear view: The HR824's electronics conceal an ultra rigid, honeycomb composite passive

DON'T SKIMP. It's amazing how many studio owners will mortgage the farm for money-is-no-object, esoteric microphones... and then monitor on

HR824

cheap, passive loudspeakers. If you aren't using some brand of ACTIVE near field monitors you're seriously compromising your creative product.

HEARING IS BELIEVING. We urge you to visit your nearest Mackie Designs Dealer and carefully audition all of their active monitors with some demanding, bass-rich program material. Judge our claims (and those of our competi-

tors) for yourself. We think you'll agree that the HR824 is truly the best of the best.



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BLUEBERRY

came across as slightly unpleasant sounding. The Blueberry, however, sounded very accurate on the tambourine, giving a pleasing, lifelike dimension to the instrument's every jangle and shake.

Used as an auxiliary microphone on an eclectic setup by percussionist Karen Stackpole, the Blueberry effortlessly documented subtleties produced by pitched stones, Tibetan singing bowls, and the tiny wires of an egg slicer. On snare drum during a jazz session, the Blueberry gave me everything I wanted to hear right off the bat, turning a mediocre-sounding metal drum into a thing of beauty.

With a ceramic dumbek, the Blueberry sounded amazingly crisp and realistic. Indeed, everyone in the control room commented on how authentically it accommodated this instrument. It did not, however, capture as much tone from the drum as I would have liked. We also tried the Blueberry on a clay udu drum. The sound lacked essential lows, though, and was too "slappy" for my tastes.

Based on the Blueberry's performance in these sessions, I could imagine it giving a stellar showing as an overhead mic for a drum set. Unfortunately, I had only one Blueberry at my disposal for most of the review period, and this application typically requires two matched mics.

The percussion tests helped reveal another thing that sets the Blueberry apart from most other large-diaphragm mics I've heard: its outstanding transient response. (The term *transient* describes the spiky, initial attack of a percussive sound such as a hand clap or wood block, which is transduced as a rapidly rising voltage.) The Blueberry is dynamically a very fast mic, capable of following extremely rapid changes in sound pressure with accuracy. In fact, this microphone's dynamic response rivals that of the best ribbon mics and small-diaphragm condensers.

LEFT AND RIGHT

The manufacturer, concerned about my assessment of the first mic, sent along a second Blueberry, which arrived toward the end of the evaluation process. Curiously, though, my independent tests and those of **EM** associate editor Brian Knave revealed the second mic to be slightly brighter sounding than the original demo unit; for this reason I didn't use the pair for critical stereo recording.

However, I did have occasion to employ both mics in a stereo XY configuration on an acoustic guitar overdub. For this rapid strumming part, the paired Blueberries contributed wonderfully airy highs to the mix. Individual EQ was required to create a tonal balance between the two mics, and I had to add some upper bass at about 300 Hz on both mics, despite placing the pair as close to the instrument as was possible.

I also performed a test comparing the BB Blueberry high-definition mic cable to a budget mic cord. To my surprise, the difference in high-end detail was easily audible. The BLUE cable provided more brightness and depth on acoustic guitar, more "smack" on a snare drum, and better definition on the attack of a kick drum. Indeed, the more I listened, the more differences I

Blueberry Specifications

Mic Amplifier	solid-state discrete Class A
Transducer	pressure gradient
Diaphragm	1", 6-micron-thick, gold/aluminum vapor-deposited Mylar
Polar Pattern	cardioid
Frequency Response	22 Hz–22 kHz
Dynamic Range	137 dB
Sensitivity	21 mV/Pa
Signal-to-Noise Ratio	69 dB (rel. to 1 Pa @ 1 kHz)
Self-Noise	7 dB
Maximum SPL (for 0.5% THD)	133 dB
Dimensions	9" (H) x 2" (W) x 1.5" (D)
Weight	1.19 lbs.

heard, including a cleaner hi-hat sound, smoother vocal sibilants, and a slight increase in sustain and coherency on bass notes.

NOTHING BUT BLUE SKIES

BLUE is to be commended for producing a mic in the \$1,000 price range without compromising its reputation for superior construction quality and mod styling, and for creating a distinctive voice rather than just another sound-alike mic. I think the manufacturer would be well advised to better highlight the specialized purpose of this mic, though. Emphasizing its unique qualities-and precisely how it should be used-would reduce confusion for potential buyers and attract the attention of recordists seeking the Blueberry's enhanced presence and superb transient response.

Specifically, the consumer should be made aware that the capsule design responsible for the Blueberry's exemplary transient response necessarily compromises the mic's low-end pickup to a degree. To make up for it, the Blueberry typically has to be positioned quite close—maybe two to three inches from the sound source. Only then does it deliver the full low end that many of us have come to expect from large-diaphragm condensers positioned, say, 6 to 12 inches from the source.

On the other hand, if you're the type of person who is constantly trimming bass frequencies on your tracks, the Blueberry could be the answer to your prayers. The mic's quick responsiveness and airy signature sound make it ideal for capturing nuance and highend detail. It is especially well suited for pop vocals and would likely be a great mic for sampling and Foley recording, as well. In addition, for percussion-based music and other styles that favor a clean, bright response (bluegrass or solo acoustic guitar, for example), the Blueberry will deliver snappy, lifelike presence every time. Moreover, an established recording facility may find its crisp attributes to be the perfect complement to a growing mic closet.

Without a doubt, the Blueberry is a superbly crafted microphone with a sound and look all its own. Considering the many copycat mics that have been released in recent years, that is something to cheer about.



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TREALE

NEUT LEVE



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CYCLING'7 PLUGGO 1.04 (MAC) Exceptional tools for

audio processing.

By Dennis Miller

ycling '74 has released a remarkable new application that can be a major addition to your audio processing resources. *Pluggo* allows Mac users to convert *MSP* patches into VST effects plug-ins, complete with attractive and versatile graphic interfaces. The initial release includes 74 *Pluggo*/VST effects, as well as instructions on how to make your own. This is among the most costeffective audio solutions to come along in some time.

Cycling '74's MSP collection of audio processing plug-ins is legendary. Winner of the 1998 EM Editor's Choice Award for best DSP software (and reviewed in the October 1998 issue), MSP does for audio what Opcode's

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2. Reverb Gain				-	0.1900
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4. AP1 Feedback				-	0.7500
5. AP2 Delay				-	34.6499 ms
6. AP2 Feedback				-	0.7200
7. AP3 Delay	-			=	24.1799 ms
8. AP3 Feedback		•		-	0.6910
9. AP4/5 Base Delay				-	17.8400 ms
10. AP4/5 Feedback		•		-	0.6400
11. AP4/5 Mod Freq	-			=	0.0000 Hz
12. AP4/5 Mod Dept.	-	•		-	0.0000
13. AP6/7 Delay				-	10.8200 ms
14. AP6/7 Feedback				-	0.6620
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FIG. 1: The native *Pluggo* plug-in interface provides sliders for changing parameter values. A meter can be enabled to show input or output levels.

Max does for MIDI. Using a graphic, modular "toolkit," you build complex audio processing routines that can be applied to your sounds in real time. There are dozens of sound-generating and processing functions you can employ and numerous tutorials with examples to get you going. Though converting your own MSP patches to *Pluggo* plug-ins is not a trivial task, the process is clearly explained in the *Pluggo* documentation (see the sidebar "Roll Your Own").

You don't have to be an *MSP* user to take advantage of what *Pluggo* has to offer; simply install the included plugins and start experimenting right away. Then keep your eyes on the Cycling '74 Web site, where new *Pluggo* plugins can be submitted by the many existing *MSP* users.

SET UP AND INSTALL

Pluggo is one of a new breed of programs that can be downloaded directly from the developer's Web site. The download starts out in demo mode, in which the plug-in's output is periodically interrupted. To convert the demo to a fully running program, you need to purchase an authorization code

> from Cycling '74. If you buy the software on CD-ROM, all authorization and IDs are included in the box.

Pluggo installs easily into the home directory of any VST host software. Currently, this can be Steinberg's Cubase, Emagic's Logic Audio, Cakewalk's Metro, Opcode's Studio Vision Pro or Vision DSP, or TC Electronic's new Spark audio editor. Prosonia has also announced VST support for a forthcoming version of its SonicWorx Studio audio editor. Depending on which host you are using, you may need to make multiple copies of the plug-ins. For example, Logic Audio can access them from anywhere on your drive, so it doesn't need to have its own copy. But Metro requires that all plug-ins be in its home directory. (Two small Lib files must be in the home folder of whatever host program you plan to use.)

You can install all or a limited number of plug-ins initially, and then use the included Plug-in Manager utility to specify which plug-ins appear when you run your host program. This allows you to keep different sets of plugins on hand for different projects. The Plug-in Manager can also keep track of any non-*Pluggo* VST plug-ins that you have.

Once you have *Pluggo* installed, you can access and use the *Pluggo* plugins like any other VST effects. Keep in mind that, depending upon where you apply a plug-in, the effect will either be mixed with the dry version or will replace it. Be sure you carefully consider where in the signal path you use the plug-in, so you get the effect you want.

INTERFACE ISSUES

Many Pluggo plug-ins incorporate a screenful of sliders, though some use other interfaces that are better suited to their functions. In fact, Pluggo developers can choose from a wide range of elements-including number boxes, toggle boxes, envelope editors, and pop-up menus-to provide the most appropriate controls for the various parameters that their effects require. Because the 74 included effects were created by several developers, you'll find some variation among their interfaces, but the vast majority of the plugins share a consistent look and feel, and you'll quickly feel at home working with any of them.

At the top of a typical screen is a small menu for choosing among numerous view options and a real-time level meter that can be disabled if you want to eke out every drop of processing power for the effects themselves (see Fig. 1). The main work area of the plug-in displays sliders for each individual parameter value. The sliders are very responsive and have a large button (or "egg") to grab on to, making it easy to fine-tune a setting with the mouse. Unfortunately, there's no way to type in numeric values for parameters that employ sliders.

There are, however, various shortcuts for setting values. For example, if you Command-click on a slider, the Parameter Change menu appears, which offers options including randomizing

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FIG. 2: You can control non-Pluggo plug-ins using all of the Pluggo functions. Either the plug-in's original interface (top) or the Pluggo slider interface (bottom) can be used to control the effect's parameters.

all settings, "evolving" (shifting) all values a small amount, and importing the settings of one of the presets for the particular plug-in you're using. The menu also offers a single-level Undo command. I found the Parameter Change menu to be very handy for jump-starting effects settings, and ended up saving a vast number of settings that it produced. Because the menu is so well integrated, you'll no doubt find yourself turning to it time and time again.

When you move your mouse over the sliders and knobs, you see a concise pop-up description of what each parameter represents. This is only one of many helpful touches the program offers, and it shows the care and detail that went into its planning. To my mind, Pluggo is truly a marvel of modern interface design.

IN CONTROL

Pluggo offers a number of other ways to control, and in some cases even automate, parameter changes. In addition to moving sliders, you can use the PluggoBus to provide the values for a parameter. The PluggoBus is an audio route or "pathway" that exists inside the program, along which you can send input and output to any Pluggo plugin. This gives you the ability, for example, to pass the output of one plug-in to another (that is, run effects in series), or to analyze some aspect of an audio signal and use the analysis data to control another plug-in. For example, you could track the amplitude of an audio signal on one track in your digital audio sequencer, and then use the amplitude levels to determine the LFO rate of another effect. The possibilities are endless, and as with other topics, the manual includes a thorough tutorial on how to use this feature.

Another tool for controlling effects parameters in real time is the Mouse Mod plug-in. Using Mouse Mod, you can assign up to four parameters of a plug-in to func-

tion under the control of mouse movements. You can determine whether the mouse's position will set, offset, or scale the values, and you can set the minimum and maximum range of the values that are sent. Using the Mouse Mod's Gate feature, you can assign a keystroke as a trigger that will enable or disable any of the four outputs in real time.

Because the Mouse Mod screen lists the parameters of every plug-in that is currently in use, you can change values in different effects simultaneously. For example, you might use the Mouse Mod as a channel insert, and then assign a granulator and a delay as master effects. Vertical movements of the mouse could control the range of the granulator's grain duration and wet/drv level, while horizontal movements might offset the delay's time and feedback amount. Of course, if you're short on desktop real estate, you might prefer to use the Key Triggers plug-in to assign parameter control to your Shift, Caps Lock, Option, Command, and Control keys.

Finally, you can use PluggoSync to provide timing references where you need them. PluggoSync has its own internal clock, which you could use to trigger a step sequencer, but it can

also detect timing information coming from an audio signal, such as a click track. One way to employ this function would be to place the included audio click-track file on any track in your sequence, and set PluggoSync to use the audio as its clock source. Next, open the Synth plug-in and set its built-in step sequencer to trigger under the control of one of the five PluggoSync outputs. By the way, each of the outputs can be a different division of the audio pulse.

ABOVE AND BEYOND

Pluggo lets you use non-Pluggo VST plugins in new and unusual ways. To use a non-Pluggo plug-in, open the generic plug-in (simply called *Pluggo*), and then load any VST plug-in you have on your system. When you open the edit screen, you can switch between the plug-in's original interface and the standard Pluggo slider screen (see Fig. 2).

In addition to the alternate interface. there are other tricks you can perform with your existing VST plug-ins. For example, you can modulate the plug-ins' parameters using any of the Pluggo modulation options, including the Mouse Mod, the LFOs, or even envelopes you create with the Breakpoint editor. You can also use the Undo command, randomize parameter values, and import settings from an effect's presets, just as you would with native Pluggo plug-ins.

OUT OF THE BOX

Pluggo

The 74 effects included with Pluggo cover an enormous range of processing functions. You'll find nearly every type of modern audio processing tool included in the set, with the exception of spectral analysis and resynthesis. There are delays, filters, reverbs, granulators, distortion effects, compressors, panners, modulators, and vocoders, in addition to other effects that simply defy description. In nearly every case, the plug-ins include numerous useful presets, which means that you'll have a vast number of effects at your disposal right out of the box. I'll give a brief

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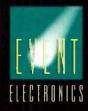
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PLUGGO

ROLL YOUR OWN

Though creating patches in *MSP* and converting them to VST plug-ins requires an investment of time, the rewards—an unlimited number of customized audio effects—are well worth the effort. You need both *MSP* and Opcode's *Max* software on hand. (See the Cycling '74 Web site for a special offer on these programs.)

The first step in the process is to add audio inputs and outputs to your MSP patch. MSP's audio inputs and outputs are called adc~ and dac~ (analog-to-digital converter, digital-toanalog converter), respectively. Normally, these modules, or "objects" in Max terminology, are used to get audio in from a microphone or out to your speakers. To move audio through your VST plug-in, however, you'll use plugin~ and plugout~ (see Fig. A). If you insert a plugin~ after the adc~ and a plugout~ before the dac~, then the adc~/dac~ will be ignored when you use your patch with Pluggo, and the plugin~/plugout~ will be ignored when

you run your patch in *Max/MSP*. This way you can easily switch back and forth between *Max/MSP* and your VST audio sequencer while designing and testing your plug-in.

With only these minor modifications, your patch will already function as a plug-in

with *Pluggo*, but if you want to have control over any of the patch's parameters, you'll need to add a "pp" (plug-in parameter) object for every user-controllable value in your patch. Each pp object will correspond to a slider in *Pluggo*'s default interface. Examples of functions you might want to control would be input gain, output gain, delay time, reverb time, or filter frequency. Naturally, these parameters will depend on the type of effect you have designed.

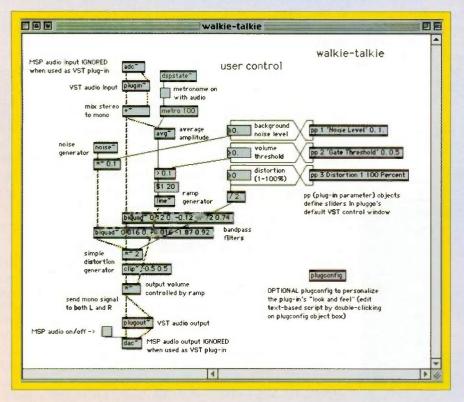


FIG. A: This MSP patch includes the two objects (plugin~ and plugout~) needed to pass audio through a VST plug-in. No other changes are required to the MSP patch, though the user might add a number of additional interface enhancements.



FIG. B: The plug-in as it would appear inside a VST host program.

The example in Figure A, made especially for this article, is a loud noise-gate with distortion called "walkietalkie." Its three parameters are background noise level, noise-gate volume threshold, and distortion level. The initial *Max/MSP* patch has three user-controllable number boxes, each of which is connected to a pp object (numbered 1, 2, and 3). These numbers define the topto-bottom order of the sliders as they will appear in the VST plug-in's editing window (see Fig. B). Notice that the pp

> objects contain additional information about the parameter's name, the values that the sliders will be scaled to, and an optional additional label to describe the values (for example, distortion is expressed as a percentage).

> Once you have a plugin~, a plugout~, and any number of pp objects in a Max/MSP patch, you can load it using the generic Pluggo plug-in and start tweaking the slider values. But at this point, you might want to convert your patch into a plug-in itself, so it can be selected directly from the list of plug-ins available within your VST host program. This is probably the easiest part of the process; just drag and drop your Max/MSP file onto the included Plugmaker application and, voilà, you've made your first VST plug-in. Finally, be sure to remove the .pi suffix attached to your plug-in's name after you move the plug-in to your audio sequencer's VST Plugins folder, or the sequencer may not be able to load the plugin correctly.-Richard Dudas

summary of several of the plug-ins; keep in mind, however, that a comprehensive review would take many more pages.

Filters. In the Filter category you'll find such gems as the Harmonic Filter, which is a set of 25 bandpass filters that can be placed under the control of an algorithmic function called Cellular Automata (CA). CA uses a set of rules to determine the current state of a collection of variables, which in this case might be controlling the frequencies of the 25 filters. It then constantly varies those values, producing a random quality to the parameters that it is controlling. The Harmonic Filter uses several types of interface elements to configure various aspects of the process, and an attractive, animated two-dimensional display shows the positions of the different frequencies (see Fig. 3).

Among the other interesting filter effects are a vocoder, available in both 10- and 16-band models; Moving Filter, which runs a signal through two parallel bandpass filters that can accept time-varying parameter changes; and Cyclotron, which employs a step sequencer to determine the filter's frequency and Q factor and provides settings to adjust tempo, glide rate, number of steps, and more.

Delays. The Delay category includes Flange-o-Tron, which provides two 16step sequencers for controlling delay and feedback levels; and a little gem called Raindrops, written by master sound programmer Jhno. Raindrops provides a fascinating assortment of

Harmonic Filter View Harmonic Filter = Meter Out = - **- - -**23 damp nate

FIG. 3: The Harmonic Filter plug-in uses several of the interface elements that are available to plug-in developers. In addition to the animated screen that shows the filters' frequencies, this plug-in employs toggle boxes, sliders, and number boxes.

functions, including two random-number generators that control the center frequency of two bandpass filters, a gain/overdrive setting, resonance, decay, and density. The presets are particularly effective.

Granulators. Pluggo serves up a good number of splicing and dicing tools. There are no fewer than seven dedicated plug-ins for granulating audio files, as well as various other tools that can be made to produce similar results. Part of this bunch is Slice-n-Dice, which chops its audio source into 32 equal chunks and then provides a graphic interface that you use to specify the

This is one of the most cost-effective audio solutions to come along in some time.

number of chunks that will play back and the order in which they will play. You can apply attack and decay times to each chunk, and sync the delay time to a PluggoSync output.

The Wheat tool also uses a graphic interface but includes a number of parameters not often associated with granulation, such as a graphic pitch envelope. Shuffler and Stutterer both provide ways to reorganize small

> chunks of looping audio, and Rye is a slider-based plug-in that includes controls for grain duration, grain period, feedback delay and amount, and grain crossfade.

> Distortion. The distortion group is another robust category of effects. Among its numerous offerings are a waveshaper, a ring modulator, a feedback network, and a "cruncher." There's also a fuzz effect, a sampling-rate and bit-depth "degrader," and a plug-in that multiplies your audio

with noise (with complete control over all of its parameters, of course). In fact, you'll probably find every type of grunge-producing effect you'll ever need within this group.

Reverbs. Only two dedicated reverb effects are included in the Pluggo package, but they provide a wide range of options. The first reverb, called Rough Reverb, is a stereo effect that has only three parameters: wet level, dry level, and reverb time. ChamberVerb, on the other hand, is far more versatile and includes 16 different parameters. The design of this effect is based on the allpass filter network described in Hal Chamberlin's seminal Musical Applications of Microprocessors but includes a few enhancements to the original. The sound, as they say, is silky smooth (or at least it can be).

Miscellaneous. A number of other effects simply defy categorizing. Plug-Loop, adapted from an application written by Ihno, loops up to three audio inputs and gives you control over the length, speed, and direction of each one (in real time, of course). I took the advice of the manual and used the Randomizer modulator plug-in to control the speed and length of the loops; I easily filled up nearly an entire DAT tape of material that I expect to use in a future composition.

The Breakpoints plug-in lets you create your own multisegment envelope and apply it to most any effect you want. Doppler effects can be found in the excellent Dr. Dop plug-in, written by Zack Settel of McGill University, and an audio-rate panner is only one of several spatialization effects. Warble, Warpoon, Vibrato, Cauldron, and Frequency Shifter perform various types of pitch-altering effects, often with humorous consequences.

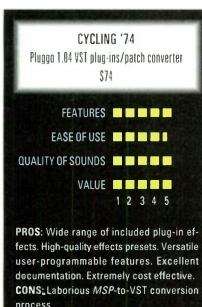
PROPS TO PLUGGO

Pluggo appears to be a remarkably efficient application. Using Steinberg's Cubase as the software host, I was able to run four effects as channel inserts and another four as master effects without choking my G3/266. Even with these eight effects running, I could use the Mouse Mod to make smooth changes to different parameters, and the MIDI data in my sequence played without a hitch.

The range of supporting documents is excellent for an application at this price. First, if you buy the CD-ROM



PLUGGO



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version, you'll get a hard-copy manual that provides a thorough overview of the program's operations. (The same document is available from the Web site if you purchase online.) When you install the software from the CD-ROM or Web download, you also get a PDF file that details every plug-in, including a description of each parameter, suggestions for usage, and an indication of the amount of processing power the plug-in will require. Yet another PDF file offers instructions for creating your own plug-ins, and tutorial files are available to ease you through that process. Finally, information about each plug-in is available via online help as you work.

If you own a Mac and are serious about audio, you must get this program. At \$74 for 74 effects (make that several hundred effects if you count all the presets!), this has got to be one of the best deals of the century. In fact, even if you don't own a host program that can use *Pluggo*, you should consider buying such a program just for that task. Moreover, Cycling '74 gives every *MSP* owner a free copy of *Pluggo*, so for only \$295 you can get it as a bonus with the massively powerful *MSP* application. One way or another, it's time to get plugged in to *Pluggo*!

Associate editor Dennis Miller has been plugged in since the age of 11, when he purchased his first Silvertone electric guitar and amp.

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GENERALMUSIC

EQUINOX 61

A synth that wants to be all things to all people.

By Julian Colbeck

ith its Equinox series of synthesizers, Generalmusic is redefining the synthesizer workstation. The Equinox 61 sports a fully editable synth engine, complete with resonant digital filters and massive oscillator power. Beyond that you get a 250,000-event sequencer, sample-loading facilities, a groove library, a drawbar section for serious organ playing, and an operating system that's upgradable with a floppy disk. You can even load text files, so you can review your shopping list during rehearsals!

Add in the multifarious options of a hard drive, a SCSI port, and up to 40 MB of memory (8 MB of batterybacked RAM, plus 32 MB of SIMMs), and you've got one big box of goodies.

Unlike many other Italian-made synths, which tend to offer the world but require a lifetime of study to operate, the Equinox 61 delivers its power in a mostly intelligible way, thanks to a well-designed display, plenty of panel hardware, and a logical operating system.

PHYSICAL ATTRIBUTES

The Equinox 61 is a good-looking instrument, professionally presented with sturdy pitch and mod wheels, recessed disk drive, and contours in the end panels that make the unit very comfortable to grab and carry around—a thoughtful touch.

The screen is smallish (only 3×2 inches), but sufficient for displaying the amount of information you need. Generalmusic has not thrown budgetary caution to the wind with the Equinox panel hardware, but the synth's many control sliders and buttons feel substantial—a good thing, because this instrument has great master-keyboard potential. (The model reviewed has 61 notes; a 76-note model is also available.)

I have some reservations about the Equinox 61's keyboard, which has too light an action for my battle-hardened fingers. (If you are similarly fussy, check out the summer-NAMM-previewed, weighted 88-note Equinox Pro model.) The synth's ins and outs are all in the right place, including a pair of frontmounted headphone sockets. The back panel includes two sets of MIDI ports, Left and Right audio out (along with an auxiliary pair of outputs), four pedal connectors, and the power switch and socket (see Fig. 1). There are also two mic/line inputs.

SOUND STRUCTURE

The most basic Equinox patch is called a Sound; more than 1,000 are provided onboard. Sounds can be stored multitimbrally in a Performance, 112 of which are offered as presets. An additional 112 locations can house your own Performance creations.

The Equinox 61 does not break any new ground in terms of sound structure. It uses a sample-plus-synthesis system that will be reasonably familiar to anyone who has programmed a Korg, Roland, or Yamaha synth in the past ten years.

But what is new—radical, almost—is the way in which the Equinox implements its various structures and systems. You find yourself wanting to do something and, presto, this synth lets you do it. You can do anything from single-sound tweaking via the panel sliders to using those same sliders to rebalance a multisound Performance. Can you save these edits? Yes. Will those sliders kick out MIDI control information? Yes. Can you tweak within a sequence? Yes. Can you instantly switch off the effects? Yes!

The Equinox comes with a thumping 16 MB of waveform ROM from which sounds can be hewn; you can also load your own samples in WAV, AIFF, and other formats. A Sound can be a three-layer, dual-oscillator sandwich if need be, although harnessing six oscillators like this will slash your polyphony from the 64-note, singlelayer/oscillator maximum.

Despite its modest size, the Equinox's screen is clear and helpful when determining the number of layers, programming the oscillators, and setting up the signal flow that you want your sound to follow. You have a choice of keeping pitch, amplitude, and filter envelopes separate for each waveform on a dual-oscillator sound or applying the same contouring to both waveforms. You can even choose to unify or separate the filters themselves. LFO, pan setting, effects, and mod routing are all common parameters per layer.

Although layering gobbles up polyphony, you could set up keyrange and Velocity-dependent layers, which can make for some highly expressive patches—flutes that develop a guitarlike edge, say, or marimbas that change in pitch—with added keyboard dynamics. Oscillators can be chosen from an Olympic-size pool of waveforms, and these can then be processed through multimode (digital)

> filtering, LFO modulation, the envelope generators, and more.

> The Equinox 61's sounds range from every imaginable squeak and hit to looped and moving waveforms that can be played forward or backward, have their start time offset, be rescaled (so that each key triggers the same note, if that's what you want), and fine-tuned in respect to their dynamic performance.



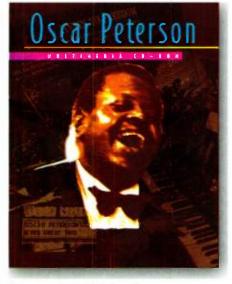
Generalmusic's Equinox 61 offers a powerful sequencer, 64-note polyphony, and more than 1,000 preset sounds. The sampling/synthesis keyboard can load many types of sound files and has a unique Grooves feature.

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EQUINOX 61

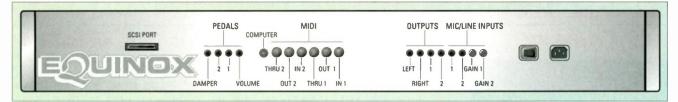


FIG. 1: The rear panel of the Equinox 61 sports two sets of MIDI ports, audio inputs and outputs, a serial connector, and the optional SCSI hookup.

This synth gives you an almost overwhelming amount of programming control. I like the fact that you don't have to download a *Unisyn* driver (or an equivalent patch editor) to edit sounds. On the other hand, any serious programmer will soon become tired of scrolling through edit pages and making adjustments on the Equinox's comparatively sluggish data wheel or keypad; this is a problem with the hardware and, to some extent, the user interface.

The Equinox software is abundantly rich in filter types (lowpass, highpass, bandpass, parametric cut, parametric boost) and sophisticated envelope generators—not just for the amplifier, but also for the filters and oscillators. Indeed, rather than offer standard attack/delay/sustain/release controls, the Equinox provides you with graphic control over the Key On and Key Off portions of your sound. You can thus draw the shape to control how your sound's volume, tone, or pitch changes over time.

The number of segments in both Key On and Key Off envelopes is your call. Well, 10 is the hard limit (just keep hitting Enter in the Add Segment pages), but I can't see anyone seriously ex-

Equinox 61 Specifications

Keyboard	61 keys, with Velocity and Channel Aftertouch
Hardware Controllers	(8) programmable sliders;
	(8) programmable buttons
Synthesis Type	subtractive sample-playback
Polyphony	64 notes
Multitimbral Parts	32
Factory Programs	1,178 Presets; 112 multitimbral Performances;
and substitution and states the	16 Drawbars
User Locations	2,048 Presets; 112 multitimbral Performances;
	16 Drawbars
Internal Sound ROM	16 MB
Internal RAM	2 MB sound RAM (expandable)
Sequencer	16 tracks; 250,000 events
Groove Patterns (factory/user)	1,024/512
Arpeggiator Patterns (factory/user)	16/16
Sample Resolution/Playback Rate	16 bit/44.1 kHz
Effects Processors	3: reverb, MultiFX, and ProEFX
Effects Types	71: 10 reverbs; 30 MultiFX, 31 ProEFX
Display	3" x 2" LCD
Expansion Options	8 MB battery-backed sound RAM; 32 MB DRAM (SIMMs); internal hard drive (2 GB); SCSI port, A-D/vocal processor card
Audio Inputs	(2) ¼" mic/line
Audio Outputs	(2) %" main L/R; (2) %" auxiliary (dry);
	(2) ¼" stereo headphone
Other Ports	(2 sets) MIDI In, Out, and Thru,
	(1) serial computer interface (Mac/PC);
	(3) programmable pedal jacks
Power Supply	120 VAC internal (IEC)
Dimensions	41.5" (W) x 4.5" (H) x 13.5" (D)
Weight	29 lbs.

ceeding that. Each segment has a Time and Level parameter (value 0 to 100) and can be looped. Fortunately, you can also delete a segment if early flights of programming fancy start to pall.

As powerful a system as the Equinox 61 is, it's not at all complex—the screen is just too small (in spite of the handy zoom feature). And I did manage to make the instrument crash several times when fooling around with the envelope generators. But luckily, the unit is blessed with an updatable operating system, and the newly released OS has apparently fixed several problems in the envelope environment (though I haven't tested it myself).

PICK QUICK

In the early days of synthesizers, an instrument's tally of oscillators, filter types, and modulation routings was the first, if not the only, port of call when deciding if it was the right model for you. Today, although they boast parameter lists far outweighing those of any old Oberheim and Sequential warhorse, modern marvels like the Equinox 61 almost hide their power under the hood.

The front panel offers an innocent clutch of Quick Edit sliders for immediate control over attack, decay, and release envelope parameters, filter cutoff and resonance, and LFO depth and rate. Just call up any of the instrument's 1,000 or more instant-gratification presets, and make adjustments on the fly. You can even overwrite Sounds thus edited, record Sound changes over time into a sequencer, or both.

MEET THE FAMILY

The Equinox's factory sounds are grouped into 11 banks (or "families," as Generalmusic refers to them), each containing 128 patches. The first four banks are simply labeled Synth A-D. Thereafter come Orchestra, GMX 1-3, Drum Kits, and Drum Sounds 1 & 2.

Sounds range from Rick Wakemanesque solo synth with portamento, to standard pianos, to multilayered of transient response to power handling the shipland over uces an extremely wide dispersion, low choorsion datial w signed for an optimal response for con-

(Non-swelded components can make a that the system exclusion about 10 rausion cal sistem-gauge steel cap encasing the dio- pass electronic crossover libers de a passive crossover used after the amplifine light pass (owneter) sections ersi advantages to using built commitmentiodulation discore used dynamic range. The MI A ersponsiver coupling for wgole errences. Easier manipulation of par-

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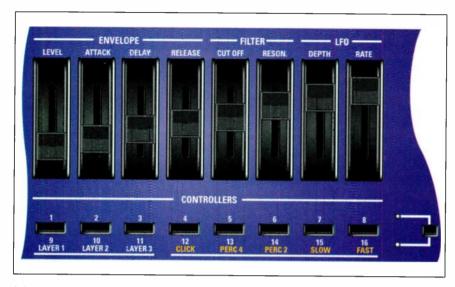


FIG. 2: The Equinox 61 offers eight fully programmable sliders that serve as drawbars when playing organ sounds; there are also eight user-definable buttons. Up to 16 different configurations for the buttons and sliders can be saved as presets.

swooshes and swizzles. Pretty much the lot. Organs? Not only do you get a bag full of blistering organ patches, but the Quick Edit slides can. at the touch of a button, turn into a set of eight drawbars providing real-time harmonic control (see Fig. 2). Similarly, several controller buttons beneath the drawbars can double as slow/fast Leslie switches, percussion switches, and click switches.

You want drums? The Equinox boasts some 40-plus drum kits, mapped using different drum sounds across the keyboard range, and you can also substitute sounds from any part of the instrument (including samples) to produce fully customized kits and subsequent loops. Individual drum sounds can be given their own pan and volume setting, and you can specify that certain drums be exclusive so that, for instance, open and closed hi-hats cannot be triggered simultaneously. If you find the General MIDI mapping too restrictive, you can spread a sound across several keys as well. Someone has really thought this out. It's good stuff.

So you get a huge number of pitched sounds (with organ a particular specialty). a competent drum sound system, and samples; what else could you ask for? Would it be churlish to want "character?" The Equinox does not have the distinctive character of, say, a Nord Lead, a Yamaha FS1R, or a Korg Wavestation. You won't hear the phrase "Just use a classic Equinox patch" very often. But in terms of range and potential, the Equinox 61 makes some other synths in its class seem uninspired and, frankly, rather tired.

EFFECTS

Ever since the Korg DW-6000 first added effects to a synthesizer's parameter list, instruments have had a lovehate relationship with this previously outboard area of signal processing. There's no question that effects enhance sounds; indeed, that's what they're supposed to do. The question is when and how, and who should perform the operation.

It's easy for effects to mask lazy programming or to smear some inherent weakness in the original waveform. Bravely and laudably, the Equinox 61 provides a dedicated FX Off button, so you can test this for yourself. Still, I'm concerned about the importance that the effects have in some of the sounds. For instance, should a sound labeled Steel Gtr, say, or Jazz Pick—relatively clean types of sounds. surely—be so reliant on not just reverb, but also on chorus and exciter processing?

What's the harm? Well, provided you plan just to play or trigger single sounds, the only harm is a tendency toward patches that sound great on their own but don't work, or can't be heard, in context. But because the Equinox 61 is a workstation, and much of its work is geared toward multi-instrument playback, there is the inevitable tradeoff between single patch effects and effects that are available simultaneously for the instrument as a whole. Even though the unit does have separate effects processors, settings still have to be apportioned among all current sounds.

Generalmusic does address this perennially thorny issue as well as it can. The Equinox 61 offers three separate effects processors: MultiFX, Reverb, and ProEFX, with effect type and attendant programming parameters individually assignable and routable for each sound.

Effects can be set up in either 3 Effects or 2xSynth + 2xSeq modes. The first maximizes all available processing, but the second is very useful if you're accompanying a prerecorded sequence and want the effects on your live playing to be different from those in your sequence. There's a catch, though: this separation nixes the ProEFX effects altogether.

Though I'm issuing words of warning about synth programming in general, I don't mean to diminish the inherent quality and flexibility of the Equinox's effects processing. Reverb, for instance, is relatively uncontentious, with a choice of ten reverb types (Hall 1-4, Plate 1-2, Matrix 1-2, Gated, and Studio Room), each offering control over reverb time and delay factor. I'm not about to split hairs between Plate 2 and Matrix 1, but the quality here is unquestioned. Ten years ago you'd pay thousands of dollars for equipment that can achieve reverbs of this smoothness. Even with comparatively silly reverb times like ten seconds, there's no degradation of the signal.

The MultiFX bank offers 30 mainly time-based effects, ranging from mono and stereo delays to chorus, flanging, and phaser. One or two rather unusual effects creep in: dubbing, dramatic hard left-right delay, Pitch Shifter 1 and 2, two rotary speaker simulations, and a few preordained EQs—Jazz, Pop, Rock, and Classic—for, I presume, the sort of Kmart hi-fi shopper who wants instant access to (what someone else deems) such sensibilities.

The really juicy effects are stationed in ProEFX, where you'll find Stereo Wah, Overdrive/Delay, Audio Exciter, Hex Chorus, 2 x PitchShift, Ring Modulator, and about 25 other effects.

There's no denying the power of the Equinox's effects. Call up almost any patch and start going through the effects options—it's almost like flipping

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through brand-new patches. The pitchshifting options are especially powerful where you have separate control over tuning (both fine and in half steps), and high and low frequency gain over both the original and pitchshifted signal.

It took me some time to find the Pro-EFX programming parameters, because I had to scroll down (out of immediately obvious view) on the LCD. For one nasty moment, I almost wondered whether the ProEFX effects must be presets. Not only are the effects themselves yummy, but their relative routings and applications are sensibly offered—once you figure out the options.

Again, this took a bit of headscratching. Even with a reverb level of zero set on the main Sound Effects page, I was still hearing reverb. Diving directly into the reverb pages themselves, I found that a level of 80 was set and, indeed, increasing or decreasing the value altered the current sound. Meanwhile the MultiFX, similarly at a level of zero on the main Sound Effects page, was not operable no matter what level or setting I tried on its own pages. Very odd.

The answer lay in the ProEFX pages, where an assigned stereo chorus was being sent to the reverb. So, the zero reverb level set on the main Sound Effects page was applying only to the dry signal. What I was hearing was reverb from the ProEFX chorus.

I hope that these mentions of my flounderings illustrate the range and flexibility of this synth's effects options.

SAMPLE OFFERING

As if the instrument's diverse preset library weren't enough, new samples can be loaded into the Equinox to form the basis of new sounds, or simply to be retriggered *virgo intacta*. The process seems admirably flexible yet direct. You can load from floppy disk, from CD-ROM over SCSI (provided you have the SCSI kit installed on your Equinox), or via MIDI Sample Dump.

I loaded in some treasured old Akai S-1000 samples from my days on the road with ABWH and Yes. Boy, did they bring back some memories. Data from disks that had not seen action for ten years loaded up just perfectly; even the mapping and loop points came through. For some reason, samples initially load with a generous coating of Generalmusic effects in place. That's why they sounded so creamy and produced! These effects can be switched off; indeed, after some minor tweaking, the effects were no longer present.

Should you need to, you can assign or reassign incoming samples. There's even a handy autoassign feature, so you can get an instant spread of samples across the keyboard. Samples can be refined in terms of tuning, loop point, truncation, and gain. And the sample-

You have an almost overwhelming amount of programming control.

rate-control parameter on the Edit Voice page gives you precise control over tempo or tuning for sampled drumbeats or vocal passages.

Given the high level of sample tinkering you can do, and the fact that the Equinox screen actually displays sample waveforms, the omission of user sampling seems odd. I imagine that Generalmusic's updatable OS may well include such a feature on a later revision.

Sample formats that the Equinox 61 can currently import include WAV, AIFF, Sound Designer, Akai, Kurzweil, and MIDI SDS. Other formats, such as Roland, Ensoniq, and E-mu, are slated to appear in due course. Though other workstations have paid only lip service to sampling, the Equinox 61 tackles the issue head-on and deserves to reap the rewards accordingly.

THE FUN STUFF

The word groove is almost as overused as *phat* these days (and I speak as a guilty party). Despite its ubiquitous moniker, the Grooves feature of the Equinox 61 is one of those "didn't know I needed it until you gave it to me" facilities.

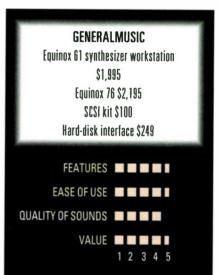
Equinox Grooves are MIDI-recorded drum loops, bass lines, and keyboard (or other accompaniment) parts that you can trigger directly from the keyboard and mix and match in seemingly limitless permutations. You can use the feature as a glorified drum machine to inspire some writing. You could even use it live by holding down a Groove or two in the bottom octave of the keyboard, for example, while playing parts or chords with your right hand.

The Equinox offers more than 1,000 Grooves to choose from, in styles that focus on dance music from hip-hop and R&B to techno/drum and bass. Grooves can be used in isolation (just call up a cool drum pattern) or amalgamated with other grooves. The splendid Shuffle feature randomly mixes different drum and instrument Grooves, allowing you to save any combination that rings your bell.

As you're grooving, sounds remain fully editable using the Quick Edit sliders, and tempo is completely flexible, of course. If you believe, as I do, that the kernel of a great song is so often the accidental (or at least unplanned) juxtaposition of parts, players, and sounds, then this feature will aid in generating a slew of hip new tracks. Less radical, but also inspirational, is the unit's powerful little arpeggiator whose pattern, shapes, and styles can be endlessly modified.

THE SEQUENCER

Workstation sequencers range from scratch-pad idea mongers to full-blown recording devices. Both of these can be handy, but both typically come with limitations of memory, editing ability,



PROS: Great effects section. Onboard organ drawbars. Excellent sample-loading options. Extensive sound-editing abilities. Unique Grooves feature.

CONS: Smallish screen. Preset sounds not cutting-edge. Keyboard action a bit light. CIRCLE #441 ON READER SERVICE CARD

EQUINOX 61

resolution, and usability. I'm nervous about sticking my neck out and saying there isn't a better workstation sequencer around than the Equinox 61's, but a more flexible and useful model doesn't come to mind.

On the face of it, this is a standard 16-track sequencer with all the usual trimmings: replace/overdub/punch-instyle recording, a lot of quantize factors, event editing, track copying and looping, and good resolution (192 ppqn). But unlike with the normal workstation sequencer model, in which sounds you play live are welded into those you've sequenced or are currently sequencing, the Equinox 61 keeps these separate. In other words, you can sequence a drum track, a bass line, or two or three keyboard parts and still physically play a full multilayered Performance patch over the top. And you can do this at any time during the process.

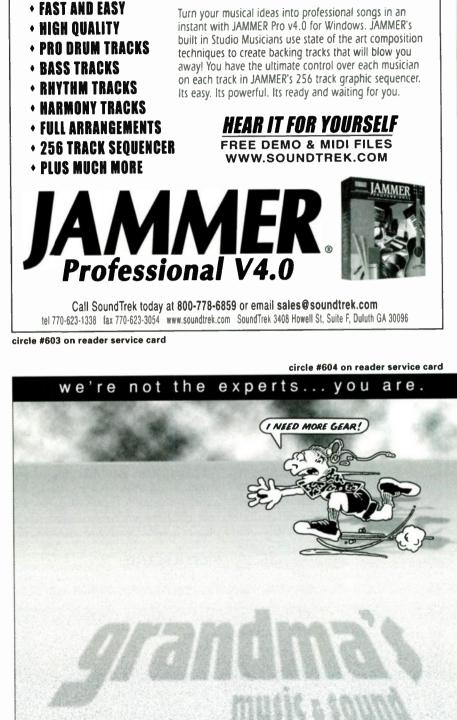
You can also spin through any of the Groove-stored drum tracks to see the effect of different drum styles on your sequence, accompanying your sequence in real time by simply holding down a key and thereby triggering a Groove. The sequencer is even intelligent enough to remain in sync with each Groove no matter what the current tempo. When you consider that each Groove defaults to its own tempo, this is a very smart and very cool ability.

SERIOUS FUN

The more I played with the Equinox 61, the more I liked it. I won't tell you to forget the computer sequencers with their fancy plug-ins, but if you actually do want to record or perform songs (as opposed to just mucking about pretending to do so), then the Equinox 61 will deliver the goods. Its screen is a tad small and sluggish; a display three times the size and twice the resolution would no doubt have improved the product. But for sheer scope, not to mention great fun with Grooves and real-time sliders, the Equinox 61 is an instrument that only fools should ignore.

Julian Colbeck has played keys for Yes, ABWH, Steve Hackett, John Miles, Alan Parsons, and others. But far more daunting was playing (the Equinox) at a recent firstgraders' production of The Ugly Duckling at his son's elementary school.

BACKING TRACKS *in just minutes*



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4

http://www.grandmas.com

RADIUS 20 AND RADIUS 30 Affordable, pro-quality tube EQ and compressor.

By Rob Shrock

HB designed the Radius series of tube processors for personal studios and budget-conscious commercial studios, aiming for a balance between affordability and professional sonic performance. Both the Radius 20 tube parametric equalizer and Radius 30 tube compressor are designed for warming up signals during tracking to digital multitrack tape or hard disk. However, these signal processors could work equally well for analog recording and live sound reinforcement.

These HHB products compete with a veritable flood of tube-based processors that have gained popularity since the ascension of affordable digital recording. It seems we all want tube gear now; let's see whether the Radius 20 and 30 fill the bill.

BIRDS OF A FEATHER

The Radius 20 and Radius 30 share many common traits, such as basic design, sonic performance, size, weight, and color (see the table "Radius 20 Specifications"). Both units employ hybrid solid-state/tube circuitry. Balanced XLR and unbalanced ¼-inch I/O connectors are provided on the rear, and ¼-inch instrument inputs for each channel are thoughtfully and conveniently included on the front. The inputs and outputs have level switches on the back that add 14 dB of gain to the signal (bringing the balanced XLR signals up to +18 dBu), which makes it easy to integrate the units into studios that mix professional and semiprofessional gear.

Both the Radius 20 and Radius 30 provide input controls that govern how much tube coloration they impart to the signal. A yellowish-orange Drive LED lights as the gain increases between 0 and +12 dB, while a red Peak LED indicates that the signal is within 5 dB of clipping. Dedicated output controls, in concert with the input- and output-level switches, provide versatile level matching between each unit and a variety of other equipment.

The manuals are brief—only a few pages long—and contain little information beyond basic operating instructions. However, most essential data can be found in the literature, and the units are easy to operate for those with experience.

EQUAL RIGHTS (AND LEFTS)

The Radius 20 provides two channels of fully parametric EQ, with four overlapping bands for each channel. All bands are peaking only, meaning there are no highpass or lowpass shelving features. However, each band has a variable Q (bandwidth) control that is adjustable between 0.5 and 5 octaves. Each channel has an EQ On button, with accompanying green LED, that engages the filters.

Operation is straight-ahead. The rearpanel balanced XLR (+4 dBu) and unbalanced ¼-inch (-10 dBu) inputs are available simultaneously. The frontpanel instrument inputs override the rear connections so you can quickly plug in a guitar, synth, or other source.

FILTER BLEND

The Radius 20 is a great-sounding and versatile equalizer with enough overlap between each band to achieve some serious tone shaping, especially given that each band provides up to 15 dB of boost or cut. The wider Q settings sound smooth and full, especially in the upper bands.

I loved the Radius 20 for opening up the top end on everything from vocals to keyboards to drum sounds. The narrow bandwidth settings aren't as surgically precise as those on a digital equalizer, but they do an excellent job of notching out problematic frequencies in instruments like bass or overly sibilant vocals. The Radius 20 is particularly quiet for a tube EQ, even when bands are boosted quite a bit.

Depending on the source, you can color signals to varying degrees just by running them through the Radius 20 and adjusting the input drive into the tube stage, even with all the EQ bands set flat. When used in moderate amounts, this results in a perceptible blossoming of the sound that flatters a lot of instruments. I noticed a fuzzy, unfocused quality to the sound whenever I used too much drive, especially when using enough gain to illuminate the red Peak LED. I don't think the Radius 20 is at its best when used as a deliberate overdrive stage, but I like the quality it imparts to most signals at lower gain settings.

Just about every instrument I tried sounded good through the Radius 20. I spent a lot of time reequalizing some acoustic guitars that had previously been cut flat to ADAT, and I really liked the sounds I got. Both male and female vocals sound great through the HHB EQ, and I think it would be an excellent budget front-end EQ for vocal tracking. I especially loved the warmth and character it added to digital synths and samples on their way to being recorded.

The Radius 20 can be subtle or bold, depending on the settings, and I like that very much. By comparison, most digital EQs don't have character, and some analog EQs have too much of a sonic signature for my taste. The Radius 20 falls in the middle and is capable of



HHB's smooth-sounding, tube-based Radius 20 offers four overlapping bands of fully parametric EQ for each of its two channels. The tube circuitry is surprisingly quiet, and the EQ is well suited to a wide variety of applications.

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benefits you won't hear:
The best RFI (radio frequency interference)
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1999 Mackie essigns. All rights eserved. "Mackie" nd the "Running Aan" figure are regstered trademarks if Mackie Designs nc. "VL2" and "IDR ure trademarks of Mackie Designs Inc. mild corrective filtering as well as unique coloration, especially when you experiment with different combinations of the I/O knobs and level switches that affect the tube drive. It's not quite up to creating extreme, techno-style synth filtering, but that's not its purpose. I would use the Radius 20 for just about anything except mastering.

A FEW COMPLAINTS

The Radius 20 sounds wonderful, and its basic operation is simple. However,

a few ergonomic issues started to bug me over time. The silk-screened panel information is small and not that easy to read (on both units), and the markings display minimal reference points. For instance, the high-frequency control of the Radius 20 displays frequency points only at 3 kHz, 6 kHz, 12 kHz, and 20 kHz. Although there are dots in between each of the designations, that's a lot of range to cover with one knob. You could easily boost at 13 kHz on one channel and mistakenly boost at 14 kHz on the other channel without



Critics agree... the YSM-1 monitor is a real contender. But the excerpts below only tell part of the story. Let your own ears be the judge... and then buy yourself a new toy with the money you save!

"...I was immediately impressed with the deep bass response."

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Pro Audio Review Lorin Alldrin, Sept/96

IN THE USA Yorkville Sound Inc. 4625 Witmer Industrial Estate Niagara Falls, N.Y. 14305 "It was a pleasure mixing on the YSM-Is, and the resultant mixes translated exceptionally well to other playback systems...ear fatigue was nonexistent."

"...The YSM-1 reproduces timbres with near pinpoint accuracy."

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"...Mids and high mids were clean and articulate."

"Stereo imaging is very good, resolution is consistent in every frequency range"

> Electronic Musician Brian Knave, July/96



knowing it. For this reason, the Radius 20 wouldn't be my first choice to equalize a finished mix or submix where left/right frequency and phase balance were critical. In addition, the knobs are all the same color, making it easier to grab the wrong parameter when you're working in a hurry.

Although the knobs appear to be of good quality, the switches feel somewhat flimsy; in fact, the power switch on the Radius 20 review unit was stuck and had to be cajoled to the off position. The same type of Alps Spun switches are found on many other pieces of audio equipment, and they would probably not pose a problem in a studio setting or gentle soundreinforcement environment, but I question whether they could withstand sustained abuse on the road.

PUTTING ON THE SQUEEZE

The Radius 30 tube compressor employs the same basic design as the Radius 20 EQ. The I/O configuration is the same. The input controls drive the tube stage and the output control affects both the flat and compressed signals equally. However, the Radius 30 thoughtfully provides a 0 to +20 dB Gain Makeup control, allowing equalenergy comparisons between the pro-



PROS: Smooth, pleasant sonic quality makes it a good choice for numerous applications. Clean, quiet operation when tubes are not driven too hard. Moderate doses of tube sound enhance audio quality. Solid knobs and front-panel instrument inputs.

CONS: Output level affects both equalized and flat settings, making it impossible to compare sounds at equal energy levels. Buttons feel flimsy. Minimal knob markings make exact settings difficult.

CIRCLE #442 ON READER SERVICE CARD

176 Electronic Musician October 1999

circle #606 on reader service card

cessed and unprocessed sounds. The channels can be stereo-linked, meaning that the signal on one channel triggers the compressor for both channels, but each channel's settings (threshold, ratios, I/O levels, and so on) remain in effect.

You can stereo-link the channels so that the same amount of gain reduction is applied to both channels. However, unlike most stereo compressors, you can set parameters other than ratio (such as threshold, attack, and release) independently for each channel.

The threshold is variable between +20 and -20 dB. The unit is sometimes finicky when you are trying to tailor the threshold, because you have to keep in mind that the settings of the input-gain knob and level switch directly affect the overall timbre as well as the operation of the compressor. A few times, I wished the threshold range extended another 10 dB in each direction to fully accommodate either very clean or very hot input settings. I could usually find settings that worked well for most instruments; you just need to bear in mind that the input knob, input level switch, threshold, and gain makeup all affect the tonal character as well as the response of the compressor.

The ratio parameters are reversed from the norm in the United States;



The Radius 20 (top) and Radius 30 have balanced +4 dBu and unbalanced -10 dBu I/O. To make interfacing even easier, rear-panel input- and output-level switches convert the balanced connections to +18 dBu and the unbalanced connections to +4 dBu. The Radius 30 (bottom) also offers a sidechain insert point for each channel.

for example, a 10:1 compression ratio is expressed as 1:10, which Americans interpret as an expansion ratio. In addition, the lowest ratio available is only 1:1.5 rather than 1:1 (which would allow you to add tube coloration without affecting the dynamic range). The highest setting, 1:30, is plenty for most limiting applications. At certain ratios, especially those under 1:5, the Radius 30 seemed milder overall than most compressors. The knob markings on the Radius 30's front panel suffer from the same minimalist approach found on the Radius 20: ratio settings of 1:1.5, 1:5, 1:10, and 1:30 are the only markings besides a dot centered between

each designation. For high ratios, that's
not a problem, but I would prefer more
precise markings between the crucial
ratios of 1:1.5 and 1:5.

INSTRUMENTS UNDER ATTACK

The attack and release times are switchable between only two settings: fast and slow (see the table "Radius 30 Specifications"). However, these are program dependent: the attack and release times are generally shortened by fast transient signals.

Although the settings sound tailored for, and work well with, instruments like guitar, piano, and voice, the available attack and release times aren't ideal for every situation. This makes the Radius 30 a dubious choice for certain instruments, especially sounds with very fast attack transients, like drums. A certain smoothing of the transient through the tube stage takes place, which can be cool, but I didn't care for the sound of drums compressed through the Radius 30.

Although I never got a synth bass sound that I truly loved, the Radius 30 fared very well with a real bass guitar using the fast attack setting and slow release. Boosting the input drive a bit also matured the bass tone, which I passed through a Countryman direct box on the way to the front-panel instrument jack of the Radius 30.

I liked the sound of vocals compressed with the Radius 30. Vocals seem to be an ideal application for this compressor, although you have to hit the unit harder and at a higher ratio than your instincts or experience might indicate. At moderate input levels enough to light up only the orange

Radius 20 Specifications

EQ Bands per Channel	(4) fully parametric
Cut/Boost	±15 dB
Bandwidth	0.5 to 5 octaves (variable)
Frequency Range	low 30 Hz–1 kHz; low-mid 100 Hz–3 kHz; high-mid 1 kHz–12 kHz; high 3 kHz–20 kHz
Audio Inputs	(2) balanced XLR line (+4 dBu nominal); (2) unbalanced ¼" (-10 dBu nominal); (1) ¼" unbalanced aux (front panel)
Audio Outputs	(2) balanced XLR; (2) unbalanced ¼"
Output Gain Control	0 dB at center; ±20 dB gain variable
Input Level Switch (rear)	converts balanced inputs to +18 dBu and unbalanced inputs to +4 dBu
Output Level Switch (rear)	converts balanced outputs to +18 dBu and unbalanced outputs to +4 dBu
Frequency Response	10 Hz-40 kHz (+0/-1 dB)
Noise	-80 dBu (22 Hz–22 kHz, line input @ 0 dB gain)
Dynamic Range	106 dB (line input @ 0 dB gain)
Power Supply	internal 110–120 VAC, 60 Hz or 220–240 VAC, 50 Hz (selectable)
Dimensions	2U rack-mount x 7.9" (D)
Weight	5.5 lbs.

RADIUS 20 AND 30



The Radius 30 includes a full-featured, 2-channel tube compressor with sidechain and a bare-bones expander/gate. Even when the channels are stereolinked, the left- and right-channel parameters remain independently active.

Drive LED—the Radius 30 can eliminate a lot of the harshness associated with some condenser microphone and digital recorder setups, and this feature alone may justify its use.

Although the Radius 30 seems to command a "turn the knobs and listen" approach rather than adjusting by the numbers, the compressor gives you enough control to shape some nice tones from a variety of instruments. I like that you can adjust the gain makeup to allow direct comparisons with the uncompressed signal (which are very important to make). The VU meters can display output level, in addition to gain reduction, making it easy to roughly match compressed and uncompressed levels.

CLOSING THE GATE

The Radius 30 includes a rudimentary expander/gate. One knob alone controls this feature, which amounts to adjusting the threshold until the Gate Shut LED lights up. The expander/ gate functions as you would expect, muting the signal entirely during silent passages.

Although I rarely employ gates when I'm recording, I've found that they are good for some applications, such as a noisy electric guitar rig. No envelope parameters are available for the Radius 30's gate, so initial attacks are sometimes lost on sharp transient material (such as drums) as the gate opens up. As with the compressor attack and release times, the gate feature seems tailored for vocals and other instruments that have mild envelope shapes.

ON THE SIDE

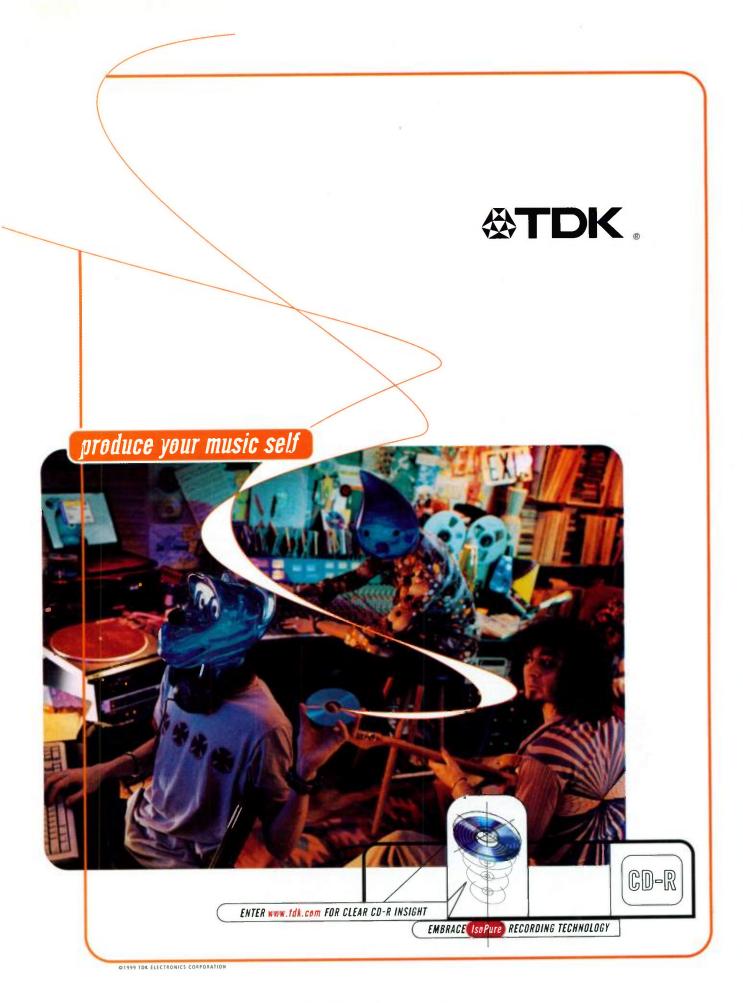
The Radius 30 provides two ¼-inch TRS sidechain inserts, one for each channel. The level of the send is affected by both the input control and inputlevel switch, so it is very versatile and can be patched to a variety of professional and semiprofessional gear. Once again, you may have to twiddle with the various gain stages to optimize the compressor's operation, send/return levels, and overall tonality of the source.

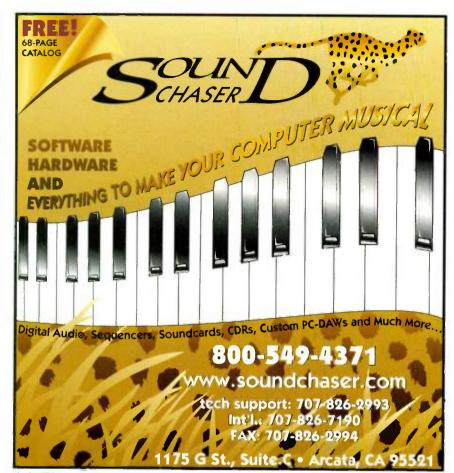
I created a very compelling vocal sound by using channel 1 of the Radius 20 equalizer to contour the compression response of one Radius 30 channel via its sidechain. I then equalized the signal through channel 2 of the Radius 20, which flowed back into the remaining channel of the Radius 30 for overall level smoothing. I kept all of the input gain levels on the low side (lightly hitting the orange LEDs), and the accumulation of tube stages gave a full and earthy character to a female vocal. Though not pristine or pure, it was a great sound for a rock or alternative voice. In a more complex signal path like this, the variety of tube coloration available at the various stages can provide some very interesting sonic characteristics.

FINAL STAGE

Both Radius processors offer a lot of processing power for the price. They are especially quiet considering that they employ tube stages. Used in moderation, the tube stages add some color and interest to most sounds, although I don't care for their sound when the tubes are pushed too hard. The various stages of I/O level control make both units very flexible for interfacing with other gear, and the front-panel instrument jacks make them even easier to hook up.

	pecifications
Threshold	-20 đBu to +20 dBu
Attack	fast (0.5 ms) or slow (20 ms), switchable
Release	fast (40 ms) or slow (2 sec.), switchable
Ratio	1.5:1 to 30:1 variable
Makeup Gain	0 to +20 dB
Display	VU meter; output level (0 VU = +4 dBu) or compression (switchable)
Audio Inputs	(2) balanced XLR line (+4 dBu nominal); (2) unbalanced ½* (-10 dBu nominal); (1) ¼* aux (front panel)
Audio Outputs	(2) balanced XLR; (2) unbalanced ¼"
Other Connections	(2) ¼" TRS send/return sidechain inserts (-2 dBu nominal)
Output Gain Control	0 dB at center; ±20 dB gain variable
Input Level Switch (rear)	converts balanced inputs to +18 dBu and unbalanced inputs to +4 dBu
Output Level Switch (rear)	converts balanced outputs to +18 dBu and unbalanced outputs to +4 dBu
Frequency Response	10 Hz-40 kHz (+0/-1 dB)
Noise	-80 dBu (22 Hz–22 kHz, line input @ 0 dB gain)
Dynamic Range	106 dB (line input @ 0 dB gain)
Power Supply	internal 110–120 VAC, 60 Hz or 220–240 VAC, 50 Hz (switchable)
Dimensions	2U rack-mount x 7.9" (D)
Neight	5.5 lbs





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HHB COMMUNICATIONS Radius 30 tube compressor \$749 each FEATURES EASE OF USE AUDIO QUALITY VALUE

RADIUS 20 AND 30

PROS: Clean, quiet operation when tubes are not driven too hard. Moderate doses of tube sound enhance audio quality. Solid knobs and front-panel instrument inputs. VU meters can display either gain reduction or output level.

CONS: Minimal attack and release settings aren't ideal for all applications. Expander/ gate clips transients' attack. Buttons feel flimsy. Minimal knob markings make exact settings difficult.

CIRCLE #443 ON READER SERVICE CARD

If I could buy only one of these two devices, I'd take the Radius 20 over the Radius 30. I think the Radius 20 is an excellent and versatile equalizer, capable of subtle to rather bold tonal shaping. There is very little to criticize, other than its delicate switches, especially considering that the street price makes this almost a no-brainer. I wouldn't hesitate to use the Radius 20 in a professional application.

The Radius 30 is good for certain applications, such as smoothing out vocals and guitars, but I wouldn't choose it as my sole front-end compressor to a digital recorder. Not only is it not quite versatile enough, it also seems too mild as a compressor unless you really pour it on. However, one of my guitarist friends liked the Radius compressor better than the equalizer. Go figure. Furthermore, the two devices work extremely well together. Considering their competitive prices, they both deserve a test-drive when you're in the market for tube processing.

Producer and songwriter Rob Shrock is the musical director for Burt Bacharach. However, being an electronic musician, he was the only member of Bacharach's band not in the Carnaby Street scene in Austin Powers: The Spy Who Shagged Me.

Let the "critics" tell you how easy the Spirit Digital 328 mixer is to use...

Spirit's Digital 328 represents a new way of thinking in digital console design—it bridges the gap between analog ease-of-use and digital sound quality and features.

George Petersen of Mix says: "There are more than a dozen entries in the 'low cost' category of digital consoles, but in terms of pricing, performance and fast, logical interface, the Digital 328 clearly sets itself apart from the pack."

Take a few moments to read what he and other "critics" say about the Digital 328. Then, go to www.spiritbysoundcraft.com on the web for more information. If you're in the market for an affordable digital console, you need look no further.

On 328's user interface:

"The 328 is a real console interface that immediately feels as close to your comfortable old analog board as you could want... the consideration that has gone into every single button, knob and interconnect is striking." – Recording

"I liked the user interface a lot, and given that the most-requested features and digital interfaces are all included, the price is excellent." – Electronic Musician

"I like this board. It has a logical interface and enough knobs for fast operation (as such it could be ideal in a live performance or broadcast situation) while its audio performance is clean enough for any recording application." – Mix

On 328's E-Strip:

"The invention of the E-Strip is a stroke of genius, [giving] instant access to all controls at once on the selected channel." – Audio Media

"The 328 is fast and intuitive, thanks in large part to its 'E-Strip' interface. There are no subroutines or hidden pages; anyone familiar with an analog console can sit down at a 328 and be working in a matter of minutes." – Mix

"With Spirit's clever E-Strip design, this digital desk has the feel of an analogue." – The Mix (UK)



three two eight

www.spiritbysoundcraft.com

On 328's equalization:

"... To my ears, this is one of the most musical sounding digital EQs I've ever heard." – Recording

"[One] of the best features of the desk: carefully tailored to provide control ranges similar to those on a top-notch analogue console, it is (dare I say) very musical." – Audio Media

On 328's effects:

"A strong selling point for this unit is the pair of built-in stereo Lexicon effects... Having quality effects in the digital domain makes for clean sounds." – Electronic Musician

On 328's automation:

"The automation is straightforward to set up and works well." — Audio Media

"Between the user setups, snapshots and dynamic automation, the 328 remembers everything except the line-input trims and 100Hz rolloff switches. It's easy to get used to this way of working." – Electronic Musician

On 328's connectability:

"Clearly, the Digital 328 provides a multitude of configuration options suitable for project studios, post-production facilities, radio stations and even live applications." – Electronic Musician

"The 328 interfaces to practically anything digital." – Recording

On 328's unbeatable value:

"All in all, the British have indeed landed with a winner. The more you use this board, the more you will discover its depth and power. With one of these consoles, you could start a musical revolution of your own." – Electronic Musician

"This mixer packs a mighty punch for \$5,000 [suggested list price]. It sounds excellent, does an excellent job of untangling all the various digital formats in use, and has an excellent interface. A bold step forward in digital console design." – Recording

"I like this desk! There's nothing better out there right now than the 328." – The Mix (UK)

On 328's mic preamps:

"The mic preamps have plenty of headroom... I was surprised at the clarity of the most subtle nuances of the percussion, including the last hint of sound from the bell trees and chimes."

- Electronic Musician

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"All in all, it is a delight to use—a real peach!" – Audio Media

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A A R D V A R AARK 20/20+ (WIN) A well-constructed recording system with

outputs to spare.

By Zack Price

he Aardvark Aark 20/20+ is a component recording system that features rock-solid construction and high-quality sound. Like other products in its niche, the Aark 20/20+ includes a multichannel sound card, outboard A/D/A converters, and a breakout cable. Though its individual features don't always place it at the top of the heap in every category, it's a worthy successor to the company's successful Aark 20/20 card.

With a large number of similarly configured systems available today, prices for such products have become fairly competitive. Still, selecting the right card for your needs can be a difficult process. I'll discuss what the Aark 20/20+ has to offer, and why it might be a good choice for you.

THE HOST WITH THE MOST

The heart of the Aark 20/20+ is a singleslot PCI board, which Aardvark calls the "host card." Unlike some audio cards, the Aark host card's circuitry is shielded in a protective casing (see Fig. 1). This shielding, along with converters that are housed in a separate external unit, reduces RF interference from within the computer. Moreover, the card's solid feel lends it an air of indestructibility. I can't say whether that impression is justified, but it does inspire confidence in the card's construction.

Installing the host card involves several steps. First, insert the card into an open PCI slot, then turn on your computer and install the low-level drivers as instructed by the dialog boxes. (Because the card is a Plug and Play device, Windows 95 or 98 should automatically detect its presence when you boot up your system. See the list of frequently asked questions on Aardvark's Web site for a discussion of known incompatibilities.)

After the low-level drivers are installed, you'll need to install the Aark Manager software. This utility is used to configure any Aark 20/20-series or Aark TDIF cards that you may have in your system. (You can use the Aark 20/20+ with other Aardvark products. such as the original Aark 20/20 card.) In fact, it is possible to use four Aark cards in one system simultaneously. Just be sure to connect the proper converter box to its corresponding card. For example, the Aark 20/20+ supports both balanced and unbalanced lines, whereas the previous version supports only unbalanced lines. Because the converter boxes are designed for use with a particular host card, any mismatch will cause the system not to work.

THINKING OUTSIDE THE BOX

A 6-foot, shielded 25-pin cable connects the host card to the back of the Aark 20/20+ interface box. The cable length is more generous than what I've seen in some other card systems, but I would still prefer that it be longer. Although Aardvark will not guarantee



The Aardvark Aark 20/20+ multichannel recording system is a well-built card with external 20-bit A/D/A converters. The card includes S/PDIF and Toslink I/O and sports a range of enhancement options.

Aark 20/20+ Minimum System Requirements Pentium-133 with open PCI 2.1–compliant slot; 32 MB RAM; Windows 95 or 98; out-

board mixer or sound source

performance if a longer interface cable is used, some users report that they have successfully used 10-foot cables with the Aark 20/20+. If you decide to build your own cable, make sure it is properly shielded and wired. Otherwise, you will seriously damage the Aark 20/20+ and quite possibly your computer as well. Also keep in mind that your warranty becomes void if you use a host cable not issued by Aardvark.

The interface box (see Fig. 2) is powered via the host card, which draws its power from the computer. This reduces the possibility of ground loops and makes for clean, quiet operation. The box's eight shielded 20-bit A/D/A converters further minimize RF interference from other elements within the computer system. Moreover, the frontpanel analog I/O connectors use TRS jacks, which also helps prevent noise. The front panel provides a pair of RCA jacks for electrical S/PDIF I/O. They can be used simultaneously with the eight analog inputs and outputs for a total of 10-channel operation.

The back panel of the interface box provides optical S/PDIF I/O, which can also be used for 8-channel Lightpipe I/O when the \$299 ADAT expansion option is installed (more details to follow). Next to the optical jacks are the MIDI In and Out ports. Though you can use the ports as your main MIDI interface, one set of 16 MIDI channels may not be enough for many users. In all likelihood, most users will employ the MIDI ports strictly for MIDI Time Code synchronization. Moreover, users with the ADAT option installed will probably use the MIDI ports with the ADAT sync adapter, which has a pair of MIDI jacks on one end and a 9-pin ADAT sync jack on the other.

The interface box also contains two BNC connectors for word-clock I/O. You can use an external word-clock signal for a timing reference, or you can link multiple Aark units in a masterslave chain to ensure that they all operate in perfect synchronization.

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KEEPING TABS

You control the Aark 20/20+ system's signal routing by way of the included *Aark Manager* software. Clicking on the *Aark Manager* icon opens the Aark 20/20+ Control Panel, which is divided into five tabs. The first is the Hardware tab, which shows, among other things, the relationship between the inputs and the selected output sources.

Most of the time, you route inputs to their corresponding outputs. However, there may be times when you want to route inputs to the Monitor Bus outputs. For instance, you may want to create a headphone mix that is separate from the signals going into the computer. Or perhaps your audio program does not let you monitor an input signal while it is being recorded. In either case, just use the Hardware tab to select the analog or digital outputs you want to use as the monitor bus, and select Monitor Left or Right from the drop-down menu in the Output Routing Source column.

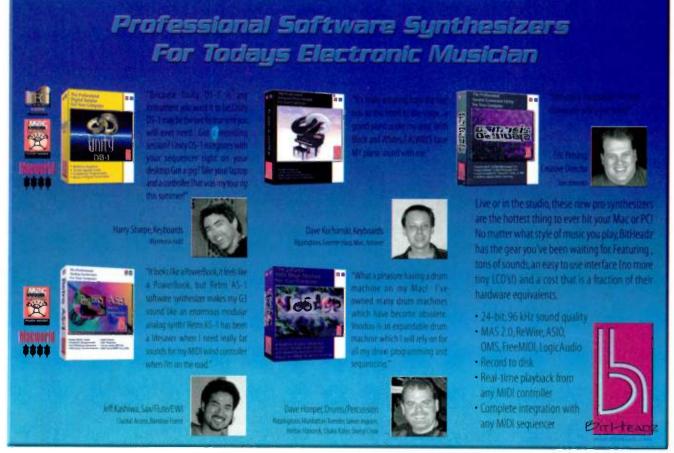
Besides letting you configure audio routing, the Hardware tab also meters

Aark 20/20+ Specifications

INTERFACE BOX	
Analog I/O	(8) ¼" inputs (+4 dBu/-10 dBV); (8) ¼" outputs (+4 dBu/-10 dBV); switchable balanced/unbalanced
A/D Converters	20 bit; 128x oversampling; 30–50 kHz
D/A Converters	20 bit; 128x oversampling; 30–50 kHz
Digital I/O	S/PDIF (RCA), Toslink (optical)
Sync	word-clock I/O (BNC); MIDI; S/PDIF digital; Toslink
Drivers	ASIO; Windows MME
Total Harmonic Distortion + Noise	0.002% @ 1 kHz
Frequency Response	7 Hz–22 kHz, ±0.5 dB
Dynamic Range	100 dB, A weighted
Modes	full-duplex simultaneous record/play
HOST CARD	
Card Type	PCI (5")
Onboard DSP	24 bit, 80 MIPS
Connection to Interface Box	6-foot, custom 25-pin shielded cable

audio levels for each analog input and S/PDIF stereo digital input. Furthermore, you can monitor any input or output source pair using the meters in the lower left corner of the Hardware tab. If you select Tone as an output routing source (and as a metering source), you can use this feature to calibrate your mixer with the Aark 20/20+. If you select Silence, you can measure the signal-to-noise ratio of your outboard setup.

There are other options for choosing analog or ADAT inputs and for ADAT Optical or Toslink (optical

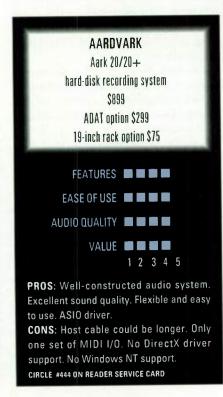


AARK 20/20+

S/PDIF) outputs. When selected, the input icons read ADAT 1 through ADAT 8. Additionally, each ADAT output appears in the drop-down menus for each output routing source. Unfortunately, you cannot combine the analog 1/O with the ADAT I/O to achieve 16-channel operation through a single Aark system. You may use the ADAT outputs while using the analog inputs, and ADAT inputs along with the analog outputs. However, when you use the ADAT outputs, the same audio is sent directly to the analog outputs.

The Software tab further extends the Hardware tab's capabilities by allowing control of the software record channels and playback/monitor routing. Although it may sound as if this tab replicates the functions of the Hardware tab, this is not the case: the Software tab manages the connections between the software device drivers (that is, the Windows MME drivers) and the hardware I/O. The presence of status indicators in this window is helpful because it lets you see precisely which wave drivers are being employed for both recording and playback at any given time. Another screen provides versatile routing and mixing options, which means you may never need to physically rewire the inputs to your system.

The I/O Levels tab is used to set any combination of the Aark 20/20+ inputs





circle #613 on reader service card

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or outputs to either unbalanced or balanced levels. (You can combine different types of inputs if needed.) The Keys tab is where the Aark 20/20+ ASIO serial number is located; you'll need this to obtain your free ASIO driver key from the company. The Advanced tab offers additional control over the Aark 20/20+'s performance—for example, switching the S/PDIF Channel Status between Consumer and Professional mode, selecting from among various warning messages that the system uses (if you attempt to play an incompatible file, for instance), and customizing the ASIO configuration settings. You can also restore all the default settings with a single mouse-click.

IF I WERE KING

The Aark 20/20+ system worked admirably with my computer and almost all of my software. I had some glitches with Cakewalk Pro Audio 8 (until I got the 8.04 update patch), but I don't perceive this as a problem originating in the Aark 20/20+ hardware or software. The sound quality was excellent in all instances, despite the fact that the unit has "only" 20-bit A/D and D/A converters.

However, I feel that a few changes would improve the quality of the system. First, I would like the inclusion of at least a 10-foot cable to connect the host card to the interface box, for greater flexibility in placing the box in a studio envi-



FIG. 2: The back panel of the Aark 20/20+ interface box has optical S/PDIF I/O---which can also be used for 8-channel Lightpipe I/O when the ADAT expansion option is installed—as well as MIDI In and Out ports and a pair of BNC connectors for word-clock I/O. The front panel provides eight channels of analog I/O and two channels of electrical S/PDIF.

ronment. I would also prefer to see at least two sets of MIDI I/O.

I'm not particularly concerned about the use of 20-bit converters. I'm sure that some readers might find my stance surprising. Excellent 20-bit performance is sonically superior to mediocre 24-bit performance, and, as I mentioned, the card sounds great. I do wish that the Aark 20/20+ had DirectX drivers, though. (DirectX drivers are expected to be available in the fourth quarter of this year.) In fact, they should be standard on any professional audio card. After all, many modern audio applications can use DirectX drivers to enhance their performance.



FIG. 1: The circuitry of the Aark 20/20+ PCI host card is well shielded in a protective casing to reduce exposure to RF interference from within the computer.

(Don't confuse the hardware performance enhancements provided by DirectX with the plug-in standard of the same name. The Aark 20/20+, like any Windows sound card, is perfectly happy if your software employs DirectX plug-ins.) The inclusion of ASIO drivers, which can improve performance, is commendable. (A special driver for NemeSys GigaSampler allows very low latency with the Aark 20/20+.)

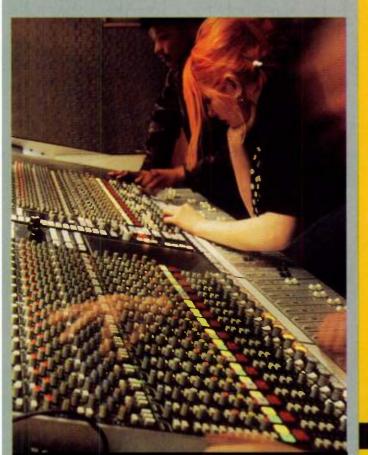
All told, I believe that the Aark 20/20+ is a solid performer for the money. Although there are other good systems on the market that equal it in features and quality, I would have no problem recommending the Aark 20/20+. It sounds great, and it doesn't have any of what I would consider to be true flaws.

The Aark 20/20+ wouldn't be my first choice if I had an all-digital studio or if I used multiple MDMs to record and play tracks—other multiport digital-only cards would be more cost-effective in this particular circumstance. Also, owners of TDIF-based systems might do better to use the Aark TDIF card instead. However, for those who use one or two MDMs and do computer-based harddisk recording, the Aark 20/20+ is a worthy addition to the studio. Best of all, it lets you take advantage of both recording technologies.

If you're looking for a well-made multichannel audio-card system for your Windows PC, then check out the Aark 20/20+. It may be just what you need.

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E T E K

A fold-and-go mixer with power to spare.

By J.J. Jenkins

tek, the professional-audio division of the Italian musical-instrument manufacturer Eko, evidently took Apple's "think different" admonition to heart when it designed the NoteMix MA 400. This cute, portable powered mixer resembles a laptop computer, folding out to provide mix controls on top, where a laptop's display would reside, and a surprisingly powerful amplifier in the bottom section. It looks like no other mixer I've seen.

LILLIPUTIAN PANELS

Constructed of beefy, gray sheet metal, the NoteMix MA 400 is a sturdy, visually understated unit with blue silk-screened control-panel graphics. The power-amp section, decorated with a large Etek logo, features air vents on the top and front and provides three small rows of yellow LEDs positioned to illuminate the control face in the dark. Except for eight prefader-listen (PFL) buttons, all the controls are small faders or sliding switches. The unit comes in a highquality, padded Cordura carrying bag, complete with shoulder strap and partition for the power cord.

Six mic/line input channels, laid out in rectangles and labeled CH1, CH2, and so on, dominate the control panel. Each channel provides a slider control for gain, 3-band EQ, three aux/effects sends, pan, and volume. Beneath each gain slider is a red clip LED.

The graphic equalizer bands—labeled Low, Mid, and High—provide 15 dB cut/boost for the low and high (shelving) bands and 10 dB cut/boost for the mid band, which is a peaking filter centered at 700 Hz. The first aux/ effects slider, labeled Aux/DRX, is dedicated to the onboard digital effects; the other two are labeled Aux2 and Aux3.

Just below the EQ section is the PFL (solo) switch. Curiously, it is a momentary switch—which is great if you're worried about leaving it on accidentally, but not too cool if you have to move around or make adjustments elsewhere with one hand while holding the switch down with the other. (There's always duct tape, I suppose.)

Another oddity is the pan slider, which is oriented vertically rather than horizontally (left is on the top, right on the bottom). This counterintuitive design will most likely take some getting used to.

Inputs 7 through 14 are controlled by two sections, Submix A and Submix B. Each submix panel provides a 3-slider aux/effects section (just like the mic/line panels), a PFL switch, and separate left and right volume sliders. The submix channels are, of course, helpful, allowing you to run a separate monitor mix, return effects, and so on. However, the modified 4-bus setup is limited by the fact that boosting channel 7 also boosts channel 9, and boosting channel 8 also boosts channel 10.

The far-right panel of the control face contains the master and monitor sections, the DRX digital effects processor, and the LED metering. You control the master section with separate left and right volume faders. The so-called inear monitor system, controlled by a single fader, has a three-way switch for changing between PFL, L+R (master), and aux 3. The LED metering is also switchable between three sources: PFL, L+R (master), and aux 2 and aux 3.

Except for the PFL setting, the monitor system is postfader. Therefore, if your master or aux 3 volume is all the way down, the monitor and headphones get no signal, either. One obvious problem with this arrangement is that you can't set up a monitor mix unless the main mix is also turned up. A prefader switch would be welcome.

EFFECTIVE IMMEDIATELY

The DRX digital effects processor provides a switch for three different effects modes: Delay, Reverber, and Surround. A little LED illuminates behind one of the three names, indicating which effect is selected. The Delay and Reverber are available individually, or you can use them simultaneously when the switch is set to Surround.



Etek's NoteMix MA 400 14-channel, notebook-style powered mixer is portable enough to tuck under one arm. It provides 3-band EQ and onboard digital reverb and delay. The effects are weak, but the overall sound of the mixer is quite good.

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The NoteMix provides six dedicated mic/line channels and four double-duty channels on TRS jacks. The ¼-inch line inputs double as inserts when the XLR channels are used for mics.

There is also an Operating Mode switch for assigning the effects, whether to L+R/Aux/Aux3, L+R/Aux1, or Aux1/Surround. A third switch lets you choose between fast, medium, and long settings for Delay and Surround and between chorus, small hall, and large hall for the Reverber. Levels for the Delay and Reverber can be set inde-

pendently using separate faders. Finally, a fader controls the effects output level, and a PFL switch is provided for the effects section.

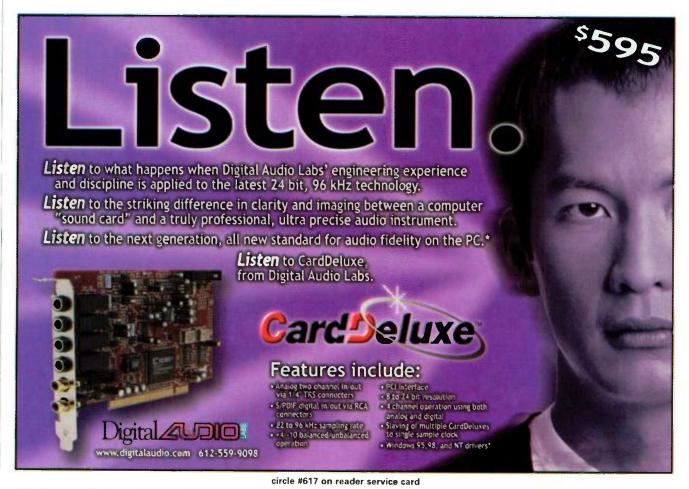
The effect billed as Surround isn't surround sound as in 5.1, nor is it an effect. Rather, it simply allows you to route cables from the aux 1 outputs on the back and send them to another amp and set of speakers. Aux 1 is posteffect, so you can apply a little delay to get a "delay tower" effect—a kind of faux surround, I suppose. (Of course, you could also bypass the main mix with sound sources going to the aux 1 output, sending them to a second pair of speakers instead, for that old-fashioned "quad" sound.)

THE BACK SIDE

Channels 1 through 6 offer both XLR and ¼-inch mono input jacks, located on the rear panel of the NoteMix. Each mic channel is equipped with "unswitched" (meaning that it's always on) phantom power, an idea I'm not fond of. I would prefer a switch, even if only a global one.

The NoteMix's mic inputs override the line inputs. Interestingly, the line inputs double as TRS insert jacks when the XLR mic inputs are being used.

Four ¹/₄-inch output jacks handle the eight submix channels. These TRS jacks are set up to accept channels 7/9, 8/10, 11/13, and 12/14. Stereo breakout connectors (or, as they are called in the manual, "Etek cables")



190 Electronic Musician October 1999

NOTEMIX MA 400

are required to access the stereo capabilities. (Stereo breakouts are available from most audio suppliers, of course. They look like Y-adapters, with a ¼-inch TRS jack on one end and two mono jacks or plugs, in your choice of format, on the other.) The four jacks can also be used as separate mono inputs using standard cables. In this case, they feed only the first channel of each jack.

There are three aux outputs, one stereo and two mono. These share two stereo, ¼-inch TRS aux-out jacks. Aux 1 (the one with DRX effects) can be used in mono with a standard cable or in stereo with a Y-adapter. Aux outputs 3 and 4 share the other TRS jack: if you use a mono cable, you'll get aux 3; a Y-adapter gives you access to both mono auxes.

The back panel also provides stereo inserts for the main mix, tape outputs on two RCA jacks, and a headphone

NoteMix MA 400 Specifications

MIXER	
Mic/Line Input	+40 dB
Frequency Response	mic: 75 Hz-20
	kHz (-3/+1 dB);
	line: 20 Hz-20
	kHz (±1 dB)
Dynamic Range	62 dB
Equalizer (highs)	high: ±15 dB @
	10 kHz (shelving);
	mid: ±10 dB @
	700 Hz (peaking);
	low: ±15 dB @
	80 Hz (shelving)
Distortion (THD+N)	<0.1%
Crosstalk	>60 dB @ 1 kHz
Signal-to-Noise Ratio	>122 dB
Mix Output Level	22 dB (max)
Aux Output Level	0 dB
Tape Output Level	-10 dB
POWER AMP	TON PROVE S
Output Power	190W/side RMS
	into 4Ω
Frequency Response	30 Hz-20 kHz
	(±1 dB)
Signal-to-Noise Ratio	>92 dB
Total Harmonic Distortion	<0.03%
UNIT	1945 C 200
Dimensions (closed)	3" (H) x 11.5"
	(W) x 9.5" (D)
Weight	17 lbs.

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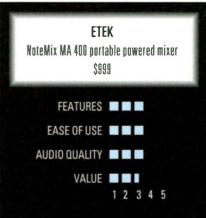
minijack that the manual mistakenly calls an "in-ear monitor." Few pro headphones have minijack connectors, so you may need an adapter to use your premium cans. There is also an "Extention Port" (sic), which is an 8-pin connector that allows you to connect the NoteMix to the Notextention, a separate 24-input module with six extra headphone jacks. However, the Notextention is not available in the United States at this time.

The main speaker outputs for the NoteMix are on Neutrik Speakon jacks, which are pretty much the standard in Europe. I prefer these connectors to others, but for many U.S. buyers, another cable purchase will be necessary. Also on the rear panel is the standard IEC power-cord socket and rocker-type on/off switch, which is conveniently located on one end of the rear panel.

THAR SHE BLOWS

The NoteMix's overall sound quality is very good, including the mic preamps and sledgehammer-style EQ. (By that, I mean that the EQ is potent, requiring only incremental fader moves to effect big changes in the sound.) But two things mar the unit's sonic pedigree: the effects and the fan noise generated by the amp.

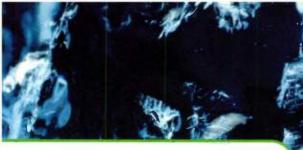
My initial reaction when auditioning the effects was that I have stompboxes that sound better. Indeed, the reverb,



PROS: Compact, cute, and powerful. Sturdy construction.

CONS: Amp produces loud, constant fan noise. Effects are limited and of poor quality. PFL uses momentary switches. Monitor is postfader. Phantom power cannot be switched off.

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NERTISER INDEX

ADVERTISER RS# PAGE . 105 Akai 557 AKG (C 414B/ULS) . . . 43 520 AKG (PT 60) . 547 93 Alesis (QS Series) 512 31 Alesis (ADAT).... 542 85 Alesis (M1). 601 169 Antares 552 99 Anthro Corporation 16 . 109 Apogee Electronics..... 560 Arboretum Systems. 554 102 Armadillo Enterprises (Nord Modular) 571 125 Audio-Technica 641 116 633 203 Audix.... B&H Photo-Video 634 204-207 Big Briar 591 154 BitHeadz Inc. 612 184 Cakewalk Music Software 514 41 Canadian Music Expo 620 192 Caruso Music 614 185 Centralis 543 87 CM Automation..... 553 101 Conservatory of Recording Arts & Sciences. . 625 198 576 131 CTI Audio..... 616 . 189 DataViz 546 92 dbx Professional Products (JT50). 517 36-37 dbx Professional Products (Quantum) 527 56-57 507 14-15 Digital Piano Buyers Guide. 183 623 Disc Makers..... 196 Earthworks 588 150 83 Edirol. 541 FGO-SYStems 597 162 Electrix 545 . 89 Emagic . . 525 53 Digital Audio Labs 617 190 Empirical Labs 530 62 E-mu Systems (E4 Ultra). 504 . 9 E-mu Systems (Proteus 2000) . 602 171 Event Electronics (PS5) 531 63 Event Electronics (Røde NT1) . 596 161 Focusrite . 594 157 Frontier Design Group 523 . 51 Furman Sound 572 126

ADVERTISER	RS#	PAGE
Gadget Labs	534 .	69
Genelec	561 .	11
Generalmusic	563 .	115
Glyph Technologies		
Grandma's Music&Sounds	604 .	173
Guitar Center		
Guitar Center's Rhythm City	619 .	191
Hafler	. 584 .	145
HHB Communications (Radius 40).	529 .	61
HHB Communications (CDR-850)	627 .	199
Interstate Music	502	154
iSong	. 535 .	71
IKI Protectional		133
JoMoX	580 .	139
Keytax Software	585 .	147
Korg (Triton)	511	22–23
Korg (N1)	518 .	39
Korg (Kaoss)	595 .	159
Kurzweil Music Systems	578 .	137
Leigh's Computers	630 .	201
Lexicon	503 .	4–5
Lexicon Product Giveaway	624 .	197
Live Sound for Musicians.	•	201
Lucid Technology	539 .	11
Lynx Studio Technology		
Mackie (D8+B)		
Mackie (HR824)	. 📷 590 .	153
Mackie (VLZ Pro)		
Mark of the Unicorn		
MARS		
Merrill's Music		
MicroBoards Technology		
Midiman/M-Audio		
MMADP	·	165
Music Business 2005.	636 .	218-219
Music Industries (StudioLogic)	540	81
Music Industries (Blue Chip)	551	
Music Industries (Quik Lok)		
Musician's Friend		
Musitek		
Neumann/USA		
NHT.	519	
Opcode		
Peavey (DPC-1400X)	546 .	94

ADVERTISER	RS#	PAGE
PG Music (Band-In-A-Box 8.0)	515	32-33
PG Music (Oscar Peterson CD-ROM)	600 .	167
QCA	631 .	202
Quasimidi		
Rane	567 .	119
Recording Industry Sourcebook . 📕		148
Reliable Music	558	106
Rocket Network	. 513	28-29
Roland	509	18-19
Samson Technologies	. 589	151
SAE Institute of Technology	615 .	187
SeaSound		
Seer Systems	. 582	141
SEK'D America	632	202
Sherman		
Shure	574 .	129
Sincrosoft USA	556 .	103
Sonic Foundry (Vegas Pro)	. 505.	10–11
Sonic Foundry (Acid)		
Sony		
Sound Chaser	564	180
Sound Quest	581	140
SoundTrek		
Spirit	610 .	181
Steinberg North America (Cubase)	. 506	13
Steinberg North America (En-Voice)	583	143
Stipko Media	550	97
Sweetwater Sound #1	510 .	
Sweetwater Sound #2	637 .	220-221
Sweetwater Sound #3	638	222-223
Sweetwater Sound #4	639	224-225
Syntrillium Software	. 622	193
Tannoy	528	60
Tascam		
Taxi	579 .	
TC Electronic	537	75
TC Works	. / 508 .	17
ток	608	
Waldorf	566	117
waves	333 .	107
WIZ00	618 .	191
Yamaha (CS2X)		
Yamaha (01V)		
Yorkville	606 .	176

RATE THE ARTICLES IN THIS ISSUE! October 1999

We want to know what you think of the articles in *Electronic Musician*! Now you can use your reader service card to give us feedback about EM's editorial coverage. We have assigned a rating number to each of the main articles in this issue. Please select a rating for each article and circle the appropriate number on your reader service card:

Please select ONE rating number per article Not So Didn't Very hat Read A. "Life in the Slow Lane, " p. 44 701 702 703 704 8. Cover Story: "Equal Time," 705 706 707 708 p. 58 C. "Mastering Continuity," p. 78 710 711 709 712 D. DIY: "Build the EM -10/+4 Level Converter,* 713 714 715 716 p. 90 E. Performing Musician: 717 718 719 720 "Music for Airports on Stage," o. 132 F. Final Mix: "Pass the Salt. Please" 721 722 723 724 p. 226

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194 Electronic Musician October 1999

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Life in the Slow Lane pp. 44-56

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Mastering Continuity pp. 78–88

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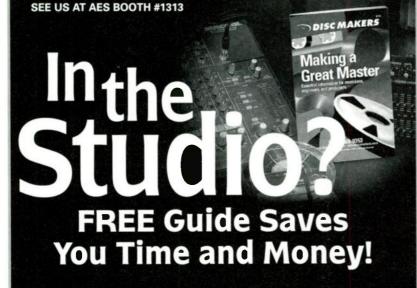
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USB/Ilio Entertainments (distributor) tel. (800) 747-4546 or (818) 707-7222; fax (818) 707-8552; e-mail ilioinfo@ilio.com; Web www.ilio.com though digital, is reminiscent of the spring reverb built into a guitar amp I own. The delay—basically a slapback sounds mediocre at best. The effects will work in a pinch, however, especially for industrial or speaking events. (Even then I would go lightly, because a little goes a long way.) For live music performances, I would rather use a good-quality external effects unit returned through one of the submixes.

The other problem is the fan noise. With almost 400 watts (190 watts per side), the NoteMix MA 400 has nearly the power of a B-29 bomber. Unfortunately, it's almost as loud as one, too. This shouldn't be a problem in clubs, considering all the other noise going on, and the MA 400 certainly has the moxie to power the average nightclub P.A. (A version with 96 watts per side, the MA 200, is available for \$849.)

In its present incarnation, though, the NoteMix is probably not the best choice for a small room or a quiet setting. I used the NoteMix for an acousticmusic show in a coffeehouse and was



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very distracted by its unrelenting fan noise. A quieter fan would definitely be appreciated. (According to Etek's U.S. marketer, Wave Distribution, the most recent version of the NoteMix has a redesigned power-amp stage with a "much quieter" fan.) And how about a thermal switch that would turn the fan on only when things got hot?

Finally, I was struck by the inadequacy of the NoteMix's user manual. Aside from the colorful turns of phrase in broken English—gems such as "the ledbars can't be missing in order to visualize the real output level" and "your system can be abreast with the times and offer your audience new sonorities"—the dual-language manual (in Italian and English) simply doesn't explain much.

FOLD IT UP

The Etek NoteMix MA 400 is a nifty powered mixer that, for the most part, delivers the goods. I love the unit's unique, notebook-type design and the

> ▼ The NoteMix MA 400 has nearly the power of a B-29 bomber.

extreme portability and small footprint it affords. For small to medium-size club gigs that require performers to bring their own P.A., the NoteMix will definitely ease the burdens of setup and teardown.

As noted, the fan noise on the review unit was excessive, and the NoteMix's market appeal would be considerably enhanced by improved digital effects. The few other grievances I have—such as momentary PFL switches and global phantom power that is permanently turned on—are fairly minor.

With a few tweaks, Etek could have a real winner on its hands. I look forward to seeing the next generation of this innovative product.

J.J. Jenkins is an independent producer, engineer, and musician who lives on an island in San Francisco Bay (no, not Alcatraz).

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UNIVERSAL SOUND BANK Raricussions (Mac/Win) By Dan Phillips

Adding some tasty percussion can give an entirely new dimension to a basic drum track. Tried-and-true standbys such as tambourines and shakers are

.

all well and good, but sometimes grooves call for something out of the ordinary. At times like these, Universal Sound Bank's *Raricussions* (\$99) might come in handy.

Raricussions, aptly named, is a collection of unusual percussion samples, including hand drums, rattles, shakers, jangles, bells, and more. My favorite is the tchango tche, which sounds like the happy love child of a particularly successful encounter between a ping-pong game and a baby rattle. Close runners-up are the floor percussion, with its flat metallic rattle, and the fedounoum, a deeply resonant hand drum.

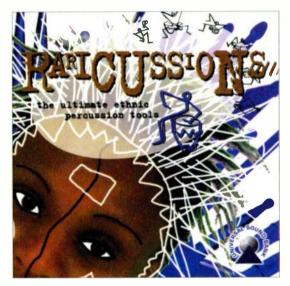
Other standouts include the twangy, percussive berimbau (imagine playing a jaw-harp

by hitting it with a stick); the dry, high slap of the cruche; and the light, almost clicking tone of the cajon. There's also a surprising selection of pitch-bending metallic instruments, including the kutu wappa, water bells, the "box," and strangely sliding glockenspiels. Shakers, rain sticks, a bass kalimba, more bells, talking drums, baja drums, and bowed metal and cymbal effects round out the set.

A two-disc package, *Raricussions* offers about 300 MB of loops, as well as individual hits for almost every instrument—all provided in Akai and audio formats. The CDs also include a set of almost 250 Standard MIDI Files in PC and Macintosh formats, each of which is matched with its own dedicated set of single-hit samples (in Akai format only).

These MIDI files are perhaps the most interesting part of Raricussions. The matched MIDI patterns and samples produce very realistic loops, with the advantage of allowing you to easily perform instant time stretching by changing the tempo in your sequencer. The package has some imperfections, to be sure. For instance, some of the individual samples include multiple hits, limiting the range of effective tempo change; a few of the samples contain some manner of grit in the recording; one set of patterns simply doesn't groove; and a couple of patterns have mild clicks. By and large, however, the MIDI files sound great and work well.

The "old style" sampled loops boast plenty of feel and energy, are skillfully recorded, and offer a wide variety of tempos.



There's a whole world of exotic percussion waiting to be heard, and USB's *Raricussions* provides a good assortment of such sounds in several formats.

The individual hits are plentiful and often include a wide range of performance nuances, but most could be better organized: they sometimes seem to be mapped across the keyboard at random. The documentation is minimal and contains a few small errors; perhaps I've been spoiled by Spectrasonics' outstanding CD booklets. These faults are minor, however.

Raricussions delivers cool, unusual percussive textures and rhythms and gives you several good ways to use them—all at a reasonable price. If you're looking for percussion beyond the pale, this collection is definitely worth checking out.

Overall EM Rating (1 through 5): 4 CIRCLE #446 ON READER SERVICE CARD

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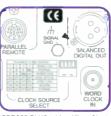
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Although readers may sometimes get the impression that reviewers enjoy criticizing gear, it is in fact far more satisfying to praise than to blame. The problem is that wholly praiseworthy products seem to come along only once in a blue moon especially among those items designed to be "affordable." The PreSonus MP20 (\$649.95) is one of those rare products. Employing Class A discrete input buffers, twin-servo gain stages, and Jensen transformers, it provides everything a 2-channel mic preamp should—and then some.

Beauty and Brains

The MP20 is logically laid out and lovely to behold. Its 1-rackspace front panel, machined from a thick slab of cobalt-blue brushed aluminum, is divided into three sections: one each for the two channels, plus a stereo-bus master-output section. Large, silver-colored aluminum knobs and a big red-backlit power button further contribute to the unit's professional styling.

Controls for each channel include a gain knob (providing 60 dB of gain), as well as green-backlit switches for polarity reverse, 48V phantom power, 20 dB pad, and 80 Hz rumble filter. A red-backlit L/R switch assigns the signal to the master output, with stereo placement controlled by a centerdetented pan knob. Each front-panel channel section also provides a ¼-inch instrument input, a 12-step LED ladder for monitoring the output level, and a unique "IDSS" control that adjusts the drain current on the transformer. This lets you add even harmonics to the input signal to yield a warmer sound.

The MP20's master section makes the unit especially handy for live 2-track recording. It has a level control for the summed stereo signal, a ¼-inch high-gain headphone jack, and a level control for the phones.

On the rear panel, each channel also has an XLR mic input, an XLR output (with gold-plated connectors), and an insert path with ¼-inch send and return jacks. The mix outputs are also on gold-plated XLR jacks.

Sound and the Jury

I used the MP20 for several months in my studio, during which time I recorded loads of different instruments through a dozen different mics. I have three other outboard mic preamps, but the MP20's pristine sound quality and wide array of features quickly made it my favorite.

I also had occasion to use the instrument input for recording bass guitar. The first time, the bass player and I had just spent 20 minutes or so dialing in EQ and compression settings on a full-featured channel-strip-type processor. Although this had given us a sound we liked, on a lark we gave the MP20 a try and both immediately preferred its sound. It was cleaner, fuller, richer—better in every way, and with no EQ or compression added.

Just to confirm my high estimation of the MP20, I compared it critically against two other mic preamps, one less expensive and one more expensive than the MP20. I recorded several sources direct to tape on an ADAT XT20 using identical signal paths (Earthworks QTC1 omnidirectional mics and BLUE Kiwi guad cables). The MP20 smoked the less expensive mic preamp. Compared with the higher-price one (a well-regarded British unit), the MP20 sounded nearly identical, with an ever-so-slightly brighter top end and not guite as much warmth in the low mids. Impressively, the MP20 was also quieter than the two other mic preamps.

A feature worth mentioning is the unit's IDSS control, which is designed to simulate analog tape saturation and tube warmth. That's not exactly how I would describe the effect, though—turning the IDSS knob to the right seems primarily to attenuate high frequencies. I usually opted not to use this feature, as it only took away from the wonderful clarity of the MP20's high end. However, when stereo-miking the top half of a Leslie cabinet with condenser mics, I dialed in a bit of the IDSS circuit to effectively reduce scratchiness in the signals.

A New Standard

The MP20 is the most impressive new mic preamp I've heard in its price range—and it sounds as good as many units costing two or three times as much. Also, whether you're recording live or in the studio, it has



The PreSonus MP20 is a dream come true for the personal-studio owner—an affordable mic preamp that sounds great, looks awesome, and provides all the features you need.

all the features you need. If you require more than two channels, you can get the same sonics from the PreSonus M80 (\$1,999.95), a 2-rackspace unit that provides eight channels of preamplification and all the same features as the MP20 (sans the fancy knobs).

Overall EM Rating (1 through 5): 5 CIRCLE #447 ON READER SERVICE CARD

MICROBOARDS

AudioWrite Pro (Mac/Win) By Carl Weingarten

N owadays, CD recorders are one of the hottest items on the technology market. Manufacturers offer a wide variety of models that boil down to two basic flavors: stand-alone units and computer peripherals. Stand-alone units are self-contained and designed to operate much like any tape deck. They are very handy for recording quick CD demos or cloning discs on the fly. The computer-peripheral types, on the other hand, are controlled by software and connected via SCSI or IDE ports. The key difference between these two kinds of recorders is that stand-alone systems provide instant gratification, whereas computer-based recorders provide detailed editing and mastering features. That difference can make for a tough choice.

Thankfully, MicroBoards Technology has come to the rescue with its new AudioWrite Pro (\$679), one of the first hybrid CD recorders that can operate either with your computer or on its own. It is a compact, well-constructed unit that includes both a SCSI connection and stereo analog I/O on RCA connectors for direct-to-disc recording.

Go Direct

On the analog side, the AudioWrite Pro's interface functions are very basic. No software is required for this type of recording, and familiar Stop, Play, FF, Rew, Record, and Finalize buttons control the process. Tracks can be recorded one at a time, or you can write an entire CD in discat-once mode, each at 1x speed. It has LEDs that indicate track numbers, system status codes, and peak levels.

As a computer peripheral, the Audio-Write Pro is top flight. The input-only SCSI interface makes for rapid data transfer and quick disc creation. My Pentium 200-MMX had no trouble interfacing with the device, and once the connection was established, there were no errors or system conflicts. The AudioWrite Pro burned CDs at 4x speed with ease, and when driven by Adaptec's *Easy CD Creator*, it recorded music and data discs with consistent quality.

MP3 to CD

MicroBoards, with a keen eye on popular technologies, has included an application called *PlayWrite MP3*. You can download MP3 files and use the software to automatically convert them to the Red Book format. The application also sports a WAV file recorder, making it easy to compile songs from vinyl, cassettes, or any other analog source.

Burning Issues

The AudioWrite Pro is a reliable CD recorder, and its hybrid SCSI/analog combo makes it a flexible tool for home and studio recording. However, the analog interface lacks several standard features found on most recorders. First, there are just two peak lights for level indication. Accurate record-level indication is critical for CD

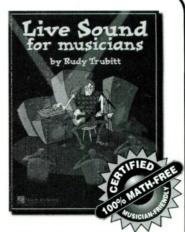


The AudioWrite Pro from MicroBoards Technology includes both a SCSI connection and stereo analog I/O for burning CDs either with your computer or on its own.

recorders since a single digital overload will ruin the entire disc. A three- or foursegment LED would make all the difference here. Also, there is no play-through to the unit's outputs during recording, so you must monitor from the audio source. Granted, these limitations are not a challenge for studio recording, where levels are more easily controlled. But for live, location, or on-the-fly recording, where signal levels are less predictable, direct-to-CD recording can be tricky without comprehensive monitoring abilities.

If you need a SCSI-driven CD recorder, though, and would also like the option of using analog inputs, the AudioWrite Pro is a good choice.

Overall EM Rating (1 through 5): 4 CIRCLE #448 ON READER SERVICE CARD



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BEATBOY

Ramon Yslas Contemporary Percussion and Drums (Mac/Win)

By Jeff Obee

Right on the heels of the Richie Gajate-Garcia disc comes another offering from Beatboy (\$49.95), this time featuring Ramon Yslas, a drummer and percussionist who has performed with Strunz and Farah, Shadowfax, and Jon Anderson of Yes. Whereas the Gajate-Garcia disc focuses on traditional Latin styles, this collection centers around hip-hop, jazz, funk, and house grooves—albeit all played with Latin percussion instruments.

Like the other collections in Beatboy's Artist Signature Series, this one comes as Standard MIDI Files (SMFs) mapped to the GM, XG, and GS formats. Yslas gives us 20 files of his playing, all unquantized for maximum feel and each between 40 and 98

bars long (though most are over 60 bars). Each song file includes various and sundry kit and percussion instruments, each on a separate track, leading in with a two-bar count-off.

Parts is Parts

Each part enters at a different time, according to its function as an intro, a verse, a chorus, a bridge, a solo, a fill, or an outro. The parts are of differing lengths, however, so they sometimes overlap. You can choose to vertically highlight any section of tracks—say, just a bridge or chorus—to create an assortment of variations on the main groove. The Beatboy discs are minimally edited to ensure that you get the entire phrase if you cut a

specified section to use elsewhere.

Tempos fall in a wide range from 66 to 162 bpm—but remember that these are SMFs, not audio files, so no time-stretching is necessary to increase or decrease tempos. Take a groove recorded at 96 bpm, play it at 136 bpm, and see if it is to your liking. Most time signatures are 4/4, although there are two files in 6/8, one in 3/4, and one in 7/4.

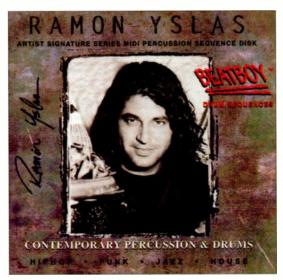
Yslas Styles

Some traditional Latin elements are infused into many of the grooves, but all are wholly original Yslas creations. He explores some cool rhythmic terrain here. One of my favorites is the 7/4 song "Lay Back in 7." It's a very natural-feeling 7, with kick and hi-hat laying an easy-tofollow foundation, crash and cuica accents, and bongo and conga patterns floating over the bar.

"Thang," a 64-bar 4/4 groove at 96 bpm, is a funky hip-hop piece—with emphasis on the funk. The song is set up by a tambourine; triangle, cowbell, and conga enter one by one. A drum kit kicks in at bar 19, and timbales solo over the pattern. The opening song, "And You Are?" is described as a swing feel for club music, but it's more of a snappy hip-hop kind of thing. Nonetheless, it kicks very nicely and is definitely dance material.

Yslas for You?

Some audio sample CDs let you match and sync different loops, but it can be a chore. In the SMF format, however, you can easily take any track, sections of a track, or



Popular dance grooves with a Latin flair are served up on Beatboy's Ramon Yslas Contemporary Percussion and Drums.

> groupings of multiple tracks that you think would work well with those from another piece and sequence them together with no sweat. Or, if the hi-hat part is all you need, copy and paste that into your composition.

> Ramon Yslas offers lots of creative, professionally performed beats right out of the box, and if you have a sampler and a bunch of high-quality GM drum kits, you can interchange the kits (or just load, say, a different snare sample) and tailor it to your needs. I like it. Give this disc a listen. @

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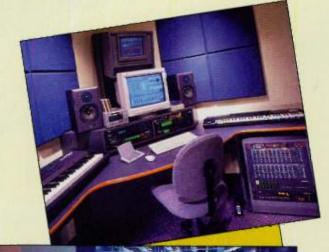
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Scheduled Speakers (subject to change)

- John Perry Barlow (Futurist, Greatful Dead Lyricist)
 Siddiq Bello
- (editor MP3 Impact) **Jack Blades**
- (artist Night Ranger, Damn Yankees) Jeff Brandstetter
- (IP/Music Attorney)
- Nicholas Butterworth (President/CEO, SonicNet/ATN)
- Ted Cohen
- (Producer, Consulting Adults/Webnoize) .
- Kevin Conroy (S.V.P. of WW Mkt. BMG Entertainment) Thomas Dolby Robertson (CEO Beatnik, recording artist)
- Mike Farrace
- (V. P. Worldwide Mkt, Tower Records) **Scott Fedewa**
- (CEO, Musicosm Rex)
- Les Garland
- (President, Sputnik 7; co-founder MTV) Marc Geiger (Principal, Artist Direct/UBL)
- Gary Gersh (Principal, GAS; former President/CEO, Capitol Records)
- Dave Goldberg (President, Launch)
- **Mike Greene**
- (CEO, NARAS) Jim Griffin
- (Founder, OneHouse)
- Bernie Grundman
- (Bernie Grundman Mastering) Thomas Hale
- (Chief Alchemist, Wired Planet) Liz Heller
- (former Exec.V.P., Capitol Records) joe Jennings
- (V.P. of Mkt., InterTrust) **Keith Johnson**
- (founder, HDCD)
- Gerry Kearby (CEO, Liquid Audio)
- Andrew Keen
- (Publisher, AudioCafe.com)
- John Kellogg
- (Gen. Mgr. Multichannel Music, Dolby)
- Jon Kertzer
- (Dir. Multimedia, Experience Music Project) Bob Kohn
- (Chairman, EMusic)
- John Meyer
- (CEO, Meyer Sound) Larry Miller (President, Reciprocal Music)
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Audio-Technica took the heart of their smash hit 4050 large-diaphragm studio condenser mic and put it in a road-worthy, handheld body. The result? The AT4055 gives you stunning clarity and definition for live vocals and extraordinary flexibility for micing instruments on stage. Also available is the 4054 with an 80Hz Bass Roll-off to eliminate unwanted rumble. Still using those old dynamic vocal mics? The AT4055 is perhaps the single most effective upgrade you can make to your sound system.

The Best Value in UHF Wireless? Think A-T!

Meet the new 7000 series UHF wireless from Audio-Technica. This robust, 100 channel, frequency agile system is everything you've wanted for bulletproof performance including 1/2 rack, true-diversity receiver, full metering, balanced output and ground lift. Select from a wide variety of mic elements, instrument cables and accessories. Finally, a touring quality wireless for under a grand. Once again, A-T delivers top quality at an unbeatable price.

Variable Weight Round Base — A Breakthrough in Mic Stands!

Mic stands don't seem to make much news when it comes to new technology. But Quik Lok is making news with their new A-300 Series. It's round base starts off lightweight — just six pounds. Add sand or water to the exact weight you want. The convenient round base takes up less room than tripods but still gives you the option for maximum stability. The pro, flat black finish looks great. Cable clips are included to keep your stage setup tidy. Select standard or short heights with your choice of optional fixed length or telescopic booms. Our Quik Lok Four-Pack nets you tremendous savings on a set of four stands. Call now for yours!

Vintage Tube Sound Live? Must be your ART Channel Strip!

Sure, ART's Pro Channel and Tube Channel rackmount "channel strips" are two of the hottest studio devices. But don't overlook their tremendous advantages for live rigs. You get genuine tube based mic preamplification (and DI), opto-compressor and parametric EQ. Warm up your vocals? Pack some punch into your bass or kick? Tweak the heat on your guitars and keyboards? Make your sax sizzle? There are so many uses for these great tube processors, you'll want a rackfull! And thanks to their remarkably low price, you can have that vintage tone without the vintage price tag!

Six Top-shelf UHF Diversity Wireless Receivers in a Single Rackspace? Only with SONY's Unique MB 806A! Easily Expand from 1-6 Devices.

You'll love the astounding flexibility and convenience. And it's a fraction of the space and weight of yesterday's wireless at a lower price! 282 selectable frequencies across 6 UHF TV channels means no worries about getting shut out by DTV (Digital Television) or other potential interference. It can even assign channels automatically, skipping any that might give you trouble!

Ready for Extraordinary Accuracy? Pick a Pair of Earthworks Mics!

You invested a lot of time and money to get great sounding instruments and amps. So why not capture those great sounds as accurately as possible? The Earthworks SR77 is a positively delicious mic for all manner of instruments and vocals. Can you say flat frequency response? And no response peaks means less feedback as well. The available Matched Pair set of SR77s is your top choice for stereo location recording. If you haven't added a pair of Earthworks mics to your live rig, you just don't know what you're missing! Plus there's Earthworks' M30 measurement mic. Want to tweak your system to perfection? Read on!

Do You SpectraFoo? We Do! Your Complete Real-Time Metering System!

What do tours by the Dave Matthews band, Lenny Kravitz and Beauty & The Beast have in common? Their secret weapon: the award winning SpectraFoo audio metering & analysis software. RTA tools like 2 channel differential FFT help you quickly get the most from any PA. You get level meters, phase scopes, oscilloscopes, spectrum analyzers, a 24 bit signal generator and much, much more! SpectraFoo runs stand-alone on MacOS®, or as a TDM or MAS plug-in. Pop it on a PowerBook®, feed it from a pair of Earthworks M30 mics and you've got more metering power than a dozen traditional devices at a fraction of the investment!

Why not enjoy the extraordinary sound and exceptional convenience of these powerful performance tools at your very next gig? Call us today!

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TASCAM

TASCAM "PERFORMANCE BUNDLE": Another Sweetwater Exclusive! This offer upgrades your TM-D1000 Digital Mixer to deliver 12 Mic Preamps and double the DSP at an amazing Sweetwater "ProNet" discount! Call for details!



CHOOSE THE "XLT POWER TRIO" FOR SUPERB SOUND AND PORTABILITY: Thanks to the XLT41E's compact 12" and the XLT51E's powerful 15" driver, you get an incredibly convenient and easy-to-carry PA that really kicks! The XLT41E even works great as a floor monitor! Call for your special Sweetwater discount on this Power Trio!

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LIVE PERFORMANCE TOOLS

Why Upgrade to SHURE PSM'700 Stereo Wireless In-Ear Monitors?

- You have the best possible protection for your hearing.
- Your monitors sound great every night, regardless of the venue.
- You have tremendous freedom of movement on stage without losing your monitors.
- You save money as multi-user systems are actually more economical than traditional, multi-speaker monitor systems.
- You drastically reduce the weight and size of your monitor system.

Why does Shure Dominate the In-Ear Wireless Monitor Category?

- Sound: Shure's unique Low Mass/High Energy E5 dual-driver earphones deliver stunning audio quality.
- Flexibility: Each transmitter delivers your choice of one stereo mix or two user-selectable mono mixes.

Use any number of receivers with a single transmitter. Everyone on stage can enjoy a clear, safe mix — all for a lot less per band member than most floor monitor rigs! Add up to 16 base transmitters for a total of 16 stereo or 32 mono mixes.

Mark of the Unicorn — the Choice for Powerful Live MIDI & Audio

Live sequencing? It's not just for keyboards and drums anymore! Automate a mix, reset effects and EQs, run your lights, even play complete audio tracks with real-time plug-in DSP effects! **Digital Performer** sequencing software has proven reliability with hundreds of live touring acts and innumerable concert performances. The **MTP AV** patches your live MIDI rig with on-the-fly setup changes —indispensable for keyboards and FOH control of effects processors. The **2408** gives you tremendous audio playback and recording capabilities and the **1224** lets you record your performances in stellar, 24 bit resolution. This combo has quickly become the standard on pro tours, both for audio "sweetening" and live location recording.

Automated Digital Mixing for Live Gigs? The Tascam TM-D1000 Performance Bundle is Here — A Sweetwater Exclusive!

No soundman? No problem! Tascam's amazing TM-D1000 Digital Mixer is perfect for the small ensemble, keyboard player or electronic percussionist that wants great sound and extensive control, without a lot of complicated headaches. Easily create preset mixer "scenes" for each song. Set all mixing functions plus built-in digital effects with a single button push! Or enjoy real-time automation when you control the TM-D1000 from a MIDI sequencer such as Digital Performer.

Sweetwater's Performance Bundle adds Tascam's MA-AD8 8-channel mic preamp/A-to-D converter and FX1000 DSP expander. You get a total of 12 balanced, XLR inputs with 20-Bit D to A conversion, enough for full band. DSP horsepower is dynamically allocatable for up to 8 dynamics processors and 4 channels of digital effects. Save all settings with scenes or automate! Why settle for manual mixing? Call us here at Sweetwater Sound today for our special "ProNet" discount on this great bundle! We'll even **pay you top dollar for your old board when you upgrade** to a Tascam Performance Bundle.

Power and Grace! A Truly Compact PA that Smokes!

What if your club PA had more volume, cleaner sound and less weight? For solo artists and small ensembles, the Community XLT41E two-way cabinet is the perfect choice, balancing top sound quality, pro durability and remarkable portability. Add an XLT51E 15" subwoofer and you've got a full range rig that really kicks, without breaking your back! From the titanium, highdispersion tweeters to the indestructible construction, Community has taken all of their knowledge and experience with arena and stadium systems and packed it into these little giants!

Enhance your live shows with these advanced tools. What's the best approach for your unique needs? Call us now to talk it over! SEE US AT AES BOOTH #1556

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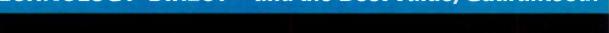


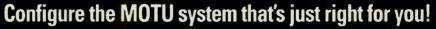
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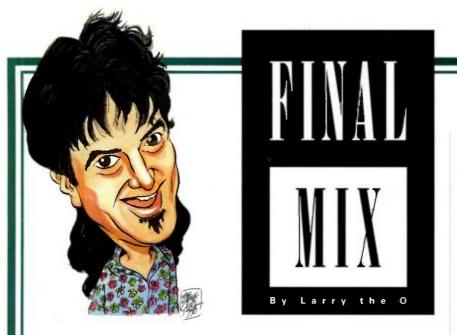
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116 dB dynamic range	With our new 1224 analog interface The 1224 gives you stunning audio specs that rival today's most expensive interfaces.
Balanced I/O	With the 1224's eight analog inputs and ten outputs All of the 1224's analog connectors are balanced +4 TRS or XLR for pro-grade I/O.
Tons of ADAT Optical I/O	24 channels of ADAT optical expandable to 72 The 2408 delivers all the ADAT optical you need for today's digital mixers, FX processors and other gea
Loads of Tascam digital I/O	24 channels of Tascam TDIF expandable to 72 If you're in the Tascam world of digital I/O, no other system even comes close.
S/PDIE and AES/EBU I/O	The new 308 gives you AES/EBU and two flavors of S/PDIF 8 channels each of optical "TOSlink" S/PDIF, RCA "coax" S/PDIF and AES/EBU — all in 24-bit glo
308 digital 10 Expansion	With the flexible PCI-324 card — the core of the system Connect up to three 1224, 2408 and 308 interfaces for as many as 72 inputs/outputs.
Sample-accurate sync	With digital transfers between your Mac and MDM's Say goodbye to worrisome phase issues and other digital audio sync problems.
Broad compatibility	With all major audio software for Mac and Windows Use your favorite audio software with your favorite native plug-ins.
Audio format conversion	Up to 24 channels at a time Own the most flexible format converters out there — without paying extra!
Sample-accurate software	with AudioDesk™, the workstation software for Mac OS Make sample-accurate transfers with ADATS. Edit tracks with sample-accuracy.
Super-easy setup	with our step-by-step Setup Wizard You'll be up and running in no time.
industry buzz	Why is everyone is talking about the 2408? Keyboard Magazine says it best: "Is the 2408 the audio interface system we've all been waiting for?the answer is yes."
Price, price and price	Did we say price? A core 2408 system with 24 channels of input/output is only \$995. Add a 1224 24-bit analog expander for only \$995 — or a 308 for only \$695. Mix and match them any way you like. At these prices, you can own just the right combination.

MOTU 2408/1224/308 hard disk recording

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Pass the Salt, Please

Ithough we in the audio and music industries do our jobs out of our love for music and sound, we still need to earn a living. To do that, we must market the products or services we have to offer. The marketing process, however, can often transform a truthful statement into something more sweeping and less accurate.

In a recent article that appeared in the San Francisco Chronicle about the staying power of vinyl records, writer Joel Selvin stated that "Records sound better than CDs....Engineers use words like warmth or bloom to describe what records have over CDs." Selvin cites "the mastering technology [that is] used to transfer analog recordings to digital" as another weakness of CDs.

I think we can all agree that the sounds of analog and digital media have a qualitative difference between them, but you'll hear more than just warmth on vinyl LPs: let's not forget about ticks and pops, surface noise, wow, flutter, and distortion from record or stylus wear. Aside from that, I've seen many more vinyl records than CDs become unplayable because of warping or wear. In fact, I may be lucky, but the only unplayable CD I ever owned only needed to be washed, and then it played fine.

As for Selvin's comment about mastering, vinyl mastering is so arcane that megaproducer Arif Mardin told me in a recent interview (see "Diva's Choice" in the September 1999 issue of EM) that in the 1960s, problems with mastering vinyl caused Atlantic Records to start mixing kick drum and bass in the center, a convention we now take for granted.

Clearly, Selvin's statements regarding vinyl were sweeping because they were intended to grab the reader and sell newspapers. After all, the title of his article was "The Vinyl Victory." I suppose it is a victory of sorts that vinyl has maintained a niche market rather than disappearing altogether.

Another example of an exaggerated marketing claim can be seen in Digidesign's recent "Is Tape Dead?" promotion. We can agree on the company's basic point: that hard-disk recording is viable and affordable for many more applications than ever before. I've used hard-disk recording literally every day for years (mostly Pro Tools, more recently MOTU 2408), making albums and soundtracks for film, TV, and games.

But the folks at Digidesign are too smart to believe that hard-disk recording is superior to tape in every application. Tapes are still much cheaper than disks, and tape machines crash less frequently (and cause less damage when they do crash) than a computer-based system—not to mention that it's much cheaper and easier to add tracks with modular digital multitrack tape recorders.

If I were to multitrack-record every night of a band's three-month tour, for example, I wouldn't use a hard-disk system unless I had a fat budget. On the road, I'd have an easier time recovering from a broken ADAT than a failed hard-disk system.

But admitting that about tape doesn't sell Pro Tools systems, which is, after all, Digidesign's business.

Even EM is sometimes vulnerable to the need to assert broad "learn everything about this hot topic from our article" statements in order to sell magazines (although we do try to avoid doing so). We know that you can't learn everything about a topic from just one magazine article, but sometimes things get pretty darn competitive on the newsstands, and we succumb.

The point is, the same caveat emptor that you hear in other industries also applies to our own: take all statements with a grain of salt, and closely examine them from all sources. Except me, of course. I'm always right, and I never lie.

Larry the 0 is a sound designer at Lucas-Arts Entertainment. He has been doing "method" sound design for the upcoming anime feature film Vampire Hunter D, which is to say that he's keeping vampire hours. (This was written at 3:30 a.m.)



In The Future, MIDI Setup Will Be Easy...

The millennium always seemed so far away. The future's here. And Opcode's new USB* MIDI interfaces for Mac and PC are part of it. In the past, setting up a computer for MIDI was a struggle. Software drivers in the right place, searching ports, IRQs, DMAs, I/O Addresses. That's history. Run Opcode's intuitive installer, plug in your interface and it's configured—don't even turn off your computer. Isn't that the way stuff is supposed to work in the future?

Go ahead, build the studio you've always dreamed of. Hook up multiple Opcode USB interfaces to integrate your entire MIDI rig. Our interfaces have a unique ID system

...Welcome to The Future

5001

JFI

to assure MIDI devices show up on the right interface every time you boot your computer (not all interfaces do).

So stop by your local dealer sometime this century and pick out an Opcode MIDI interface with USB. After all, you live in the future now—you'll be pleasantly surprised how easy it is to set up your studio.



News

Vision USP

\$59.95**

Software Download

Fermata notation software Revolve MIDI pattern sequencer

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Why use Digital Performer's effects automation?

1. Beat/tempo-based automation.

Automate plug-in effects in perfect time with your music, from filter sweeps that land on downbeats to multitap delays that echo in triplet 8ths. Your beat-based effects always stay in rhythm, even through meter and tempo changes. You'll never waste time wondering things like, "how many milliseconds is a 16th note at 126 bpm, anyway?" Rhythmic effects are now just a few clicks away.

2. Sample-accurate ramp automation.

Digital Performer's plug-in automation isn't a kludge — it calculates true ramps in 32-bit floating point glory. And it's sample-accurate, not quantized to buffer boundaries, so you'll never hear weird artifacts or zipper noise in your audio. Instead, your moves will be as smooth as silk...

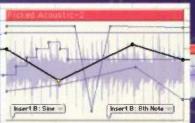
3. Discrete events and stair-step automation.

Some effect changes are discrete events, I ke changing an LFO from a sine wave to a square wave. Others require a stair-step approach. Digital Performer has all three: ramps, events and steps.

4. Graphic editing

View all automation data directly on the audio waveform. Work fast with descriptive icons and convenient control points.

- View all automation data at one time. Clearly view all automation data at one time. Easily control the interaction of multiple FX parameters.
- Units of measurement that actually make sense. Digital Performer's automation data is always displayed in the correct unit (like milliseconds or percent), instead of arbitrary number ranges like other programs. (0-127, yipee!)
- Five advanced automation modes. Tweak your heart out with advanced automation modes like Touch, Latch, Overwrite, Trim Touch and Trim Latch. Want to bypass the effect? You can automate that, too.
- Mackie " HUI" support. Tweak FX parameters in real time with real knobs. Record your moves. Feel the power.



Digital Performer includes more than 50 automatable MIDI and audio plug-ir effacts.

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