RECORDING VOCALS: CHOOSING THE RIGHT GEAR

Electronic Musician®

November 1998

Sequencing Secrets

Record stellar tracks with these insider tips

ART OR THEFT?

When does sampling cross the line?

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Mixing TV music in your personal studio

13 *must-read reviews, including Panasonic DA7 digital mixer, Lexicon Studio DAW, Sonic Foundry Acid*

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The only 24x4 mixer with

Built with advanced technology & premium components,

SR24•4

After you've gone store-to-store, checking out mixer after mixer, they can start to appear pretty much alike.

But if you could "look under the hood," you'd discover that all 4-bus mixers aren't created equal in many critical areas. These important differences can affect sound quality, noise floor, mix head room and durability.

Greg Mackie initially intended the SR24•4 to be a very competitively priced live sound mixer.

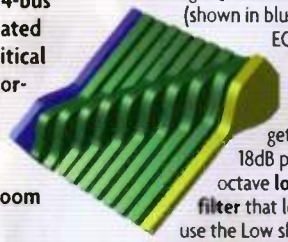
But having been a Musician On A Strict Budget himself, he knew that few bands have the bucks for a separate studio mixer. So instead of cutting corners, he made the SR24•4 a "downsized" Mackie 8•Bus with much of its circuitry and many of the same cool features.

The result is a compact console with premium mic preamps, natural-sounding equalization, ultra-low noise floor and EIGHT tape outputs. Just the thing for recording demos — or whole albums — on a limited budget.

Call toll-free for a comprehensive tabloid brochure or log onto our Web site for the full story of the SR24•4 and its big brother, the SR32•4. They look good outside. But more imporant, they SOUND good inside.

Flexible, creativity-enhancing equalization. Mono mic/line channel's swept midrange has a super-wide 100Hz-8kHz sweep range (and a broad, natural-sounding 1.5-octave wide curve (shown in green below). Low shelving EQ is fixed at 80HZ (shown in blue); high EQ is at 12kHz (yellow). Plus you get a sharp, 18dB per octave **low-cut filter** that lets you use the Low shelving EQ to enhance vocals, floor toms, etc. without boosting unwanted mic thumps and stage rumble.

VERY useful.



Six aux sends per channel. Auxes 1 & 2 are **pre**-fader (for live sound monitors). Auxes 5 & 6 are **post**-fader (for studio effects). Auxes 3 & 4 are switchable to either **pre or post** so you can always have four of the kind you want most.

Trim control has a 10dB "virtual pad" that tames ultra-hot line inputs; 60dB total gain range lets you boost timid vocalists and low level line inputs.

In-place stereo solo on channel strips & sub buses. Master section has solo level control & AFL/PFL global mode switch.

60mm logarithmic-taper faders. Many conventional faders "give up" about ¾ of the way down. Fades sound sorta like this:

LA LA LA LA LA.....?

The log-taper faders on the SR24•4 and SR32•4 have extra screened resistance elements that provide a linear-sounding fade, throughout the full travel of the control. Something like this:

LA LA LA LA LA LA LA LA LA

Your LA LAs may vary.

Special pan controls maintain the same apparent loudness even when you pan a channel hard right or hard left — a must for accurate studio mixes.

Super-twitchy Signal Present LEDs on every channel are so responsive that you can differentiate between vocals, rusty chainsaw samples, percussion, etc. All channels also have an overload LED.

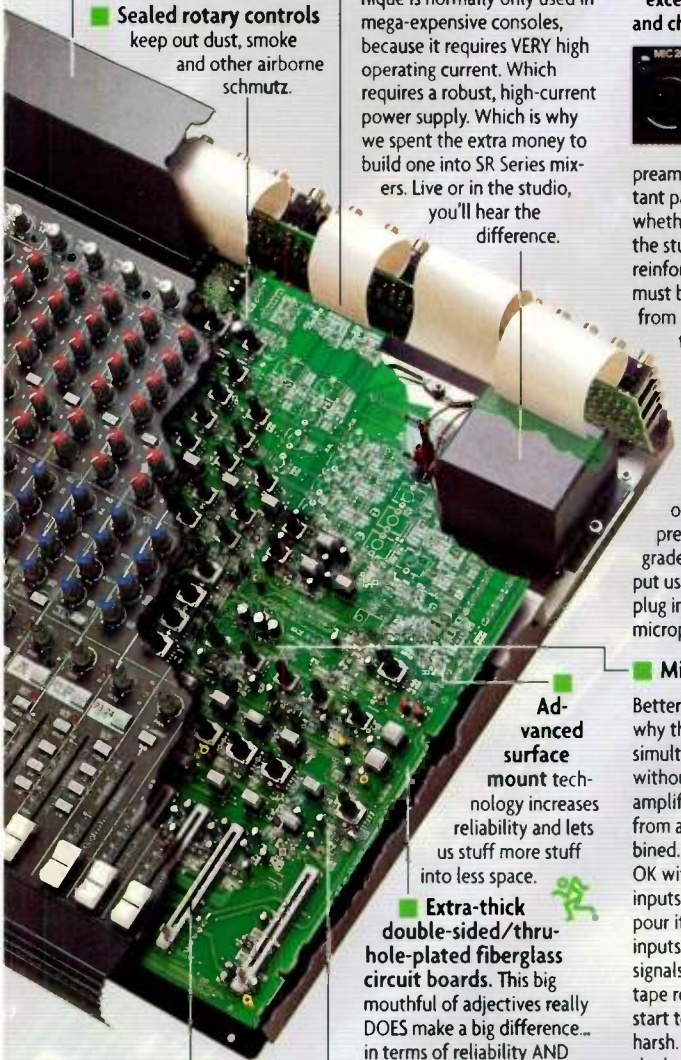
Mute/Solo LED on every channel.

Ultra-high "AIR" EQ on submix buses centered at 16kHz. As one magazine review put it, *"The AIR controls turned out to be effective in adding top end clarity... it's almost an 'exciter' kind of effect, except without the harshness."*

Also available in a family-size 32-channel model!

enough guts to strip in public.

the SR24•4 is equally at home in the recording studio or on the road.



Solid, cold-rolled steel chassis. Not aluminum or plastic. Monocoque design resists flexing and bending.

Gold-plated internal interconnects remain corrosion free for perfect electrical contact, even if used repeatedly in industrial sections of New Jersey.

Sealed rotary controls keep out dust, smoke and other airborne schmutz.

Large, high-current internal power supply lets us use VLZ® (Very Low Impedance) circuitry at critical points in the SR24•4 and SR32•4. **VLZ®** significantly reduces thermal noise and crosstalk by using extremely low resistor values in certain circuits. This innovative technique is normally only used in mega-expensive consoles, because it requires VERY high operating current. Which requires a robust, high-current power supply. Which is why we spent the extra money to build one into SR Series mixers. Live or in the studio, you'll hear the difference.

All inputs and outputs are balanced* to eliminate hum and allow extra-long cable runs (they can also be used with unbalanced connectors). Tight-gripping 1/4" jacks are solid metal; XLR's are genuine Neutrik®s with internal ferrite beads to reduce radio frequency interference. * **except RCA-type tape jacks and channel inserts.**

Low-noise, high-headroom discrete mic preamps. It can be argued that the preamps are the most important part of a mixer — whether you're recording in the studio or running a sound reinforcement system. They must be accurate and free from coloration...yet be able to handle screaming vocalists and close-miked kick drums without overloading. And, they have to be ultra-quiet. Nowadays, we're not the only ones to claim our mic preamps are "studio-grade." So we invite you to put us to the test. In the store, plug in a good, high-output microphone and a pair of headphones and decide for yourself whose preamps have the most headroom, the least noise and the best sound.

Double tape outputs eliminate repatching during tracking. Okay, we'll be the first to admit that eight buses are a nice feature. But if you're on a tight budget, the SR24•4's "double-bussing" feature is a great solution (and besides, how many times do you REALLY track more than four channels to tape at a time?). Each of the SR24•4/SR32•4's four submix buses feed two different outputs. For example, Sub Bus 1 feeds Tracks 1 and 5; Sub Bus 2 feeds Tracks 2 and 6. etc. Instead of repatching, you route the bus' destination by what tape tracks you put into Record.

Dual headphone outputs with enough level to satisfy even most drummers. And a separate input for a talkback mic (so you don't tie up a mixer channel).

Inserts on all mono channels. Plus submix and main stereo mix inserts, separate control room outputs, extra RCA-type tape inputs and outputs, both 1/4" and XLR stereo outputs, and XLR mono output with its own rear panel level control.

The SR Series in a proverbial nutshell.

	24•4	32•4
Total Channels	24	32
Mono Channels	20	28
Stereo Line Inputs	2	2
Mic Preamps	20	28
Submix Buses	4	4
EQ (mono chs.)	12kHz HF	
	80Hz LF	
	100-8kHz Swept Mid	
	18dB/octave low-cut filter	
EQ (stereo chs.)	12kHz HF	
	80Hz LF	
	800Hz Lo Mid	
	3kHz Hi Mid	
Aux Sends/Ch.	6	6
Stereo Aux Returns	4	4
Tape Outputs	8	8
Channel Inserts	20	28
Width (inches)	31.0	39.25

Advanced surface mount technology increases reliability and lets us stuff more stuff into less space.

Extra-thick double-sided/thru-hole-plated fiberglass circuit boards. This big mouthful of adjectives really DOES make a big difference... in terms of reliability AND sound quality. The expensive thru-hole plating process maximizes electrical conductivity and eliminates the possibility of intermittant contact. The SR Series' flexible fiberglass main board soaks up downward impacts that would shatter brittle phenolic circuit boards.

Ultra-wear-resistant fader wiper surface derived from automotive sensor technology won't develop "the scratchies" even after years of use.

100% genuine name brand electronic parts throughout. Nuff said.

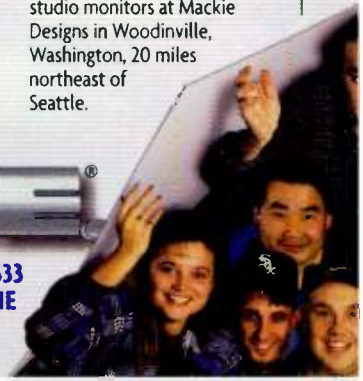
Mix amplifier headroom. The SR24•4's inside story.

Better mix amplifier design is why the SR24•4 can handle 24 simultaneous HOT inputs without distorting. The mix amplifier is where signals from all channels are combined. Some mixers sound OK with just a couple of inputs...but when you pour it on with lots of inputs — particularly signals from digital tape recorders, things start to sound pretty harsh. Backing off on the bus or main faders doesn't help, since the mix amp comes before these gain controls.

The mix amp funnels multiple channel inputs into a single bus.

The SR24•4 and SR32•4 use Mackie's innovative **negative gain mix amplifier architecture.** Instead of mixing at unity gain where headroom is quickly used up, our mix amps operate at -6dB. At this negative gain level, SR Series mixers are capable of summing **FOUR TIMES** the number of channels before clipping. That nets out at **DOUBLE** the amount of mix amplifier headroom compared to any competitive mixer. It's a critical difference that you can plainly hear.

Below: A few of the 500+ folks who build the SR Series, our other mixers, amps and studio monitors at Mackie Designs in Woodinville, Washington, 20 miles northeast of Seattle.



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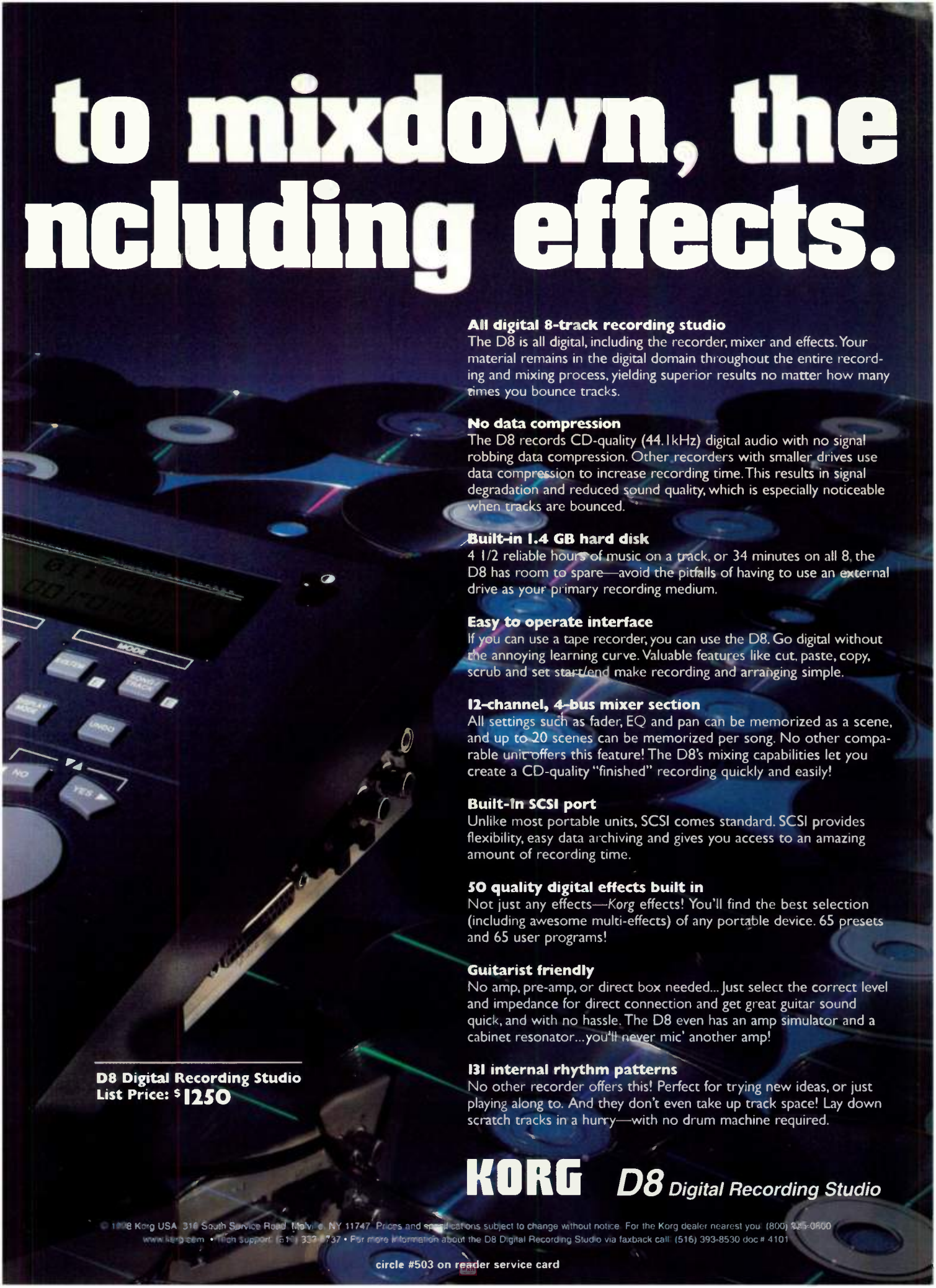
From recording
D8 does it all, i

to mixdown, the ncluding effects.



All digital 8-track recording studio
The D8 is all digital, including the recorder, mixer and effects. Your material remains in the digital domain throughout the entire recording and mixing process, yielding superior results no matter how many times you bounce tracks.

No data compression
The D8 records CD-quality (44.1kHz) digital audio with no signal robbing data compression. Other recorders with smaller drives use data compression to increase recording time. This results in signal degradation and reduced sound quality, which is especially noticeable when tracks are bounced.

Built-in 1.4 GB hard disk
4 1/2 reliable hours of music on a track, or 34 minutes on all 8, the D8 has room to spare—avoid the pitfalls of having to use an external drive as your primary recording medium.

Easy to operate interface
If you can use a tape recorder, you can use the D8. Go digital without the annoying learning curve. Valuable features like cut, paste, copy, scrub and set start/end make recording and arranging simple.

12-channel, 4-bus mixer section
All settings such as fader, EQ and pan can be memorized as a scene, and up to 20 scenes can be memorized per song. No other comparable unit offers this feature! The D8's mixing capabilities let you create a CD-quality "finished" recording quickly and easily!

Built-in SCSI port
Unlike most portable units, SCSI comes standard. SCSI provides flexibility, easy data archiving and gives you access to an amazing amount of recording time.

50 quality digital effects built in
Not just any effects—*Korg* effects! You'll find the best selection (including awesome multi-effects) of any portable device. 65 presets and 65 user programs!

Guitarist friendly
No amp, pre-amp, or direct box needed... Just select the correct level and impedance for direct connection and get great guitar sound quick, and with no hassle. The D8 even has an amp simulator and a cabinet resonator...you'll never mic' another amp!

131 internal rhythm patterns
No other recorder offers this! Perfect for trying new ideas, or just playing along to. And they don't even take up track space! Lay down scratch tracks in a hurry—with no drum machine required.

D8 Digital Recording Studio
List Price: $1250

KORG *D8 Digital Recording Studio*

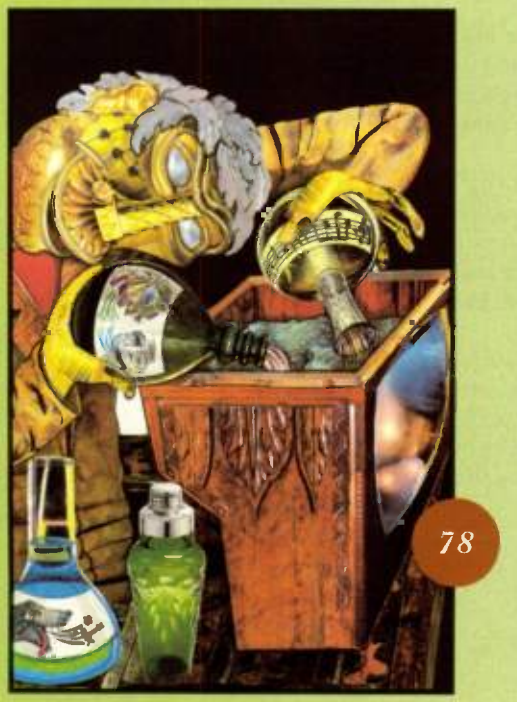
I N S I

FEATURES

DEPARTMENTS

Electronic Musician®
NOVEMBER 1998 VOL. 14, NO. 11

COLUMNS

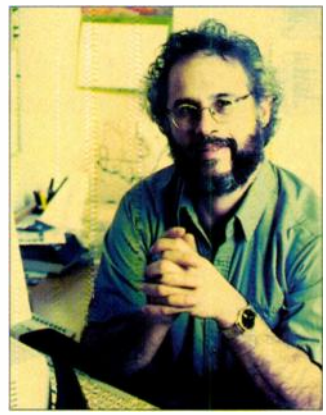


REVIEWS

Cover: Illustration by Michael Morrow

Baked, not Fried

Oh boy, an open-faced crow sandwich!



ANTHONY PIDGEON

In the September 1998 "Letters" column, reader Geoff Hardy sang the praises of Linux, a Unix-based, computer operating system that is free, has an open source code that users can modify, and can run on many different hardware platforms. When Hardy challenged us to get with the program (so to speak), I responded that Linux was interesting but was likely to remain a fringe OS and wasn't something most musicians were likely to use.

It didn't take long to find out I was wrong; several knowledgeable friends informed me that Linux was happening stuff for musicians. I put Contributing Editor Scott Wilkinson on the case, and you'll find his coverage of Linux in this month's "Tech Page" (on p. 36). From here on, we'll keep a close eye out for music-related products for Linux; if you have relevant Linux news of interest to **EM** readers, please e-mail me (steve_oppenheimer@intertec.com).

Meanwhile, I'll have that crow baked, please—not fried. And because Linux is an open OS, make that an open-faced sandwich.

While I'm discussing computer products, we have a special treat for you this month. Digital audio sequencers are the core software for the desktop musician, combining MIDI and digital audio recording and editing. We've put together "Sequencing Secrets" (see p. 54), a master class that delivers both general techniques and product-specific tips for programs from Steinberg, Emagic, Opcode, MOTU, and Cakewalk. Admittedly, we weren't able to include all the top sequencers, but we included most of the big ones.

This project brought out my Perry White side: I abused authors, editors, and manufacturers mercilessly. Fortunately, all responded with excellent work. (Did you know that Superman's real secret identity is Dennis Miller?) "Sequencing Secrets" is an article you'll come back to again and again. My thanks to all; bottles of salve for your bruises are in the mail.

Finally, we recently lost Assistant Editor Joe Humphreys to the freelance life, and that's a big loss. Joe was an integral part of our editorial team. As I noted in the August "Front Page," Diane Lowery also departed not long ago. The good news is that we have now brought our staff back to full strength with the addition of two very promising new editors.

Editorial Assistant Matt Gallagher is an experienced proofreader and copy editor who comes to us from *Music & Computers* magazine, which passed into history recently. Matt is a drummer, which means we have four drummers on our editorial staff. (It scares me to think about that!) In addition to proofreading and copyediting, he is handling a wide variety of essential tasks, and already he has made a difference.

Copy Chief Patty Hammond, whom we shanghaied from *Red Herring* magazine, is going to make a huge difference for us, too. Patty is a first-rate language wizard who will ensure that our editorial style is consistent and our writing is clean, properly punctuated, and grammatically impeccable. She is also a bass player, which not only gives her a personal interest in our subject matter but also helps balance the staff musically. I couldn't have survived with five drummers.

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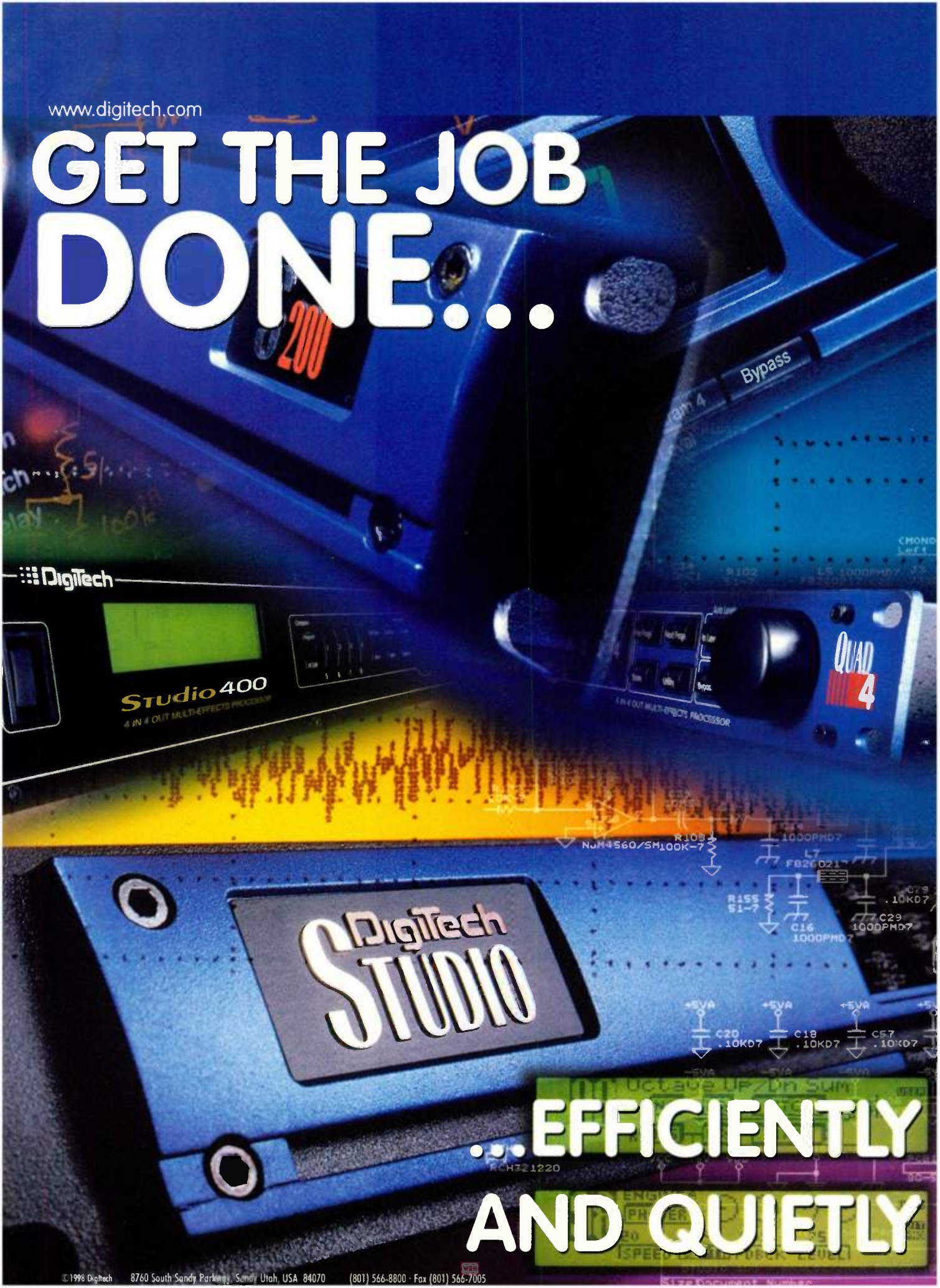

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THINK BEFORE YOU LINK

In your August 1998 review of the PreSonus ACP-22 compressor, Myles Boisen states that he prefers to run stereo program material through a compressor without using the Link mode. Would he mind explaining the reason for this?

Steve Vincent
Portent Music
vincents@harbornet.com

Author Myles Boisen responds: *To be completely accurate, I should say that my first choice is usually to compress a stereo program without using the Link mode, while setting all parameters identically on both channels. Many compressors, when stereo linked, are controlled by the input to one channel only (typically the left). In this case, a transient that goes above threshold on the left—for example a floor-tom hit—may cause a momentary pumping of the entire mix. The same floor-tom hit, if panned to the right channel, might not be compressed at all. Dialing in both channels with the same parameters eliminates some of these inconsistencies but may also cause a noticeable stereo panning shift if one side suddenly dips down in response to an asymmetrical transient. For multitrack mixing, this usually seems like the lesser of two evils to me, and it's almost always my choice in the unpredictable realm of live-to-DAT recording. Let your ears be your guide!*

TEMPEST IN A TEAPOT

My impression has been that the majority of the **EM** staff labor under the sadly mistaken assumption that the Mac is the one truly professional computer. This feeling is evidenced not only in your articles, but in the whole vibe of the magazine. Although you have some folks working there who don't subscribe to this folly, others seem to be stuck in 1993. Wake up, y'all! The new millennium is almost upon us!

Like it or not, the majority of computer users now are running PCs with Windows operating systems. Alas, for the time being, most pro studios still have Pro Tools and a Mac. But I believe this will change drastically over the next year and exists today not as a testament to the "mighty Macintosh" or any belief in the future of Apple itself, but because most musicians don't want to invest their precious time in learning another platform's operating system.

A musician thinking about putting together a home studio shouldn't have to read an article ("Build a Personal Studio on any Budget," July 1998) that proudly proclaims that the "$32,000 studio with computer" consists of a machine that, although very fast (the G3), only has three available slots for expansion. For way less money, that person could have a "faster than lightning" Pentium II 400 MMX with a dual processor ready for the future and six expansion slots. Your magazine has a huge influence on its readers, more so than I think you realize. The young readers, who are very impressionable, pick up on any editorial slant you consistently espouse. Most software companies that wish to stay in business today have become cross-platform; your magazine, the loudest voice out there for our community of electronic musicians, needs to also become more cross-platform in all areas, including its editors.

Ted Perlman
President, "Buffalo Sound" Studios
tedperl@pacbell.net

Ted—First of all, I think we cover Windows computing issues at least as much as, and probably more than, we do Macintosh issues. In the music market, the Mac has a much higher market share than it does in the general computing market, especially in the higher-end studios, but Windows is the most popular platform among **EM** *readers, and we cover it that way. Some of our editors use Windows computers to make music, some use Mac, and a few use both.*

When we designed "Build a Personal Studio on Any Budget," we decided to design half of our studios with computers and half without, and we decided to design half of our computer-based studios around a Mac and half around a Windows PC. In short, we made a conscious effort to give a balanced picture.

As the article clearly states, each of our eight studios reflects the personal preferences and hands-on experience of its designer, and each has a distinctive and overt purpose (e.g., multimedia production, CD production, demos, and so on). We told you what we would choose, given the budgets—neither more nor less. We gave the two $32,000 studios (with and without computer) to Jeff Casey because he has engineered CD, film, and broadcast projects at numerous major studios in New York City, so he is well qualified to discuss the higher-ticket items. He has used Pro Tools quite a bit, so that's what he chose for his computer-based studio.

We have received a number of letters from readers who perceive a platform bias in

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EM—but they're not all on the PC side of the fence, as you will see in the next letter. In fact, they are split about 50/50 between Mac and PC proponents. That indicates we are probably balancing our coverage nicely.—Steve O.

IN THE OTHER CORNER

I agree with all of Larry the O's comments about Apple and the Macintosh in "Final Mix: Whither Ghost Apple" in the July 1998 **EM.** I have agreed with *all* the articles I have read from various sources about the same topics. But none of your statements was new. In fact, they were very old.

I, too, own a G3 desktop. Yes, the slots are too few; yes, I would like more. But I don't blast these comments into the media. We, as Mac supporters, should stop the public complaints and boost the great things about these machines. We must stop being negative and quit telling the PC world that Apple stinks. Apple does not stink. The company is dealing with Microsoft, as is the rest of the world, and we all know that story.

When it came time to renew my subscription to **EM,** I almost didn't. The primary factor in my decision would have been a continuing bolstering of the horrible PC platform and the degrading of support of the Mac from people like you.

Please stop complaining, Larry, and use your influence to remind readers of the fantastic capability of the Macintosh and the tremendous amount of fun the machine is over the Windows interface. Unfortunately, if Windows users hated Windows, too, they would be afraid to change to Mac because of articles like yours.

Jeff Newman
totalsound@earthlink.net

Jeff—I agree with you that it is even more important to laud positive developments than to decry negative ones, and I fully intend to do so as I become aware of them. I consider my purchase of a G3 (and my public declaration of love for it) to be such a statement.

Indeed, none of the statements in my column were new. My feeling is that Apple, over the years, has repeatedly mismanaged opportunities to maintain their dominance in our market, and the only difference now is that the situation has reached a critical juncture. Our market is quickly and strongly moving to Windows, a fact that is easily documented.

I am fortunate enough to have a "bully pulpit" from which to express opinions, and, given that, feel a duty to speak out on the needs of audio and music professionals. I am sorry if you feel that my column, as well as other press, is so negative, but if you agree as fully with my statements as you indicate, I encourage you to send your comments to Apple.—Larry the O

PHASED-OUT MIDI

I am vexed by a problem with my digital audio recording system. I am using Macromedia's *Deck II* as a multitrack digital audio workstation. My recordings typically consist of a MIDI sequence made up of drum and keyboard tracks, a bass guitar track, an acoustic or electric guitar track, several vocal tracks, and possibly a few other acoustic instrument tracks. I have a Mackie CR-1604 analog mixer for mixing the non-MIDI tracks into the DAW. Most of the non-MIDI instruments are recorded with microphones.

Usually, I record the MIDI sequence to digital audio first, since it provides a nice foundation for recording all of the other tracks. However, I have noticed that, as I add additional tracks to the recording, I get a strange phase-shifted effect on the earlier tracks. The earlier tracks seem to be bleeding into the tracks that are recorded later on, and I have tested this by muting the later tracks and noticing how this cleans up sound of the early tracks.

I've tried reducing the volume of the monitor mix in the headphones and increasing the distance between the headphones and the microphone, but this hasn't eliminated the phase-shift effect. What can I do to solve this problem? Am I missing something obvious?

Doug Faber
Naperville, IL
dfaber@xnet.com

Doug—It's tough to troubleshoot this based on your letter, so I'm going to have to do some guesswork. It seems likely that somehow you are recording, or at least hearing, multiple versions of the same source. That can happen in several ways, but we only have space to discuss two of them.

One possibility is that you are playing back the MIDI sequence while monitoring the audio recording of the same sequence. That wouldn't cause the old tracks to literally "bleed" into the new ones, but I wonder if that is really happening. Channel crosstalk isn't possible in the digital domain within Deck, *and you apparently have already eliminated the possibility of acoustic bleed-through into the mics. At any rate, the audio versions of your MIDI tracks would be slightly delayed relative to the original MIDI tracks because the recording process takes a few milliseconds, so you probably would hear phase cancellation and reinforcement when playing both together. To solve this, just mute the MIDI tracks in your sequencer.*

Another possibility is that you are recording the same tracks multiple times. When you bounce ("ping-pong") tracks in Deck, *all active (unmuted) tracks are mixed. So let's say you record MIDI tracks to audio, then record some acoustic tracks, and after that decide to bounce the tracks down. If you don't mute the set of MIDI-cum-audio tracks, they will be included in the bounce. When you play back your tracks, you will hear the MIDI-cum-audio tracks twice: the originals and the bounced version. Mixing two copies of the same tracks would cause the symptoms you describe.—Steve O.*

HARD DRIVE SIZE?

While discussing Cakewalk *Pro Audio* with a sales rep from a mail-order catalog company, I was told that, when recording digital audio to a computer, it is better to use a smaller hard drive to reduce possible latency problems. Obviously, a SCSI bus is a given, but I'm wondering whether I'm going to notice any synching problems when running digital audio with MIDI. The computer setup I have has an Ultra-2 SCSI interface (80 MB/s), and I was planning on getting a Seagate Barracuda or Quantum Atlas 9 GB hard drive for recording (with a standard ultra-DMA drive for software, etc.). Should I get a smaller drive?

Matt Greer
gree@computron.net

Matt—Actually, the opposite is usually the case. Provided your drive meets your software's required data-throughput rate, your system performance should not be affected by the size of the hard disk. What can slow playback performance is a fragmented or full hard disk.

Many digital audio recording programs are nondestructive, so for every take you record, a new file is created, regardless of whether that file ends up in the final mix. This can quickly eat up available disk space, slowing down playback performance and creating a potentially unstable synchronization environment.

In addition, you are constantly writing and erasing files from the target disk,

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resulting in fragmentation, so you should check and optimize the disk regularly. For this reason, you are better off not recording audio files to the disk on which your operating system and applications reside. If possible, use a dedicated hard disk for audio recording.

Obviously, the smaller the hard disk, the more quickly it fills up and fragments, which is why most professional recording studios and editing suites employ larger hard disks for day-to-day production (often at least 9 GB) and rely on removable media for backup and long-term data storage.—Jeff C.

EAR-OPENER

I have a CD of music that has piano, violin, flute, and a little guitar. Is there any software out there that can transcribe this? I figure that it's probably out of the realm of possibility to get the different instruments separated, but I wonder if I can at least get the whole piece of music onto paper, where I could separate the parts myself on the master score. Is this possible with the CD-ROM drive on my computer?

Dan Varwig
djvarwig@mindspring.com

Dan—You can't get there from here with current technology. In general, to notate your CD mix, the audio has to first be converted to MIDI data, which allows a MIDI notation engine to create the score. Several pitch-to-MIDI algorithms are available, but they're all monophonic, so not only can't they convert an audio mix to MIDI, they can't even deal with one polyphonic instrument (e.g., guitar or piano).

So you first would have to separate out the instruments in the stereo mix, saving each on its own track, then translate each track to MIDI, and display the MIDI data as notation. The only software I know of that is said to be able to separate the instruments in a composite stereo mix is Prosoniq's Pandora *music-decomposition software (www.sonicworx.com), which is extremely expensive, and I honestly don't know how well it works. But even assuming it works, you couldn't convert the piano and guitar tracks to MIDI and notatate them because they are polyphonic.*

I suggest doing the job the old fashioned way: transcribing the music by ear. Sure, it takes some work, but all it costs you is time (or maybe a few bucks for a course in ear training and a pad of music paper), and it's a great way to improve your musicianship.—Steve O.

PRO TOOLS PERPLEXITY

The feature "Build a Personal Studio on Any Budget" (July 1998) was far and away the best article to appear in **EM**. I think it would be worthwhile to make it an annual feature.

One question: the "$32,000 Studio with Computer" section (which described a system built around Digidesign's Pro Tools 24) discussed the pros and cons of using Digidesign's *Pro Tools* 4.1 software versus Opcode's *Studio Vision Pro* 4.0 software. I'm a little confused by this, because I thought that *Pro Tools* 4.1 was an integral part of the Pro Tools 24 system. What exactly is the role of *Pro Tools* 4.1 in the Pro Tools 24 system? Is *Studio Vision Pro* 4.0 a completely compatible substitute?

David Eide
Lockport, Illinois
deide@ngelaw.com

David—The Pro Tools 24 system includes Pro Tools *4.1, a professional, multitrack, audio editor.* Pro Tools *4.1 does not have MIDI sequencing features, though it can import and play back MIDI files. So if your project does not involve MIDI, it might be all the software you need.*

However, if your projects involve both audio and MIDI tracks, you might find it best to edit, mix, and process everything in a digital audio sequencer, which integrates MIDI and audio editing. All top-of-the-line digital audio sequencers for the Mac (including Studio Vision Pro*) are completely compatible with Digidesign's Pro Tools 24 hardware.*

Even if you have a digital audio sequencer, you might prefer to edit audio tracks in Pro Tools *4.1 because it is a slick audio recording, editing, and mixing environment. For instance, it has features that help you manage large numbers of audio files, which is one reason it's a popular choice for editing audio to picture. In short, it makes sense to work in the environment you prefer and to use each program for the things you feel it does best.—Steve O.*

SUPPORT FOR STREAMING

I am responding to Geoffrey Newcomb's letter, "Stop in MIDI Stream" (September 1998). I am also a musician and computer aficionado who loves MIDI. However, I must contend some of his statements.

First of all, "no one wants to land on a site that plays background music": what an outrageous statement! How can he speak for the other two billion people who surf the Web? If you don't want to listen to other people's music that's embedded in their pages, you can turn off that option in your browser.

Second of all, not everyone has a high-speed modem or ISDN connections that allow MIDI files to be downloaded "faster than you can blink your eyes." Streaming is a viable, new technology that is quickly coming into its own—not only for MIDI but for other audio and video formats, as well.

As for his remark that one cannot make real music with General MIDI alone, the quality of playback is wholly dependent upon the quality of the playback device! And yes, you can change the palette of available sounds to any range you desire with higher-quality GM devices, such as a Edirol's Sound Canvas. The point of GM files is to allow the listener the opportunity to hear a fairly representative palette of voices.

I feel that Mr. Newcomb has a right to his opinion—just don't try to dictate how the rest of us are supposed to listen to and create music or surf the Web.

John Purkey
Owner, the MidiFactory
midifactory@usa.net
www.midifactory.net

ERROR LOG

1999 Personal Studio Buyer's Guide, *Dynamics Processors table, p. 70: The CLX 51 and CLX 52, listed under Kensington, are actually made by Ashly Audio.*

1999 Personal Studio Buyer's Guide, *Keyboard Workstations, p. 84: The Kawai 5000W was incorrectly listed as "K500W."*

1999 Personal Studio Buyer's Guide, *Power Amps table, p. 126: The seven power amps listed under Kensington—FTX-1001, FTX-1501, FTX-2001, CFT-1800, MFA-6000, MFA-8000, and SRA-120—are all products of Ashly Audio. Also, the CFT-1800 was incorrectly listed as "GFT-1800."*

WE WELCOME YOUR FEEDBACK.

Address correspondence and e-mail to **"Letters,"** Electronic Musician, *6400 Hollis Street, Suite 12, Emeryville, CA 94608 or emeditorial@intertec.com. Published letters may be edited for space and clarity.*

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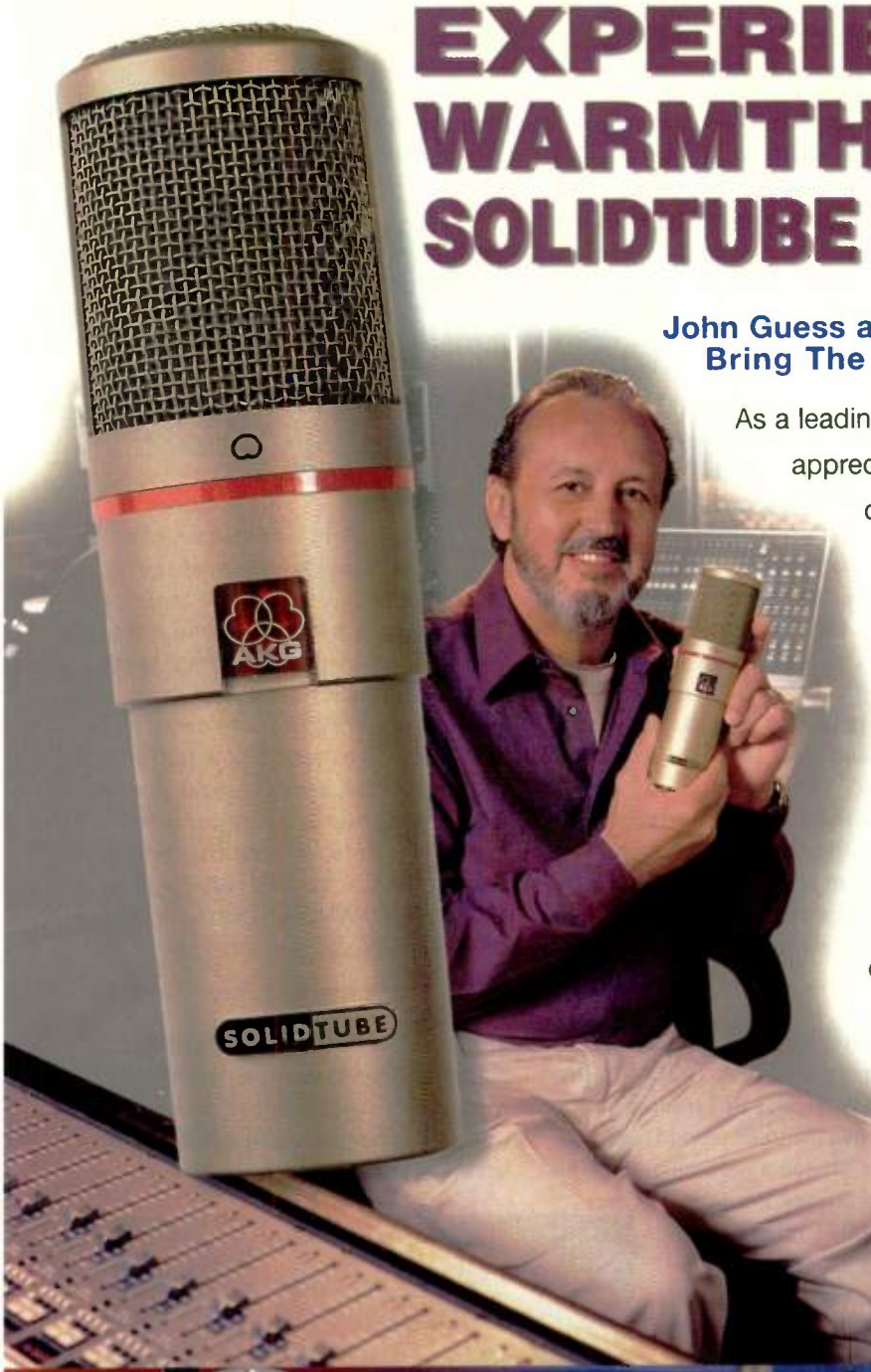

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WHAT'S NEW

By the EM Staff

AUDIX TR40

Audix has released the TR40 ($249; $559 for a matched pair), a small-diameter, omnidirectional, electret-condenser mic that is the first in the company's new series of test and recording mics. Featuring a ¼-inch capsule, the TR40 is designed to accept high sound-pressure levels and deliver a phase-coherent signal. The mic is designed primarily for testing and measurement purposes. Due to the microphone's omnidirectional polar pattern, the proximity effect is lowered, which allows you to position the mic close to an instrument without the boominess inherent in close-miking with cardioid microphones.

The TR40 is machined brass, features a gold-plated XLR connector, and comes with a carrying case and clip. Audix rates the TR40's frequency response at 20 Hz to 19 kHz (±1 dB), its maximum SPL at 140 dB, its signal-to-noise ratio at 77 dB (A weighted), and its self-noise at 17 dB (A weighted). Impedance is 200Ω. Audix; tel. (503) 682-6933; fax (503) 682-7114; e-mail info@audixusa.com; Web www.audixusa.com.

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ROLAND JP-8080

Last year, Roland brought out a new digital synthesizer, the JP-8000, which featured the company's Analog Modeling technology. Roland's new JP-8080 Analog Modeling synth module ($1,595) provides all of the "phat" sound-building features of the JP-8000, and more.

Like its keyboard synth partner, the new module has two oscillators, two LFOs, seven waveforms, and filter and amplifier sections, as well as an onboard arpeggiator. The unit also features Motion Control, allowing for automation of slider and knob movements. A selection of 384 preset Patches and 192 preset Performances, in addition to 128 user Patches and 64 user Performance locations, are part of the package.

The 10-voice polyphonic JP-8080 not only provides the tools for sound creation; it also allows real-time processing of audio, instrument, and mic signals. External audio can be processed as Oscillator 2 waveforms or passed through the new Voice Modulator. The latter offers 26 editable parameters and three modes: the Formant Filter, which applies the frequency of vocal sounds to another audio signal; the Filter Bank 12-band stereo filter; and the Vocal Morph Control for modifying synth sounds with mouth movement.

Onboard effects include delay and tone control, as well as multieffects (chorus, flanger, and distortion). A new Unison mode allows you to combine all ten voices into one Patch. Roland Corporation U.S.; tel. (323) 685-5141; fax (323) 722-0911; Web www.rolandus.com.

Circle #402 on Reader Service Card

TASCAM DA-45HR

With the release of TASCAM's DA-45HR ($1,999), DAT machines have joined the trend toward high-definition, 24-bit audio recording. In 24-bit mode, the 3U rack-mount professional DAT recorder runs at double speed and can record 60 minutes of 24-bit audio per 120-minute tape. The unit is backward compatible with 16-bit files; in 16-bit mode, it can record and play back 120 minutes of audio per tape. It supports sampling frequencies of 44.1 and 48 kHz.

The analog-to-digital converters are 24-bit, and analog input is provided on both balanced XLR and unbalanced RCA jacks. The DACs are 20-bit, and analog output is also on XLR and RCAs. The DA-45HR supports 24-bit AES/EBU and S/PDIF digital I/O.

The DA-45HR has word sync I/O, and a 15-pin parallel interface adds control via fader starts of General Purpose Interface (GPI). It has a menu-driven operating system that includes an LCD. In addition, the front panel features a data/shuttle wheel, L/R input-level knobs, phone jack with level control, and a 2-point Memory/Locate function.

Other features include Auto ID (which can be set at -48, -54, -60, or -66 dB), Copy ID Select copy protection, Reference Level (-20, -18, or -16 dB), and Record Mute (1 to 8 seconds in 0.5 sec increments). A Repeat function lets you repeat play from two to fifteen times, or choose free or unlimited repeat. Single, Skip, and Program Play modes are also implemented. TASCAM; tel. (323) 726-0303; fax (323) 727-7635; faxback (800) 827-2268; Web www.tascam.com.

Circle #403 on Reader Service Card

▶ LINE 6 POD

The modeling technology that is at the heart of Line 6's *Amp Farm* plug-in for Pro Tools is now available in POD ($399), a desktop, direct-recording unit. POD provides stereo simulations of miked guitar-amp tones and built-in digital effects. As an independent DSP unit, POD can process a signal directly from your guitar *or* be used as an outboard effect during mixdown.

The top of POD has eight knobs for output, drive, volume, effects levels, and EQ, as well as selector knobs for amp style and effects. Sixteen digitally modeled amp styles, such as "Brit Blues," "Black Panel," and "Jazz Clean," can be selected from the Amp Models knob; ten other amp models are available through POD's editing software, which ships in both Mac- and PC-compatible versions.

Along with a dedicated knob for reverb, the Effects knob offers six different effects, with or without delay. There is a single-parameter Effects Tweak knob on the unit, and additional parameters can be controlled via MIDI; effect edits can be saved within the unit. POD also provides a guitar tuner.

Around the perimeter of POD are a ¼-inch headphone out jack; balanced, ¼-inch TRS left and right outputs; MIDI In and Out jacks; and a single, ¼-inch instrument input. A pedal output is also provided for use with Line 6's pedal boards. Line 6; tel. (310) 390-5956; fax (310) 390-1713; e-mail sales@line6.com; Web www.line6.com.

Circle #404 on Reader Service Card

▲ ALESIS DM PRO

Designed for drum sequencing, live performance, and drum overdubs, Alesis's 1U rack-mount DM Pro drum module ($899) features 64-voice polyphony and 20-bit DACs. Loaded with 16 MB of sounds, the DM Pro is expandable via a slot for an 8 MB PCMCIA card.

Alesis boasts lightning-fast trigger response for immediate output from the DM Pro's sixteen ¼-inch inputs, which can be triggered by MIDI drum pads or acoustic drum triggers. Gain, crosstalk, noise, decay, and Velocity curve can be controlled independently for each input. In addition, pitch, volume, panning, layering, and MIDI Note Assignment are programmable for all the patches.

The DM Pro is loaded with 1,536 presets recorded at 48 kHz, with another 128 user-definable voices. ROM storage is provided for up to 64 programmable drum kits, and each kit can contain up to 64 sounds.

In addition to the sixteen trigger inputs, the back of the DM Pro packs in six ¼-inch, -10 dBV audio inputs; two RCA, -10 dBV audio outputs; and MIDI In, Out, and Thru. Alesis Corporation; tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com.

Circle #405 on Reader Service Card

▼ CREAMWARE PULSAR

Pentium PC–based desktop musicians who are on a quest for an all-in-one recording solution may want to take a look at Creamware's new Pulsar ($1,298), a PCI board that ships with DSP effects, software synthesizers, a MIDI interface, and twenty channels of I/O.

Input and output includes two ADAT Optical interfaces located on the card, in addition to breakout cables with S/PDIF on RCA connectors; two channels of analog I/O on unbalanced, ¼-inch connectors; and MIDI In and Out connectors. A/D and D/A conversion is on 24-bit converters, and a 32-bit internal bus resolution is used. Four Analog Devices SHARC DSP chips can process audio with a 96 kHz sample rate.

Using the Pulsar, your digital audio sequencer can handle up to 32 tracks of material. Every channel has one fully parametric digital equalizer, and up to four more effects can be added. Automation and synchronization are accomplished via MIDI Time Code. The software includes compressors, limiters, and effects such as delays, flangers, chorus, a vocoder, and more.

In addition, an FM synth is included, along with three analog-style synths (including a TB-303-style synth and a virtual modular synth). An Akai-compatible sample player ships with the Pulsar, as well. The card features MME and Steinberg ASIO drivers. Creamware is also working with several third-party developers on creating Pulsar-compatible plug-ins.

The card has an S/TDM Bus that allows multiple Pulsars to be linked. It will also provide compatibility for the company's forthcoming Scope sound-design workstation (see the July 1998 "What's New"). A Plus version is in the works that will have two balanced, XLR connectors and an AES/EBU interface on an XLR connector, in addition to ADAT I/O. Creamware; tel. (800) 899-1939; fax (604) 435-9937; e-mail steve@creamware.com; Web www.creamware.com.

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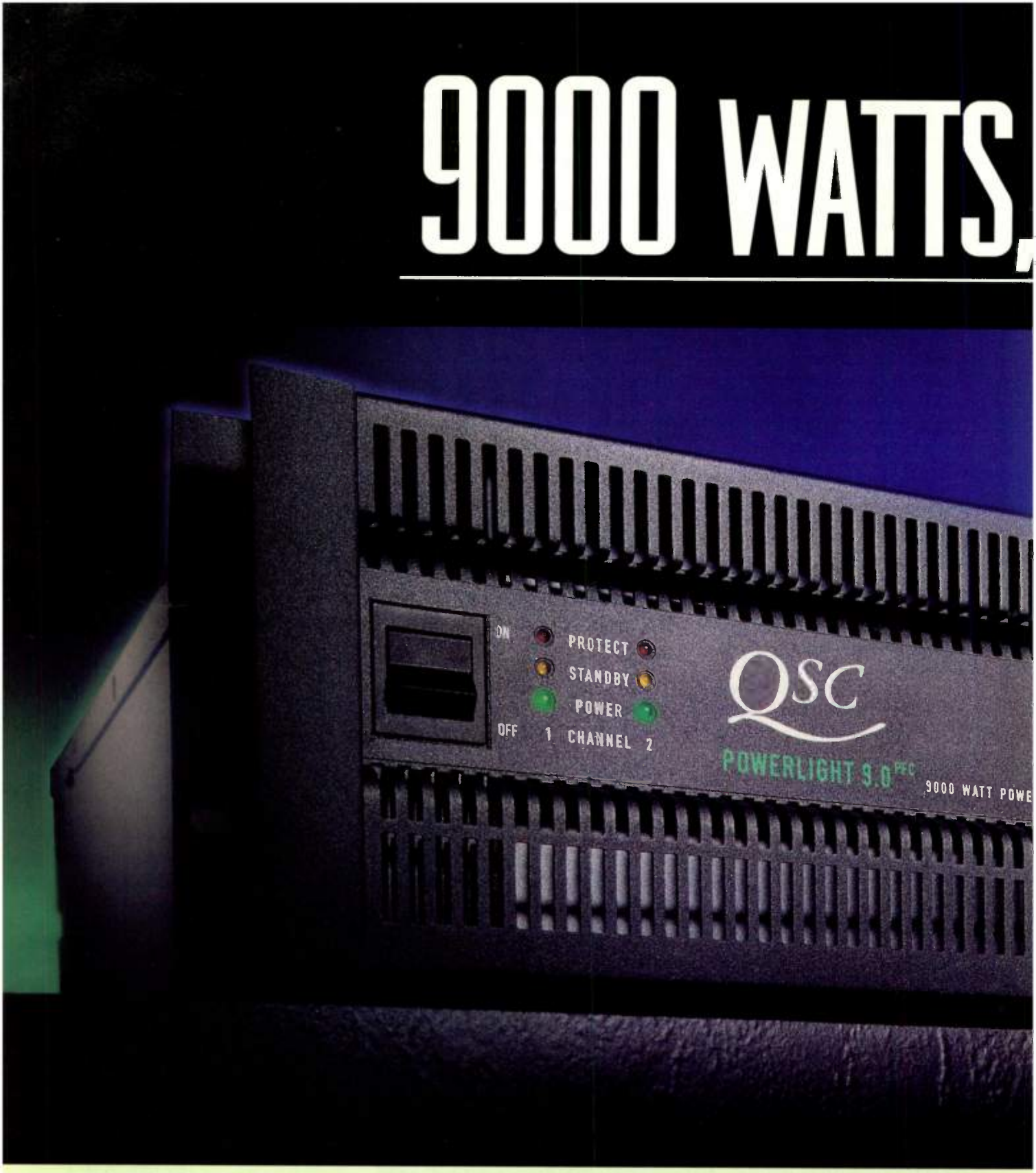
9000 WATTS,



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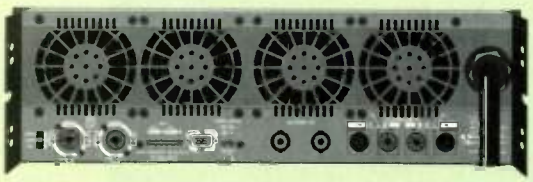

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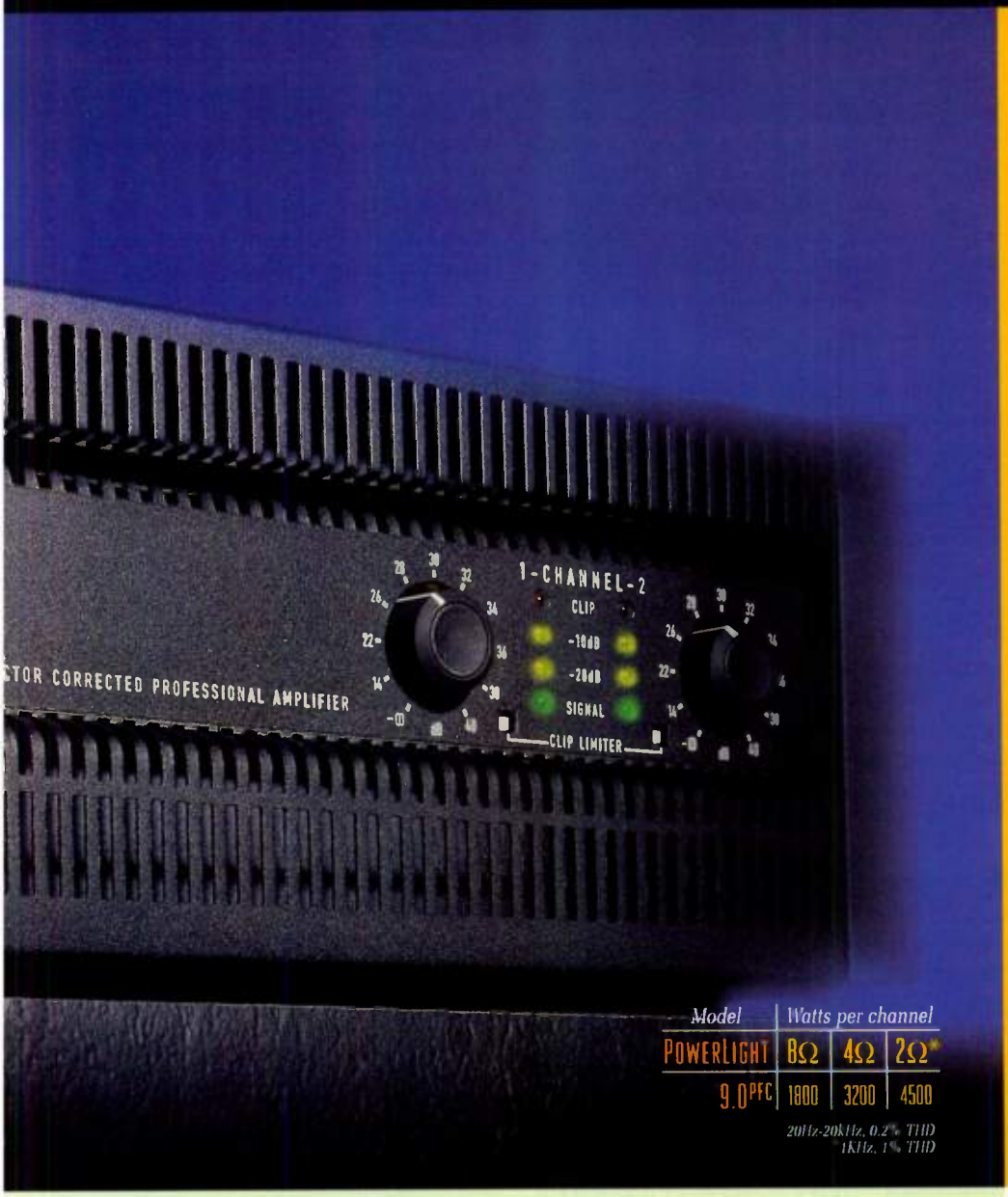

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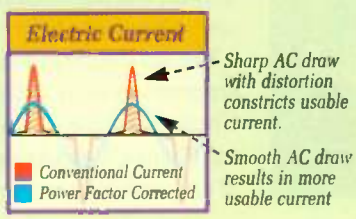

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QSC™ AUDIO

HEAR THE POWER OF TECHNOLOGY.

▶ SONY DPS-V55

Has the attempt to get the perfect Tori Amos piano sound completely tried your patience? Do you lust after Frank Zappa's guitar sound? Steely Dan's? Well, Sony has consulted with some of the recording industry's legendary producers and commissioned them to develop the presets on the new DPS-V55 ($550) multi-effects processor. Roger Nichols, Snuffy Walden, Joe Chiccarelli, Tom Jung, and Michael Bernard have all lent their ears and their expertise to the making of this 4-channel, 2U rack-mount device.

Two hundred presets, 200 user-editable programs, and SysEx librarian software can be downloaded from Sony's Web site. Effects algorithms are divided into 45 types, of which 9 are 4-channel, 27 are 2-channel, and 9 are mono. There are reverbs, delays, choruses, pitch shifting, compressors, phasers, and flangers. A Tap Tempo button allows users to set effects times and panning rate. You can search for various effects by type using a button, and you can search through individual effects using a velocity-sensitive data wheel. The effect is displayed on a 16-character, 2-line LCD. The unit's four channels can be used in mono, as two stereo channels, or for 4-channel "surround sound."

The unit's rear panel features MIDI In and Out/Thru connectors. Each of the processor's channels features analog I/O on unbalanced, ¼-inch connectors, with 20-bit converters. Internal processing is 52-bit. The DPS-V55 is switchable between +4 dBu and -10 dBV operating levels. Sony rates the unit's frequency response at 20 Hz to 22 kHz (+0/-1 dB, signal-to-noise ratio at >93 dB), and distortion at <0.005% (at 1 kHz). Sony Electronics; tel. (800) 635-SONY, ext. V55; fax (201) 358-4907; e-mail bob_tamburri@mail.sel.sony.com; Web www.sony.com/proaudio.

Circle #407 on Reader Service Card

▶ MIDIMAN SAM

A new, compact, stereo signal converter is available from Midiman. SAM ($399.95) converts ADAT to S/PDIF and vice versa, supporting sample rates from 39 to 51 kHz.

When converting ADAT to S/PDIF, you can send your signal through SAM's onboard mixer. For each of the eight ADAT channels, the mixer provides pan and gain knobs with 128x resolution. The mixed signal is then sent to left and right S/PDIF channels. In S/PDIF-to-ADAT mode, SAM directs the left channel to all odd-numbered ADAT channels, and the right channel goes to all even-numbered ADAT channels. Alternatively, there are four modes for direct, one-to-one channel transfers: ADAT 1/2, 3/4, 5/6, and 7/8 to S/PDIF left/right.

SAM offers 56-bit internal processing, and 24-bit output. The unit has ADAT Optical I/O for ADAT, and RCA jacks for S/PDIF. Along with the sixteen mixer pots, SAM's petite control surface includes left and right LED clip indicators, as well as LED mode indicators. Midiman; tel. (800) 969-6434 or (626) 445-2842; fax (626) 445-7564; e-mail info@midiman.net; Web www.midiman.net.

Circle #408 on Reader Service Card

▼ DIMENSION ARC REVOLVE 100M

Dimension Arc Software offers *Revolve 100m* ($145), a new pattern-based MIDI sequencer for Windows. The modular sequencing system is composed of two 103 Bassline monophonic sequencing modules, the 109 Drummer polyphonic drum sequencer, and the 183 Sequence Recorder for chaining patterns from the sequencers.

The two 103 modules each give you room for 32 patterns with 16 steps per pattern. Six assignable knobs that send Control Change messages are available on each module. (Dimension Arc plans an upgrade that will enable these knobs to be assigned to SysEx messages.) A keyboard display is included in the 103 interface, and the software can be controlled from a MIDI controller.

The 109 Drummer also lets you program up to 32 patterns with 7-note polyphony. There is a Control Change/SysEx knob for each of the seven voices in a pattern.

For chaining together your 103 and 109 sequences, the 183 module provides a tape deck–style interface and will record up to 1,000 measures with 24 ppqn resolution. All edits can be performed in real time and recorded.

Revolve 100m can import songs from Propellerheads' *ReBirth* and can convert them to MIDI. Songs can be synched to MIDI Clock through two sync ports, and sequences and songs can be saved as Standard MIDI Files. Minimum system requirements for *Revolve 100m* are a Pentium 60 with Windows 95/98; 16 MB RAM is recommended. Dimension Arc Software; tel. (604) 664-0403; fax (604) 684-3656; e-mail revolve@dimension arc.com; Web www.dimensionarc.com.

Circle #409 on Reader Service Card

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STOP THE PRESSES! ▲ ▲ ▲ ▲

THE EAGLE HAS LANDED

"Houston, Tranquillity Base here. The Eagle has landed." With these words, Apollo 11 astronaut Neil Armstrong announced to the world that the first manned lunar module, Eagle, had successfully touched down on the moon's surface. Although the impending arrival of Generalmusic's Falcon ($1,395) and Eagle ($2,595) digital mixers won't carry the historical impact of a lunar mission, they nonetheless could turn out to be of great interest to electronic musicians.

The 10-channel Falcon features six mono channels with both balanced, XLR mic/line inputs (including phantom power) and unbalanced, ¼-inch line inputs. You also get two stereo channel strips with ¼-inch L/R inputs. All mono-channel inputs have 20-bit, 64x oversampling converters. The stereo channels have 18-bit, 64x oversampling inputs. The mixer samples at 48 kHz (fixed).

Both pairs of L/R mix outputs (one on XLRs, one on ¼-inch jacks) have 20-bit DACs, while the two ¼-inch auxiliary outputs use 18-bit DACs. The headphone output is through a stereo ¼-inch jack.

You get one stereo pair of S/PDIF (switchable to "unbalanced" AES/EBU) digital I/O, provided on RCA jacks. This gives you a total of twelve inputs and eight outputs. In addition, the Falcon can accept the LEM ADAT Extension card (price tba), which adds eight more mixer channels with Lightpipe I/O, as well as twelve effects processors.

Each channel has a 3-band, fully parametric EQ; four aux sends (two with virtual controls for routing to the internal effects); pan; and 60 mm faders. You get one motorized, assignable fader in the master section, which allows you to write automation and to check the null positions of any function.

The mixer also has up to eight configurable processors that can be used serially or in parallel and assigned as needed to channels and outputs. These assignments are made using the Environment structure, which is similar to an effects rack inserted into a signal path. One Environment can contain a maximum of four processors, connected in serial or parallel. Processes include dynamics control (compressor, limiter, gate, de-esser); graphic and parametric EQ; aural enhancer; and 3D processing.

You get two internal effects processors; one dedicated to reverb, the other for multieffects. Signals are routed to these using the virtual Aux 3 and 4 channel sends. All parameters on the entire Falcon can be recorded in a snapshot; up to 128 snapshots can be stored in onboard memory.

The Falcon has a 128 x 64 LCD; MIDI In, Out, and Thru ports; a programmable footswitch connector; and an RS-2323 serial port for connection with a Mac or PC. The operating system resides in Flash memory and can be updated via the serial port or MIDI.

For those who need more than the Falcon can provide, Generalmusic offers the Eagle ($2,595), which has the same operating system, automation, and other basic features as the Falcon. This bigger bird has twelve mono mic/line inputs with balanced XLRs (with phantom power); balanced, ¼-inch jacks; Gain controls with 30 dB pad; and 3-band, fully parametric EQ. As with the Falcon, there are two stereo channels with 4-band graphic EQ.

All input channels have four physical aux sends, which can be assigned as pre- or postfader, and two virtual aux sends that address the internal effects. You also get Pan (mono)/Balance (stereo), Peak LED, View, Cue, and Mute controls, as well as 60 mm, motorized faders.

The output section includes two sets of stereo mix outs (on XLR and ¼-inch) and four aux outputs. These outputs are software-configurable. As with the Falcon, one stereo, digital I/O pair is provided on RCAs, which can be S/PDIF or AES/EBU. This gives you a total of eighteen inputs and ten outputs. You can cascade two Eagles or an Eagle and a Falcon together to form a larger system.

In addition to two multieffects processors for reverbs and delay-based effects, the Eagle offers 30 digital effects processors that can be inserted, four at a time, on any channel. The big bird also has a 1.44 MB floppy-disk drive for saving snapshot scenes, global machine setups, and effects libraries (reverb, delay, and dynamics processing).

The Falcon and Eagle user interfaces are designed to work like analog mixers in that all EQ, Aux, Volume, Cue, and Mute controls are physically located on the control surface. You can opt to disable automation for any control, allowing you to manually control some features while using snapshot automation for others. In addition, all parameters can be MIDI controlled via SysEx for dynamic automation from a sequencer. Generalmusic; tel. (800) 323-0280 or (630) 766-8230; fax (630) 766-8281; e-mail gmail@generalmusicus.com; Web www.generalmusic.com.

—Steve Oppenheimer

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▲ TC ELECTRONIC FINALIZER EXPRESS

TC Electronic's Finalizer Express ($1,599) is a slimmed down, less expensive version of the company's Finalizer Plus studio mastering processor. While the Finalizer Plus has more advanced features and flexibility, the Express provides numerous processes that are designed for faster, easier use. Like the Finalizer Plus, the Express employs TC's proprietary DARC3 DSP-chip technology, which yields 80 million instructions per second.

The heart of the Finalizer Express is its multiband compression and limiting functions. It has three bands, with fixed crossover points, of 24-bit digital compression and limiting. A level-normalizing function optimizes the gain structure and reduces noise.

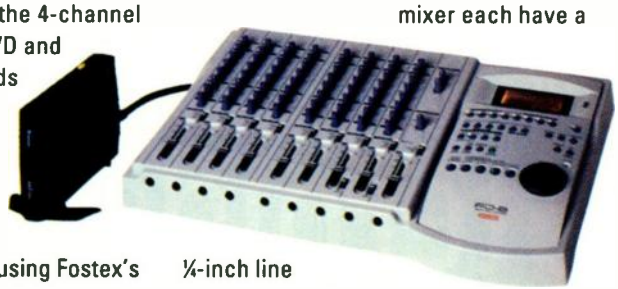
Analog I/O is on balanced, XLR connectors. The Express features 24-bit ADCs and DACs with 24-bit internal processing. You can dither a signal down to 16- or 20-bit resolution and switch between 44.1 and 48 kHz sampling rates.

AES/EBU digital I/O is provided on an XLR connector, and you get S/PDIF I/O on both optical and RCA connectors. There are MIDI In, Out, and Thru connectors, and an input is available for use with TC's Master Fader, which lets you record fades into a digital audio sequencer. Frequency response for the unit's analog output is rated at 10 Hz to 20 kHz (+0/-0.5 dB). The analog inputs have a dynamic range rated at >106 dB (A weighted) and THD at -95 dB @ 1 kHz. TC Electronic; tel. (805) 373-1828; fax (805) 379-2648; e-mail tcus@tcelectronic.com; Web www.tcelectronic.com.

Circle #411 on Reader Service Card

▶ FOSTEX FD-8

Fostex has taken the ideas implemented on its FD-4 4-channel digital multitrack recorder and produced the new FD-8 ($895). Like the 4-channel unit, the FD-8 has 16-bit A/D and D/A converters and records uncompressed, 44.1 kHz, 16-bit audio to an external SCSI hard-disk or magneto-optical drive. But the new digital multitrack also lets you record using Fostex's new Advanced Digital Audio Acoustic Coding (ADAC) compression scheme, which allows for more track time (up to 160 track minutes on a 230 MB SyQuest EZ-Flyer disk) than in uncompressed mode. A Mastering mode lets you record up to sixteen virtual tracks.



Channels 1-6 on the FD-8's analog mixer each have a ¼-inch line input, 3-band EQ with sweepable mid, monitor send, and pan control. Channels 7 and 8, besides providing these features, each have a balanced, XLR input and TRS ¼-inch inserts, with a 3-step trim level for gain adjustment. Each channel features two aux sends and two stereo aux returns.

The FD-8 has S/PDIF optical I/O and a ¼-inch footswitch jack for punch-in recording. There are RCA tape returns, main outs, and monitor outs. It can slave to MTC and sync to MIDI Clock. Fostex rates the unit's dynamic range at 105 dB (mixing section) and >90 dB (recording section), frequency response at 20 Hz to 20 kHz, and THD at <0.008% (all at 44.1 kHz). Fostex Corporation of America; tel. (562) 921-1112; fax (562) 802-1964; e-mail info@fostex.com; Web www.fostex.com.

Circle #412 on Reader Service Card

▶ CYCLING '74 M 2.5

Back in 1987, Intelligent Music came out with the novel, interactive composition program *M*. This program has been revived and updated, and *M* 2.5 (Mac; $74) is now available from Cycling '74.

Written by David Zicarelli, who coauthored Opcode's *MAX* (with IRCAM's Miller Puckette) and several other music programs, *M* 2.5 provides a variety of algorithms for turning musical fragments into complete compositions, as well as many tools for manipulating the output in real time, all controllable via MIDI. The music you produce can be saved in SMF format.

You can input melodies, chords, and rhythms as Patterns, using your computer keyboard, mouse, or MIDI keyboard. Using these Patterns, you create Voices, which include settings for an array of playback variables, such as transposition, MIDI Velocity ranges, and algorithmic parameters for cyclic variations in articulation. A unique Conducting window allows you to control tempo or other variables by dragging your mouse around an onscreen grid.

The new version supports both OMS and QuickTime Musical Instruments. In the MIDI Assignment window, you can assign each of *M*'s sixteen tracks to any input device and MIDI Out channel. *M* can also exchange data with *MAX*. An Input Control System allows you to assign a variety of functions to MIDI notes for easy, real-time control from your MIDI keyboard.

M can operate on any 680X0 Mac or PowerPC with Mac OS 7.5 or higher and 16 MB RAM. Cycling '74; tel. (415) 621-5743; e-mail info@cycling74.com; Web www.cycling74.com.

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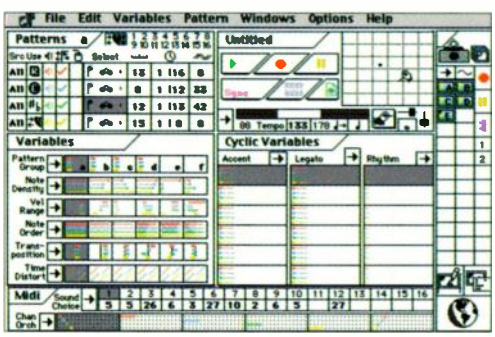
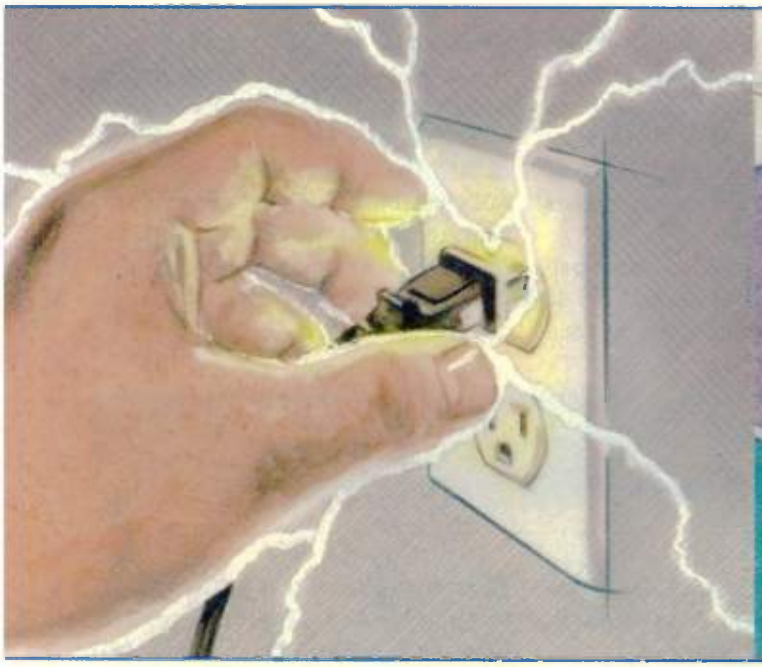
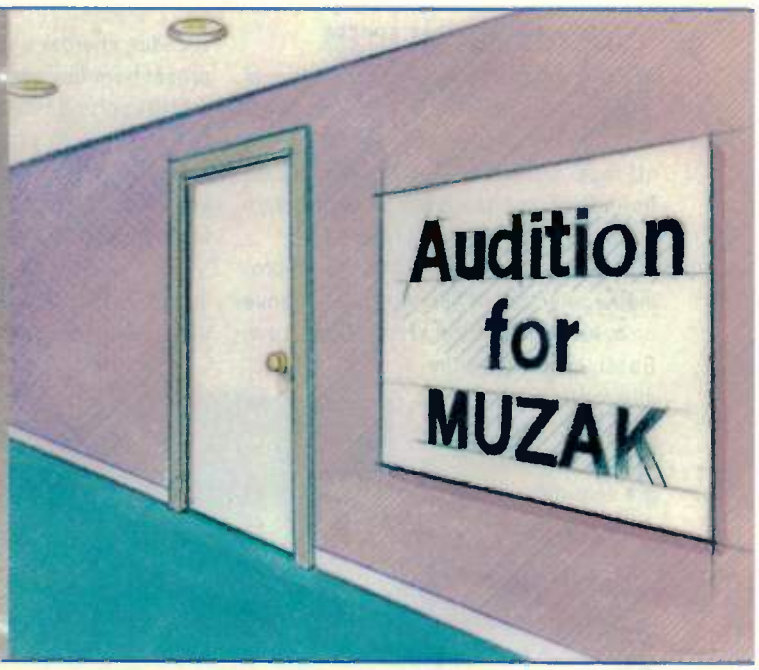
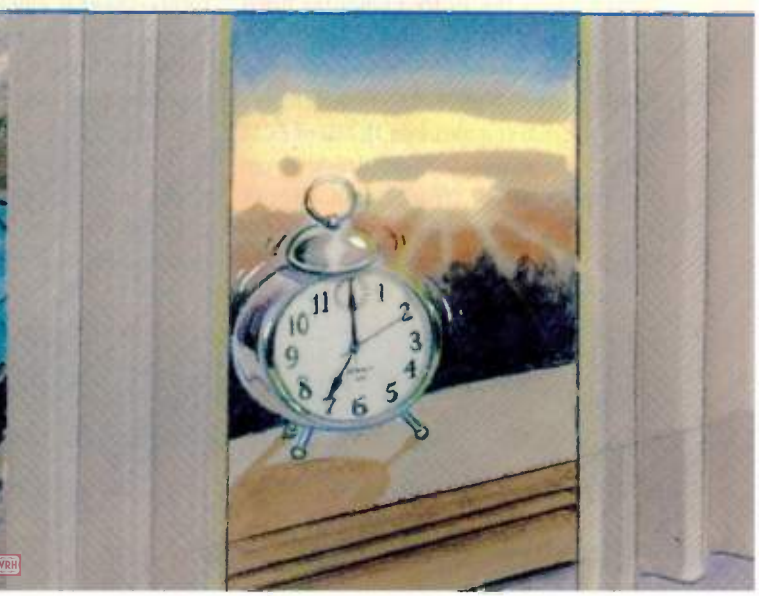


There are many things





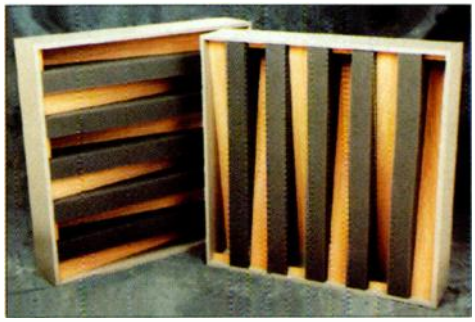
a musician can live without.



SOUND ACOUSTICS D/A BOX

As personal-studio owners seek to optimize the sound quality of their recordings, they usually focus first on types and use of microphones, the console, and other gear. Once aware of how equipment can affect sound, the next frontier is the tracking room itself. While some recordists may choose to build elaborate contoured walls and such to tune their rooms, many either don't have that option or simply want an acoustic treatment that they can put away at the end of the session. Sound Acoustics' D/A Box and I-Lite Box ($175 each; two for $299) are portable, stackable acoustic treatments that are designed to lend pleasing reverberation characteristics to your existing space.

The D/A Box, which weighs 29 pounds and measures 3' x 3 'x 8", is made from alternating foam strips and ¾-inch, oak plywood slats. The foam sections provide sound absorption, and the wood slats are positioned at an angle of eight degrees, compared with the plane of the standing box, to minimize flutter echoes and standing waves.



Also available is the I-Lite Box, which features a sheet of ¼-inch Plexiglas set into the frame, which is the same size as the D/A Box, again at an angle of eight degrees. Sound Acoustics; tel. (630) 832-3064; e-mail sound@ezhost1.ccm.net; Web www.soundacoustics.com.

Circle #414 on Reader Service Card

DIGITECH VOCALIST ACCESS

DigiTech has brought out a new vocal harmony processor and reverb unit. The Vocalist Access ($479.95) is a 1U rack-mount unit that provides 4-part harmonies and ten types of reverb with an onboard analog mic preamp.

Backlit buttons on the front panel provide a selection of harmony levels above or below your melody (e.g., High, Low, Bass), as well as a unison option for voice doubling. Four harmony processing algorithms are available: detuning, vocoder, chordal, and scalic. There are 50 preset harmony and doubling programs.

MIDI ports on the back enable you to harmonize with your MIDI keyboard or Standard MIDI Files, and they also allow you to use MIDI continuous controllers for harmony parameters.

Vocalist Access provides both a balanced, XLR input and an unbalanced, -10 dBu, ¼-inch input; left/mono and right outputs are on ¼-inch, unbalanced connectors at -10 dBu. The frequency response is rated at 20 Hz to 20 kHz, signal-to-noise at >92 dB (A weighted), and THD at <0.04%. The unit has 16-bit sigma-delta ADCs and DACs and a 48 kHz sample rate. DigiTech; tel. (801) 566-8800; fax (801) 566-7005; e-mail customer@digitech.com; Web www.digitech.com.

Circle #415 on Reader Service Card

KORG N1R

Korg has added to its family of AI^2 synths by rolling out the N1R synth module ($850), a 1U rack-mount version of its N1 synth. The N1R is 64-note polyphonic and 32-part multitimbral (when using the PCI/F computer port) with GM, GS, and XG sound-map support for optimal integration into a desktop music system. It packs 563 multisamples and 304 drum samples into 18 MB of PCM ROM, including samples drawn from Korg's M1, O1/W, Trinity, and SGproX keyboards. The module features 1,269 programs and 402 combinations, and users can store 100 of each. More than 1,700 sounds are available as programs, combinations, and drum kits.

Two independent stereo effects processors offer 48 types of effects, including resonant filter, chorus, delay, and rotary-speaker emulation. The N1R also features a 16-part Performance mode for access to layers and splits, as well as polyphonic portamento and other performance functions.

The front panel sports four dedicated knobs that control attack time, release time, filter cutoff, and effects parameters, as well as up to four user-assignable parameters for real-time adjustments in Performance mode. These settings can be stored as one of 32 Performances, enabling the user to save custom sound combinations. An onboard arpeggiator provides twenty preset patterns with editable parameters, and it can sync to a sequencer.

The 144 x 40-pixel backlit LCD displays most editing and performance information as icons. In Master mode, the user can program display colors to respond to GM, GS, or XG messages. Connectors include ¼-inch, L/R stereo outs; two ¼-inch aux outs; MIDI In, Out, and Thru; and a ¼-inch headphone output. A serial port allows connection to a Mac or PC (cable not included). Korg USA, Inc.; tel. (800) 645-3188 or (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

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*Number of simultaneous recording tracks is dependent on speed of hard drive and recording mode used.

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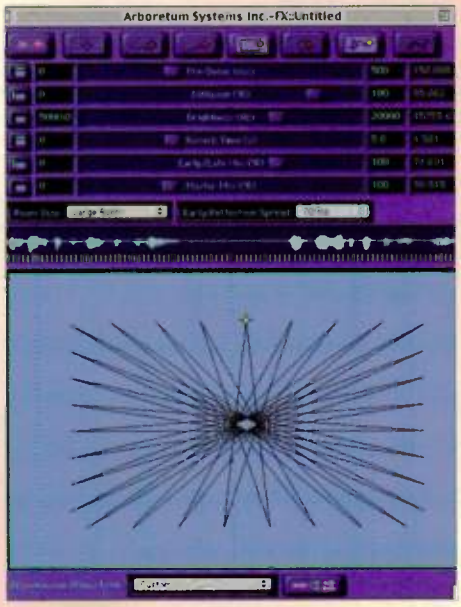
REV UP



ARBORETUM

Arboretum's *Hyperprism 2* (Mac; $249), the newest stand-alone version of the company's flagship product, adds an analog-style vocoder and Arboretum's new Hyperverb reverb algorithm for a total of 36 real-time effects. The 26-band vocoder, which works with AIFF and SDII files, can be used with two sound files, one file and a live signal, or two live signals. Also new to the program is the Z-Morph effect, which is similar to the vocoder but has filters whose frequency can be changed over time, creating a sound much like a talk box.

The ability to alter a live signal in real time, without the need to create or read a file or use your computer's hard drive in any other way, is made possible by *Hyperprism 2*'s inclusion of the company's HyperEngine, the framework within which *Hyperprism* effects run. HyperEngine, which uses 32-bit internal processing, comes with drivers for Sound Manager, Korg's 1212 I/O card, and Digidesign's Audiomedia II and III cards.

Also included with the program are compressors, ring modulators, flangers, chorus, tremolo, parametric EQ, and more. The number of simultaneous real-time effects is dependent on your computer's processor speed.

The program requires a 120 MHz Power Mac (200 MHz recommended), Mac OS 7.6 or later, and 16 MB RAM. Arboretum Systems; tel. (650) 738-4750; fax (650) 738-5699; e-mail sales@arboretum.com; Web www.arboretum.com.

Circle #417 on Reader Service Card

SONIC FOUNDRY

A new upgrade is available from Sonic Foundry for the *Sound Forge* audio editor (Win $499). Version 4.5 includes all the music and multimedia production and sound-design features as the last version—including an array of built-in processors and effects, DirectX support, and video synching—with several additions.

Two plug-ins that had been sold separately are now bundled with *Sound Forge*. These are the Batch Converter, which can process and convert thousands of files at once, and the Spectrum Analysis plug-in, which is a tool for interpretation and real-time, graphic display of waveform characteristics.

A new set of loop-editing tools has been added, featuring support for Sonic Foundry's *ACID*. For enhanced Internet authoring capability, *Sound Forge* now supports NetShow 3.0 and RealAudio/Video 5.0 audio streaming.

Sound Forge 4.5 requires a PC with Windows NT, 95, or 98; a minimum of 16 MB RAM is recommended. Sonic Foundry; tel. (800) 577-6642 or (608) 256-3133; fax (608) 256-7300; e-mail sales@sonicfoundry.com; Web www.sonicfoundry.com.

Circle #418 on Reader Service Card

SEER SYSTEMS

Responding to feedback from users, Seer Systems has introduced version 1.5 of its *Reality* software synthesizer (Windows; $495). *Reality* is 128-note polyphonic and employs several synthesis techniques, including physical modeling, FM, analog, and samples. The previous version only supported Creative Labs Sound Blaster cards; *Reality* 1.5 supports any Direct Sound–compatible sound card or motherboard.

Seer has beefed up the sonic selections with a new set of multisampled instruments and several new or improved algorithms. In addition, sample access has mushroomed because *Reality* now supports SoundFonts, and the new package includes a shareware version of FMJ Software's *Awave*, which converts more than 140 different sampler formats (e.g., Kurzweil, Roland, and Ensoniq) to SoundFonts. Several other shareware demos, including the *Seq-303* simulated-analog sequencer and the *ArpX8* 8-layer arpeggiator, both from TechnoToys, are also bundled with *Reality* 1.5.

With Seer's own *SeerMusic* plug-in now part of the package, you can post your audio files on your Web site. They can then be downloaded in real time, without compression or a high-speed modem, using Seer's own streaming technology.

Several refinements to the synth's interface and MIDI controls round out the program's new features. *Reality* 1.5 requires a Pentium 133 running Windows 95, with at least 40 MB RAM. Seer Systems; tel. (650) 947-1915; fax (650) 947-1925; Web www.seersystems.com.

Circle #419 on Reader Service Card

THE NEW YAMAHA MSP5 BI-AMP SPEAKERS WILL, LITERALLY, BLOW YOU AWAY!

The jet stream that shoots from the MSP5's vents tells you these speakers are powerful. The sound they project is nothing short of astonishing. And the $599 MSRP price per pair ices it—you ABSOLUTELY MUST get these speakers.

Inside the magnetically-shielded enclosures, smooth Waveguide horns and dual amplifiers—one 27W for the tweeter and one 40W for the woofer—create natural, distortion-free sound from 50 Hz to 40 kHz. You get more headroom and dynamic range than any other speaker in this price range, making the MSP5 perfect for monitoring or post-production. And, with a 4-position low frequency EQ switch and 3-position high frequency EQ switch, you can tailor the MSP5 output to your own environment.

The Yamaha MSP5. No other speaker can hold a candle to it. So, get to your nearest Yamaha Pro Audio dealer and prepare to be blown away!

YAMAHA

FATAR SL-990

Fatar's StudioLogic SL-990 ($1,095.95), is an 88-key, weighted-action MIDI master keyboard that weighs in at 44 pounds. The controller sends MIDI messages on channel 1, and its keyboard is mapped as a single zone. There are Bank Select and Program buttons on the front panel, and a Velocity Curve button allows users to choose from among eight Note On Velocity curves (which apply envelopes that shape the keyboard's Velocity response).

With Fatar's Contoured Strike Force Action weighted, hammer-action keyboard, when a key is depressed, the hammer is thrust up and forward on a pivot, approximating the way a piano hammer moves toward the strings. The Contoured Strike Force hammers are slightly heavier in the bass keys than in the middle octaves and are lighter in the treble. The design mimics the comparative weight and feel of a piano's lowest and highest keys. The instrument's keys are colored to more closely resemble those of a piano keyboard.

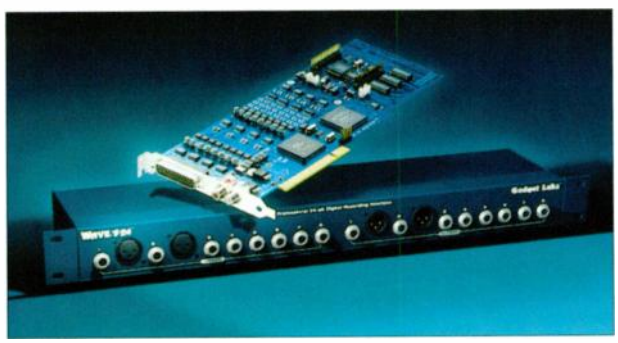
The rear panel sports a MIDI Out connector, a ¼-inch sustain-pedal input, and a connector for the 9 VDC adapter. StudioLogic by Fatar/Music Industries Corp. (distributor); tel. (516) 352-4110; fax (516) 352-0754; e-mail fatar@aol.com; Web www.musicindustries.com.

Circle #420 on Reader Service Card

GADGET LABS WAVE/8 • 24

Gadget Labs is offering PC-based musicians a new 24-bit, multichannel audio I/O solution, the Wave/8•24 ($499). The system consists of a full-size PCI card; drivers for Windows 95/98, Windows NT, and ASIO; and an 8-channel audio patch bay/interface. The hardware can be used with any software that supports the Microsoft WAVE device-programming standard.

The patch bay's front panel mounts eight balanced/unbalanced, ¼-inch inputs and eight balanced/unbalanced, ¼-inch outputs, which can operate at -10 dBv or +4 dBu levels (software switchable). Alternative, balanced, XLR inputs and outputs are provided for channels 1 and 2. All analog inputs and outputs use 24-bit, 128x oversampling converters. The system supports 11.025, 16, 22.05, 24, 32, 44.1, and 48 kHz sampling rates. An optional S/PDIF card ($129) adds 24-bit digital I/O on RCA connectors.



Up to three Wave/8•24 systems can be combined for up to 24 channels of audio I/O. Rather than having onboard DSP chips, the Wave/8•24 utilizes MMX technology, so it requires an MMX-capable, 166 MHz or faster, Pentium (or equivalent) CPU. The system also includes MIDI In and Out ports.

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PRO FILE

Eighty Miles High

Inclement Weather *is cool and casual.*

By Rick Weldon

Eighty Mile Beach, named for an isolated stretch of Australian coast, makes music that embraces a broad range of textures and styles. The band is made up of Beth Custer, a longtime Bay Area artist who holds a master's degree in clarinet performance, and Christian Jones, an engineer at San Francisco's Mobius Music. On their debut album, *Inclement Weather,* the pair evidence a strong compositional sensibility, with Custer singing and playing clarinets, trumpet, and piano, and Jones in charge of drum programming, turntables, guitars, samples, and bass. Numerous Bay Area stalwarts also lend their talents to EMB's breezy, electronics-laden mix of funky jazz and hypnotic balladry.

The album's tracks were recorded at Mobius and in the home of Patti Clemens, using Clemens's Mackie 24•8 mixer and some borrowed ADATs. For "There Are No Right Angles Found in Nature," Custer close-miked her clarinet with an AKG C 414 in order to capture the sound of the instrument's key clicks as she played. At mixdown, she gave the instrument more bite by running the track through the "Metal Guitar" effect on a Boss SE-70.

Sequencing drum patches from an Ensoniq ASR-10 into Passport's *Master-Tracks Pro* and Opcode's *Studio Vision Pro,* Jones often layered multiple loops and sequences, giving many tracks a shifting, propulsive feel. "Usually, rather than step-record, I play the drums into the ASR-10 so the track has some feel to it. Later, I can arrange the parts however I want in the sequencer."

On the aptly named "5 Loop 7," the syncopated drums play a loop consisting of three bars of 5/4 followed by one of 7/4. The inspiration for the song, to a degree, was ZZ Top's "La Grange." "We were trying to make a dirty, southern-rock kind of song," says Jones. "But we also thought it would be cool to program something in an odd meter, so we said, 'Well, how about a bar of seven in there?' We started to program it that way, and it worked."

The wonderfully thick, atonal sound of an instrument called the Crawdad further distinguishes the odd-meter boogie. Invented by Mobius owner Oliver DiCicco, who builds instruments as musical sculpture, the Crawdad comprises a cylindrical wood resonating chamber with two graphite rods rising from the front and two pedals at the bottom. To play the Crawdad, you pluck the rods with your hands, causing them to rattle against the sound board. This produces a sustained, guttural sound, the pitch of which can be altered by pushing on the pedals with your feet.

Other less-than-orthodox sounds appear on the song "Afterlude." "Talk about a geeky tech story," laughs Jones. "There was this old, messed-up organ sitting out in the hallway of our rehearsal space. The thing sounded awful. I banged on it one day and it made this really weird sound. It had an output, so we took a cable and ran the banging sound through a wah and a ProCo Rat distortion pedal. Then we ran that through an amp, miked and sampled it, and used it on 'Afterlude.'"

"If we're going to use a sample," Jones continues, "we really tweak it out, to make the sound as original as possible. Beyond that, we like to juxtapose textures and put in a lot of ear candy. But I've got to keep myself in check with the samples because, ultimately, it's the song structures and Beth's vocals that are most important."

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E-MU SYSTEMS

TECH PAGE

DIY OS

Linux means freedom for computer musicians.

By Scott Wilkinson

Most computer users love to hate Microsoft. Of course, it develops lots of helpful software that's used throughout the world. However, its proprietary grip on the source code for that software, especially its Windows operating systems, makes it impossible for anyone who is not employed by the company to fix bugs or adapt the code for their own purposes. They must wait for the company to release updates that might or might not address their specific concerns.

Of course, most software companies follow the same basic approach for software development and distribution. However, a growing movement within the information-technology community is taking an opposing view. This movement asserts that software should be free in two respects: it should cost end users nothing to obtain, and they should have free access to the source code in order to fix bugs and adapt it.

One centerpiece of this movement is Linux, a clone of the Unix operating system. Originally developed by Linus Torvalds in Finland, Linux is free to anyone. In addition, the C source code is also freely available. (Several companies assemble a CD-ROM with the kernel and various utilities and sell it for a nominal fee, which allows them to offer technical support, but you can download it for free from the Linux Web site at www.linux.org.) Users who find a bug or want to add a new feature can implement and share their idea with anyone who is interested. As a result of this ongoing, free-for-all development, there are always two versions of the Linux kernel: a stable version and a development version.

No official organization coordinates these unbridled development efforts, but such an anarchical approach requires someone to screen bug fixes, additions, etc. Coordinators include Torvalds and Alan Cox in England, who is in charge of updates to the stable version. He collects bug fixes, verifies that they work, and releases a new version of the stable kernel. Others maintain different aspects of development.

This concept is sometimes called *open source* (www.opensource.org). Interestingly, Netscape recently joined the open-source community and released the source code for *Communicator* 5.0, including a version for Linux. In addition, Corel has contributed significantly to the Linux kernel and compiler, and the company is rumored to be talking about releasing all of its software as open source, including *WordPerfect Suite*.

The Linux kernel, available for PowerPC, 680x0, 80x86, and many other hardware platforms, includes support for audio devices, and there are several audio drivers to choose from. Even better, numerous audio applications are available, including synths, audio editors, MIDI software, hard-disk recorders, music notation, and DSP software. In addition, many music and audio applications developed for Silicon Graphics machines are now being ported to Linux (see **Fig. 1**).

In addition to being free of cost and restriction, the stable version of Linux is exactly that: stable. Some Linux systems have up times of more than three years. In addition, Linux multitasks much better than the Mac or Windows, which means you can start an audio batch-processing job and then do something else without worrying that the system will slow to a crawl.

As the open-source movement grows, more software of increasing quality will become available. Electronic musicians are bound to benefit greatly from the work of millions of fingers on millions of computer keyboards around the world. And for those who aren't afraid to get their hands dirty in the code, it's hacker's heaven.

Thanks to Russell Pickett (emerson@hayseed.net) and Dave Phillips (www.bright.net/~dlphilp/linux_soundapps.html) for their help with this article.

COURTESY DAVE PHILLIPS AND *LINUX JOURNAL*

FIG. 1: *Mix* is one of several audio applications that have been ported from SGI to Linux.

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Art or Theft?

By Paul Myers

SAMPLING OPINIONS ON COPYRIGHT

The debate over intellectual property ownership is not a new one. Artists have always borrowed from other artists, and there has long been a fine, if somewhat blurry, line between influence and plagiarism. Jazz soloists frequently "quote" other artists' melodies within their solos. Decades before the digital age, incorporating others' licks and riffs into one's own material was considered essential to the journeyman blues player's trade. Sometimes, an influence or a quotation becomes plagiarism in the eyes of the law. Certainly, Led Zeppelin learned an expensive lesson when Willie Dixon successfully sued them over "Whole Lotta Love" and its striking similarity to Dixon's "You Need Love."

A forum on sampling and the legal definitions of "fair use."

In more recent years, the digital sampler has become as ubiquitous in popular music as the electric guitar was in the '60s, from Vanilla Ice's 1990 hit "Ice Ice Baby," which borrowed heavily from the David Bowie and Queen song "Under Pressure," to James Brown's frequent virtual appearances with Public Enemy, Janet Jackson's recent "digital duet" with Joni Mitchell on "Got 'Til It's Gone," and Puff Daddy's use of Bowie's "Let's Dance" in "Bad Boys."

In the early days of sampling, publishing companies were slow to understand both the legalities and the process of sampling, if they were aware of it at all. Nowadays, thanks to some high-profile lawsuits, most musicians are at least vaguely aware of basic copyright rules. The common sense and legally safe approach is clear, as in "clear all your samples." Even so, there's still plenty of confusion about what is fair use and what is not.

Illustration by Barry Fitzgerald

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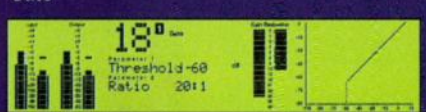
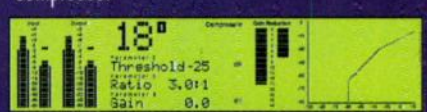


Start with the gate. Set parameters for threshold, ratio, attack, hold, release, and output gain. See the effect of your settings on the graphical display, as well as on the gain reduction and audio level meters, they all interact in real time with your manipulation of the parameters. Start with a threshold setting of about -60dB to clean off the noise in between the vocal takes. You can save your final gate settings as a "gate preset" building block and recall it into any other setup you do.

Compressor



Then move to the compressor. The effects of the gate settings are still visible on the graphic display, so let that help you determine where to set your compressor threshold. The parameters you change here will also effect the curve on the graphical display in real time. Move through all the regular parameters, like threshold, ratio, attack, release, and output gain. For vocals use a threshold of about -25dB, a ratio of about 3:1 or 4:1, and a slow attack and fast release for the most natural sounding effect. Your compressor settings can also be saved off as a building block to be called up into any other preset.

Limiter

On to the limiter. Changes you make to the limiter settings are also seen on the graphical display. Adjust the level up or down to suit your needs. The flat top line of the display moves up and down as you adjust the level. You can also set the speed at which the the limiter lets go of the signal as it goes below the threshold. This is truly smooth limiting, with patented dbx PeakPlus™ algorithms, so rest assured that where ever you set your threshold level, your tape will not distort, and your signal will not get butchered as it goes across the threshold. And like the other parts of the processor, your limiter settings can be named and saved for later recall.

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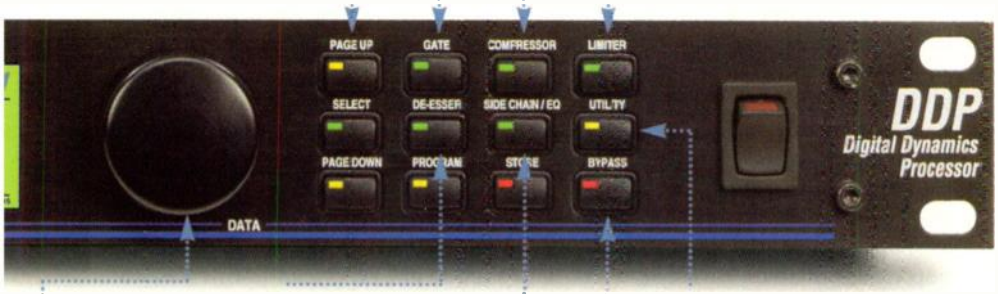

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De-essing works the same way: see the effects of your settings displayed on the graph. Parameters here are the common ones: threshold in frequency, 800Hz to 8kHz, and amount in percent. Other available processing includes EQ - both in-path and sidechain, for special-effect types of processing. When you are editing any of the building blocks, its icon is visible on the display, and the parameters are shown on the graph, so it's always easy to know where you are. Parameters are easy to see in this page driven operating system. When it's as complex as this, it's nice to know somebody was thinking when it was put together.

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You can also work in stereo, or set up a completely different and independent processing chain for the other channel. Also notice that the audio meters are capable of showing both peak and average levels for input and output. Optional digital output with the TYPE IV™ Conversion System with TSE™ (Tape Saturation Emulation) provides up to 24-bit output in either AES/EBU or S/PDIF formats with the trademark digital processing of TYPE IV™. The DDP also has full MIDI/Automation capability, with separate midi in and thru jacks. Entire processing setups may also be saved into one of 50 user defined presets, or use one of the 50 factory setups.

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DAWN OF THE SAMPLE

Any history of found sounds would have to mention *musique concrète*, the term given by French composers Pierre Schaeffer and Pierre Henry in the late 1940s for their compositions made largely from edited and rearranged tape recordings. Although their work bears little resemblance to today's popular sample-based music, Schaeffer and Henry definitely influenced notable sound sculptors such as Karlheinz Stockhausen, Negativland, and John Oswald. And, of course, tape loops were employed by the Beatles on "Tomorrow Never Knows" and "Revolution No. 9."

In case you missed it (and where the heck were you, anyway?), it was the emergence of hip-hop culture—especially its use of scratching—that introduced the notion of the record player as a musical instrument and recordings as raw materials to be dropped in by a DJ behind a rapping MC. As these crews developed their technical prowess using nothing more than "two turntables and a microphone," custom-pressed 12-inch dance records emerged that were built entirely from bits of sound nicked from existing recordings. Often issued with a plain, white label and mainly circulated among DJs, these singles were a phenomenon that most copyright professionals had never even heard of, let alone knew how to license. But when the Sugar Hill Gang's "Rapper's Delight" (essentially Chic's "Good Times" with added dialog) crossed over to the pop charts in 1979, this style of music began to make a good deal of money. And if there's one truth in the music business, it's that people always take notice when there's money involved.

The beginning of the '90s saw the digital sampler become affordable to more musicians. Now, sampling is everywhere, and the genre boasts true virtuosos like DJ Shadow, Prodigy, and the Dust Brothers, whose groundbreaking work with the Beastie Boys on *Paul's Boutique* presaged still more innovative work, such as Beck's *Odelay*. Thanks to these artists and others like them, samples and loops are everywhere; and although many publishers will admit that it's not quite the same as bootlegging, most view uncleared samples as theft or infringement of copyright.

ALL CLEAR

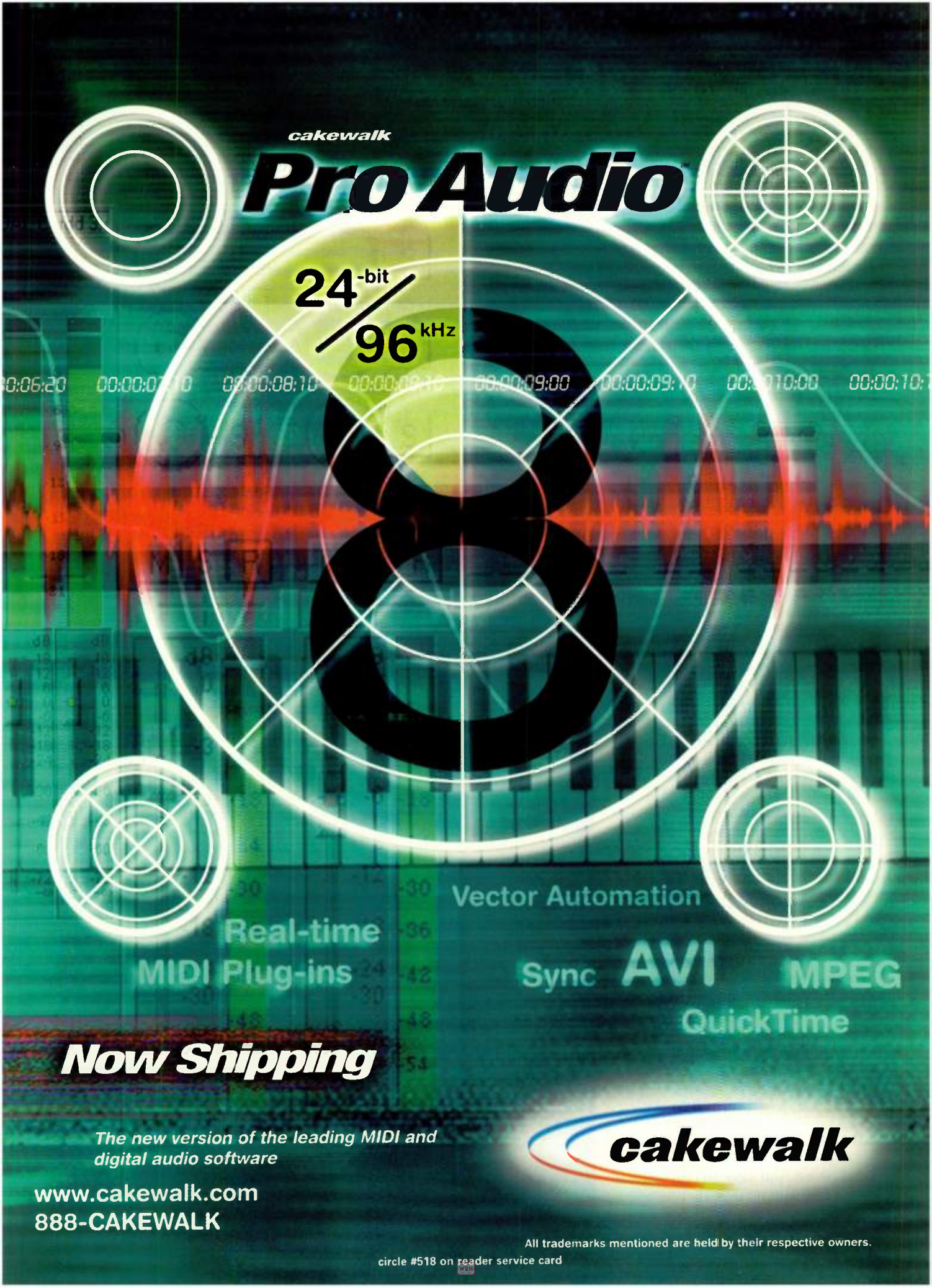
One thing many musicians find confusing is that there are *two* copyrights to clear for every sample they use. The sample must be cleared with the owner of the copyright of the composition (a publishing company) *and* the owner of the copyright of the master (a record company). For example, when you buy an album, it's yours to listen to, use as a frisbee, or wear as jewelry. You do not, however, own the rights to the songs or compositions on that disc, and you do not own the rights to the original sounds encoded on that disc. Record companies pay licenses and artist advances in exchange for the exclusive right to manufacture—and profit from the sale of—copies of the original sound recordings of the artists they release on their labels.

Ask musicians if they clear all their samples, though, and many will admit that only the big-name samples (i.e., those recognizable ones that come with lawyers attached) get the clearances. That's a reasonably safe game to play, but let the sampler beware. With the exception of certain specific circumstances, any copyrighted recorded sound superimposed onto your original recording without the appropriate license is considered an infringement of that copyright.

David A. Basskin, general manager of CMRRA, Ltd., a Canadian mechanical rights licensing agency based in Toronto, has seen many misconceptions from musicians, producers, and record labels alike. "People think there's this magic number of notes you can take before it's considered infringement," he explains. A frequent panelist and guest speaker at copyright conferences and artist forums around the world,

ROSS HARRIS

The Dust Brothers, Mike Simpson (left) and John King (right), have cultivated a production style that is heavily influenced by the use of samples. King says that the duo makes it standard practice to include clauses about sample clearances in every contract.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Basskin points out, "The general rule of thumb has been that, if enough of the underlying work has been taken that someone could recognize it, then it needs to be authorized." In other words, it's infringement if the artist who created the original work can recognize it. So even if you're sampling James Brown's scream from the sample made by Public Enemy, you are responsible for finding out who owns the rights to the original.

Jill Alofs of Total Clearance, Inc., a rights-clearing organization based in Mill Valley, California, regularly researches and tracks down those rights holders and negotiates sample clearances. "My job," explains Alofs, "is to let people know what the legal ramifications are so that they can be well-informed about their options, because as the attorneys get involved, it can get expensive, even if you have only received a cease-and-desist order and have to retain an attorney to write letters back."

Certain artists, such as Michael Jackson, the Beatles, and particularly the Rolling Stones, are harder to clear and more expensive to license. But Alofs stresses that the need to get clearances and pay for samples shouldn't dissuade an artist from pushing the envelope creatively. "You don't want there to be so much paranoia among artists that they just give up on using samples."

BITTER SWEET STORIES

However, to validate one's own paranoia about using uncleared samples, one needs to look no further than current Britpop stars the Verve. In a much-publicized case, the band learned the hard way not to mess with the Rolling Stones. The centerpiece of the Verve's breakthrough hit, "Bitter Sweet Symphony," is a looped passage of an orchestral recording of the Stones' "Last Time" as originally played by the Andrew Loog Oldham Orchestra. (Oldham, incidentally, was once the Stones' manager and producer.) Verve singer Richard Ashcroft sang his own original lyrics and melody over the orchestra sample, and the band added a modern

SAMPLED QUOTATIONS

Following is a sampling, if you will, of quotes about sampling, copyright, and audio collages.

A28 YNE

"Maybe he ought to write his own riffs...I guess if he's going to nick something, he might as well nick something good."—*Robert Plant commenting on Rick Rubin's sample of Led Zeppelin's "The Ocean" for the Beastie Boys' "She's Crafty" (as told to David Fricke in* Rolling Stone*)*

"You want the thing, not the almost thing."—*sampling pioneer Steve Steinski*

"The best way you can protect yourself against infringement is to become famous and successful. Because unless you're famous and successful, it's unlikely that anything you have will be subject to infringement, and you won't be able to afford to do anything about it."—*David Basskin, General Manager of CMRRA, Ltd.*

ation for

"A good composer does not imitate, he steals."—*Igor Stravinsky*

"We were the biggest nickers in town; plagiarists extraordinaire."—*Paul McCartney (Musician magazine)*

"When history is replaced by stories, the curator becomes a storyteller, her path an adventure through the cultural landscape, creating meaning and resonance by combination and juxtaposition. Think of modern sampling musicians like Howie B., whose work is essentially the 'curating' of his record collection, creating new music out of the juxtapositions of existing musics."—*Brian Eno, from his memoir* A Year with Swollen Appendices *(Faber and Faber)*

"It's just amazing what can happen. It carries one step further the idea of splicing together different cultures: words, phrases, textures, everything. With a kind of technical perfection, anything can be mixed right in with anything else. You get this massive rhythm going, and it's bigger than any song or musical entity."—*David Byrne (as told to Robert Farris Thompson in* Rolling Stone*)*

"If creativity is a field, then copyright is the fence."—*John Oswald*

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What turned the music technology industry on its ear?

beat to complete the track. When a seemingly routine clearance was requested from the Stones' publishing company, the band encountered a major snag. His name was Allen Klein.

A legend in the music business, Klein had briefly managed the Stones (and the Beatles) during the 1960s and still controls most of the Stones' copyrights through his company, Abkco Music, Inc. Klein also reportedly disdains sampling in principle and wasn't enthusiastic about granting a sampling clearance for one of his coveted Rolling Stones copyrights. But after EMI Records executive Ken Berry played Klein a dub of "Bitter Sweet Symphony," Klein is said to have told Berry that, because he liked the new record, he would allow the sample to be used and quoted a 50/50 publishing split for the new work. The Verve assumed that Klein had agreed to let them have a 50 percent cut of the new song; after all, Ashcroft had made up his own melody and lyrics. But what Klein meant was that, as writers of "The Last Time," Mick Jagger and Keith Richards would not only be credited as sole writers of "Bitter Sweet Symphony," they would each receive a full 50 percent cut. Ashcroft, meanwhile, would earn a mere $1,000 for his lyrics.

"They rung up and said we want 100 percent or take it out of the shops, you don't have much choice," the Verve's Simon Jones later told Betsy Powell in the *Toronto Star*. "At the end of the day," Ashcroft added, "that song, whoever owns it...has opened up many doors around the world for us. On [our] next album, when we don't go into any of Allen Klein's back catalogue and we make our own symphonies, there'll be the Verve and that's it. [The Rolling Stones] just had the biggest hit they've had since 'Brown Sugar.'"

Ironically, the Stones recently turned to sampling kings the Dust Brothers to give them a groovy makeover for the loop-crazy '90s. John King, one half of the ace L.A. production duo behind hits by celebrated "borrowers" Beck and the Beastie Boys, says that one way he and partner Mike Simpson stay out of trouble is by being creative in the way they manipulate their sounds. And they also make it standard practice to get sampling clauses in every contract and make sure every recognizable sound is cleared. If for any reason they can't, they bite the bullet and start again.


ALISON DYER

Beck (pictured above) is renowned for the use of samples in his work, most recently on his breakthrough hit album, *Odelay*. The "cultural terrorist" group ®™ark has released an album called *Deconstructing Beck*, which uses unauthorized samples of Beck's work.

"We don't use things without permission," King affirms, "so if we've got a sample that we have to get permission for, that's when we make the decision as to whether or not it's worth it." King says that, compared with the early days of hip-hop, producers and artists are generally wiser to the laws of copyright. "Back then, nothing was getting cleared," King remembers. He recalls a sample from an early Tone Loc record that the duo produced. "The publisher said 'no,' so we pulled it off and put in some other thing that worked just as well."

It's a shame that the Verve didn't consult with the Dust Brothers before their bittersweet debacle. King might have warned them that, if you sample large, identifiable chunks of another artist's record, there's a good chance you'll end up forfeiting 100 percent of your own copyright in the bargain.

"If you take a hit song, the original artist could ask for all of it," King explains. "It's like having that artist on your record. I mean, Hammer just basically rapped over 'Super Freak' [for his hit "Can't Touch This"]. Yeah, he composed all the lyrics, so technically, he's like a 50 percent writer on the song. If he could get that deal, good for him. But, if I were Rick James, I wouldn't give him that deal because before MC Hammer came along, 'Super Freak' was already a huge hit. Anyone could write new lyrics over a hit song."

FAIR USE

In case you're interested in the legal language behind the concept of fair use in U.S. copyright law, we have excerpted below the Fair Use provision of the revised Copyright Act of 1976.

Limitations on exclusive rights: Fair use.

Notwithstanding the provisions of sections 106 and 106A, the fair use of a copyrighted work, including such use by reproduction in copies or phonorecords or by any other means specified by that section, for purposes such as criticism, comment, news reporting, teaching (including multiple copies for classroom use), scholarship, or research, is not an infringement of copyright.

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(2) the nature of the copyrighted work;

(3) the amount and substantiality of the portion used in relation to the copyrighted work as a whole; and

(4) the effect of the use upon the potential market for or value of the copyrighted work. The fact that a work is unpublished shall not itself bar a finding of fair use if such finding is made upon consideration of all the above factors.

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As do many people in the business, King cites the saga of Biz Markie as the first instance of an uncleared sample drawing legal fire. A lawsuit was filed after the rapper appropriated the 1972 Gilbert O'Sullivan hit "Alone Again (Naturally)" as the foundation of his own song "Alone Again," from the 1991 album *I Need a Haircut* (Cold Chillin).

"In the Biz Markie case, the judge not only barred any further sale of the album, but he also referred the case to the U.S. Attorney for possible criminal prosecution," says Alan Korn, a San Francisco attorney who specializes in entertainment, media, and intellectual property law. Korn notes that the U.S. Attorney may have declined to prosecute the case, as there was no subsequent record of any prosecution noted in legal journals nor in the mainstream media. Markie made his feelings clear about going through the legal wringer when he released his next album for Cold Chillin, 1993's *All Samples Cleared.*

After the Biz Markie case, one would assume that full disclosure, at least for the most blatant of samples, would be the rule. But in 1990, Robbie Van Winkle, aka Vanilla Ice, sampled the signature rhythm, bass line, and piano from David Bowie and Queen's international chart topper "Under Pressure" for his megahit "Ice Ice Baby." Whereas Hammer had negotiated a deal with Rick James up front, Ice neglected to even mention the source of his glaringly pilfered main hook, let alone get the necessary clearances. The liner notes credited Earthquake, M. Smooth, and Vanilla Ice but failed to even thank Queen or Bowie. When legal threats heated up, the Ice melted, and an out-of-court settlement for an undisclosed sum was negotiated.

King suggests that, rather than generating lawsuits, the sample business should be run more like a musicians' union. He offers a few ideas about how to go about clearing samples, based on this scenario.

"If you use a bit of someone else's music," he says, "it's fair for that person to point to it and say, 'Hey, wait a second, I deserve something for that!' Whether they deserve a session fee, master license use fee, or writing credit should depend on the usage. If you work with someone on a record, everybody gets paid differently depending on whether they produce, engineer, play bass, or sing. It should be the same way for samples. It shouldn't be super-expensive for a tiny beat."

King offers a case in point, James Brown's "Funky Drummer." One of the most sampled grooves of the early '90s, the piece wound up on hits by Fine Young Cannibals, George Michael, and Sinead O'Connor to name but a few. "That drummer," King says, "should have gotten a session fee, James Brown should have gotten an artist fee, and the label should have gotten a master license use fee." Sounds fair, doesn't it?

ALL'S FAIR

Fair is also the first word in a significant legal exemption that affords the reuse of copyrighted works, free of infringement fear, for educational purposes or news reporting. The Fair Use provision of the revised Copyright Act of 1976 (see the sidebar "Fair Use") is rarely cited outside of academic circles, but for our discussion, we'll look at a couple of noteworthy exceptions. The first was a landmark case involving a notorious rap bad boy and a pretty woman. The second involved the letter *U* and the numeral *2*.

In July 1989, 2 Live Crew released "Pretty Woman," a version of the Roy Orbison hit "Oh Pretty Woman," on their album *As Clean as They Wanna Be* (Li'l Joe Records). Luther Campbell, the leader of 2 Live Crew, assembled the signature riff from the Orbison song but substituted much of the lyrics with his trademark brand of verbal free expression. Orbison and cowriter Bill Dees were credited on the Crew album as the source of the original song, to which Campbell and his Crew added their customary scratching, breakbeats, and assorted out-of-key melodies. But when Acuff-Rose Music, owners of the song, attempted to sue Campbell for infringement, the suit made legal history in 1994. Campbell's lawyers invoked the Fair Use provision, which had not previously been claimed on a commercially released recording. Asserting that the 2 Live Crew version was, in fact, a parody—and given that the audience for his product was substantially different than that of the original—Campbell's legal council successfully argued that the song posed no threat to the market for the traditional work and was therefore within the legal definition of fair use.

Out of necessity, musical sound sculptors Negativland, whose recorded works

CHRIS FLOYD

The battle between the Verve (above) and Allen Klein over the use of the Stones' "The Last Time" left the Verve little revenue from their hit song "Bitter Sweet Symphony."

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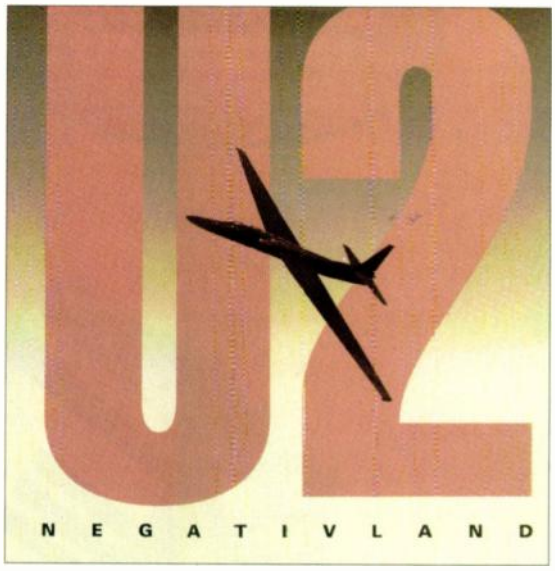
revolve around "found sounds," have become reluctant activists in the cause of fair use. The collective is as well known for their legal dispute with Island records as for their own work. The lawsuit cost them $45,000 and the good will of their label at the time, SST Records. The controversy centered around their record *U2*, which featured samples from the well-known Irish rock group of the same name. (After some bad blood with SST that stemmed from the legal dispute, Negativland formed their own label, Seeland.)

Out of their misfortune, the group wrote and published a book entitled *Fair Use: The Story of the Letter U and the Numeral 2*, which has become something of a manifesto for them. It is their belief that copyright laws are controlled by a corporate culture that has colonized the arts. They use the sound bites of mass culture as the raw materials for their collage works that, in a parodic way, comment on and react to the noise pollution that they feel is thrust upon them without their choosing. The mulched fragments end up bearing little resemblance to their original sources. Negativland's latest album, *Dispepsi*, was created from fragments of actual Pepsi Co. advertising and corporate training recordings. Surprisingly, Pepsi didn't take any legal recourse.



Reluctant activists and sound sculptors Negativland were forced to recall all copies of the album *U2* and return them, along with the original artwork, to Island Records as part of the legal settlement in the sampling lawsuit against them.

Negativland's Mark Hosler says he prefers to address the aesthetic and ethical issues over legal ones. "I'd rather try," proffers Hosler, "to encourage people to think about what's right or wrong as opposed to just making laws. The band Deep Forest have taken pygmy samples and made millions worldwide. I think pygmies should get paid, not for legal reasons but for ethical reasons."

Negativland also promotes the idea that corporate ownership of a culture spells the death of that culture and that current copyright laws are just a symptom of today's corporate control.

"You cannot enter the gates of culture," Hosler suggests, "unless you pay for this sample or little chunk of sound." But Hosler foresees a kind of decriminalization in the wake of growing public support for creative appropriation.

"You can see that the impulse to reuse things creatively is really mainstream now," he continues. "It's fun to take chunks of things and rearrange them; it's just another thing to do with music. So many people are doing it that it's getting harder for the lawyers to keep up. A sort of common law is almost emerging because so many people are developing work this way and acting like they are allowed to. The lawyers end up having more time to go after the real criminals, the bootleggers." Hosler stresses that he and his colleagues in Negativland are adamant about the distinction between bootlegging and the "transformative reuse" of fragments of other works.

Canadian sound-collage artist John Oswald also takes minute portions of recognizable sonic textures to create abstract musical mosaics that often bear no relation to their source material. Although his work is not remotely mainstream, he has been commissioned by Elektra Records to produce an aural collage of their catalog. Also, the Grateful Dead hired him to produce a sort of audio documentary, called *Gray Folded*, culled from hours of concert recordings of the Dead song "Dark Star."

Like Negativland, Oswald found himself embroiled in a legal matter involving his *Plunderphonics* CD, a collection of aural collages created with uncleared samples of everything from Stravinsky to Michael Jackson that was given away for free. Free or not, the Canadian Recording Industry Association, representing Sony Music and Michael Jackson, seized all copies of the disc and destroyed them. Oswald assumed that he would

MICHAEL BENABIB

Biz Markie's 1991 album *I Need a Haircut* landed him in so much legal hot water that he titled his subsequent album *All Samples Cleared.*

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not have had the capital resources to mount a winning legal defense and says that the incident, which some critics have compared to Nazi book burnings, may have ended up being more embarrassing to his opponents. "I think everybody realized that they were on shaky ground as far as public support. I mean, you've got Michael Jackson going after some guy with no money who's giving these CDs away for free." Like Negativland, Oswald sees a distinction between bootlegging and what he terms "the creative rearrangement of a piece of music in time."

WHO STOLE THE SOUL?

Even samplers get sampled. Hosler recalls the time a sample from Negativland appeared on a million-selling record by Marky Mark and the Funky Bunch. And even though they were never given any money, or even asked for permission, their complaints were anything but legal.

"I thought it was a stupid track, but we didn't care that he did it," remarks Hosler. "Curiously though, on that same album, Marky Mark sampled Lou Reed's 'Walk on the Wild Side' and was telling everyone in interviews how good they were about making sure all the samples were cleared and credited. We were quite amused because the very first sound you hear on his CD is Negativland. So it comes down to high profile versus lower profile."

That may be, but even the high-profile Dust Brothers say they've noticed a few high-profile artists culling uncleared samples from *Paul's Boutique.* "People like Bobby Brown and Janet Jackson have sampled us without our permission," says King. "A lot of times the producer, not the artist, will grab a sample off of our records the same way we might grab sounds off other records."

Also interesting is that Beck, an ardent appropriator himself and one of the Dust Brothers' more high-profile clients, sits at the center of a recent controversy that has drawn Negativland back into the fray. Like U2 before him, Beck's legal representatives were none too pleased to hear about the release of *Deconstructing Beck,* a CD that is based on unauthorized Beck samples.

The disc, funded by self-described "cultural terrorist" group ®™ark on the Illegal Art label, has prompted legal threats toward Illegal Art founder Philo T. Farnsworth (not his real name), who claims that his label exists to provide an outlet for artists interested in exploring what he terms "an illegal palette." In a press release, Farnsworth calls *Deconstructing Beck* an act of "digital graffiti" and justifies his lack of legal clearances by contending that existing copyright laws do not allow for this type of expression and therefore need to be amended.

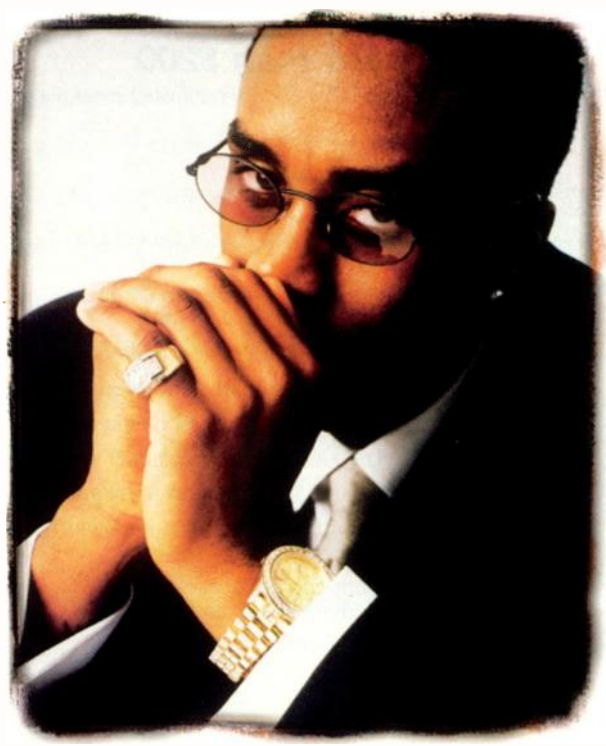
Farnsworth says, "Corporations invade our lives with product but forbid us to use it in our art or in any way they don't want. This just doesn't make sense." Still smarting from their own battles with corporate culture, Hosler says that he and Negativland, whose Seeland label is distributing the album, find the *Deconstructing Beck* situation somewhat amusing but are frustrated that an artist whose career was built from sampled works refuses to comment publicly about the CD.

"We've heard that Beck himself doesn't like it, but he doesn't want to say anything publicly about it," Hosler reveals. "I say there's a bigger issue involved. He could publicly say that he hates it or finds it boring. But his management just says that they don't want to be involved. I'm fairly certain that Beck doesn't clear every single sample on his own records, only the identifiable ones. We need to see someone at Beck's level take a stand and say that there is a distinction between bootlegging and transformative reuse. *Deconstructing Beck* is an interesting example because it's so mutilated and messed up, you're not going to think you're hearing a remix of a Beck song, not for a second."

CAVEAT SAMPLER

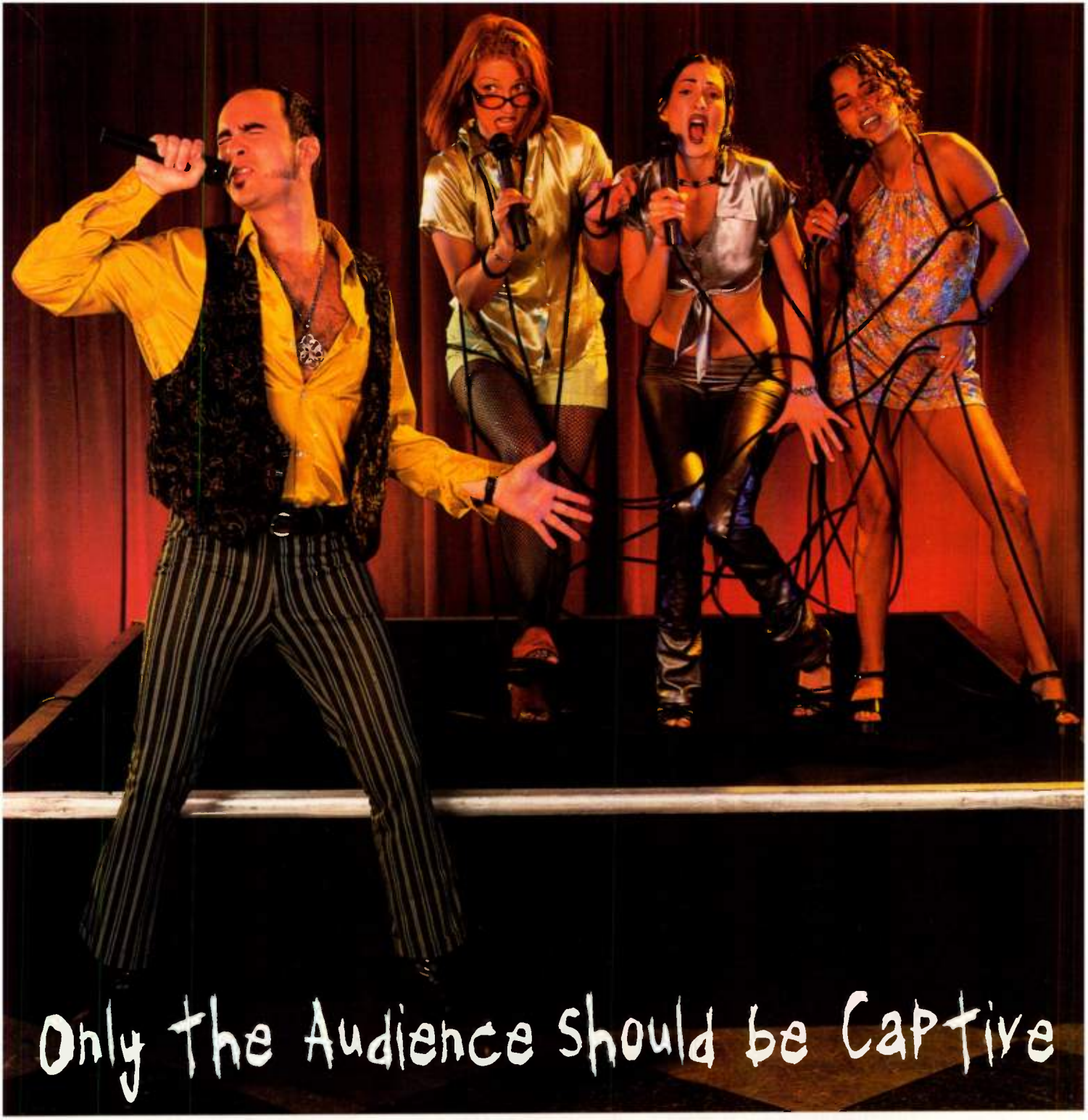
The Recording Industry Association of America (RIAA) recently stepped up their campaign against illegal sampling. The organization has introduced its *Anti-Piracy Good Business Practices for CD Mastering and Manufacturing Plants* guidelines, which states that replication plants may be held liable for pressing discs that contain illegal samples. While acknowledging that many artists feel it is a strong-arm tactic on the part of the recording industry, the RIAA maintains that the guidelines are a necessary step in antipiracy efforts.

Clearly, there are many sides to this issue. And although we may one day see the laws amended to accommodate "audio quotation," as it stands now, unauthorized sampling is still considered infringement. In the closing words of the CMRRA's David Basskin, "If you take it, you've got to pay for it."



Puff Daddy's work is synonymous with sampling. Rather than just sampling a simple beat or riff, he takes most of the melody of a song, then records his own lyrics over it, as he did with "I'll Be Missing You," which used the melody from the Police's "Every Breath You Take."

Paul Myers *is a songwriter, musician, and producer who frequently gets behind the word processor to ramble on about music and related topics. He lives in San Francisco with his wife, his cat, and his record collection.*



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Sequencing SECRETS

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BY SCOTT R. GARRIGUS

Today's digital audio sequencers are packed with so many features that nearly any feat you can imagine is possible. But just because you can dream up a great concept doesn't mean you have the technical chops to pull it off. In order to help you exploit your digital audio sequencer to the fullest, we've put together a number of intermediate to advanced tips and tricks that we hope will provide new possibilities for your work and help you reach your musical goals. In the pages that follow, you'll find numerous suggestions from the major manufacturers of both Mac and Windows sequencing programs, including ideas from their in-house gurus and from professionals working in the field.

Master CLASS

3-D ILLUSTRATION BY MICHAEL MORROW

Before we begin the journey, we'll present some generic tips that should apply to nearly any of the modern sequencing programs on the market. Because this is a master class, we assume that you already know the ins and outs of MIDI and digital audio. If not, read along and discover what you can look forward to as you master your software. (And of course, don't forget to look over back issues of EM!) Ready? Set? Let's go...

Effects without Effects

Many of today's sequencers include built-in MIDI and audio effects, but these features aren't the only way to add excitement to your tracks. With liberal use of the copy, paste, and pitch-shift commands, you can easily simulate a number of different effects, including delay, flange, and chorus. This can free up some of your computer's precious processing power for handling more complex tasks, such as compression and reverb.

Digital delay. This technique actually covers a number of different effects and can be applied to both MIDI and audio tracks. First, copy the MIDI or audio clip that you want to process and paste it to a new track. Then, slide the copied clip forward in time to create a delay. Depending on how much of a time offset you use, different delay-related effects can be achieved. Slide the clip between five and ten milliseconds, and you get a thickening effect. Slide the clip between ten and twenty milliseconds, and the thickening turns into doubling. Using anything above that amount will begin to create discrete echoes.

If you want to create a delay that is synchronized in time with your music, just move the copied clip ahead by snapping it to a note value, such as a sixteenth. To make things more realistic, make a few clip copies and slide each one a sixteenth note apart. Then lower the volume of each successive echo so that the effect fades over time. As an example, for three echoes, lower the volume of the first to 75 percent, the second to 50 percent, and the third to 25 percent of the original. You can even get a little crazy by sliding each copy to a different, uneven time offset and giving it a different pan position, as well. It would be hard to get that kind of precision using a real-time effects plug-in.

Flange. This effect can be applied to both MIDI and digital audio tracks but will require a slightly different procedure for each. As with delay, simply copy your audio or MIDI clip to a new track. Then move the copy forward in time from one to five milliseconds to produce a static flange effect. Though this creates an interesting sound, it doesn't have that authentic analog sweep that we all love and cherish. To achieve that, you'll need to add a little pitch shift.

For audio, instead of moving the clip copy, just shift its pitch anywhere from –12 to +12 cents. This will make the copy longer or shorter by a very small amount, causing it to drift out of sync with the original. Of course, depending on how long the clip is, there will eventually be so much of a delay that the flanging will turn into doubling. To get around that, just shift the pitch of small segments of the clip, alternating the direction of the shift with each segment. For example, shift the first two seconds of the clip by –9 cents, the next two seconds by +9 cents, and so on. This is also how you can control the rise and fall of the sweeping characteristic.

For MIDI, the procedure is similar, except that you need to apply Mod Wheel or Pitch Bend controller changes to the clip copy. Just draw in some data over the duration of the clip that varies by rising above and falling below the zero axis (see **Fig. 1**). It's best to stay within approximately 7 percent of your total range, or +600 to –600 (assuming that you're using a 14-bit controller, which has ±8,192 possible values), because any changes larger than that start sounding like detuning rather than flanging. To apply precise changes, many sequencers allow you to insert a series of controller messages that change evenly from a start value to an end value over a set amount of time. This will give you more precise-sounding effects, but you may prefer the more random fluctuations you can get from just drawing the controller changes by hand. Either way, the effect is not quite as authentic as with audio. Many times, it sounds more like chorusing than flanging.

Chorus. Audio chorusing is created in almost the same way as flanging except you need to stretch the pitch, rather than shift it. This means that you need to change the pitch of the audio clip without changing its length. Most programs have this capability. To apply chorusing to an audio clip, simply copy the clip to a new track and stretch its pitch by plus or minus nine to twelve cents. The more you stretch it, the deeper the effect. You can also use this technique to create a pseudostereo effect from mono parts by simply panning the original clip and the clip copy to different locations. The more pan, the wider the stereo image.

If you really want to go wild, you can create something we like to call the Super-Combo Chorus Effect. This time, instead of creating just one clip copy, make three, four, or more duplicates.

FIG. 1: Flanging and chorusing effects can be simulated using Pitch Bend data applied to MIDI notes. Varying the amount of pitch change only slightly will often produce the best result.

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Stretch the pitch of each copy using slightly different values from the rest. Now slide each copy in time using the delay techniques described earlier, with slightly different values. For the final touch, evenly pan each copy throughout the stereo field. You can also give each copy a different volume level for more subtlety. When you're finished, hit Play to experience one very cool chorus simulation.

Processing the Effects

When destructively applying plug-in effects to digital audio, many sequencers let you replace the original material or leave the original material intact and create a new track (or set of tracks, in the case of stereo) for the new, processed material. If you adjust the levels of the effect module so that the dry mix level is 0 percent and the wet mix level is 100 percent, your new tracks will contain the effect output minus the original signal. Why would you want to do this? Because it gives you some very interesting possibilities.

For instance, you can control your effects track in a variety of ways. Using Volume controllers, you can bring the effects tracks in and out to add an artistic flair to the mix. You can also use panning to move the processed signal around the stereo field while the unaltered, original signal remains stationary or moves around independently. Even more possibilities arise when you consider that you can add effects to the processed tracks. How about applying different EQ to the original and processed tracks? If your processed tracks contain reverb, how about applying another reverb effect on top of that for some very far-out spatial displacement? If you happen to have multiple sound cards or multiple outputs, you can assign your effects tracks to their own outputs. This way, you can even use your outboard modules to add effects to the already processed tracks. Experimentation is the key here.

Next we'll present a number of techniques that have come to us from the in-house specialists at the major sequencer manufacturers. Though these ideas are organized by specific product, you'll find that many of the tips can be applied to any of the major programs. Keep in mind that these techniques are intended for use with the latest versions of the programs listed, though a few of them will apply to older versions of the software, as well.

OPCODE STUDIO VISION 4.0

Solving Timing Delays

Though most current electronic instruments deliver reasonably accurate timing performance, some older units, especially analog synthesizers, suffer from triggering delays that can spoil an otherwise rhythmically tight mix. Luckily, Opcode's *Studio Vision* 4.0 provides an easy way to compensate for this troubling problem.

Studio Vision 4.0's Play Shift feature can move a recorded track any number of clock ticks forward or backward in time. This feature works well for setting offsets by ear, but if the next song you work on is at another tempo, the ticks will occur at a different rate, thus throwing off the accuracy of your previous offset.

To set perfect track offsets every time, regardless of tempo, you'll need to figure out, in SMPTE time, the amount you need to advance your sluggish tracks. This can best be determined by setting up a sequence in *Studio Vision* to trigger a chord or note on the unit in question at the start of a bar, while recording its audio output onto a free track. Open up the audio track you've just recorded and move the pointer over the start of the actual waveform. When you click this location with the mouse, *Studio Vision*'s counter will display the difference in time between the transmitted MIDI note and the resulting audio signal in SMPTE frames. For instance, the MIDI-controlled piano in our studio has a delay of around 500 milliseconds. At 30 frames per second of SMPTE, that equals 15 frames.

Next, set *Studio Vision* to the tempo at which you'll be working and type your offset time in SMPTE form into the counter in the Control Bar. *Studio Vision* will automatically display the equivalent offset in bars and beats above the SMPTE readout. Take the right-most digit in the bars and beats display and enter it as a negative number in the offset spot for the track you wish to advance. Voilà! Instant tempo-independent track offsets! —*Ken Freeman, recording engineer for jazz pianist Bob James.*

Assignable Key Commands

Studio Vision 4.0's assignable Key Commands provide an added level of control and convenience to this already full-featured program. On my particular setup, I have the F6 key assigned to bring up the Audio Crossfades window, F7 calls up the Pitch-Shifting feature, and F8 brings up the Adjust Audio Tempo option. Using these assigned keys allows me to select any audio track in the Main Track overview and, with the push of a single key, apply the DSP processing of my choosing. Because almost any of the editing functions available in the program can be assigned to a key, the sky's the limit as to what can be done with this feature.—*Sascha Konietzko, member of the band KMFDM.*

HARRISON FUNK

Sascha Konietzko

For manufacturer information, please see Contact Sheet, p.163.

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And of course, because there are often several ways to reach the same goal, you may have devised your own approaches to accomplishing these same tasks. Hopefully, you'll find these tips to be productivity boosters and techniques that you might not otherwise have thought about.

Cakewalk Pro Audio 7.0

From Cakewalk to CD

You've just created a masterpiece in Cakewalk *Pro Audio*, full of tracks—some that send MIDI out to your cool samplers and sound modules, some that play the SoundFont banks on your Sound Blaster card, and even some audio tracks complete with Volume, Pan, and real-time effects settings. All your hard work has paid off. There's one problem: it's going to be tough carrying around your whole setup so everyone can hear your masterful WRK file. So why not just record it to a cassette or, even better, a CD! Fact is, your CD recorder will only accept WAV files. That's no problem with *Pro Audio;* with a few simple steps, you can transform your WRK file into a WAV file that will be ready for your CD-R software.

Once your file sounds just the way you want it, you'll need to create audio tracks from your MIDI tracks. The first step is to record any MIDI tracks that use your internal synth sounds (FM, wavetable, or SoundFonts) with the sound card they are playing from. To do this, arm two open tracks for recording audio and assign the Source columns to your sound card's Left and Right inputs. If you have more than one sound card installed, make sure you select Left and Right for the sound card that is playing your internal MIDI sounds. Also, make sure that you don't have any tracks assigned to a MIDI Source in *Pro Audio*, or you could overwrite them by accident.

Next, open your sound card's mixer device. Make sure you are adjusting the Recording Control settings and not Playback controls, because you want to adjust the recording options, not the playback options (see **Fig. 2**). Drop the volumes of all faders on the mixer except the MIDI fader (or enable only the MIDI fader) to ensure that you only record the outputs of the internal synth. Hit Record in *Pro Audio*, and all the MIDI tracks assigned to the internal synth will be recorded as a stereo pair of audio tracks. Your new audio tracks will automatically be panned hard left (0) and hard right (127). To avoid confusion, you should mute the MIDI tracks that you just recorded, because you'll only want to listen to them as audio tracks from here on.

Next, record the MIDI tracks that use your external sound modules, if any. Connect the audio outputs of your sound modules to the line inputs of your sound card. If you have several modules, you may already be using an external mixer. If this is the case, connect the stereo output of your mixer to the line inputs of your card. In this case, all your modules should already be mixed the way you like. If they're not, then you can record your modules one at a time, creating stereo tracks in *Pro Audio* for each one. (Keep in mind that you may end up with a large number of audio tracks this way, which could bog down your system.)

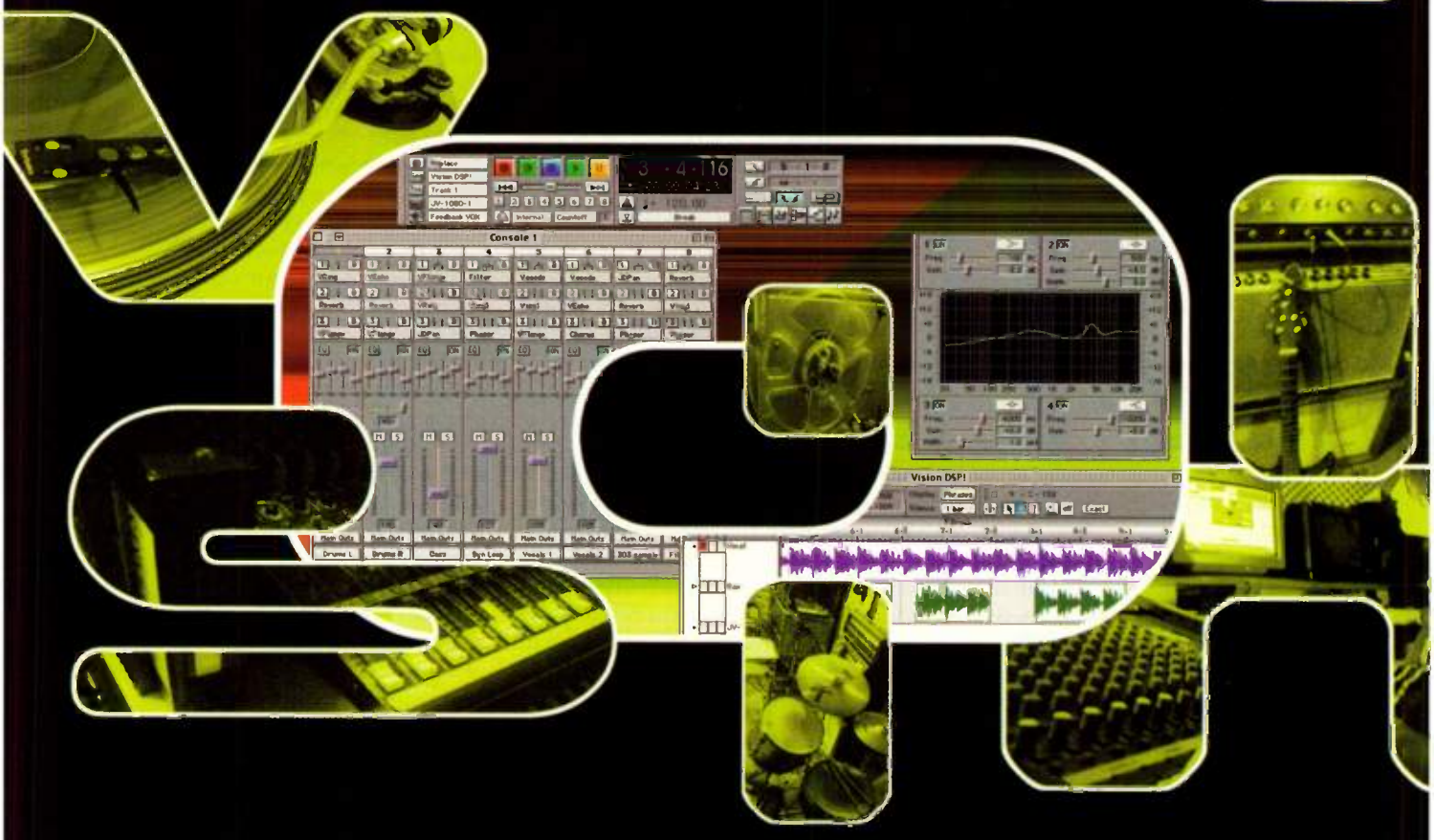
Again, enable two open tracks for recording audio, just as you did to record internal synth sounds. But this time, select only the Line In fader of your mixer device's recording controls. This fader will adjust the input levels in *Pro Audio*. To make sure you're recording at a good level, keep an eye on *Pro Audio*'s Audio Meters, which are found in the Console view. Then hit Record in *Pro Audio*, and all the MIDI tracks assigned to external sound modules will be recorded as a stereo pair of audio tracks. Again, you'll want to mute the MIDI tracks you just recorded. Now you should have two audio tracks representing your internal synth sounds, two audio tracks representing your external sound modules, and any additional audio tracks that you might have created previously.

Keep in mind that you could have recorded your internal synth sounds and your external modules at the same time by selecting both the MIDI and Line In faders on your sound card's mixer. However, by doing that you would be limiting yourself to one pair of stereo tracks, and you would not be able to adjust the relative volumes of the two sound sources. This could be your best bet if you are running low on audio tracks or are hitting the limits of your CPU processing.

Now that all of your tracks are audio, you can tweak them further

FIG. 2: To record MIDI tracks directly into Cakewalk *Pro Audio*, you'll need to enable the MIDI section on your sound card's Record mixer.

vision dsp



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with Cakewalk *Audio Effects* or other DirectX plug-ins loaded on your system. Then, you could create a WAV file using the Export Audio command, but this command does not recognize controller events (Volume, Pan, etc.) or any real-time effects that you may have applied. Therefore, you'll want to use Cakewalk's Mixdown Audio command, found in both Cakewalk *Professional* and Cakewalk *Pro Audio* versions 6.01 and higher.

Using Realtime/Mixdown Audio, you can create a stereo pair of tracks that will preserve all the dynamic changes in your tracks and will apply any effects placed in *Pro Audio*'s Effects Loops or Track Inserts. First, rewind the song to the beginning; then, play your file to make sure it sounds just the way you like. Go to the Mixdown Audio command box, and pick a destination track that is empty and also has an empty track right below it. Click Start; then click Stop when you are finished. You should now see a new pair of audio tracks that represents all of the tracks contained in your file. These are the only tracks you'll need to create your final WAV file. (Check out the *Pro Audio User's Guide* for a more in-depth description of the Mixdown Audio feature, if needed.)

The last step is to create the WAV file for your CD software. Highlight the stereo tracks you just created in the Mixdown Audio box, and then go to Tools/Export Audio. Choose the directory to save your file, name it, and click Save. You can also create a RealAudio file of your mix for Internet applications in this dialog box by selecting RealAudio (RA) under File Type.

One thing to keep in mind when creating your WAV file using the Export Audio to Wave command is that *Pro Audio*, like some other programs, places at the start of the file summary information that can't be recognized correctly by some CD-writing programs. Using a program like *StripWav*, which you can find via the links at www.cakewalk.com, you can easily remove this information. Then open your CD-R software, load the WAV file you just created, follow the instructions, and burn!

Steinberg Cubase VST

Spicing Up Your Drum Loops with Cubase VST and ReCycle

Triggering sampled grooves or drum loops in a sampler can be a great way to get good-sounding drum tracks, especially if you find the right sample CD-ROM. Unfortunately, there are many drawbacks to using loops this way. You might find yourself with a great groove but the wrong drum sound, or perhaps great drum sounds but a terrible groove. Tempo is another problem. Who wants to time-correct a sample every time you need a new tempo? Talk about disrupting the creative flow! Using Steinberg *ReCycle* in conjunction with *Cubase VST* can change all that, and what's more, you don't even need a sampler to use this technique. Here's how to create great drum loops and use them right in your *VST* audio tracks.

First, slice up the loop of your choice in *ReCycle*. (Be sure to tell the program the number of bars you want so it can automatically calculate the tempo for you.) Instead of sending the sound to a sampler, save it as a REX file using *ReCycle*'s Export command. Next, load *Cubase VST*, select an audio track, and choose the "any" channel. Now, when you import the REX file from the File menu, a dialog box will pop up and ask you how many channels you would like to use for dividing or splitting your loop. Pick "3" for now, and hit Do It. Next, set the left and right locators around the part, make sure Cycle is on, and hit Play to listen to the loop.

While the loop is still playing, double click on Tempo in the Transport bar and enter different values with the numeric keypad. (Be sure to press Enter on the numeric keypad each time.) Try different values, such as 120, 80, 130, or 135, and you'll notice that the loop speeds up and slows down without a glitch. Why does it do this? Because instead of the continuous stream of audio data that you'd normally find in a sampler or audio track, *ReCycle* cuts the audio into numerous segments that can

FIG. 3: Audio segments that have been created in Steinberg's *ReCycle* program can be imported into *Cubase* (shown here) for additional processing.

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be played back at a different rate for each tempo (see **Fig. 3**).

Because the loop is split among three channels, you can use your *VST* mixer and set separate pan levels for each of the channels, or you can solo, mute, EQ, or add effects to any of the channels to really spice up that old loop. For example, try sending channel 1 through a reverb and channel 2 to a delay. Then sweep the EQ on the third channel as it plays back, and record the sweeps by clicking Write in the *VST* mixer. For even more variety, you can apply different types of groove quantization or feels to the loop. This is possible because each piece of audio has a Q-Point (in case you didn't notice!), which allows you to snap the segments to the closest 16th or 32nd note on the grid.

Keep in mind that, if you're using a lot of channels and effects, you can go to the Master Section at any time, click on Export audio, and mix everything down to a new file that includes EQ, effects, and automation. Then simply bring this mix back into a track. This will free up some tracks, so you'll have more available.

Even if you don't own *ReCycle*, you can get many of these same effects using *Cubase*'s M-Points (Match Points). Check the manual for information about this option, and remember, no groove is so old that you can't give it new life!

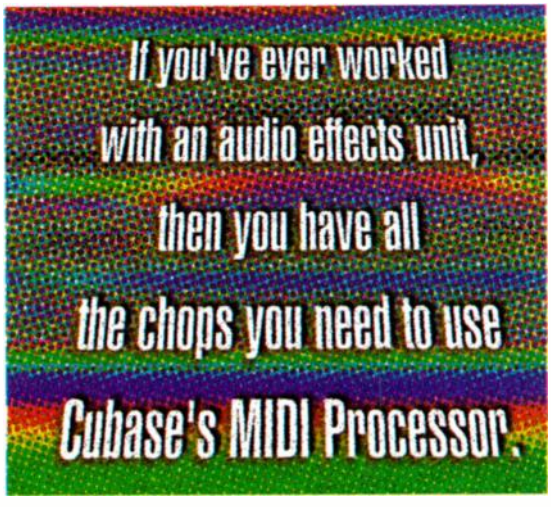
Maxing MIDI with Cubase's Processing Options

In the area of MIDI processing, *Cubase* is hard to beat. There are no fewer than three separate areas of the program that allow you to alter MIDI data as it enters the program or once you've got tracks recorded. Here's a quick summary of the three features.

Interactive Phrase Synthesizer. The Interactive Phrase Synthesizer (IPS), found in the Options menu, offers features you won't find in most other sequencing programs. In fact, it combines dozens of MIDI processing options into a single screen. Open the IPS and have a look around: you'll see a number of drop-down menus and dialog windows, which can be used to alter MIDI data in dramatic ways. Need a powerful arpeggiator? The IPS has the power of multiple arpeggiators that you can run in series or in parallel. Want to generate a truckload of variations on a melody you've come up with? As the saying goes, "It's in there!"

The IPS works by using the parameters you specify to alter the data you've loaded into it. These parameters allow you to change the pitch, rhythm, and dynamics of your phrase, among other options, using static adjustments and time-varying controls. It's easy to apply randomness to a parameter—you could easily have every note play on a different MIDI channel or use a different patch on your synth. And what's more, you can even set up two different sets of parameters and change between them with the click of your mouse! This is one "option" that will keep you busy for a long time—don't say we didn't warn you!

The MIDI Processor. If you've ever worked with an audio effects unit, then you have all the chops you need to use *Cubase*'s MIDI Processor. This feature is accessed from the PC's Module menu (or Options menu on the Mac) and provides six faders that control how your data is altered. The basic concept is that any note that the Processor receives is transformed according to the values you specify. The Processor settings, which include Repeat, Echo, and Quantize, allow you, for example, to turn a single note into an endless stream of echoing, decaying notes. It all works in real time, of course, and as you might have guessed, you can record everything the Processor cooks up right into a *Cubase* track.



The Input Transformer. If you think the two features above are all *Cubase* has to offer, you've got another thing coming. The Input Transformer is yet another way to modify your MIDI data. This tool can be thought of as a "transfer function" in that it takes input from any source and transforms that data to something else entirely. Using the Input Transformer, you can switch all incoming notes, or only those on a certain channel, to Program Changes, for example. Or you could trigger sounds on a tone module using the Mod Wheel on your controller. You'll find the Transformer awaiting your every command in the Options menu, and once you've enabled MIDI Thru and initialized the function, you're on your way to some very interesting results!

Emagic Logic Audio

Key Commands

One of the most powerful, user-friendly features in *Logic Audio* is its user-definable Key Commands. The traditional "mouse and menu approach" can accomplish almost every task in *Logic Audio*, but true speed and depth-of-use are attained by learning or defining Key Commands for your most-used edit functions. Also, some edit functions are only available by assigning a Key Command.

Logic Audio users can use the factory preset assignments or customize the presets, which allows you, for example, to emulate other programs' Key Commands. If you've never looked through the Command list, open the Key Command window from the main Windows menu and scroll the list to review the various presets (see **Fig. 4**). Press any existing Key Command on your computer keyboard and you'll find the definition of that Key Command. You can also find a Key Command by copying down any menu function, typing it into the Find box, and then clicking the Find button to see if a factory preset exists.

Here are some of the more useful Key Commands and descriptions of what they do.

Capture Last Take as Recording. *Logic Audio* is always in "MIDI record" mode, even during regular

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playback. If you're playing along with your arrangement and realize that you just played *the* magic take, use this Key Command before you hit Stop.

Set Rounded Locators and Cycle & Play. This command lets you define the cycle points from any selected arrange sequence, audio region, or MIDI events, and then initiate play with just one command.

Copy Value to All Following Events. While in the Event List, this Key Command allows you to select the current note Velocity or note length and copy this value to all the following events in that sequence.

Go To.... This is a set of Key Commands that are very useful for quick, "one step" navigation.

Roll Your Own

Creating your own Commands is easy and involves only a few steps. First, open the Key Command window and select (or do a Find for) the Key Command Play. (*Logic Audio*'s default for Play is the "0" key on your numeric keypad.) Now, turn on the Learn button on the left side of the window, and choose any other Key Command on your computer keyboard (the Spacebar, for example). If this command is already used (the Spacebar is *Logic Audio*'s default for Record Toggle), *Logic Audio* will prompt you and you can use the Delete key to clear your last choice.

Next, press the Spacebar again, and *Logic Audio* will scroll to this Key Command. Delete the default preset and

EMAGIC LOGIC AUDIO

Getting the Best Vocal Take

Recently I had the pleasure of working with the Italian pop singer Marina Rei. We had just completed the Italian version of the song "Heal Me with Your Love," and Virgin Records decided they would like an English version of the song, as well. Because Marina is such a great vocalist, recording the Italian version had been no problem. Marina always hits her note no matter what key I ask her to sing in. The problem now was that we were asked to record in a language Marina speaks slightly, but with a definite accent.

What I decided to do was turn on the Cycle Record function in the *Logic Audio* Transport window. My song settings were set to Create New Track in Cycle mode. This means a new track is created each time the loop passes. I set the start and end points of the Cycle function to cover an 8-bar range. I began recording and had Marina sing the same line seven or eight times. The entire song would be recorded using this method so I could get her best takes from each section. By recording in this manner, not only could I take her best vocal parts, but if I needed to, I could replace individual words from different takes, combine them, and create a single, perfect vocal part. (Here's a hint: Using *Logic Audio*'s Key Command to Cut at SPL, or "song position line," you can cut out unwanted sections in real time without stopping the sequencer.)

The reason I record using this method is that even the best singers change their expression and pronunciation when forced to sing straight through. And for me, the point of producing is to create polished and, if possible, perfect products, regardless of the obstacles. *—Ashley Ingram, platinum record producer and songwriter whose credits include Des'ree, Emel, Marina Rei, and Jesse Powell.*

Tuning Samples via Time and Pitch Machine

In preparation for the Mary J. Blige World Tour, I was told the background singers were going to be singing in a different key from what was on the original recording. The problem was that certain parts (which were samples) now had to be converted from one key to another. The samples were created using the Akai S3000, so we tried to use the sampler to alter the pitch. But because tuning on the S3000 is done solely by numeric value, trying to tune with this device became painful. We soon came up with a work-around, which was to import the samples into *Logic Audio*.

In *Logic Audio*'s Arrange window, I selected a track and chose my Roland JV-1080 as my MIDI device. Then I went to *Logic*'s Time and Pitch Machine feature and began altering the pitch of my samples. If I needed to alter the pitch of the sample to match the note C, for instance, I would simply tap on my controller to get that reference pitch from the JV-1080. Using my controller, I could audition my pitch changes in real time until the sample matched the correct note. Perfect match every time!

After processing the file, we sent the samples into *MESA*, Akai's sample import/export program. From *MESA*, we could import the samples into the S3000, and the job was done. *Logic Audio* and *Sound Diver* together took literally hours off our production and programming time. No exaggeration, they were life savers! And because everything worked so well with Mary's rig, I used the same process with the New Edition and Luther Vandross tour, and I plan on using this combination with future projects. *—Jeff "Thunder" Mays, tour programmer and technician whose credits include Mary J. Blige, TLC, Luther Vandross, Jodeci, and New Edition.*

JEFFREY MAYER

Ashley Ingram (left) with Danny Stick, president of BMG Music, Inc.

Roger Sanchez – world renowned remixer/producer and Sony recording artist.

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go back to step one. That's it! By the way, always remember to turn off the Learn button when you are finished.

Importing and Exporting Audio Regions

Though many aspects of MIDI and digital audio have now become standardized, one area that has not yet reached this level is audio regions. There is simply no way to easily move audio regions or playlists from one application to another. Fortunately, *Logic Audio Platinum* 3.5 on the Mac is now able to import and export time-stamp information via SDII regions. This feature makes the task of moving compositions between applications, such as *Logic Audio* and Digidesign's *Pro Tools*, much simpler, because it allows you to easily restore the time position of a region once it's imported.

The following procedure is for moving files between *Logic Audio* and *Pro Tools*, though it will also work with any application that can read and write time-stamped SDII regions.

Start *Logic Audio* by opening the Audio window and selecting the audio files that you wish to export. (If you are moving all the files and regions of a song, simply choose Select All.) Next, select Audio Window/File/Export SDII Regions, and *Logic* will time stamp the regions according to their position in the Arrange window. Keep in mind that, if the same region is used multiple times in the song, then *Logic* will use the position of the first instance of the region. For this reason, you will generally want to make all regions in your song "individual regions" before starting this procedure. Use *Logic*'s Arrange/Functions/Convert Regions to Individual Regions menu option to perform this task.

Next, load *Pro Tools* and add the audio regions using the Import Audio option found in the File or Region List menus. Enable Spot mode and drag the regions onto the desired tracks in the Edit Window. When the Time Stamp window appears, load the Original Time Stamp value into the Start Point window and hit OK. The region will now be located at the same position as it was in *Logic Audio*.

You can accomplish the task in reverse to move data from *Pro Tools* to *Logic Audio*. Here's how. In *Pro Tools*, select all the Regions in the Region Bin that you wish to export; then select Export Region Definitions from the Region List menu. Back in *Logic Audio*, choose Audio Window/File/Add Audio File to add the files that were exported from *Pro Tools*. Now select the files you've imported in the Audio window and select File/Import SDII Regions. The regions that now contain time-stamped info will show up with small clock icons to the left of their names.

Drag the Regions into the Arrange window onto the desired tracks, and with the Regions selected, choose the Functions/Regions to Original Record Position option. If the region has both an Original and User Time Stamp (see your *Pro Tools* documentation), then *Logic* will ask which time stamp you wish to use. Other than that, mission accomplished!

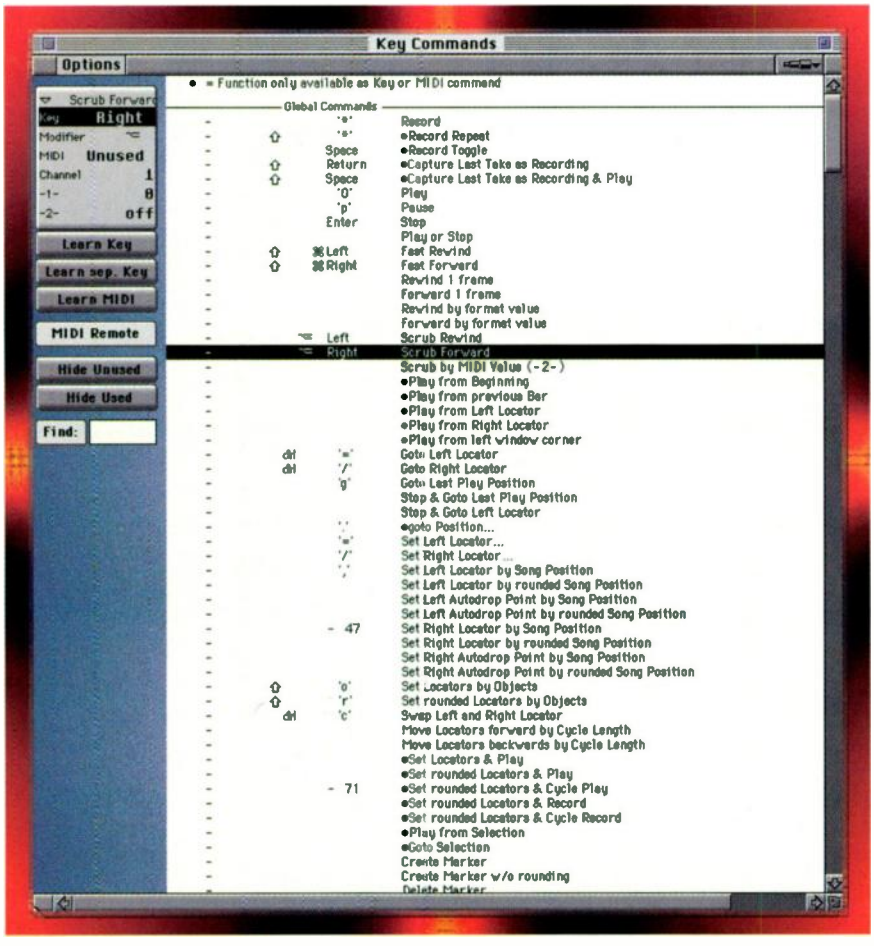

FIG. 4: The preset Key Commands in Emagic's *Logic Audio* are the key to many useful shortcuts. You can also create your own to customize the program.

Mark of the Unicorn Performer and Digital Performer

Clippings

One recent addition to the drag-and-drop functionality of the Mac OS (7.1 and above) is clippings. Clippings allow you to drag and store selections of data in one quick, easy step. For example, open SimpleText, select a portion of text, and then drag the selection off the window into the Finder. You have just created a clipping! When you want to reuse the text, just drag and drop the clipping back into SimpleText (or any text editor that supports clippings).

Performer and *Digital Performer* are currently the only sequencers that support this handy tool. You can store just about anything as clippings: MIDI

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data, audio data, musical phrases that consist of several tracks worth of MIDI and/or audio, and more. You can even store an entire sequence as a clipping. To create a clipping, you simply select some data (MIDI or audio) and then choose Copy to Clippings Window from the Edit menu. In some cases, you can simply drag objects, such as folders in the Finder or a sound bite in *Performer*'s Soundbites window, directly into a Clippings window.

Listed below are a number of ways clippings can be used to help you create music quickly. *Performer* and *Digital Performer* have built-in Clippings windows that are globally accessible to any *Performer* sequence or can be attached to a particular sequence (see **Fig. 5**). New Clippings windows are created by using the Clippings submenu under the Windows menu.

Phrase scrapbook. If you find yourself constantly re-creating "bread-and-butter" grooves, start moving them into a Clippings window instead. Drum grooves are ideal for this. Useful bass lines can be stored and then transposed for any harmonic context. The same goes for any MIDI and audio phrase libraries, such as the incredibly useful Twiddly Bits library. This is a huge time saver.

Drum loop library. Clippings windows are an ideal way to organize a drum loop library in *Digital Performer*. Be sure to assign a tempo to each drum loop before storing it as a clipping. (Select the loop and choose Set Soundbite Tempo from the Audio menu.) Then, when you drop it into a sequence, *Digital Performer*'s PureDSP time-stretching feature can match the tempo of the sequence in one easy step. (Just choose Adjust Soundbite to Sequence Tempo from the Audio menu.) Store all your tempo-mapped drum loops in a Clippings window and forget about tedious beat-matching when inspiration strikes.

Lost ideas. Sometimes you'll create a short, useful sequence that is not a complete composition in itself, but may get lost in the shuffle as a stored file. It could consist of MIDI tracks, audio tracks, or both. Drop it into the Clippings window with the rest of your phrases. At a later date, if you have a deadline and you're short on ideas, start pulling out your "lost ideas" clippings, and you'll have great-sounding music in no time.

Effects configurations. MIDI and audio are not the only types of data that can be brought into a Clippings window. Effects combinations can be clippings too. Once you have created the "perfect" vocal chain for a particular vocalist, Shift-click each effect and then drag the selection into the Clippings window or a folder on your Mac desktop. To reuse the combined effects settings, just drag them from the Clippings window (or Mac folder) onto the desired inserts in *Performer*'s Mixing Board window. Now, every time you work with this vocalist, your setup time will be greatly reduced.

This technique can be applied in other circumstances, as well. For example, using plug-ins like the Sonic Modulator and PreAmp-1, it's easy to create a clippings library of ready-to-use guitar tones. MIDI multi-effects can be created using this same technique.

Using Custom Consoles to Create MIDI Data

One of the joys of MIDI is creating new ways to input note data. Rather than just tapping on your keyboard, you can use *Performer*'s Custom Console in combination with your keyboard's controller sliders to create a new way to input notes.

Here's an interesting way to simulate string or harp flourishes using the Custom Console and a Pitch Bend or Mod Wheel controller from your keyboard. First, create a slider in the Custom Console and select the source as a Continuous Controller or the Pitch Bend controller. Now select the Target as Notes and allow the value to be 0 to 127. (You can narrow this number down to get more precise note values if you prefer.) When you select the slider you've just created, it should be activated by your controller. Move the controller up and down, and you will hear a chromatic scale that will play as fast as you move the slider.

You can add sliders to associate with volume, reverb depth, and sustain of a track. With this combination of elements you can create an exciting tremolo or flourish effect with your keyboard's Pitch Bend wheel. Want to drive your listeners insane? Create a Custom Console slider for notes and patch changes to generate a frantic wash of sounds. That will do the job!

Instant Percussion Sample Trimming

If you get tired of trimming drum samples for export into your sampler, you'll love this tip. Take a drum or percussion track and use the Dynamics/Gate plug-in. Set the Attack to 0.10 ms and adjust the Release and Threshold until you hear individual drum sounds with a reasonable decay. When this is accomplished, select the track and use the Bounce to Disk feature under the Audio menu. Be sure to check the option Add to Sequence. Once this operation is finished, select the newly created track and choose the Strip Silence option under the Audio menu. This will create individually trimmed, ready-to-go samples for export into your sampler. Now you can select all your new Soundbites and drag and drop them to your sampler via SCSI using *Digital Performer*'s Drag and Drop sample system.

To create some really strange samples, try this technique with a nonpercussion audio track. One interesting trick is to insert the Sonic Modulator plug-in before the Gate on a sustained vocal track. Set the Sonic Modulator to modulate the amplitude of the track with a square wave. Now the Gate will create samples of a fixed, identical length.

Live MIDI Effects

Performer and *Digital Performer* support an open MIDI effects architecture that operates in real time. Many people enjoy the nondestructive nature of MIDI effects in a production environment but overlook the real-time performance opportunities this feature also presents. For example, by using the Echo MIDI plug-in, it's possible to create instant

FIG. 5: MOTU's *Performer* and *Digital Performer* both support clippings, which are small segments of data that you can insert anywhere in a sequence.

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layers of MIDI percussion on any MIDI instrument, including MIDI guitar. Here's how to use this nifty feature.

In the Tracks window, record-enable a MIDI track. Select your MIDI input and output devices using the pull-down menus next to the Record Enable button. This will route live MIDI data from your controller to the destination MIDI device. Next, open the Mixing Board and you'll see five Inserts per MIDI track. Clicking on an insert will reveal a pop-up menu. Select Echo, and the MIDI Echo/MIDI Effect plug-in dialog box will pop up. Select a dotted eighth note so the plug-in will generate auto-syncopated rhythms. Depending on the percussion sound you choose, a little transposition will help add variety to the results you generate. For percussion with a definite pitch center, you might want to choose a harmonically congruent interval like a perfect fourth or perfect fifth. Nonpitched percussion can get by with smaller intervals, like a major second.

Additional MIDI effects can be inserted "in-line." Placing the Arpeggiator plug-in after the Echo, for example, will yield interesting results. You can also use the Bypass buttons, located on the top of each plug-in window, to instantly defeat any effect. By turning different effects on and off throughout your composition, you can create different-sounding sections, which may help give shape and structure to your music.

Opcode Studio Vision and Vision DSP

The Substitute Command

In both *Studio Vision* 4.0 and *Vision DSP*, the Substitute window under the Do menu provides creative possibilities that not many users are aware of. To access this feature, open the Select/Modify window and select Substitute from the window's pull-down menu (see **Fig. 6**).

Substitute exchanges selected data of any kind in *Studio Vision* with the current contents of the Clipboard. For example, if you copy the note sequence "A, B, C" to the Clipboard and substitute a note sequence of "E, G, G" for the original sequence, the notes will be replaced in one of two ways: you can tell *Vision* to replace each selected event with the entire contents of the Clipboard, or you can have it sequentially replace the selected events with independent events from the Clipboard. The Substitute command can also be used to perform some other unusual tricks. Here's one of our favorites.

Record a track with a rhythm tapped out on a MIDI device. Then copy a short audio file, such as a snare drum or vocal snippet, into the Clipboard by selecting it and pressing Command-C. Open up the Select/Modify window by pressing Command-G on the keyboard, and select Substitute from the pull-down menu in the window. Next, press Enter. Your MIDI track will now be replaced with the audio snippet on the clipboard playing the exact same rhythmic pattern. This is a great way to quickly create drum patterns composed of sampled sounds, even if you don't own a dedicated sampler.

RAM Disk Recording and Playback

If you're using *Studio Vision Pro* on an older system that lacks a fast, dedicated

STEINBERG CUBASE VST

Matching a Sequencer to Tape

One type of project that my company, Noisy Neighbors, is involved with is dance remixes for various artists. Typically, these artists bring us a master tape of their song and ask us to do a remix version for use in clubs or for inclusion as an alternate mix on a CD. The process involves playing their master and doing all sorts of overdubs (drums, loops, synths, etc.) in perfect synchronization. In many cases, I'll use only the vocals from the master and create a whole new music track for accompaniment. The songs we're remixing are often originally recorded by other producers using different sequencing systems, and I need an easy way to synchronize my sequencer, *Cubase Audio VST* (on the PC), to the master tape.

There are two ways that *Cubase* makes my job easier. Because we're often remixing dance tracks, we use a lot of drum and rhythmic loops. But things can get tricky when you're trying to match the tempo of a song that was recorded using a different sequencer than your own, or the same sequencer running on different hardware. For example, on the original computer, the song might have been recorded at 120 bpm, and on a different computer, you might have to set the tempo to 120.02 bpm for it to play in sync.

One way to deal with this is to build a tempo map so you can actually record clicks into the computer and have it figure out a tempo map, beat by beat, that matches what's on tape. In the above example, the map might start with a beat of 120.03 bpm, then 120.01 bpm, then 120.02 bpm. (There are always tiny fluctuations.) The problem with this approach is that, if you want to use 1- or 2-bar samples of drum or rhythmic loops in your remix, they're not going to sound like they flow into each other each time they're triggered. It's amazing what a difference 0.01 bpm can make when you're triggering long samples. A better way to approach this is to find a single average tempo that matches the master tape all the way from the top of the song to the end, and *Cubase* has a very interesting way of helping with that.

I find an average tempo by setting a SMPTE start point in *Cubase* that matches the SMPTE start time of the song on the master tape. I then have *Cubase* slave to SMPTE and generate a click so I can hear if it's in sync relative to the audio on the master. As *Cubase* plays, I adjust its tempo up and down with the Plus and Minus keys. Here's the coolest part: Every time I change the tempo, I'll hit the Spacebar. This causes *Cubase* to stop momentarily, recalculate its bar-position based on the new tempo, and then immediately start up synch-

hard drive for audio recording, consider using a RAM disk. This technique can be used with any audio system that does not require a dedicated external hard drive: Apple's Sound Manager, Digidesign's Audiomedia II and III, and Opcode's DATport, for example.

First, open the Memory control panel found in the Control Panels folder inside your System folder. Activate the RAM Disk by clicking the appropriate radio button, and then set its size in the space provided toward the right side of the window. Make sure you set it to a size that your system can realistically handle, while still leaving enough space in RAM to run whatever applications you may need open.

At this point, you'll need to restart your computer. After rebooting, the Mac OS will prompt you to format your new RAM Disk, which will appear on the desktop as a disk icon with an image of a RAM chip on it. Once you click OK and the disk is formatted, you can record and play back files from your new superfast RAM disk at transfer rates faster than those of any hard drive. This technique can come in handy if you're constantly plagued with "Hard drive too slow or fragmented" error messages or if you intend to play back audio tracks while working with a movie in *Studio Vision*'s QuickTime window.

In the latter case, try copying the movie to be played back to the RAM disk while keeping any audio tracks used in your production on a separate drive. Don't worry, you won't lose the contents of your RAM Disk if you crash or restart. The RAM disk *will* be erased if you power-down completely, though your system will prompt you first with a warning. To save all the files on your RAM disk before powering down, simply drag them onto your hard drive.

Calculating the Tempo of an Audio Segment

Working with audio segments and loops has become a mainstay of the production process for many contemporary musicians. It is often difficult, though, to determine the correct tempo of a loop. Thankfully, *Studio Vision* and *Vision DSP* make this a simple process through the Scale Time function found under the Do menu.

First, import the audio for which you wish to determine the tempo into an empty track and select it. Then choose the Scale Time function from the Do menu. At this point, a window listing the various parameters for this command should open onscreen. Simply enter the length of your selected audio in bars and beats into the field for duration and make sure that the radio button next to Insert Tempos to Maintain Timing is checked. Now, when you click on the Scale button, the tempo of your sequence will match that of your loop.

Recording without a Metronome

In a sequencing environment, it can often be difficult to take a musical passage recorded without a timing reference and lock it into bars and beats for easy editing. To overcome this obstacle, *Studio Vision* and *Vision DSP* provide the Reclock function.

To begin, connect a non-latch-type footswitch to the Sustain pedal input of your MIDI controller. Turn off *Vision*'s metronome and record your performance while tapping out time in quarter notes on the footswitch. When you've finished recording, select Reclock under the Do menu. The Reclock dialog window should now be open.

Select Same Click on Every Beat from the radio buttons. Further down in the window is the area where you determine what kind of MIDI message will

ing to the SMPTE time code again.

For example, if I'm playing at a tempo of 120.0 bpm (which is in sync at the beginning of the song), and it seems to be falling behind the master tape by the time we get to bar 50, I'll change the tempo to 120.1 bpm while still playing, and then hit the Spacebar. *Cubase* will momentarily stop, recalculate its position, and restart. Or, if *Cubase* seems to be ahead of the beat, I'll try the same thing with 120.05 bpm. It's an easy, fast way to zero in on a perfect tempo. I can then set my loops (in a sampler) to this tempo, and they'll all play perfectly and retrigger smoothly.

Triggering Drum Loops

Here's a method I use when I need to trigger lots of drum loops from my sampler. In *Cubase*, create an empty part that's one bar long, then double-click on it to open the Editor. Use the Pencil tool to insert a middle C that's also one bar long, select that note, and shorten its length by one MIDI tick. This ensures that your "trigger note," which you'll be using many times, will turn off immediately before the next one is sent.

If your sampled loop has a long release time (fade out), shorten it or shorten the trigger note further to avoid hearing overlap. Set the Velocity of your trigger note to 100 (to allow some headroom) and then close the Editor. Now, from the Arrange window, duplicate the 1-bar part throughout the song using *Cubase*'s Repeat command (Control-K) with the Ghost option selected. This creates copies of the part that will reflect any changes you make to the original, such as Velocity and delay.

If you plan to use more than one loop, copy the track to as many other tracks as you have loops, and use the Transpose command to make each track trigger a different note. In your sampler, put all your loops on the same MIDI channel, but limit their key ranges to the different notes you plan to use. (Also, be sure that you "tune" each loop to play in the correct key when it receives the note you're using as its trigger.) This way, you can audition each loop or combination of loops from your keyboard very quickly. Another benefit is that you've only used up one MIDI channel for up to 128 loops (or however many keys you have on your controller). You'll find this technique can make loop organization easy and save you lots of time.

—Rob Arbittier, composer-producer who has scored hundreds of TV and film trailers; his most recent trailers are for Godzilla *and* Les Miserables. *His recording credits include Stevie Wonder and Michael Jackson.*

be used as reference points to determine your new tempo map. Select the MIDI device and the type of message being used as a "click-point" (i.e., the Sustain pedal) in the window's pull-down menus. Finally, click on the Re-clock button, and *Vision*'s bar lines will now line up correctly with your recorded material.

Cakewalk Metro 4

The Rhythm Explorer

Metro 4.0 for the Macintosh provides some killer methods for building a song quickly. It offers extensive looping capabilities for assigning multiple loops inside tracks and allows you to build up a song with a drag-and-drop, pattern-based method.

But one of the coolest ways to beef up your song is to use one of *Metro*'s brainstorming features, like the Rhythm Explorer. The Rhythm Explorer is a great tool for playing around with notes in your sequence, trying variations, exploring ideas, and breaking through creative roadblocks. (*Photoshop* users can think of it as a "plug-in filter" for MIDI tracks.) These tips will get you up and running on the Rhythm Explorer in minutes.

Metro 4 has an enhanced Rhythm Explorer—it's now a floating window, so it's easier to experiment with different notes in different tracks, solo and mute tracks, and loop the playback. We'll use examples from *Metro* 4, although this tip applies to some earlier versions of *Metro* as well, where the Rhythm Explorer appears as a dialog box.

Start with any groove you like—step enter it, play it in, or load it up. Next is the fun part: you can select notes, audition variations on those notes, and blast the changes into your track. Although the Rhythm Explorer works well on harmonic passages (especially using its arpeggiators) or melodies, the Rhythm Explorer is also perfect for percussion tracks. Let's experiment with a drum track.

CAKEWALK PRO AUDIO

Fixing and Fattening Vocals

John Mahar, production coordinator of the Minnesota Vikings, called me with a dilemma this past July. The Vikings wanted a new version of their theme song, "Skol Vikings." John had all the tracks recorded, but the vocals sounded weak and were not in sync with the music. The song is played at every home game to get thousands of Vikings fans riled up after a big play. Big plays need a big song, and weak, out-of-time vocals just don't cut it.

"Don't worry, I can fix your problem fast with Cakewalk *Pro Audio*," I told John. "Just send me your tape."

The first thing I needed to do was fix the timing problems. I started by recording all the music tracks into *Pro Audio*. I then opened the Audio view and split the vocal track at the beginning of each phrase to create separate audio events. Once the vocal track was split into separate events, it was easy to drag the different phrases back and forth until they aligned with the beat of the music. When doing this, it's helpful to have the music tracks open in the Audio view, side-by-side with the vocal tracks. That way, its easy to visually align the start of vocal phrases to the beat.

The next step was to fatten up the vocals. For this I used a process called "layering." This technique works great for any tracks that need thickening, especially vocals and rhythm guitars.

First, I copied and pasted the vocal track until I had six tracks of the same vocal part. I then used *Pro Audio*'s Time/Pitch Stretch function to drop the pitch down 0.1 semitone on track 2 of the vocals, down 0.2 semitones on track 3, down 0.3 semitones on track 4, up 0.1 semitone on track 5, and up 0.2 semitones on track 6. By subtly changing the pitch of vocal tracks, you give the illusion of multiple voices singing the same part. Time/Pitch Stretch offers options for different audio types. For this application, I chose Vocal as the Source Material and set the Advanced Settings to High Accuracy and Formant Preserving Algorithm.

I now had a crowd singing, but it still sounded a little contrived. So to warm it up, I sent vocal tracks 2 through 5 to one of the aux buses in *Pro Audio*'s Console View. Here I placed FX Stereo Chorus with the Thick Group Vocals preset and FX Stereo Reverb with the Arena preset on the tracks. I then sent the 100 percent wet submix to a separate output on my sound card.

With the wet and dry vocals combined, I now had the sound of a full stadium of raging fans singing "Skol Vikings." The song was ready. I sent it off to the Vikings' main office in Winter Park.

Then came the surprise: Winter Park called to say the song sounded "too new" and asked if I could try and make it sound a little more like the old version. (So much for state-of-the-art, slick production!) To solve the problem, I decided to add the old vocal track from the original version (circa 1970). But first I had some things to deal with.

When I recorded the original track into *Pro Audio*, there were two issues. The original recording was noisy and at a slightly different tempo. No problem. From within *Pro Audio*, I ran Sonic Foundry's Cakewalk DirectX Noise Reduction plug-in on the track. Then I used *Pro Audio*'s Fit-to-Time command, and the old singers were singing right in time with the new ones.

There was still something missing, however. After listening to the original version a few times, I sensed that what was needed was more bass drum and tuba. I'm not much of a tuba player, so in came MIDI to the rescue. I was able to quickly add MIDI bass drum from a Korg NS5R and the tuba from a Yamaha MU90R. Now the song was complete. I sent it over to the Vikings and they loved it.

If the Vikings can perform as flawlessly out on the gridiron as *Pro Audio* did on this project, we'll see them at the next Super Bowl! —*Tom Pearson, audio director for the Minneapolis Metrodome.*

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The easiest way to work with percussion tracks in *Metro* is to view them in the Graphic Editor with the drum grid. All the notes in your track can appear with note names and will show up in the grid in different sizes—the louder the note, the bigger the dot (see **Fig. 7**).

Pick one or two bars of music and select a particular percussion note in those bars. For example, take several bars in a drum track and select the hi-hat notes. The easiest way to do this is to click the measure number to select the bar and then click the note name to select only those events. You could also drag over the measure numbers to select more than one bar and Shift-click the notes to select more than one note type.

Now open the Rhythm Explorer window by choosing it from the Window menu. The Rhythm Explorer offers numerous processing algorithms in a pop-up menu. These include two types of arpeggiators, functions to create random Velocity and groove patterns, options to add trill or flam effects, and different methods for randomly deleting notes. (There are seventeen in all.)

Pick several algorithms and decide what amount of randomness you want:

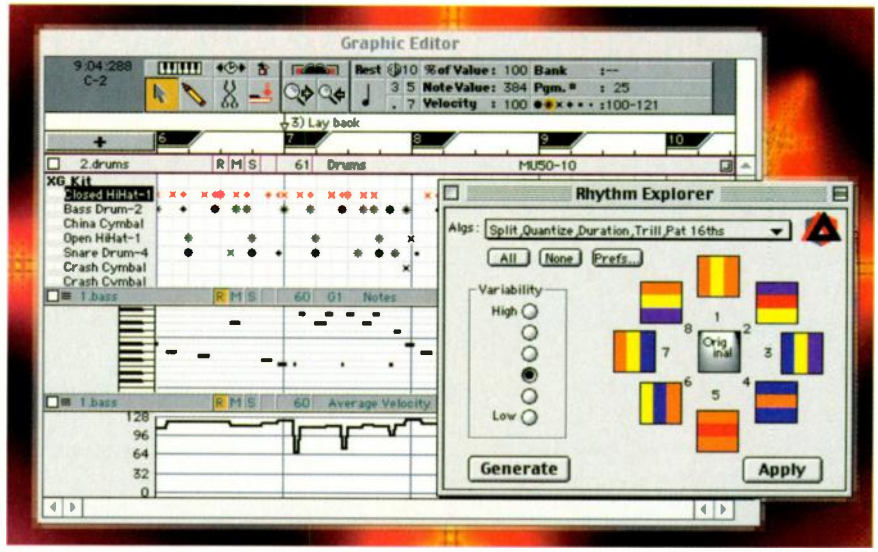

FIG. 7: The Rhythm Explorer in Cakewalk *Metro* 4 allows you to alter existing material in numerous ways. It is especially useful when used with percussion notes.

Low variability all the way up to High. Then, click Generate, and *Metro* will immediately concoct eight variations of your original notes, which are represented by colorful squares. Each color represents one of the algorithms in the pop-up menu, and if there's more than one color in a square, it means *Metro* used more than one algorithm in that variation.

Audition the variations one-by-one by clicking them—you can use *Metro*'s ability to solo and mute individual drum notes to hone in on the right groove while you audition. Once you find one you like, click Apply to replace the original notes with the new groove. If you don't like the variations, click Generate again and get eight new ones to preview. If you use *Metro*'s Cue Looping feature, which will loop selected notes for playback, you get a great sense of how the pattern will develop and groove over time as the measures repeat themselves.

Using the Rhythm Explorer and other *Metro* features, like Groove Quantizing and Human Feel, you can build a stunning groove in minutes, even from square, step-entered notes. It's a perfect way to put a spin on a Latin percussion track or build a pulsing techno beat, ready for dragging and dropping in patterns for fast song building.

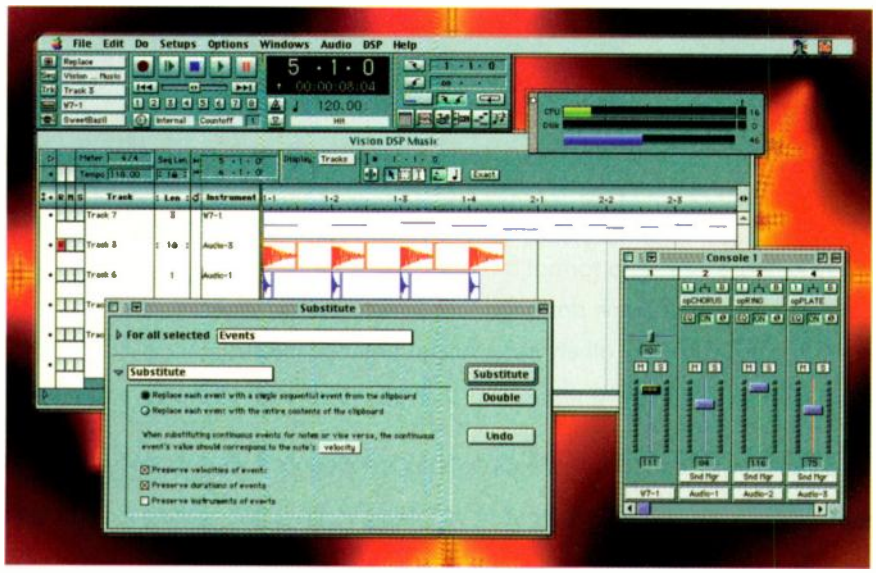

FIG. 6: Opcode *Vision DSP*'s Substitute window exchanges selected data of any kind with the contents of the Clipboard.

There You Have It

We've brought together these tips and tricks with the hope that they'll enhance your sequencing skills and give you new ideas about how to reach your goals. Though technology alone can't guarantee that you'll be able to make great music, you certainly don't want it to slow you down. The sooner you learn how to utilize the incredible tools that are available to every desktop musician, the sooner you'll be on your way to creating the audio of your dreams.

Scott R. Garrigus *is such a music technology fanatic, he has just about every issue of* **EM** *since June 1986. You can reach him on the Web at www.garrigus.com or via e-mail at scott@garrigus.com.* *Special thanks to all the manufacturers who helped us put this article together, and in particular to Costa Kotselas, Carl Jacobson, and Rich Thurow.*

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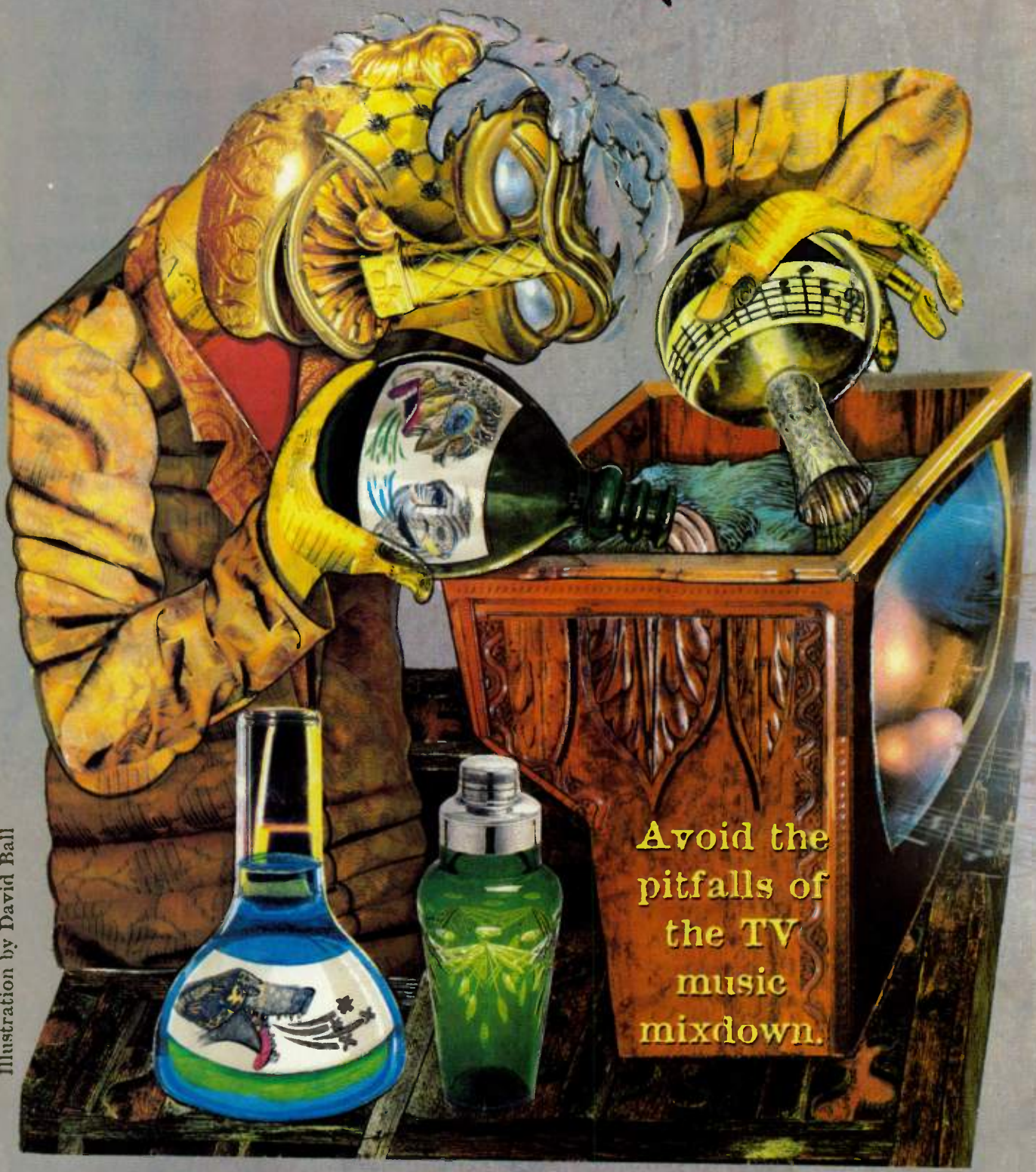


Illustration by David Ball

SMALL SCREEN

By Mike Levine

Over the past few years, there has been a dramatic increase in the number of television audio projects turned out from personal studios. In much the same way that music performers and producers have taken advantage of the personal studio phenomenon, so television composers, sound designers, and advertising executives have also been reaping the benefits of working in noncommercial facilities. Many smaller-studio owners with experience only in music production have been crossing the threshold into television audio. In short, this is a booming business.

Though the methods for recording instruments and vocal talent for TV don't really differ much from those employed in music production, the television mixing process has a distinct set of problems and constraints that must be factored into the equation. These include mono compatibility, a limited dynamic range, and the addition of a voice-over track. If you're going to mix a commercial or soundtrack, but most of your experience has been with music projects, it's important to understand some of the differences between traditional record-style mixes and those done for television broadcast.

One Mix or Two?

Unlike a music recording, a television project involves two distinct mixes. The first is a mix of the music from multitrack to 2-track, which, although quite similar to a record mix, has significant differences that need to be addressed during the mixdown.

The second mix is referred to as the "post mix" (or "film mix"), during which the music, effects, voice-over, and location sound get combined. Because it's the last chance to add any necessary EQ or compression before the material is broadcast, the post mix also serves as the television equivalent of mastering. This kind of mix would probably not be carried out

in a personal studio, as it's often performed during the "lay back" session (the combining of the audio with the video), which requires specialized video equipment.

Every element of the music mix must gel coherently with the various parts of the post mix, so even if you're only involved with the music portion of the project, you need to work with the final product in mind.

Listen Like Your Audience

Despite all of the good-sounding stereo televisions on the market, a large percentage of the TV audience still hears programs in mono through 3- or 4-inch speakers. According to Joe Arlotta, an engineer at New York's Back Pocket Studios, the pitfall of many poor television mixes is that they don't take into account the sonic limitations of these speakers.

"I've think we've all heard those underscores where the only things you hear are snare drum and tambourine," he explains. "You know that there's a full band there, but because of the way it's mixed, you only hear the high-end instruments." Not only can TV speakers make bass frequencies disappear, they can also accentuate the levels of instruments that reside in the high and upper-mid ranges. Tambourines, shakers, and hi-hats are particularly prone to this problem and must be handled with care in a TV mix.

To avoid the potential problems caused by the small size of TV speakers, most engineers switch back and forth between their mid-size, close-field monitors (often Yamaha NS-10Ms or Genelecs) and small speakers similar to what you'd find in a typical television set. (Auratones are often used for this purpose.) The larger speakers are employed for the bulk of the mixing, but the mixes are constantly referenced on the smaller monitors to ensure that the relative instrument levels remain the same, regardless of the size of the speaker.

No matter what speakers you're monitoring with, it's generally a good idea to keep their volume relatively low. One axiom of mixing is that you should monitor (at least some of the time) at levels close to those at which the mix will ultimately be heard. Therefore, for a television project, your level should be representative of where the audience will set the TV volume, which probably won't be as loud as a typical music mix.

Selling the Product

A television mix usually has a voice-over (known as a "VO") or dialog that sits on top of the music. In the case of commercials, the audio team's directive—be it from the production company or the ad agency—is to deliver that voice loud and clear, even if it means turning the music down quite low in the post mix. Jingles that are sung deal with the same issue: the singers are generally mixed much louder in relation to the music than they would be on a record.

Consequently, during the music mix, it's essential to make sure that the lead instruments don't interfere with any VOs or lead vocals, which may or may not already be recorded. To do this, you'll need to check your mix alongside any other audio that will accompany the music. This won't be as big a concern if you're working on something other than a commercial. For example, in a documentary, there may be long stretches of film with no voice, only music and location sound. In those cases, your lead instruments can be as hot as you want them to be.

Take It to the Limit

Almost all television stations employ multiband audio limiters to prevent the program material from overmodulating the transmitters and to keep the signal within FCC guidelines. More importantly, these limiters are also used to maintain consistency of volume and equalization, as television stations typically broadcast audio from a number of different sources. Therefore, any signal containing loud peaks will trigger these limiters to lower the overall level (see **Fig. 1**). For a commercial, this can be devastating, as it will make the ad sound quieter than the spots surrounding it—a big no-no in the advertising business.

To avoid such a faux pas, it's important to tailor your mix so that it triggers those limiters as little as possible. According to Tony Volante, an engineer at New York's Soundtrack Studios, you're more likely to sneak your music through unscathed if *you* apply the compression to keep the dynamic range under control. "The broadcast limiters are like security guards," he says, "and you don't want to leave it up to them to take care of compressing your tracks."

In addition to being subject to compression, signals containing an excessive amount of frequency information (especially in the high end) will be automatically equalized by the stations' processors. You can help prevent this by keeping your mix smooth across the frequency spectrum. However, it would be wise not to do too much equalizing on the small speakers; because of their limited frequency response, what seems like a modest EQ change could actually be a fairly sizable one. Volante agrees: "On smaller speakers, you'll find that turning EQ knobs doesn't change the perceived tone that much, and you just wind up overequalizing things."

Applying EQ during a television mix is a balancing act. On one hand, you're

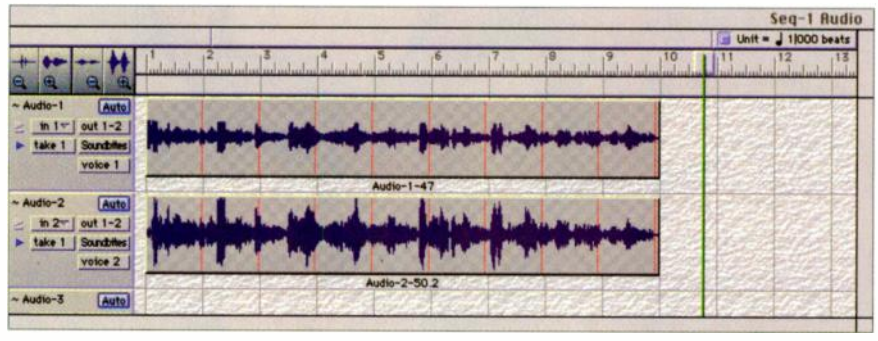
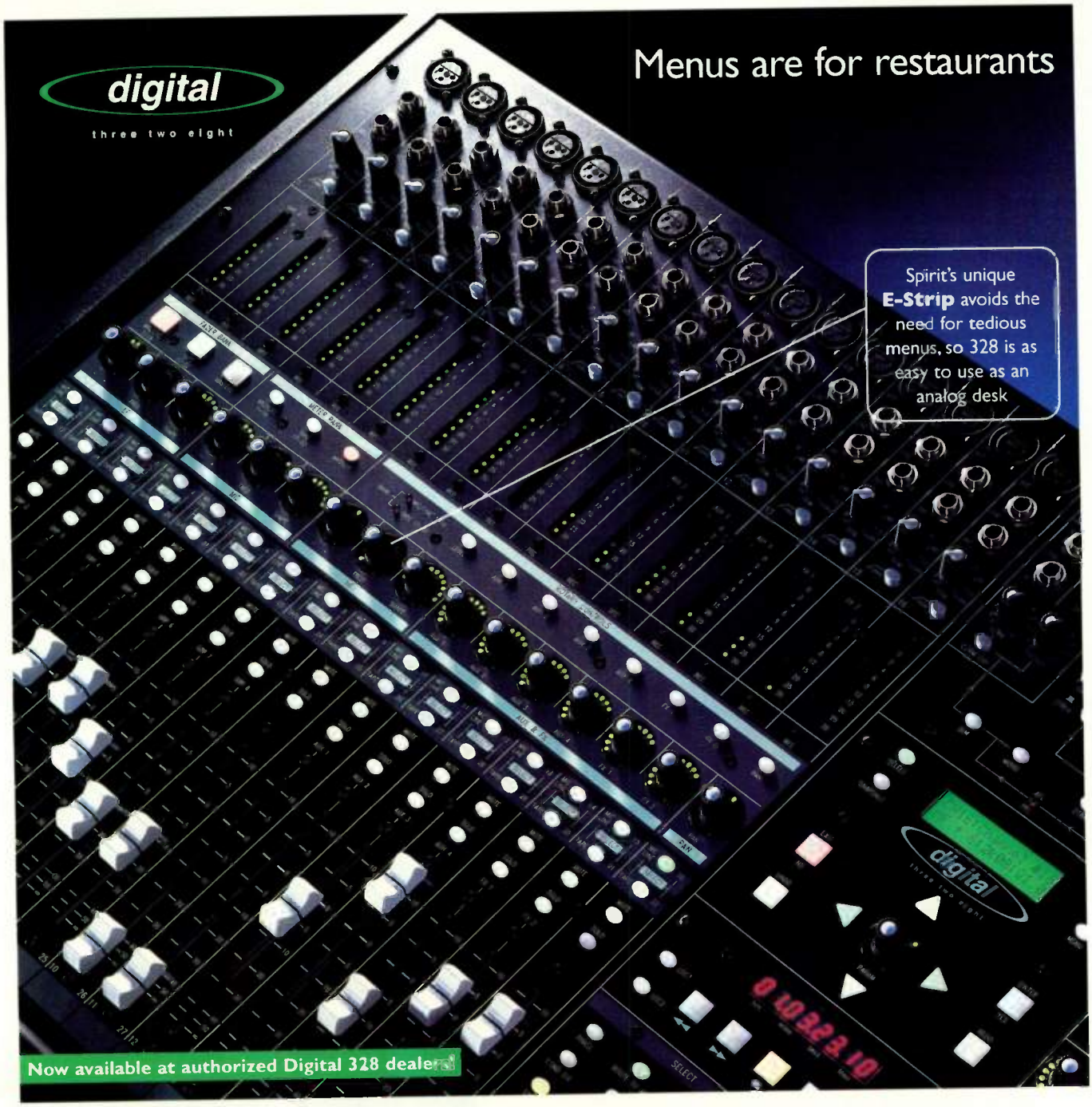


FIG. 1: This screen shot from Mark of the Unicorn's *Digital Performer* illustrates how broadcast limiters can squash the dynamic range of a signal. The bottom waveform represents the mono mix of a television jingle, taken directly from the master DAT. The top waveform shows that same mix after being broadcast, recorded from a conventional television set.



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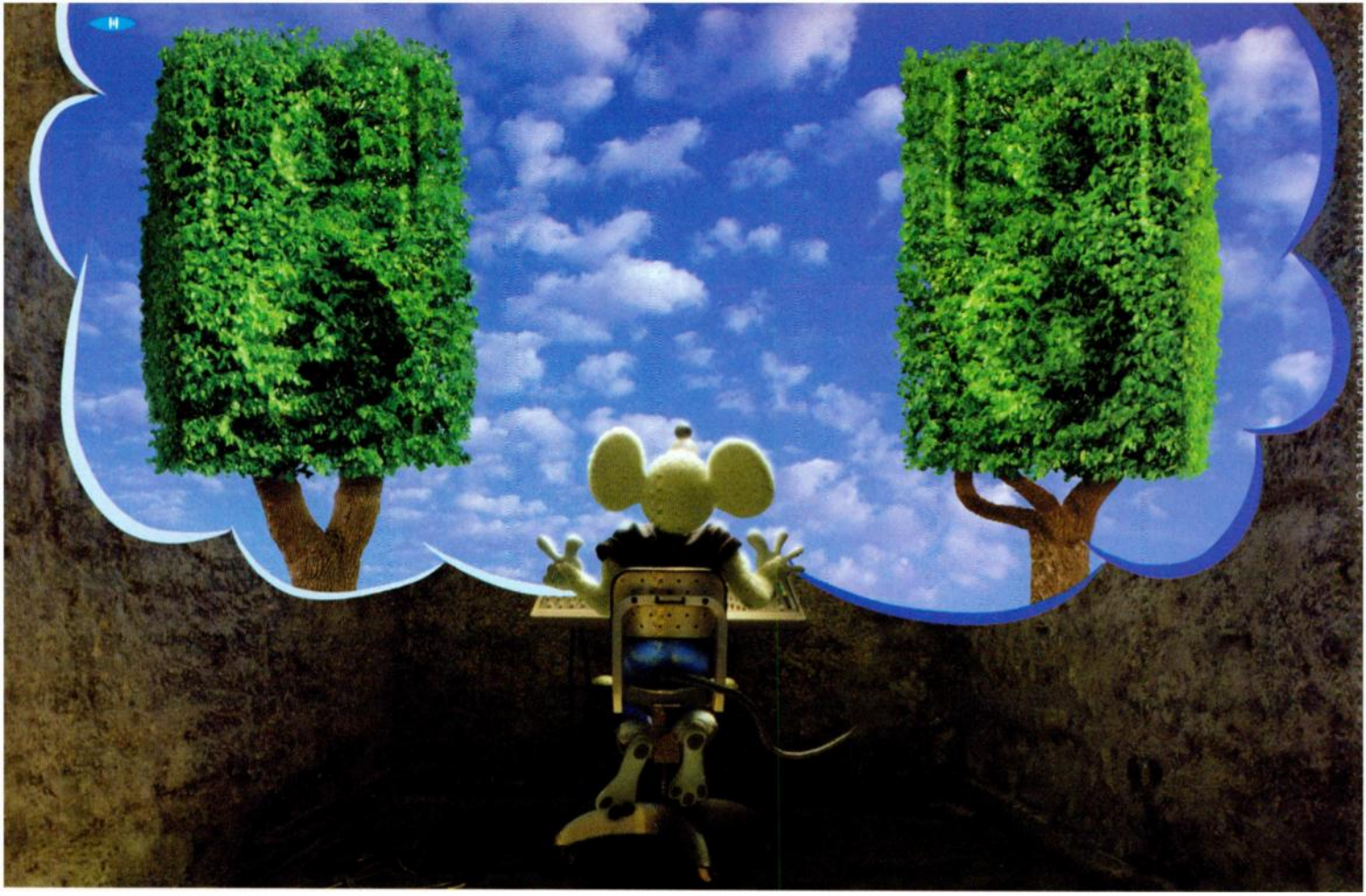

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trying to make things sound good on a cardboard-sounding TV speaker; on the other hand, if you goose the top or bottom too much, you'll have a mix that sounds lousy on bigger systems and also run the risk of triggering the broadcast limiters. Sometimes you can use compression, rather than EQ, to get an element to cut through a mix better on smaller speakers. Bass guitar tracks are a case in point: it may be tempting to boost the track's lower frequencies or increase its level, but unless you make subtle changes, your mix will sound unbalanced on larger monitors. Alternatively, if you compress the bass and squash down its peaks, you can get its overall level louder relative to the rest of the track. That way, the track stands out more without any change in frequency content.

Commonly used for mixing television audio, Auratone T5V monitors do a good job of simulating the typical speaker found in a television set.

So Little Time

During record mixes, producers and engineers have been known to spend many hours (or even days) perfecting the mix of a single song. In television, you're almost never afforded that luxury. In fact, many TV commercials are mixed in an hour or less. Because of these time constraints, engineers often use compression as a substitute for the

slower process of fine-tuning levels using an automation system. A compressor may not sound as natural, but it will get the job done alot faster.

If you're in a pinch for time, or if your studio has a shortage of good compressors, you can try compressing the entire mix, instead of processing each track separately. If you go this route, however, be mindful not to overly compress the mix, as this can cause pumping and breathing.

These by-products of compression are generally unwanted and viewed as a sonic detriment. But, according to Volante, sometimes you can use the pumping of a heavily compressed track to your advantage. "Compressors are going to pump—there's no way around it," he notes. "Sometimes you can utilize that effect to enhance what you're doing by making the pumping happen rhythmically in time with the music. Now, all of a sudden, sounds that were flat and dull start to come alive. The listeners won't perceive this as something wrong, they just hear it as a very live, aggressive sound."

Mono Compatibility

As a rule of thumb, you should assume that the majority of the audience will be listening to your mix in mono. Sometimes, when you sum a stereo signal to mono, elements that are panned too far left or right can change in level due to phase cancellation, thus generating flanging or comb-filtering effects. To avoid these problems, some engineers pan elements (including effects returns) only to a maximum of three and nine o'clock.

Not everyone agrees that you need to do this, but no matter how you approach the panning issue, it's prudent to periodically check your mixes in mono to make sure that they're holding up. Some studios will have a single Auratone hooked up in mono—not a bad idea if you're in this business for the long haul.

Doing a mono-compatible mix also means that you'll have to work harder to keep the mix uncluttered from a frequency standpoint. If a mix is going to be listened to only in stereo, you have the luxury of panning elements that occupy the same frequency range to opposite sides. In mono, however, these elements will be pushed together, and the result can be a muddy one. You should keep this in mind and carefully utilize equalization to ensure that each piece of the mix fits into its own sonic space.

SIX STEPS TO A BETTER TV MIX

1. Monitor at low levels to simulate the average listening environment.

2. Switch back and forth often between your regular monitors and smaller TV-type speakers (such as Auratones) to make sure your mixes will translate well. Be wary of equalizing when using the small monitors, because their limited frequency response can mislead you into overcompensating.

3. Make sure your mix is mono-compatible by periodically summing it to mono. One way to ensure compatibility is to pan no farther than three or nine o'clock.

4. Be sure to keep the dynamic range relatively small. Strong peaks are likely to trigger the broadcast limiters.

5. Keep lead elements lower than normal in mixes that will be combined with voice-overs or dialog; otherwise the entire music track may have to be turned down during the post mix.

6. Be deferential when working with clients, and try to handle absurd suggestions diplomatically.

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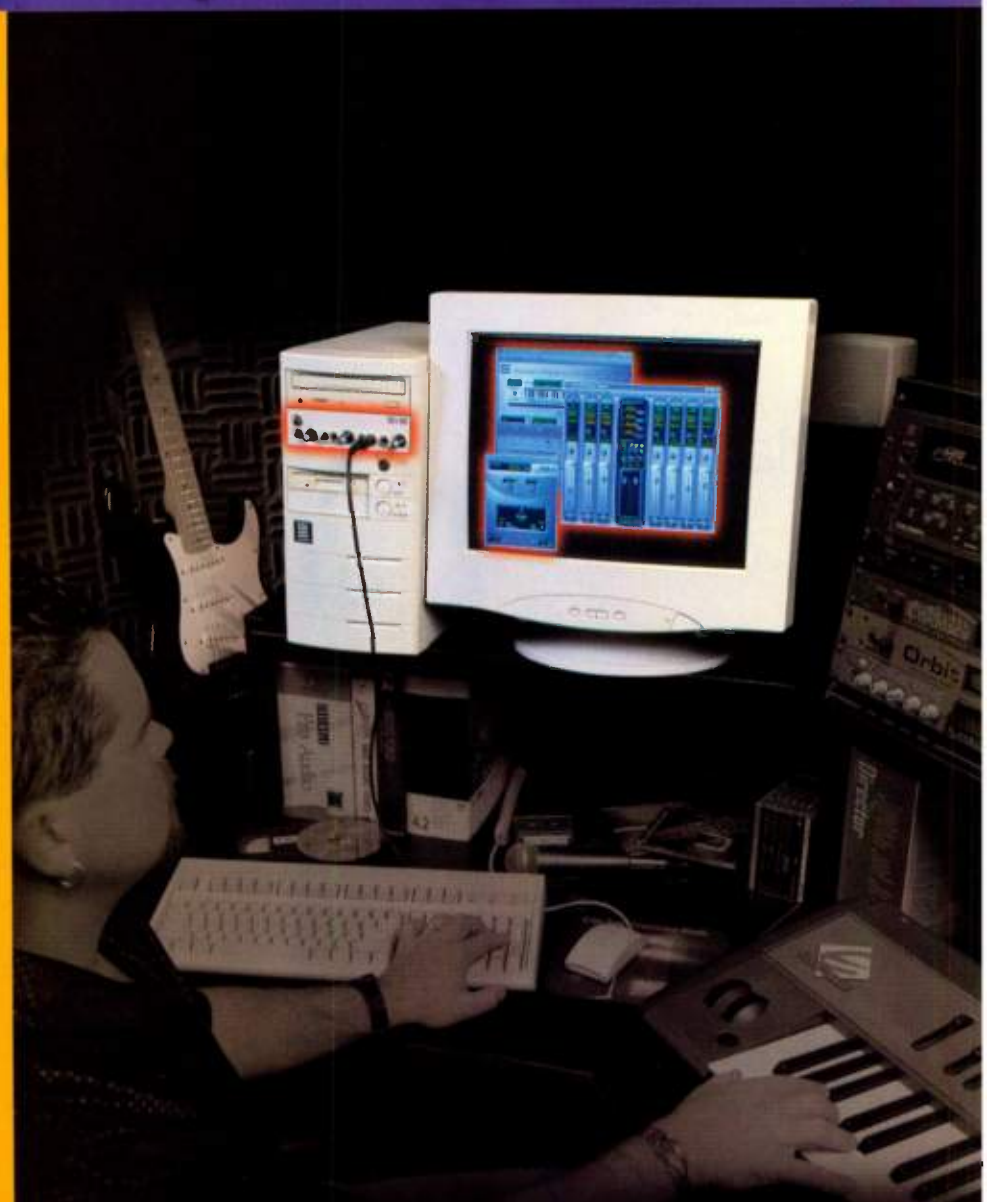

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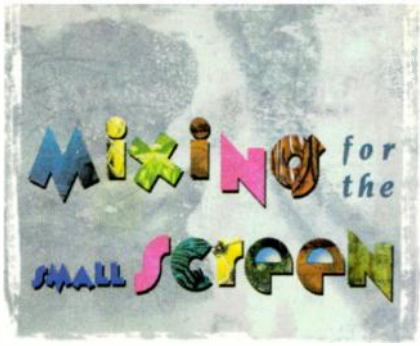

Using Effects

Reverb and time-based effects, such as chorus and delay, are handled in pretty much the same way as they would be in any kind of music mix. One caveat is that, because the mix must be mono-compatible, you'll probably want to steer clear of autopanning, ping-pong delays, and any other effect that will not translate well to mono.

In fact, you'll probably notice that *any* stereo effect will diminish somewhat when summed to mono. This is especially true of reverb: because reverb is the simulation of an acoustic space and generally requires a stereo image to sound convincing, you'll have to find a happy medium so that there's enough effect in mono, but not too much in stereo.

The Post Mix

Up until now, we've been concentrating on mixing multitrack pieces of music down to two tracks. It's also important to look at some of the issues you'll run into if you're doing a post mix. From an audio standpoint, you have the same concerns regarding dynamic range and equalization as you did during the music mix, and you should follow the same procedure regarding monitoring and mono compatibility.

A bigger issue, though, is the heightened level of involvement from the clients during this stage of the process. Because many clients have no musical background, there will be times when they make somewhat inane or impossible suggestions, and you will need to have excellent diplomatic skills to deal with them. What you must not do in this situation is ridicule, patronize, or mislead them. Arlotta describes how he handles these dilemmas. "I never say that anything is a stupid idea, even if I know that it is. You shouldn't make anyone feel silly, because they are the clients and you're working for them."

The best thing to do in those situations is explain to the clients why their suggestion might cause problems from a sonic standpoint and hope that, after hearing what you have to say, they will agree. Volante feels that knowing how to deal with clients is a very important skill for post engineers. "I think there are a lot of people who are good mixers," he says, "but there are fewer people who know how to deal well with clients."

Air It!

Despite all the little things that differentiate a television music mix and a record mix, they're still very similar in many respects. If you're good at doing music mixes, you can become equally skillful at mixing for the small screen. The trickiest things to adapt to are working with the short turnaround times and dealing with a new type of client—especially advertising people. If you can get used to those things, you'll find that mixing for the small screen isn't that big an adjustment after all.

Mike Levine *is a composer, session player, and author of four books, including* How to Be a Working Musician, *recently published by Billboard Books. Visit his Web site at www.mikelevine.com.*

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RECORDING MUSICIAN

Gearing up for Critical Vocals

Test procedures for determining the optimal signal path.

By Brian Knave

Part of the allure of the personal studio is low overhead. Rather than paying exorbitant hourly rates at a commercial facility, where the stress of the clock can make it hard to deliver an inspired performance, personal studio owners can relax and take plenty of time to perfect their parts. The muse, after all, is not a circus animal, ready to jump through hoops at the crack of a whip.

On the other hand, there are definite advantages to recording in a commercial studio. These include great-sounding rooms, a wide selection of premium gear, and—this is important—experienced engineers who know how to get a great sound quickly.

So here's the million-dollar question: Is it possible to get as good a sound in a dinky little home studio as can be achieved in a full-blown commercial facility? Well, as my grandmother used to say, It all depends. Obviously, if you're recording, say, an orchestra, big band, or gamelan, then probably not. Even a 4-piece drum kit can prove a bear to record well in a garage or bedroom studio.

Many instruments do, in fact, lend themselves to small-space recording, and the voice is one of them. After all, small, dead-sounding vocal booths are often employed for recording vocals, even in major studios, at least for pop applications. And it's relatively easy to construct a dead-sounding space in a closet or garage. After that detail is taken care of, quality of sound depends on two things: the signal path and your knowledge of recording.

This month, I'll detail the test procedures I use to determine the best signal path for recording a particular vocalist. This is an elaborate process that I usually reserve for critical vocal tracks only. (By "critical," I mean final tracks for a CD.) However, it's very worthwhile, especially if you've never done it before. It teaches you how each piece of gear affects the final sound and how various components work

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The Sennheiser MD421 dynamic microphone is typically thought of as a tom mic, but it's also an excellent choice for certain vocalists.

together—the key knowledge required to ensure an optimal signal path for the application at hand. Each time you perform the tests, your knowledge base grows, which ultimately pays off by enabling you to work faster and smarter.

QUALITY COUNTS

When you're going to record critical vocal tracks, make every effort to get your hands on the best gear you can. Aside from whatever recording medium you've decided to use, the requisite gear includes a microphone, a mic preamp, a dynamics processor (which may not even be needed), and perhaps a D/A converter, if the recording medium is digital. Those are only a few things, and the fewer the better: in the interest of audio quality, it's wise to keep the signal path as short as possible. Let your mantra be Direct-to-tape whenever possible.

Even among those few items, there are countless products to choose from. Of the four, the microphone is the most critical. However, in the end, the recording chain is only as strong as its weakest link, so it's important to search out quality equipment in each category. What's more, despite the reputation of any particular piece of gear, it's impossible to know in advance how any two products will sound together, short of giving them a try. Unfortunately, this luxury is seldom available to the personal-studio recordist.

So what's a fellow to do? Scrounge. Borrow. Beg. Rent. Do whatever it takes to amass the gear you need. Try to get at least three different mics, preamps, and dynamics processors to make the tests worthwhile. Then, your task is to compare. (I skipped converters, but if you're really fastidious, you'll want to compare a few of those, as well.)

THE MICROPHONE

Start with the microphones, using the stock mic preamps on your console. Your goal is to find the mic that flatters the voice the most—and, ideally, allows you to record with no EQ. Usually, large-diaphragm condenser mics are best for recording vocals, but not always. Sometimes a condenser is just too brittle sounding or revealing, especially on a harsh voice. In those cases, a dynamic or ribbon mic may provide a better sound. Again, you never know until you compare.

Don't fall into the trap of thinking that the most expensive mic is necessarily the best. The choice of mic should always depend on how it sounds on the voice in question (unless, of course, you're using the mic to create a special effect). Bonnie Raitt, who can afford whatever microphone she wants, has been known to use a relatively inexpensive dynamic mic, the Electro-Voice RE20, for her studio vocal tracks.

To compare microphones, set them up in a tight cluster with the capsules aligned on a horizontal plane facing the singer. (Depending on the size of the mics, six is about the most you can group together for a practical test.) Set any multiple-pattern mics to the cardioid position, and disengage all pads and high-pass filters. Likewise, on the board, make sure all pads, filters, and effects are disengaged for the channels handling the mic signals (both send and return). The point is to even the playing field so you can hear each mic in its flat, "unaffected" state.

Position the singer about two feet back from the mics. This allows enough time and space for the sound to open up so that each mic gets hit with the same information. Next, while the singer rehearses the tune, adjust the gain for each mic preamp until all track levels are the same. This is critical to the process, as even the slightest disparity in levels can skew your perception, possibly leading you to favor, unfairly, one sound over another.

Now it's time to record a pass and listen to the results. Initially, at least, use studio monitors rather than headphones to listen to the tracks. First, mute each channel. Then, unmute one at a time to compare the signals. Listen to each track both with the music tracks and soloed (to check for mic self-noise, room noise, etc.). Discerning which mic sounds best is a somewhat subjective matter, but hopefully it will be readily apparent. (Of course, the paying client always gets the final say.) Once you've selected the microphone, it's time to match it with a preamp.

THE PREAMP

The procedure for comparing mic preamps is much the same as for comparing mics. If you have the resources to split the mic signal and send it through several preamps at once, great. In lieu of that, however, you will have to record a separate pass for each preamp. (This also means that the singer will have to match his or her tracks one to another.)

Again, use each preamp in its "flat" state, with no pads, filters, phase-reverse switches, or dynamics processing activated. (You can play with that stuff later, once you've selected the preamp.) Be sure to match the levels as closely as possible. If one track is recorded hotter than the rest, you can compensate for the difference by using the faders during playback.

The differences between preamps can be subtle, so listen carefully. Often I find it helpful to pan the tracks hard left or right, so I can listen through one speaker only. Things to listen for include openness, airiness, body, clarity, coloration, and detail. Personally, I

Of the fifteen presets offered by the affordable PreSonus Blue Max Smart Compressor, three are ready-made for vocal applications. There is also a manual setting to allow for user-specific input, ratio, attack, release, and output settings.

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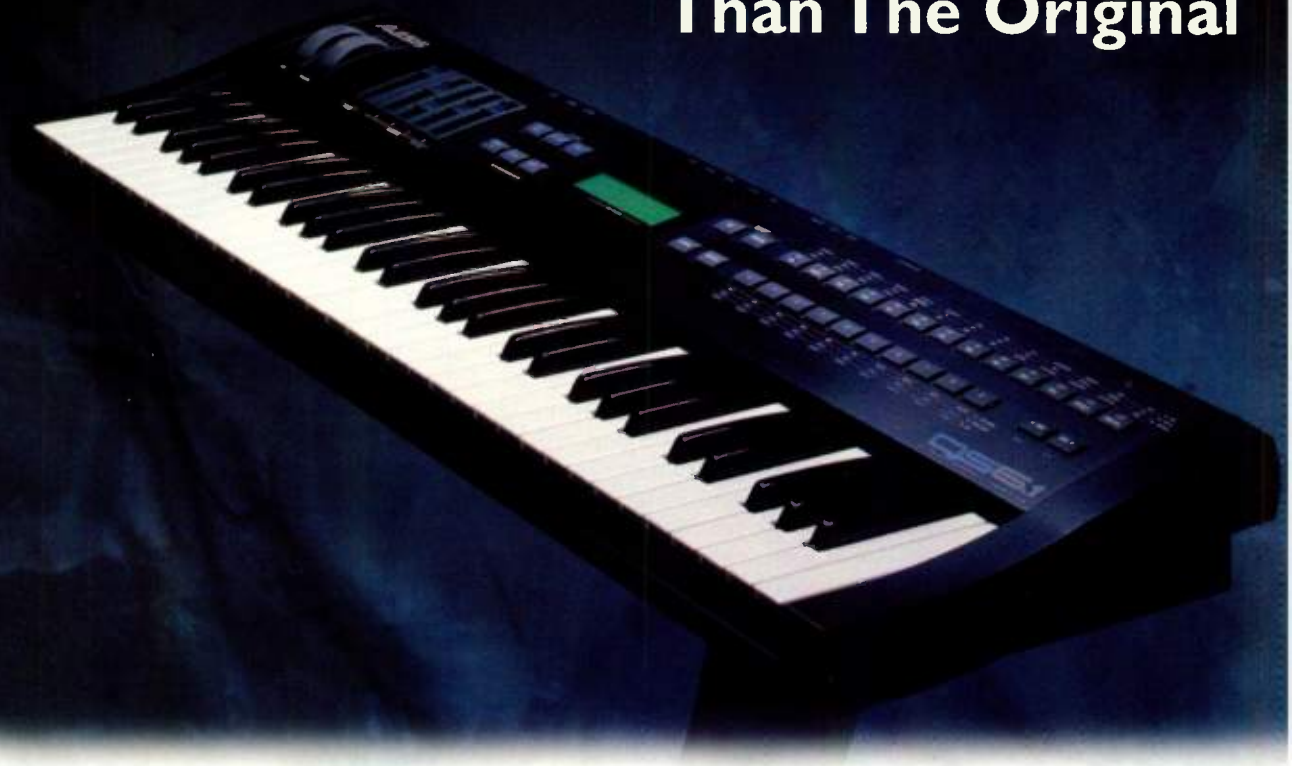

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ALESIS synthesizers

tend to favor a very open, transparent (i.e., uncolored) sound with good detail (hard, clean edges rather than soft, smeary ones) and full body (i.e., with plenty of lows, and not thin-sounding).

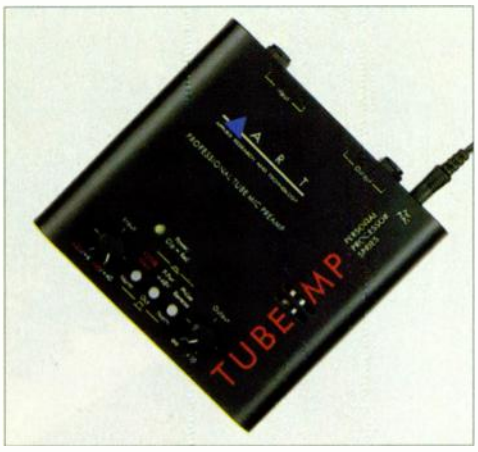
Of course, sometimes coloration is what you're after: for example, when you're matching a tube mic preamp with a solid-state mic in hopes of "warming up" the signal. (I've found that a quality tube mic adds a more desirable "tubiness" to the sound than a solid-state mic going through a tube mic preamp. And here's a tip: if you really want a lot of tube coloration from a tube mic pre, try a lower-priced unit rather than a high-end one.)

THE DYNAMICS PROCESSOR

After selecting the mic pre, you can compare dynamics processors. Is this starting to sound like a hassle? Well, it *is* a hassle! But I guarantee that if you compare several different compressors head-to-head, you'll hear a surprising amount of variance in the types and amounts of coloration that each imparts to the signal.

For vocals, I generally prefer the sound of tube compressors with optical (rather than VCA) control, because they tend to sound fatter, fuller, and more luscious than solid-state, VCA-based units. On the other hand, optical compressors may not give you as much precision control as VCA-based models. Again, comparison is the only way to learn what sounds and works best for the particular singer, song, and signal path.

Other useful dynamics processes for recording vocals include expansion and peak limiting. Usually, my first approach is to attempt recording without *any* dynamics control. But various factors often make this impractical. For example, in a studio plagued by environmental noise (such as trucks roaring by just outside the door), an expander can keep down the noise between sections. Or, for a song with a broad dynamic range, a compressor will allow you to bring up the quieter parts without increasing the loud parts too much.



The affordable A.R.T. Tube MP is practically a must-have for the budget-challenged personal-studio recordist. Not only can it add some serious warmth to vocal tracks, but it also works great as a DI for recording electric bass guitar.

I do everything possible to avoid peak limiting (including coaching the singer on working the mic). However, when recording digitally, it's better to catch overshoots with a peak limiter than to allow an otherwise stellar performance to be ruined by digital distortion.

THE ROOM

In a sense, the recording space is also a part of the signal path because its ambient sound is critical to recording great-sounding vocal tracks. Even with the best gear in the world, you won't get a good sound in a space that's riddled with unwanted echoes, standing waves, or excessive environmental noise.

Usually, a fairly dead-sounding space is preferable for recording vocals. A vocal booth is ideal, but you can get similar results by deadening a corner of the room with studio foam and then setting up the mic so that the capsule faces into the corner. If you still hear unwanted reflections, position a foam-covered partition in front of the corner, effectively walling it in. If the recording gear is set up in a bedroom, try recording in a clothes-filled closet. Rugs, carpet, padded furniture, and draperies can also be used to help deaden the acoustics of a space. (For more information on acoustics treatment, check out "Silent Partners" in the August 1996 **EM.**)

If house-induced environmental noise is a problem, deal with it. For example, refrigerators, freezers, hot-water heaters, and telephones can be unplugged temporarily to quiet the recording space. To quell outside noise, close windows tightly and cover them with foam or some other material.

ALL TOGETHER NOW

One alternative I haven't discussed is the use of multifeatured, "channel-strip" type units, commonly called voice processors. These typically comprise a mic preamp, dynamics processor, and EQ section in a single box. Some also contain exciter circuits and A/D converters.

Obviously, a voice processor can greatly simplify assembly of the signal path. In my experience, however, I've found that a unit with a great-sounding mic preamp doesn't necessarily have the best dynamics section (or vice versa). For that reason, it still pays to compare the individual sections, any of which can usually be bypassed if necessary.

In an upcoming "Recording Musician," we'll cover specific tips and techniques for recording critical vocal tracks. In the meantime, set up that gear and start comparing.

Brian Knave *is an associate editor of* **EM.**

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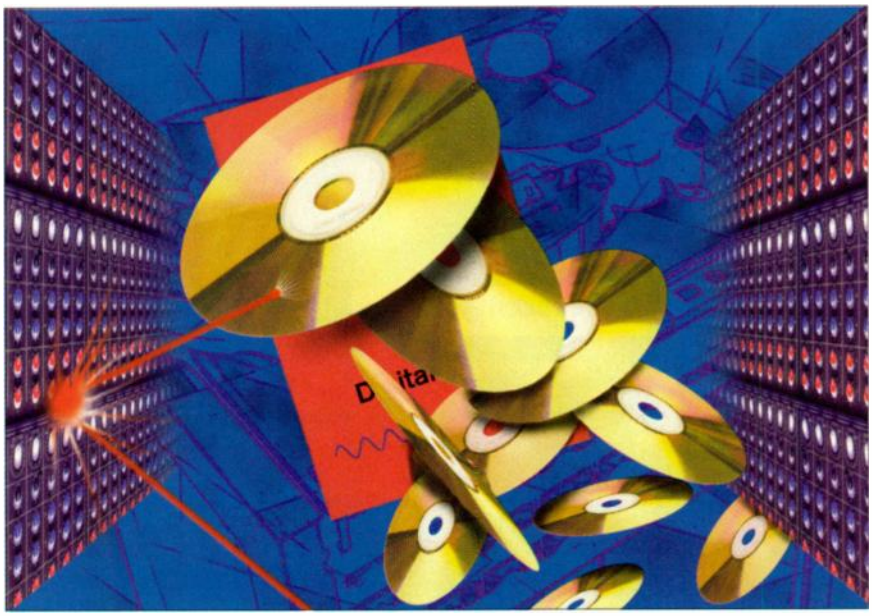
Going by the Book

The Red Book specification is the last word on audio CD.

By Markus Fest and Rob Griffith

As CD-Recordable (CD-R) technology becomes a standard feature in every professional and project music studio, musicians are becoming increasingly concerned that their CD-R software and hard drive be able to produce audio CDs suitable for mastering and playback. To accomplish this, it helps to understand the Red Book specification for CD-DA (Compact Disc Digital Audio). This article will present an explanation of this format, which should be useful to studio musicians and engineers who want to create audio CDs on the desktop.



While the actual name of this specification is "System Description Compact Disc Digital Audio," it has come to be known as "Red Book" because of the color of the booklet in which the specification was first published. The spec, which was finalized by Philips and Sony in 1980 and later published as IEC standard 908, defines a physical and logical format for storing audio and other data on a compact disc. (You can purchase a copy for $115 from the American National Standards Institute. Visit their Web site at www.ansi.org.)

Much of the Red Book specification defines physical characteristics such as the size and thickness of the CD, the wavelength of the laser, and the type of error correction used. Some characteristics, such as the wavelength of the laser light, were based on engineering considerations, while others were based on marketing decisions. For instance, the physical diameter of a CD was based on diagonal measurements of a cassette tape.

The most important portion of the Red Book specification defines logical structures. These include Lead In, Lead Out, tracks, gaps, Index Points (which mark time within a track), sampling rate, audio channels, and subcode channels P through W. (A subcode channel is an additional stream of reference data that will be discussed shortly.) It's very helpful to understand how data is formatted on a disc.

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TAKE THE LEAD

The Red Book spec says that a CD must contain a Lead In Area, Program Area, and Lead Out Area (see **Fig. 1**). Contained in the Lead In is a Table of Contents (TOC) that lists track numbers; the location of each track in terms of minutes, seconds, and sectors (75 sectors per second); and the end point of the CD. A disc can contain up to 99 tracks (tracks 1 to 99). A track can contain up to 100 time markers, called Index Points (0 to 99). The Program Area is the area that actually contains audio tracks or data. The Lead Out Area is 90 seconds long and is encoded as a silent track. It is used as a reference point in case the CD player overshoots the Program Area.

Each track of audio must contain at least one Index Point that indicates the start of the track (Index 1). Often, there's a gap prior to the start of a track, in which case another Index Point (Index 0) is required. The TOC, however, only references the start of each track (i.e., Index 1) and tells where each track begins in terms of Absolute Time. (Time is measured on a CD as either Absolute Time, counted from the beginning of the Program Area, or Relative Time, calculated from the beginning of the track.) The TOC, then, will include each track number and the Absolute Time where Index 1 is located in each track.

According to the Red Book standard, CD audio is stored as 16-bit stereo data, which can also be calculated as 4 bytes (2 for each channel) per sample. Because time is divided into 75 sectors per second, each sector contains 2,352 bytes, so each second contains 176,400 bytes. To put it another way, 176,400 bytes divided by 4 bytes per sample equals 44,100 samples per second, which is how we arrive at the standard sampling rate for CD audio.

Red Book allows several audio channels to be interleaved and played simultaneously. Of course, everyone is aware that audio CDs contain a left and right channel. In fact, Red Book allows up to four channels of audio—quadraphonics revisited! However, it would be difficult to find a CD that actually contained four channels of audio and just as difficult to find a CD player that could play it.

CHANNEL SURFING

In addition to the audio data format, Red Book defines subchannels P through W. Channel P is a single-bit flag originally intended to provide simplified playback systems with a way to find the gaps between tracks. No playback systems actually use this method, though.

▼

The Red Book specification defines a format for storing audio and other data on a compact disc.

The contents of the P channel are completely determined by the Q channel, so CD mastering software requires no special user input for the P channel. The Q channel controls the gap, tells the current time within a track, indicates the current track number, and allows for the placement of Index Points within a track (see the table "Contents of the Q Channel").

Finally, the Q channel can contain other useful data, including ISRC (International Standard Recording Code), UPC/EAN (Universal Product Code/European Article Number) codes, and the start time of the Lead Out. The UPC/EAN is the catalog number of the disc and does not change from track to track. The ISRC consists of twelve characters representing the country, owner, year, and serial number of the recording. Each track can have a unique ISRC, and the ISRC can be used by radio stations to track which songs have been played.

When we talk about PQ subcode editing, we are really talking about editing the Q channel, especially the start and stop points of tracks.

INDEX POINTS

A track usually has at least two Index Points, Index 0 and Index 1 (see **Fig. 2**). Index 0 marks the beginning of the gap that precedes the track. As mentioned earlier, the location of each track is stored in the TOC. The TOC doesn't store Index 0, but it does store Index 1. It is sometimes useful to use a nonsilent gap (containing applause, for example). To play this gap, you have to start at Index 1 of the previous track and play through the entire track, because CD players do not start with Index 0.

Index 1 defines the Relative Time as 00:00:00 in a track. Therefore, Index 0 is always a negative Relative Time. For example, if a gap is two seconds long, Index 0 is –00:02:00. However, the pause can actually be any length, including zero seconds. In that case, no Index 0 is required.

Red Book allows the option of storing up to 100 Index Points within a track, but most audio CD players can't see Index Points other than 0 and 1. The TOC only describes Index Point 1, so there is no practical way for most CD players to jump to any other Index Point without searching the disc. People sometimes ask, "What good are they?" As a matter of fact, Index Points within a track are useful in complex music forms, such as classical music, to mark different

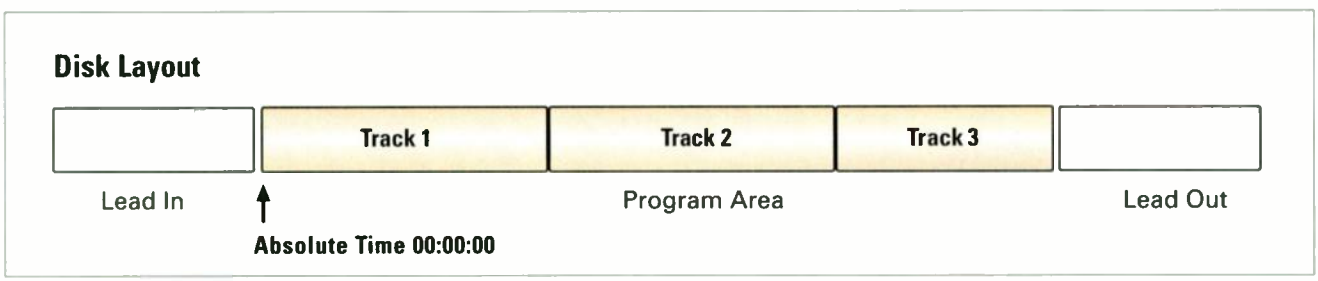
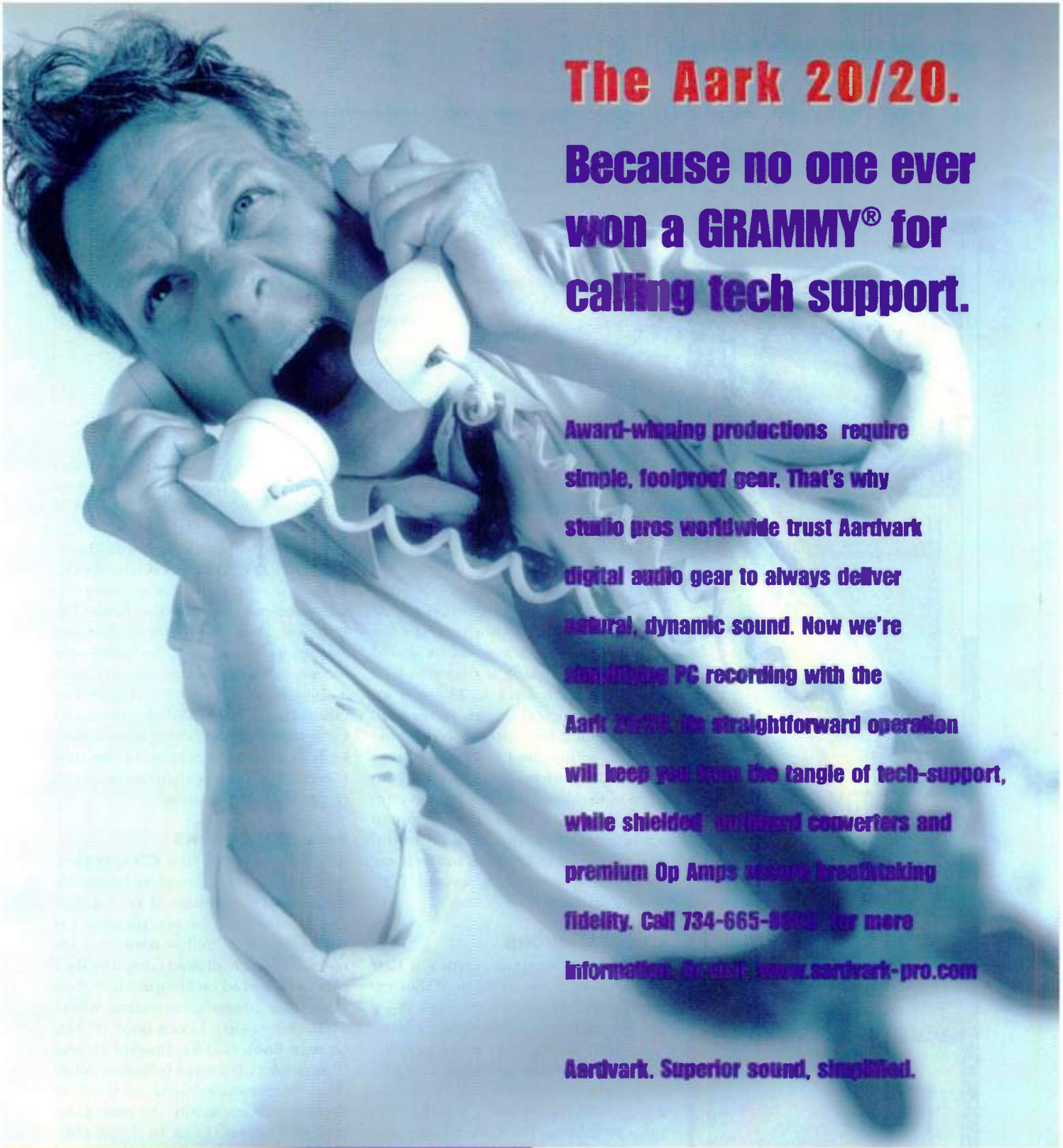


FIG. 1: The arrangement of data on a CD is detailed in the Red Book audio specification. All discs will contain a Lead In, Program Area, and Lead Out.



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Contents of the Q Channel

CONTROL	2- or 4-channel audio, copy bit, preemphasis bit
ADR	ADR = 1: Position information as described here ADR = 2: Remainder of data is UPC/EAN code (if present at least 1 out of every 100 subcode frames) ADR = 3: Remainder of data is ISRC code (if present at least 1 out of every 30 subcode frames)
TNO	Current track number
X	Current index number
MIN, SEC, FRAME	Current track relative time
AAMIN, ASEC, AFRAME	Current absolute time

sections within a larger movement. Also, because Red Book limits a CD to 99 tracks, sound designers sometimes use Index Points to mark where multiple sound clips are stored within a single track.

It is unfortunate that the TOC only includes Index 1 for each track. It would be useful if multiple Index Points within a track could be stored in the TOC, making it easier to find sections within a track. There is no physical or logical reason for excluding this data, because there is plenty of room within the Lead In section to store more information about the contents of the disc. However, it is very unlikely that the specification will open up to allow this, and even more unlikely that audio CD players will be re-engineered to support a revised TOC specification.

OTHER FLAVORS

Red Book also supports types of CDs that are seldom or never used. For example, subchannels R through W include graphics modes, called CD+G. Graphics modes allow some graphic and text data to be included on a CD. This would allow a computer to display graphic data along with the audio. Pictures and video effects, such as cut and dissolve, are possible. CD+G is sometimes used for karaoke discs.

CD+MIDI allows a channel of MIDI data to play in sync with the digital audio. In the early 1990s, companies such as Warner New Media put considerable research and development into this format. Some CD players, such as the old JVC XL-G512, actually have a MIDI Out port. Unfortunately, it was difficult to get the MIDI signal to synchronize properly with the digital audio signal, and the market at the time was not ripe for CD+MIDI. Today, you would be hard-pressed to find a CD player with a MIDI Out.

Although not part of the original specification, CD-Text allows the song title, author, producer, and other information to be encoded onto Red Book CDs. Presently, audio CD players do not usually recognize such text, but in the near future, you will probably see a crop of audio CD players that can display this information. Some CDs being produced today have this text information already included, in anticipation of the new players. CD-Text is encoded in the R through W subchannels.

PRIMARY COLORS

Red Book was the first CD specification to be widely accepted by the industry and consumers alike, but not the last. In 1983, the specification for CD-ROM called "Yellow Book" was announced, which allowed computer data files to be stored on compact disc. Several other formats, including White Book (Video-CD), Green Book (CD-i), Orange Book (CD-R), Photo CD, and Blue Book (CD-Extra) followed. All of these formats are based on the Red Book specification, with the same physical parameters, Lead In, Lead Out, and Program Area. They all support audio tracks in addition to data tracks (called *mixed-mode disc*).

Of particular interest to musicians is the Orange Book specification for CD-R. Two of the four recording methods specified in the Orange Book can be used to create audio CDs. These are the Disc-at-Once (DAO) and Track-at-Once (TAO) modes. With DAO, the entire disc is recorded with one pass of the laser. TAO, on the other hand,

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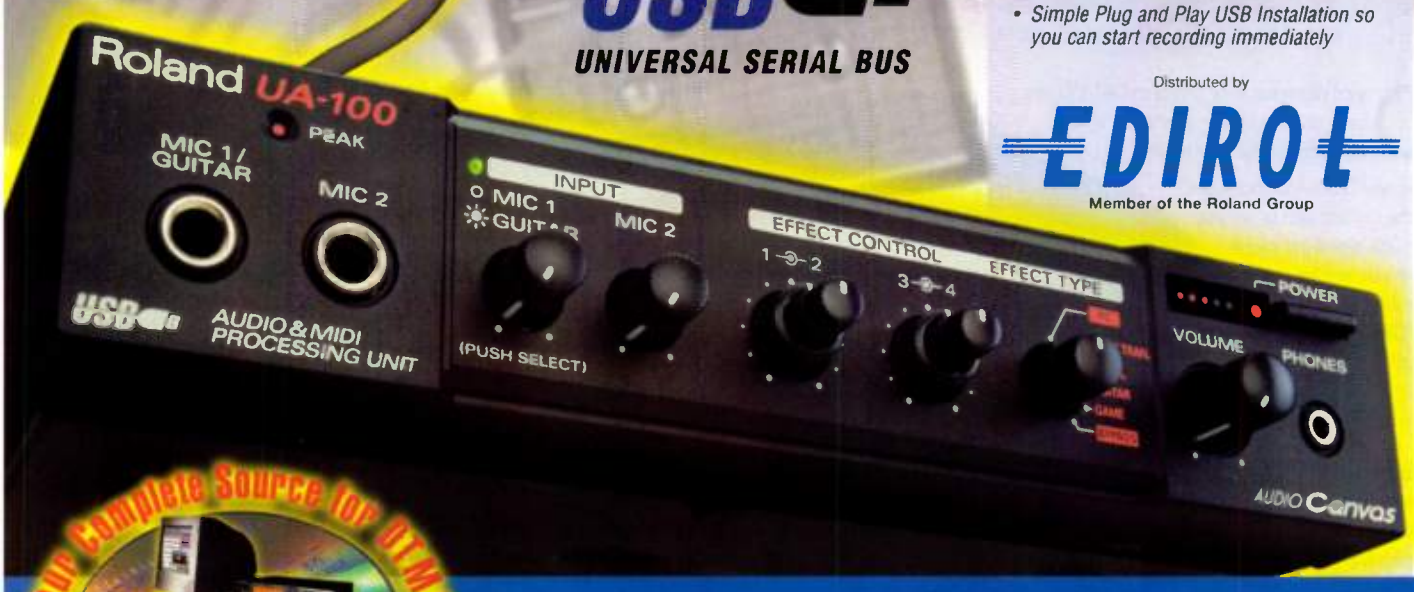

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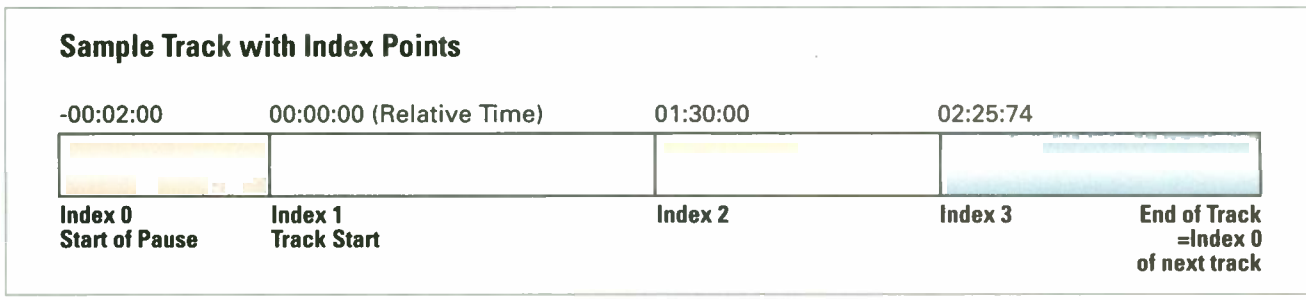


FIG. 2: Indices are used to mark internal time points within a track but cannot normally be accessed directly by CD hardware.

ramps the laser down at the end of each track, creating "link blocks" of useless data.

Both methods can be used to create Red Book–compliant discs that are suitable as masters for mass replication. In the case of TAO mode, the Red Book rules for the P subchannel are slightly relaxed, but this should not have any practical consequences. A much more significant restriction is that, when using TAO mode, it is impossible to create seamless, zero-length pauses or crossfades between tracks because of the link areas between tracks. Some mastering equipment even flags these link areas as "E-32 errors" and rejects the disc. Although newer equipment generally can create flawless discs from TAO masters, DAO is clearly the way to go for professional audio CD premastering.

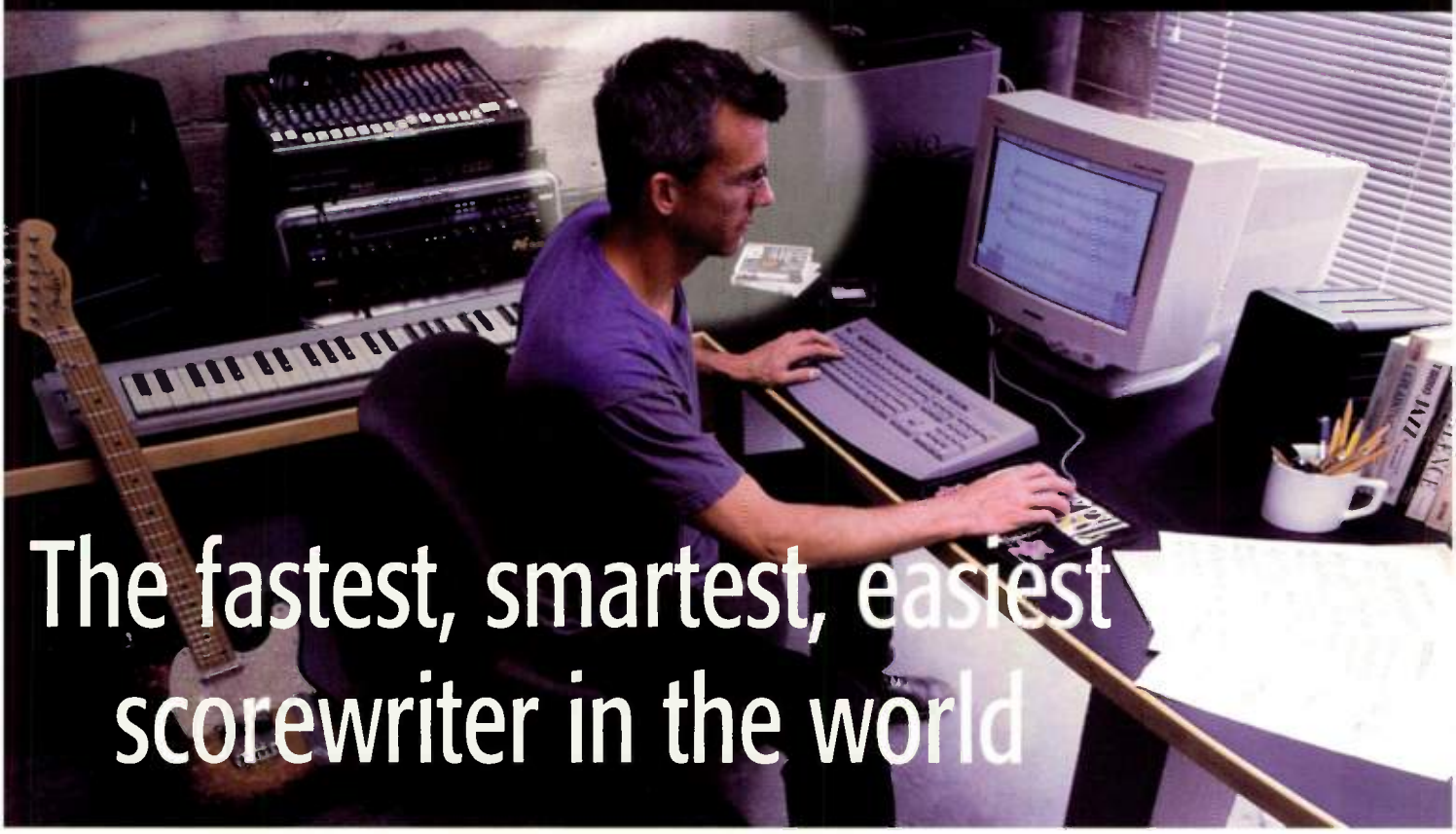
THE FINAL PICTURE

At this point, the desktop musician may ask, "How do I know if my CD-R software records true Red Book CDs?" The answer is, if your CD burning software allows you to create audio CDs, they will be Red Book–compliant, barring software or hardware bugs. Of course, your software may not support all the features of Red Book. For example, you may not be able to add Index Points, but you may not need to do that anyway. Nor is it likely to support CD+G or CD+MIDI, but, again, there's no point in supporting formats that are rarely or never used.

Red Book is not a perfect format, and in hindsight, many things could be improved. However, because of its universal acceptance, Red Book audio CDs will be with us for years to come.

Markus Fest *is the lead engineer and* **Rob Griffith** *is a product manager for Adaptec* Toast *and Adaptec* Jam.

SIBELIUS



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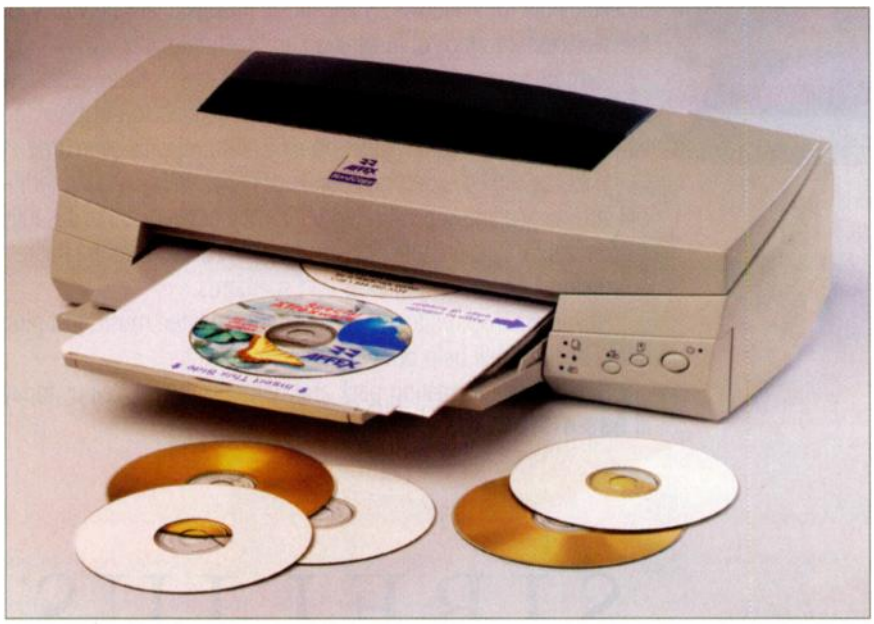
Check out these tools and tips for creating your own CD-R packaging.

By Erik Hawkins

So you've bought a CD recorder, and you're now giddily burning CDs to send out to A&R folks. Congratulations, you're sure to impress. But wait a minute—are you just throwing those CD-Rs in their jewel cases with some chicken-scratch handwriting on their covers? (From what I've seen, a lot of people are.) Even though your CD-R is a demo, or "one-off," treat the packaging as if your disc were a full-fledged commercial release. Compelling artwork, good design, and legible writing are key to informing and impressing listeners.

CD-Rs are beginning to appear more frequently on the desks of A&R reps, publicists, music directors, and producers. Therefore, packaging is critical in the game of making your disc stand out from the pack. This doesn't mean you need to spend an arm and a leg creating the mother of all jewel cases. With a little ingenuity and originality, combined with the right tools and some tips I've collected from the A&R dugouts, you can inexpensively design and print a killer CD-R package at home.



CD-R printers, such as the Affex SK7 shown above, are more expensive than labeling kits, but they make it easier to process large quantities of CDs and the results are more professional looking.

NO EXCUSES

Just a few years ago, the tools needed for creating jewel-case covers or booklets, tray cards (the back-panel insert), and CD-R labels were either nonexistent or astronomically priced. It was easy to get away with just scribbling on your CD-R with a permanent marker because that's all that was available to musicians doing it themselves. Today, you can find CD-R labeling kits at office supply stores, at some of the larger music stores, and through computer mail-order houses. The prices for high-quality printers and scanners have fallen dramatically, and systems are now available that let you print directly onto a CD-R. Combine these items with some desktop-publishing software, and you have a DIY graphics house.

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A&R folks and others who receive CD-R demos know these tools are available. If you submit a CD-R demo with a sloppy packaging job, it looks like you either are uninformed or just don't care. Even though many A&R reps say packaging is not that important, a well-conceived and well-assembled jewel case tells them "this band [or artist] has its act together."

"You send little signals about how serious you are in the way you present things," advises Doug Minnick, Taxi's vice president of A&R. "Even the way you address your mailing envelope is important. A&R people watch for these things."

Even if you're not burning your own CD-Rs, you still need to make sure your jewel case and CD-R are presentable. If the one-off service you're working with doesn't do design and printing, take your undecorated CD-R home and finish the job there. If you only have a computer and you're not in a financial position to spring for a color printer and a scanner, do your design and layout work at home, and do the scanning and printing somewhere that rents desktop-publishing stations by the hour.

THE BASICS

There are two basic types of music demos: the ones used to shop songs and the ones used to shop acts. Demos for shopping songs don't need to be as ornate as demos for shopping a band or a solo artist. Both types need to convey essential information (e.g., name, address, phone number, song names, song order, and copyrights), but the artist demo also needs to convey something about the act's style and image. Music directors and publishers usually don't care what a songwriter looks like, but an A&R person looking for a new act is always interested in an artist's image.

If you're designing the jewel case for your act, try to make the artwork mirror the kind of music you play. If your music evokes peaceful fields of yellow flowers, don't use rusting chain saws in your artwork. If you're unsure where to begin, you can get some ideas by asking your friends and colleagues what kind of images your music evokes for them. Also, check out the booklets in your own CD collection to see what the pros do.

Make sure there is at least one clear picture of yourself and your band somewhere in the jewel case, because A&R people need to associate the music with the performer. If you're not comfortable having your mug on the cover, put a photo inside the booklet. It's likely that your demo will become irrevocably separated from its promo pack. In the process of being played in a car, knocked off the desk, and handed off to other A&R people, this separation is inevitable. Having photos and lyrics in the jewel case booklet makes the jewel case more complete and independent of the promo package. Don't print your song lyrics and photos too small—text smaller than 8 points is usually not a good idea. Also, make sure that the text is well written and printed in a color that stands out clearly from the background (e.g., don't use yellow text over an image of a sunset).

Neato's CD Labeler kit includes a Windows- and Macintosh-compatible CD-ROM that has graphic design software and clip art for fashioning your own CD labels.

Now, you've heard this before, but its importance can't be overemphasized: Remember to put your contact information on your demo. No matter how great your music is, you won't hear from anyone if nobody can figure out how to contact you. This might seem obvious, but I've heard stories about music directors who received songs they wanted to use but couldn't figure out who sent them. Make sure the contact information on your demo is always current. If you move around a lot, get a permanent P.O. box and phone number. (Many telephone companies offer voice-mail numbers.) Getting a song placed can take a while—I have friends who've received callbacks from music directors more than a year after submitting something. Lastly, as the demo gets separated from the promo pack, so too does the CD-R from its jewel case. Therefore, it's a good idea to put your contact information on all parts of the CD-R package, including the disc, the booklet, and the tray card—this way, you can't go wrong.

MAKE IT CONSPICUOUS

While contact information and legibility are applicable to all demos, CD-R or otherwise, a few labeling tips apply exclusively to CD-Rs. Over the past year, I've heard more than one person mention the importance of having your song titles listed in order on both the CD-R and the jewel case (either on the back of the booklet or on the tray card). This is in addition to having the song lyrics inside the booklet.

The spine is another part of the demo that's important to label. I realized the value of this when Minnick showed me a shelf in his office that was full of demo CD-Rs without spine labels (some of them were mine). If you're looking for a particular CD-R and it doesn't have a spine label, you need to pull them all out until you find the right one. That can be very annoying, especially if you're in a rush. The last thing you want is to be passed over because a music director couldn't find your CD-R or just plain forgot about it. (Out of sight, out of mind!) Label the spine and make your CD-R more conspicuous, an advantageous trait for a demo.

LABELING KITS

Two popular CD labeling systems are the CD Labeler kit ($79) from Neato and the CD Stomper ($69) from Stomp, Inc. Each of these kits includes a CD-ROM, jewel-case booklets, tray cards, CD-R labels, and a tool for aligning the labels on your discs. The CD-ROMs contain software (Mac- and Windows-compatible) for designing your CD-R, lots of clip art for your pasting pleasure, templates for third-party graphics programs (such as *QuarkXPress* and Adobe *Illustrator*), and various resource templates (e.g., EPS, PICT, TIFF, and BMP).

The design programs are rudimentary but adequate for creating nice-looking inserts and labels. (If you're

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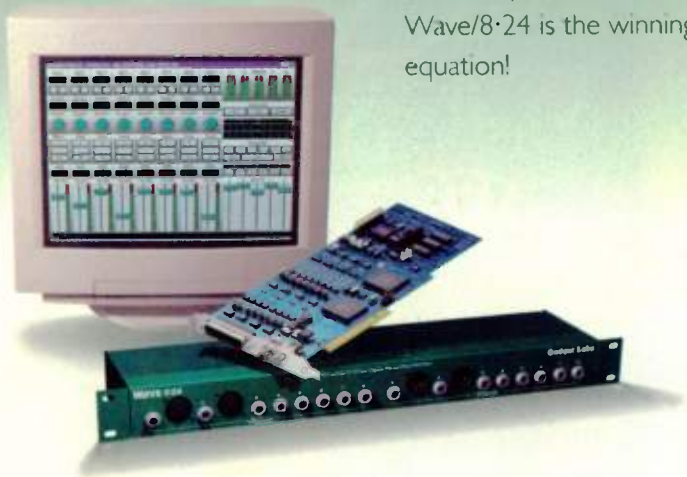

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new to design software, you'll find Stomp's program to be a bit more intuitive.) If you want to create custom artwork and get really tricky with text creation and layout, you should use a dedicated graphics program in combination with the templates; I recommend Adobe's *Illustrator* and *Photoshop*.

There are a few minor differences between the two kits. Neato's jewel-case covers are 2-page booklets (four printable surfaces: front, back, inside left, and inside right), while Stomp's covers are just single-page inserts (two printable surfaces: front and back). By placing the Neato covers inside one another, you can make booklets that contain several pages. Neato's paper stock is a heavier weight than Stomp's, giving Neato's inserts a more expensive feel. Neato's CD-R labels come in a variety of colors: matte white, photo gloss white, mustard, and gray for inkjet printers; metallic silver and metallic gold for laser printers. Stomp's CD-R labels (for inkjet printers only) are matte white, and the company says photo gloss white will be available soon. As for inserts, both companies have matte white; Neato also has photo gloss white. (Again, Stomp says photo gloss white is on the way.) Extra paper and labels are sold separately for $12 to $24, depending on what you need. (In general, Stomp's prices are lower than Neato's.)

Neither product's CD-R label-alignment tool is perfect, but they both get the job done. Neato's tool is composed of a round base (the diameter of a CD and about two inches tall) and a device that looks something like a dreidel (but I'll call it a plunger because that's more descriptive of what it does). There is a hole in the center of the base with the same circumference as the plunger's head. Placing a label on the platform, sticky side up, you line up the label's hole with the hole on the base. You then put a CD-R, face down, over the handle of the plunger. The handle fits snugly into the CD-R's center. This leaves the CD-R resting on top of the plunger's head. Holding the plunger's handle, you drop it headfirst through the base's hole. The plunger aligns the label with the CD-R as it goes through the hole, while the label's sticky back adheres to the CD-R's face.

Stomp's tool works similarly, but instead of the plunger being a separate piece, it's attached to the base via a spring mechanism. (The attached plunger is cool because you don't have to worry about misplacing it.) You put the label on the base and the CD-R on the plunger, and push down. The spring's recoil action pops the CD-R up after the label is stuck on. The bottom of Stomp's tool has a rubber backing, which prevents the device from sliding around—a nice touch if you're working on a slanted surface, such as a drafting table.

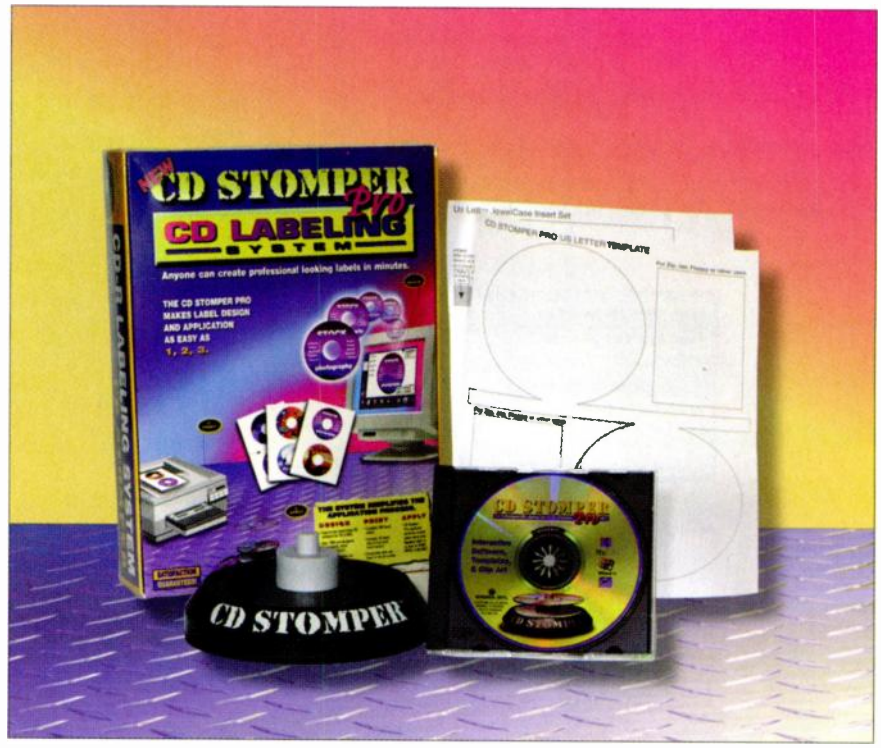
Both tools leave between 1/16 and 1/32 inch of play in the alignment process, so the labels don't always end up perfectly centered. But they get very close. Once the labels are stuck on with the alignment tool, they need to be smoothed on by hand for complete adhesion. Remember not to press too hard and to make sure the CD-R is not sitting on an abrasive surface, or you will damage it.

Always put your labels on *after* burning the CD-R, because the heat generated by the laser during the burning process can melt the label's adhesive and damage your recorder. (The adhesive is designed to handle playback temperatures, which are less intense than burning temperatures.) In addition, the extra weight from a label can cause write errors during the delicate burning process.

One problem I had with both systems was that, once I peeled a label from its backing, it would curl up like a burrito. Because of this, it was nearly impossible to get the label to lie flat on the alignment tool. There is a trick, I learned, to peeling the labels off their backing so that this curling doesn't happen: lay the sheet of labels face-down, and then pull the backing away from the label (rather than pulling the label away from the backing, as I had done initially). After struggling with the curling labels, I tried this approach, and it worked perfectly! (This tip was culled from a call to Neato; neither company mentions this peeling technique in its manual.)

PRINTING RIGHT ON IT

A few companies make printers that print directly on CD-Rs. The two main companies that manufacture these devices are Affex and MediaFORM. The printers are actually modified versions of color inkjet printers made by Epson. Affex's SK7 ($1,695) uses Epson's Photo 700, and MediaFORM's CDP-CP2 ($1,495) uses the Color 600. (At press time, MediaFORM was planning to discontinue the CDP-CP2 and had not yet introduced the new model intended to replace it.)



The CD Stomper system from Stomp, Inc. allows you to label your CD-Rs and design single-page jewel-case covers.

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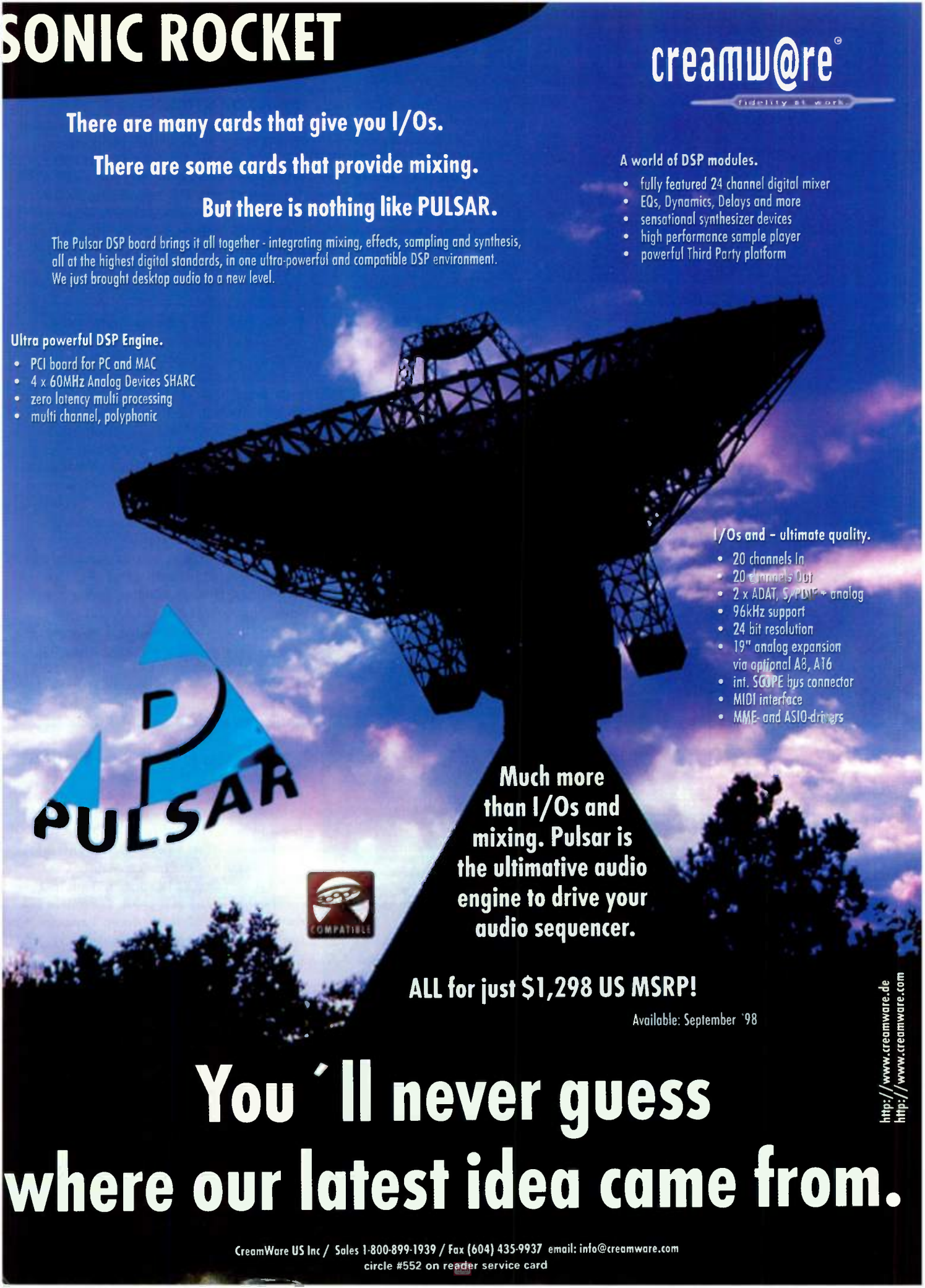

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In order to do this type of printing, special CD-Rs must be used. They are more expensive than regular CD-Rs, averaging at least a dollar more per disc. The face of a printable CD-R is covered with a textured surface similar to photo gloss paper. You'll find printable CD-Rs with both white and gold finishes. The white finish is preferable because the gold bleeds through certain colors.

Printable CD-Rs vary widely in quality, with some yielding better prints than others. Questionable finishes are identified by off-white or dull surface coatings and ragged edges along the inner and outer perimeters of their printable surfaces. I was unable to test all the available printable CD-Rs, but in general, I did notice that the cheaper the CD-R, the more likely it is to have a low-grade surface coating. The three brands I tried, from least to most expensive, were CMC, Mitsui, and Ricoh New Media. The Ricoh yielded the nicest print (nearest to photo quality) and the CMC the worst, with the Mitsui's quality somewhere between the two.

PRINT OR STICK?

Inkjet CD-R printing systems are cool, but they are an expensive proposition. In all honesty, I'm not sure they're worth the extra bucks. You can pick up a regular Color 600 Epson printer for $249, along with a Neato or Stomp kit, and get basically the same results. Why spend $1,700 (the retail price for the Affex printer), plus extra money for printable CD-Rs, just to avoid sticky-back labels?

One argument is that printable CD-Rs offer a more photo-quality finish than regular paper labels. However, now that photo gloss labels are available, this argument doesn't stick (so to speak). In fact, because the printable CD-Rs' finish isn't actually photo quality, you get nearer to photo quality with the new labels (though it's still not as good as actual photo gloss paper, such as Hewlett-Packard Premium Inkjet Glossy Paper). Then there are the arguments that labels are difficult to put on (i.e., difficult to center) and that they might eventually peel off. I've been using Neato's labeling system for more than a year; my CD-Rs have been all over the world, and so far I've never had a problem with a label coming off. However, the photo gloss labels are brand-new, and it is difficult to formulate an adhesive for such slippery stock. Only time will tell how well these labels hold up.

As for putting labels on, the process is a little tedious, so if you're running more than 50 CD-Rs a week, a CD-R printer might be a good idea. In addition, if you need to burn CD-ROMs, it's essential to have a perfectly weighted CD-R because of the high speeds of CD-ROM drives (e.g., 24× speed). A slightly off-center label may cause a problem during playback. But this doesn't matter for audio CDs because they don't require high-speed drives.

Lastly, printing an image that completely covers the surface of a printable CD-R presents a problem. In order to create such an image, you need to make a design that is a bit larger (at least a sixteenth of an inch) than the actual face of the CD-R. This overlap area is necessary because precise print alignments are virtually impossible with inkjet printers. The ink from the overlapping design bleeds from the nonprintable surfaces (the extreme edges and the center) onto the printable surface. The bleeding results in

For manufacturer information, please see Contact Sheet, p.163.

a jagged, oversaturated area that appears along the outer and inner edges of the printable surface (about a sixteenth to an eighth of an inch thick). It is especially noticeable with lighter colors and on CD-Rs whose outer and inner printable edges are ragged, so it's best to steer clear of designs that require an overlap. With CD-R labels, this problem never arises because the entire label sheet is printable. Simply peel the label out and you get sharp, clean, edges.

A FEW MORE TIPS

In addition to jewel cases, there is a variety of packaging options for CDs. Most of these alternative CD holders were originally meant just for CD-ROMs, including flat cardboard mailers, clear binder inserts with CD-sized pockets, and clear seashell CD holders (a plastic contraption that holds the CD snugly in its center until you turn it upside down and squeeze its sides, which makes the CD fall out). Even though holders like these are quite eye-catching, they can be a hassle. They don't fit on a shelf with other jewel cases, and, as a result, they get lost in a stack of CDs. In addition, there usually isn't enough space for photos and text. So your interests are probably best served by using a regular jewel case.

▼

A professional-looking demo package sends the message that you have your act together.

Sticking with traditional jewel cases doesn't mean your CD has to look boring. Most office supply stores now carry a variety of empty jewel cases, from clear to neon-colored, and they usually cost about a dollar each. The clear cases offer some great opportunities to get creative with tray cards. Some jewel cases have textured covers that generate double images in combination with the booklet's cover picture. (Remember those cereal-box prizes that had "3-D" pictures, whose images moved depending on your viewing angle?)

Unwrapping things is always fun (even for A&R people), but most of us don't own our own shrink-wrap machines to go along with our CD-R burners. An alternative is to create your own "assurance seal" (like those that store-bought CDs have on their edges to prove that they're new) using clear mailing labels, available at office supply stores. Print your company logo or band name on the labels and use them to seal the edges of the jewel cases.

Finally, even though the labels that come in the CD-R labeling kits are essential, you don't have to use the provided booklets and tray cards. If you want heavier stock, different colors, or custom finishes (e.g., marbleized paper), just use the provided templates with your own paper. Granted, the perforated edges on the stock booklets and tray cards make them easy to tear out and fold, but it is often difficult to line up your artwork with them. (The templates don't line up with your printer's output without some serious nudging.) Using your own paper alleviates this problem because you cut the booklets and tray cards after the printing is done; you just have to be very careful to cut them to the precise measurements.

DON'T MESS AROUND

You've put so much time and energy into your music, it seems a shame to release it in a shoddy package—so don't. With about a $100 (not including a printer) and a little extra effort, you can now cook up a great CD-R package, complete with pictures and a booklet. Show the powers that be that you mean business; do yourself and your music right by presenting yourself professionally.

Finally, although I've expounded on the virtues of a great packaging job, don't think for a minute that a super-cool packaging job can make up for lame music. The objective is to enhance your music, not distract from it. The images and text on the package are simply there to inform and entice the listener; your music must make the final sale. We all know that the music business is a gamble. Putting the time and energy into your CD-R's presentation helps you hedge your bets.

Erik Hawkins *is a musician and producer in Los Angeles County and the San Francisco Bay Area. You can check out his music at www.muzicali.com.*

REVIEWS

PANASONIC/RAMSA

WR-DA7

A powerful new digital mixer puts musicians in the driver's seat.

By Barry Cleveland

Digital mixers are probably the hottest new products for the personal studio, and the debut of Panasonic/Ramsa's WR-DA7 digital console has been eagerly anticipated. However, Mackie, TASCAM, and Yamaha already are shipping digital mixers for the personal studio, so the DA7 has lots of competitors. Furthermore, Spirit by Soundcraft is preparing to release its new digital board in the U.S., and Generalmusic recently announced two digital consoles for release late this year. Clearly, this is becoming a buyer's market.

To be successful, then, a digital-mixer manufacturer had better do the job right, and indeed, Panasonic's design reflects a lot of careful consideration by people who have obviously recorded a lot of music. (Ramsa, by the way, is a division of Panasonic that is known for its mixers, while Panasonic is better known for professional-quality digital technology, so the company decided to use both names on this product, at least for the American market.)

At $5,000 for a basic unit, the Panasonic board would appear to plot a course right down the middle of the price range for affordable boards. That said, you will almost certainly want to add several options that increase the price significantly. Once you have added these options, though, the console

Panasonic/Ramsa's WR-DA7 digital mixer delivers great sound, thanks to quality mic preamps; transparent, flexible EQ; 24-bit converters on most inputs and outputs; and clean design all around. The user interface is friendly, and many expansion options are available.

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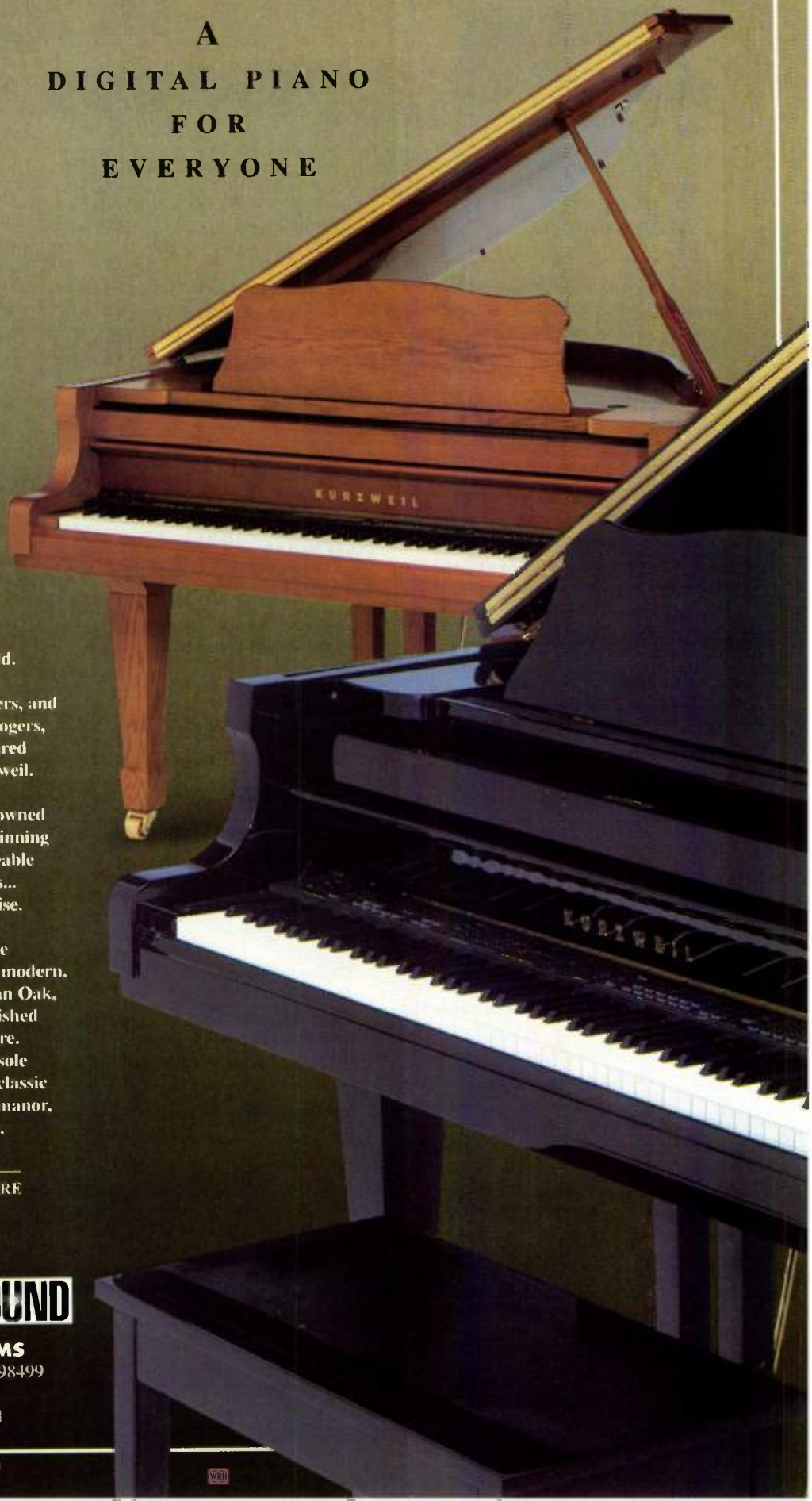

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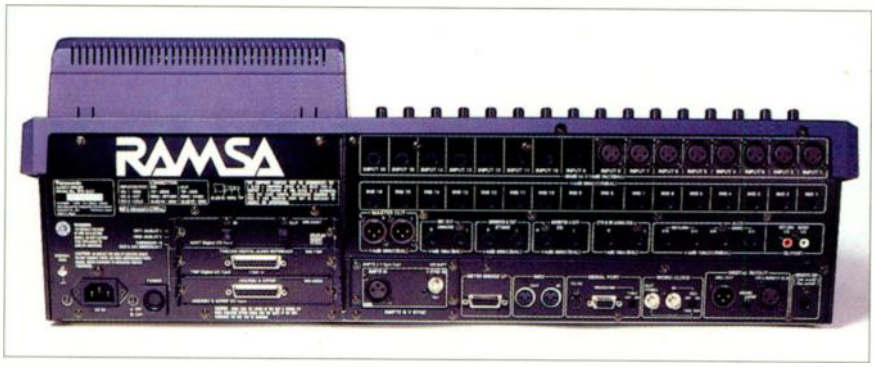

The DA7's top row of connectors includes XLR inputs for channels 1 to 8 and balanced, ¼-inch inputs for channels 9 to 16. Row 2 holds the channel inserts, while row 3 contains the outputs and aux buses. The bottom row sports the meter-bridge port, MIDI ports, serial ports, word-clock I/O, digital Record Out, digital 2-track In, and the footswitch jack. The expansion slots are on the left.

can compete with boards costing several thousand dollars more.

THE STOCK MACHINE

The DA7 is essentially a 32-channel, 8-bus mixer, with six aux sends and returns for a total of 38 inputs. Just what goes into and out of those channels is determined by the optional I/O cards you put in the console's three I/O-card slots. Before we get into the options, though, let's examine this board in its stock condition.

This is an extremely powerful machine, boasting 32-bit, floating-point processing. The sampling rates are switchable between 44.1 kHz and 48 kHz. The sixteen analog-input channels, the master output, and both monitor outputs have 24-bit converters, while the four analog aux sends and returns use 20-bit converters. All of the converters sound very good; to my ears, they are superior to those found on most MDMs and digital workstations. As a valuable bonus, onboard redithering to any resolution from 23 to 16 bits is provided on all digital outputs.

The first sixteen channels may be thought of as the analog-input channels, but as we shall see, this is not strictly accurate. Channels 1 to 8 have balanced, XLR connectors and individually switchable phantom power but no phone jacks. Channels 9 to 16 have balanced, ¼-inch TRS phone jacks but no XLR connectors. Other than that, these sixteen channels are identical and include TRS insert jacks. Despite the difference in connectors, all analog-input channels accept mic or line inputs from -60 dBu to +24 dBu.

Stereo digital input (switchable AES/EBU or S/PDIF) is provided on an XLR connector. These digital inputs can be routed to channels 15/16 for mixing, to 2-track A/monitor A for monitoring, or to the L/R bus.

The six aux sends and returns are arranged in groups of two. Auxes 1 and 2 are S/PDIF digital, allowing you to connect an outboard processor without leaving the digital domain. Auxes 3/4 and 5/6 are on unbalanced, ¼-inch stereo phone jacks and require splitter cables. Although inconvenient, this conserves valuable real estate on the rear panel. Unfortunately, you cannot route aux returns to aux sends, making it difficult, for instance, to feed effects into a headphone mix taken from an aux send. However, you can accomplish the same thing by routing the signal back to an input.

The master stereo mix is routed to analog and digital outputs, both of which are on XLR connectors. The digital main outputs can be switched between AES/EBU and S/PDIF formats, but the latter requires an XLR-to-RCA adapter.

WR-DA7 Features

Number of Input Channels (stock/fully expanded)	16 + 6 aux/32 + 6 aux
Faders	(21) 100 mm motorized, + 10 dB to -∞ dB
EQ	4-band parametric on 32 channels, 8 buses, and master outs; 2-band parametric on aux returns. Low and low-mid: 20 Hz to 20 kHz. High-mid and high: 500 Hz to 20 kHz
Dynamics Processors	(42) on 32 input channels, 8 buses, and master stereo mix out
Memory	(50) snapshot scenes, (50) channel library, (50) EQ library, (50) dynamics library, (4) automation mixes
Display	320 x 240-pixel, backlit LCD
Level Meters	(32) bar graph (assignable to inputs 1–32, bus/aux, and slots); (2) master output
Channel 1–8 Inputs	(8) balanced XLR
Channel 9–16 Inputs	(8) balanced ¼" TRS
Channel 17–32 Inputs	accessed via optional cards
Master Outputs	(2) XLR balanced/unbalanced
2-Track Analog Input/Output	(2) balanced ¼" TRS/(2) balanced ¼" TRS
Monitor Outputs	(4) ¼" balanced (monitor A & B, L/R)
Digital 2-Track (Ch. 15-16) In	(1) XLR, stereo AES/EBU and S/PDIF (hardware switchable)
Digital L/R Record Out	(1) XLR, stereo AES/EBU and S/PDIF (hardware switchable)
Digital Aux 1-2 Send/Aux Return	(1) RCA in/(1) RCA out, stereo S/PDIF
Other Ports	MIDI In and Out; (1) RS-422/485 PC serial interface; (1) DIN8 Mac serial interface; meter bridge connector; (2) BNC word-clock In and Out/Thru ports; (1) ¼" footswitch jack
Dimensions	27.5" (W) x 9.6" (H) x 21.6" (D)
Weight (stock)	51 lbs.

For manufacturer information, please see Contact Sheet, p. 163

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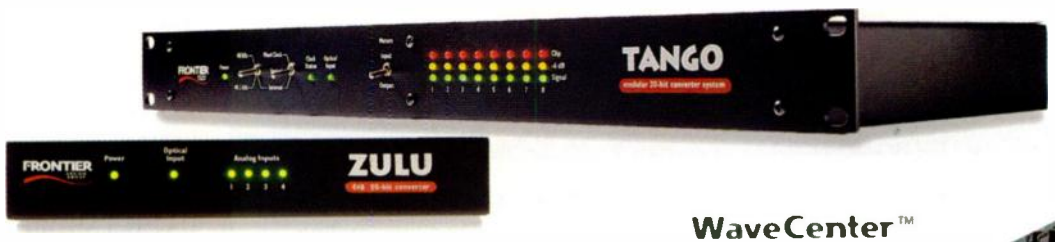

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Balanced, ¼-inch, TRS jacks are provided for Record L/R, monitor A, monitor B, and 2-track B analog outputs.

Up to 300 ms, or 14,400 samples, of delay is available on all channels. The delay can be used for very basic effects, but if you want to tweak the delay time in real time, you must adjust the samples parameter—not the milliseconds one—or you'll get an audible pop.

The stock DA7 has several extra amenities in the form of an RS-422/485 serial port for interfacing with a PC, an 8-pin DIN serial port for Mac, and word-clock I/O on BNC connectors. Of course, you get a headphone jack, which is conveniently located on the front corner; the headphone amp sounds okay, but not as good as some I've used. The rear-panel footswitch jack accepts a momentary switch (not included) that can be used to control the talkback function or for punching the automation in and out. By the way, talkback can be latched or unlatched and can be routed to slate, monitor B, or all auxes.

THE LOADED MACHINE

For most users, outfitting a DA7 to taste means adding optional expansion boards. For example, channels 17 to 32 can only be addressed via an I/O card. The good news is that you have a lot of options, and with an expandable system, you only buy the extra features you need.

As noted earlier, the first eight analog channels have XLR inputs, and the second eight have ¼-inch inputs. If you want more than eight XLR or ¼-inch phone connectors, you must use adapters or pay $799 a piece for 8-channel, AD/DA cards. That's a lot of bread, but you add a lot of functionality: not only does the AD/DA card provide analog inputs for channels 17 to 24 or 25 to 32, its outputs can be configured as analog bus outs (which the DA7 does not have otherwise). Aside from interfacing with outboard processors, this is the way to go if you wish to use a digital mixer with an analog multitrack recorder or for sound-reinforcement. It's important to note that these cards require breakout cables.

In addition to the AD/DA card, you can get digital I/O cards in ADAT Optical ($295), TASCAM TDIF ($350), and AES/EBU (hardware-switchable to S/PDIF; $360) digital formats. Like the AD/DA card, the AES/EBU card requires breakout cables. The DA7's ADAT Optical card can pass 24-bit digital audio, thanks to Panasonic's custom controller chip. This is noteworthy because, although the ADAT Optical specification is 24-bit, some other controller chips that are used in many Lightpipe devices can handle no better than 20-bit resolution. By designing its own chip, Panasonic has upped the ante for ADAT transfers.

Any combination of cards can be placed in the DA7's three I/O-card slots. However, signals coming in via the third slot are routed to channels 9 to 16, replacing the analog inputs. Signals can be routed from one card to another, facilitating conversion between digital formats (e.g., ADAT to TDIF).

Your options are not limited to I/O cards. Two DA7s can be cascaded with complete bidirectional functionality, using a pair of Tandem connection cards that are inserted in slot 3 of both mixers. The downside is that you have to buy one card for each mixer (whereas Yamaha, for instance, gives you both 02R or 03D interface cards in one package, and the 01V doesn't even require a card).

A fourth, differently shaped, card slot admits Panasonic's SMPTE/V card ($495). This card equips the mixer to accept SMPTE time code on an XLR connector and to receive a composite video signal (NTSC or PAL) on a BNC connector and use it to drive the internal word clock.

Although the DA7 can display input levels for channels 1 to 32 using tiny bar graphs on its Meter window (see **Fig. 1**), sooner or later you will probably want to add the optional meter bridge, which will cost you $1,000.

Building a loaded system, then, can add up to a good bit more than the base price. To get a 16-track ADAT system up and running with dynamic automation and meters larger than your thumbnail, you are looking at $6,600 plus external sync hardware. If you would like to have the remote-control software necessary to operate your DA7 with your computer, you'll need to also purchase the WR-RC/M (Mac) or WR-RC/W (Win) package, for $495.

However, as of this writing, the control software isn't shipping. Panasonic informed me that when it does arrive, the software will enhance the DA7 in a number of ways, providing custom views (closer to an analog layout), a librarian for cataloging patches and scenes, and exapanded automation features. Keep in mind, though, that the DA7 is a MIDI-controllable device, so sooner or later, someone will hack a slick control panel for a digital audio sequencer or write a *MAX* patch that will do the trick for free.

SOUND ALL AROUND

Panasonic has equipped the DA7 for surround sound, allowing you to reconfigure the buses to output 5.1 or LCRS signals. If one of the optional I/O cards is used, the surround signals can be sent directly to your digital recorder, though a more typical scenario would be routing them to an external surround processor.

WR-DA7 Audio Specifications

A/D Converters	Inputs 1–16: 24-bit, 64x oversampling Aux send 3–6: 20-bit, 64x oversampling
D/A Converters	Master, monitor A: 24-bit, 64x oversampling Monitor B: 24-bit, 128x oversampling Aux returns 3–6: 20-bit, 128x oversampling
Frequency Response	20 Hz–20 kHz (+1/-2 dB)
Dynamic Range	110 dB typical, AD + DA (analog in to analog out)
THD (input gain minimum)	<0.1% (input = +10 dB/1 kHz; output = +4 dB/RL 600Ω)
Equivalent Input Noise	-128 dB (Rs = 150Ω, input sensitivity = 60 dB typical)
Crosstalk (adjacent channels)	90 dB typical (1 kHz)
Sampling Rate	44.1 or 48 kHz (switchable)
Internal Processing	32-bit (192 dB internal dynamic range)
Signal Delay	<2.5 ms, mic/line input to master out

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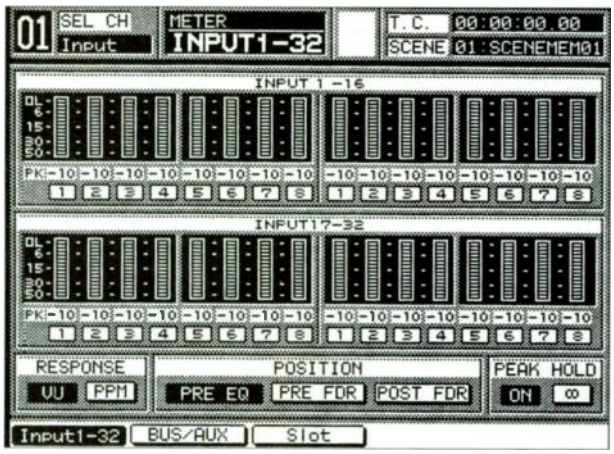

COURTESY PANASONIC

FIG. 1: The DA7 shows audio levels in three Meter windows. Shown here is the window for channel inputs 1 to 32; another window displays levels for the buses and aux sends and returns, and a third window monitors the option-card inputs.

Surround mode is selectable for individual channels, and signals can be panned in one of three ways. You can use the jog/shuttle wheel and the master fader; the pots in the EQ, dynamics, and panning sections, which do double-duty; or the pattern generator, which allows you to set a dynamic panning velocity and choose from three dynamic surround-sound placement patterns (a circle, a straight line, and an arc). Surround panning can be MIDI-controlled on the DA7 with standard MIDI messages, but not directly—the console is designed to work with JLCooper's MIDI Controller, which handles the translation.

VISITING THE Q CONTINUUM

The DA7 has 4-band parametric equalization on each of the 32 input channels, eight buses, and master outputs. The EQ is extremely versatile and easy to use, and it sounds very pleasing and musical. It does not have much personality (read: coloration), like some analog EQs have, but in my estimation that is not a bad thing.

The four bands are similar in range: the low and low-mid bands can be swept from 20 Hz to 20 kHz, and the high-mid and high bands span 500 Hz to 20 kHz. The low and high bands can also operate as highpass/lowpass and low-shelf/high-shelf filters. Two-band EQ, switchable to highpass and lowpass filters, is available on the aux returns.

The Q can be adjusted from a pinpoint, for surgical accuracy, to extremely wide, covering a range of several octaves. EQ curves are shown on an instantaneously updated graphic display on the EQ window (see **Fig. 2**), and 50 EQ programs can be stored in the onboard EQ Library. There are no factory presets, as on some other digital mixers, but those are arguably of limited use to most engineers, anyway.

DEALING WITH DYNAMICS

Dynamics processing is available on all 32 input channels, the eight buses, and the master outputs. Up to 50 dynamics settings can be stored in the onboard Dynamics Library. Each dynamics processor can be used as a compressor/gate—the two functions can be used simultaneously—or expander and can be positioned pre- or post-EQ. There is no dedicated limiter, and I was not thrilled with the results I got when using the compressor as a limiter.

Threshold, ratio, attack/release times, and gain are all adjustable, but unfortunately, there is no soft-knee response. There is also no sidechain (key) function, and consequently, no ducker, which is an odd oversight. It would also have been nice to have some sort of analog-compression emulation, but you have to draw the line somewhere.

I found the compressor to be quite useful on individual channels and on the master mix, though I had to use different time values than I would have on an analog compressor, in order to get similar results. The gate is pretty good, but it is sometimes slightly clumsy when dealing with subtler sounds. The expander worked just fine, even when I used fairly radical settings.

WONDERS OF AUTOMATION

The DA7 features snapshot (scene) automation with programmable crossfade times between scenes for each channel. You also get full dynamic automation with mechanized faders. Up to four automated mixes (depending on available memory) can be stored internally, and up to 50 scenes can be stored in the onboard Scene Library.

Scenes include all mixer settings except the analog trim pots and analog switches. The dynamic automation allows you to recall presets from the scene and other libraries, and to record all fader movements, including aux send levels and bus levels, as well as mute status, dynamics processor on/off, EQ changes, channel on/off, and panning/surround moves. You can also automatically recall channel, EQ, and dynamics libraries. A Select/Manual parameter decides whether automation changes are recorded when you make a manual change on a given channel. A handy Undo feature allows you to return to your previous automation settings.

The automation synchronizes to MIDI Time Code, MIDI Clock, SMPTE time code (with the optional SMPTE/video-sync card), or internally generated time code. The automation memory can store up to 32,000 events; by the time you read this, the controller software should be available, with which automation memory is limited only by available hard-disk space. The current memory capacity is sufficient for most mixing tasks, especially if simple conservation strategies are employed.

Offline event-list editing is thorough and easy to use, with filters available to zero in on specific events. Automation data and all library data can be transferred to a MIDI librarian program, or to another DA7, via MIDI Bulk Dump.

You will need to have a BRC controller or other time-code generator to synchronize an Alesis ADAT to the DA7's dynamic automation, but once you're so equipped, the two devices should get along fine.

I used the SMPTE/video-sync card to synchronize the DA7 to my Fostex G-16S analog 16-track recorder. I used the G-16S's SMPTE time-code generator to stripe time code onto track 16 and routed track 16's output to the DA7's SMPTE sync input. The mixer locked up beautifully, and at no time did I experience any wobble, offset creep, or other funny business.

FUN WITH FADERS

The mixer's 21 100 mm, servo-controlled, moving faders are arranged in three groups: sixteen channel strips, four buses, and the master fader. All except the master fader can be switched between four "layers" of control with the push of a button.

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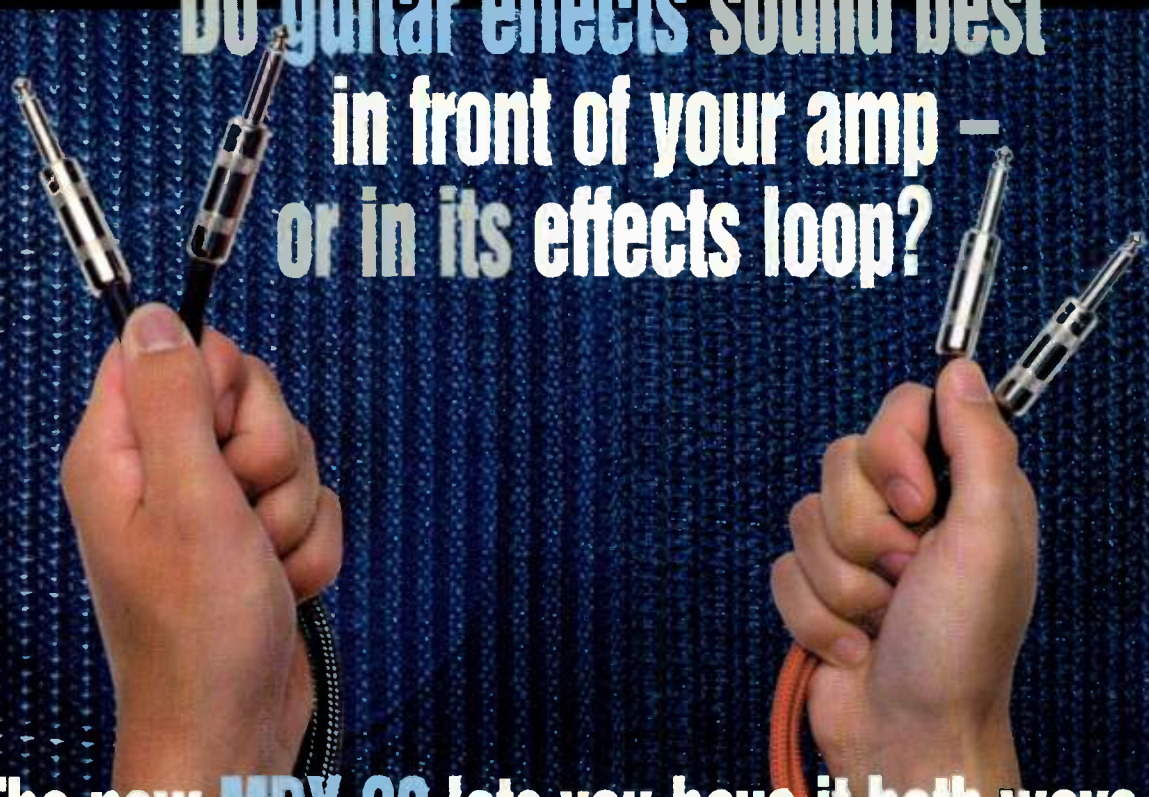

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The first layer assigns the faders to analog-input channels 1 to 16 and the four odd-numbered buses. Selecting the second layer assigns them to the four even-numbered buses and channels 17 to 32, which are activated when 8-channel I/O cards are installed in slots 1 and 2. The third layer assigns the faders to the six aux sends, six aux returns, and the eight buses. The fourth layer allows you to custom-configure each fader independently: you can have any fader serve any function implemented in the other layers. Fader groups 1 to 8 and 9 to 16 can be switched to send MIDI messages on any MIDI channel. Very cool! Faders can also be assigned to any of four VCA-style fader groups and four mute groups.

Faders assigned to adjacent even/odd-numbered channels can be linked in either of two ways: with regular linking, each channel retains its original setting but changes are done in tandem; with stereo linking, the odd-numbered channel's settings—gain, EQ, dynamics, mute, etc.—overwrite the settings of the even-numbered channel.

When the board is in dynamic-automation Record mode, a preference allows you to merely move a channel fader to punch in, but you have to push the channel's Select button to punch out. A programmable Fadeback function returns the fader to its prepunch level in a graceful manner. Relative Trimming raises or lowers overall level while maintaining the movement curve.

Fader movement is fairly firm yet smooth, but paired faders have a somewhat spongy feel that was not eliminated by calibration. Speaking of fader calibration, it is easily accomplished by simultaneously pushing two buttons. The faders do a little calibration dance and then return to their previous positions.

EASY TO LOVE

Getting around the user interface is easy. Parameters can be easily selected and adjusted using the four cursor buttons, jog/shuttle wheel, and numerical keypad. (The DA7 does not have a mouse port.) The cursor buttons also serve as MIDI Machine Control transport controllers.

Controls for the various functions are arranged logically and ergonomically,

PANASONIC/RAMSA
WR-DA7 digital mixer
$5,000

FEATURES ■■■■
EASE OF USE ■■■■▮
AUDIO QUALITY ■■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Intuitive user interface. Excellent overall sound quality. Great 24-bit A/D and D/A converters. Onboard dither. Footswitch-controlled talkback system. Expandable, with lots of options. Extensive MIDI implementation. Dynamic and scene automation.

CONS: Limited onboard automation memory. Only eight XLR and eight ¼-inch analog inputs. No analog bus outputs without option card. Requires numerous options to be configured for most uses. No onboard effects. Limited aux I/O and routing. Control software is overpriced.

CIRCLE #437 ON READER SERVICE CARD

with the sixteen channel-fader strips occupying the center and left side of the console and the remaining five faders, the Display Bridge, and controls on the right side. Note that all of the controls are on the right, which is not so great for southpaws. The channel strips have nonprogrammable, analog trim pots on top, with spaces for tape strips just below them. Having a separate space to label what's coming into the analog inputs is a very thoughtful touch, especially because keeping track of what's on the fader layers can sometimes get confusing.

The channel Peak/Signal LEDs glow green when signal is present and red just before clipping. Six LED Status Indicators toggle between showing which aux sends are active and monitoring the dynamic-automation record/playback status. Automation mode has priority over Aux mode, and when the DA7's dynamic automation is in Record, LEDs flash red for all automated functions on all selected channels. You get a nice light show when lots of parameters are being recorded on lots of channels! The top row of LEDs flash green when dynamic automation is in Playback mode.

Four LED-equipped buttons show Solo, Flip (for toggling individual channels between the two top mixing layers), Channel Select, and Channel On status. These buttons have additional functions, such as calling up screens on the LCD and pairing channels. Finally, you get a second area for labeling tape, just above the channel faders. Up to 50 complete channel-strip settings (except for the analog trim pots) can be stored in the onboard Channel Library.

Once you have selected a channel, it is easy to make adjustments to its EQ, dynamics, delay, aux assignments, and bus assignments. For example, to assign a channel to the master outs, a direct output, or any of the eight buses, you simply push one or more of the corresponding buttons in the Pan/Assign/Surround section. As the name suggests, you can also make panning and surround panning adjustments in this section. Aux assignments, EQ changes, and dynamics changes can be made just as easily.

Pushing data-encoder knobs in these sections causes the screens related to their various functions to appear on the LCD. The mixer also offers a preference that allows you to call up the appropriate screens when any adjustments are made.

At the heart of the DA7's control surface is the raised Display Bridge section, which features a 320×240-pixel, backlit LCD that looks remarkably similar to those found on the Yamaha 03D and 02R mixers. Directly associated with the LCD are two buttons that make up the Master Display section. Pushing the Meter button cycles

WR-DA7 Add-On Options

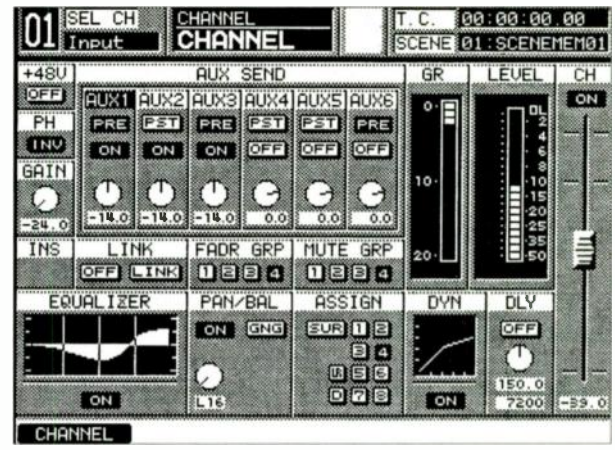
AD/DA card (8-ch., ¼" or XLR)	$799
ADAT Optical card	$295
TDIF card	$350
AES/EBU S/PDIF card	$360
Tandem (cascade link) card	$539 ea. (2 required)
SMPTE/video-sync card	$495
Meter bridge	$1,000
Remote-control software	$495

through the Input 1-32, Bus/Aux, and Slots screens, where input levels are shown on tiny bar graphs.

Input levels can be monitored pre-EQ, prefader, or postfader, with a choice between VU and PPM response. Peak hold (selectable between 0.3 seconds and infinity) is an additional option. The overlap of the Input 1-32 and Slot screens is apparently for convenience, so no matter how the inputs are configured, they can be viewed from a single screen. Of course the most convenient way to monitor the DA7's input levels is by using the optional meter bridge.

Pushing the Channel button in the Master Display section recalls the Channel window (see **Fig. 3**), which displays the current status of the selected channel. This is "command central" for all channels, where you can make adjustments to all of the channel's primary functions. This screen tells you just about everything you need to know at a glance. Despite the massive amount of information presented here—including graphics for EQ, dynamics curve, gain reduction, and input level—it is organized in a clear and easy-to-comprehend arrangement. The screens for the DA7's other functions are similarly organized.

Alongside the LCD are a knob for adjusting its contrast, large master-output meters, a numerical display showing the number of the current scene memory, LEDs indicating Solo and Console Lock status, and a button that selects Multi Channel View. Pushing the Multi Channel View button brings up a split window with a condensed version of the Channel window on the left (sans graphics) and an identical window on the right that can be assigned to any channel, bus, aux send/return, or the master outs. From this window, you can quickly copy all channel settings from one channel to another, all aux send settings from one aux send to another, and so on. Similarly, pushing the Multi Channel View button while viewing the EQ, Dynamics, or Surround screens for a selected channel (or bus, send/return, etc.) calls up a split window with condensed versions of those screens on both sides. This useful feature allows you to cut to the chase quickly and easily when you want to copy settings from one place to another without having to access the individual library screens.



COURTESY PANASONIC

FIG. 3: A variety of settings can be made within the Channel window. Here, aux send 1 on Channel 1 has been selected, turned on, and set as prefader.

MIGHTY MIDI, MUDDY MANUAL

The DA7 features an extensive MIDI implementation. It can send MIDI messages via faders, buttons, and pan controllers from its Custom/MIDI fader layer, and all automated parameters can be controlled with a MIDI sequencer. You can also back up information to external storage devices via bulk dumps, output MMC commands from the cursor buttons and other controllers, and more. Unfortunately, the mixer cannot route incoming MIDI Clock to its MIDI Out.

However, the manual leaves something to be desired. It is generally helpful, and most of the information you need is included, but it is spotty in places and would have benefited from better organization and additional proofreading. Fortunately, considerable support—both from Panasonic staff and from fellow users—is available at the unofficial Panasonic/Ramsa WR-DA7 User Group Web site (www.nc17.org/ramsa).

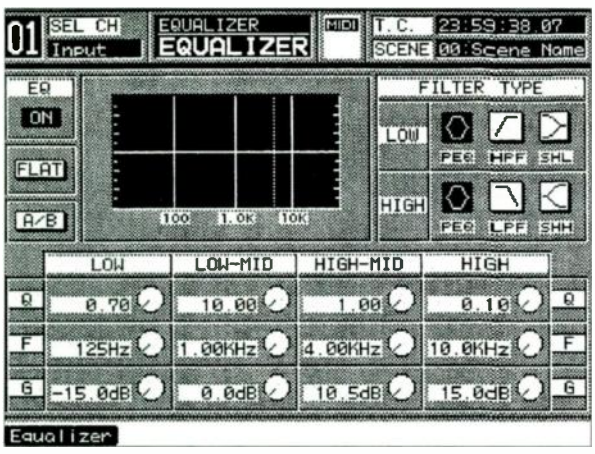


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FIG. 2: The Equalizer window shows the Q, frequency, and gain for the 4-band EQ. The bands can be fully parametric, highpass, lowpass, or shelving filters.

A TIGHT MIX

The DA7 has a spacious, clean, and exceptionally transparent sound. The mic preamps sound remarkably good, particularly for a digital mixer at this price. The board is jam-packed with features, is relatively easy to operate, and is flexible enough to expand with your needs. If you are inexperienced in working with digital mixers, you will have to do some homework, especially when it comes to learning the many "power user" shortcuts and alternative ways of doing things. But your efforts will be repaid many times over once you learn your way around.

Although I have leveled a number of criticisms at the DA7 during this review, and was disappointed with the stock unit's lack of certain features (analog bus I/O and onboard effects, for example), this mixer is definitely a strong contender in the digital mixer market. It's an excellent choice for anyone who wants to mix sixteen channels of digital audio, and it can be configured to handle analog recording, sound reinforcement, and even 24-track digital recording if you are willing to make some compromises that I consider acceptable.

In short, this product deserves to be wildly successful. If you are shopping for a digital mixer, you owe it to yourself to give the DA7 a long, hard look and listen.

When he isn't playing guitar in his band, Cloud Chamber, **Barry Cleveland** *is the editor of the* Mix Master Directory, *the* Recording Industry Sourcebook, *and* **EM***'s* Personal Studio Buyer's Guide.

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AUDIO-TECHNICA

AT3525

An affordable, versatile, and great-sounding condenser mic.

By Brian Knave

Audio-Technica has introduced several new condenser mics this year, but the especially good news for personal-studio owners is the economical 30 Series, which includes the AT3525, as well as two "pencil" condensers, the omnidirectional AT3527 and the cardioid AT3528.

The AT3525 is a side-address, electret-style, cardioid condenser mic designed for both studio and stage applications. (The element in an electret has a permanent charge—in this case, applied to the backplate; a "true" condenser element requires polarization voltage from an external source.) With its ¾-inch–diameter diaphragm, the AT3525 might be more properly termed a medium-diaphragm microphone rather than a large-diaphragm mic, as "large" typically refers to diaphragms that are one inch or more in diameter.

I tested the AT3525 extensively in a variety of studio applications, and I also tried using it at a local club gig. Here's the scoop.

SMALL FRY

The AT3525 is a small microphone—only four and a half inches long—with a gun-metal gray finish and a sturdy wire-mesh grille. It's a cute microphone, but it's made with a quality of craftsmanship that clearly shows that it's no toy.

On the bottom of the AT3525 are two small switches: one to activate a 10 dB attenuation pad and the other an 80 Hz highpass filter. The switches are easy enough to move with a thumbnail, and their location helps prevent accidental switching. The mic's XLR connectors are gold plated.

A matching shock mount is included with the AT3525. The shock mount is simply and sturdily constructed, employing two rubber bands stretched triangularly over either end of a metal cylinder. A U-arm holds the cylinder in place via two finger bolts and connects to a mic stand on the other end. The microphone is easily inserted into the shock mount, and the design of the mount allows for quick, secure, and—in the studio, at least—versatile positioning of the mic.

Also included with the AT3525 is a black vinyl zipper pouch for storage. Actually, this is just a standard handheld-mic pouch, and the AT3525 barely fits inside it. If I owned this mic, I would more likely keep it in the cardboard box it comes in, which is lined with foam rubber cut specifically to accommodate the mic and custom shock mount.

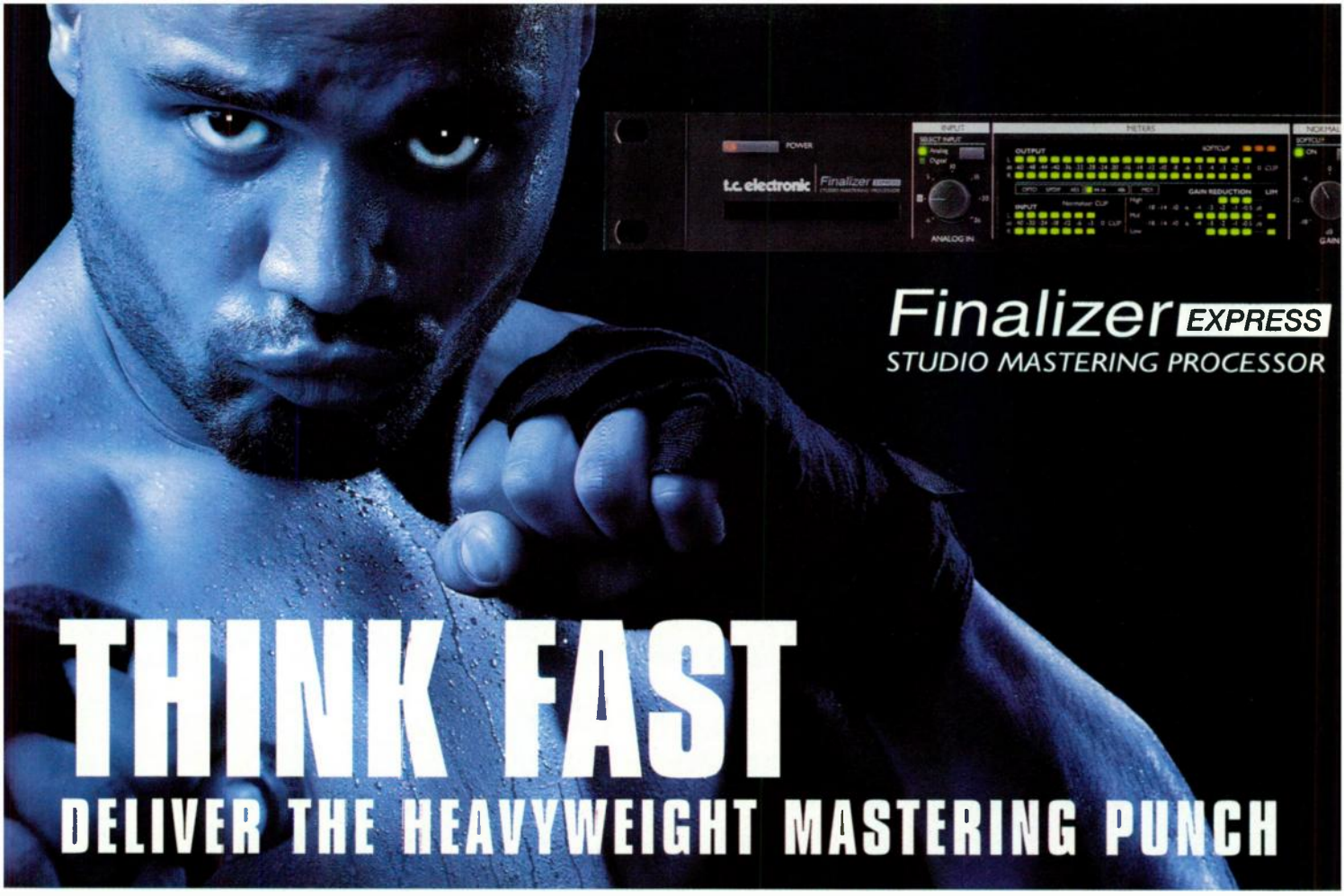
HANDY DANDY

In a 6-week period, I recorded more than twenty different instruments with the AT3525, including both male and female vocals, acoustic and electric guitars, Andean pan pipes and wood flutes, harmonica, and various percussion instruments. I also used the mic on a drum set, both on kick drum and as an overhead. I tested the AT3525 with different mic preamps, too, including a dbx 1086, A.R.T. Tube MP, and the stock preamps in my Mackie 8•Bus board.

The AT3525 proved very versatile. It sounded good to exceptionally good in each application, excelling on acoustic strings, vocals, and percussion. Despite its medium-sized diaphragm, this mic has a big sound—you'd be hard-pressed to tell from listening that it's not a large-diaphragm condenser. Overall, the sound is detailed and very present, with a bright but relatively smooth high end and mildly rolled-off low mids.

According to the manufacturer's frequency-response graph (see **Fig. 1**), the AT3525's response is virtually flat from 200 Hz up to 2 kHz. A mild "presence boost" starts at 2 kHz, peaks by 4 dB at 4 kHz, and drops off smoothly somewhere after 13 or 14 kHz. On the low end, the microphone's response dips slightly from 200 Hz down to 60 Hz, then rises slightly again from 60 to 40 Hz. Both the presence boost and low-mid dip correlate with what I heard when I tested the microphone—a bright finish and mildly attenuated low mids.

The latest low-cost condenser mic vying for a piece of the personal-studio market, Audio-Technica's AT3525 (shown here with the AT3527 and AT3528 "pencil" condensers that round out the 30 series) is a great-sounding, very versatile microphone complete with 10 dB pad and 80 Hz bass rolloff.



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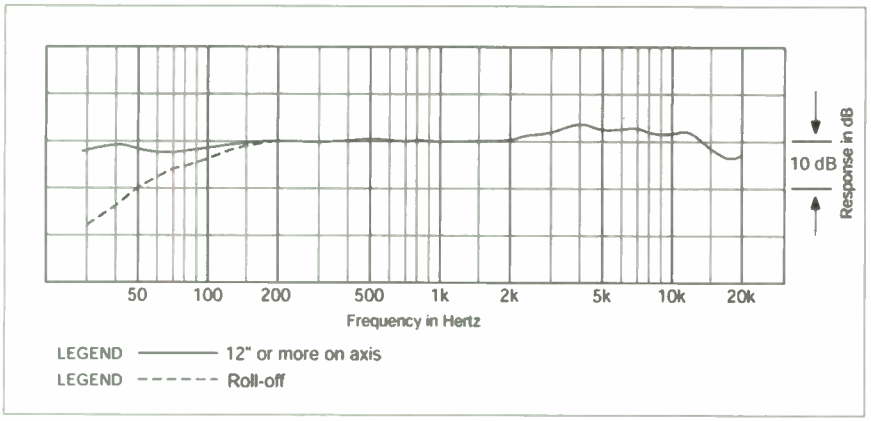

COURTESY AUDIO-TECHNICA

FIG. 1: The frequency response of the AT3525 is impressively flat up to the presence boost.

TOUGH COOKIE

During the test period, I happened to be recording tracks for South American musician Enrique "Quique" Cruz, a master of the Andean "pipes" (e.g., antara, toyos, mocxeño, kena, and kenacho) and an accomplished string player, as well, equally adept on guitar, cuatro, charango, and tiple. Cruz was working on songs for his Ph.D. dissertation on the role of cultural memory. On one of these songs, he played the ronroco, an oddly tuned, 10-string Bolivian instrument that looks like a cross between a beetle-back mandolin and a bouzouki, but sounds more similar to a lute.

The AT3525 did a beautiful job of capturing the soft, lilting sound of the ronroco. It also sounded great on Cruz's classical-style guitar. The mic's presence boost increased the clarity of the two instruments' nylon strings, while the slightly scooped-out low mids helped avert muddiness during heavy strumming.

To get a sufficiently hot signal from the relatively quiet ronroco, I had to crank the input gain on my Mackie board all the way up. That is to say, the AT3525 is a very low output microphone—an intentional feature of the mic's design. According to Audio-Technica, the high output of its 40 Series mics has proven problematic for many personal-studio and live-sound engineers, simply because budget recording mixers and lower-end live consoles tend not to have input-attenuation pads. By designing the AT3525 to have a low output, Audio-Technica hoped to increase the mic's usability in the personal-studio and performing-musician markets.

Impressively, Audio-Technica also managed to give the AT3525 quite high SPL handling (156 dB with the 10 dB attenuation pad engaged), which makes it suitable for a wide range of applications, including live-sound reinforcement. I recorded with the mic positioned only a few inches

AT3525 Specifications

Element	fixed-charge backplate, permanently polarized condenser
Diaphragm	¾", 2-micron thick, gold vapor-deposited Mylar
Polar Pattern	cardioid
Frequency Response	30 Hz–20 kHz
Dynamic Range	124 dB, 1 kHz @ max. SPL
Signal-to-Noise Ratio	72 dB, 1 kHz @ 1 Pa
Sensitivity	-48 dB (3.9 mV) re 1V @ 1 Pa
Maximum SPL	146 dB (156 dB with 10 dB pad)
Highpass Filter	80 Hz, 12 dB/oct.
Attenuation Pad	10 dB
Weight	8.1 oz.
Dimensions	4.7" (L) x 1.89" (D)

For manufacturer information, please see Contact Sheet, p. 163.

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AT3525

from the grille of a cranked-up guitar amp and, even without engaging the pad, experienced no distortion. I also recorded a 22-inch kick drum with the AT3525 stuck inside a hole in the front head of the drum. That time, of course, I engaged the pad. The mic was quite robust, capturing a desirable kick-drum sound with no hint of distortion.

The AT3525 has a fairly broad cardioid pattern, a smooth off-axis response, and minimal off-axis coloration. Self noise, too, is impressively low.

TO EACH ITS OWN

Although dimensionality is almost always a desirable trait, accuracy often is not. We usually choose a microphone precisely because of the way it alters or flatters a source (something that all microphones do to some degree). On certain instruments, for example, I preferred the sound of the AT3525—especially when it came time to mix. The AT3525's tracks tended to be bright and present, which helped them cut through the mix. Also, they didn't have too much low-mid content—something that can quickly gunk up a mix.

On a tambourine track, for example, the AT3525 captured a great balance of jingle and thwack. It also gave me the crispness and snap I want to hear from a wood shaker to bring it out in a dense mix.

Of course, no microphone is ideal for every sound source. But on the whole, the AT3525 proved very versatile, and, tonally at least, it compared

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quite favorably to a premium microphone.

LIVE THANG

The literature that comes with the AT3525 says that the microphone is suitable for a "wide range of applications," including "high-quality sound reinforcement." Thinking I would be remiss not to test the mic in a live-sound environment, I asked live-sound engineer Paul Trutnau of Command Sound Company if we could try the mic out on the Flatirons, a Portland-based "rockabilly-surf" band that was scheduled to play at the Starry Plough, a local pub.

This, I thought, would be a good reality check for the AT3525. The Starry Plough, a busy music venue, is fairly typical for a medium-sized pub: high ceiling, concrete floor, and enough other reflective surfaces to wring feedback from even the tightest-patterned dynamic stage mic.

During sound check, we set up the AT3525 for the lead vocalist. Of course, we immediately noticed an improvement in sound over the house mic (a standard dynamic); unfortunately, the AT3525's cardioid pattern proved too wide for front-of-stage work—particularly on a small stage with tightly crammed floor monitors. The AT3525 simply couldn't provide enough gain before feedback in this application. Even with no one else on the stage, the mic began to shriek as soon as I sang loudly into it. Finally, Trutnau and I agreed that there was no way the AT3525 would make it through the first set—if even the first song—so we changed back to the dynamic mic before the band came on.

During the process, I noticed something else that suggested that this was not a particularly appropriate microphone for live vocal work: the shock mount, when positioned beneath the mic (rather than above, which is the common studio method), became hard to adjust because the mic cable got in the way of the U-arm. My sense is that the AT3525 would work just fine on a guitar amp or as a drum overhead, especially in a much larger venue. But a small-venue vocal mic it is not.

The AT3525 compared favorably to a premium microphone.

REAL DEAL

The Audio-Technica AT3525 is a great-sounding, inexpensive, and very versatile condenser mic. It has a bright, open, detailed, and very present sound, and its high SPL handling extends its versatility both in studio and stage applications. The mic comes with useful amenities, too: the switchable 10 dB pad, 80 Hz bass rolloff, and sturdy shock mount.

The AT3525 is a great choice for the personal-studio owner looking to buy a single, low-cost condenser mic for recording a variety of instruments and sounds. It would also be a fine grab for the professional live-sound engineer who prefers to use condenser mics for certain sound sources.

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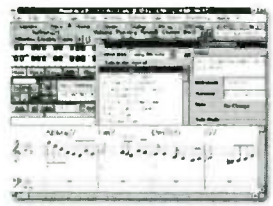

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LEXICON

STUDIO (MAC/WIN)

High-end sound and superb reverb in a mid-priced DAW.

By Mike Collins

With the introduction of its Studio for the PC (and soon for the Mac), Lexicon clearly intends to lay claim to the mid-priced DAW market. With a retail price of $2,999, this package is positioned midway between budget PC cards and high-end professional systems, such as Digidesign's Pro Tools.

Although its price is moderate, Studio's features are calculated to appeal to serious recordists: the system combines a bucketful of I/O and sync options with Lexicon's highly acclaimed DSP technology, 24-bit A/D converters, and 20-bit D/A converters.

WHAT YOU GET

The system consists of the Core-32 digital audio card (a full-size PCI card), the PC-90 reverb module (a daughterboard that clips onto the Core-32 card), and the 1U rackmount LDI-12T interface. A more full-featured interface, the LDI-16S, is due for release around the end of the year (see the sidebar "The Full Monty"). One Core-32 card can simultaneously support one LDI-12T and two LDI-16S interfaces, and you can install multiple Core-32 cards to expand the system.

The Core-32 card is a 24-bit PCI bus master that controls the clocking of the digital data down the PCI bus. The company's proprietary LexiPowerCore technology not only speeds audio transfers but relieves the CPU of some of its number-crunching chores, allowing your computer to run other tasks faster. According to Lexicon, this enables the Studio system to deliver at least 32 audio I/O channels with excellent stability.

The front panel of the LDI-12T interface provides lots of I/O connections, including RCA and XLR analog I/O (not simultaneously available) and an S/PDIF I/O pair on RCAs. A balanced XLR jack serves as a time-code input (see **Fig. 1**). The rear panel provides a BNC word-clock input, a set of ADAT sync connectors (see **Fig. 2**), and a pair of Toslink optical I/O connectors that can be switched between 8-channel, 20-bit ADAT format and 2-channel S/PDIF format.

The LDI-12T thus provides a total of twelve simultaneous inputs and outputs: two analog and ten digital. You can simultaneously use the coaxial S/PDIF and optical ADAT inputs and outputs, but you cannot simultaneously use coax S/PDIF and optical S/PDIF input (for a total of four S/PDIF input channels) because there is only one S/PDIF input driver. The optical and coax S/PDIF outputs are simultaneously available, however, so you can have four S/PDIF output channels.

WHAT YOU NEED

As of this writing, the Lexicon system only runs on the PC with Steinberg's *Cubase VST* as the software front end. Mac drivers are being developed for *Cubase VST*, however, and they should be available by the time you read this. Opcode's *Studio Vision Pro* and *Vision DSP* will also support the Lexicon Studio on the Mac.

Furthermore, by the end of September, the new multichannel Windows 95 Multimedia I/O (MMIO) code will hit the street, which will allow you to launch the Control Panel without launching *Cubase VST*. The basic features of Lexicon Studio will then work with a wide range of audio programs, including Steinberg's *WaveLab*, Sonic Foundry's *Sound Forge*, Syntrillium's *Cool Edit Pro*, SEK'D's *Samplitude*, and pretty much anything else that supports a Creative Labs Sound Blaster.

For optimum performance, Lexicon recommends a 300 MHz or faster Pentium II and two separate drives: a boot disk and a drive for storing audio. Lexicon also recommends that you use an AGP (Accelerated Graphics Port) video card for your PC because the AGP card doesn't use the PCI bus, which leaves slots free for the SCSI controller and Lexicon Studio.

You don't have to use SCSI drives, though: according to Lexicon, the latest IDE drives work quite well and are reaching speeds comparable to those of SCSI drives that were available a couple of years ago. Still, SCSI has its advantages, and if you are considering buying a high-quality DAW, the additional investment in SCSI is probably worthwhile. (For more information on storage media, see "Mass Storage for Musicians" in the July 1998 **EM**.)

GETTING STARTED

The review system was sent preconfigured from Wave Distribution (tel. 973/728-2425; e-mail gil@wavedistribution.com; Web www. wavedistribution.com) and included a Pentium II/300 MHz PC with 128 MB of RAM, a 2 GB internal drive to store the software, and Windows 95. An additional 4.3 GB Ultra Fast and Wide SCSI-3 internal drive was installed for the audio.

The Lexicon Core-32 System PCI card had the PC-90 daughterboard attached and was preinstalled in the computer. All I had to do was hook up the multipin connector from the back of the card to the LDI-12T interface, wire the interface's S/PDIF I/O to my Yamaha 02R digital mixer, connect the 02R's stereo analog outputs to the LDI-12T, and boot up the computer.

The first thing I noticed was the lack of indicators on the interface to show the presence of signals coming into the unit. I wanted to make sure that the cables were good and that the levels were set somewhere in the ballpark before setting up *Cubase VST*, but I couldn't. It took me a few attempts before everything was routed correctly; a set of indicators on the LDI-12T could have helped me troubleshoot the situation.

FIG. 1: The front panel of Lexicon Studio's LDI-12T interface provides an assortment of RCA and XLR jacks for S/PDIF and analog I/O.

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FIG. 2: The rear of the LDI-12T interface offers optical Toslink connections for 8-channel ADAT or S/PDIF formats and provides serial connections for ADAT sync and machine control.

After properly setting up *Cubase*, I tested the Lexicon Studio's analog inputs by transferring a drum track from a Phil Upchurch album via the 02R into the analog inputs on the LDI-12T. Then I transferred the same drum track via S/PDIF. I couldn't hear any difference between these recordings, which told me that the A/D converters in the Lexicon Studio sounded at least as good as those in the 02R.

Then I transferred eight tracks of a jazz-rock piece that I was working on in Pro Tools, so I could see how the PC-90 sounded in a mix. I didn't have the 8-channel ADAT interface hooked up yet, so I decided to transfer two tracks at a time via the LDI-12T's stereo S/PDIF inputs. After all, not everyone who buys this system and wants to transfer multiple digital tracks will have ADAT Lightpipe-equipped gear.

GETTING IN SYNC

Once I had the first two tracks recorded, I needed to make sure that any subsequent tracks would be recorded alongside them in sync. The only way to

THE FULL MONTY

The Lexicon Studio system is expandable in several senses. You can install several Core-32 cards, and each card can simultaneously support one LDI-12T (described in the body of our review) and up to two optional LDI-16S interfaces (price tba). In addition, the LDI-16S interface accepts up to three optional add-ons. When fully expanded ($3,599), it is jam-packed with connectors.

The stock LDI-16S includes eight channels of analog I/O on balanced XLRs and one pair of 8-channel, TASCAM TDIF digital I/O connectors. Thus, a stock LDI-16S has sixteen I/O channels. Unlike the LDI-12T, the higher-end interface's analog I/O features 24-bit A/D converters, as well as 24-bit DACs. Sync connections on the LDI-16S include word-clock loop-through and out-through (on BNCs), Sony RS-422 serial 9-pin for machine control, and a port for daisy-chaining a second LDI-16S.

However, the LDI-16S really comes into its own when you expand it. Lexicon has announced three optional packages: the AES-8, MDM, and STC-1.

The AES-8 option (price tba) supplies eight channels (four stereo pairs) of AES/EBU digital I/O on balanced XLRs. In addition, you get one stereo S/PDIF I/O pair on RCAs, which replace channels 7 and 8 of the AES/EBU I/O (selectable in the host software). These AES/EBU and S/PDIF connectors can transfer 24-bit digital audio. The AES-8 also provides real-time sample-rate conversion for all formats.

By the way, those AES/EBU ports are capable of double-wide, dual-spigot operation, which means an AES/EBU stereo pair can be used as a single channel ("dual spigot") capable of recording and playing back at 96 kHz, rather than the system's usual 48 kHz (hence, "double wide"). That means a Studio with one LDI-16S and an AES-8 option can handle up to four channels of 24-bit, 96 kHz audio.

Many studio owners will want to add the MDM option (price tba). This add-on supplies a second 8-channel TDIF connector, two 8-channel ADAT Optical inputs, and three 8-channel ADAT Optical outputs. You also get ADAT 9-pin sync I/O. If you have a digital mixer, an Alesis or TASCAM MDM, or other such gadgets, this "option" might prove a necessity.

Summing up the channel count, with one LDI-16S and both audio I/O options, you have sixteen channels of TDIF I/O, sixteen ADAT input channels, 24 ADAT output channels, eight channels of AES/EBU I/O (two of which can be S/PDIF), and eight analog I/O channels, for a total of 48 inputs and 56 outputs. And if that's not enough, remember that you can use two of these per Core-32 card, along with one LDI-12T (twelve channels). Mind you, the card itself can handle only up to 32 channels at once, but with all these ports, you get, in essence, a software-configurable patch bay.

If you are using a lot of digital audio devices—and especially if you create audio for picture—you have serious sync needs. To address these, Lexicon offers the STC-1 option (price tba), which gives you longitudinal time code (LTC) input and output on balanced XLRs; vertical-interval time code (VITC) input and output on BNC connectors; and a loop-though to house sync. The last includes mechanical automatic termination, so you don't have to worry about your house-sync system destabilizing if a device is turned off or disabled.

One part of the STC-1 has potential for lots of unusual and creative applications: the General Purpose Interface (GPI). This D-25 connector supplies eight contact closures so you can trigger external devices from the Lexicon Studio System. These triggers will work with 24V or transistor logic level (TLL) signals, so they can be used in many ways. Veteran post engineers might use the GPI to trigger Foley effects and automatic dialog replacement (ADR), but we clever EM types will probably find lots of strange things to do with it. Personally, I plan to trigger my espresso machine just when we're getting close to the end of the mix.

—Steve Oppenheimer

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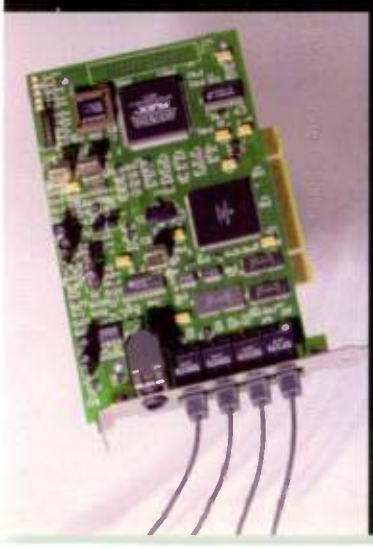

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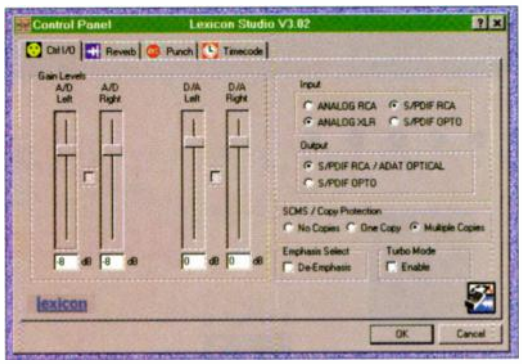

FIG. 3: With Lexicon's ASIO Control Panel, you can set up a variety of I/O configurations. Note that you also can add SCMS copy protection if you so desire.

do that is to have word clock and time code connected between the two systems to keep them locked to a common clock reference and a common SMPTE positional reference. I use a MOTU Digital Timepiece (DTP) as the timing reference for my system, so I connected the DTP's SMPTE Out jack to the SMPTE In port on the front of the Lexicon interface. I also ran a word-clock signal from the DTP to the Yamaha 02R and then to the BNC input on the back of the LDI-12T. I also had a 256× (Superclock) word-clock connection to Pro Tools from the DTP. I used an Alesis LRC remote controller plugged into the front of the DTP to start and stop the whole system using MIDI Machine Control commands.

The sync system worked perfectly. Hitting Play on my LRC started the clocks running in the DTP to output word clock to the 02R and Lexicon Studio. SMPTE was sent to the Lexicon Studio to provide positional information for *Cubase VST*, and MTC was sent to the 02R to run its automation. Superclock and MTC went to the Pro Tools system.

Of course, I could have avoided the hassle of making multiple synchronized recordings by simply hooking up the ADAT ports on the back of the interface to my 02R. I could then have routed the eight Pro Tools tracks to separate outputs from Pro Tools into the 02R and from there to the ADAT outputs. I could then have transferred all eight tracks in one pass into *Cubase VST*. The moral of this story is that, unless you are working with a very basic studio system, you'll probably want to use the ADAT 8-channel I/O with a digital mixer like the 02R to gain the flexibility that you really need when combining the Lexicon system with all the other gear in your studio.

THAT FAMOUS REVERB

Once I had my eight tracks in the Lexicon Studio, it was time to check out the reverb. The PC-90 daughterboard is based on the same processing engine as Lexicon's acclaimed PCM 90 reverb processor (which retails for $2,999) and features Lexicon's latest custom, nonlinear LexiChip DSPs.

Five classic Lexicon reverb algorithms are included in the PC-90: Chamber, Inverse, Room, Ambience, and Concert Hall. These can be loaded into either of two reverb engines on the card. Two banks of preset reverb and delay effects all sounded first rate.

I couldn't help but notice, however, that the user interface of Lexicon's *LexiVerb* TDM plug-in, which I use on my Pro Tools system, is vastly superior to the PC-90 user interface provided from within *Cubase VST*. *LexiVerb* has a great visual display and attractive, pop-up parameter controls. These are all clearly displayed along with a representation of the currently selected algorithm.

LexiVerb sounds quite good, but the PC-90 sounds even better. According to Lexicon, this is because the LexiChip DSP chips offer greater depth and quality than the plug-in running on the Pro Tools DSP Farm's Motorola DSP chips. This is the fabled Lexicon reverb; make no mistake about it.

Lexicon has provided flexible routing options into and out of the PC-90 processors. For example, if you just want to use the PC-90 as a stand-alone outboard effects unit, you can simply route a pair of stereo inputs and outputs from the LDI-12T interface directly to and from the card using Lexicon's ASIO Control Panel (see **Fig. 3**).

Normally, though, you would set up the PC-90 to work within *Cubase*. By default, the outputs of the PC-90 are summed with the outputs of *Cubase VST* and routed to the stereo outputs of the interface; so the output level is determined by the settings in the PC-90 preset.

Unfortunately, the PC-90 effects do not work either as inserts into the channels or as inserts across the stereo mix in the *VST* Master window. However, Lexicon has overcome this limitation by enabling you to route the PC-90 (using the ASIO Control Panel) to any output of the LDI-12T interface. You can even cascade the reverb effects using the Control Panel. It is also possible to route the effects returns from the PC-90 into the *Cubase* mixer's channel inputs so

Lexicon Studio Specifications

Analog Inputs	(2) balanced XLR (+4 dBu or -10 dBV, switchable); (2) unbalanced -10 dBV RC
Analog Outputs	(2) balanced +4 dBu XLR
Digital I/O	(1) S/PDIF stereo pair RCA (coaxial); (2) optical Toslink connectors (8-channel ADAT or 2-channel S/PDIF format)
Other Connectors	(1) BNC word-clock input; (1) balanced XLR time-code input; (1 pr.) ADAT 9-pin, D-sub sync I/O; (1) Sony RS-422 serial 9-pin for machine-control; (1) 68-pin port for host computer
Converters	24-bit A/D; 20-bit D/A
Sampling Rate	44.1 and 48 kHz
A/D THD	<0.005%, 20 Hz to 20 kHz
D/A THD	<0.01%, 20 Hz to 20 kHz
A/D Dynamic Range	104 dB typical, 20 kHz bandwidth; 106 dB typical, A weighted
D/A Dynamic Range	94 dB typical, 20 kHz bandwidth; 97 dB typical, A weighted
Power Requirement	9 VAC (wall transformer provided)
Dimensions (LDI-12T)	1U rack-mount x 4" (D)

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Lexicon Studio
Minimum System Requirements
PC and Mac: PCI-equipped computer, host software that supports ASIO or Windows Wave drivers; audio-capable hard drive. Other requirements depend on host software.

you can set the effects-return levels and EQ the signals before feeding them into your mix.

However, if you route the signals in this way, you must consider the problem of latency. Latency is the processing delay that you get with any input that you monitor though software. When you record or play back, this isn't a problem, but if you overdub new instruments—or feed effects returns from the PC-90 into the mixer—and you monitor these signals through the software, rather than through an external mixer, you will hear a delay between the incoming signal and the tracks playing back from the hard disk.

To avoid this delay a Punch mode lets you audition just about anything (including effects returns) directly through the Studio card. That's a very handy feature. You can also record the effects-return signals onto your hard disk and replay them along with your mix to avoid the problem.

THE BOTTOM LINE

The Lexicon Studio's excellent audio quality, the PC-90's professional-level reverb, and the system's Punch mode are some of its greatest strengths. With its time-code, ADAT-sync, and word-clock features, and its digital and analog I/O options, you've got a dynamite audio-production system with great flexibility.

Cubase VST users who want truly professional results should seriously consider a Lexicon Studio instead of a budget audio card. As mentioned earlier, by the time you read this, Opcode's *Studio Vision Pro* and *Vision DSP* should also support the Lexicon Studio on the Mac, and other companies (both Mac and PC) are slated to join the band around the same time. With its audio quality and versatility, this mid-priced system is a clear winner.

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AUDITY 2000

This tempo-based synth can rip your head off.

By Jeff Burger

Diehard synthesizer aficionados will remember a creature in early analog history (circa 1979) called the E-mu Audity, scientific name *Dromaius novaebollandiae auditus.* This 16-voice behemoth was controlled via a digital scanning keyboard, and only one prototype and one custom unit (commissioned by ex–Tangerine Dreamer Peter Baumann) were built before E-mu Systems decided that, with its price tag of approximately $50,000, the Audity was not going to be viable.

Nearly twenty years later, E-mu has released the Audity 2000, and the only similarity worth talking about is the name. This newcomer is a thoroughly modern synth module, replete with in-your-face industrial sounds and clock/tempo-based modulation sources, including sixteen (yes, sixteen!) simultaneous arpeggiators.

INTERFACE EVOLUTION

At first glance, the single-rackspace Audity 2000 module might be mistaken for a member of the Proteus family, like Carnival, Orbit, and Planet Phat. Indeed, some Proteus-family traits are present. For example, Proteus users will feel right at home with the 2 × 24-character LCD, negotiated via cursor buttons and data wheel.

However, with its patch-cord modulation architecture and Z-Plane filtering, the Audity draws at least as much from the Emulator IV's EOS operating system. The OS is in Flash memory, which provides for easy updates. In addition, this is the first E-mu offering to utilize Motorola's recent 32-bit Cold Fire processor, which is 2.5 times faster than the brain at the heart of the EIV. There's nothing remotely sluggish about this bird.

Proteus users will also recognize the familiar rear-panel complement of MIDI In, Out, and Thru, along with main stereo outs and two stereo pairs of sub-outputs that can double as effects send/return (see **Fig. 1**). (In a pinch, you could even use the returns to make the Audity 2000 serve as a line mixer.) All the audio ports have 18-bit DACs. New to the lineup is an RCA S/PDIF output that can route the main stereo signal in the digital domain to S/PDIF-equipped devices such as DATs, digital mixers, and hard-disk recorders.

The front panel sports four real-time controller knobs and a switch that assigns the knobs to three different sets of functions. In QuickEdit mode, these knobs allow you to control common synth parameters, such as filter cutoff and resonance, as well as core arpeggiator functions, without having to enter Edit mode. (The LCD does not show the parameter values in QuickEdit mode, however.) The controller knobs can optionally send MIDI data to virtual controllers A through H, the assignments of which are user-definable.

When you are editing a preset, the four knobs correlate to the four parameters displayed on the current LCD screen, thus sparing you one more round of cursor moves. Of course, you'll need to position the module within easy reach if you want to take advantage of these real-time features.

SONIC ARCHITECTURE

The Audity offers 32-voice polyphony, and E-mu expects to debut a software upgrade this fall that will double the polyphony to 64 voices. Presets are organized into seven banks, with a total of 640 in ROM and 256 for user programming.

Proteus users are accustomed to two layers of sounds per Preset, but Audity Presets employ four layers. Each layer comprises a sampled waveform, processed through a Z-Plane filter and amp; three multistage envelopes; two LFOs; and copious matrix-modulation options. Each layer can be zoned, as well as crossfaded or cross-switched by key position, Velocity, or an assigned controller, allowing for very interesting combinations.

As with the Proteus series, each of the sixteen channels can be assigned a single Preset for multitimbral work. Although there is no dedicated Performance mode, as found on some products, each Preset can be linked to two others—each with its own key range, Velocity range, volume, pan, transpose, and delay settings. Between the options for layers within Presets and the linking of multiple Presets, you can set up some serious performance scenarios.

The Audity ships with more than 200 waveforms in 16 MB of ROM. An internal slot is designed to accept a 16 MB ROM card, which is still in development. The sound set is made up primarily of sampled analog waveforms, modern drum kits, and a healthy seasoning of industrial noise and similar types of rude effects—nothing for the squeamish. The sounds are all 16-bit waveforms sampled at 44.1 kHz, a step up from the Proteus's 39 kHz. A chorus effect is available at the oscillator level, but it steals half the polyphony.

The unit also includes a resonant Z-Plane filter that evolved from E-mu's Morpheus (see **Fig. 2**). This unusual type of filter is capable of morphing or interpolating between two completely different filter states under control of an

E-mu's Audity 2000 is not for the faint of heart, nor is it likely to attract those who play traditional music. But it's very cool if you want to create futuristic effects or play rude and rhythmic industrial, acid, techno, or rave.

For manufacturer information, please see Contact Sheet, p. 163.

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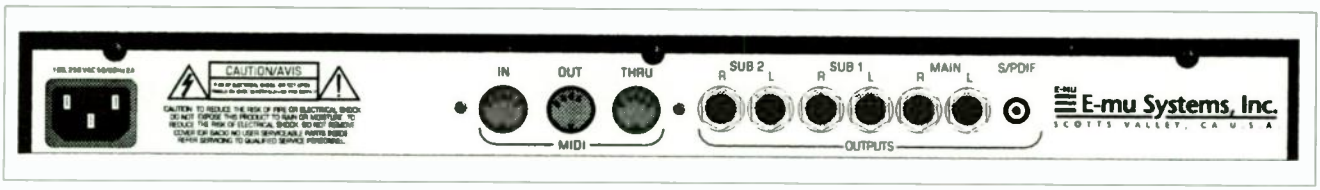

COURTESY E-MU SYSTEMS

FIG. 1: In addition to the usual ¼-inch analog outputs, the Audity 2000 has S/PDIF digital output.

envelope, LFO, Velocity, Pressure, various real-time controllers, and more. (For more on Z-Plane filtering, see the review of the Morpheus in the May 1994 **EM**, available on the **EM** Web site at www.emusician.com.)

The incarnation of Z-Plane filtering employed in the Audity incorporates eighteeen of the sixth-order filters found in the EIV sampler, plus 32 new twelfth-order filters, for a total of 50 filter types. You'll find all the classic low-pass, high-pass, and band-pass types, and the filters can be two, four, or six poles. You also get vowel formants, wild phasers, flangers, and combs. There's a good helping of EQs, which produce sounds ranging from cheesy radio to DJ rings to bombastic bass boom.

The Audity's three envelope generators depart from the Proteus family's standard fare, offering six stages with level and rate controls for each stage (see **Fig. 3**). There are two stages each for attack, decay, and release, with the decay stages falling under the jurisdiction of key sustain. Each envelope can be either time based (like we're used to) or tempo based, which is unique, in my experience. The tempo-based version syncs the envelope to the master clock and compresses or expands the time parameters according to the clock speed—a great feature when working with techno and related music. As you'll see shortly, this is just the tip of the iceberg when it comes to the Audity's clockcentric features.

The two LFOs offer the standard sine, triangle, square, and pulse waves; the latter is locked into intervals at 12, 16, 25, and 33 percent. The LFOs can be free-running or set to start over at the beginning of their cycles when a key is pressed. You can program an optional delay amount, and a Variation control allows each note played to have a slightly different LFO rate from its sibling, providing an ensemble effect. Each LFO can be triggered by the master clock, which resets the LFO to zero every time the clock goes to zero. When the rates of the two elements are similar, the LFO effectively syncs to the clock. Disparate speeds can reshape the LFO waveform, which is sometimes useful, sometimes not. This triggering can also be applied as any of 25 different metric multipliers or subdividers of the beat, including triplets!

MODERN MODULATION

The matrix modulation at the heart of each layer uses the patch-cord metaphor borrowed from E-mu's EIV. Unfortunately, you have to visualize that metaphor because the Audity's LCD provides no way to do justice to the concept. Clearly, this synth is a very good candidate for computer-based patch editing.

You'll find 24 general-purpose patch cords that can be used for each layer, with a total of 64 per Preset. Some modulation paths are hard-wired, such as pitch LFO, filter envelope, and amplifier envelope, thus freeing up some patch cords. However, this doesn't prevent you from routing those modulation sources to other destinations.

There are a total of 66 modulation sources and 64 destinations. Positive or negative amounts are programmable for each connection. There is no

AUDITY'S AWESOME ARPEGGIATOR

In addition to the wealth of features described in the main article, the awesome Audity 2000 arpeggiator has a long list of global functions that affect the selected arpeggiator mode. Here are some highlights.

1. Note value can be set to anything from double whole notes down to 32nd notes, including all the metric values and triplets in between. This parameter is available for the individual steps in User Pattern mode and is applied globally in all of the other modes.

2. Velocity can be specified or can follow the performed Velocity.

3. Gate time is described as a percentage of the note value.

4. The Extension Count and Extension Interval parameters work together to determine the number of additional iterations and the range, respectively. For example, an Extension Count of "1," teamed with an Extension Interval of "4," would play the performed notes, followed by the same intervals transposed up a musical fourth. Upper limits are 15 and 16, respectively.

5. Sync determines whether the first note in the arpeggio sounds immediately on performance or is quantized to the next clocked metronomic value.

6. Delay allows you to specify a lag time before the arpeggiator kicks in. This resets after all keys are released.

7. Duration shortens the number of steps that are triggered. This function can increase the perceived variations in patterns when combined with a delay.

8. Recycle repeats the pattern immediately after the duration value has been reached, otherwise the pattern stops until manually retriggered.

9. Keyboard Thru merges normal note performance with arpeggiated notes for a layered effect.

10. Latch forces notes to hold until they are either played again, or the channel arpeggiator or latch function is turned off.

11. Key range allows each arpeggiator to respond only to a specified key zone.

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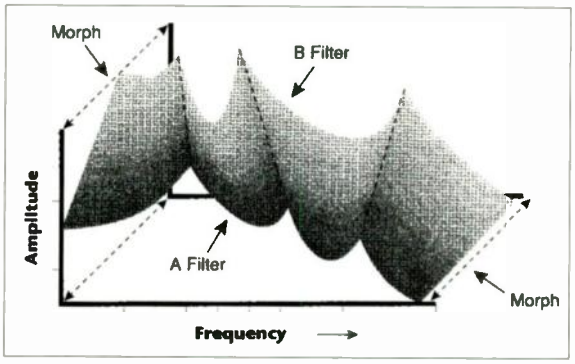

COURTESY E-MU SYSTEMS

FIG. 2: E-mu's Z-Plane filter changes its function over time by morphing between two different filter types. It accomplishes this by interpolating between parameter values in the two filter types.

restriction on how many times a source or destination can be patched within a layer, and you can even modulate other modulators in some cases. Some examples of common modulation sources include key number, LFOs, envelopes, Velocity, MIDI Control Changes, and noise. Common destinations include fine and coarse pitch, filter cutoff and resonance, volume, glide, envelope parameters, LFO rate, and pan. I've ragged on other synth manufacturers in recent reviews for not providing panning as a modulation destination for added spatial dimensionality, but the Audity comes through!

There are also some special modulation processors designed to be patched between sources and destinations, including conditional switches, diodes, lag processors (see **Fig. 4**), quantizers, summing amps, and other esoterica rarely seen since the halcyon days of modular synthesizers. Ironically, amid all that, there's no provision to modulate an old synth staple—pulse width—because the waveforms are samples.

One of the most exciting modulation sources is the system clock, regardless of whether it is the Audity's internal clock or an external MIDI Clock source. (You can also program an external MIDI controller to govern clock speed. There's no provision for Tap

Audity Specifications

Polyphonic Voices	32
Multitimbral Parts	16
Presets (ROM/user RAM)	640/256
Waveform ROM	16 MB (expandable to 32 MB)/200 waveforms
Sample Rate/Resolution	16-bit/44.1 kHz
Filters	(50) Z-Plane: (18) sixth-order and (32) twelfth-order
Envelope Generators	(3) 6-stage
Modulation Sources/Destinations	66/64
Arpeggiators	16 (independent)
Arpeggiator Patterns (factory/user)	200/100 (32 steps ea.)
No. of Effects Processors	2
No. of Effects (Processors A/B)	44/32
Analog Audio Outputs	(6) ¼" unbalanced
Digital Audio Outputs	(1) S/PDIF stereo on RCA
Submix Inputs	2 (analog; sum to main outs)
Other Ports	MIDI In, Out, Thru
Digital-to-Analog Converters	18-bit linear
Internal Processing Resolution	24-bit
Signal-to-Noise Ratio	>92 dB
Dynamic Range	>90 dB
Frequency Response	20 Hz–20 kHz (+2/-1 dB)
THD + Noise	<0.02% (1 kHz sine wave, A weighted)
Intermodulation Distortion	<0.05%
Dimensions	1U rack-mount x 8.5" (D)
Weight	6 lb., 14 oz.

If you're serious about your music, don't play games with your computer sound card.

You need a MIDI/audio sound card that's as committed to making great music as you are. The Yamaha SW1000XG fits the bill perfectly. Actually, calling it a sound card doesn't do it justice—IT'S A COMPLETE, TURNKEY MIDI PROJECT STUDIO THAT FITS IN YOUR COMPUTER. It has a 64-note poly XG synthesizer, 6 fully programmable effects processors, 12 channels of accelerated hard disk recording and a robust package of sequencing, recording and editing software, all for just about $699 MSRP. After all, when it comes to creating music, Yamaha doesn't play around.

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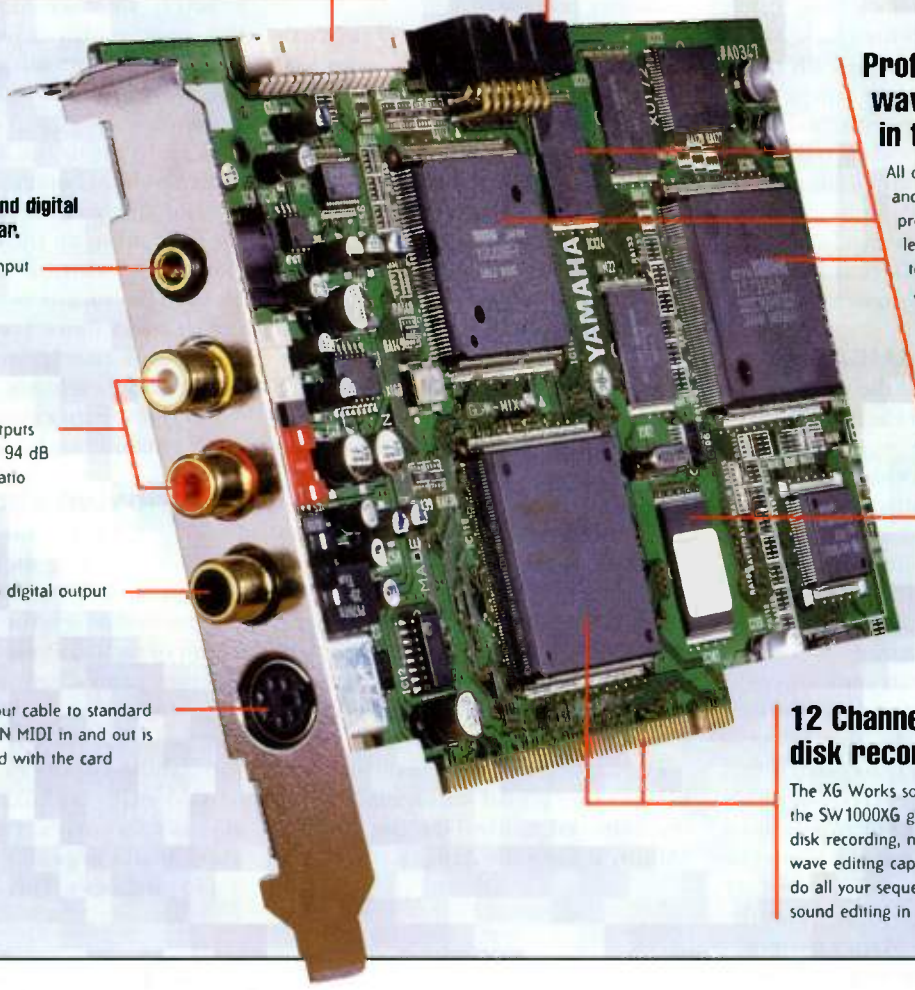

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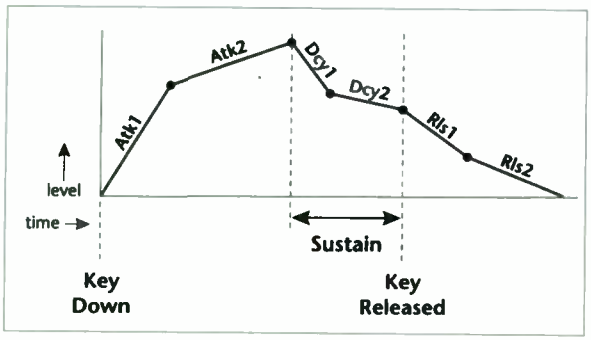


COURTESY E-MU SYSTEMS

FIG. 3: With the Audity's 6-stage envelopes, when you trigger the sound from a MIDI keyboard, the envelope begins at zero and moves to the Attack 1 Level at the Attack 1 Rate, then procedes to the Attack 2 Level at the Attack 2 rate. If you hold down the key, the envelope continues through the two decay stages. As soon as you release the key, the envelope moves to the release stages.

Tempo, however.) As mentioned earlier, the LFOs and envelopes can all be driven by the clock. You can also modulate anything else that's a valid Patch Cord destination with the clock. Moreover, the clock can be metrically divided or multiplied for each arpeggiator, resulting in some fantastic "synchrosonic" rhythm textures when more than one is used. Potential applications, such as retriggering a 6-stage modulation pattern by synching an envelope to the clock and cycling different layers on and off, show Audity's true colors.

ARPEGGIATOR ALLIGATOR

So far, we've discussed the tremendous rhythmic potential of the Audity's ability to sync many different parameters to internal or external clock. The icing on the cake is the instrument's sixteen parallel arpeggiators, one for each of the sixteen MIDI channels. When played from a multichannel controller or sequencer, the possibilities are incredible—and dangerous!

Each Preset stores its own arpeggio settings, which are accessed via the Proteus-like Edit button. An identical master arpeggiator menu is accessed with a dedicated button. The screen in which Presets are assigned to channels has a field that is used to turn the arpeggiator on and off and to select whether the channel's arpeggiator will play the Preset pattern or the Master pattern.

How an arpeggiator responds is determined by a comprehensive series of parameters. Pattern types include up, down, up/down, random, forward (order of performance), backward, forward/backward, and user-programmed. There's memory enough for 100 user patterns, each of which can be up to 32 steps in length. This is a step-entry job: you increment the display through each step in the pattern and set the desired parameters, including Key Offset/Tie/Rest, Velocity, Duration, and up to 32 repeats.

Individual steps can also be turned off, so you don't have to start over if you find you've botched a step. There's also a provision to copy patterns. Given the advanced concepts found in this instrument, it's surprising not to find a pattern-entry option that resembles a more intuitive, real-time performance process.

In QuickEdit mode, the four real-time knobs can control the resolution, extension, Velocity, and gate of the current channel arpeggiator. This sweet feature allows variation of the arpeggiator effect during performance.

Whew! As you can see, this plethora of parameters employed in various combinations provides for an incredible amount of flexibility within each of the sixteen arpeggiators. Combining two or more arpeggiators can get nuts pretty quickly. This unique Audity feature requires a different mind-set than grabbing a chord and taking a cosmic ride. It is often much better to play individual notes and get into the effect generated by the synchronized interaction of different arpeggiators playing different patterns at different clock divisions using different voices on different channels. Got it? With all due respect, you likely won't until you test-drive the beast yourself. It's one of those instruments you just have to play.

The arpeggiator even features a battery of other global settings that affect the selected mode (see the sidebar "Audity's Awesome Arpeggiator").

MULTIPLICITY OF EFFECTS

The Audity has two stereo effects processors. The A processor generates 44 effects dedicated to reverb and delay, with user control over decay and high-frequency damping. The B processor offers 32 chorus, flange, and distortion effects with programmable feedback, rate, and delay time. You can even send a given amount of effect B to effect A, which is a nice touch. However, Audity lacks some effects one expects to find, including rotating speaker, exciter/enhancer, and compressor.

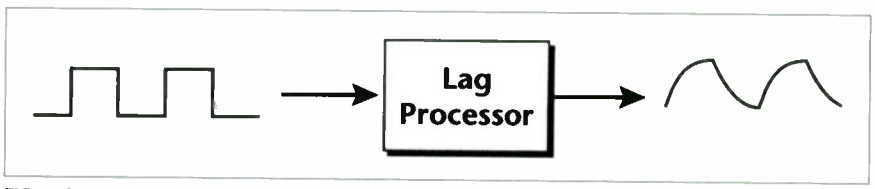
The two effects are joined at the hip, and there's not enough horsepower to have different effects on different multitimbral channels. The effects settings are derived either from the master settings or from the settings of a specified preset. Although there's a reasonable amount of flexibility in routing effects to the three audio output buses, the sonic quality is just average, and there are fewer user parameters than can be found in many competing products.

In addition to the effects section's rather ho-hum implementation, there's a major faux pas: the effects are almost the only thing on the instrument that can't be synchronized to the master clock. That means you'll have to dial in the effect times manually to match all those gloriously synched arpeggios, LFOs, and envelopes, and the delays won't track tempo changes. Hey E-mu, you're hurtin' me here!

AUDITIONING AUDITY

Playing Audity's factory demo sequences is nothing short of a headspin. It's absolutely nuts and makes a brash statement that sets the stage for exploration of factory Presets.

Preset names have prefixes such as "arp," "edg," "syn," and "pad" that help categorize them. The first two banks (and their copies in the user RAM banks) offer a shotgun smattering across categories. Fortunately, the other banks organize Presets according to categories. This approach makes



COURTESY E-MU SYSTEMS

FIG. 4: A lag processor slows down rapid changes in the input signal, so that the output "lags" behind the input at a preprogrammed rate.

it easy to find most Presets by style, but it would still be nice to have a method of viewing *all* sounds according to category.

Many of Audity's sounds are rudely industrial and would have been relegated to the novelty department only a few years ago. In today's anything-goes music scene, this may be just what the doctor ordered. Although there is a reasonable amount of analog synth sounds, the unit's sample playback technology doesn't seem able to deliver the fatness and warmth of true analog. In addition, E-mu intentionally left off compression, limiting, and other processing that is normally applied to protect speakers and eardrums. This allows you to create some really nasty sounds, but it also means you have to be very careful when using extreme settings.

I found some excellent examples of Audity's tempo-based features, including "Clocking Out" and "Modify." These are patches you can just grab and groove on, even sans arpeggiator. I also liked sequencing with "Env Kit," in which blocks of keys are tuned to the same notes, with subtle differences in filter setting. The hip-hop crowd will find plenty of resources with drum kits like "Cooleo" and "Drumatix," along with basses such as the subsonic "Submarine." Dance mavens will appreciate chording Presets such as "Ice Dreams" and "House Cord." Leads range from the delayed, Minimoog-esque "Tangerine" to the fried static of "Short Circuit."

There's also decent fodder for more traditional sequencing with a lighter touch, such as the spacey "Fundamentals," blippy "Prophetics," and bell-like "Ballerina." Alas, many—perhaps too many—sounds fall in the category of "too quirky to even begin to describe"!

HORSE OF A DIFFERENT HUE

The Audity 2000 could help drive a whole new wave of electronic music. A tempo-based instrument is an idea whose time has come, and E-mu has implemented that concept well. The innovation of having sixteen discrete arpeggiators takes the Audity over the top. The sounds are crisp, and the interface is snappy and nicely evolved for a single-space rack. The MIDI implementation is comprehensive, and the four real-time control knobs add a lot of performance flair. The only real weak area is the effects—the omission of synchronization in this department is hard to ignore.

Is Audity for you? That depends. If you're into cutting-edge music in the industrial/acid/techno/rave genre, it's a real kick. Audity will also be at home in commercial production environments, not so much as the sonic centerpiece as for tastes of futuristic effects. In either case, using it to trigger other MIDI sound sources can certainly extend the arpeggiator's possibilities. For more traditional applications, the sound set is rather rude, and the multiple arpeggiators may find limited use. (This box would send Barry Manilow running for the hills!)

Regardless of whether you like Audity, it is a harbinger of the future. I'd be surprised if E-mu didn't put Audity's technological advances into their next round of samplers. One thing is certain: Audity is audacious.

Jeff Burger *is a songwriter, digital artist, author, and multimedia producer based in Sedona, Arizona.*

E-MU SYSTEMS
Audity 2000 synthesizer module
$1,795

FEATURES ■■■■■
EASE OF USE ■■■■
QUALITY OF SOUNDS ■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Entirely tempo-based synthesizer architecture with synchable envelopes and LFOs. Sixteen simultaneous arpeggiators. Excellent matrix modulation and four real-time control/edit knobs. Flexible Z-Plane filters. Expandable sample-playback ROM. High fidelity combined with S/PDIF output.
CONS: Effects processors mediocre and not synchable to clock. Factory sound set too brash for more traditional musical genres. Small LCD display for the programming power. Too bright and thin for true analog emulation.

CIRCLE #440 ON READER SERVICE CARD

SONIC FOUNDRY

ACID (WIN)

An intuitive approach to arranging sample loop–based songs.

By Scott R. Garrigus

If you've ever created a song by piecing together digital audio sample loops, then you're aware of the problems that can arise during the production process. More often than not, your source material will differ in tempo, pitch, or both. Even though today's advanced digital audio editing software allows you to stretch or transpose loops to make them match, it's tedious to have to go through all the steps manually for every sample in a project, and this can easily stifle creativity. Sonic Foundry aims to remedy the plight faced by all you loop jockeys out there by offering a new loop-based arrangement tool called *ACID*.

ACID is a sequencing program in which you arrange audio loops using simple drag-and-drop editing. No matter what the original tempo or pitch of your source samples, *ACID* automatically stretches and transposes them to fit the current project's tempo and key. It also allows you to apply dynamic envelopes for manipulating volume and panning, and the program supports DirectX-compatible plug-in effects. What's more, it does everything in real time.

If you think that you'll need a powerful machine to tap *ACID*'s full potential, you're partly right. During my tests with the program, I found that the only time that raw power was really needed was when applying effects. Otherwise, even an old Pentium 75 with 24 MB of RAM can have you hip-hoppin' your way through your next tune. And because *ACID* works with any Windows-compatible sound card—even multiple sound cards at once—you can rest assured that your existing hardware is probably all that you'll need.

THE ACID PARADIGM

The *ACID* interface is divided into three main sections: the Track List, Track View, and a multifunction accessory panel (see **Fig. 1**). The accessory panel occupies the lower third of the main window and provides direct access to the Media Explorer, Properties window, Mixer, and Effects utilities through a set of tabs at the bottom of the panel. The Track List and Track View sections appear above the accessory panel on the left and right, respectively.

When you first open the program, the Track List and Track View are empty and the accessory panel displays the Media Explorer. The Media Explorer shows you a directory tree (left pane) of all the storage devices on your system, including floppy drives, hard drives, and CD-ROM drives. It also displays a list (right pane) of any WAV, AIFF, and ACD (*ACID* project) files that are present in the currently selected folder.

You can preview a WAV or AIFF file using the playback controls located at the bottom right of the Media Explorer. If the Auto Preview option is enabled, simply selecting a file will make it play. What's cool about this feature is that you can preview files while the current *ACID* project is playing. *ACID* will do a quick read of the file properties (more about this later) and then stretch and transpose the file to fit the current project's tempo and key. Once you've found a file that you like, you can add it to the project by dragging and dropping it into either Track area or by simply double-clicking it.

When you add a file to an *ACID* project, a new track is created. In the Track List, controls appear that let you set common track parameters, such as name, output device, mute, and solo. One of the controls is a multifunction slider. Its purpose is determined by the adjacent drop-down list. The list lets you choose settings for volume, panning, and effects send levels, which are displayed in dB from –infinity to +12 (±100 percent from center for panning). I'd rather see all the values displayed at once and just have pop-up sliders appear during adjustment. I'd also like to be able to type in specific values, but that's not possible in the current version.

Unlike the tracks in a typical multitrack sequencer, each *ACID* track only contains a single audio file. As with most audio sequencers, the tracks don't actually contain any audio data. Instead, they contain playback events that tell *ACID* how and when to play a track's associated sample file. The Track View displays the events in each track along a time line so you can see when each event occurs and how it relates to the other events. The program works in a way that's similar to the way a sampler works with MIDI. Just as a MIDI Note On event sent from a sequencer instructs a sampler to play an audio loop at a certain time, an event in the *ACID* Track View tells the program to

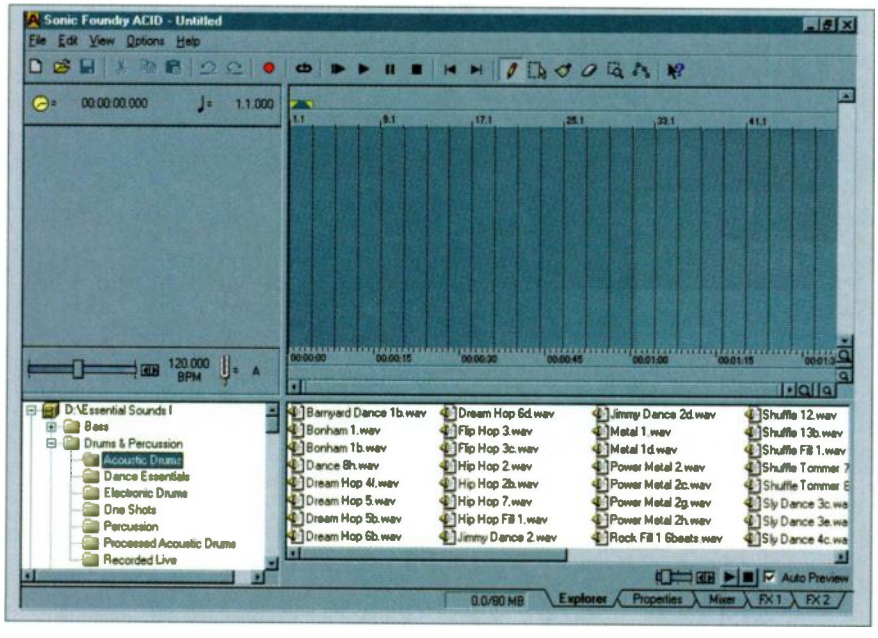


FIG. 1: *ACID*'s interface is divided into three main sections: the Track List (left), the Track View (right), and a multifunction accessory panel (bottom).

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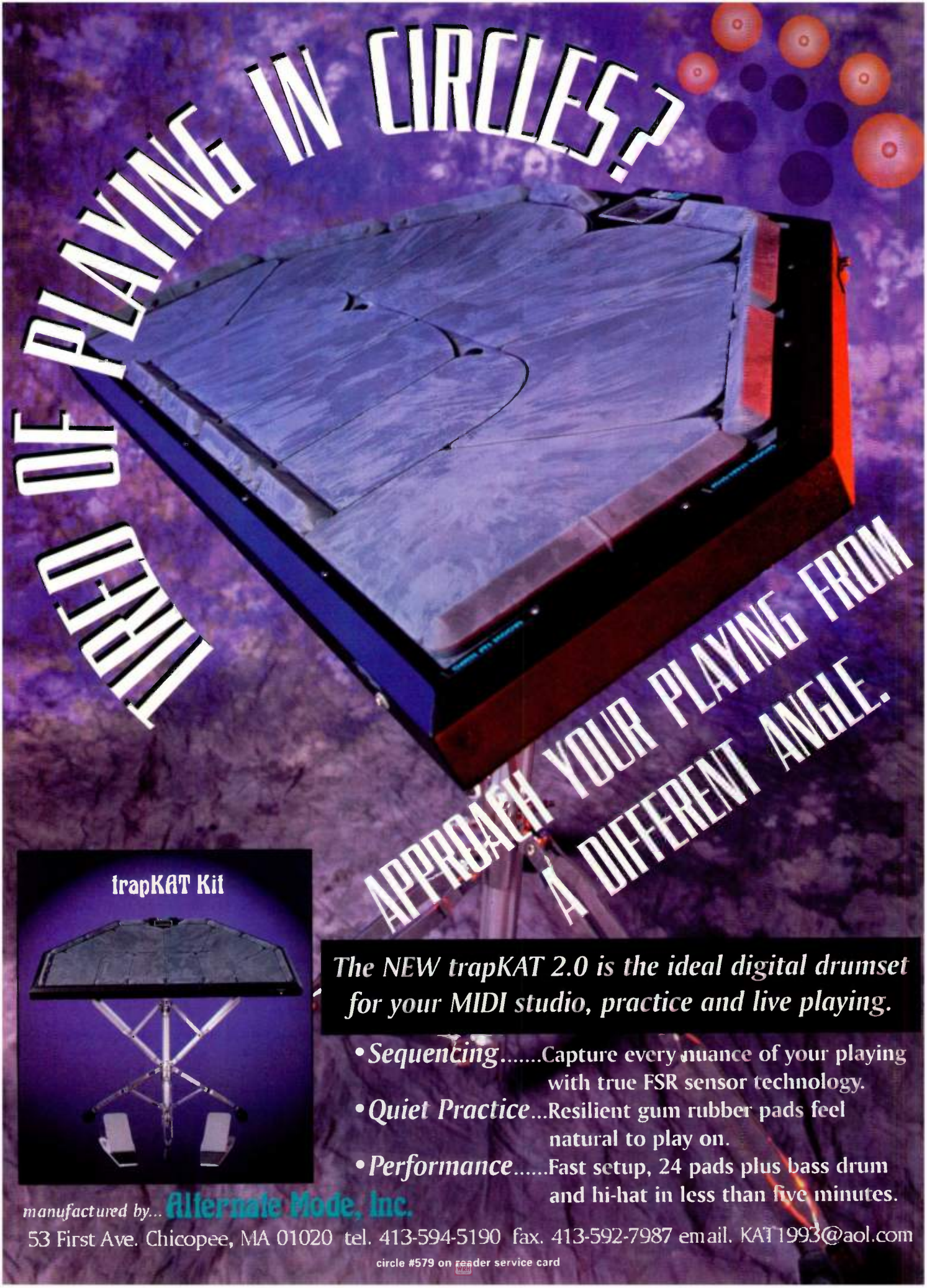

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LET'S GET TRIPPIN'

To create a new track, drag or double-click an audio file from the file list, and the Track View will display an empty event area. Then, to add audio events to the track, use the set of tools located above the Track View. This set, which is also used for editing, includes the Pencil, Paint, Erase, Envelope, Select, and Zoom tools. Erase, Select, and Zoom perform the obvious functions. The Pencil tool allows you to add events to a single track, but it also provides some editing capability. For example, you can use it to change an event's position and length by simply dragging on the event. The Paint tool, on the other hand, doesn't provide any editing functions, but it lets you add events across multiple tracks. Just click and drag anywhere in the Track View and you can paint some interesting random grooves using any or all of the loops you've loaded.

You can navigate a project with the usual horizontal and vertical scroll bars in the Track View. Zoom In and Out buttons are provided as well. As you navigate, two rulers indicate your position in a project. The ruler at the top of the Track View uses bars and beats as units. This ruler provides a fixed measurement no matter what changes are made to a project, including tempo. On the other hand, the Time Ruler at the bottom of the Track View can be set to display a variety of measurements, including Samples, Time, Seconds, Time & Frames, Absolute Frames, and various SMPTE formats. *ACID* also provides a Snap-To function for greater accuracy when positioning events. Unfortunately, it can only be set to snap to the Beat Ruler marks or to musical note divisions—from whole note to sixty-fourth note. It would be better if you could also snap to any of the Time Ruler formats, especially considering that *ACID* can generate and chase to MIDI Time Code.

For quick navigation, *ACID* provides Markers, which are quite simple to use. Just hit the M key on your keyboard, and a Marker is placed at the current cursor position. You can also give each Marker a name and assign it to a number on your computer keyboard. Hit a key (1 through 0), and the cursor jumps to the associated Marker. An unlimited number of Markers is available, but only ten are assignable to hot keys. Although this is a useful feature, it's also a bit limiting. A better solution would be the addition of a cue list and perhaps a play list as well. Being able to set up different playback sections with names and repetitions would not only boost the program's navigational features, but would provide an easy way to test out different arrangements of a tune. As it is now, you can set up a single loop region within a project, but only for editing purposes when you need to listen repeatedly to a specific section and make event adjustments.

ACID
Minimum System Requirements
Pentium 133 or Alpha microprocessor (Pentium II microprocessor recommended for real-time effects previewing); 32 MB RAM; Windows 95/NT 4.0; CD-ROM drive

Because of its real-time nature, *ACID* lets you build and edit your tune as you listen. The Playback controls, located at the top of the window, provide the usual Stop, Play, and Pause buttons. A Loop button continuously plays the loop region mentioned earlier. During playback, you can manually adjust the tempo and key of the project using the controls displayed at the bottom of the Track List. You can also have the tempo and key change dynamically during a tune by placing Tempo Change and Key Change Markers along the bottom of the Time Ruler. Whenever *ACID* reaches one of those Markers during playback, it automatically adjusts the tempo and key of the project, and it stretches and transposes its samples to match.

As I tested the program, I gave the tempo- and key-change functions a good workout. Overall, the results ranged from good to excellent. Depending on the material, even large changes often sounded good. Key changes on instrumental samples sounded great when transposed down as much as a fifth. Transposing up as little as a second, though, introduced some slight, echolike anomalies. Vocal

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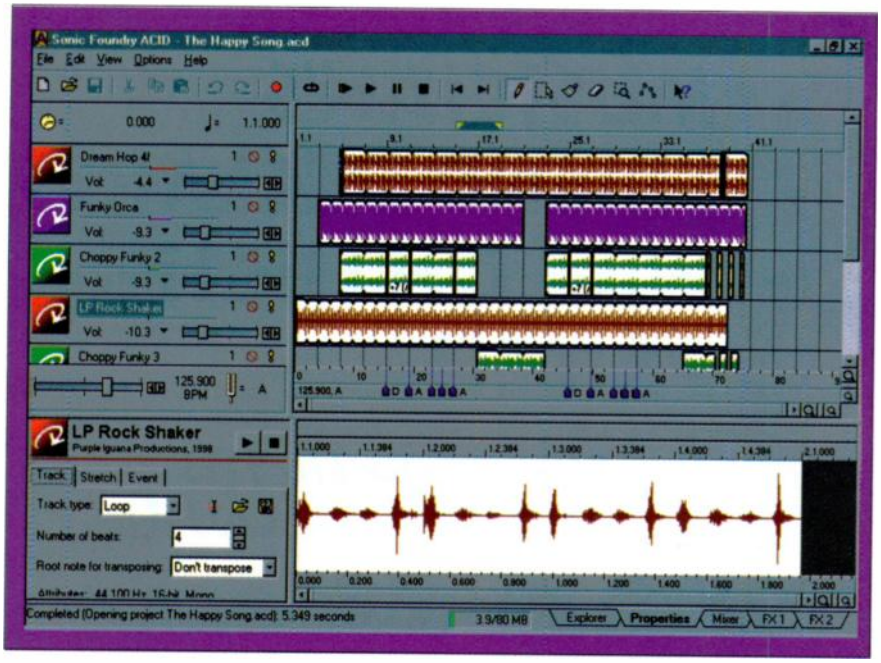

FIG. 2: To *ACID*–ify an audio file, an extra chunk of data is added that contains settings specific to *ACID*. These settings are accessed via the Properties tab at the bottom of the accessory panel.

samples exhibited the usual "munchkin" and "Darth Vader" effects, no matter how little they were transposed up or down. With tempo changes, I attained similar but opposite results. Samples sounded great when boosted by 10, 20, or even 30 bpm. Slowing down some samples (such as those with added effects or ambience) by any more than 5 bpm, however, introduced slight echoing that was especially noticeable on percussive samples.

Dry samples performed much better. They didn't exhibit any echoing until the tempo was reduced by about 12 bpm, and even then, the echo was very slight. As a point of reference, most of the echo effects that I've mentioned were difficult to detect. Furthermore, no matter how much the tempo and key were changed, they never affected each other. And that's just the way it should be.

ACID-IZATION

Though *ACID* can work with any sample that you throw its way, it's better if that sample has been "*ACID*–ified" first. This means that an extra chunk of data is added to the file. The data contains settings specific to *ACID*, and these settings can be examined and edited by clicking on the Properties tab in the accessory panel (see **Fig. 2**). Settings include the type of sample, its number of beats, its root note for transposing, and its "stretching properties." Samples can be either looped or one-shot samples that are loaded into and played back from RAM. The more memory you have, the more loops you can use in your tune. You can also use disk-based samples, which are played back directly from disk and are best suited to long, nonrepetitive audio such as a vocal track.

ACID comes with a wide selection of unprocessed samples, which you can use in your tunes. Sonic Foundry also has a new line of CDs that contain nothing but *ACID*–ified sample files from third-party developers. I tested a few of the latest releases, and a couple of my favorites are *Street Beats* by poogie bell and *Syntonic Generator* by Sound Werx. (You can check out some of the Sound Werx samples on the *ACID* disc.) Of course, you can also create your own *ACID*–ified files by loading an existing sample or by recording a new one.

If you load an existing sample, *ACID* analyzes the file and makes a guess as to the number of beats, the tempo, and the stretching properties. Most of the time, *ACID* guesses correctly, and all you need to do is set the type of sample (loop or one-shot) and root note. If you hear audio anomalies, you may need to tweak the stretching properties a little (see **Fig. 3**). The stretching properties markers are set automatically by *ACID* throughout the audio file to designate detected subdivisions of beats. The stretching properties function is the key to *ACID*'s great tempo adjustment abilities. You can add, edit, or delete the stretching markers if you think that *ACID* has detected any transients incorrectly.

If you want to record a new sample, all you have to do is hit *ACID*'s Record button. A dialog box appears, enabling you to enter the name of the new file,

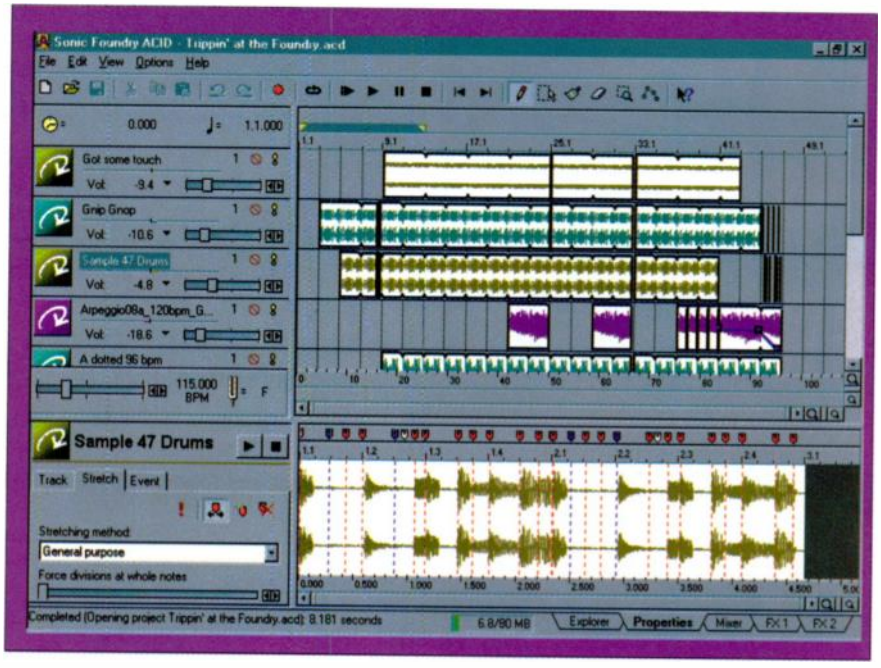

FIG. 3: The stretching properties are the key to *ACID*'s excellent tempo adjustment abilities.

A real pro is pro Reality™

"I'm a big fan of the concept of software, the concept of synthesis, and the concept of Reality."

Rob Arbittier, co-founder of Noisy Neighbors Productions, writes and records music for TV and for movie trailers and has numerous album production credits. Much of this work is done in his state-of-the-art digital home studio. With a busy schedule of music production at multiple levels, Rob counts on the best tools to complete his work. He needs power, reliability, and performance. That's why he chose Reality as a crucial part of his studio.

"I've always been fascinated with the original synthesizers, the modular patchable synthesizers, and the evolution from those giant Moog modular systems to what we have today. The thing that got me excited about Reality was that I knew it was going to be very flexible and that it would model in software the approaches to synthesis that I've always loved in hardware. It's a new approach to the kinds of sound generation that I've used for many years.

"I love the fact that it's so easy to design sounds on the screen. I use it the way I would have used a modular synthesizer 20 years ago. It's a very quick, easy way for me to design a sound from scratch.

"I use Reality as a stand-alone synthesizer because I don't like having my computers multi-task during a session. I like being able to look up at any one screen and see everything that computer is doing during the session. I keep it on the PC that's right next to the Cubase VST™ PC so while the sequencer is playing I can be fiddling with the sounds on the Reality PC. I have Reality installed on a machine that has a high quality digital output. When you hear Reality through a digital out, the richness of the sound it generates is just amazing."

Rob's work can be heard on albums by many artists, including Stevie Wonder, Michael Jackson, Whitney Houston, Ray Charles, Stevie Ray Vaughan, and Diana Ross. He has scored hundreds of television commercials for companies such as Coca Cola,® Anheuser-Busch,® Kodak,® and the four major television networks. Recent film trailers include *Godzilla*, *Les Miserables*, and *One Night Stand*.

For more information about Reality professional synthesizer software and Rob, visit our website at www.seersystems.com or call us toll-free at 888.232.7337.

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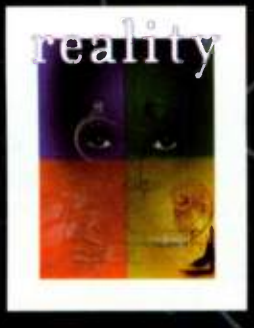

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the record-from position, the recording device, the sample size (16- or 24-bit), the number of channels (mono or stereo), and the disk folder in which to store the new file. A meter displays your input level. Hit the Start button, and *ACID* begins simultaneous playback and record (if you have a full-duplex sound card or multiple sound cards). The only disappointing thing about this feature is that, during the recording, the dialog box remains on the screen and the rest of the program is unusable. It would be more useful if you could see the tune scrolling along as you record.

After you finish recording, you can edit the file in an external audio editor, such as Sonic Foundry's *Sound Forge*, by clicking on the Edit button in the Properties display. Owners of *Sound Forge* version 4.5 can *ACID*–ify files from within *Sound Forge* itself by using its new *ACID*-specific functions in the Special menu. That's a very cool feature.

DIRECTX EFFECTS

In addition to its other real-time features, *ACID* allows you to apply DirectX-compatible plug-in effects to any of the tracks in a project. You apply effects to a track using its associated multifunction slider (mentioned earlier) in the Track List. A total of eight plug-ins can be loaded at once, and their parameters are accessed with the FX tabs at the bottom of the accessory panel. Which parameters are displayed depends on the plug-in, but the standard effects settings include the plug-in selection itself and the effect preset. There are also faders and meters to adjust the effect input level, and you can assign an output device. That's a great feature if you have multiple outputs and want to do some outboard processing on the effect itself.

Of course, the number of effects that you'll be able to use depends on the amount of processing power that a specific plug-in needs and how fast your CPU is. *ACID* doesn't actually come with its own plug-ins, but Sonic Foundry has several plug-in packs available for purchase. *XFX 1*, *XFX 2*, and *XFX 3* combine some of the best effects from *Sound Forge*, in DirectX format. If you want to try them out before you make a purchase, there are demos on the *ACID* CD. During my tests, I was able to get only two effects running on my Pentium 150/MMX with 80 MB of RAM before I started to get some serious skipping during playback. Suffice it to say that Sonic Foundry's minimum recommendation of a Pentium 133 is nowhere near enough if you plan on using multiple effects.

MIX IT UP AND POUR

After you've finished adding and editing events in your new project, you can polish it up with some envelopes. Envelopes give you dynamic control over volume, panning, and effects-send levels. You add envelopes to an event by

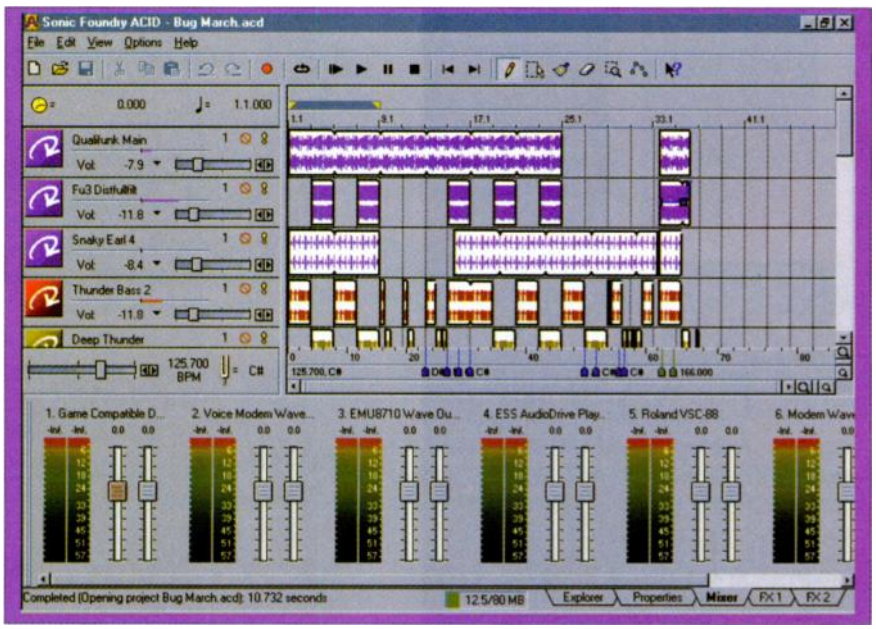

FIG. 4: The Mixer displays faders and level meters for all of the available sound devices.

right-clicking on an event with the Envelope tool and selecting the type of envelope you want from the pop-up menu. A blue line representing the envelope appears inside the event. You can then adjust the envelope by dragging on it with the mouse, and you can add as many envelope points as you like by simply double-clicking directly on the event.

You can adjust the final mix of your project even further by manipulating the actual output levels of your sound devices. These are accessed with the Mixer tab at the bottom of the accessory panel (see **Fig. 4**). The Mixer displays all of your available sound devices along with faders and level meters for each.

Finally, you can save your project as an *ACID* (ACD) file, so you can do more editing in the future. It's also possible to export the tracks in an *ACID* project as separate audio files for use in other programs such as digital audio sequencers.

For a stereo version of your mix, you can save it as a mixed AIFF file, a mixed WAV file, or even a mixed Netshow Active Streaming Format (ASF) file for posting on the Internet. I'd like to see a RealAudio output option added here, too. It's available in *Sound Forge*, so why not *ACID*?

FINAL BEAT

Sonic Foundry calls ACID "a breakthrough loop-based music production tool," and they're right on the beat. *ACID* provides a unique and yet familiar environment for building songs with audio sample loops. A few areas could be improved, such as the documentation. (The software includes a 20-page Quick Start Guide; the full manual is available only on disc as an Adobe *Acrobat* PDF file.) The included tutorial is a description of the available features rather than a step-by-step guide for using the tools to build an actual project.

Nonetheless, *ACID*'s ease-of-use and powerful audio stretching and transposing capabilities far outweigh the small concerns I've presented here. If loop-based arranging is what you do, then *ACID* is the program you should be doing it with. You'll have to hear it to believe it.

Scott R. Garrigus *has* ACID*–ified his whole collection of audio samples. You can hear some of his* ACID *tunes on his Web site at www.garrigus.com.*

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FEATURES ■■■■
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PROS: Intuitive user interface. Excellent real-time functionality. Automatic sample stretching and transposing. Support for DirectX plug-ins. Support for multiple sound cards.

CONS: Limited number of navigational Markers available. Requires a powerful CPU for multiple effects. Pitch shifting vocal samples causes "munchkinization." Snap-To function doesn't work with the Time Ruler.

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SPL

STEREO VITALIZER JACK

An affordable psychoacoustics equalizer puts killer mixes within reach.

By Brian Knave

The original SPL Vitalizer got rave reviews all around, including one in this magazine (see the December 1994 **EM**). Our reviewer's only gripe was the price: at $1,299, the unit was simply too expensive for the average personal-studio buyer.

Since that time, Sound Performance Laboratory, a German company, has been refining and expanding its Vitalizer line and now offers seven different models (see the sidebar "All in the Family"). These include two updates of the original; a premium, all-tube version; two Vitalizers optimized for broadcast; another optimized for hi-fi and home-theater applications; and the Stereo Vitalizer Jack, SPL's least expensive Vitalizer.

Priced at less than a third of the cost of the original Vitalizer, the Stereo Vitalizer Jack puts SPL's much-touted psychoacoustic processing within reach of the personal studio. Let's take a look at how this unit performs and see if it's worthy of devouring a chunk of your studio dollars.

VITAL INFORMATION

SPL Vitalizers work their psychoacoustic magic principally via reconstruction of phase relationships between various instruments in a mix. According to SPL, when multitracked (i.e., individually recorded) tracks are combined in a stereo field, sonic losses are incurred because individual phase responses add up or get canceled. The Vitalizer's primary processor analyzes the spectral content and amplitude of the input signal and dynamically assigns a very slight delay (1.5 to 2.5 ms—technically a phase shift) to the louder parts. This unmasks and helps clarify previously obscured sounds, increasing detail and depth.

Vitalizers also employ filters for emphasizing fundamentals (bass) and upper partials (high harmonics) and for balancing odd and even harmonic content. Unlike "exciter" circuits, however, which artificially generate "missing" harmonics, Vitalizer processing uses only information drawn from the original signal. According to SPL, this approach "produces a more natural, inherently nonfatiguing sound" than exciters.

The Stereo Vitalizer Jack utilizes the same principles as the more pricey Vitalizers, and, according to SPL, is manufactured to the same quality standards (including the use of 1 percent metal-film resistors, MKT capacitors, and a torroidal transformer). The control topography has been streamlined as compared with the more expensive Vitalizers, but the unit operates in basically the same fashion.

JACK IN THE BOX

The Stereo Vitalizer Jack comes in a 1U rackmount box decorated with Jackson Pollack–like "splatters" of deep purple over a black faceplate. Considering the sophistication of the processing power within, the Jack's user interface could scarcely be simpler. You get five knobs, an active/bypass switch, and an illuminated on/off switch. (On my test unit, the LED behind the on/off switch came on sporadically after power-up, but the processor itself worked without fail throughout the 2-month test period.)

The unit's rear panel (see **Fig. 1**) is also a no-brainer. It provides stereo inputs and outputs, both on ¼-inch and RCA jacks (hence the name Jack), and a jack for connecting the power cord (no wall wart!). There's also a switch for toggling between 115V and 230V operation.

The Stereo Vitalizer Jack is easy to operate. There are no level meters, configuration chains, or buried menus to mess with, and the one set of controls handles processing for both channels, ensuring that both sides are processed identically. I was concerned about the lack of an input control and clip LED (amenities offered on the higher-priced Vitalizers), but their lack didn't prove problematic. I threw dozens of hot mixes at the thing and never heard any distortion, so evidently, there's plenty of headroom.

The manual, though intelligibly written and useful for explaining what's going on with each process, is practically extraneous: you can pretty much plug the unit in and start enhancing your mixes right away, simply by listening to the results. However, the manual does come in handy as there are a few counterintuitive control interrelationships that are not immediately obvious.

KNOB KNOWLEDGE

The five knobs, from left to right, are labeled Bass, Mid-Hi Tune, Process, Brilliance, and Stereo Expander. The Brilliance and Stereo Expander processes operate independently via their respective knobs, just as you would expect. The Process knob, however, does double (or quadruple) duty, simultaneously controlling both the ratio and amount of Bass and Mid-Hi Tune processing.

The detented Bass knob, which is flat (no process) at the twelve o'clock position, offers two sound colors: "soft" to the left and "tight" to the right. Settings in the soft range cause the bass to sound warmer and fuller with more resonance and sustain. Settings in the tight range create a drier, tighter, more percussive sound. Generally, the Process control needs to be positioned around twelve o'clock or higher before the bass process becomes audible. Differing qualities of either sound can be obtained by varying the relationship between the two knobs.

The Mid-Hi Tune knob determines the shelf frequency of a broadband shelving filter above which all frequencies are processed. The shelf control ranges from 1 kHz (hard right) to 20 kHz (hard left).

The SPL Stereo Vitalizer Jack is an inexpensive, easy-to-operate psychoacoustics equalizer that can take your mixes to another level. It works by manipulating harmonic content and reconstructing phase relationships between instruments.

Stereo Vitalizer Jack Specifications

Stereo I/O	RCA and ¼-inch jacks
Frequency Response	10 Hz–150 kHz
THD and Noise	0.002% @ 1 kHz; 0.02% @ 10 kHz
S/N Ratio	-95 dBu
Dimensions	1U x 6.5" (D)
Weight	3.4 kg

Again, the amount of processing is determined by the Process knob. In addition to determining the ratio between Bass and Mid-Hi Tune processing, the Process knob also controls the damping intensity of dominant mid frequencies that fall below the Mid-Hi Tune shelf frequency. This midrange damping helps compensate for the inherent nonlinearity of human hearing (see the sidebar "Jack of All Volumes").

The Brilliance knob controls a coil filter as well as phase relationships of high frequencies and harmonics. Turning the knob to the right emphasizes the perception of high and harmonic frequencies. The process is inactive when the knob is positioned hard left.

The Stereo Expander knob controls the width of the stereo image, widening the subjective soundstage of any stereo source. This circuit employs phase principles, too. Basically, it detects off-center signals, inverts their phase, and then feeds them back to the opposite channel.

PROCESS PARTICULARS

The relationships between the Stereo Vitalizer Jack's knobs allow for practically any combination of the four processes, as well as single-process applications. For example, if you want to use the Stereo Expander by itself, simply turn off the Brilliance and Process controls (assuming, of course, that the bypass switch is set to Active). The Bass process is the only one that can't be used in isolation, because as soon as Process is turned up, the Mid-Hi Tune process is activated, as well. This limitation, however, didn't prove problematic in any of my applications.

It was hard to get used to the way that the Mid-Hi Tune shelving filter works in relation to the signal and to the other processes. Basically, the lower the setting, the brighter the sound becomes, because an increasing number of frequencies are included in the process. At higher settings, the sound gets darker.

I generally used a Mid-Hi Tune setting somewhere between 1.5 kHz and 3.5 kHz, which helped brighten and articulate both mids and highs. But you can also use the filter to attenuate overly bright-sounding mids by setting a higher shelf frequency, say, 8 kHz or above. The only problem then is that vocals (and other midrange instruments) get darkened and lose some presence. That's where the Brilliance circuit comes into play. It recovers certain frequencies lost in the mid-damping process and lets you bring them back into the mix, thereby restoring brightness and presence to the vocals.

GET BACK, JACK

As the name implies, the Stereo Vitalizer Jack is designed for processing stereo signals, and this is what it does best. (The manufacturer recommends connecting the Stereo Vitalizer Jack via insert cables patched into your console's main or subgroup inserts.) However, I also got good results using it to process single (mono) sources during both tracking and mixdown.

For example, I used the Brilliance control alone to add an airy presence while recording vocals with my AKG C414. Also, on mono voice-overs I recorded for a client's Web site, I used the Mid-Hi Tune and Brilliance circuits to sweeten the voice track during mixdown. Curiously, the Stereo Expander circuit also enhanced the track. I attributed the enhancement partly to gain added by the Stereo Expander circuit, but there seemed to be something else going on, as well. Just be sure, when using the Stereo Expander in mono, that the other insert cable isn't patched into a channel with a different instrument on it, because the circuit will return that instrument (phase inverted) to your mono track.

JACK OF ALL VOLUMES

As demonstrated by the Fletcher-Munson "curves of equal loudness," the human ear is better attuned to midrange frequencies than to lows and highs, particularly at low listening levels. That is, the lower the volume of a signal, the more difficult it becomes to hear low- and high-frequency content. Conversely, as the signal becomes louder, we are better able to hear the lows and highs.

The curves depicted in **Figure A** indicate the sound pressure levels (SPLs) required for different frequencies to sound equally loud to most people with normal hearing. Each curve is identified by the SPL at 1 kHz (also known as the *phon* of the curve).

Essentially, the Vitalizer's amplitude-controlled phase shifting helps "linearize" the Fletcher-Munson curves. The result is a better balance between the frequency ranges at different monitoring levels. This improves the clarity, strength, and fullness of the audio signal, making it easier to hear all parts of the music—no matter where the playback volume is set.

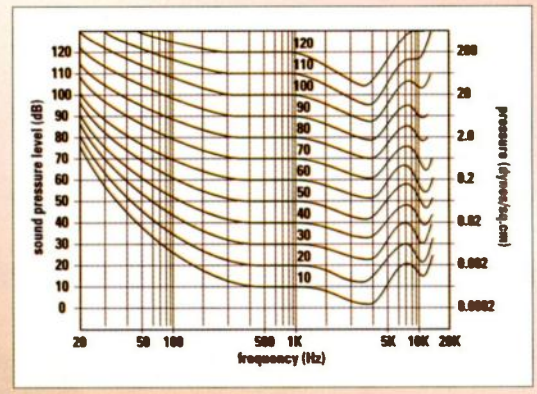
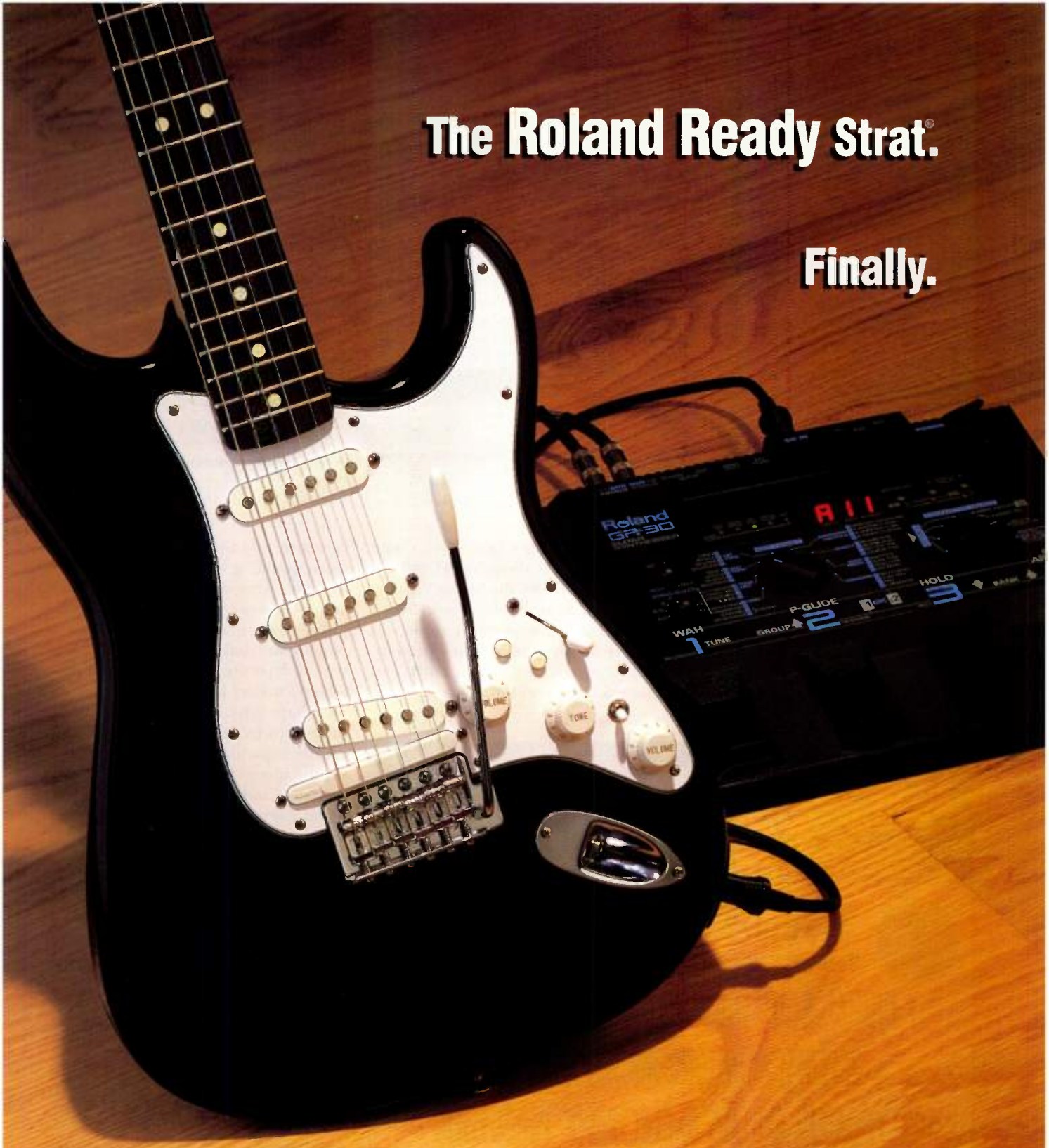


FIG. A: The Fletcher-Munson curves show optimum SPLs for the average person to hear all frequency ranges equally.



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ALL IN THE FAMILY

The SPL Vitalizer line includes three other studio models: the Stereo Vitalizer MK2 ($699), the Stereo Vitalizer MK2-T ($899), and the Tube Vitalizer ($3,299). Shown is the Stereo Vitalizer MK2-T, which is the same as the MK2 but includes a tube-output stage and a special LC Filter (said to provide extra warmth, punch, and presence). All three models provide more control than the Stereo Vitalizer Jack by inclusion of an input-level control (Drive) with clip LED, a compressor section for the bass, and a tunable High EQ control with dedicated Intensity knob. The Tube Vitalizer goes even further by providing a tube-line preamp, a compressor for the highs, switchable LC Filters, dual VU meters, and dual output-attenuator knobs.

I had a chance to compare the MK2-T to the Stereo Vitalizer Jack and, not surprisingly, found that it did all the same things (and more), only better. The extra knobs allowed for more extensive and precise spectral sculpting and the overall sound was sweeter, tighter, and more focused.

In a 2-month period, I used the Stereo Vitalizer Jack on practically every mix that came out of my studio, always with impressive results. (After hearing what the unit could do, my repeat clients *insisted* on Vitalizer processing from there on out.) Among stereo enhancers, this is the closest thing to a "magic box" I've come across, at least in the below-$1,000 category. Properly set, the Stereo Vitalizer Jack was able to improve every mix I threw at it—in ways that I could not achieve by any other means I know of. Everything sounded richer, fuller, and more detailed, yet with no added artifacts and only a bit of hiss. Highs were more sparkling without sounding brittle, mids were dramatically clarified, and bass, depending on the setting, sounded warmer and fuller or more punchy and defined. The amount and type of bass control afforded by the Stereo Vitalizer Jack is very impressive.

Perhaps most dramatic, though, is the Stereo Expander. This circuit can greatly expand the soundstage, making the spread of instruments blossom magically and appear almost 3-dimensional. Consequently, by creating a bigger "space," it allows for increased discernibility of individual instruments, including any effects applied to the instruments. After applying the Stereo Expander, I often found myself turning down the effects returns to compensate for the increased clarity—a move that could help reduce the amount of noise being added to a mix.

The Stereo Vitalizer Jack is also very useful for stereo sources other than complete mixes. For example, I used it to process stereo-miked instruments (this worked great on drum overheads) and subgrouped vocal harmonies. You could also use it to enhance samplers and synths—an application that would work well live, too. I also used the Stereo Vitalizer Jack to "remaster" a favorite old cassette tape that was on the brink of collapse. Impressively, I was able to make a second-generation tape that sounded as good as or better than the original (better bass and overall clarity, but, naturally, with additional hiss).

I suspect the Stereo Vitalizer Jack would prove a formidable weapon in

FIG. 1: If you can't figure out the rear panel of the Stereo Vitalizer Jack, perhaps your future is not in audio.

any DJ's arsenal, considering that it can readily enhance the sound of CDs or vinyl. I didn't get a chance to audition the unit in a live sound-reinforcement application, but I bet it would kick butt there, as well.

DON'T JACK ME AROUND

When an audio tool performs this well, one can't help but get suspicious. Was it possible, I wondered, that I was being hoodwinked by psychoacoustic sleight of hand?

One thing I worried about, of course, was how well the mixes would translate to other systems. So I ran off several processed mixes onto cassette tape and played them in various boom boxes, car cassette players, and friends' stereo systems. Happily, the mixes sounded fine no matter where I played them.

My other big concern was mono compatibility. After all, you never know when a mix might end up on AM radio or played through a ceiling speaker at the dentist's office. After carefully comparing mixes processed by the Stereo Vitalizer Jack (using both individual and multiple processes) to a "dry" mix (i.e., an identical mix but without Vitalizer processing), I discovered that the Stereo Expander was the only process that consistently led to a noticeably different tonal content when the mix was collapsed to mono.

I found that a mix processed with the Stereo Expander, when played back in mono, suffered a reduction of midrange content and clarity. For example, the very clear, forward mids discernible on an acoustic guitar in the stereo mix seemed to shrink or step back a bit when the mix was switched to mono. The enhanced sense of depth was diminished, too.

Interestingly, though, when I compared the Vitalizer mono mix to the dry mono mix, the Vitalizer mix still sounded better. From this, I deduced that the losses incurred from collapsing a "Stereo-Expanded" mix to mono were losses *only* in relation to the "Stereo-Expander" mix played in stereo. In other words (and somewhat obviously), the Stereo Expander process creates an effect that largely disappears in mono, yet its disappearance doesn't detract from the original signal content. But it appears that Vitalizer mixes are otherwise mono-compatible.

ENOUGH'S ENOUGH

One problem inherent to enhancers in general, and psychoacoustics in particular, is how quickly the ear acclimates to the altered sound. After twenty minutes or so of sustained listening to a processed mix, even extreme settings can start to sound "normal." Although the Stereo Vitalizer Jack is less dangerous in this regard than some other stereo processors I've used, moderation is still advised. The key is to make frequent use of the active/bypass switch (to A/B the processed and original signals) and to take regular breaks from mixing so you can come back to a session with fresh ears before finalizing a processed mix.

After using the Stereo Vitalizer Jack on numerous mixes, I concluded that the various processes tend to sound best at medium settings. Indeed, it would appear that SPL designed the unit to achieve a kind of one-size-fits-all sound when the knobs are all set at around twelve o'clock (excepting the Bass knob, of course, which is flat at that setting). Sometimes I used a little more or less of one process or another, depending on the mix; but in general, 50 percent gets you there. How's that for ease of operation?

JACKPOT

Here's one reviewer who is sold on psychoacoustics—at least the type being peddled by Sound Performance Laboratory. The Stereo Vitalizer Jack has made a really tangible, musical difference in my final mixes. Highs have more sparkle, mids are more intelligible, and bass is fatter and warmer sounding. Also, the stereo field is magically expanded, leaving more "space" in which to hear the increased clarity and detail of the music. At this point, it would be hard to go back to mixing without a Vitalizer around.

For the money, I don't know of a better spectrum-enhancing, home-mastering processor on the market. The Stereo Vitalizer Jack not only sounds great, it's also a cinch to operate. Of course, there's always the risk of overkill with psychoacoustic processors. But used judiciously, the Stereo Vitalizer Jack will confer sonic delights on your mixes without sizzling them to a crisp. ●

SOUND PERFORMANCE LAB
Stereo Vitalizer Jack psychoacoustic equalizer
$399

FEATURES ■■■■
EASE OF USE ■■■■■
AUDIO QUALITY ■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Quick, easy, and musical spectral-enhancement and home-mastering tool. Improves clarity, detail, and depth. Exceptional bass control and stereo-field enhancement. Very affordable.

CONS: No input control or metering. Subtle psychoacoustic process is easy to overdo. Adds modicum of noise to mix. Not completely mono-compatible.

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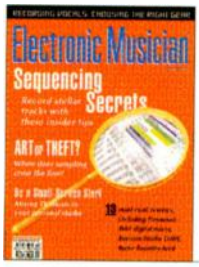

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Steinberg tel. (818) 993-4161; fax (818) 701-7452; e-mail info@steinberg-na.com; Web www.us.steinberg.net

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VOYETRA TECHNOLOGIES

DIGITAL ORCHESTRATOR PRO 3.01 (WIN)

Affordable, easy-to-use audio and MIDI sequencing.

By Zack Price

Desktop musicians always seem to be on the prowl for that elusive software package that offers plenty of powerful features without costing an arm and a leg. You usually get what you pay for, but some programs deliver a lot for a modest sum.

Voyetra's *Digital Orchestrator Plus* (reviewed in the December 1996 issue of **EM**) was a surprisingly good digital audio sequencer for about half the price of its high-end competitors. Now, Voyetra has released its heir apparent, *Digital Orchestrator Pro*. This latest incarnation continues Voyetra's commitment to providing quality digital audio sequencing in an easy-to-use, cost-effective package.

SET UP AND ENTER

Installing *Digital Orchestrator Pro* is a snap. Just insert the CD-ROM, run the setup program, and type in the product ID number when asked. (The CD-ROM also contains demo files, MIDI rhythm files, and WAV files that you can import into the program.) Installation is fast and nearly automatic.

After you install the program, set up the MIDI interface and digital audio device that you intend to use with *Digital Orchestrator Pro*. To assist you in that endeavor, Voyetra provides a little utility called *SoundCheck* (see **Fig. 1**). With this application, you can test whether your MIDI interface, sound card, and CD-ROM drive are working properly.

SoundCheck works fine with most standard multimedia cards; however, like many utilities of its type, it doesn't always work as well with more specialized devices. For example, *SoundCheck* stated that my Frontier Designs WaveCenter card wasn't working properly when, in fact, it worked fine with *Digital Orchestrator Pro*. On the other hand, *SoundCheck* did a good job of recognizing my Creative Labs Sound Blaster AWE 64 Gold card. Unless you're the overly cautious type, I wouldn't worry much about the utility, because if your sound card worked before installing *Digital Orchestrator Pro*, I can't imagine it not working with the program.

To set up for audio, access Digital Audio Options and select the audio devices that you want to use. *Digital Orchestrator Pro* lets you choose up to twelve stereo audio output "ports" and one stereo input port for a theoretical total of 24 physical channels of audio output and 2 physical channels of audio in. (The actual number you can use depends on your sound card, of course.) Despite the program's limitation of 2-channel input, *Digital Orchestrator Pro* is quite capable of dealing with multichannel cards, such as the Event Darla, SEK'D ARC 44, and Gadget Labs Wave/4. These cards use drivers that group their multiple outputs as stereo pairs, which appear to digital audio programs as separate stereo devices. This can be quite handy when using *Digital Orchestrator Pro*, as we will see later. More often than not, though, the typical *Digital Orchestrator Pro* user will have only a stereo audio card, which provides two audio channels in and two out.

MIDI MY WAY

After setting up your audio card, you'll want to select your MIDI devices. The program supports 256 MIDI channels, divided among sixteen MIDI ports. That's probably more than necessary for the average user, but it's nice to have that many MIDI ports in case they're needed.

There are numerous options for configuring MIDI usage. For example, those users with parallel-port MIDI interfaces will want to use the Close Drivers While Printing option if they print out musical notation from *Digital Orchestrator Pro*. Another handy option is Keep Drivers Open While Inactive. Ordinarily, *Digital Orchestrator Pro* deactivates the MIDI In and Thru portion of the drivers if you switch to another program while *Digital Orchestrator Pro* is open. This option allows you to keep the drivers open even when you switch. However, this may cause potential conflicts and system instability if the two programs try to access the same MIDI ports at once, especially when using MIDI interfaces whose drivers are non-multitasking.

Once you've selected your MIDI devices, you can assign your instrument patch lists to the appropriate MIDI channels and ports. Just select Patch Map Setup from the Options menu, and select the desired MIDI port and the appropriate instrument list for that port. Unfortunately, the Patch List feature has a couple of genuine weaknesses. First, *Digital Orchestrator Pro* lists the

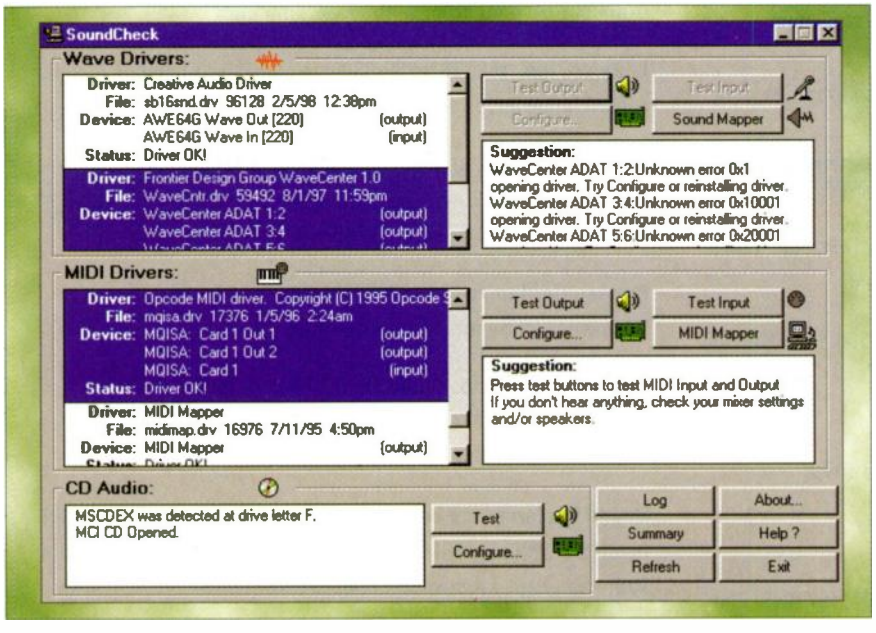

FIG. 1: The *SoundCheck* utility tests whether the MIDI and audio devices in your system are working properly.

The world's most dazzling workstation and the wildly acclaimed Z1 synth have just joined forces. The new V3 gives you Trinity's PCM and Z1's sound modeling architectures in one keyboard. Plus, new combinations and an additional 64 programs. The Z1 board is also available as an option for current Trinity owners.

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main sounds of the Roland GS and Yamaha XG sound sets but doesn't list all of their variations. Although it's easy to add patches with a text editor, many users will probably be using GM, GS, and XG devices, so it's only reasonable to expect that the complete patch lists (with variations) should be included as part of the package. There's no point in making people do more setup work than is absolutely necessary.

Of course, even if these were included, *Digital Orchestrator Pro* would still be unable to automatically call up the variations. Currently, *Digital Orchestrator Pro* lets you access program variations by inserting the proper values in the Bank Most Significant Bit and Bank Least Significant Bit columns in the Main screen. Although entering these values does switch the sound, it doesn't change the name in the Patch column. That means, for instance, that calling up variation 64 in an XG device will still display the basic name of "SynBass1" instead of the variation name of "Oscar." Listing patch names instead of numbers makes the selection process more convenient, so why not make it truly convenient by ensuring that all patches and variations are directly accessible from the Patch List Selector?

Aside from instrument and module patches, *Digital Orchestrator Pro* also supports the loading and listing of SoundBanks and SoundFonts, which are sample banks found on Turtle Beach cards and Creative Labs Sound Blaster cards, respectively. (Several other systems also support SoundFonts, such as the new E-mu APS card and Seer Systems' *Reality* software synth.)

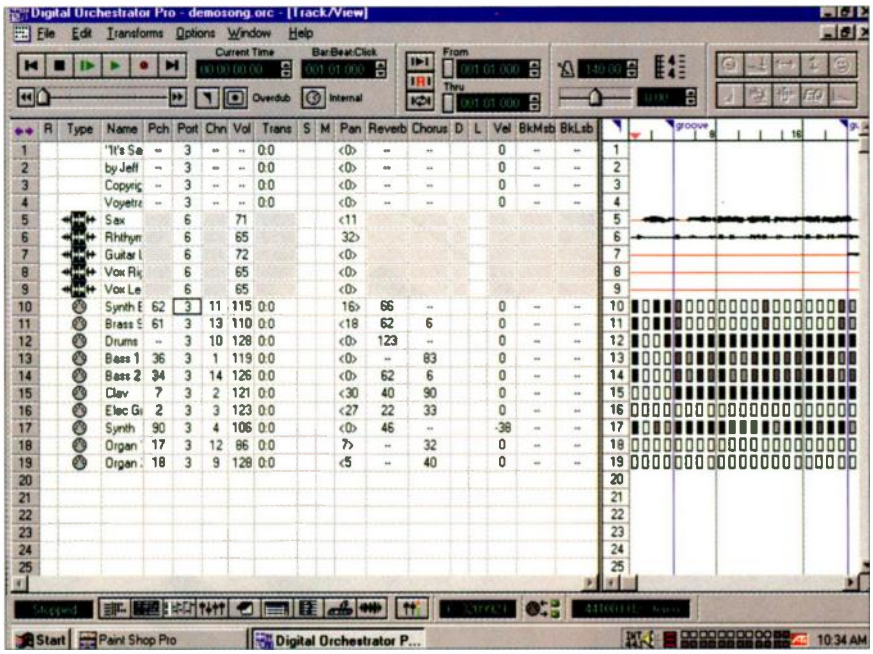

FIG. 2: The program's main screen has been redesigned to improve program navigation and to display information in a more concise manner.

My only complaint is that SoundBanks and SoundFonts are not located in the same menu. SoundBanks for the Turtle Beach cards (which, incidentally, are made by Voyetra) can easily be found under the File menu. SoundFont banks, on the other hand, are located in the Options menu. It seems to me that it would be more intuitive to have both types of patch banks in the same menu.

MORE THAN A PRETTY FACE

Digital Orchestrator Pro retains many of the MIDI and digital audio features of *Digital Orchestrator Plus*. Rather than recite them here, I suggest that you refer to the review in the December 1996 issue of **EM**. You may also want to check out the "Fab Five" digital audio sequencer face-off that appeared in the March 1997 issue of **EM**. For now, I'll focus mainly on the new features along with a few other features that deserve special mention.

Digital Orchestrator Pro sports a new main screen (see **Fig. 2**) that makes program navigation faster and easier. The most visible aspect is the Track/View window itself, which, as before, is split between the Track view on the left and the Measures pane on the right. You can now see an expanded Track view by clicking on the arrows at the corner of the Track grid. Furthermore, you can expand the Measure pane so that it fills the whole window area, which is handy when you want to do large-scale cut-and-paste editing.

In addition, the other program windows (Piano Roll, Mixer, Notation, etc.) can be opened with Quick View buttons located in the Status bar at the bottom of the screen. And some of the digital audio and MIDI editing functions can now be selected from Transform buttons in the upper right corner of the window. The top row of buttons is activated whenever MIDI data is selected for editing; the second row becomes active when digital audio data is highlighted. However, your monitor must be set to at least 800 × 600 resolution to see these buttons. If set to a lower resolution, you'll have to access those functions from the Transforms menu. Also, if you do set your monitor to a higher resolution, be sure not to use the Large Fonts option. *Digital Orchestrator Pro* will not work with large fonts; in fact, the program won't even open.

Digital Orchestrator Pro includes features that were either lacking or not fully implemented in *Digital Orchestrator Plus*. These include an Overdub/Replace/Punch-In recording selector, Marker flags in the Measure view, and

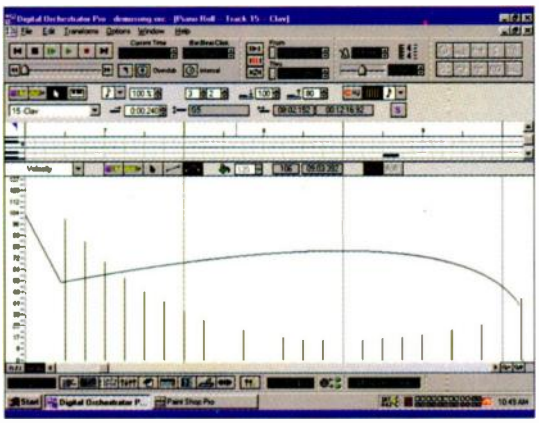

FIG. 3: *Digital Orchestrator Pro* now includes graphic editing of controller and Velocity values. The Curve tool is shown graphically shaping the Velocity values of the notes in this section.

For manufacturer information, please see Contact Sheet, p. 163.

an improved Transport control. Additionally, *Digital Orchestrator Pro* now supports step-time recording, which is available from within the Piano Roll editing window. The Piano Roll window also includes a Snap-To-Grid function that can be set to a variety of quantization values. This same Snap-To-Grid option appears in the Digital Audio editing window, and it's helpful for moving audio data into the precise spot where needed.

The Piano Roll window now allows the user to see and graphically edit different types of controller and Velocity messages. These message values appear as vertical lines or as solid blocks that extend to the next event. You can change individual values by selecting the desired event with the pointer: moving the pointer up increases the value; moving the pointer down decreases it. With the Pencil tool, you can enter precise individual values or freely generate multiple values that change over a period of time.

The Line tool is used to make linear changes in values, and the Curve tool

▼

A new main screen makes program navigation faster and easier.

can shape data in other ways. With the Curve tool, you create a Bezier curve by first selecting the start and end points of a line. Mouse clicks set the curve's length and location. You then drag the pointer up, down, forward, or backward to set the shape of the curve itself. Click the mouse again, and the program changes the values in that section to match the shape of the curve (see **Fig. 3**).

Finally, it's possible to adjust the fill rate of any controller data you insert by using the Pencil, Line, or Curve tools. This helps keep down the number of unnecessary event changes when adding or editing data.

DIGITAL AUDIO DITHERS

As mentioned earlier, *Digital Orchestrator Pro* supports multiple digital audio outputs. Users with multiple-output

sound cards can route their digital audio tracks anywhere they choose to take advantage of outboard effects gear. In fact, it's almost mandatory to have this type of setup with *Digital Orchestrator Pro*, given its limited number of built-in effects.

This is not to say that *Digital Orchestrator Pro* is lacking in basic digital audio editing tools. For instance, you can scale, normalize, fade in/out, reverse, and remove the DC offset from digital audio. *Digital Orchestrator Pro* also retains the Compressor/Limiter/Gate and Digital Delay functions found in *Digital Orchestrator Plus*. Furthermore, Voyetra has added Pitch Shifting, Time Compression/Expansion, and Graphic Equalizing tools to the list. The quality of these effects is adequate.

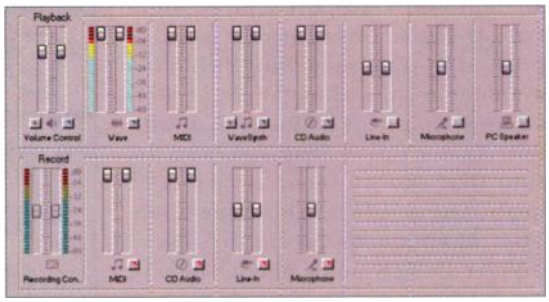

FIG. 4: *Digital Orchestrator Pro*'s Audio System Mixer uses a sound card's multimedia mixer for controlling input/output settings and for monitoring audio record and playback levels.

Unfortunately, *Digital Orchestrator Pro* still lacks a true reverb processor. That can be a problem for users with stereo sound cards who want to apply reverb to some tracks but not to others. If you have a multiple-output card, however, it is possible to route selected audio tracks to an outboard reverb unit, then control your tracks through an outboard mixer.

Digital Orchestrator Pro also lacks a few tools that may not be mandatory but are certainly helpful to desktop musicians. First, it has no tool for extracting tempo from digital audio. Though the manual recommends that you start your sessions working with the MIDI sequence first, there are times when you'll start a session working with audio. You can use *Digital Orchestrator Pro*'s Tap Tempo feature to get a rough idea of the tempo you're working with, but it's not as elegant as extracting the tempo directly from the audio itself.

Second, *Digital Orchestrator Pro* could use a noise-reduction tool. Recording audio in a computer environment does not always produce the most pristine results. It would be nice if the program had noise-reduction capabilities so that tracks could be made as clean as possible.

Next, I wish *Digital Orchestrator Pro* could record single stereo tracks. As with *Digital Orchestrator Plus*, the program records only mono tracks. Considering that some styles of music rely on using stereo samples and loops that are played as digital audio tracks, Voyetra should seriously consider offering the user this option.

Finally, Voyetra should find a better way to monitor the recording and playback levels of audio tracks in *Digital Orchestrator Pro*. Currently, the program creates a Mixer window based on your sound card's existing multimedia mixer (see **Fig. 4**). That may be fine if you have a multimedia card with a mixer applet, but specialty cards don't often have multimedia mixer applets, so you may not be able to monitor levels from within the program.

THE PRO FROM DOVER

Despite my complaints, I really like *Digital Orchestrator Pro*. The MIDI tools, for example, are among its best features,

VOYETRA TECHNOLOGIES
Digital Orchestrator Pro 3.01.05
digital audio sequencer
$199.95

FEATURES ■■■
EASE OF USE ■■■■
DOCUMENTATION ■■■■■
VALUE ■■■
1 2 3 4 5

PROS: Easy to use. Excellent MIDI capabilities. Good documentation. Supports multichannel sound cards.

CONS: Audio editing features a bit too basic. Only records mono tracks. No built-in reverb. No noise-reduction tool. No support for third-party plug-ins.

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Digital Orchestrator Pro 3.01.05
Minimum System Requirements
80486 DX2 66 MHz PC (Pentium 90 recommended); 8 MB RAM (16 MB recommended); Windows 3.1/95/98/NT; full-duplex sound card.

and though it has only a small set of audio processors, it's easy to get around most of the program's audio limitations, as long as you have a multichannel sound card and a couple of outboard effects units. *Digital Orchestrator Pro* is also well integrated, so moving from screen to screen is straightforward. The difficulty with reviewing this program is that it occupies the middle ground between basic and high-end digital-audio sequencers. Because it does so much, there's a tendency to dwell on the things that it can't do, while ignoring the many things that it does quite capably.

Judging the program by its cost/benefit ratio, however, is more of an issue than with the previous version. At $149.95, *Digital Orchestrator Plus* was a bargain. At $199.95, *Digital Orchestrator Pro* is inching into the territory occupied by some mid-level audio sequencers with features that it can't match. Many other programs, even those with a limited number of audio tracks, provide DirectX plug-in capability for real-time effects processing. *Digital Orchestrator Pro* is especially handy for users with less-powerful computers that may not be able to access numerous effects in real time; eventually, however, the program will have to match its competition.

Is *Digital Orchestrator Pro* worth checking out? Definitely, if for no other reason than because it's so easy to use, and that seems to be its strongest selling point. The 354-page User's Guide is clearly written, nicely designed, and well indexed (although it lacks tutorials), making this package an appealing choice for less-experienced users.

Despite its "pro" designation, *Digital Orchestrator Pro* is not particularly well suited for a truly professional environment. However, when it comes to general usage, *Digital Orchestrator Pro* is a fine integrated audio/MIDI sequencer available for a modest price.

Zack Price *doesn't have to do anything he doesn't want to do.*

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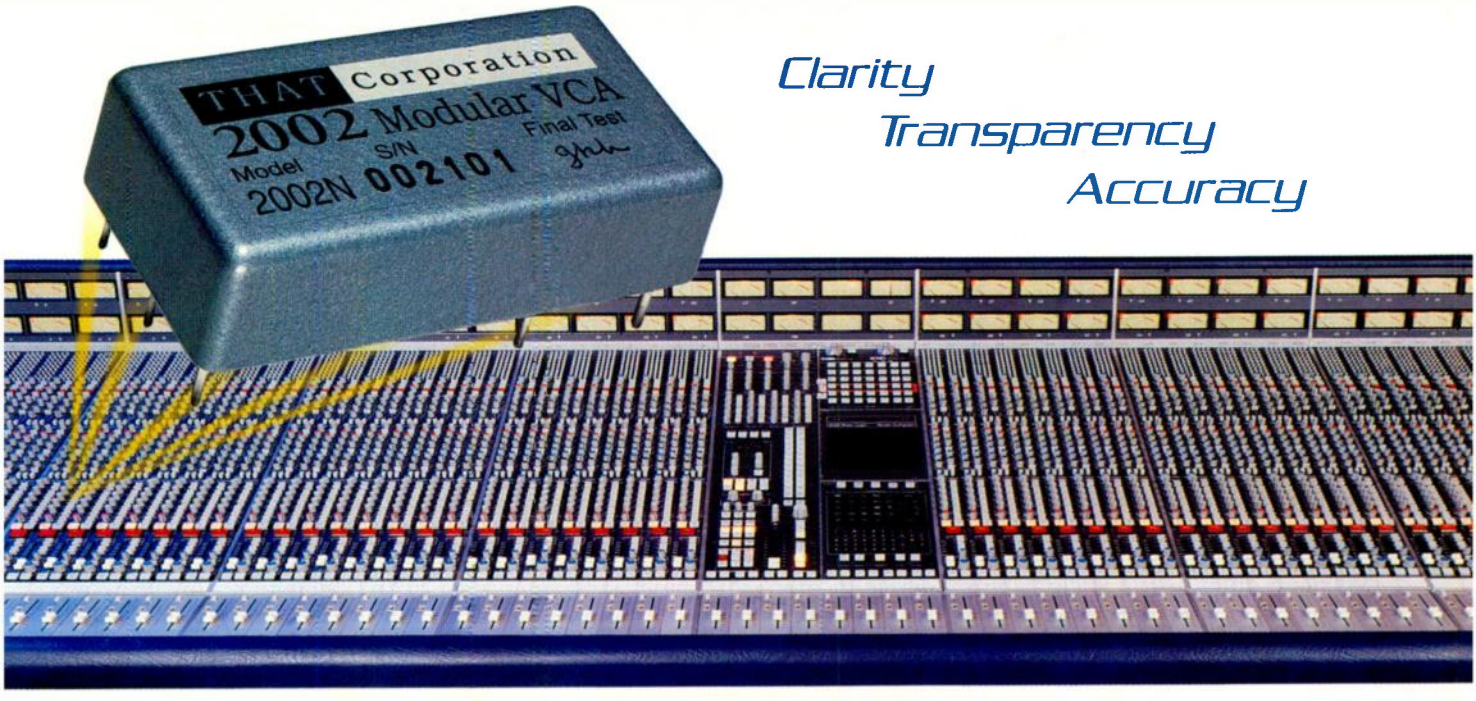

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A. R. T.

TUBE PAC AND TUBE EQ

Quality tube processing on a budget.

By Rob Shrock

FIG. 2: Featuring two shelving and two sweepable bands of tube-based equalization, the Tube EQ is a great companion to the Tube PAC.

Following up on the success of the Tube MP, A.R.T. has added two new models to its line of affordable, tube-based processors. Taking up only a half rack-space each, the Tube PAC (Preamp And Compressor) and Tube EQ are single-channel, hybrid solid-state/tube processors. The units can be used separately or together; together, they provide extensive control of the signal path from source to tape, complete with variable tube warmth and semi-parametric equalization.

TUBE PAC

Like many other hybrid mic preamps on the market, the Tube PAC employs a 2-stage preamp design. The input gain section of the unit is solid-state, and the tube section is isolated to allow controlled overdriving of the tube. The unit is typically very quiet and clean except, of course, at extreme settings.

In addition to a rotary Gain pot that provides 40 dB of input gain, the Tube PAC also has a +20 dB Gain button. With the button engaged, the Tube PAC can provide 60 dB of gain at the XLR input (57 dB for the ¼-inch input). But if you don't need that much gain, it's better to leave the +20 dB button disengaged (0 dB), which will result in improved headroom and a lower perceived noise floor. Fortunately, the 40 dB of gain provided by the rotary pot is sufficient for many applications.

A dedicated Output knob provides an additional 10 dB of gain or 20 dB of cut after the tube stage and compressor. The relationship of the Gain control, +20 dB button, and Output control makes it possible to get varying amounts of tube saturation out of the Tube PAC, while still regulating the level going to tape or to the next device in your signal path. (You can also invert the phase of the output signal with the Phase button, located next to the Output control.)

Metering of the preamp is very basic, yet surprisingly informative and easy to use (see **Fig. 1**). Four LEDs indicate the input level hitting the tubes: a green LED labeled Cln (for Clean), two yellow LEDs labeled Warm, and a red LED labeled Clip that lights when the signal is 6 dB below audible clipping. Of course, audible clipping with a tube device is subjective, but the meters do act as a useful guide for setting input levels.

According to the user's manual, steadily lit Warm LEDs indicate the unit's optimal operating range; however, it's also OK to regularly light up the Clip LED, as this is where much of the more noticeable and desirable soft-clipping of the 12AX7a tube occurs. Finding the just-right relationship between input and output settings can be tricky—beyond a certain point of pleasing saturation, things get ugly fast—but after using the Tube PAC a few times, you'll get the hang of it.

One note of caution: the Phantom Power switch is located directly beside the +20 dB Gain button. On one occasion, while attempting to push the Gain button, I inadvertently switched off the phantom power to one of my condenser microphones. This caused an audible pop through the monitors, and although no damage was done, it's something I don't want to do again. However, this could prove a big problem, especially if you have a device connected to the Tube PAC that does *not* want to see phantom power (e.g., a ribbon mic). Although the buttons are clearly labeled on the silk-screened front panel, the Tube PAC is only a half-rack unit, and the buttons and script are relatively small as compared with standard 1U rack devices. It would have made more sense to me, had the rotary Gain pot been positioned between the Phantom Power and +20 dB Gain buttons.

FIG.1: The new Tube PAC from A.R.T. crams a tube preamp and a tube compressor into an ergonomic, half-rackspace unit.

SOUND OFF

I recorded with the Tube PAC directly into an original Alesis ADAT and was pleased with the results. For one demo, a single vocalist sang both a doubled lead and six additional background parts. To help create tonal distinctions

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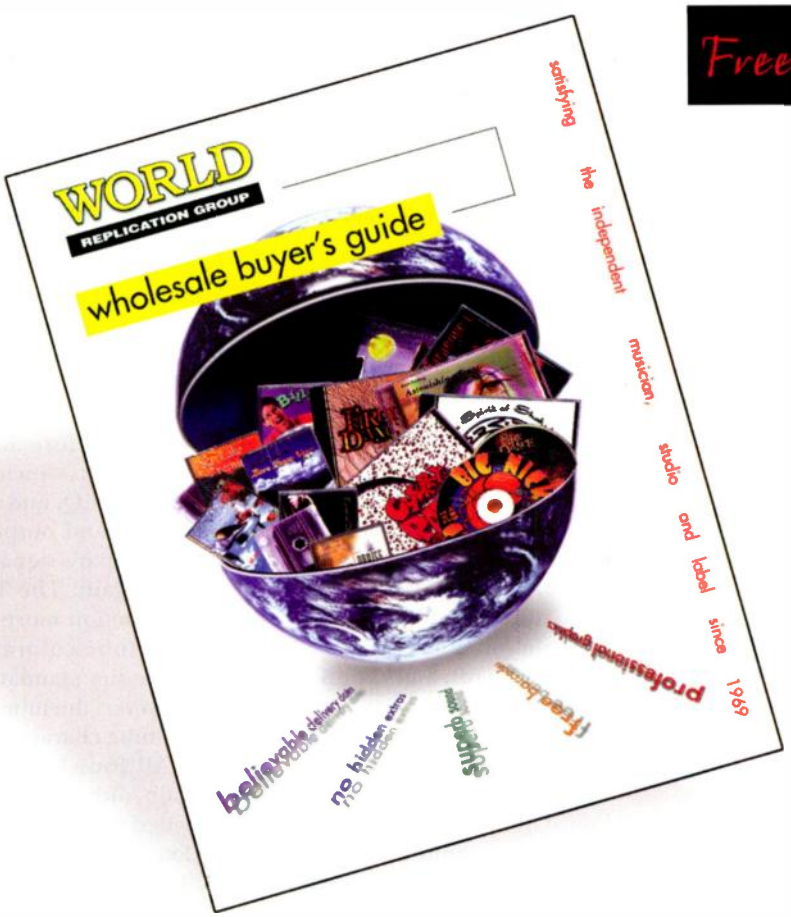

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Tube PAC Specifications

Input Connectors	(1) XLR, (1) ¼" TS
Output Connectors	(1) XLR, (1) ¼" TS
Maximum Gain	70 dB (XLR to XLR), 67 dB (¼" to ¼")
Maximum Input Level	+15 dBu (XLR), +21 dBu (¼")
Maximum Output Level	+27 dBu (XLR), +22 dBu (¼")
Dynamic Range	>90 dB (unweighted)
Equivalent Input Noise	-129 dBu (XLR to XLR, A weighted)
THD	<0.1% (typical)
Phantom Power	+48V DC (switchable)
Tube Type	12AX7a (2)
Compression Ratio	2.3:1 (comp); 6:1 (lim)
Attack Time	6.5 ms (fixed)
Release Time	200 ms (Fast), 70 ms-1 sec (Auto)
Dimensions	1.65" (H) x 8.5" (W) x 5.25" (D)
Weight	6 lbs.

between the background and lead parts, I employed different microphones and different settings on the Tube PAC.

I used a Neumann KM 86 condenser mic for the two lead tracks. To avoid much coloration on the voice, I kept the Tube PAC's input gain fairly low, staying in the green for the most part, and only occasionally hitting the second yellow LED. A slight boost on the output achieved the desired level to tape, and we were off and running.

For the backup parts, I switched to a Shure SM57 dynamic mic and hit the tube stage harder. The red LED lit up quite a bit for these tracks, and the overall tone was a bit thicker and less focused. In the end, though, the cumulative weight of the six tube-saturated tracks somewhat overpowered the lead vocals. (If I were doing it again, I would probably reverse the process, thickening the leads with more tube saturation and keeping the backup parts cleaner and thinner.)

The Tube PAC also sounded very good as a DI for bass guitar. I'm not particularly fond of DI-bass sounds to begin with, so I was impressed when the Tube PAC made a low-cost Ibanez sound fuller and seemingly more expensive. It won't replace an SWR bass cabinet, but the Tube PAC will impart some character and coloration to a bass signal before it goes to tape.

SQUEEZED IN

The Tube PAC's compressor directly follows the tube preamp stage and can be switched in or out of the signal path with a Bypass button. Though not full featured, the compressor is useful and intuitive. Four LEDs display gain reduction introduced by the compressor. The resolution of the meters is rough: 0 dB, -3 dB, -6 dB, and -12 dB.

A Threshold knob sets the point at which the compression kicks in on the input signal, and it is fully interactive with the Gain knob. That is, the threshold must be set according to how hard you hit the preamp's tube stage. Adjust the threshold until the yellow 0 dB LED lights achieve unity gain into the compressor, and from there you can adjust for the desired amount of compression.

A Slope button toggles between compression and limiting. When set for compression, the ratio is fixed at 2.3:1, a mild amount of compression useful for many types of signals (including vocals). In the limiting mode, the ratio is fixed at 6:1. Although that's not technically a limiting ratio (10:1 is commonly regarded as the onset of limiting), it is suitable for a variety of sounds. Given only two choices of ratios, A.R.T. chose well.

The Tube PAC compressor does not offer an adjustable attack rate, but there is a switch for toggling between Auto and Fast release times. Auto seemed to work for most signals, while the Fast setting proved better suited to quick, percussive sounds. Compression is a subjective area, but overall I was not as knocked out by the compressor as I was by the preamp. Still, it's a useful feature, especially when used in moderation.

TUBE EQ

If you're in need of a single-channel EQ with a bit of tube character, the Tube EQ is a good choice (see **Fig. 2**). Utilizing the same 12AX7a tube as the Tube PAC, the Tube EQ provides high and low shelving EQ and two sweepable mid-range bands. Not only is it a great companion to the Tube PAC, but a pair of Tube EQs would be a terrific choice for processing a stereo line output (e.g., from a keyboard) going to tape.

A Gain control provides 15 dB of boost and more than 25 dB of cut. Greatly appreciated is an Output control, which ensures that you get not only the tone you're looking for, but the proper level, too. A clip LED lights up just before any stage of the Tube EQ distorts—including the input, each band of EQ, and the output. A Bypass button and output control allow you to compare signals while maintaining unity gain. The Tube EQ is designed to function more as an equalizer than as a tube-coloration device (that is more the mandate of the Tube PAC); however, the tube stage does add a subtle tube character to the sound.

All four bands provide ±12 dB of gain, although gain markings are provided only on the shelving bands. The low shelving band is switchable between 40 Hz and 120 Hz, and the high shelving band is switchable between 6 kHz

A.R.T.
Tube PAC tube preamp and compressor
$249

FEATURES ■■■▮
EASE OF USE ■■■■
AUDIO QUALITY ■■■▮
VALUE ■■■■▮
1 2 3 4 5

PROS: Clean, quiet preamp section. Tube stage provides a range of tonal coloration. Compressor works well with many types of signals. Output control allows for optimal settings of overdrive.

CONS: Phantom power switch located next to Gain button. Coarse metering of gain reduction from compressor. Compressor has minimal control settings.

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Tube EQ Specifications

Input Connectors	(1) XLR, (1) ¼" TS
Output Connectors	(1) XLR, (1) ¼" TS
Max Input Level	+20 dBu
Max Output Level	+27 dBu (XLR), +22 dBu (¼")
Dynamic Range	>90 dB (unweighted)
THD	<0.1%
Tube Type	12AX7a
Gain/Band	±12 dB
Low (shelving)	40 Hz/120 Hz (selectable)
Low-Mid (sweepable)	20 Hz–200 Hz/200 Hz–2 kHz (selectable)
High-Mid (sweepable)	200 Hz–2 kHz/2 kHz–20 Hz (selectable)
High (shelving)	6 kHz/18 kHz (selectable)
Dimensions	1.65" (H) x 8.5" (W) x 5.25" (D)
Weight	5 lbs.

and 18 kHz. (Interestingly, the photograph on the cover of the Tube EQ manual shows different frequencies for the Low and High shelving points.)

The Lo-Mid and Hi-Mid controls are dual-concentric pots, with the outer control setting the frequency and the inner control setting the gain. The bandwidth (Q) is fixed for both midrange bands, but the sweepable mids are partially overlapping. The Low-Mid can be switched to cover either 20 Hz to 200 Hz, or 200 Hz to 2 kHz. The High-Mid repeats the 200 Hz to 2 kHz range but can also be switched to cover 2 kHz to 20 kHz.

The Tube EQ is easy to operate—if you know how to use a parametric equalizer, you're in business. If you don't, A.R.T. provides some helpful advice in the manual, along with a nice little chart illustrating generic frequency ranges of a few instruments.

I liked the Tube EQ. It can be a little noisy when overused, but, fortunately, a little goes a long way with the Tube EQ. If you have a boxy-sounding microphone or preamp, this unit can breathe some life into the signal path. I also liked the character and sparkle it added to tired-sounding presets on mono keyboard tracks. (Unfortunately, I had only one unit to test and therefore wasn't able to try it on stereo material.)

COME TOGETHER

Both the Tube PAC and Tube EQ sound great and perform their designated jobs well. Both units have XLR and ¼-inch inputs and outputs, making them flexible for use in a variety of recording and performance situations. Also, they are ruggedly designed and very affordable.

I would compare the Tube PAC and Tube EQ to units costing four times as much money, because they deliver audio quality at that level. In addition, the two units can be combined into a sharp-looking package that occupies only a single rack-space.

Producer-composer **Rob Shrock** *recently worked with Dionne Warwick on her upcoming album. He was previously in the studio with Burt Bacharach and Elvis Costello. He is also Bacharach's musical director and keyboardist.*

A.R.T.
Tube EQ tube equalizer
$199

FEATURES ■■■▮
EASE OF USE ■■■■
AUDIO QUALITY ■■■▮
VALUE ■■■■▮
1 2 3 4 5

PROS: Four bands of EQ with overlapping, sweepable mids. Tube stage adds subtle warmth. Output control provides proper level setting.
CONS: No bandwidth control for midrange bands. Can be noisy when overused.
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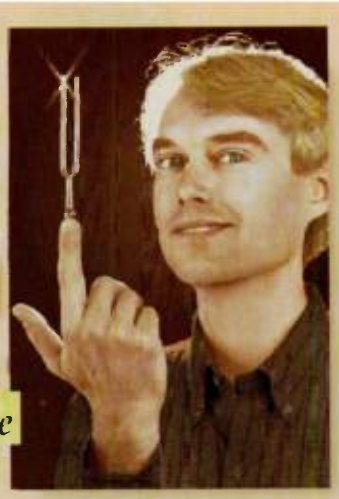

► The **TRUE STORY** *by David L. Burge*

It all started in ninth grade as a sort of teenage rivalry...

I'd slave at the piano for five hours daily. Linda practiced far less. But somehow she always had an edge that made *her* the star performer of our school. It was frustrating. *What does she have that I don't?* I'd wonder.

Linda's best friend, Sheryl, sensed my competition. One day she bragged on and on about Linda, adding more fuel to my fire.

"You could *never* be as good as *Linda*," she taunted me. "*Linda's* got *Perfect Pitch*."

"What's Perfect Pitch?" I asked.

Sheryl gloated over a few of Linda's uncanny abilities: how she could name any tone or chord—*just by ear*; how she could sing any pitch she wanted—*from mere memory*; how she could play songs—after only *listening* to them on the radio!

My heart sank. *Her fantastic EAR is the key to her success* I thought. How could I ever hope to compete with her?

But later I doubted Sheryl's story. How could anyone possibly know F♯ or B♭ just by *listening*? An ear like that would give one a *mastery of the entire musical language!*

It bothered me. Did she *really* have Perfect Pitch? I got up the nerve, approached Linda, and asked her point-blank if it was true.

"Yes," she nodded to me aloofly.

But Perfect Pitch was too good to believe. I rudely pressed, "*Can I test you sometime?*"

"OK," she replied cheerfully.

Now I'd make her eat her words...

My plan was ingeniously simple: I picked a moment when Linda least suspected. Then I boldly challenged her to name tones *by ear*.

I made sure she had not been playing any music. I made her stand so she could not see the piano keyboard. I made certain that other classmates could not help her. I set everything up perfectly so I could expose her Perfect Pitch claims as a ridiculous joke.

Nervously, I plotted my testing strategy. Linda appeared serene. Then, with silent apprehension, I selected a tone to play. (She'll *never* guess F♯!)

I had barely touched the key. "F♯," she said. I was astonished. I played another tone. She didn't even stop to think. *Instantly* she announced the correct pitch. Frantically I played more tones, skipping here and there all over the keyboard. But somehow she knew the pitch each time. She was SO *amazing*. She knew tones like colors!

"Sing an E♭," I demanded, determined to mess her up. She sang a tone. I checked her on the keyboard. She was right on! Now I was starting to boil. I called out more tones for her to sing, trying hard to make them increasingly difficult. Still she sang each note perfectly on pitch.

I was totally boggled. "*How in the world do you do it?*" I blurted.

"I don't know," she sighed. And to my dismay, that was all I could get out of her!

The dazzle of Perfect Pitch hit me like a ton of bricks. My head was dizzy with disbelief. Yet from that moment on, I knew *Perfect Pitch is real.*

I couldn't figure it out...

"*How does she DO it?*" I kept asking myself. On the other hand, why can't *everyone* recognize tones by ear? It dawned on me that most musicians can't tell a simple C from a C♯, or the key of *A major* from *F major*! I thought about that. A *musician* who cannot tell tones by ear?! That's like a painter who can't recognize the rainbow of colors on his palette! It seemed odd and contradictory.

I found myself more mystified than ever. Humiliated and puzzled, I went home to work on this problem. At age 14, this was a hard nut to crack.

You can be sure I tried it myself. I would sweet-talk my three brothers and two sisters into playing tones for me, which I would try to identify by ear. It became just a guessing game; my attempts were dismal failures.

I tried playing the tones *over* and *over* in order to memorize them. I tried to feel the "highness" or "lowness" of each pitch. I tried day after day to learn and absorb those elusive tones. But nothing worked. After weeks of struggle, I still couldn't do it. Sure, Linda had an extraordinary gift—the ultimate ear for music, the master key to many talents. I wished I had an ear like that. But it was out of my reach. So I finally gave up.

Then it happened...

It was like a miracle. A twist of fate. Like finding the lost Holy Grail. Once I had stopped *straining* my ear, I started to listen NATURALLY. Then the incredible secret to Perfect Pitch jumped right into my lap.

I began to notice faint "colors" within the tones. Not *visual* colors, but colors of *pitch*, colors of *sound*. They had always been there. But this was the first time I had ever "let go"—and *listened*—to discover these subtle differences within the musical tones.

Soon—to my own disbelief—*I too could recognize the tones by ear!* It was simple. I could hear how F♯ sounds one way, while B♭ has a *different sound*—sort of like "hearing" red and blue. The realization struck me: THIS IS PERFECT PITCH! This is how Bach, Beethoven, and Mozart could envision their masterpieces—and know tones, chords and keys all by ear—*by tuning in to these subtle "pitch colors" within the tones.*

It was almost childish—I felt sure that *anyone* could unlock their own Perfect Pitch with this simple secret of "color hearing."

Bursting with excitement, I went and told my best friend, Ann (a flutist), that she too could have Perfect Pitch. She *laughed* at me.

"You have to be *born* with Perfect Pitch," she asserted. "You can't develop it."

"You don't understand what Perfect Pitch *is* or how it works," I countered. "I couldn't recognize a single note before. Now it's *easy*."

I showed her how to listen. Timidly, she confessed that she too could hear the pitch colors. With this jump start, it wasn't long before Ann had *also* acquired Perfect Pitch.

At school we became instant celebrities. Classmates would test our ears, endlessly fascinated with our "supernatural" powers. Yet to us, our hearing was nothing "super"—just natural.

Way back then, I never dreamed I would later cause a stir among college music professors. But when I got a little older, I eventually started to explain my discovery to the academic world.

They *laughed* at me. Many told me: "You must be *born* with Perfect Pitch; you can't develop it." I would listen politely. Then I'd reveal the simple secret—*so they could hear it for themselves.* You'd be surprised how fast they would change their tune!

As I continued with my own college studies, my "perfect ear" allowed me to progress far faster than I ever thought possible. I even *skipped over* two required courses. Perfect Pitch made *everything* easier—performing, composing, arranging, sight-reading, transposing, improvising—and it skyrocketed my *enjoyment* of music as well. I learned that music is definitely a HEARING art.

And as for Linda?

Oh, yes—I'll backtrack. Time found me at the end of my senior year of high school. I was almost 18. In these three and a half years with Perfect Pitch, my piano teacher insisted I had made ten years of progress. But still I wasn't satisfied. I needed one thing *to beat Linda*. Now was my *final chance*.

Our local university sponsored a music festival each spring, complete with judges and awards. To my horror, they scheduled me as the last person to play—the *grand finale* of the entire event.

Linda gave her usual sterling performance. I knew she would be tough to match, let alone surpass. My turn came, and I went for it. Slinking to the stage, I sat down and played my heart out.

Guess what? I scored an A+ in the most advanced performance category. Linda only got an A. *Sweet victory was music to my ears—mine at last!*

Now it's YOUR turn!

For 17 years now, thousands around the world have proved that my Perfect Pitch method works, including research at two leading universities. Now I'd like to show YOU how to experience *your own* Perfect Pitch! You only need a few basic instructions. I've put everything I know into my **Perfect Pitch® SuperCourse**.

It's fun—and it's *guaranteed* to work for YOU—regardless of your instrument, your playing style, or current ability level.

Try it for yourself: Order your own **Perfect Pitch® SuperCourse**—AT $25 OFF!—and play the first two cassettes. I promise you will *immediately* hear the Perfect Pitch colors that I'll start you on, *or return the Course for a full refund.* You've got my word on it. Or you can check out your progress with no risk. You'll find a *dramatic improvement in both your pitch and your playing* in only 40 days, or return the Course and *I will personally ensure you get your refund—no questions asked.*

Think of the possibilities that Perfect Pitch can open for YOU and your music. Imagine how it can improve *your* playing, *your* singing—*your* creativity and confidence.

And picture your friends' faces as YOU name tones and chords with laser-like precision! Please—don't *you* laugh, too! At least not until you see how simple it is to discover *YOUR VERY OWN* Perfect Pitch!

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❑ I'd like more info. Send me Perfect Pitch Lesson #1 *(written)* with research at two universities—**FREE** with *no obligation.*

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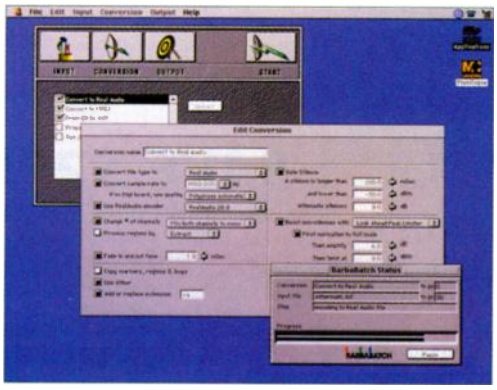
QUICK PICKS

AUDIO EASE

BarbaBatch 2.4 (Mac)

By Rudy Trubitt

Need to exchange raw audio files between different DAW formats? Or crunch a passel of files for a CD-ROM project? How about putting all the songs from your last three CDs on your Web site, without stuffing your hard disk in the process? Any audio-processing task that requires repetitive steps is a candidate for *BarbaBatch* ($395) from Netherlands-based developer Audio Ease.



***BarbaBatch* can convert the same files into multiple formats in a single batch job. Clicking the big buttons from left to right guides you through the conversion process.**

Recipe for Success

BarbaBatch can change sample rates; convert file formats; extract regions; transform sounds to mono, stereo, or split stereo; change bit resolutions; encode audio for Web streaming; extract and convert audio from CDs; and manipulate the dynamic range for virtually any number of sound files. *BarbaBatch* takes the drudgery out of file-processing chores. It enables you to define custom file-transformation "recipes" and use them again and again. Even better, you can apply more than one recipe at a time. That can be a real time-saver, because some projects require multiple formats for the same audio (i.e., both 22 kHz, 16-bit *and* 11 kHz, 8-bit files), which *BarbaBatch* can generate in a single pass.

BarbaBatch 2.4 reads and writes over two dozen Mac, PC, and Unix sound-file formats, including AIFF, AIFF-C, System 7 sound, MPEG-1, Microsoft and IMA ADPCM, WAV, µLaw, and RealAudio. It also writes to QuickTime 3 format along with PureVoice and QDesign Music codecs. That versatility comes in handy because you never know when you'll be asked to read or write an obscure file format. For example, I recently needed to create VOC files for a sound-generating chip, and to my delight, it was on *BarbaBatch*'s list.

Above and Beyond

The program goes beyond the call of duty in many cases. For instance, *BarbaBatch* reads and writes markers, loops, and regions for formats (such as Sound Designer II) that support those attributes. It even copies regions and markers into the files it creates, which is a big help to me. Now I can "annotate" high-resolution source recordings with markers, and these notes are preserved in the sample-rate-converted copies.

Speaking of sample-rate conversion, *BarbaBatch* will convert audio to any rate from 1 to 100 kHz and squish bit depth from as much as 32-bit linear PCM to 4-bit ADPCM. And all the conversion options are set via check boxes and sliders within a single large dialog box.

Sound Good?

I generally use Tom Erbe's excellent shareware program *SoundHack* for sample-rate conversion. I'm happy to say that, in my tests, *BarbaBatch* produced results equal to *SoundHack*'s best setting. As an added bonus, *BarbaBatch* processed the same files several times faster. The program's optional dither worked well and was unobtrusive in loud, 8-bit sound files.

My only complaint was with *BarbaBatch*'s built-in limiter, which is similar to the Lookahead limiter on Waves' popular *L1 Ultramaximizer*. The *BarbaBatch* limiter provides control over the amount of limiting, which adds overall gain as limiting is increased and sets the maximum peak level in the file. This kind of peak limiting is really essential when making 8-bit files. Unfortunately, *BarbaBatch*'s limiter lacks *L1*'s transparency—you can hear it pumping on music tracks when moderate to aggressive limiting is applied.

That quibble aside, *BarbaBatch* has quickly worked its way into my audio production process. Those who wrangle a lot of sound files will certainly want to add it to their bag of tricks. The informative documentation includes several tutorials to get you started. Best of all, the program is quick and easy to use, it produces great-sounding output, and it's the most efficient batch-processor I've used.

Overall EM Rating (1 through 5): 4.5

CIRCLE #446 ON READER SERVICE CARD

BIG FISH AUDIO

Didgeridoo and Other Primitive Instruments

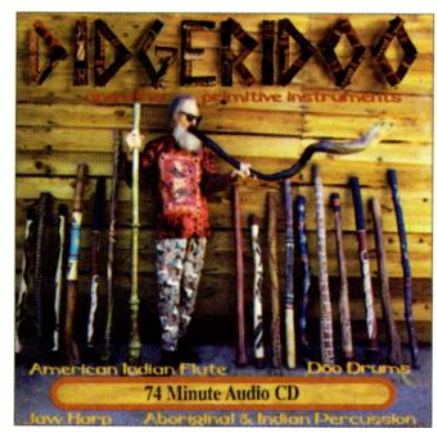
By Julian Colbeck

I've never understood the fascination with the didgeridoo, but I do understand the appeal of obscure musical entities. Produced by Peter Spoecker, *Didgeridoo and Other Primitive Instruments* ($299.95) is an adventure into the soul of seldom-recorded tribal musical instruments.

Primitive Orchestra

This Akai S1000/3000 CD-ROM is filled with 512 MB of single-note samples and mini-performances of didgeridoo drones and rhythms, jaw harp, bull roarer, doo drums, Indian drums, rattles, and flute. The samples, which are universally dry, average about 5 to 10 MB, and all except the flutes are in stereo.

The mapping presented a bit of a challenge due to an inconsistency: some instruments are sampled through an octave and are presented as such, while others are sampled at the same pitch but with different performances triggered by each key through an octave range. The sample



Is that buzz a swarm of termites? No, it's Big Fish Audio's *Didgeridoo and Other Primitive Instruments*.

For manufacturer information, please see Contact Sheet, p. 163.

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naming is clear, usually indicating pitch and—where relevant—tempo. I appreciated the fact that multiple programs in a group default to sequential MIDI channels.

The Didgeridoos

Didgeridoo is the main dish, accounting for more than 200 MB of the disc's contents. "Didge Wild" is a collection of random didgeridoo samples spanning an octave, with each note a completely different performance. Assessing sound quality is tricky with so rough-hewn an instrument. Occasionally, the lowest sample within a program seemed to contain what sounds like light rain—is that buzz a swarm of termites, bits of eucalyptus bark flying out of the end, or just excessive amounts of wind? The rest of the samples were clean and consistent, but they could all use a splash of reverb.

None of these samples could ever be deemed unobtrusive. You'd have to really want to feature didgeridoo if you put "Didge Wild" in the mix. The "Drone Simple" samples are a little more innocuous, though they are also somewhat thin. Many full-blown, rhythmic didgeridoo performances are featured later on the disc, played at a variety of pitches and tempos.

Percussion and Others

Under the moniker "Indian Drums" comes a large number of Native American handdrum samples. The "Indian Rattles" are dry, but they could add a nice texture to a percussion section. A category entitled "Wood Percussion" (which I presume is wood blocks) contains more than 100 samples of what reminded me of Monty Python's "False Teeth Symphony."

The included flute tones are enormously variable, which is fine. However, the performances contain very idiosyncratic riffs and trills, which is only fine if you're happy with chunks of predigested ethnic playing.

My favorite sound is definitely the bull roarer: a low, eerie swooshing sound. "Doo Drums"—short slices of didgeridoo that are hit with the open palm—would provide an interesting alternative to log drums, marimbas, or other tuned and semituned percussion instruments. The evocative jaw harp is offered in both static and "groove" varieties.

The Wrap

This sample CD has considerable appeal. Unfortunately, hard facts about individual maps, tunings, and applications are scant. Of course, there is a premium to be paid for special-interest items such as these. The price seems awfully high for a collection of tribal bonks and squawks. But if that's what you're looking for, your choices are few, and you may find this collection worth the price.

Overall EM Rating (1 through 5): 2.5

CIRCLE #447 ON READER SERVICE CARD

CLOCKWORK MUSIC

CAL Tutor 3.0 (Win)

By Scott R. Garrigus

CAL (Cakewalk Application Language) is a programming extension found in the professional line of MIDI sequencing products from Cakewalk Music Software. CAL allows users to create their own custom editing commands that can insert, delete, and otherwise modify MIDI events. Unfortunately, because of its complexity, CAL can be difficult to use. *CAL Tutor 3.0* ($30, or $15 upgrade from version 2.1 found on Musicians Toolbox II) from Clockwork Music may give you the knowledge you need to start writing your own CAL programs.

Hard Lessons

Delivered in a 3-floppy-disk set, *CAL Tutor* is accompanied by a single installation instruction sheet and runs on any Windows-based PC. Even though the program is somewhat interactive, the bulk of the presented material is onscreen text that is categorized in the familiar chapter-based book format. When first run, *CAL Tutor* presents you with a table of contents listing all seventeen chapters (see **Fig. 1**). These cover everything from programming basics to writing full-blown CAL programs. Unfortunately, like the concepts that it is trying to teach, the material in *CAL Tutor* can be a bit intimidating. Though the program does a decent job of

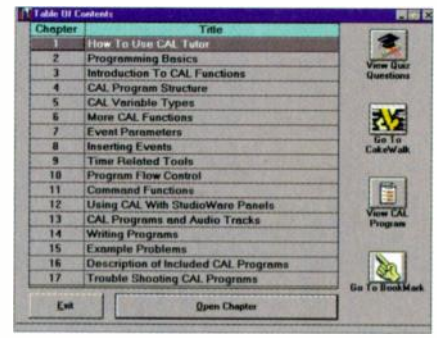

FIG. 1: *CAL Tutor* offers seventeen chapters of information that can help enhance your programming skills.

describing how to work with CAL, its wording is not quite as clear as you might find in a professional publication. The description about the order of parameters in functions, for example, is particularly difficult to grasp.

CAL Tutor has an interactive nature in the form of context-sensitive quiz questions. While reading through the text material, you will occasionally happen upon a button with a question mark logo. Clicking on the button opens up a quiz question that tests your understanding of what you're currently reading. You simply type in your answer, and the program will inform you of whether you are right or wrong. The questionnaire can be a bit picky though; for example, it does not recognize the word "one" for the number "1" and therefore will consider that answer incorrect. Another shortcoming is that, when you do happen to be wrong, the program presents you with the correct answer but fails to give an explanation. There are a total of 45 questions, and you can see how well you scored by using the View Quiz Questions menu option.

Test Your Skill

CAL Tutor gives you a nice supply of prewritten CAL programs that you can use as starting points for your own work. A number of these programs are referenced throughout the text material to give you examples of proper programming form and are useful demonstrations of some of the techniques that are discussed. To run the programs, you will of course need to use your Cakewalk MIDI sequencer, which can be launched with the click of a button from within *CAL Tutor*. In order to test the programs though, you will have to provide your own MIDI data. *CAL Tutor* doesn't contain any specific MIDI files to use for testing the included CAL programs.

Though the product could stand a bit more polish, advanced users who have some previous programming experience with computer languages such as Basic, C, or Pascal, should easily be able to wade through *CAL Tutor*'s material and piece together their own CAL programs in a reasonable amount of time. Intermediate users will also find enough information here, but it may take them a little longer to get through. Unfortunately, beginners who are hoping *CAL Tutor* will help them utilize the Cakewalk Application Language are left out in the cold.

Overall EM Rating (1 through 5): 3

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QUICK PICKS

MASTERBITS

Project X

By Jeff Obee

After listening to the first couple of tracks on *Project X* ($39; audio CD), I had to light candles in my studio to provide the proper ambience for listening to the rest. The disc is rich with sensuous pads, panning textures, squirrely arpeggiations, liquid filter sweeps, and percolating electronic rhythms that are thoroughly enjoyable.

Ambience in Spades

Project X is composed of 49 short synth compositions, ranging in length from 39 seconds to almost three minutes. They are described as "paranormal" soundtracks, and although that seems like a somewhat vague and clichéd label, it does suit this collection. Most of the pieces are titled with "space" names, like "Mothership Connection" and "Stargate," and there is a definite spacey, trance/ambient approach to the music. You won't hear traditional melodies, and the music is heavily laden with reverb and other effects.

This is not a sampling CD, per se, although you could certainly sample snippets of it for use in your own compositions,

"Paranormal" compositions make up the sounds on Masterbits' *Project X*. The sample's out there.

as long as you credit the disc as your source. The material is rich with potential uses, from video post-production, games, and environmental soundscapes to looping portions of the disc for extended ambient collages or background textures for various electronic styles of music. Masterbits even suggested using it for short meditations—mini–relaxation tapes, if you will.

How'd They Do It?

Along with a Lexicon PCM-90, the main effects processor used in the creation of these tracks was the Roland RSS-10, which takes a mono source and moves it around in a 3-dimensional space. This gives a wonderful depth to the music—by all means, listen with headphones! The synths used were an Oberheim Xpander, a Kurzweil PX-100, a Roland JD 990 and S-760, a Yamaha TG77, and a Studio Electronics SE-1. All the recording was done using Digidesign's Pro Tools.

Shining Stars

Delineating some of my favorites is a tad difficult, because frankly, I really liked the whole disc. "Alien Factory" captured my attention with its industrial yet harmonically pleasing backdrop and its "pointy," metallic sounds that rolled across the sound plane. The NASA samples from

Masterbits' *Add Lips Vocals III* CD, which I recently reviewed (see the September 1998 **EM**), were used sumptuously in a piece called "Moon Talk."

Track 18, "Mindscanner," is one of the longer pieces on the disc and is a fine ambient sound experience. Long, unfolding textures are the backdrop, with some bloopy droplets panning across and sawtooth, sample-and-hold–type sounds and other analog goodies moving around the 3-dimensional space. The five "Waterworld" pieces also present some great textures—watery patches that lay nicely in the 3-D environment, with some slightly Doppler-esque effects to give the pads an underwater sound.

My only slight misgiving about the collection as a whole is that the tracks have a certain "sameness" to them, but perhaps that sameness is the point of a disc like this. It was entirely written and recorded in four days by producer Hans-Jörg Scheffler, a fact that undoubtedly contributes to the similarity among the pieces but also adds an immediacy that I liked. Scheffler's use of texture is excellent and his choice of sounds superb. His soundscapes are truly worthy of the "paranormal" description.

They're Out There

For a great price, *Project X* gives you a lot of delicious synthesizer music with lots of potential uses. You can't go wrong! The audio quality is fabulous and the music intriguing—I heartily recommend it.

Overall EM Rating (1 through 5): 5

CIRCLE #449 ON READER SERVICE CARD

RADIAL ENGINEERING

JDI and JDV

By Jim Miller

Direct-injection (DI) boxes are among the most underrated and underutilized tools available to today's studio owners. I built direct boxes myself back in the 1970s, so I feel uniquely qualified to evaluate a couple of new DIs from Canadian-based Radial Engineering, a division of CableTek Electronics, Ltd.

Inject Me

The two boxes I received for review are the Radial JDI ($239.99)—a passive direct box equipped with a premium Jensen transformer—and the Class-A, active Radial JDV ($299.99). All external switches on the units are recessed to protect them and prevent accidental switching. These things are built like tanks, but the units do have some rather sharp corners that can give you a nasty nick if you're not careful.

The JDI and JDV direct-injection boxes from Radial Engineering are quiet, transparent, versatile, and built to last.

The dark, gray-green JDI is a passive DI box designed to take any audio source and send it directly to the mixer or recorder. In the studio, this means you can take the audio signal from a bass guitar, for instance, and send it to the console

without having to worry about the extra noise generated by an amplifier. This also gives you the pure, uncolored sound of the instrument, allowing you to do your tone shaping at the console. In live applications, the bass signal can go out to the house mixer, while the original signal can be passed directly through to an onstage amplifier via the JDI's Thru ¼-inch output connector.

A -15 dB pad can be inserted for taming exceptionally hot signals, and there's a ground-lift switch to resolve those annoying ground loop hums and buzzes. By using the Merge switch, the unit's input and throughput can be used to passively mix two channels or devices (i.e. two synths), thus saving valuable console inputs. Finally, a Filter switch, which cuts out ultrasonic noise caused by electronic devices (i.e., computers), can be used in the Pickup position when a guitar is plugged in or in Line position when taking the hotter signals from synths, drum machines, CD players, and so forth. The main output is a balanced XLR.

The dark blue JDV is designed to be powered via internal chargeable batteries, an AC wall adapter, or +48 volt phantom power. Like the JDI, there is a Pad switch, a Ground-lift switch, a Merge switch, and a Thru jack (but no Filter switch). The JDV has a balanced, XLR mic-level output and balanced, TRS line-level out. Being powered, the JDV is perfect for taking low-output gear, such as guitars with passive pickups, direct to the board, though it can also handle hot signals from, say, a synthesizer.

They're Great

The best things you can say about products like these are that they are quiet, transparent, versatile, and built to last. Both Radial DI boxes fit these descriptions. The unit's construction is rock-solid, and each can perform a wide variety of useful functions with a minimum of hassle. Most importantly, signals pass through both with outstanding linearity and phase accuracy that reflects the manufacturer's obviously high standards.

There are many DIs available today, but after using the Radial boxes regularly for several weeks now, I can say without hesitation that you won't find anything out there offering better performance or more durability for the money than the JDI and JDV.

Overall EM Rating (1 through 5):
JDI 4.5, JDV 5

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DIGITAL MIXERS

MACKIE
Digital 8 Bus Mixing Console

New

Well, it's finally here and just like the analog 8 bus a few years back, it's everything you've anticipated! Great sound quality, full recording and mixdown capabilities, motorized faders and an array of digital features geared to take you flying into the next century.

FEATURES-
- 48 channels of automated compression, gating, EQ and delay
- Built in 3-way meter display keeps you on top of your mix.
- Built-in meter bridge.
- Ultramix II automation for complete control, hook up an S-VGA monitor and you'll feel like you spent a lot more money
- All functions can be automated, not just levels and mutes. Store EQ, reverb, compression, gating and even Aux send information
- Fast SCENE automation allows you to change parameter snapshots on every beat
- Reads Standard MIDI tempo maps, displaying clock info on the built-in position counter
- Truly the cutting edge of mixing technology.

Panasonic
WR-DA7 Digital Mixing Console

New

Stop dreaming about your digital future, it's here! The Panasonic WR-DA7 digital mixer features 32-bit internal processing combined with 24-bit A/D and D/A converters as well as moving faders, instant recall, surround sound capabilities, and much more. Best of all, it's from Panasonic.

FEATURES-
- 32 inputs/6 AUX send/returns
- 24-bit converters
- Large backlit LCD screen displays EQ, bus and aux assignments, and dynamic/delay settings
- 4-band parametric EQ
- Choice of Gate/Compressor/Limiter or Expander on each channel
- 5.1 channel surround sound in three modes on the bus outputs
- Output MMC
- Optional MIDI joystick

TASCAM
TMD1000 Digital Mixing Console

New

You want to see what all the digital mixing buzz is about? The NEW TMD100 from Tascam will have you smilin' & automatin' in no time. It features fully automated EQ, levels, muting, panning and more in an attractive digital board with an analog 'feel'. Your digital future never looked, or sounded, so clear.

FEATURES-
- 4 XLR mic inputs, 8 1/4" balanced TRS inputs.
- 20-bit A/D D/A conversion, 64x oversampling on input, 128x on output.
- Store all settings, fully MIDI compatible.
- Optional **IF-TD1000** adds another 8 channels of TDIF and a 2-channel sample rate converter.
- Optional **FX-1000** Fx board adds another 4 dynamic processors and another pair of stereo effects.

MIC PREAMPS

Focusrite
Green 3 *"Voicebox MKII"*

New

The Voicebox MKII provides a signal path of exceptional clarity and smoothness for mic recording, combining an ultra-high quality mic amp, an all new Focusrite EQ section optimized for voice, and full Focusrite dynamics. The new MKII now includes a line input for recording and mixdown applications.

FEATURES-
- Same mic pre section as found on the Green Dual Mic Pre includes +48V phantom power, phase reverse, and a 75Hz high-pass filter. Mute control and a true-VU response LED bargraph are also provided
- EQ section includes a mid parametric band with frequency and gain control as well as a gentle bell shape to bring out the character of the voice.
- Dynamics section offers important voice processing functions of compression and de-essing combined with a noise reducing expander
- Single balanced Class A VCA delivers low distortion and a S/N ratio as low as -96dBu

EFFECTS PROCESSING

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ULTIMATE SOUND MACHINES
Finalizer Plus

New

Improving on the multi-award winning Finalizer platform, The Finalizer Plus delivers an unprecedented level of clarity, warmth and punch to your mix. Inserted between the stereo output of your mixer or workstation and your master recording media, the Finalizer Plus dramatically rounds out your material, creating that "radio ready" sound.

FEATURES–
- Balanced Analog as well as Digital outputs including AES/EBU, S/PDIF, & TOS.
- 24-bit precision A/d & D/A Converters
- 5-band 24-bit stereo EQ
- Enhance - De-essing, stereo adjust or digital radiance
- Real-time gain maximizer
- Variable slope multi-band expander
- Multi-band compressor
- Word Clock Sync
- MIDI section useful for controlling sequencer fades or any of the Finalizer's parameters from a remote MIDI controller.

Lexicon
PCM81 Multi-Effects Processor

New

The PCM-81 has everything that made the PCM80 the top choice among studio effects processors, and more. More effects, more algorithms, longer delay and full AES/EBU I/O

FEATURES–
- 300 Presets include pitch, reverb, ambience, sophisticated modulators, 20 second stereo delays, and dynamic spatialization effects for 2-channel or surround sound applications
- 2 digital processors including Lexicon's Lexchip for the reverb and a second DSP engine for the other effects
- 24-bit internal processing
- Dynamic patching matrix for maximum effects control
- PCM card slot

EQUALIZERS

Focusrite
Green 2 *"Focus EQ"*

New

The Green 2 Focus EQ is suitable for a variety of applications combining a Focusrite equalizer section with a multi-source input section. Use it as a high-quality front end for recording applications or patch it into the send/return loop to upgrade a single channel of console eq, either way, it sounds great.

FEATURES-
- XLR & 1/4" inputs are similar to the Dual Mic Pre but have been adapted to cope with a wider range of levels.
- VU metering via a 10-LED bargraph
- EQ section derived from the Red and Blue range processors for superb audio quality.

COMPRESSORS

JOE MEEK
VC1 Studio Channel

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The Joe Meek Studio Channel offers three pieces of studio gear in one. It features an excellent transformer coupled mic preamp, a great compressor and an enhancer unit all in a 2U rackmount design. Find out why more and more studio owners can live without one.

FEATURES-
- 48V phantom power, Fully balanced operation
- Mic/Line input switch
- Mono photo-optical compressor
- High pass filter for large diaphragm mics
- Extra XLR input on front makes for easy patching
- Compression In/Out and VU/compression meter switches
- Twin balanced XLR outputs with one DI XLR output for stage use
- Enhancer In/Out switch and enhance indicator
- Internal power supply 115/230V AC

dbx
Blue Series 160S Stereo Compressor

The dbx 160S combines the best features of all the great dbx compressors in a well-built unit where the craftsmanship is as stunning as the engineering is innovative. This is truly a desirable compressor.

FEATURES-
- 127dB dynamic range • Program dependent "Auto", or fully variable attack and release
- Hard knee/OverEasy switchable.

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MIDI HARDWARE

Mark of the Unicorn

MIDI Time Piece™ AV
8x8 Mac/PC MIDI Interface

The MTP AV takes the world renowned MTP II and adds synchronization that you really need like video genlock, ADAT sync, word clock sync, and even Digidesign superclock!

FEATURES–
- Same unit works on both Mac & PC platforms.
- 8x8 MIDI merge matrix. 128 MIDI channels.
- Fully programmable from the front panel.
- 128 scene, battery-backed memory.
- Fast 1x mode for high-speed MIDI data transfer.

Digital Time Piece™
Digital Interface

Think of it as the digital synchronization hub for your recording studio. The Digital Timepiece provides stable, centralized sync for most analog, digital audio, and video equipment. Lock together ADATs, DA-88's, ProTools, word clock, S/PDIF, video, SMPTE, and MMC computers and devices flawlessly. It ships with "Clockworks" software which gives you access to its many advanced features and remote control of some equipment settings such as record arm.

OPCODE

Studio 64XTC
Mac/PC MIDI Interface

The Studio 64XTC takes the assorted, individual pieces of your studio-your computer, MIDI devices, digital and analog multitracks and even pro video decks, and puts them all in sync.

FEATURES–
- 4 In / 4 Out, 64 channel MIDI/SMPTE interface/patchbay with powerful multitrack & video sync features
- ADAT sync with MIDI machine control
- Simultaneous wordclock and Superclock output, 44.1kHz or 48kHz for perfect sync with ADAT, DA-88 and ProTools
- Video and Blackburst in (NTSC and PAL)
- Cross-platform Mac and Windows compatibility

SAMPLING

AKAI
S-Series
Rack Mount Samplers

Starting with 64X oversampling, Akai's S-Series Samplers use 28-bit internal processing to preserve every nuance of your sound and the outputs are 18- and 20-bit to ensure reproduction of your sounds entire dynamic range. These three new samplers add powerful capabilities, ease-of-use, expandability and affordability to set the standard for professional samplers.

KEYBOARDS & SOUND MODULES

Roland

XP60 & XP80 Music Workstations

The XP-80 delivers everything you've ever wanted in a music workstation. An unprecedented collection of carefully integrated features provide instant response, maximum realtime control and incredible user expandability. The XP-80 features a pro-quality 76-note weighted action keyboard while the NEW XP-60 features the same sound engine in a 61-note keyboard.

XP80 FEATURES–
- 64-voice polyphony and 16-part multitimbral capability
- 16 Mbytes of internal waveform memory; 80Mbytes when fully expanded (16-bit linear format)
- 16-track MRC-pro sequencer with direct from disk playback. Sequencer holds approx. 60,000 notes
- New sequencer functions like "non-stop" loop recording and refined Groove Quantize template
- Enhanced realtime performance capability with advanced Arpeggiator including MIDI sync and guitar strum mode and Realtime Phrase Sequence (RPS) for on-the-fly triggering of patterns
- 40 insert effects in addition to reverb and chorus
- 2 pairs of independent stereo outputs; click output jack with volume knob
- Large backlit LCD display

SR-JV80 Series Expansion Boards

New

Roland's SR-JV80-Series wave expansion boards provide JV and XP instrument owners a great-sounding, cost-effective way to customize their instruments. Each board holds approx. 8Mb of entirely new waveforms, ready to be played or programmed as you desire.

Boards Include–
Pop, Orchestral, Piano, Vintage Synths, World, Super Sound Set, Keys of the 60's & 70's, Session, Bass & Drums, Techno & Hip-Hop Collection.

KURZWEIL

K2500 Series Music Workstations

The K2500 series from Kurzweil utilizes the acclaimed V.A.S.T. technology for top-quality professional sound. Available in Rack mount, 76-key, and 88 weighted key keyboard configurations, these keyboards combine ROM based samples, on-board effects, V.A.S.T. synthesis technology and full sampling capabilities on some units.

FEATURES–
- True 48-voice polyphony
- Fluorescent 64 x 240 backlit display
- Up to 128MB sample memory
- Full MIDI controller capabilities
- 32-track sequencer
- Sampling option available
- Dual SCSI ports
- DMTi Digital Multitrack interface option for data format and sample rate conversion (Interfaces with ADATs or DA-88s)

KORG

Trinity Series Music Workstations DRS

Korg's Trinity Series represents a breakthrough in sound synthesis and an incredible user interface. It's touch-screen display is like nothing else in the industry, allowing you to select and program patches with the touch of a finger. The 24MB of internal ROM are sampled using ACCESS which fully digitizes sound production from source to filter to effects. Korg's DSP based Multi Oscillator Synthesis System (MOSS) is capable of reproducing 5 different synthesis methods like Analog synthesis, Physical Modeling, and variable Phase Modulation (VPM).

FEATURES–
- 16 track, 80,00 note MIDI sequencer
- Flexible, assignable controllers
- **DRS (Digital Recording System)** features a hard disk recorder and various digital interfaces for networking a digital recording system configured with ADAT, DAT recorder and hard disk.
- 256 programs, 256 combinations
- Reads KORG sample DATA library and AKAI sample library using optional 8MB Flash ROM board

TRINITY — *88 Weighted-key/Solo Synth*
TRINITY — *76-key/Solo Synth*
TRINITY — *61-key/Solo Synth*
TRINITY — *61-key*

(Digital IF, SCSI, Hard Disk Recorder, and sample Playback/Flash ROM functions are supplied by optional upgrade boards)

MONITORS

Hafler

TRM-8

Winner of Pro Audio Review's PAR Excellance Award in 1997, Hafler's TRM8s provide sonic clarity previously found only in much more expensive speakers. They feature built-in power, an active crossover, and Hafler's patented Trans-nova power amp circuitry.
- 45Hz - 21kHz, ±2dB
- 75W HF, 150W LF
- Electronically & Acoustically matched

MACKIE.

HR824

These new close-field monitors from Mackie have made a big stir. They sound great, they're affordable, they're internally bi-amped. "What's the catch?" Let us know if you find one.

FEATURES-
- 150W Bass amp, 100W Treble amp
- Full space, half space and quarter space placement compensation
- Frequency Response 39Hz to 22kHz, ±1.5dB

TANNOY

Reveal

New

The latest playback monitor from Tannoy, the Reveal has an extremely detailed, dynamic sound with a wide, flat frequency response.

FEATURES-
- 1" soft dome high frequency unit
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SOUND CARDS

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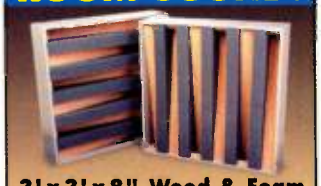

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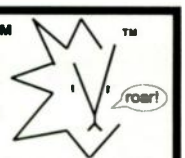

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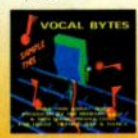

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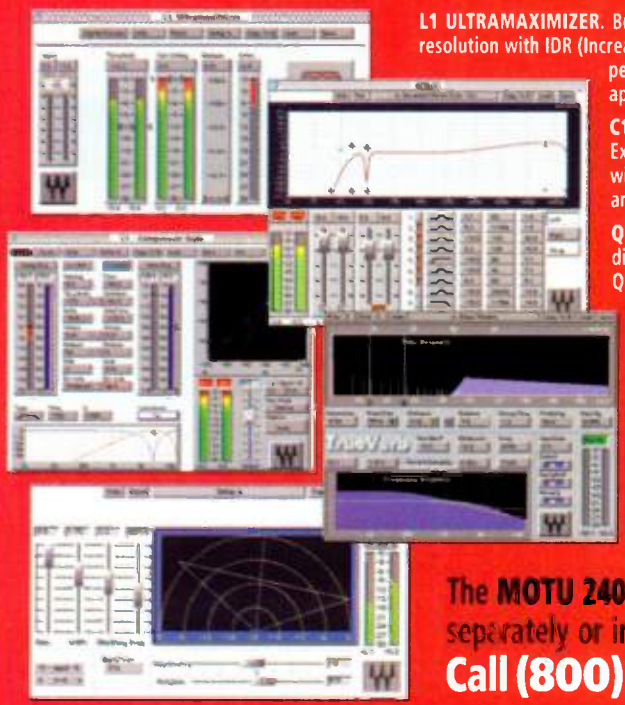

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FINAL MIX

By Larry the O

Both Sides Now

At some point, all of us who work with electronic equipment level criticisms at manufacturers, and often rightfully so. Manufacturers have been known to make announcements, institute policies, and release products that show, shall we say, a less-than-complete grasp of their customers' needs and desires. Sometimes it seems almost inconceivable that a company could do some of these phenomenally stupid or greedy things and yet survive and even prosper.

There is often, however, another side to the matter. I'm not about to be an apologist for manufacturers here, and anyone who has read my articles knows that I never hesitate to hit a company for something that I think misses the mark. But it is always worthwhile to attempt to gain more understanding by seeing the world through another's eyes.

The fundamental premise underlying almost everything a manufacturer does is that they are in business to make money. In and of itself, this is not a bad thing; it's the way our society and economy are structured. Besides, the Bible does not say that money is the root of all evil; it says the *love* of money is a root of all evil (1 Timothy 6:10). We all have to make a living. Furthermore, on the whole, there is very little money in our industry in comparison with pharmaceuticals, for example, or computers, or even video production for broadcast.

Add to that the degree to which electronic music and audio are technology-driven and the astonishing speed at which technology changes. Not only does this present the problem that a product may be virtually obsolete the day that it ships, but it also highlights the larger difficulty of predicting where technology is going. Companies must also commit resources to products before knowing whether the products will be accepted by consumers.

Where do we see this played out? One place is software upgrades. Software can require fewer physical resources to develop than hardware, and its costs can be completely amortized over time, whereas hardware costs never go away. But software development is time-intensive and difficult to accurately schedule. For a small company, there may be only one person working on software development, and that person spends absurd amounts of hours trying to keep up with the competition. That, along with the small quantities they can expect to sell into our little market, are frequently the reasons that software upgrades are late, buggy, and cost more than you want to spend. These are also the motivations for using hated copy-protection schemes.

Large companies, too, are subject to development pressures. Investors behind the scenes push management to release new products ahead of the competition. A company that raised money to develop an innovative and ambitious product must announce a product that is barely more than a plausible concept in order to drum up market interest. Then the company receives heavy pressure from its backers to push the product out the door. The result? Products are often released as much as a year or more after being announced, sometimes with a veritable nest of nasty bugs hiding inside.

It is possible to successfully ride the economic and development tiger. Management and engineering teams that achieve this can separate great companies from mediocre or awful ones. Of course, even top manufacturers don't often provide us with perfect dream tools at dirt-cheap prices, but they do earn a reputation for quality, value, and vision. That reputation is the most important asset a company can have.

Larry the O *is a musician, producer, engineer, and sound designer whose San Francisco–based company, Toys in the Attic, provides a variety of musical and audio services. He does not have a TV, VCR, cell phone, or Web page, but he holds the distinction of introducing the term* stud muffin *to audio writing.*

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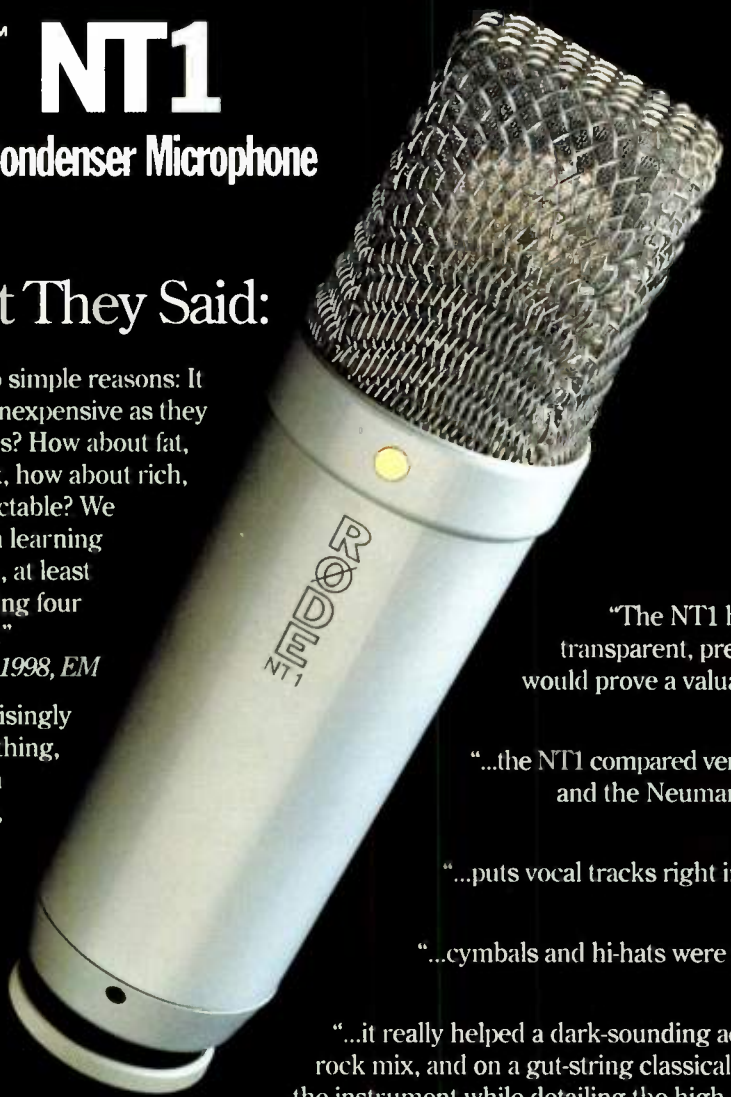

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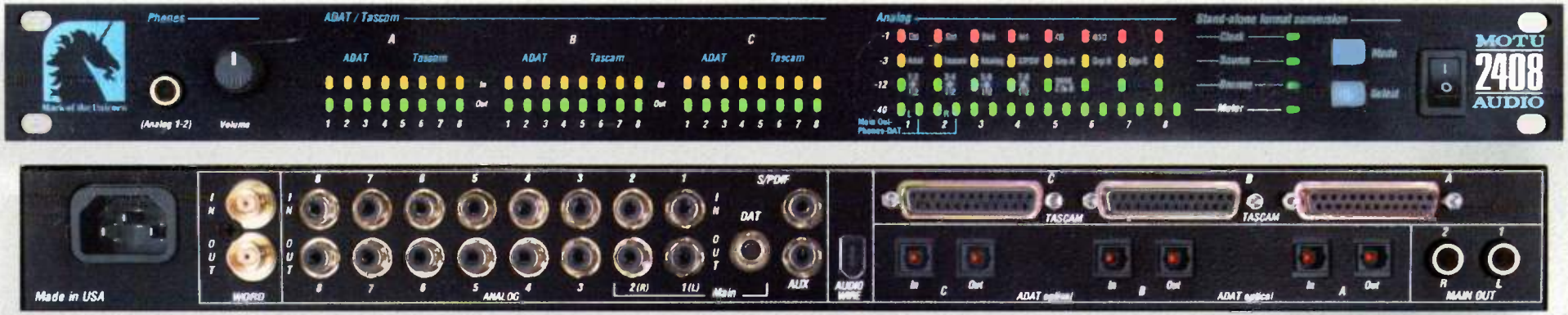

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