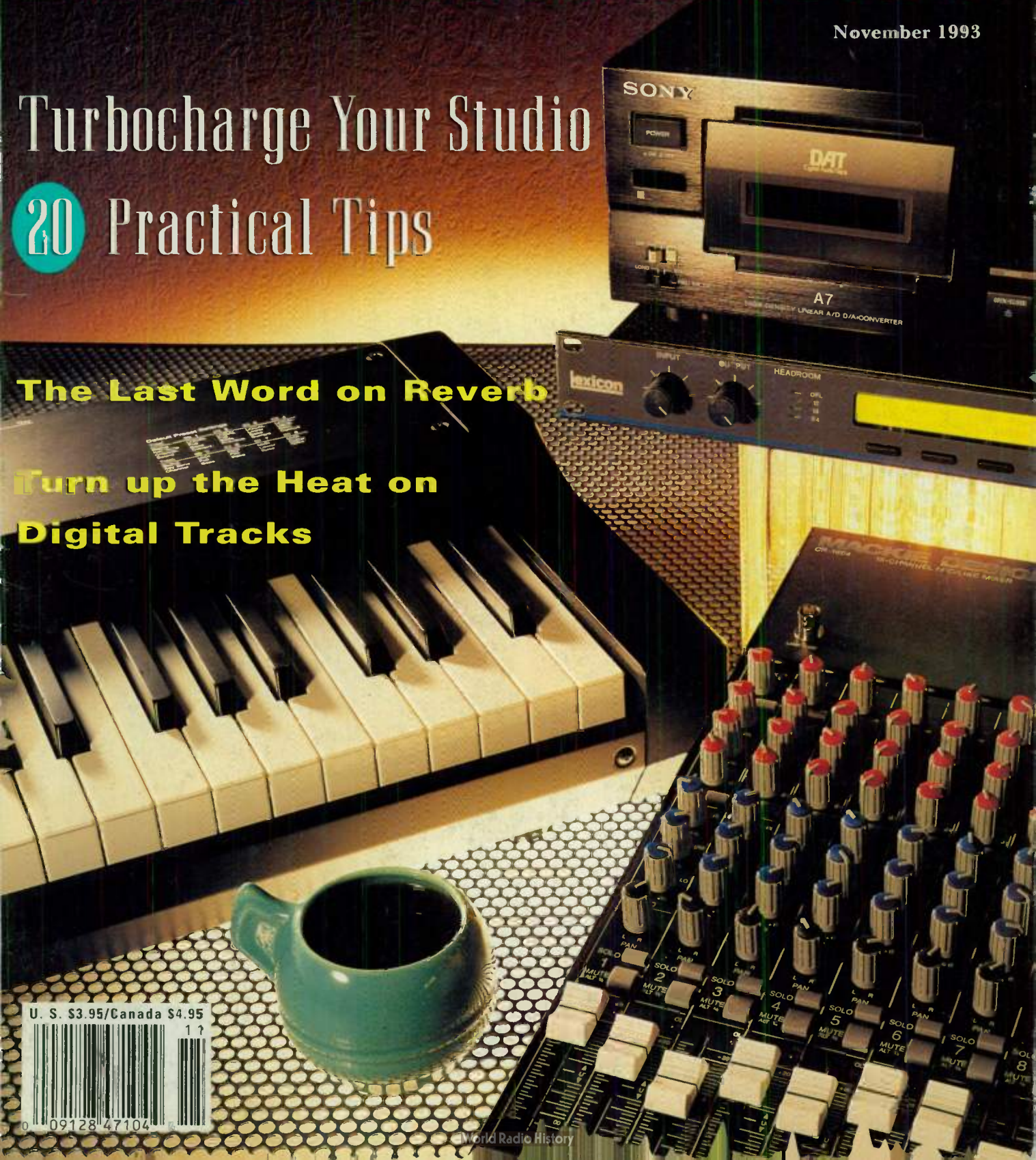
Ensoniq TS-10, Alesis BRC, and 7 More Reviews

Electronic Musician

November 1993

Turbocharge Your Studio

20 Practical Tips



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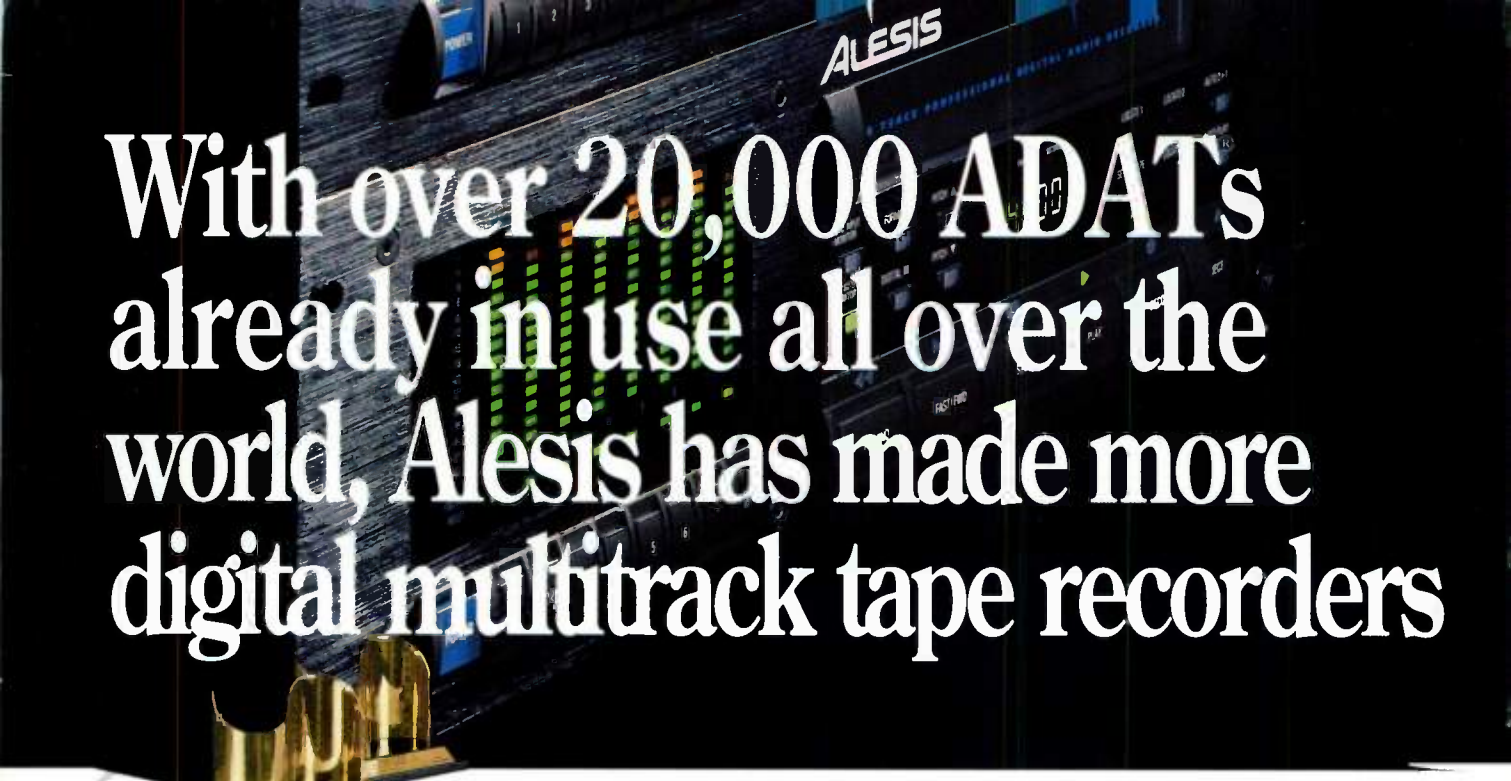
With over 20,000 ADATs already in use all over the world, Alesis has made more digital multitrack tape recorders



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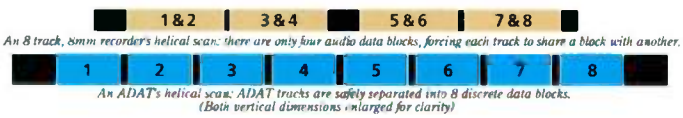
Its revolutionary impact on the recording industry has made ADAT the de facto standard in digital multitrack. The enormous number of ADAT users worldwide, the fact that Fostex has licensed the ADAT format for their own digital recorder, and the growing list of leading companies focusing on industry compatibility by becoming members of The ADAT Group™, all mean that when you choose ADAT, you're compatible with a vast array of music and audio equipment, now and in the future. And, you're supported by a network of professionally trained Authorized ADAT Service Centers worldwide.

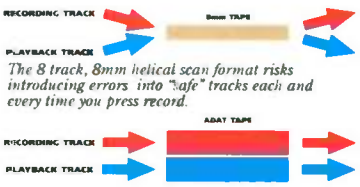
The ADAT Format – made for multitrack

ADAT records eight tracks of 16-bit linear, 48 kHz sample rate audio, with no data compression "tricks" or channel sharing. We chose Super VHS® (S-VHS®) tape as a foundation, then designed ADAT's data structure and heads specifically for the rough-and-tumble, back-and-forth, punch-in-and-out environment of multitrack recording. To make sure that recording one track wouldn't disturb any other track, we divided each helical scan into eight separate data blocks. Some digital recorders combine data from two different channels into the same data block on tape, which means that each time you record a track, another track must be read into a buffer and actually re-recorded even though it is in "safe" mode.



An 8 track, 8mm recorder's helical scan: there are only four audio data blocks, forcing each track to share a block with another.

An ADAT's helical scan: ADAT tracks are safely separated into 8 discrete data blocks. (Both vertical dimensions enlarged for clarity)



The 8 track, 8mm helical scan format risks introducing errors into "safe" tracks each and every time you press record.

The ADAT format records each track discretely, as all professional multitrack recorders should.

Bigger is Safer

Microscopic contaminants in the studio aren't just probable, they're statistically inevitable. If the format can't overcome them, they'll cause mistracking, noise, distortion, even total muting of the audio. Formats smaller than S-VHS are more vulnerable to contaminants, dropout, and misalignment, especially when exchanging tapes between machines. One 8mm digital format attempts to squeeze the same amount of sound into one-tenth the tape area that ADAT does. ADAT's S-VHS tape offers more total surface area to meet the demands of digital recording, and its wider 100 micron tracks are five times less vulnerable to being derailed by dust. Because even though technology makes it possible to make formats smaller and smaller, dust stays the same size.

Comparison of tape areas for 1 second of audio: ADAT (1,211 mm²) and the 8 track, 8mm helical scan format (133 mm²).

Actual microscopic comparison of the ADAT tape format and the 8 track, 8mm helical scan format (enlarged approximately 100 times).

ADAT's wide 100-micron tracks offer an extra margin of safety for digital audio.

The 8mm's 20-micron tracks squeeze more data into the same area, with little room for error.

than any other company. More than Sony. More than Mitsubishi. More than Yamaha, Akai, and Tascam combined.

More than just a tape recorder– The ADAT System

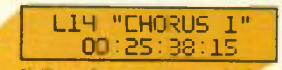
ADAT, when combined with the BRC™ Master Remote Control, is a complete digital recording and digital editing system with features that no other recorder, analog or digital, can match. The BRC is a full-function autolocator and MIDI/SMPTE time code chase-lock synchronizer. Plus, it controls digital copying between ADATs, like a disk-based recorder, but much simpler to use.

The ADAT MultiChannel Optical Digital Interface digitally transmits up to eight ADAT channels at once over a single fiber optic cable to any track on any ADAT in the system without repatching, all in the digital domain. Now you can "fly in" that perfect vocal part to multiple locations in seconds, with absolutely no generation loss. And our new QuadraSynth™ keyboard has an ADAT digital interface so you can record it without ever leaving the digital domain.

A fiber optic cable for digital connection is included with every ADAT.

The BRC Master Remote Control, shown with optional RMB™ Remote Meter Bridge, supercharges your ADAT System by adding SMPTE and MIDI synchronization, storable autolocation points, copy and paste digital editing and more.

ADAT/BRC digitally stores important session notes



Unlike analog autolocators, the BRC can recall 400 points, storable on each ADAT tape for later recall, so you can keep your mind on the project instead of having to remember minutes, seconds and frames.

Instead of scribbling notes on cumbersome studio track sheets, the BRC lets you store 400 autolocation points, 20 Song start points, punch in and out points, MIDI tempo maps, SMPTE offsets, and more in the two-minute data header of the ADAT tape. The BRC's alphanumeric display lets you name each cue point and song. It even has a handy built-in list of 16 standard cue point names you can edit.

The ADAT Worldwide Network

Thousands of ADAT Worldwide Network™ multitrack recording group members are reaping the benefits of choosing The ADAT System. As WWN members, they are able to collaborate and exchange ADAT tapes with other talented musicians, producers, composers and engineers throughout the world. Alesis is proud that so many creative people worldwide are using this American-made product, making ADAT the most popular digital multitrack tape recorder in history. The recording professionals below don't endorse ADAT, they *use* it every day. Their credentials speak for themselves. Visit your Authorized ADAT dealer and see what the new standard in digital multitrack recording can do for you.

Dave Rouze
Technical engineer for Larry Carlton, currently using ADAT to record all Larry's live concerts. 2 ADAT's and a BRC.

Jay Graydon
Two time Grammy® Award winning (twelve nominations) producer, engineer, writer and guitarist. 4 ADATs and a BRC.

Owen Bradley
Country Music Hall of Famer. Producer of many legendary country music artists. 9 ADATs and a BRC.

Francis Buckley
One of the top dance and pop engineers in Hollywood. 4 ADATs and a BRC.

Web Staunton
Grammy-nominated chief engineer and studio owner. 3 ADATs and a BRC.

Mick Guzauski
L.A.'s leading platinum mixdown engineer. 4 ADATs and a BRC.

Andy Hilton
Owner and Chairman of the largest pro-audio equipment-for-hire company in the U.K. and Europe. Plenty of ADATs.

Ray Benson
One of country music's hottest producer/arranger/writers. 3 ADATs and a BRC.

Tom Size
Has engineered and mixed a wide range of music from rock to legendary jazz. 3 ADATs and a BRC.

Russell Brower
Two Emmys (eight nominations). Sound designer and producer for film, television and major theme parks. 2 ADATs and a BRC.

Tim Wilson
Consultant and system designer for leading recording artists and songwriters. Has installed more ADATs than he remembers.

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World Radio History

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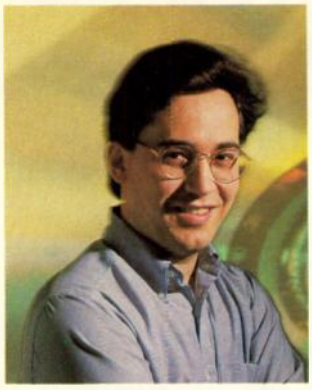
Cover: Photo by Stephen Tang. Special thanks to Ensoniq, Lexicon, Mackie Designs, Sony, and Foundesign.

The Joy of Tinkering

Building a high-performance studio requires spending some time "under the hood."



ANNE HAMERSKY

I've never been much of a car fanatic, but I am intrigued by folks who spend weeks, months, or even years tinkering with their automobiles. Weekend after weekend—and sometimes night after night—they toil under the hood, behind the dash, and beneath the chassis to perfect their car's appearance and performance.

Many electronic musicians take a similar approach to their personal studios. They spend months lovingly piecing together the studio of their dreams, adding new synth modules, high-quality mics, signal processors, and so on. Depending on the budget and goals involved, this process can continue for years, leading to the creation of home-based private studios that put many commercial facilities to shame.

Like car hackers, many **EM** readers start out as casual tinkerers—changing a connection here and synth preset there—and graduate into full-blown electronic-music gurus. Along the way, they develop an appreciation for the fine details that separate one piece of gear from another and one studio from another.

Of course, before you can tweak, it's important to have the right gear for your studio. In fact, music sessions are little more than exercises in frustration without essential pieces, such as a multitimbral synth or two; a computer-based sequencer; a few high-quality mics; a solid mixer; accurate monitors; some effects and dynamics processors; and a multitrack tape deck. But once you have the appropriate gear and have read all the manuals, any time spent refining your existing system's performance can pay for itself many times over.

The type of tinkering you do will depend on the gear you own and what you want to do, but there isn't a piece of gear in your studio that wouldn't benefit from some in-depth probing and creative experimentation. In addition to the immediate benefits you can derive from these efforts, the work you do will help you troubleshoot more difficult problems that may arise down the road.

To point electronic tinkerers in the right direction, this month's cover story, "Home Improvement," on p. 36, offers twenty tips on how to get the most from what you already own. You'll learn that maximizing your studio sometimes requires "directing" your gear to do things it wasn't designed for. We also discuss a few pieces of equipment you may want to consider adding to your rig. Whether you're interested in creative applications, practical tips, ingenious problem-solvers, or stretching a limited budget, you're bound to find a few ideas you can put to immediate use.

This issue also has a few other articles dedicated to helping you improve the performance of your gear. "The Proper Ambience," on p. 54, offers an in-depth look at digital reverbs and how to program them. The article explains all the important reverb parameters and offers loads of useful tips for creating better sonic environments.

This month's "Recording Musician" column, on p. 82, discusses the fundamentally important, but often misunderstood, concept of gain stages. One of the easiest ways to reduce the sea of hiss that may be drowning your recordings and live sound is to apply proper gain structuring. This article tells you how.

Putting together a high-quality music system seems like a quixotic, never-ending quest. And if you're not careful, it will be. But if you spend some time fine-tuning your system, the high-performance you build into your music machines can speed you into the winner's circle.

Bob O'Donnell

Electronic Musician

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tel. (212) 909-0430
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Subscription Services Office
(Address changes and customer-service inquiries)
PO Box 41525
Nashville, TN 37204
tel. (800) 888-5139
or (615) 377-3322

Electronic Musician (ISSN: 0884-4720) is published monthly by ACT III Publishing, 6400 Hollis St. #12, Emeryville, CA 94608.

This is Volume 9, Number 11, November 1993. One year (12 issues) subscription is $24; outside the U.S. is $49.95. Second Class postage paid at Oakland, CA, and additional mailing offices.

POSTMASTER: Send address changes to Electronic Musician, PO Box 41525, Nashville, TN 37204. Editeur Responsable (Belgique): Christian Desmet, Vuurgatstraat 92, 3090 Overijse, Belgique.

An ACT III PUBLICATION
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In fact, the DP/4 gives you more sound variation potential than any other signal processor available. For about half the price.

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Disc One contains a complete product description, and a variety of wet-vs.-dry comparisons so you can hear most of the DP/4's effects algorithms and Presets.

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To hear the two CD set just stop into your local Authorized ENSONIQ Dealer. For the one nearest you call 1-800-553-5151. Or to order a set for yourself, mail a check or money order for $5.00 to ENSONIQ. And then kick back and listen to what the DP/4 can do for your next recording. Who knows, maybe we'll be featuring you on our next CD...

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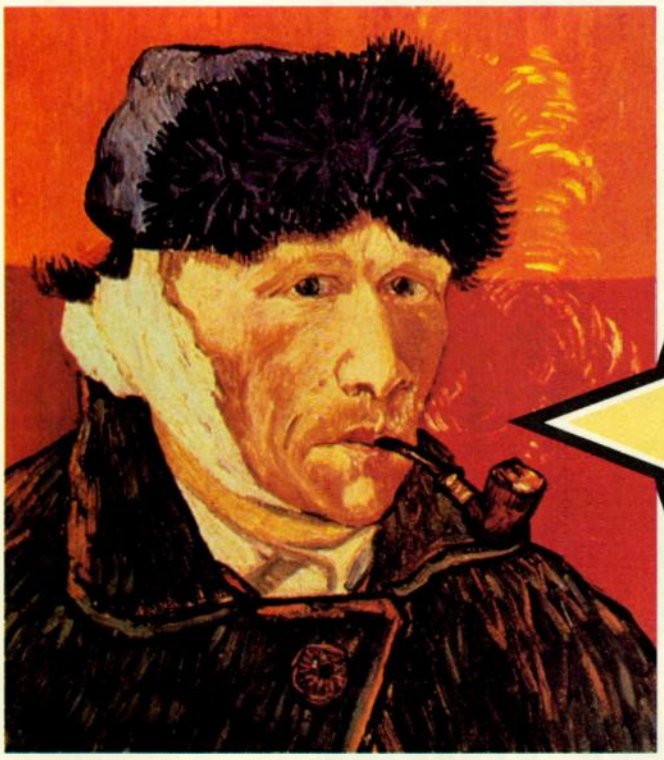

If I'd known about this, I would have cut off my nose instead.

Being the visual person he obviously was, Vincent Van Gogh would have instantly appreciated the new JV-1000. Because unlike so many instruments that look exactly like so many other instruments, this particular synthesizer workstation looks unlike anything you've ever seen before. And as you'll learn in a moment, it also performs unlike anything you've seen before. But, we're getting ahead of ourselves.

See the LCD display? The one on the left or the one on the right, you ask? And that's the point, because the new Roland JV-1000 actually has two of them—one for the synthesizer and one for the sequencer.

With the JV-1000 we've integrated our remarkable synthesis, waveform and sequencing technologies into a single unit that provides unrivalled music production and performance capabilities along with a 76-note keyboard.

The synthesizer section has 4Mbyte of ROM waveforms, which encompass everything from breathtaking acoustic instruments to dynamic synthesizer textures to an extraordinary array of drum and percussion sounds.

And if you'd like to expand the waveform memory

Roland Corporation US,

further, get your hands on any of our SR-JV80 series of 8Mbyte Expansion Boards or PCM waveform cards.

If you wish, you can also take advantage of a user-installable Roland VE-GS1-01 Expansion Board and in the process, add a complete GS synthesizer module. You'll be rewarded with 226 sampled sounds, drum kits and digital effects, as well as an additional 28 voices of polyphony and 16 part multi-timbral capability—giving you an extraordinary 56 voices of polyphony and 24 part multi-timbral performance literally at your fingertips.

The sequencer on the JV is our widely acclaimed Super MRC with eight tracks, each of which has 16 channels. A staggering array of editing capabilities gives you easy access to every event on every track.

The 3.5" floppy disk drive can save and load both your Super MRC sequencer files and SMF, or Standard MIDI Files, thereby giving you access to the extensive Standard MIDI/GS library that's now available. And your sequences can easily be loaded to and from any other sequencer using the SMF format.

The 76-note keyboard is both velocity-and after-touch-sensitive. It's capable of controlling up to eight external MIDI channels simultaneously, each with its own independent key zone and volume, panning, velocity curve and program change.

You'll find eight control sliders on the front panel which can be used either for editing sounds, for mixing volume and panning on sequenced tracks, or even for external MIDI control. Consequently, the JV-1000 works beautifully as a MIDI master keyboard.

By now you no doubt appreciate that the new Roland JV-1000 is a truly remarkable workstation. All that's left is to play one at your music store. You'll appreciate your ears as never before. **Roland®**

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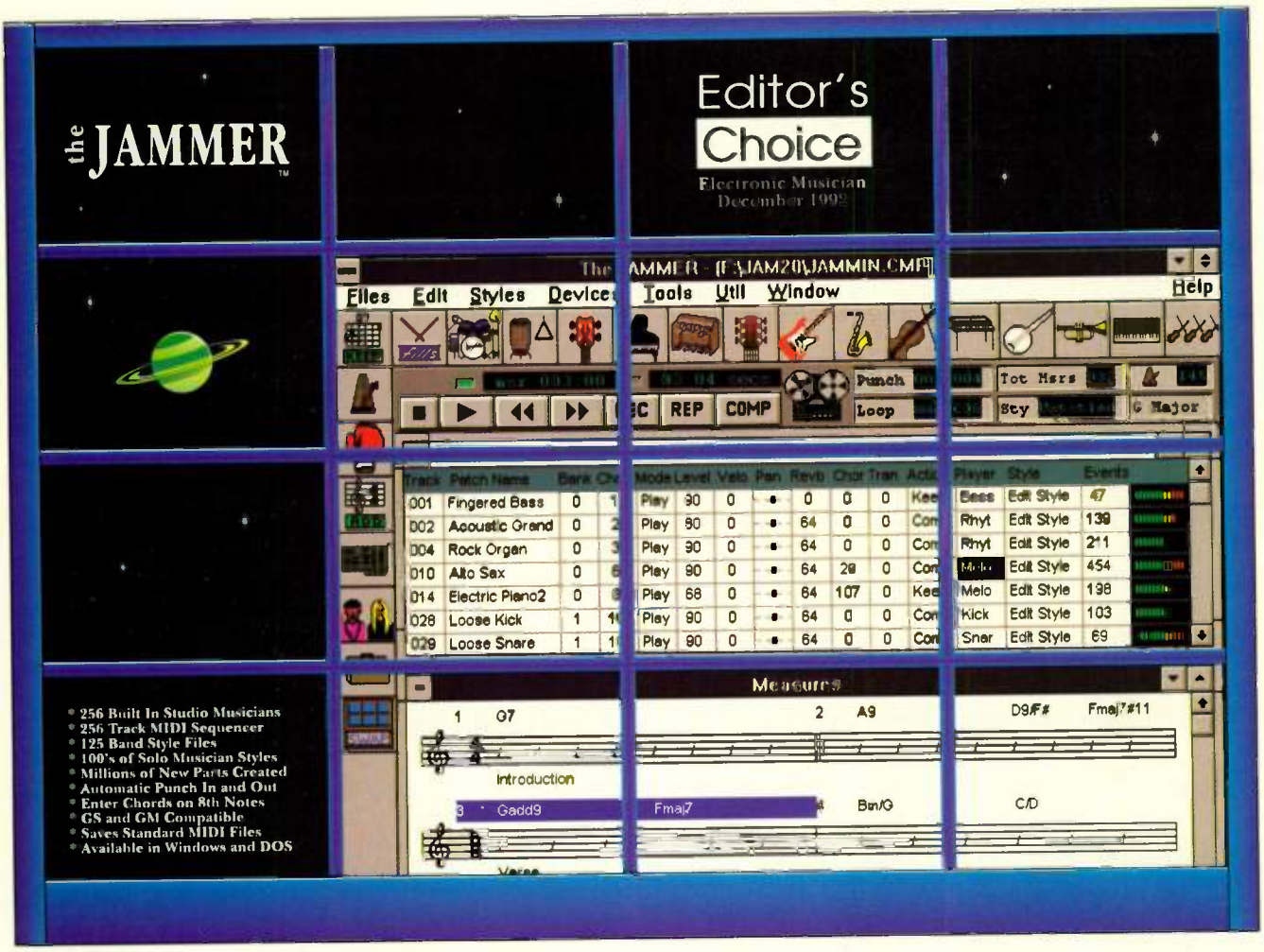

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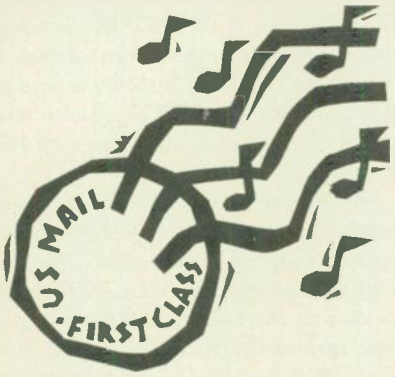
LETTERS



GOOD NEWS/BAD NEWS

Although I've read this magazine since it was called *Polyphony*, I've never felt the need to write a letter. But the September 1993 issue was a fine example of **EM** at its best, and it convinced me it was time to give you feedback. **EM**'s strength has always been in its how-to pieces, and the articles "The Electronic Orchestra" and "Maximizing Sample RAM" in that issue were excellent, lucid, and practical guides.

Another thing I've liked is your probing of the esoteric. The pieces you've done on subjects such as fuzzy logic and algorithmic composition always inspire excitement about the future of electronic music.

While I'm praising you, I'd also like to offer up some constructive criticism: **EM** tends to fall down when it comes to its product-comparison articles (sampler shootouts, etc.). They tend to be little more than laundry lists of features. I've *never* come away from one of these pieces with a feeling of which products were the best for particular applications. If you aren't going to draw any conclusions, you needn't bother with the comparisons. Be more like *PC Magazine* and give outstanding products the credit they deserve.

Richard Kaczynski
Royal Oak, MI

Richard—First, thanks for the kudos and the loyal readership. Second, your criticism about product comparisons is well-taken. When we began doing buyer's guides on a regular basis three years ago, the goal was to provide readers with a thorough listing of available product choices and an explanation of salient features. We left most of the analysis up to the readers, believing that with the appropriate information, they could make the best choice for themselves. Over time, we've realized this theory hasn't always translated well into practice and readers want more critical analysis. In response, we've made an effort to increase the number of product shootout articles, in which we compare similar products head-to-head. Recent examples include "Windows Shootout," on Windows*-based sequencers in the November 1992 issue; "State-of-the-Art Sequencing," on Mac-based sequencers in the May 1993 issue; and "On the Beat," on drum machines in the July 1993 issue. Buyer's guides still fill a critical need for many readers, particularly when a large number of product choices are available, but you can look forward to more product-specific comparisons in the future.—BO'D.*

WORKING TOGETHER

I read Dave Snyder's letter in the September 1993 **EM** with great interest. I am in a similar situation, having advanced muscular dystrophy.

I became computer capable in early 1991 through a sip-and-puff device used in conjunction with a Morse code program. This system operates all keys on my IBM keyboard. That year, after being musically disabled for nineteen years, I discovered the world of MIDI. I bought a Yamaha TG-55 synthesizer for straight piano music. Through my own research, I found *MusicPrinter Plus* notation and sequencing software. It works like a dream for reading a composer's notation in the most literal sense, and it often reveals some startling results. Most of my overall dynamics and rubatos are worked into the MPP file. Then I turn it into a Standard MIDI File and export it to *Cakewalk Pro*. I have to imitate manual pianistics by manipulating the velocities and durations during chordal activity. It works!

I would like to contact other disabled MIDI musicians. I am interested in developing some real-time expression inputting devices because I can't do any real-time sequencing. Perhaps we could establish an organization or user group (including you, Dave Snyder), so we can get our heads together and help one another. Anyone interested, please contact me at 2494 W. Grand River Ave., Okemos, MI 48864.

Richard Carlson
Okemos, MI

K2000 UPGRADES

I usually enjoy reading Alan Gary Campbell's "Service Clinic," and I normally find the information he presents accurate and interesting. Unfortunately, the answers he gave in the September issue regarding the K2000 and P/RAM were full of inaccuracies or incomplete information.

First, there is only one P/RAM kit sold for both the keyboard and rack, not two separate kits as he stated. Second, although it is correct that installation of the FK-1 fan kit is required if adding P/RAM, it should be noted only the K2000 comes without a fan. The K2000R, K2000S, and K2000RS all come with the fan installed.

Alan states version 1.30 is included with the P/RAM kit. This is incorrect. It is sold separately. (It should be noted the majority of K2000s are 1.30 or higher. Only the units manufactured during the first nine months of 1992 had versions lower than 1.30.)

Regarding the pricing of the 1.30 firmware, Alan says the price changed from $10 to $60 after a few months. Again, this is incorrect. Until September 1, 1993, our policy was to charge $10 for the upgrade and to add a $50 charge for the EEPROMS, which was credited back to the service center's account once the previous version's chips were returned. Starting on September 1, 1993, we changed our policy so that the chips are no longer sent back to us, and we now charge $50 for the upgrade. The reason for this change is the administrative effort required to process the credits became too costly.

Finally, Alan states the "1.30 kit is intended for user installation." This is incorrect. All options involving the

DAVID SMITH

installation of anything inside the unit should be done by an authorized service center.

Dave Fox
Senior Product Specialist
Young Chang/Kurzweil
Los Angeles, CA

Alan Gary Campbell responds: *Obviously, I had communications problems with Kurzweil's service department. I apologize for any inconvenience this may have caused.*

SEEKING STRANGENESS

My volley is a response to Steven Slaughter's letter, published in the August 1993 **EM**.

I am a contemporary classical composer and a composition teacher. I know nothing about Primus, who Mr. Slaughter tells me is strange, and I have not heard their work, which I suppose is also pretty quirky. But I do know the most difficult thing to teach beginning composers is to *be* themselves.

If being yourself means a little bit of strangeness, so be it. When you're doing something that counts, strange is far better than commonplace, and the history of our art bears me out on this score.

Wake up, Mr. Slaughter: It's the strange people who make it into the encyclopedia. The fellow who figured out the wheel probably had a few pretty unpleasant social habits, too.

David Tcimpidis
Livingston Manor, NY

LET'S COMMUNICATE

In regard to William Holt Penninger, Jr.'s letter in the June 1993 issue, there are so many of us creating our "ArtMusiK" here in the underground, it's amazing no one has devised a system of communication between these far-flung artists. Perhaps the time has come for a newsletter.

Computer bulletin boards are fine, and I use them myself, but there are those of us who aren't set up with the computers and modems required to access them. Radio programs are all right, too, but again, they go only so far. These avenues allow little or no tech talk between musicians, personality profiles, etc., that a newsletter would. If such an item is available, I want to know about it. If not, I want to know who is interested in a publication aimed solely at the underground market; if the response were favorable, I want to get the ball rolling!

Considering what we pay for a subscription, what's the small cost of a newsletter aimed at our group's unique situation? Maybe a classified section or "trading post"? Contributing authors? Any ideas? I want to hear from you folks! Drop me a line at this address: 210 E. Vilbig, Irving, TX 75060.

Jim Devault
Irving, TX

NON-STANDARD STANDARD

I recently added a Roland SD-35 Sound Canvas to my Peavey C8-equipped home studio. Unfortunately, I didn't find out until I hooked the two together that these great pieces of equipment, adhering to the MIDI standard, do not work well together.

The problem I ran into is this: The MIDI spec says controller 0 followed by controller 32, followed by the program change number will select the desired bank, hence the desired variation of your base instrument on the GS-compatible SD-35. No problem, right? Wrong! The Peavey C8 sends out controller 0 with a value of 0, followed by controller 32 with the bank select number, followed by the program change number. All you tell the C8 is what bank you want, and it handles the rest. However, the Roland SD-35 wants something different. It requires controller 0 with the bank select number, followed by controller 32 with a value of zero, followed by the program change number.

So, it seems, the Peavey C8 uses controller 32 as the decision maker while the Roland SD-35 requires controller 0 as the decision maker. As a result, there are over 90 variations of sounds on the SD-35 I can't select because it doesn't like what the C8 has to say. I have talked to both companies about this problem, and I got the impression neither was going to change anything.

I would appreciate any suggestions on how to work around this problem.

Robert B. Stocking
Miles City, MT

Robert—When is a standard not a standard? When different manufacturers implement it in different ways. You have hit upon one of the ambiguities of MIDI; even the MIDI spec itself is unclear on this point. If you have 128 or fewer banks to select from in a sound module, the original MIDI spec says to use only the MSB (Control Change 0, or CC0), which is what the SD-35 expects. The addendum to the MIDI spec in which the Bank Select message is described seems to indicate the LSB (CC32) should be used for bank numbers 0 through 127, which is what the C8 sends. Personally, I agree with Peavey's approach over Roland's.

Unfortunately, the C8 cannot be programmed to send CC0 with different values in its present incarnation. The only solution I know of requires a MIDI processor such as the Yamaha MEP4, or a program such as Opcode's MAX *on the Macintosh. Run the MIDI from the C8 through the processor, which can be programmed to change the message string:*

[CC0, Value=0] [CC32, Value=Bank #] [Program Change, Value=Pgm #] *into the string:*

[CC0, Value=Bank #] [CC32, Value=0] [Program Change, Value=Pgm#] *while passing all other messages through unchanged.*

I know this is an inelegant (and potentially expensive) solution, but it is the only one I could come up with. Apparently, Peavey is looking into updating the C8's software to address this problem.—Scott W.

ERROR LOG

August 1993, "Roland SP-700 Sample Player" review, p. 114: We stated that the Peavey DPM SP sample player lacks filters; in fact, the Peavey SP has digitally controlled, non-resonant filters.

August 1993, "Modern Manuscripts," p. 34-35: The chart states Temporal Acuity Products' *Nightingale* has no user preferences. On the contrary, *Nightingale* has a special utility, *NightCustomizer*, that controls user preferences, MIDI preferences, instrument tables, PostScript output preferences, spacing tables, and remapping the Mac keyboard.

September 1993, "The Electronic Orchestra, Part 1," p. 44: We accidently stated that when transposing, "If your piece is in D, the trumpet part should be in C." The trumpet part should actually be in the key of E.

Address correspondence to "Letters," Electronic Musician, 6400 Hollis St. #12, Emeryville, CA 94608. Published letters may be edited for space and clarity.

MACKIE DESIGNS' EXCLUSIVE, MODULAR 16/32/48 CH. MIXING SYSTEM. THINK OF IT AS A GROWTH EXPERIENCE.

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COMBINE IT • $249

A favorite of "power user" film scorers and session keyboardists, the **MixerMixer** ($249*) effectively turns up to three 1604s into "one big mixer" without "cascading" or using AUX sends. Three CR-1604s and a MixerMixer yield 48 line inputs (24 of which are mic inputs), 12 stereo AUX returns, 24 direct outs and 3 stereo/6 mono submasters.

FADE IT • $75

All combining is done at unity gain, so no level or headroom is lost and no additional noise is introduced. Matching 100mm **Remote Fader** ($75*) controls master level of all mixers plugged into the MixerMixer (existing master faders become submasters). Comes with its own 6-ft cable and can be bolted to the side of a CR-1604, handheld or attached to any surface. Not shown: The **CordPack** ($69*) all patch cords needed to combine 3 CR-1604s.

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Instead of cheap, integrated circuits, the CR-1604 has six totally discrete preamplifiers with conjugate pair, large-emitter transistors. The result is ultra-low noise (–129dBm E.I.N. @150 ohms), low distortion (0.005%), astonishing headroom and 300K bandwidth that contributes to the preamps' transparent accuracy. At any gain level, you can handle everything from a close-miked kick drum to a flute with exceptional sonic fidelity and freedom from overload. These preamps have made the CR-1604 legendary among pros who are used to $150,000+ mega consoles. But what if you need more than six mic inputs? Simply add ten more of the same with our **XLR10 Mic Preamplifier Expander** ($199*). It attaches in minutes to form an integrated, mechanical/electronic whole and includes its own +48V phantom power switch. Plus you can still use the line inputs on Chs. 7-16!

AUTOMATE IT • $849

Never again will you lose that elusive "perfect mix." Save it, recall it and fine-tune it over and over with our **OTTO-1604 MIDI Automation Retrofit System** ($849*). Consisting of an internal gain cell board and external MIDI control box, OTTO provides realtime fader and muting automation of CR-1604 channel inputs, AUX Returns 1-4, ALT 3/4 buses and master outputs. The system works with any Macintosh, Atari or PC sequencing program which supports graphic faders. FREE **OTTOmix**™ Mac® automation software that precisely duplicates CR-1604 controls, and adds features such as subgrouping is also included.

ROTATE IT • $25

Yet another twist to the CR-1604's unique rotating pod! The **RotoPod** bracket set ($25*) creates a 10-rack-space, jacks-to-top configuration with all inputs and outputs on the same surface as the mixer's controls.

* Suggested Retail Prices. Slightly higher in Canada. Actual prices at your local Mackie dealer will vary depending on a variety of economic factors including phase of the moon and barometric pressure.

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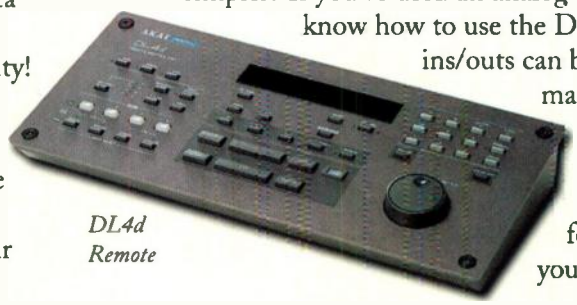
Ten Reasons Why You Should Cho

1. TAPELESS EDITING The DR4d can simultaneously record 4 tracks directly to standard SCSI-compatible hard disks, not tape. Tape recorders which use a cassette format (VHS, 8mm, etc.) have a huge problem: without at least two machines, you can't edit. But even a single DR4d allows random access editing that tape recorders just can't offer. Move, Copy, Insert, Copy + Insert, Move + Insert, Erase, and Delete with ease. Edit with complete confidence, because if you try an edit but change your mind, the Undo function will instantly restore the previous arrangement. It's a breeze to copy any part of a track and paste it anywhere on any track, even with a specified number of repeats. Or perhaps use the Insert commands to instantly slide track data in time against other tracks. This editing power encourages experimentation, and thus, your creativity! Imagine it. Do it.

2. NO WAITING Another problem with tape is the time required to physically move from one point on the tape to another. Concentrating on your music is what's important, not waiting for tape to shuttle back and forth. Never again waste such precious time: the DR4d allows you to instantly move to 108 different locations. Set up repeat sections, jam along with your tracks, then drop into record to capture it all while it's still immediate, fresh.

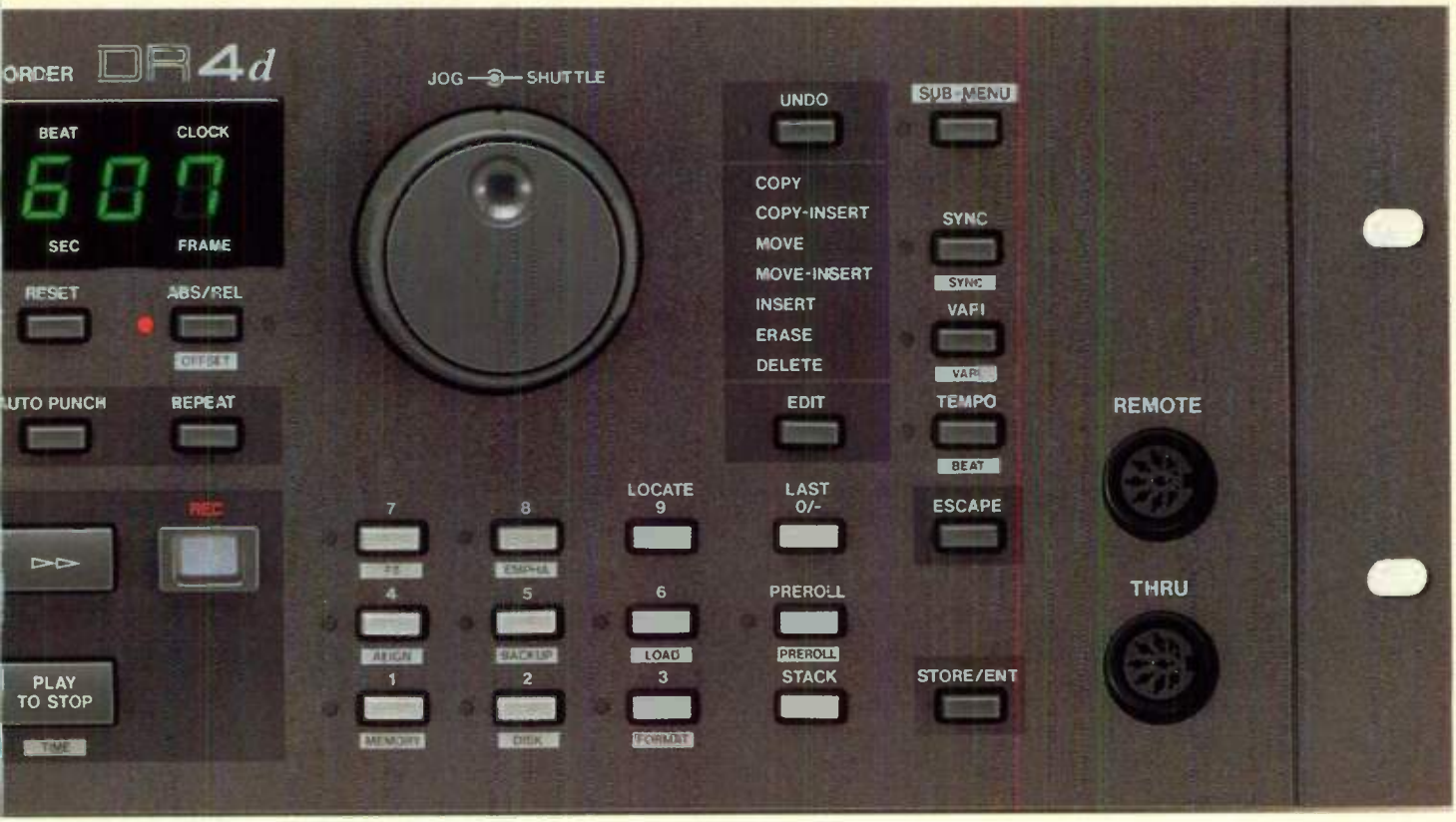
3. JOG/SHUTTLE Another cool DR4d advantage is the ability to offer scrubbing of audio, like "reel-rocking"on analog decks - only with much better quality. Our Jog/Shuttle wheel lets you scrub through the audio at various speeds, forwards or backwards. So finding precise editing points is only as complicated as using your ears.

4. FAMILIAR OPERATION One concept we *did* want to carry over from tape recorders is the user interface. Friendly, tape machine-style controls make the DR4d by far the easiest hard disk recorder to use. With dedicated buttons for Play, Stop, Rewind, Fast Forward, and so on, what could be simpler? If you've used an analog deck, then you know how to use the DR4d. Punch-ins/outs can be performed manually or automatically from the front panel, or via footswitch. Like you'd expect.



DL4d Remote

5. EXPANDABILITY Up to four DR4ds can be chained together to create a 16-track system, simply by plugging an optional cable between units! And the optional DL4d Remote makes it a snap to



ose the DR4*d* Hard Disk Recorder

control all of them. An optional, factory-installed 200 MB internal hard disk offers 32 track minutes of recording right out of the box. The DR4d can handle up to seven hard disks and supports seamless overflow recording across multiple disks. With enough disk storage space, you can actually record on all four tracks for an incredible 24 hours!

6. EXCELLENT CONNECTIONS Four balanced TRS 1/4" Input and Output jacks, easily switchable between -10 and +4 dBu levels, simplify interfacing with any type of console. The DR4d's pair of digital I/O ports allow communication with other digital devices in the form of both XLR and RCA connectors (AES/EBU or Type II selectable), as well as provide DAT backup. And then there's the supplied SCSI port for access to external hard disk drives. Just plug and play!

7. YOU'VE GOT OPTIONS And affordable ones, at that. For digital access to all four channels simultaneously, the IB110D provides the two additional AES/EBU ports. For SMPTE timecode applications (slave or master), the IB112T is installed in seconds. The IB113M interface gives you MIDI In, Out, and Thru, and the IB111S is a second SCSI port which will allow connection to computers for visual waveform editing and magneto optical drives for data backup.

8. DEDICATED DESIGN The DR4d is a dedicated digital audio product rather than an add-in board for a computer. It's a tool designed for a single purpose: to record and edit audio precisely, effortlessly, and affordably. We think you'll agree that it succeeds on all counts beautifully.

9. SOUND QUALITY The DR4d contains Akai's own advanced digital technology, including super-clean 18-bit 64x oversampling A/Ds and advanced single-bit 8x oversampling D/A convertors with 18-bit resolution. Industry standard sample rates include 48, 44.1, and 32kHz. In short, the quality is superb and with a full 96dB dynamic range, you can rest assured of always sounding your best.

10. $1995.00 Simply put, the DR4d is the best value in digital recording today. For the first time, the nucleus of a professional quality 4-track hard disk recording system can be yours for only $1995.00! Just add internal or external hard disks, and you're ready to use our latest masterpiece for creating your next masterpiece.

DR4d

AKAI DIGITAL

Specifications subject to change without notice
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P.O. Box 2344 Ft. Worth, TX 76102
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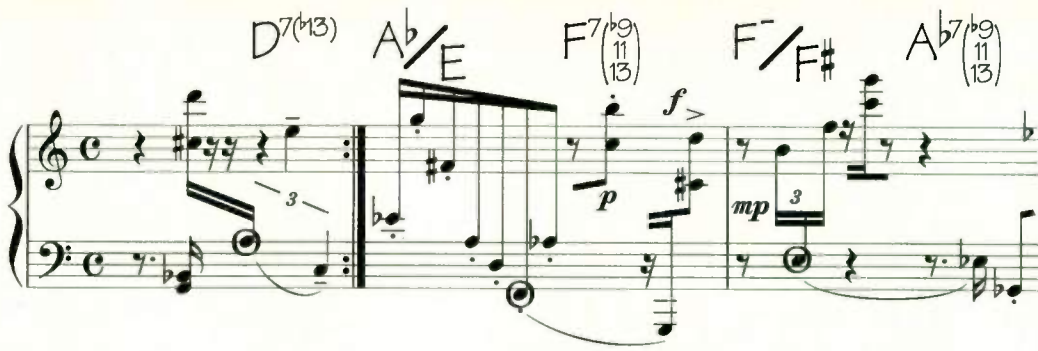
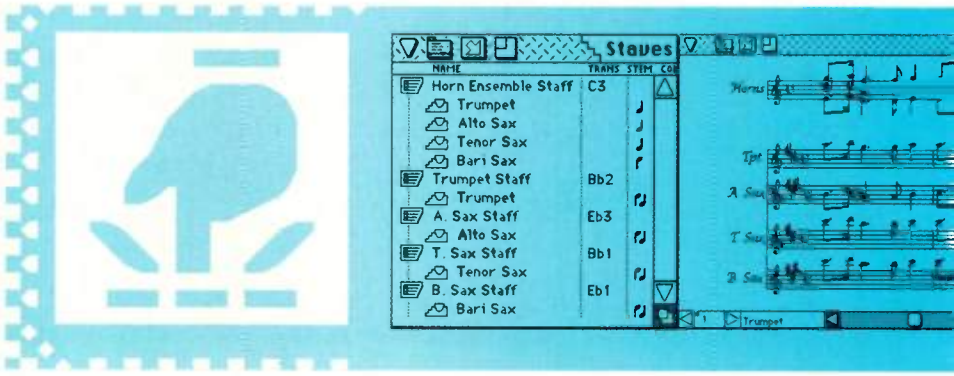


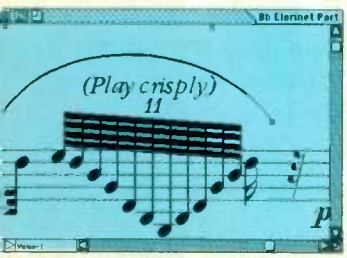
"Printed output is excellent, and on Quickdraw printers, the best I've seen." **Alan Belkin, Computer Music Journal**

"...it's going to be the notation software to own." **Jim Aikin, Keyboard**



"Interface details have been meticulously refined to perform various tasks more easily." **Alan Belkin, CMJ**

Mosaic Scores Big with Composers and Critics Alike.



Ultimately, the music notation software to own. Flexible, friendly and fast. Mosaic strikes a balance between exhaustive programs that are difficult to use and simple applications that lack the flexibility needed for professional results. **"We've seen other notation programs undergo update after update without getting this good."(Jim Aikin, Keyboard).** Move notes, dynamics and symbols simply by dragging. Mix any combination of voices your muse desires. Mosaic's multiple views have linked music and multiple formatting. Edit your music in any view and all other views are updated, automatically. **"Mosaic's 'Views' window - the best approach I have ever seen..."(Alan Belkin, Computer Music Journal).** Breeze through every part with freedom of expression.

Mosaic works like many well known Mac drawing programs. Simply click and drag, shift-select regions, shift-click to constrain movement, etc. Zoom from 20%-800% for precise editing. Work in true WYSIWYG. And with unlimited undo/redo, you can experiment to your mind's desire. **"[Infinite undo] is such a cool feature, we started fantasizing..."(Jim Aikin).** Mosaic's key bindings feature puts the tools you use most at your fingertips, so entering music is natural and intuitive. **"Automatic enough to allow novices to get notes on paper..."(Ken Gruberman, MacUser).** Mosaic, the program to own, so you can spend your time on something more important...your music. Call Mark of the Unicorn for your authorized dealer. And if you already own notation software, ask about our competitive upgrade.

COMPOSER'S MOSAIC

4½ **MacUser Magazine**

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WHAT'S NEW

▲ DBX 296 SPECTRAL ENHANCER

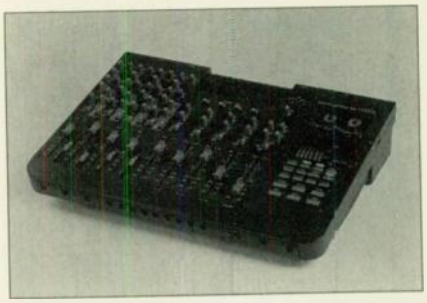
Dbx announced the 296 Spectral Enhancer ($349), the latest addition to the company's Project 1 signal-processor series. The dual-channel processor provides user-selectable, high-frequency detail-enhancement; low-frequency detail-enhancement; and hiss reduction (unusual in enhancers). Low frequency Detail is said to add bottom while reducing mid-bass mud. An LED meter indicates the amount of hiss reduction. AKG Acoustics; tel. (510) 351-3500; fax (510) 351-0500.

Circle #401 on Reader Service Card

▶ FOSTEX 380S

Fostex released the 380S Multitracker ($995), its latest top-of-the-line, multitrack cassette ministudio. The 380S combines a double-speed, 4-track cassette recorder with a 12-input mixer and defeatable Dolby S noise reduction.

Each of the four full-featured mixer channels offers a ¼-inch, mic/line input; an input-trim slider; 3-band EQ with sweepable midrange; two aux sends; pan; a monitor/stereo assign switch; an input-select switch; and a fader. Two of these channels also have XLR mic inputs and ¼-inch TRS insert points. The other eight ¼-inch mixer inputs are configured as four stereo channels and selected by a L/R Balance pot. The stereo channels can access either aux send and offer an assign switch and a fader, but no EQ or trim. The monitor section can audition the stereo bus, monitor bus, or both, and there are two front-panel headphone jacks. The mixer



also has stereo aux returns and a master fader.

The tape deck offers pitch control, manual and auto punch-in/out, and a Rehearsal function that lets you practice punches without recording the edits. In addition, you can set up punch-in/out points within a larger loop, effectively allowing pre- and post-roll for punches. A 7-segment LED ladder indicates levels for the four tape tracks and the L/R main bus. Fostex Corporation of America; tel. (310) 921-1112; fax (310) 802-1964.

Circle #402 on Reader Service Card

▶ BEYERDYNAMIC MV100

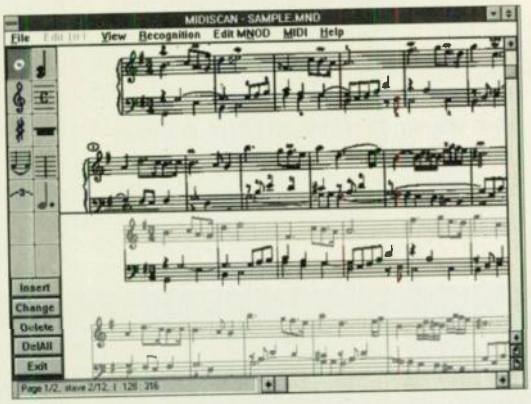
The MV100 stereo mic preamplifier ($1,200) fits its many features in a small (145 × 85 × 40 mm), lightweight box. It offers two balanced XLR inputs; mini-jack and phono line outputs; an on/off switch with indicator LED; overload and battery-status LEDs; a 3-position, input-attenuation switch; and a 2-position, low-frequency roll-off switch. The unit's phantom power supply is switchable between 8 to 12 volts and 48 volts.

Power is provided by two PP3 batteries or from an external supply. beyerdynamic; tel. (516) 293-3200; fax (516) 293-3288.

Circle #403 on Reader Service Card

▶ MUSITEK MIDISCAN

Musitek has released *MIDISCAN for Windows* 1.0 ($379), the first commercially available, optical character-recognition (OCR) software for music. The program analyzes scanned music in the form of TIFF files and recognizes note pitches, note and rest durations, chords, accidentals, ties, bar lines, clefs, and key and time signatures. The manufacturer claims up to 98% recognition accuracy. *MIDISCAN* can automatically process multiple-page scores, correcting skewed images while ignoring non-note objects such as titles, guitar tablature, lyrics, and other markings that don't translate into MIDI messages. Each page is processed in approximately three to five minutes.

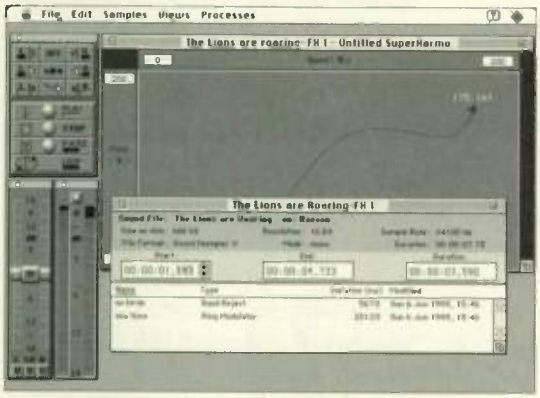


A graphic editor lets you correct the recognized score. The original TIFF file is displayed in one window, with the converted version below it. Notes are displayed with the same spatial relationships as in the original. You make corrections with an interactive graphic toolbox and save the final output as a multitrack Standard MIDI File. A Macintosh version is under development. Musitek; tel. (800) 676-8055 or (805) 646-8051; fax (805) 646-8099.

Circle #404 on Reader Service Card

▼ ARBORETUM SYSTEMS HYPERPRISM

Arboretum Systems' *Hyperprism* ($395) is the first real-time, audio signal-processing program for the Macintosh computer. Utilizing any Digidesign digital-audio card (Audiomedia I/II, Sound Tools I/II, or Pro Tools), the program lets you apply non-destructive, dynamic effects "on the fly" to an AIFF or *Sound Designer II* file, or to real-time audio routed through the sound card's digital or analog inputs.



You can apply lowpass, highpass, bandpass, or band-reject filters. Modulation effects include phasing, flanging, chorusing, tremolo, and vibrato. Special effects include ring modulation, frequency shifting, and simultaneously independent variation of speed and pitch. Other effects include envelope and pitch followers, stereo dynamics (rotates the sound in virtual space), and panning. These effects are accomplished on the Digidesign card, and delay effects can be simultaneously achieved using the Mac CPU.

The action takes place in a blue window with X and Y axes. The parameters assigned to the axes are determined by the user-selected effect. With the audio playing, you use the mouse to draw variations in the parameter settings in the blue window; the program applies these settings to the sound as you draw. You can draw curves and lines of any shape and length, limited only by available RAM.

The resulting algorithm can be saved as a *Hyperprism* file, and can be applied to the sound file to create a new, processed AIFF or *Sound Designer II* sound file. (A user-created algorithm is tied to a sound file and cannot be edited or applied to another file; however, editing and MIDI support are promised in a future upgrade.) You can reprocess the file to add more effects as many times as desired. Arboretum Systems; tel. (415) 931-7720; fax (415) 931-7725.

Circle #405 on Reader Service Card

▼ ALESIS MONITOR ONE

Alesis is shipping the Monitor One near-field monitor speaker ($395/pr.). The 2-way, vented system uses a proprietary, 6.5-inch woofer with a mineral-filled, polypropylene cone; a linear, rubber surround; and a 1.5-inch voice coil. The company's SuperPort speaker-venting technology is said to produce solid, high-power bass transients and extended low-frequency response. The 1-inch, ferrofluid-cooled, silk-dome tweeter is fed by a passive crossover network divided at 2.5 kHz.

The 8.5 × 15-inch Monitor One handles 120W program power (200W peak) into 4 ohms and offers a frequency response of 45 Hz to 18 kHz (±3 dB). The outside of the cabinet is covered with a rubber polymer, providing a non-slip surface. The cabinets come in a mirror-image, L/R pair. Alesis Corporation; tel. (800) 5-ALESIS or (310) 558-4530; fax (310) 836-9192.

Circle #406 on Reader Service Card

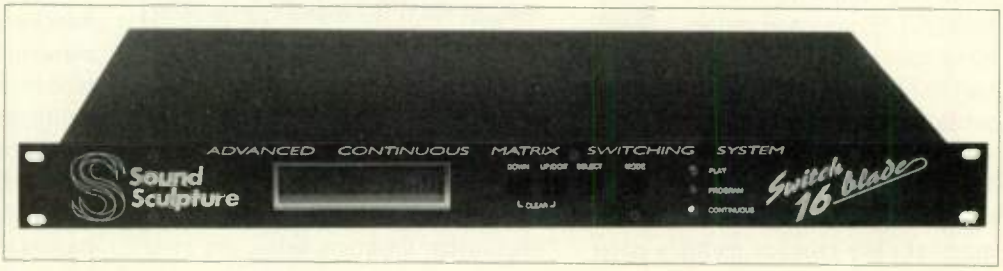
▼ SOUND SCULPTURE SWITCHBLADE 16

Sound Sculpture introduced the Switchblade 16 ($2,299), a 16-in/16-out, MIDI-controlled, audio switching system. The device uses true matrix switching to route any combination of inputs to any combination of outputs, electrically mixing inputs that are patched to the same output. This lets you custom-configure the signal path, placing signal processors and preamps in any order (including in parallel), and reconfigure the chain with a Program Change command. A Preset Manager also lets you create and step through custom chains of Switchblade presets. Each Switchblade preset also can send separate Program Change commands on each of the sixteen MIDI channels. Eight relay-based switches provide footswitch-simulation for switching amp channels and non-MIDI effects.

The signal level for each input-to-output path is programmable from -42 dB to +6 dB, so you can control the balance between effects, optimize the gain staging to minimize noise and distortion, and achieve image-placement effects. Each level can be continuously controlled via MIDI or an internal oscillator (continuous Auto Sweep or single, timed sweep).

The device provides 75 named user presets, in solid-state, non-volatile memory (no battery). Preset, setup, and Preset Manager data can be saved via SysEx. Frequency response is rated at 10 Hz to 25 kHz (±3 dB). An optional footpedal is being developed. Sound Sculpture Musical Instrument Products; tel. (303) 442-1954; fax (303) 447-3502.

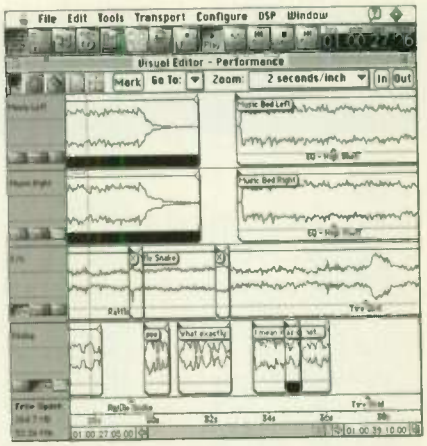
Circle #407 on Reader Service Card



▶ DIGITAL EXPRESSIONS SOFTSPLICE

SoftSplice ($2,395), from Digital Expressions, is a 4-track digital audio recorder/graphical editor that uses a Macintosh (68020-based or better) front end. The system consists of the Mac software and a SCSI-based hardware brain built around a 33 MHz, Motorola 56001 DSP chip. The hardware also includes both AES/EBU and S/PDIF stereo digital-audio connectors; no analog I/O is provided. SCSI hard-drive options include 240 MB (40 track-minutes at 48 kHz; $315), 520 MB (85 track-minutes; $755), and 1.05 GB (170 track-minutes; $1,155) drives. DAT backup is supported.

SoftSplice provides graphical, non-destructive, sample-accurate editing. The system supplies non-destructive track-bouncing, sample-accurate gain and pan control, and unlimited-length crossfades. It offers fully automated mixing and supports SMPTE chase-lock. The parametric EQ has unlimited scene changes. Other features include named markers, varispeed playback, sample-rate conversion, and scrubbing. Digital Expressions; tel. (800) 868-3434 or (206) 389-9895; fax (206) 643-3844.



Circle #408 on Reader Service Card

▶ LAKE BUTLER MSI-400 SERIES

Lake Butler Sound unveiled the Micro Mitigator MSI-400-series of MIDI-controlled, analog switchers. The first three members of the series are the MSI-408, MSI-414, and MSI-420 ($279 ea.). Each contains eight relays, connected to rear-panel ¼-inch jacks, which operate under microprocessor control.

The MSI-408 and MSI-414 primarily use the relays to provide analog switching functions, such as changing amp channels and controlling non-MIDI effects. They also can also be used as MIDI-controlled, audio mutes. In addition to their switching functions, each model includes four pedal inputs for a volume pedal or footswitch (momentary or click on/off). The pedals' status is converted into MIDI messages (Control Change 0-120, Velocity, Aftertouch, and three kinds of Pitch Bend—sharp, flat, or both), which can be sent on any MIDI channel.

The MSI-420 also has two stereo effects loops, and the MS-414 has one. The effects loops use the relays to provide a transparent Bypass feature for effects processors. Each stereo effects loop has separate control of the left and right channels, which can be used as two mono loops. The effects sends can be muted. The loops' status can be stored to any of the 128 user memory locations and can be recalled via MIDI Program Change. The loops also can be assigned to MIDI controllers and note numbers for real-time MIDI control.

The MSI-408 offers eight switching relays and the continuous-pedal interface but lacks effects loops. The MSI-420 provides two effects loops and the pedal interface but lacks the regular switching functions. The MSI-414 has four switch relays, one effects loop, and the pedal functions. With an optional bracket, up to three Micro Mitigators can be mounted in one rack-space. Lake Butler Sound; tel. and fax (407) 656-5515.

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▶ ROLAND P-55

Roland announced the P-55 Piano Module ($695), a half-rack-space sound module that includes sampled sounds from several previous Roland instruments. The 28-voice polyphonic, 3-part multitimbral unit contains 4 MB of waveforms, producing 32 factory-preset sounds, including grand piano, CP-70 electric grand, classic Rhodes, vibes, clavinet, harpsichord, and celeste. Velocity response, layer and split setups, and stretch and Baroque tuning can be programmed for each of sixteen user presets. A digital effects processor provides delay, chorusing, flanging, and reverb. Stack mode lets up to eight P-55 modules be combined for up to 224-voice polyphony. Roland Corporation US; tel. (213) 685-5141; fax (213) 722-0911.

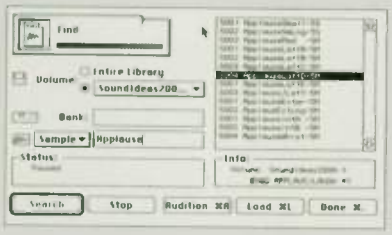
Circle #410 on Reader Service Card

REV UP ▲ ▲ ▲ ▲ ▲

OSC

OSC has upgraded *DECK II* to version 2.1 ($399; upgrades $50). The new version provides four tracks of 16-bit, 44.1 kHz, direct-to-disk recording and editing on the Macintosh Quadra 840AV and Centris 660AV. The program also works with Spectral Innovations' NuMedia card, which uses the same AT&T 3210 DSP chip and Apple Real Time Architecture (ARTA) as the AV series. OSC; tel. (800) 343-DECK or (415) 252-0460; fax (415) 252-0560.

Circle #411 on Reader Service Card



▲ E-MU SYSTEMS

E-mu Systems is shipping its *Emulator IIIx Remote Controller/Librarian* software ($295) for the Macintosh. The program provides complete computer control of all E-IIIx functions and adds sound-library catalog, search, and Bank-creation features. It can control up to seven E-IIIx units simultaneously through a front-panel emulation on the Mac screen. Pop-up menus allow quick access to functions and an onscreen virtual MIDI keyboard, pitch wheel, and mod wheel let you audition Presets without using an external controller. The librarian's Browser function allows you to view the contents of entire libraries, while a Find command locates Banks, Presets, or samples via keywords. An Audition From Disk feature allows previewing of samples directly from disk, without loading them into RAM. E-mu Systems; tel. (408) 438-1921; fax (408) 438-8612.

Circle #412 on Reader Service Card

MICRO TECHNOLOGY UNLIMITED

Micro Technology Unlimited released version 2.1 (upgrades $79) of *MicroEditor*, its *Windows*-based software for the MicroSound hard-disk audio-editing system. The new version offers multiple-mode video and SMPTE varispeed sync-lock, even when the code is striped on video or analog tape. It provides an 8/16-bit Multimedia Sound Device driver and can mix up to 50 Segments from 16-bit Wave files (sampled at between 8 and 48 kHz). The new version lets you play the mix while recording in sync; overdub; and punch in/out. Other features include automated effects sends and networking. Support for the new MicroSound 32 MHz DSP board cuts multitrack mix times in half. Micro Technology Unlimited; tel. (919) 870-0344; fax (919) 870-7163.

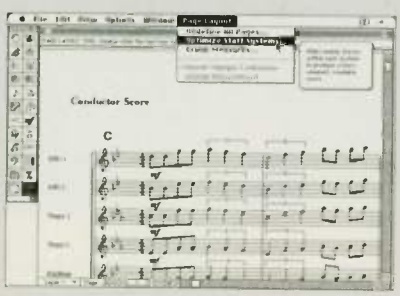
Circle #413 on Reader Service Card

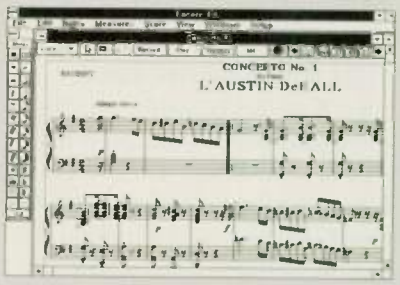
▼ CODA MUSIC TECHNOLOGY

Coda announced *Finale Allegro* for Macintosh ($349; upgrades $99), a major upgrade to and replacement for *Music Prose*. The mid-level notation software supports 32 staves and provides 24 tools from *Finale*. The upgrade adds full System 7 support, including TrueType fonts, Balloon Help, 32-bit addressing, and virtual memory. Tool palettes are movable and resizable.

Feature enhancements include built-in part-extraction and the ability to select partial measures, apply articulations (e.g., staccato marks) to groups of notes simultaneously, and rebeam and transpose multiple notes at a time. Articulations are positioned automatically on the notehead and move with it when the stem direction changes. HyperScribe real-time entry has been improved to more accurately capture your performance. Coda Music Technology; tel. (800) 843-2066 or (612) 937-9611; fax (612) 937-9760.

Circle #414 on Reader Service Card





▲ PASSPORT DESIGNS

Passport Designs announced *Encore* 3.0 for the PC and Mac ($595; upgrades $149). The major upgrade supports up to 64 staves, with up to eight voices per staff, and offers an enhanced user interface with new menus. A new expression palette lets users save their expressions for personalized markings in any font, size, or style.

The program graphically displays and plays back nested repeat structures, including variable barlines and multiple endings. A score can now contain multiple clefs, tempo changes, and key and time signatures. A percussion staff has been added. You can automatically notate guitar tablature for up to eight strings, in any tuning. There's cross-staff beaming and a new slur function. Dynamic marks, crescendi, and decrescendi affect MIDI playback, and MIDI Control Changes can be inserted in the score. Text and lyrics can be entered directly into the program. *Encore* now supports EPS files for export into desktop publishing programs. Passport Designs; tel. (415) 726-0280; fax (415) 726-2254.

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INNOVATION AT WORK.

PRO STANDARD DIGITAL AUDIO SERIES

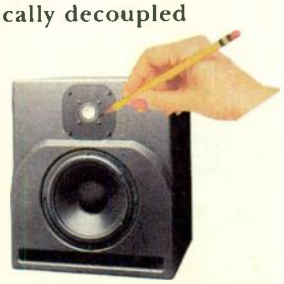
Three top studio-monitor designers sound off.

By George Petersen

SPEAK OUT!

Hundreds of companies throughout the world build loudspeakers, and each has its own distinctive philosophy of loudspeaker design. Even if we narrow the field down to businesses whose products cater to the recording-studio market, we're still left with dozens of companies and hundreds of models. A treatise on studio-monitor design could easily fill several textbooks, so I limited myself to interviewing speaker-system designers at three leading companies with divergent approaches to the art and science of studio-monitor design.

Among the participants in this forum was Bill Calma of Tannoy, a British company whose reputation was built on high-performance coaxial loudspeakers. A few years ago, Tannoy entered the low-priced studio market with its PBM-6.5/PBM-8 series, which uses discrete (separate tweeter and woofer) components. More recently, Tannoy debuted its Monitor Series, which features a Dual Concentric, coaxial driver, mounted within a space-frame skeleton (explained shortly) that supports acoustically decoupled wall panels.



John Meyer of Meyer Sound Laboratories pioneered the use of active electronics to control every aspect of loudspeaker performance, a concept that is now becoming accepted practice throughout the professional-sound industry. Perhaps best known for live-performance loudspeakers that are in use with dozens of top acts, John Meyer created a sensation in 1989 with the HD-1 studio monitor. This high-definition system packs 230 watts of biamplification, with full electronics control, into compact, 2-way speakers capable of reproducing bass down to 30 Hz.

I also spoke to Ted Telesky of JBL, a world leader in home, auto, sound-reinforcement, and studio speaker systems, and a leading supplier of raw speakers. While considered a conservative and

Illustration by Dennis Irwin

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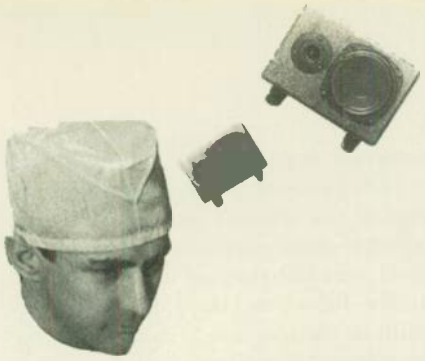


Speaker Design

traditional company, over the past few years, JBL has debuted a number of radical new designs, from the injection-molded cabinets of its Control Series to the "Buddha-belly" front-baffle shape of its 4200-series monitors (reviewed in the June 1992 **EM**). Not content to rest on its laurels, JBL recently unveiled the 4400 series, an upgraded version of its most popular professional studio-monitor line.

THE IDEAL

The topic of monitor design begins with basics, so we asked the participants to define some of the qualities that define an ideal loudspeaker system.

"Neutrality is important," says JBL's Telesky. "There are two different trends in engineering preferences: Speakers that overemphasize different frequency bands and speakers that try to impart very little to the mix. I believe the more neutral, the better. This includes response, phase information, and distortion levels. I've measured monitors that have a lot of second-harmonic distortion. They may sound like they have a lot of bass, but it's not real bass. A monitor should be as natural as possible and impart as little to the mix as possible."

Tannoy's Calma agrees, adding, "Aside from the basic things—good frequency response, good sensitivity, and good dynamic range—look for a well-balanced monitor. A good speaker system must have good articulation: A transparent monitor may not be the best speaker in a recreational listening environment, because it's meant to be revealing for mixing. Don't buy a studio monitor for consumer recreational listening; monitors should be purchased as a working studio tool. Forget about quick A/B/C-comparison listening tests; try out monitors in a mixing situation. Don't just listen to monitors with prerecorded music to see which sounds the most beautiful.

"And don't always take a frequency-response plot or specification at face value." Calma warns. "For instance, if a manufacturer claims to have a phase response of ±15 degrees, and the measurement was taken at one point in time, at one frequency, the spec doesn't tell you much about the history of the monitor; it just indicates a measurement at one point in time, at one frequency. Those types of measurements can be deceiving, so you need to use the best reference tool, which is an educated ear.

Distortion is another area of concern in defining the ideal monitor, according to John Meyer. "In order to achieve purity without artifacts, a speaker should have very low harmonic distortion or intermodulation distortion, so it's not producing new frequencies." (Intermodulation distortion occurs when input signals interact to form new output frequencies that are *not* harmonically related. IM distortion is more unpleasant to the ear than harmonic distortion.)

Bill Calma, Tannoy

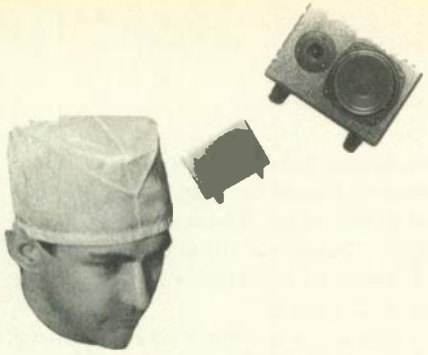
"Speakers can have properties that make them sound like they've been put through a comb filter," Meyer continues, "almost like adding little reverbs. And speakers can mask a signal in the same way you get reverberation when you put a speaker in a room [see sidebar "Reviewing the Basics: Near-Field Listening"]. Very inexpensive speakers may have high degrees of masking and could be difficult to understand speech on, even though they may have a good frequency response. A 20 Hz to 20 kHz (±3 dB) frequency-response spec may not tell you enough about a product, as it may have 100% distortion and still have a good frequency response.

"It's important to have a full-range monitor, so you can hear low-frequency hums and buzzes," Meyer explains, adding that "all the frequencies you put in should come out at the same time. This can be termed 'minimum phase distortion,' or 'time alignment,' or a 'good impulse response' [the ability to reproduce a burst of energy in the order it was received]. It will show

REVIEWING THE BASICS: NEAR-FIELD LISTENING

Room acoustics can play a major role in the way speakers sound to the listener. A room with little furniture and hard, reflective wall and floor surfaces emphasizes bass frequencies, while a heavily furnished, carpeted room tends to have the opposite effect. And a space that is acoustically reflective (sometimes referred to as a "live" room) bounces sound waves around, creating a reverberant environment where sounds become indistinct, and intelligibility is lost. When a studio monitor is placed in such a room, the listener may have trouble separating the sound of the source material from the changes added to the signal by the room acoustics.

One effective solution to the problem is listening to monitor speakers in the close, or near, field, meaning distances of approximately one meter away. As the ratio of direct to reflected sound is increased, the effects of poor room acoustics become less of a problem. When listening at close quarters, it is important that the combined image of the individual drivers in each studio monitor be coherent at short distances. Therefore, systems designed for such applications have closely spaced drivers, or employ a coaxial design in which a tweeter or other high-frequency unit is mounted in the center of a bass reproducer.



Speaker Design

up as the delicacy of a piano or the timbre of a harpsichord; a lot of instruments are affected by the transient response of a speaker."

COAXIAL VERSUS DISCRETE

The ability to avoid such phase errors is a major advantage of coaxial speaker designs, such as Tannoy's Dual Concentric speakers. Explaining the advantages of this approach, Bill Calma comments, "The primary principle in Dual Concentric technology is 3-dimensional time-alignment; in other words, phase coherency at all angles at all frequencies. We refer to our drive units as having coincident time-alignment, meaning that the two acoustic sources coincide at the same point in space behind the front baffle of the loudspeaker. The benefit is providing 3-dimensional time-alignment characteristics, which frees the listener from massive phase and time errors, so ear fatigue is reduced.

"The two things that are most fatiguing to the listener are phase error, which is unnatural to the ear, and high-level, second-harmonic distortion. Odd-order distortion [third, fifth, and seventh harmonics, etc.] is also fatiguing, but second-harmonic distortion is particularly fatiguing, and it can occur even when the specs show low amounts of first-order distortion."

Asked to describe the effect of phase errors, Calma says that they are perceived as a "smearing of the image placement and an unnatural timbre of natural sounds. For example, the note 'E' on a clarinet or violin has a certain personality. The acoustic signatures are created by waveforms that are a combination of the fundamental note with second, third, fourth, and fifth harmonics, and so on. Phase errors occur when these harmonics are not reproduced simultaneously; such time-arrival errors drastically alter the natural timbre of the instruments."

"It is important to note that coaxial designs are not the only valid approach to speaker design," adds Telesky. JBL, a company that builds monitors using discrete speaker components, studied both approaches carefully and decided each offers benefits and tradeoffs. Ultimately, the company decided to offer both types. Its UREI division specializes in coaxial studio monitors. "The choice of components in the coaxial side has always been limited," Telesky points out. "Certainly coaxials give you the ability to have a point source, but you have to deal with the problems of time-offset and the possibility of having components in front of your woofer."

According to Telesky, horn-based coaxials, such as UREI monitors, offer the designer more control over dispersion than coaxials which use the woofer flare as an extension of the high-frequency horn. "If you don't have a horn, you have to deal with the polar-pattern limitation of the particular coaxial driver. Power-handling can also be an issue with some smaller coaxial designs, depending on the amount of space you have for mounting your coaxial driver.

Tannoy's System 12 DMT is part of the Monitor series, which features Dual Concentric drivers.

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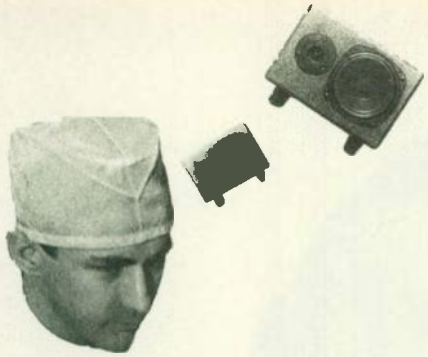


Speaker Design

The UREIs have a 1-inch [throat diameter] compression driver mounted on the back of a woofer with a 4-inch voice coil, which is a good combination for high SPL."

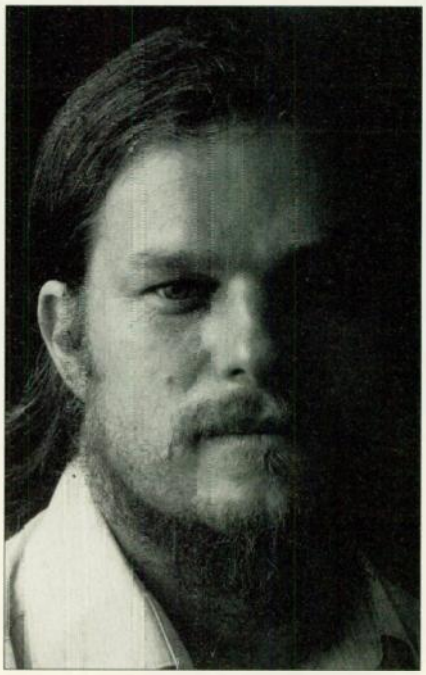
Another advantage of the discrete approach is the possibility of combining various types of components, such as a cone woofer, dome midrange, and horn tweeter. "There are many tonal characteristics available in separate components," Telesky continues. "Discrete components also allow you to try different placements of components, such as our 4410A, which has its drivers placed in a line array for a wide horizontal soundfield. And with multiple components, you can play with the dispersion and select components for dispersion control."

"There are inherent tradeoffs in considering whether to use a coaxial driver," Meyer observes. "A coaxial arrangement can provide fairly good phase characteristics, but at the expense of increased intermodulation distortion. Physically separate components yield lower distortion, but their phase response must be corrected. In the HD-1 monitor, we use an electronic correction technique to attain very flat phase and true point-source performance from a 2-way, discrete-component configuration."

ELECTRONICS

Crossovers are a major area of interest to monitor manufacturers, and the past decade brought major improvements in the loudspeaker/electronics interface. One of the earliest to integrate electronic control of loudspeakers was John Meyer. "My early research in the 1970s made it clear that we would not be able to get a speaker to move quickly enough to get a good transient response, no matter how much magnetic power we put behind it," Meyer recalls. "It's an inherent air/size delay problem: Even if you could make a woofer move absolutely instantaneously, the air wouldn't respond. It's analogous to punching a big bubble with your finger: It won't move, and we don't have enough size to move it correctly.

"Electronics became a way of compensating for this problem," Meyer explains. "Instead of trying to make the drivers do something they can't do, we looked at the frequencies and how they came out, and we rearranged the order. Even ten years ago, it was proven that there wasn't a passive network solution to this problem. With all the passive elements between the power amplifier and the speaker, you lose a lot of power, so it makes sense to move all these controls into the electronic domain, which frees up the designer. This path leads to more precise results."



John Meyer, Meyer Sound Laboratories

While JBL has been incorporating full electronic control into some of its large concert-speaker systems, the company uses passive crossover technology in its studio monitors (see sidebar "Reviewing the Basics: Crossover Design"). The approach is decidedly more cost-effective than requiring multiple amplifiers and sophisticated electronic control in the studio-monitor chain.

"Over the years, JBL has made constant improvements, but it has been evolutionary, rather than revolutionary," Telesky states. "When the 4311 came out, it had a fairly straightforward first-order (6 dB/octave) network, and it actually sounded good because it had little phase shift. In our attempts to become more neutral, we're using the network to better control the drivers, not only in the response slopes, but with conjugate networks built into the networks to fine-tune the response of the drivers.

"Drivers are usually measured on a flat, large baffle," Telesky continues. "If you put a perfectly flat driver on a speaker baffle, it will have a completely different curve because of the directivity changes caused by the baffle. So you need to design components for use

REVIEWING THE BASICS: CROSSOVER DESIGN

Crossovers (sometimes called frequency-dividing networks) take a full-range input signal and break it up into several signals, each containing a different frequency band (e.g., lows, mids, and highs). These signals are then routed to the appropriate loudspeakers within the system.

Crossovers fall into two basic types: Passive crossovers are usually built inside speaker cabinets, where they divide an amplifier's output signal and route the parts to different speakers. Active crossovers divide a line-level output signal from a mixer or other sound source and route the resulting signals to individual amplifiers that drive different speaker components.

The simplest crossover in a 2-way system is a passive circuit consisting of a simple highpass filter. The filter is really no more than a capacitor, in series with the tweeter, that prevents the low-frequency energy from blowing out the tweeter.

Similarly, an inductor (coil) in series with the woofer acts as a lowpass filter. These two filters combine to form a first-order filter with a gentle slope that provides 6 dB of attenuation per octave. By combining more inductors or capacitors, more complex crossovers can be created to provide slopes of -12 dB, -18 dB, or -24 dB/octave.

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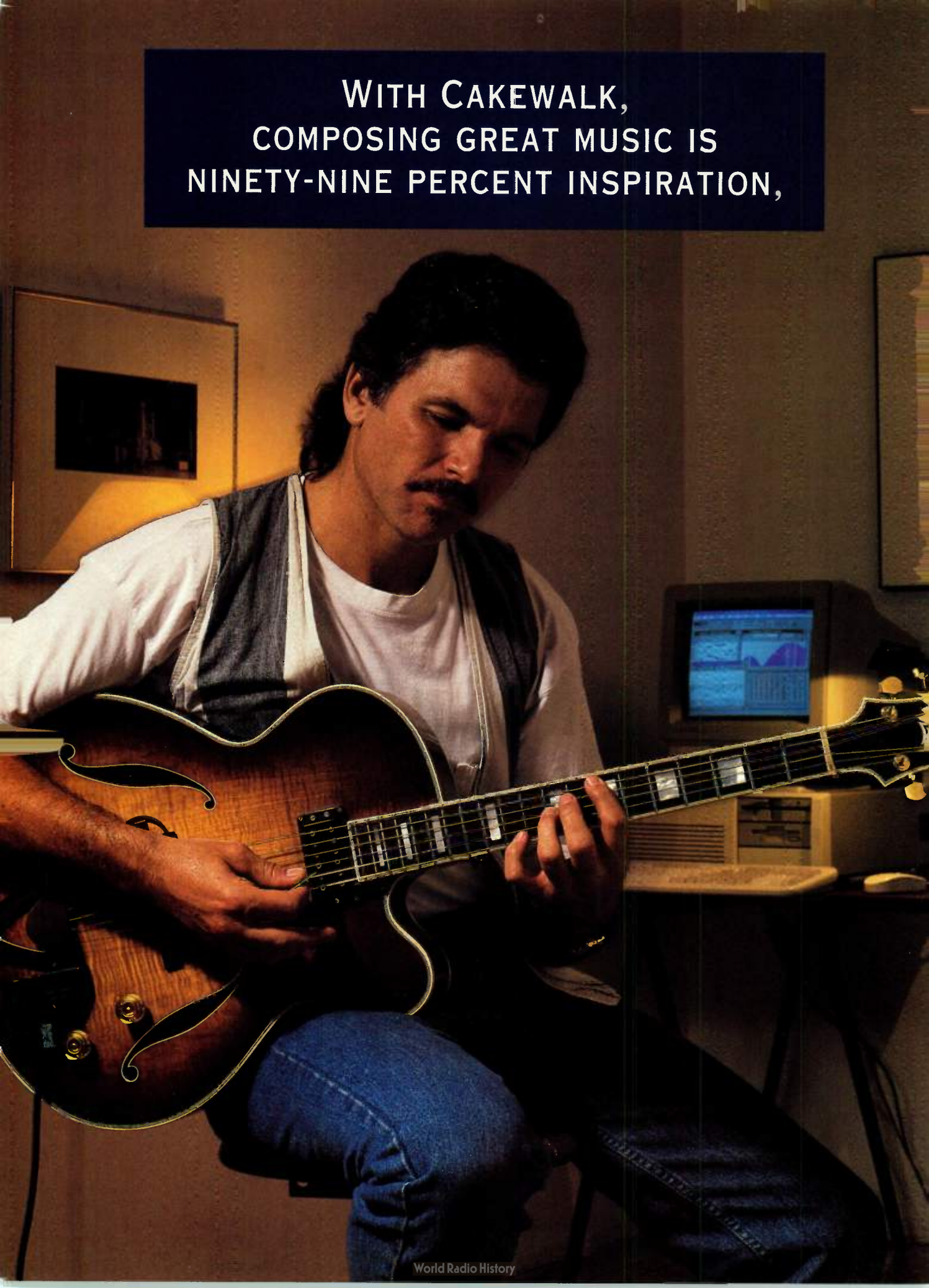

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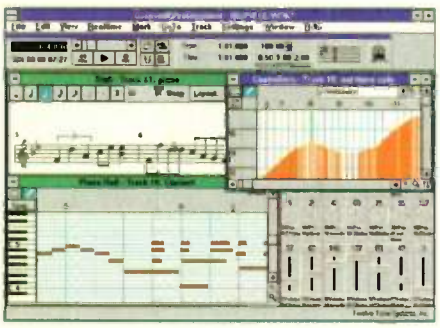

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TEC 1993 NOMINEE

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Speaker Design

on a speaker baffle and then fine-tune them with a network. Our networks are getting more complex; we're using computer modeling to design networks to augment the drivers."

"Tannoy's general approach to crossover design is a minimalist topology with the fewest components in the signal path," says Calma. "We engineer the drive units to be well-behaved to begin with, and then we modify them to work together within acceptable parameters without adding a lot of extra components. We also use hard-wired configurations; we don't run high-current signals [i.e., speaker feeds] through thin film surfaces, such as circuit boards."

According to Calma, Tannoy recently completed an 18-month study on crossover components and layouts. "We studied the effects of the large magnetic fields and the acoustic environment that components are subject to within the enclosure. For example, it is critical to keep certain components away from other components. We also looked at the crossover components themselves. Inductors can ring, so they're now injected with resin, similar to the way that toroidal transformers in high-end amplifiers are potted. And capacitors are generally built for electronic—not passive analog—applications, so we've begun manufacturing our own capacitors. A single-strand wire floating in a loudspeaker cabinet is a hostile environment. If the wires that run from the crossovers to the high-frequency drivers are allowed to vibrate in the magnetic field, they can induce signal into the cable. It's an audible distortion."

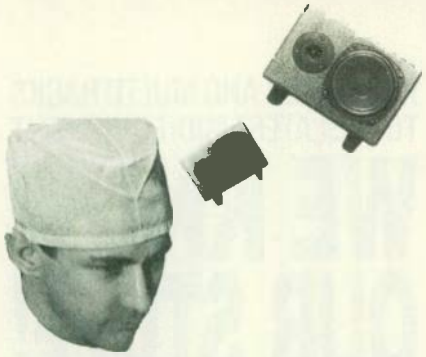
CABINET DESIGN

Probably the most active area of interest in speaker manufacturing involves cabinet design. (For an overview, see sidebar "Reviewing the Basics: Enclosure Types.") Once limited to a few variations of a simple box with a couple of holes for mounting the drivers, enclosure design is now an area of intense interest and research.

JBL's Telesky explained some of the thinking behind the company's successful 4200 series, which uses a molded front baffle to align the voice coils of the tweeter and woofer so frequencies from the two reach the listener simultaneously. "With our 4206/4208 series, we wanted to fix some problems that could be accomplished in the physical domain," Telesky notes. "In a near-field speaker, where the listener is close to the monitor, time arrivals are very important. So the 4200s were designed to optimize that and still use discrete components. Through a tooling medium, we managed to control costs, so we could offer advanced technology at an affordable price. You can do things in tooling that would otherwise be expensive to do."

In addition to offering time-domain manipulation, the molded baffle approach has other advantages, according to Telesky. "The blending—the coherency of getting both drivers to work together—is important. The shape of the 4200 baffle allowed us to do some control and blending of the wavefronts of both drivers.

Meyer Sound Labs' 2-way, HD-1 studio monitor is remarkable for its custom-designed internal power amps and groundbreaking electronic-correction technology.



Speaker Design

It's still a wide-dispersion device; we try to control the vertical more, and the molded baffle does a good job of reducing diffraction effects."

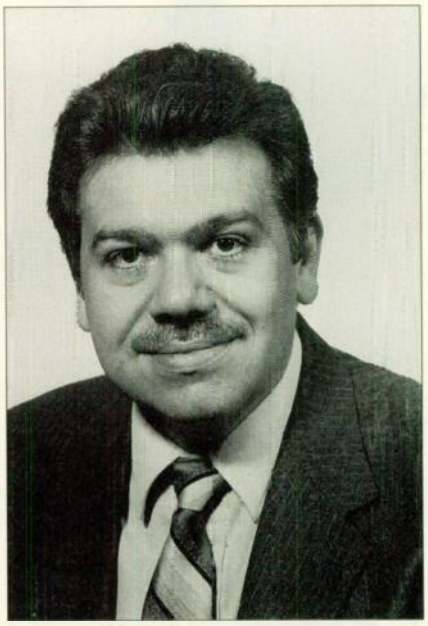
While the enclosures of its PBM-6.5 and PBM-8 monitors are quite conventional, Tannoy took a more advanced approach in its high-end Monitor Series speakers. "Part of the Differential Materials Technology used in our Monitor Series is what we call 'space-frame' construction," Calma notes. "The space-frame construction acts as a lossy coupling, an energy-absorbing panel. It's similar to the floating "room-in-a-room" principle that studios use, but applied to loudspeaker construction. We use Medite front and rear baffles, four corner posts joining the four side panels together, and in each one of those, the carcass is fitted and is rubber-lined around the space-frame construction. A cross-brace matrix runs horizontally and vertically and supports the drive unit, and the entire matrix is suspended on rubber boundaries, so the wood side panels never touches the carcass itself.

"The carcass utilizes a high-pressure laminate on both the inside and outside panel surfaces, ensuring the panels do not flex when locked into the space frame, even with low frequencies at high sound-pressure levels. At higher frequencies, it acts as a lossy construction that breaks resonant frequencies into different areas of the cabinet, dampening them at every edge with a rubber boundary and lessening the effective result of the transmitted frequencies, so they don't accumulate, as they would in a rigid cabinet."

THE CRYSTAL BALL

After nearly 100 years, speaker systems are relatively unchanged: Air is moved by cones or diaphragms that are controlled with magnets and coils. Why haven't we seen more radical approaches, such as compressed-air modulation?

"You can play with technology, but you have to make sure it's robust," warns Telesky. "At JBL, we sell tools to people earning their living from our products, so reliability needs to be very high. We haven't gone into esoteric areas such as flame-modulating loudspeakers. We're involved in a fair bit of aerospace research on materials; but for cones, which need to be stiff, light, and well-damped, the combination of paper and felt is still one of the best choices. There are lighter and stiffer materials, but they tend to ring, because they are not damped enough. At the same time, highly damped materials can't move fast enough without buckling and bending. There are some materials on the horizon that offer hope, but some of these tend to reduce sensitivity by adding mass to the cone. We need to balance sensitivity, stiffness, and damping. Paper still gives us the best combination of the three."



Ted Telesky, JBL

So gazing into the crystal ball, what lies ahead five, ten, or twenty years from now? "The integration of electronics is something we've been looking at to solve minor problems in loudspeakers," Telesky answers. "Crossing over loudspeakers is always an issue, one that DSP can solve easily. The ideal is to stop a driver at a brickwall and turn on another driver, while making it seamless. Doing that without altering the phase curve is difficult to do in the analog domain; it's much simpler using DSP. We've also been testing some ways of reducing and removing distortion in the digital domain. The place to start is in the transducer, but it has to be

REVIEWING THE BASICS: ENCLOSURE TYPES

Speaker systems for studio applications fall into two basic categories: Those with sealed enclosures and those with vented cabinets. Acoustic-suspension enclosure (sealed) designs mount the speaker components on the outside of an air-tight cabinet, which prevents the escape of the air displaced by the woofer's rearward motion. This increased air pressure within the cabinet acts as a springboard to push the woofer's cone forward.

Vented systems—sometimes called bass-reflex systems—have one or more vents or ports that create a small amount of phase cancellation at the woofer's resonant frequency, while increasing bass response at a lower frequency. The resonant frequency is the point where a speaker's mechanical efficiency is at its highest level, as determined by the size, mass, and stiffness of the speaker cone.

Ports can be mounted on the front or rear of the speaker enclosure. When rear-vented monitors are used near walls or other sound-reflecting surfaces, the sound waves exiting the port are redirected elsewhere in the room. This occasionally creates an unpredictable situation where the direct sound from the woofers combines at the listener's ears with the reflected signal arriving a few milliseconds later. This usually results in emphasized bass frequencies, although the effect can be a loss of bass response if the reflected signal is out of phase with the direct sound. Obviously, some experimentation is required in making placement decisions when rear-ported studio monitors are used near reflective surfaces.

JBL's 4200 series features a molded, "Buddha-belly" front-baffle shape that reduces diffraction effects and aligns the voice coils of the tweeter and woofer so frequencies from the two reach the listener simultaneously.

both sides: You have to do as much as you can on the mechanical side, and then do what you need to do on the electronics side to improve things. I'm not an advocate of using bad components and trying to fix them with electronics."

"We'll probably get involved in powered monitoring in the not-too-distant future," says Calma, who also notes that Tannoy is involved in DSP research. "Quality A/D and D/A converters are expensive now, but with digital processing, you can push existing parameters, making systems do things they currently cannot do. But whether we get involved in active analog electronics, or DSP control, the object is to achieve optimum, linear phase and amplitude, while keeping the loudspeakers musical."

In terms of long-range forecasting, "We have dreams of going to an ion-type [charged air-particle] speaker," says Meyer, "but that's going to be extremely problematic. What's more likely to occur in the near future is learning more about materials to increase power-handling, such as going to hard materials like non-resonating domes. There are some wonderful materials for building speaker domes, such as sapphire crystals, but they happen to be expensive, and pretty soon you have a very nice $3,000 tweeter.

"The next level is to look for new materials that we can push more power through. If this were the military, we'd have it tomorrow, but in audio it's hard to generate the money for research."

In addition to contributing articles to **EM**, **George Petersen** *is the products editor for* Mix *magazine.*

By the EM Staff

Home

The idea of a personal studio is simple. You buy a few pieces of gear, hook everything together, and start making music. In practice, however, the scenario is far from seamless. Equipment limitations, sonic gremlins, and other electronic annoyances are the constant bane of recordists. And technical distractions can demolish creativity faster than writer's block.

Getting the most from your musical playground requires freeing yourself from any impediments to music-making. For example, a common creativity obstruction is "just-one-more-itis." This is a wrenching, nearly universal condition characterized by outbursts, such as "If I just had one more _____ [fill in the blank with your choice of a hardware or software toy], then I could *really* do what I want."

Twenty productive personal studio tips, from the essential to the esoteric.

Of course, there are situations when purchasing a new piece of gear is the only solution for a technical or creative problem. However, you might be surprised at what you can do with what you already have.

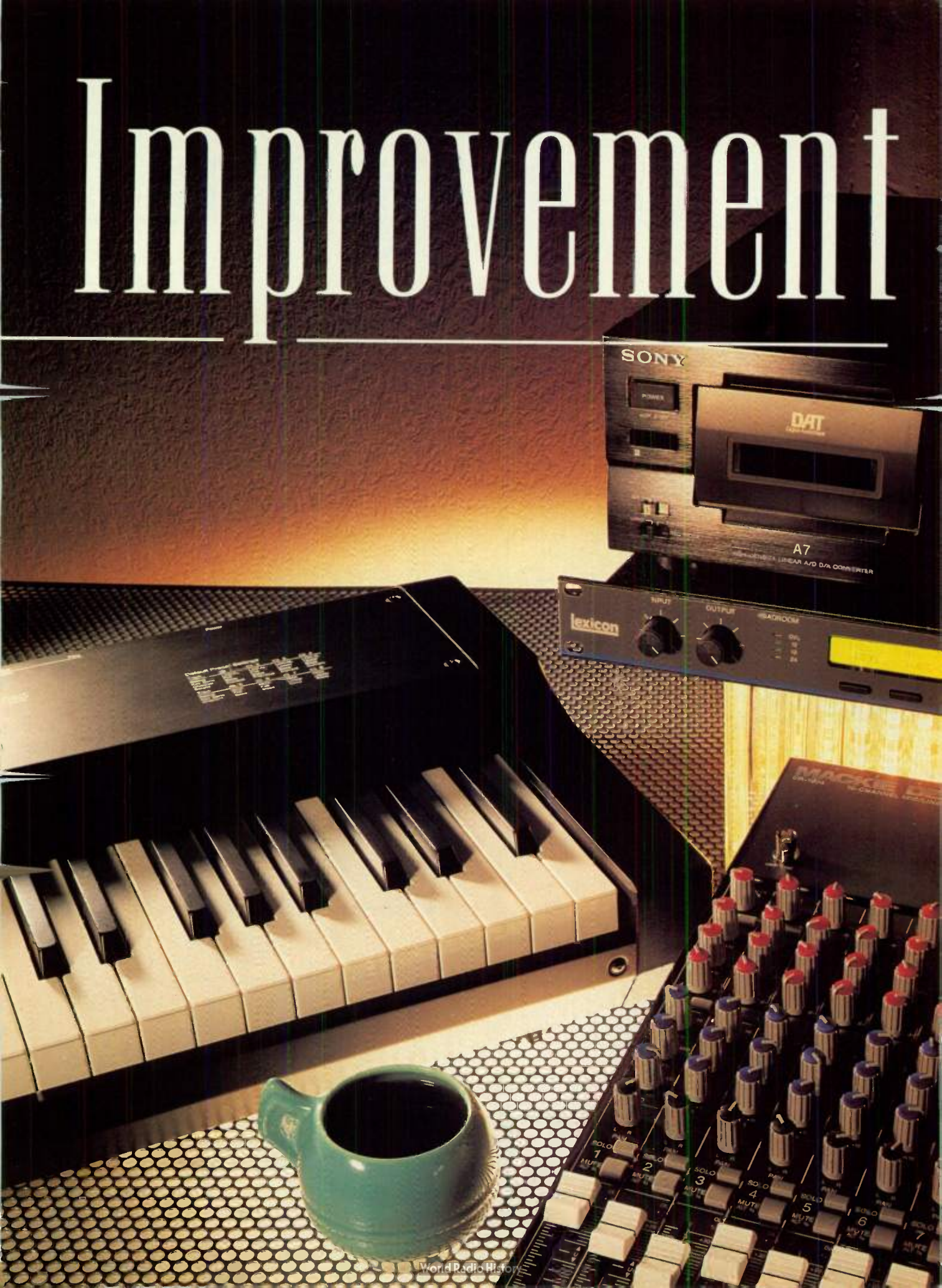
To help you unlock the possibilities within your studio, we've put together twenty of our favorite suggestions, tips, and tricks for overcoming common problems. Some of them may require you to part with convention, or look at your equipment in new ways. But if you can keep in mind that "rules are made to be broken," you'll discover some cool work-arounds. Enjoy.

K.I.S.S. IT GOODBYE

I can't tell you how many horror stories we've heard from very technical people who've wasted hours and days troubleshooting their medium-to-large MIDI systems. Figuring out why the fifth channel of your 8-part multitimbral synth module connected

Photograph by Stephen Tang

Improvement



Announcing a much

Encore's new user-defineable expressions palette lets you give your music the exact coloring you desire.

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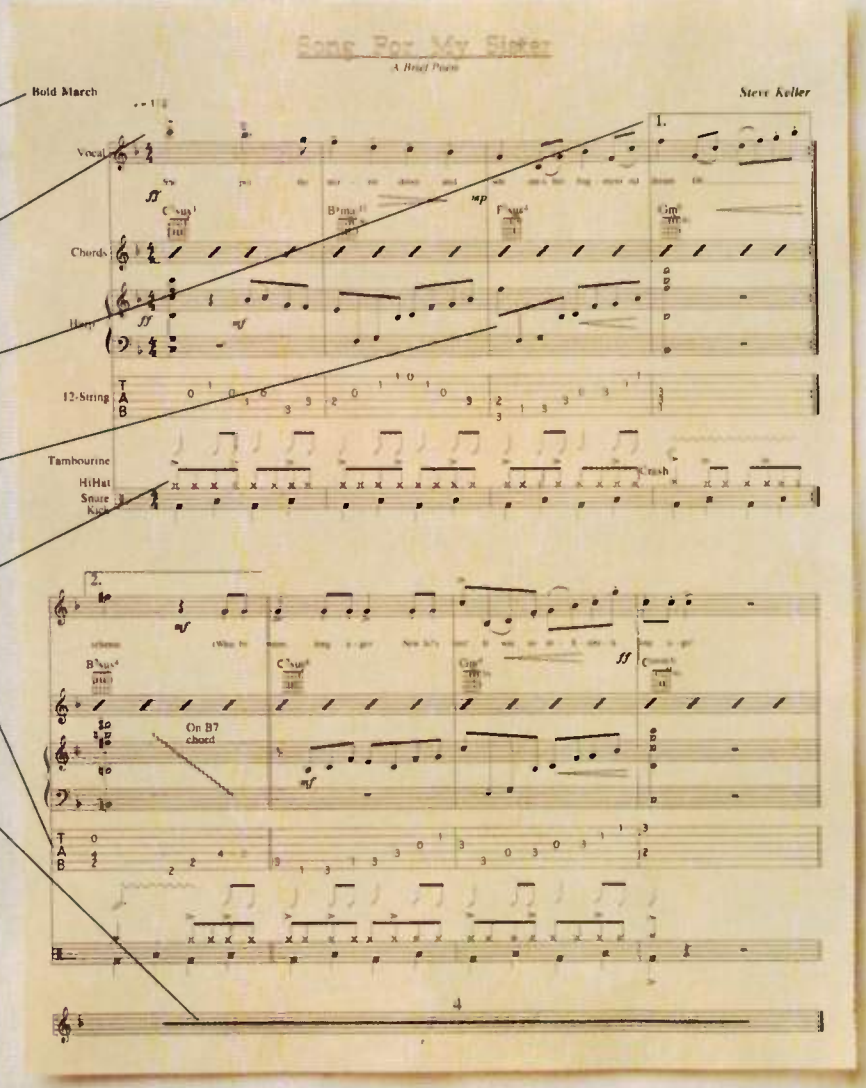

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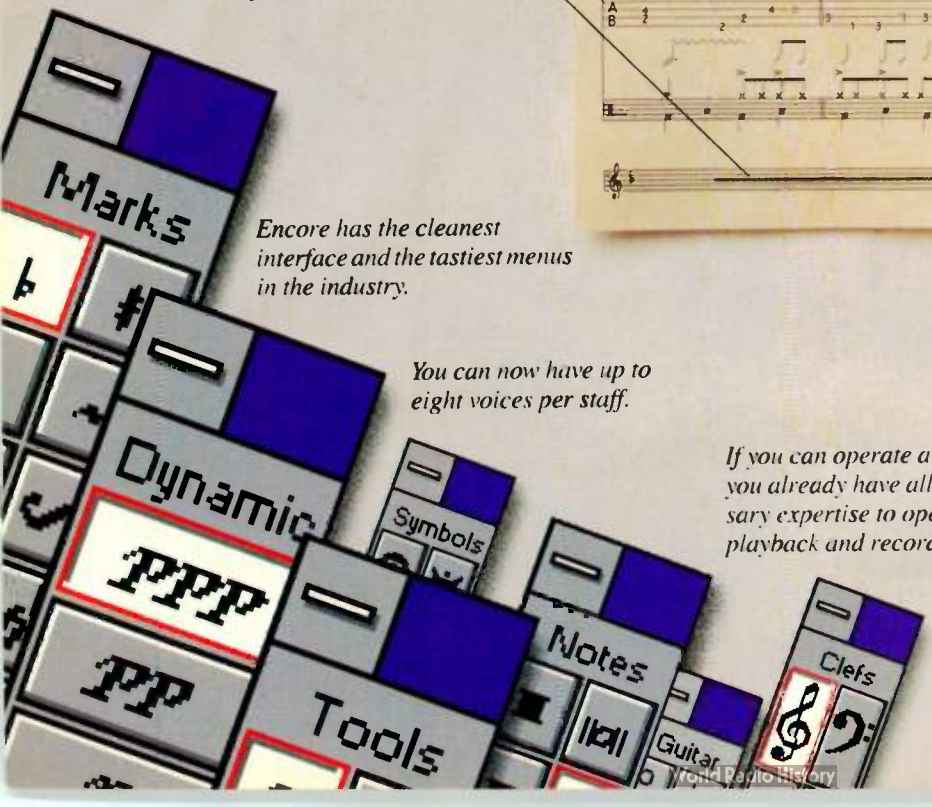

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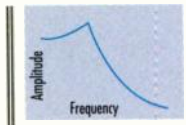

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Home Improvement

to the fourth output of your multiport MIDI interface isn't responding to the MIDI Volume messages from the MIDI fader box sending on channel 3 into the second input of the same interface is not a good way to spend a Saturday afternoon.

You can circumvent the whole problem by creating a simple songwriting/idea subsystem within your large setup. Create a patch within your MIDI patch bay or interface that allows a single connection between one multitimbral keyboard—GM-compatible, if possible—and your sequencer. (Or use the simple hardware sequencer on a keyboard workstation.) Save this as your working setup.

Stick with this limited system until you develop rough ideas, and then—and only then—switch to a fully loaded patch-bay setup that lets you tap into your complete arsenal. Not only will this save you headaches, it could help your music by forcing you to think about the song first.

NEIGHBORLY VOCALS

For most of us, it's incredibly difficult to record rock vocals in an apartment studio. This hardship has nothing to do with technical limitations, either. The problem is in our heads. It's somewhat intimidating to cut your voice loose when you know the neighbors can hear every bad lyric and off-pitch note. And if you really wail—or find it necessary to punch-in the same line 700 times—your next public notice may be an eviction warning.

Simulating passionate vocals at a very quiet volume can alleviate "Apartment Vocal Anxiety." Soft dynamics and rock vocals may sound incongruous, but bear with us. Set up a dynamic microphone outfitted with a foam windscreen, *and* place a mesh pop screen about one inch in front of the mic. Route the mic input through a compressor set to a -10 dB threshold and a 10:1 ratio.

Now, put your mouth about two inches from the mesh pop screen and sing softly into the mic. Produce screams and other vocal tricks by using phrasing and voice timbre adjustments, rather than volume. The heavy compression should evoke a tough, "in-your-face" sound at any volume, and a slight midrange boost from the console EQ will toughen the tone even more. Many heavy metal singers actually cut album tracks at low volumes to save their throats. (A producer once observed that AC/DC's gruff vocalist practically *whispers* his trademark shrieks.)

If you're not used to these wimpy vocal techniques, the whole exercise will sound pretty strange at first. But once you master the art of soft singing, you'll be able to cut blistering vocals in a tiny apartment without fear of ridicule or eviction.

SPACIOUS STRINGS

To add depth to sampled or synthesized strings, layer two slightly different string patches, perhaps from different synths. Set the amplitude envelope for one patch to a fast attack and short release, and set the other patch for a slightly slower attack and longer release. Then add different amounts of reverb, or even different reverbs, to the two patches, with more reverb on the faster patch. Finally, pan the two sounds slightly apart.

The resulting variations in time and frequency between the parts will give you a simulated front-to-back space, while the panning delivers the traditional left/right image.

If you have access to several reverbs and delay lines, and you really want to get wild, split the two original signals so you have two versions of each. Put delay lines on one version of each signal, and set short delay times that are slightly different for each. Then send the results to as many reverbs as you can spare (up to four). Return all four signals to your mixer, and pan them slightly apart. Finally, experiment with the delay, reverb, and pan settings to get the best effect.

SHARE AND SHARE ALIKE

If you record in your studio and play live gigs, you can economize on gear by selecting equipment to serve double duty. For example, dynamic mics can be used on gigs more easily than condensers, because they are generally less delicate and require no phantom power. Install sound modules, signal processors, etc., in manageable road racks (four rack spaces if you're a solo act, six to eight rack spaces if you have grunt help). If you normally take only certain things out on a gig, rack them together with short internal audio and/or MIDI cables and a separate power strip; minimize external connections to and from the road rack.

Many compact mixers, such as the Mackie 1604, work quite well in the studio and on the road (see "Command Centers" in the August 1992 **EM**). Most of these mixers can be rack-mounted for increased portability, as well; a slanted mixer rack on the top of the case lets you install other gear, such as your power amp and signal processors in addition to the mixer. Of course, you still need P.A. speakers and possibly stage monitors, which are decidedly different from studio monitors, but that's the price you pay for fame and fortune on the road.

NOISY MASTERS

Cassette ministudios are great musical

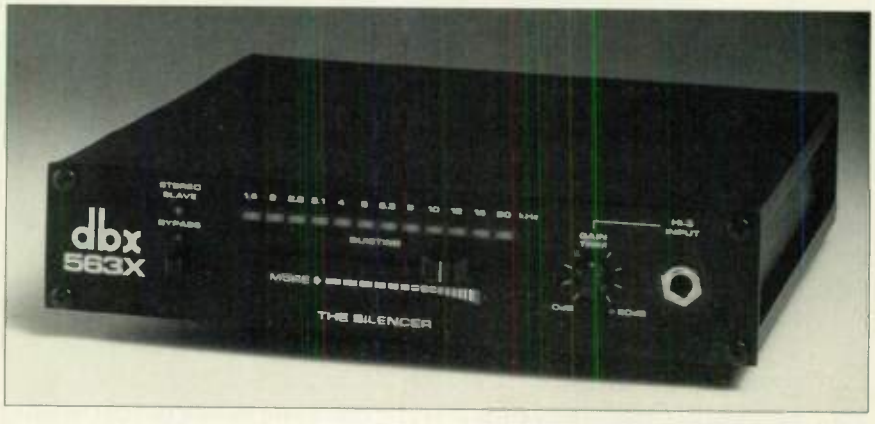
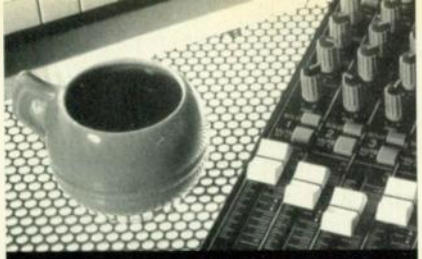


FIG. 1: If noisy tapes are a problem, you can sanitize your signal by inserting a single-ended noise-reduction unit, such as the dbx 563x Silencer, into the picture.



Home Improvement

notebooks, but the "hiss factor" of cassette tape often makes playing tracks for others an uncomfortable experience. A single-ended noise-reduction processor can help you avoid the discomfort.

Devices such as Rocktron's Hush and the dbx 563x Silencer are frequency-dependent tonal processors that subtly filter out any high-frequencies not present in an audio signal (see **Fig. 1**). The attenuation ebbs and flows with the frequency fluctuations of the material being processed. Because audible hiss exists beyond the frequency range of most sounds, single-ended noise reduction can greatly diminish the noise zone of your master tapes. (A drastic EQ cut at 10 kHz can also diminish hiss, but such "hard limit" processing might dull the sonic quality of your tape.)

To clean your audio house, simply insert a dual or stereo model into your master stereo bus. This routing ensures that the entire mix is "hiss sanitized" before the stereo signal reaches the mixdown deck. Then, adjust the amount of processing to diminish hiss, while still allowing brilliant sounds (such as cymbals and synth bells) to shine through.

SEQUENCERS AS LIBRARIANS

If you've always admired the patch-storage convenience and cost savings offered by universal librarian software, but haven't had the inclination—or scratch—to pop for one, you'll be pleased to know you don't have to. Lurking within the inner recesses of most software sequencers is a librarian that lets you store and transmit single patches or banks of patches to almost any of your connected synths, either individually or collectively. You won't find it in a hidden menu; it's no further away than the Record button. As long as your sequencer can record MIDI SysEx, you can use it to store patch banks from your synths by recording a bulk dump or individual patch dump initiated from the synth's front panel as a stand-alone sequence.

You won't have the renaming and bank-arranging convenience offered by a true librarian, and a few older MIDI synths require tedious handshaking messages before they spill their guts. But as long as you use sensible naming conventions for the resulting sequencer files, you'll get a lot of the functionality for none of the cost. If you save the sequence files in Standard MIDI File (SMF) format, you can use them as universal message carriers to send patches to other musicians, regardless of the computer platform or software they own. All they have to do is load the file into their sequencer, set the synth to receive SysEx (if necessary), and hit Play.

GATED REVERB

Gated reverb has been a popular effect for drum parts. Rather than relying on the preset gated-reverb programs in your effects processor, try creating your own with a real noise gate.

Using a patch bay or audio splitter, split the unprocessed drum signal *before* it goes to the reverb. Route one part to the Key input on the noise gate; it will act as a trigger for the gate (which is why you use a dry signal). The other signal goes to the reverb, then through the gate's audio chain. The gate can now control the reverb decay time.

If your gate lacks a Key input you can still gate the reverb, albeit with less control over the gate time. Simply send the wet source straight through the gate's audio path and use the threshold setting to trigger the gate.

MULTIFACETED DAT

Although it fizzled in the consumer market due to legal delays and recording-industry pressure, DAT has become the mastering medium of choice in studios of all sizes. However, DAT decks have many applications in addition to mastering (see **Fig. 2**). For example, portable DATs are indispensible for gathering samples in virtually any location. These recordings can then be digitally transferred to your sampler or hard-disk recorder with no generational loss of fidelity. If the sounds are destined to end up on a CD, use 44.1 kHz to avoid sample-rate conversion.

Another important application is data backup. There are data DAT machines designed especially for this task, but they are relatively expensive. If you use any of Digidesign's workstation products, such as Sound Tools, Pro Tools, or Session 8, you can use your audio DAT to back up digital audio and session files with their *DATa* utility software. You can also use your DAT machine's A-to-D and D-to-A convertors to shuffle digital signals into and out of various pieces of gear in your studio. In fact, several hard-disk recording systems require the use of a DAT to perform this function. Make sure your DAT can stay in Record Pause mode for any length of time because some machines automatically switch out of

FIG. 2: A portable DAT, such as Sony's TCD-D10 Pro II, can be used for more than just recording stereo masters. You can field record samples and live gigs, do computer data backup, and use it as a stand-alone A/D or D/A converter.

this mode after a certain period, which then cuts off the A-to-D conversion of the incoming signal.

PROCESSING SHORTAGES

The Fates that control final mixdowns are mischievous and mean-spirited. This can be the only explanation why you always need one more signal processor than you have available. Luckily, there's a way to foil destiny. If you plan ahead, you can record some tracks *with* effects and save your critical signal processing for the mixdown.

For example, if a guitar lick begs to be chorused, why wait until the mix? Find a complementary chorus patch on your effects box, and mix in the effect with the source sound during the actual performance. Percussion, background vocals, instrumental sweetening, brass, and rhythm parts can all benefit from predetermined (and premixed) signal processing. Of course, once you record the effect/source sound balance, you're committed. You can't change either the effect or the effect level, unless you re-record the part. (If you're the wishy-washy type, this application may cause you nothing but trouble.)

Premixing effects can also add impact to your submixes. If track limitations require you to submix individual drum tracks to a stereo pair, try adding a little reverb to the snare and toms during the bounce. This produces a more three-dimensional stereo image and saves submixed percussion from sounding dry and flat.

GROUNDING

One of my synths kept dumping its programs, and several other high-tech devices in my studio were acting flaky for no apparent reason. I thought I faced a massive repair bill until a friend advised me to check the grounding in my apartment's AC outlets.

A Radio Shack 3-pin outlet tester revealed that my 3-conductor outlets weren't grounded; the third pin wasn't even connected. So I had an electrician rewire my outlets. While he was at it, he put my studio on a separate breaker so my AC wouldn't get trashed by household appliances. Establishing a solid ground cleared up some, but not all, of the problems. Lesson 1: Check your outlets, and if they aren't faultlessly grounded, call a professional electrician. It's cheaper than fried

Home Improvement

gear (or even burnt flesh, if you become the path to ground).

FLUCTUATING POWER

Even with the outlets grounded, an ancient parametric EQ and one elderly synth were still misbehaving, and Old Reliable, my faithful power amp, just wasn't pumping out the juice I expected. The AC power is supposed to be clean in my area, but brown-outs and other local and regional power conditions (not to mention home appliances) can produce voltage sags, spikes, and surges that can wreak havoc with microprocessors and compress the output of power amps.

The answer was to add an AC line regulator (see **Fig. 3**). This device smooths voltage surges and spikes, and filters out electromagnetic and radio-frequency interference, but that's just the start. Line regulators track the incoming voltage and raise or cut it to supply a consistent, safe 110 to 120 VAC. As I write this, the display on my line regulator reveals that it is receiving 123 VAC, but my computer will never see a millivolt more than 120 volts. The EQ's finicky power supply no longer balks, the synth is stable, and Old Reliable's reputation has been restored. I later added a second line regulator for insurance on my most critical rack.

Final confirmation came onstage. A RAM-based sampler was my only instrument for the gig; one bad power drop, and it might empty its RAM, murdering the set. The club had an ice machine on the same circuit as the stage outlets, and the line regulator's display showed a puny 103 VAC input voltage with the ice machine off. With the machine on, the voltage dropped to a perilous 90 volts, then surged up again as the ice demon returned to its "off" cycle. The sampler never saw anything but clean, steady juice. Since then, I never play electronic instruments on a gig without bringing a line regulator.

There's one more step you can take: an uninterruptible power supply (UPS). A UPS contains a storage battery that takes over if the AC power fails. The battery charge won't last long, but it gives you enough time to quickly save your work (say, a computer file, sample, or synth patch) and shut down your gear.

THE SLOW FADE

Probably the most overlooked and underappreciated piece of gear in today's MIDI/digital-audio world is a dedicated fader box. Products like JLCooper's FaderMaster, Peavey's PC-1600, and Lexicon's MRC let you do everything from automated mixdown to editing synths and signal-processors in real time, yet most musicians aren't motivated enough to buy a dedicated device.

Synthesizer owners can get some of the functionality of these devices by using their instrument's volume and data-entry sliders to send a stream of continuous controller messages; these can be remapped within their sequencer to the controller message of choice. For this to work, you have to make sure both sliders generate MIDI messages that can be sent out the instrument's MIDI Out port, and your sequencer can remap incoming messages in real time, or its faders can be externally controlled from a controller number of choice.

Two faders aren't enough for a sophisticated automixdown—though many controller keyboards and certain synths give you access to four or more sliders—but if you concentrate on a few tracks at a time and make multiple passes, they can be surprisingly useful.

ACOUSTIC ISOLATION

Many home studios are set up in the garage or an extra (or not so extra) bedroom without a separate control room and recording booth. Nevertheless, when you record vocals or acoustic instruments, it's important to isolate the engineer from the artist. If they are both in the same room, the engineer must wear headphones to prevent the monitor signal from bleeding into the microphone. Even wearing closed headphones, however, the engineer will probably still hear the artist's performance in the room, which can alter perception of tone and audio anomalies.

Run a long, well-shielded mic cable and headphone cable (for the artist's monitor signal) from the mixer to a nearby bathroom or bedroom. Not only does this solve the control-room monitor problem, it also prevents sound contamination from computers, fans, and other equipment. Experiment with the acoustic properties of different rooms; you can also try different acoustic treatments such as wall foam, bass traps, and gobos to affect the sound of the artist on tape.

ALTERNATE INPUTS

Your vast collection of MIDI sound modules allows you to blend almost limitless timbres. That's good. However, the number of inputs on your mixer is overwhelmed by the number of outputs from your sound modules. That's bad. Especially when your latest electronic opus employs every synth in your collection.

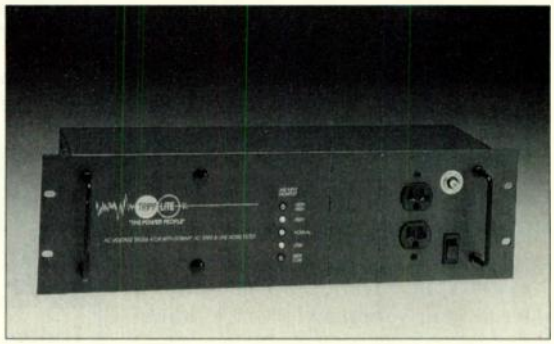


FIG. 3: Having problems with flaky gear? Perhaps the problem stems from inconsistent AC power levels. Try using an AC line conditioner, such as Tripp Lite's LCR-2400, which provides a constant 120 volts to all connected gear.

But before logistics usurp your tonal creativity, remember that aux and group returns can be used as additional inputs (see **Fig. 4**). In some cases, depending on the mixer construction, a split monitor section can be wired to allow line-level inputs for each monitor pot.

Although none of these alternative inputs have the processing power of a full-channel input module, they *do* have level and pan controls. Other parameters can usually be worked around. For example, sometimes you can program your synth patches to achieve desired tonal ranges without console EQ. Also, take advantage of any internal DSP to process your sounds, so you can free up effects returns to use as inputs. In this instance, pre-planning is essential because if all

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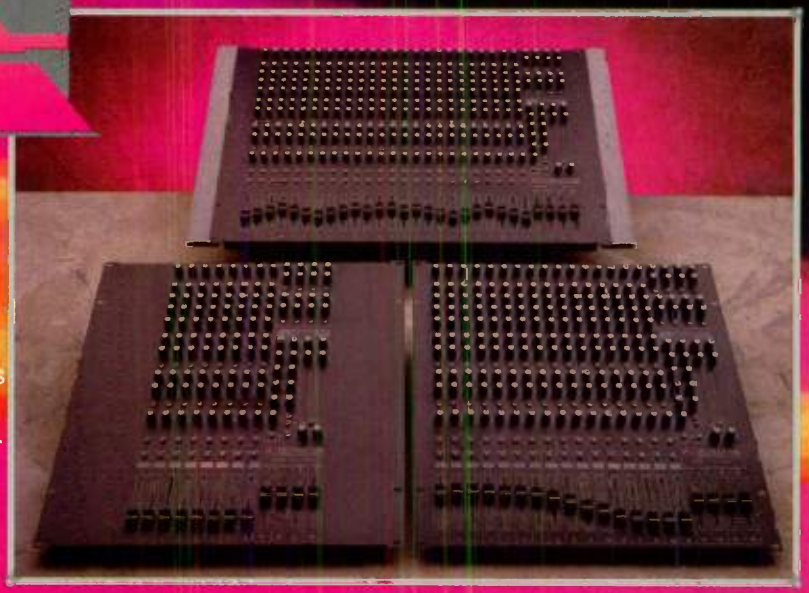

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Every once in a while a product comes to market that offers a brilliancy in design that seems beyond human engineering. The Phantom Series consoles offer the performance and features of mixing boards costing thousands of dollars more. They are rugged enough to take the pounding of steady live use. They are also so transparent and utterly free of noise that they are the first choice for precision multitrack recording! From a four track home studio to 32 channel digital, the Phantom consoles offer a level of performance that is inspiring. A.R.T. has taken the fidelity of world class recording mixers and made the technology available in a professional console that is as silent as its' name implies. Production unit will vary slightly from photo.

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Home Improvement

the aux returns are committed as instrument inputs, the sound modules must bear responsibility for effects such as reverb, delay, and modulation.

FLANGED AND CHORUSED SAMPLES

Sampling flanged and chorused sounds can be problematic because the sounds tend to take up lots of memory. On the other hand, sometimes you won't have extra signal processors available at mixdown, and you'll want the effect to be part of the sampled instrument.

Instead of sampling the processed sound, layer two identical versions of the dry sample. Then, detune one slightly flat and the other slightly sharp. Shazzam! Instant chorus and flange. An alternative (especially for chorusing) is to modulate the pitch of one sample with an LFO, instead of detuning the samples.

CABLE LABELS

Don't forget to label your cables at both ends, so that if you have to repatch them, you know where they go. Following tradition, I mark the mixer's channel inputs on a strip of masking tape below the faders, and the aux buses and other connections on a tape strip along the side. With the mixer labeled, I mark both ends of each audio cable according to the source device, e.g., "From K2000RS L. Main Out" and "To K2000RS L. Main Out," rather than "To Mixer Channel 5." I follow this procedure throughout the studio, including labeling all computer and MIDI cables. A short strip of masking tape, folded around the cable about four to six inches from the connector (for easy viewing in the rear of a rack), makes an inexpensive, easily marked, and easily removed label.

To keep it all straight, I post on the wall a set of word-processor tables that show the patching scheme and related information (such as MIDI channel assignments) for every cable and device in the studio. You can take this further by using a drawing program to create a simple signal-flow diagram for your studio. It's not as much work as it appears, and once completed, it saves a lot of time and frustration.

PROCESSING THE WIND

Wind controllers and keyboard breath-controllers offer expressive potential unavailable on keyboards alone. A MIDI processor such as the Yamaha MEP4 or Opcode's *MAX* on the Macintosh can help you realize more of that potential. For example, set the wind controller to send all Note Ons with a fixed Velocity of 127, or use the processor to perform the same task. Set the controller to send MIDI Volume (CC07), or use the processor to convert Breath Controller messages to Volume. This lets you perform a *sforzando-piano-crescendo* with any sustaining sound, which is not possible if the note is sent with low Velocity.

Most new instruments let you route Volume messages to control timbre and other sound parameters, but many older instruments are more limited in their controller-routing capabilities. If you are controlling such an instrument with a wind or breath controller, use the processor to add Breath Controller or Aftertouch messages to the Volume messages (or vice versa, if you are using a keyboard breath controller). The processor sends both streams of messages to control volume and timbre simultaneously with breath pressure. This fills up the MIDI bandwidth and a sequencer's memory rather quickly, but the resulting wind-like expression is worth it.

INDUCED HUM

Hum can enter the signal chain because of ground loops, poor cables, and improper cable-placement. Ground loops are a bit too involved for a short "tips" article, (see "On Solid Ground, Parts 1 and 2" in the September and October 1992 **EM** for more), but there are a few simple rules you can follow to improve overall audio quality in your home studio.

Your first step should be to make sure you're using appropriate, high-quality cables. The shielding in audio cables protects them from electromagnetic interference (EMI). Cheap cables usually have poor shielding, and cheap guitar cables are the worst; don't even think of using them. Some cables (especially those with plastic ends) don't even have the shield connected. If in doubt, replace them.

Next, clean up your act. Isolate all audio cables from sources of EMI and radio-frequency interference (RFI). Keep them as far as possible from AC lines; MIDI cables (a source of RFI); power transformers, including those in power amps and "wall-wart" power supplies; TV and computer monitors; etc. Don't coil audio cables around a metal table leg or rack part, lest you create an electromagnet. Snake your audio cables together, but away from everything else. Keep speaker cables (which are often unshielded) away from everything else, too. Even a small distance makes a big difference, as magnetic fields drop off exponentially.

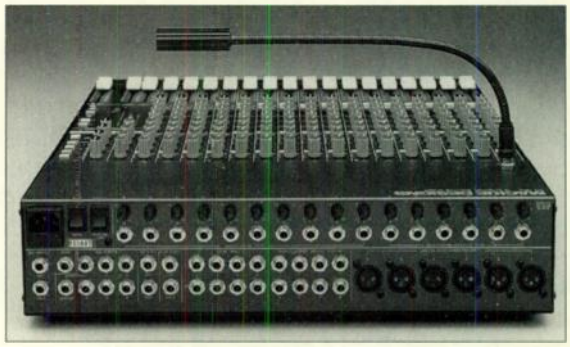

FIG. 4: Running shy on mixer inputs? Don't forget that, in a pinch, aux and group returns can be used as alternative inputs for some of your synth outputs.

MORE SYNTH VOICES

One of the most common complaints voiced by MIDI musicians is their lack of instrument voices. In spite of the increasing oscillator count on most of today's instruments, everyone seems to run out of synth voices, because as soon as you start using the really cool-sounding layered patches, your 24-voice synth dives down to 8- or even 6-note polyphony pretty quickly. One way around the dilemma (without simply buying another instrument) is to sync your sequencer to tape and record your first set of sequenced tracks onto one of the tape's empty tracks. Then you can hear your first synth parts and free up your synth voices to play other parts.

The trick to maintaining flexibility with the sequencer tracks you record to tape is to mute them, not erase them.

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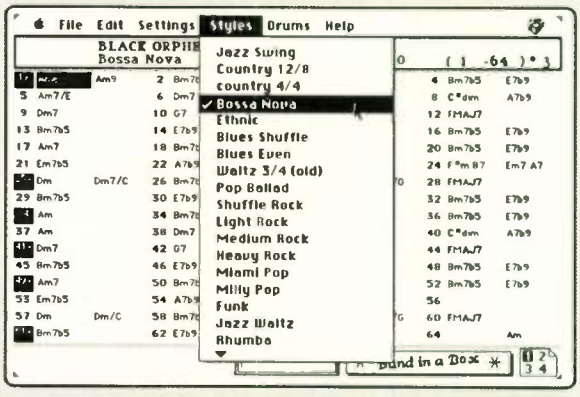

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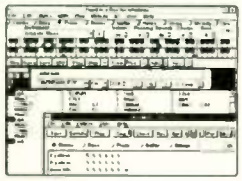

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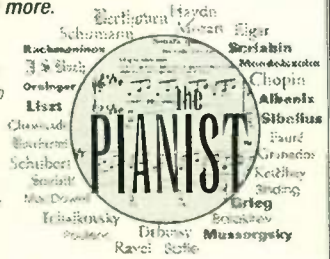

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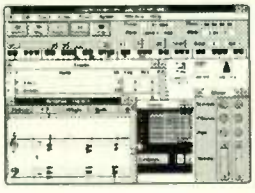

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But here's a cheap way to kludge "snapshot" mixer automation: Buy a $29.95 Polaroid camera. (Models such as the Cool Cam have a close-up feature and built-in flash.) Then, simply take pictures of your mixer EQ settings, fader levels, and aux send and return levels. You'll still have to manually document the parameters of any signal processors employed—no system is perfect! The snapshots even have a 1-inch bottom border onto which you can write the mix date, song information, photo description, and other pertinent data.

Although Polaroid film isn't a bargain (approximately $13 for ten shots), it's *way* cheaper than outfitting your mixer with an automation package. And after a long mix session, I'd gladly part with $13 to avoid spending an additional half-hour jotting down technical notes.

TIP OFF

Once you've checked out a few of these ideas, the real trick to making your studio a better place to be is to just dive in and do it. Nothing can substitute for exploring your gear in greater detail and trying a few things that you're not supposed to. You never know; it might just work.

WHAT DO

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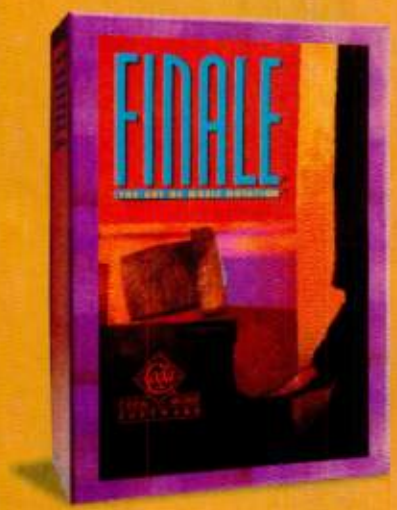

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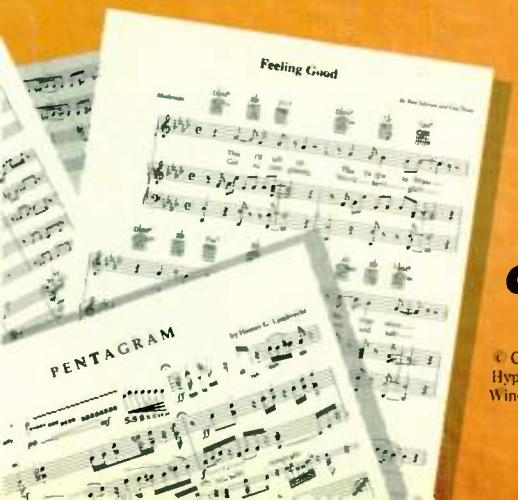

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By Larry "the O" Oppenheimer

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At first, reverb programming seems like an easy ride: Simply drop into edit mode, tweak the necessary parameters, and you're in business.

But the problem is knowing

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which parameters are "necessary" and how to tweak them.

DIGITAL REVERB BASICS

The phenomenon of *reverberation* occurs when sound waves encounter objects or boundaries and bounce off them. In an enclosed space, such as a concert hall, the sound continues to bounce from boundary to boundary—changing a little with each bounce—until its energy is dissipated.

A listener in the concert hall hears not only the original sound, but many of the reflections. The first few reflections off the boundaries closest to the source sound are distinguished individually by the brain, which extrapolates a great deal of information from them about the room and the listener's position in it. But as the sound bounces around, the number of reflections increases, and the time between their arrival at the listener's ear decreases. The brain quickly loses the ability to differentiate individual reflections and hears just the dense, decaying sound.

The listening environment almost completely determines the nature of the reverberation it produces. The room's shape and materials used in its construction define the number, direction, and spectral content of the discrete early reflections; the speed and manner in which the reflections build into reverb; the length of the reverb decay; the speed with which high frequencies decay in comparison to low frequencies; and the timbral nature of the reverb decay.

In practice, digital signal processors cannot match the complexity and detail of natural reverb. The trick is to create the most reasonable simulation possible by mastering common digital-reverb parameters. (It's important to note that terminology is often inconsistent between manufacturers, so it's worthwhile to read the parameter descriptions in the owner's manual.)

The most basic level of control is determining the type of reverb being simulated. Concert Hall (or Hall), Plate, and Room simulations are the most common selections. Halls tend to be spacious and somewhat slow to build in density. Plates have explosive attacks and very high density, while Room programs feature prominent early reflections and short decay times.

A parameter called *predelay* determines the amount of time between the original sound and the onset of reverb. The idea here is to simulate the distance between the sound source and the listener. Also, digital reverbs usually have some facility for simulating critical early reflections. Frequently, the delay time and level of anywhere from one to six reflections can be independently specified. For realistic simulations, the early reflections are set to values appropriate to the size of the space being simulated (easily calculated from the dimensions of the space and the speed of sound), and the predelay is set just a little longer than the latest early reflection.

Many reverbs have programs that generate only early reflections, with no later reverb. In some Lexicon reverbs, the early reflections (also known as *preechoes*) are not single delays, but clusters of delays. Lexicon claims these clusters more accurately simulate what occurs in nature, and the company even developed Spread, Spin, and Wander parameters to govern the behavior of these early sound reflections.

In most cases, *diffusion* and *density* parameters control the fashion in which reverb builds in density, while *attack* usually refers to the rate at which the reverb builds in volume. A space with distant walls would have a slower reverb attack, while your bathroom would have a very fast one. Some reverbs have a room-size feature that strongly effects every other reverb parameter.

Most digital reverbs offer control over *decay time*. The longest times are usually available on Hall programs, the shortest on Room programs. Plate programs typically cover the widest range of decay times. Many units offer separate decay times for high (HF) and low (LF) frequencies. Other parameters affecting tonal characteristics of the reverb include a high-cut filter (basically a post-reverb equalizer) and/or an HF damping factor. The damping factor could almost be thought of as a third decay setting, because it affects how quickly the high frequencies in the reverb die away.

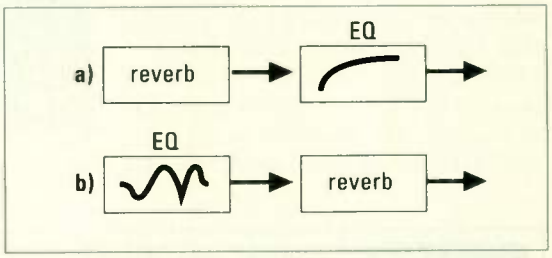

FIG. 1: A gentle, post-reverb low-frequency rolloff can keep reverb from muddying a mix (a), while selective EQ on the send can decrease unwanted resonances in the reverb (b).

APPLICATIONS

Digital reverb can be applied to achieve one of three goals: realism, enhancement, or transformation. If realism is your goal, consider the environment you want to create from an acoustical point of view. Visualization is helpful when trying to emulate real acoustic spaces. The parameters for creating reverb in a box are much different than those needed to simulate reverb inside the Taj Mahal. Trying to put a soft voice into a bedroom? Bedrooms are small rooms (at least where I live), so call up a small Room program, set the decay time as short as it gets, program a few early reflections (these come from the walls), and add a fair degree of HF damping, because there are usually a lot of absorptive materials, such as carpeting, bedding, curtains, etc. Be sure to mix the reverb very low, because you don't usually notice much reverb in a bedroom.

Enhancement is a broad area that I define as adding something to the

Lexicon's LXP-15 offers in-depth control over its reverb algorithms.

source signal that may not be true to life, but is not so radical as to render a sound unrecognizable. I often use reverb to enhance less-than-thrilling sounds, such as drum-machine samples. Drum-machine kick drums can be surprisingly flat and lifeless. An early reflections program—or a very small room with prominent early reflections—mixed judiciously with a touch of EQ, produces a kick with depth and body. This setting also works well with real drums (especially toms) that were originally recorded in dead rooms.

My most creative reverb trick is adding life to drum-machine cymbals. Call up a Plate program, and set the LF decay very low, the HF decay to approximately five seconds, bandwidth to full, and HF damping to zero. Be sure to feed only the cymbals into the reverb. Mix in the reverb level carefully, using the processor's programmable output level, an aux return, or a free channel fader on your mixer. If the wet/dry mix is right, it's almost impossible to tell that the "cymbal tail" is really reverb.

WHAT'S WRONG?

Sometimes the best way to use a reverb is the "wrong" way. Early Reflection programs are generally intended to simulate the primary and secondary bounces that happen in a space. However, the early reflections delay parameters often go up to completely unnatural settings of 100 milliseconds or more. These delays can be programmed to act as discrete echoes that may be blended into the basic reverb, or set to stand out from it.

Sometimes a reverb's worst sound can be just what you're looking for. Most reverbs start breaking down sonically at the extremes of their parameter settings. The shortest and longest decay times, the highest and lowest diffusion, and so forth, can make really strange, metallic sounds. All that's needed is to find the right source material, and these bizarre, ugly sounds might make your mix next week's chartbuster.

For example, if a reverb is set to a very long decay time, very low diffu-

DigiTech's new TSR-24 performs true stereo reverb processing and many other effects.

sion, and no damping or bandwidth limitations, it will probably give you a sound with obvious periodic fluctuations and a somewhat gritty texture. This effect is exacerbated by transients and high-frequency material. Capturing a sharply hit, sustained piano note in this reverb resulted in the most beautiful glittering, shimmering, fairy-dust sound I've heard.

Another successful "wrong" technique I've used concerns HF and LF decay times. In music mixes with many elements, muddiness in the low-midrange is often a problem. Reverb can worsen the problem considerably. To clarify low-mids, I frequently set the low-frequency decay time much *shorter* than the high decay to reduce the amount of reverb in that part of the spectrum. (In nature, such a condition is very rare.) This trick is particularly useful in live-performance situations, where many venues have boomy acoustical environments.

Combining equalization with reverb is a time-honored way of solving problems and devising clever effects. The effect of equalizing the reverb send is different than equalizing the returns, but both are useful (see **Fig. 1**). Some reverbs even include onboard equalization. You can smooth out a sound that has funny resonances by adding reverb with the resonances radically EQ'd out (usually on the reverb send). I made a great dance-mix snare sound by applying a "telephone filter" (all mids, no highs or lows at all) and some serious damping to a fairly normal Plate reverb.

The reverb's tonal controls can also be useful for solving EQ-type problems. For example, I often add Room reverb to drums recorded in a dry room. The problem is that I usually want substantial reverb on the toms, but little or none on the cymbals, and the tom and overhead mics are often submixed to save tracks. If I cut the HF decay time to nothing, all I get from the toms is body, no slap. My solution is to equalize the send, rolling off steeply at around 8 kHz, then adjust the HF cut until the cymbal grunge is not evident.

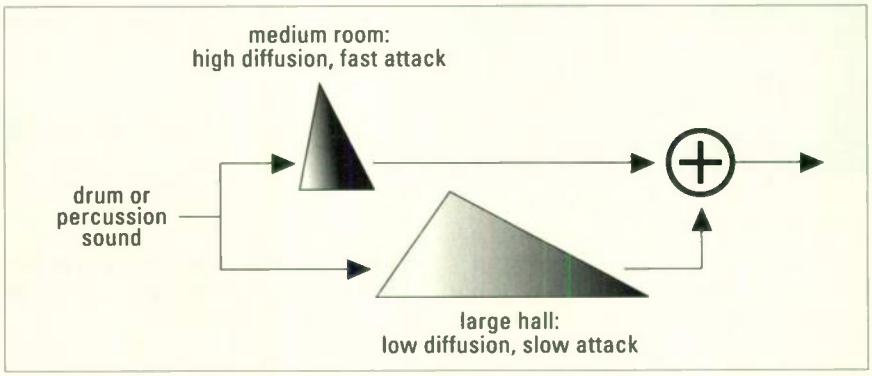

FIG. 2: You can smooth out the sound of percussion placed into a large space by using two reverbs in parallel and mixing their outputs.

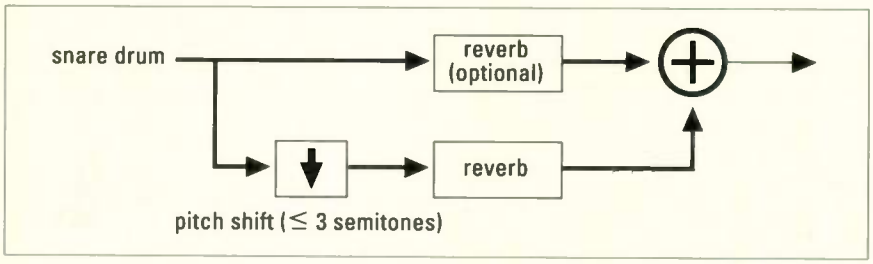

FIG. 3: One technique for beefing up a thin snare sound is to send a downwardly pitch-shifted version of the sound through a reverb, and mix that with the original. Pitch-shifting the reverb output could cause undesirable artifacts.

Compression is another valuable addition to reverb. On the send side, compression makes it easier to avoid input-level problems. It's a drag when an input level is adjusted to avoid clipping on peaks, but then barely lights up the level indicators for average signal levels. (In fact, I'd like to see onboard compression/limiting at the input of every digital processor.) On the return side, compression makes reverb more consistently audible, because it doesn't disappear into the mix quite as quickly as it decays. I've found compression most useful on genuine room reverberation, particularly when laying drum tracks in a studio.

MULTIPLE EFFECTS

Many fine effects can be produced by applying two reverbs to a sound. For example, you can use this trick when trying to put a drum or other percussive source into a very large space (see **Fig. 2**). Generally, large spaces have relatively low diffusion; that is, it takes a while for the reverb density to build. On a voice or a violin this can be beautiful, but on percussion it tends to sound like a lot of clicking in the early part of the reverb buildup. To get around this, I feed a second reverb from the same source, set it to a small to medium Room setting (a second or less decay time), and mix it in along with the large Hall (or whatever large space I've programmed).

The key here is to keep the reverbs from treading on each other. I program a fairly long predelay with no early reflections for the Hall, so that it doesn't really kick in until the room sound is dying. Some amount of tweaking is usually necessary to ensure a smooth transition. If the snare still isn't "in your face" enough, try adding some rather loud and very short (no more than five milliseconds) early reflections to the room program.

On the other hand, it can also be useful to use two reverbs in parallel for the express purpose of layering their outputs. With one reverb set to a fairly large, dark Hall and the other to a shorter, bright Plate, the sound has extra zing at the beginning of the reverb,

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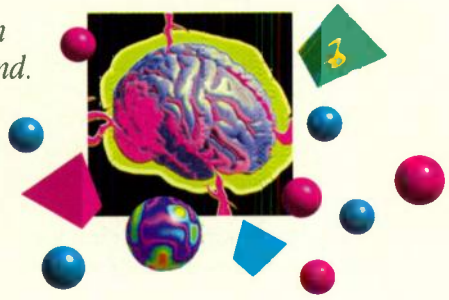

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then mellows into a darker, more spacious tail. I've even experimented with programming two identical units to exactly the same setting and mixing them together. This "twin tweak" resulted in a shimmering, phasing type of reverb that was almost certainly not mono-compatible. I have also tried using two mono reverbs, feeding the left channel into one and the right channel into the other. Because these two reverbs have no correlation, the sound is not really natural, but not so weird as to be disturbing.

Some newer multi-effects processors, such as the Ensoniq DP/4 or the Zoom 9200, let you feed the output of one reverb to the input of another, without external patching. This feature presents a host of possibilities, such as feeding an early reflection program into a Hall to get higher density. Of course, you can also use another effect instead of a second reverb. Feeding a reverb into a flanger or a delay can yield some terrific stuff. I made one great effect by feeding a fairly short reverb (1.2 second decay) into a multitap delay program with the delays set a few hundred milliseconds apart. I suppose one could call this "bunched reverb."

Pitch shifters mated to reverbs can also produce some excellent effects. As with equalization, either the send or the returns can be pitch-shifted with differing results. One example involved beefing-up a snare drum. I tried pitch-shifting the drum and it didn't sound good. I tried mixing the pitch-shifted snare with the original and that didn't cut it, either. Then I thought of feeding the shifted copy into the reverb in place of the original, which worked very well, indeed (see **Fig. 3**).

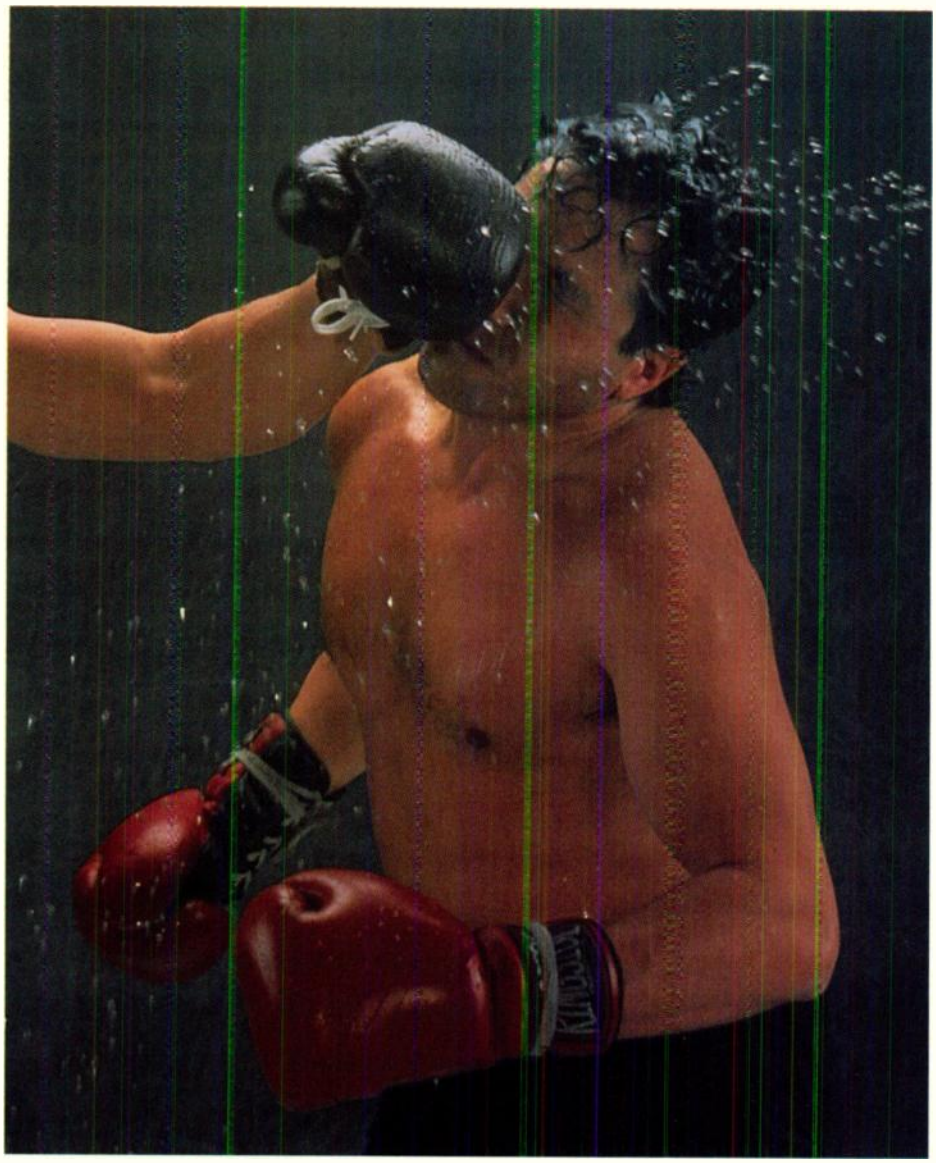
However, my favorite pitch-shift/reverb exploit was taking a single-line piano lead, feeding it into an "intelligent" pitch shifter programmed to produce diatonic harmonies, then sending only the shifted version into the reverb. (The reverb was set to a conservative chamber sound.) The client was completely tickled at the juxtaposition of a normal reverb sound, mixed at an unremarkable level, but carrying harmonies instead of the original. We then sent the original track into another reverb and mixed just enough of that in to keep the original from sounding apart from the mix.

Unfortunately, space does not permit me to even begin to discuss the vast possibilities of MIDI control of reverbs, but I will say that I automate reverb parameter and program changes on virtually every mix that I do.

FINAL DECAY

Creative reverb programming comes down to clarity. When you know what you want and what your equipment has to offer, you can quickly get the sound you need. Then, it's time to forget all that and fool around aimlessly for a while. You never know what lies around the next knob-twist.

Larry the O *recently added The Lazy Bums to the far-reaching collection of groups with whom he performs. No, really.*



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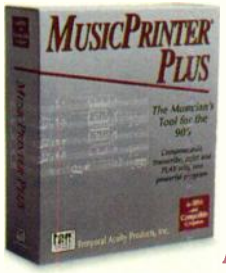

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By Michael Molenda

NAUGHTY Bits

Digital is one cool cucumber, but there *are* ways to heat things up.

The socialization of digital recording has done more than allow personal recordists to compete with big studio productions; it has also given us the right to complain about the quality of digital sound.

For years, a significant percentage of professional artists, producers, and engineers have rebuked what they consider the cold, sterile nature of the digital domain. Now, thanks to affordable hard-disk systems and modular digital multitracks (MDMs), such as the Alesis ADAT and Tascam's DA-88, personal recordists are joining the debate.

Of course, not everyone is critical of digital audio. But if you own an MDM or hard-disk recorder and find yourself getting nostalgic for the warmth and fatness of analog tape, fear not: There are ways to have your cake and eat it, too!

TEMPT GOOD BITS TO GO BAD

John Wayne never gave anything except his name, rank, and serial number when captured by enemy forces, and digital is just as tough. You can't cajole, seduce, or torture the medium into delivering anything but pristine audio. And if you *force* a digital signal to overload, it answers your brutality with self-annihilation (muting) or self-mutilation (distortion).

But there's more than one way to fry a bitstream. Just as a judo expert can turn opponents' greater size and weight against them, wiley recordists can exploit digital's precise resolution. While it's true that you can't coax warm, analog coloration from the medium, you can record wonderfully distorted sounds *into* it.

The following processing tricks can "de-sterilize" any signals recorded into an MDM or hard-disk system. And because the digital recorder plays back the roughed-up signals exactly as they were captured, your tracks can exhibit some of the warmth and fatness usually

restricted to the analog realm. Of course, these tricks are more illusion than miracle: If you want *real* analog sounds, you still must use analog tape. However, the illusions are authentic enough to stop even the most fervent analog loyalist from cursing the digital revolution.

THE HIGH-QUALITY SHRED

Unfortunately, the first trick isn't cheap: Record with a tube microphone. Long revered for their lush audio quality, tube mics add a pleasingly warm coloration to audio signals. This fatness is pretty much maintained when signals are recorded into a digital medium. Additionally, routing the mic through a compressor and assigning the signal to a subgroup allows you to slightly overdrive the various gain stages (see "Recording Musician: Gain Stages," p. 82) to produce a subtle thickness that emulates analog tape coloration.

The bad news is that tube mics often cost thousands, rather than hundreds, of dollars. Owning a classic used Neumann U47 requires parting with $3,000 or more. And although the U47 hasn't been manufactured since the 1950s, advances in technology haven't really lowered the prices for current tube mics. A Microtech Gefell UM935 costs "only" $2,495, and Sony's ballyhooed C800G lists for $6,000.

But you don't have to risk tenancy in the poor house to get your hands on a tube mic. Many classic and current models can be (temporarily) acquired through rental companies that cater to the recording industry. If there are no rental firms in your area, contact a local recording studio. The recession has forced many commercial facilities to scramble for revenue, so some operators might consider renting out a few microphones when bookings are light. Be forewarned, however, that they'll probably require a hefty deposit, as a prized tube mic is a valuable—and fragile—commodity. (Some engineers cherish their old Neumanns as much as their children.)

TUBE INTIMIDATION

If securing a U47 or other tube mic proves impossible, you can still reap the benefits of tube coloration by plugging into a tube microphone preamp. Both dynamic and condenser mics can be processed using mic preamps. Some preamps even accept high-impedence inputs for semi-pro mics and direct signals from guitar, bass, or keyboards.

Setup is easy: Just connect your mic or instrument to the preamp—for line signals, you may need a direct box if the preamp accepts only balanced XLR connections—and route the preamp's output to a channel input on your mixer.

Traditionally, the signal is then routed direct to tape (or disk) to attain an optimum signal-to-noise ratio. However, because we are attempting to *add* coloration, assign the signal to a subgroup. This detour introduces another stage of analog electronics that can help warm things up. The preamp also may have adjustable input and output levels—or a master level control—that can be used to further barbecue source sounds.

Of course, multiple gain stages often produce audible hiss. (And tubes aren't exactly silent critters, either.) Inserting a noise gate into the signal path during recording keeps the input quiet during non-performance passages. During performance, the hiss is usually masked by the vocal or instrument. If not, try backing down some of the fader or preamp levels. If all else fails, insert a single-ended noise-reduction device *after* the noise gate. These frequency-dependent processors diminish hiss—even during a performance—by attenuating high frequencies that are not part of the source signal.

If all this routing and patching and tweaking seems like a lot of work, it is! And because you are willingly adding noise and other artifacts to a source signal, you'll have to trust your ears to

Tube microphones are prized for their warm reproduction. Using a mic such as the Neumann U67 (shown) can help digital tracks sound more full-bodied. The classic Neumann tube mic, the U47, hasn't been manufactured since the 1950s, and used models command a price of $3,000 or more.

Valley Audio's Dynamite 2 is a dual compressor that can be used to heat up digital submixes. You can also reclaim the distorted glory of (analog-recorded) electric guitars by using extreme compression when recording in the digital domain.

ascertain what is, and what isn't, "good" distortion. However, the fat sounds this process yields are often well worth the trouble.

Tube mic preamps are typically less expensive than tube mics, although primo models such as Summit Audio's TPA-200B ($1,950) don't exactly nurture a tight budget. If you record predominantly instrumental music, you can save a few bucks by using line-level preamps such as Hughes & Kettner's Tubeman ($299). These tend to be guitar-oriented products, but they can also heat up synth and sampler tracks.

Also, don't forget to look for hidden treasures in want ads and at studio equipment sales. The mic preamp I use is an old, rebuilt Altec 1567A mixer amplifier. This forest-green reject from a 1950s sci-fi epic (it has BIG knobs) has six balanced XLR inputs, a master volume, and treble and bass controls. The total cost, including the refurbishing, was $250.

SONIC SQUEEZE PLAYS

Even if budget or lack of availability knocks you out of the tube-processing arena, you can still heat up digital recordings with a common solid-state compressor. For example, I'm often disappointed with the timbre of electric guitars recorded to digital. It seems that no matter how much the source sound rages, the recorded signal often appears stiff and cold. To compensate for the absence of analog tape coloration—a factor I believe is critical to achieving slamming guitar tones—I brutally compress guitar signals before they reach the digital deck. A ratio setting of 10:1 at a threshold of -10 dB is usually a good starting point for rough-and-ready timbres.

Digital drum tracks can also benefit from heavy compression. If you typically submix drums to a stereo perspective (to save room for other instruments), a dual compressor can crunch both tracks to produce a fatter sound. Be sure to activate the unit's stereo link function so that the two tracks are compressed equally. All parameters should be controlled by the master channel. A threshold setting of -5 dB or more often smooths out the sound to a Phil Spector-type rumble, without smearing individual elements. Dial in the ratio to taste—I usually use a 2:1 ratio—listening critically to avoid cymbal splattering. (Higher compression

ratios tend to limit input signals to the point where crash cymbals wash over everything.)

Recordists with 16-track or larger systems don't usually submix drums, but compressing individual drum tracks is seldom practical unless a platoon of compressors is available. However, you can still warm up your drum tracks by using a single (dual or stereo) compressor to process a stereo tom submix. Simply assign all tom tracks to a stereo spread and compress the subgroup. Because the other tracks—kick, snare, hi-hat, overheads—remain unaffected, you can get some nasty timbres by going a little overboard with the compression on the toms. I often hit the submix with a 10:1 ratio and a -10 dB threshold. During mixdown, adjust the level of the squashed tom tracks in relation to the unaffected tracks to produce the desired punch.

A TRULY DUMB IDEA

No treatise on engineering methods is complete without a couple of exquisitely stupid applications. The history of audio recording is chock full of absurd techniques that produced wonderful and bizarre sounds. Here's my contribution.

On a recent project recorded with ADATs, I wanted the chorus background vocals to sound huge and out of control. (The recording was just that: I exhausted twelve singers by stacking up ten tracks of group vocals.) Unfortunately, the digital resolution proved too clean for the desired bloated ambience. To rough things up, I mixed the ten background vocal tracks to a ¼-inch, analog 2-track. During the mixdown, I compressed the submix at a ratio of 15:1 with a -10 dB threshold, ran the mastering deck at 7 ½ ips (slower tape speeds increase coloration and noise), and pegged the recording levels into the red. The resulting mix sounded exquisitely anarchic. Now, the problem was getting the stereo vocal mix back onto the digital multitrack.

The analog 2-track didn't have center-track time code, so I couldn't lock up the analog tracks to the ADAT. I was forced to manually fly-in the mix.

Guitar processors can come to the rescue if pro audio products are too rich for your budget. For just $299, the Hughes & Kettner Tubeman can fatten up line signals from guitars, keyboards, and even drum machines.

A fine tube mic preamp, such as Summit Audio's TPA-200B, helps your mic collection deliver lush timbres. When recording in the digital domain, you can also overdrive the preamp to emulate analog tape coloration.

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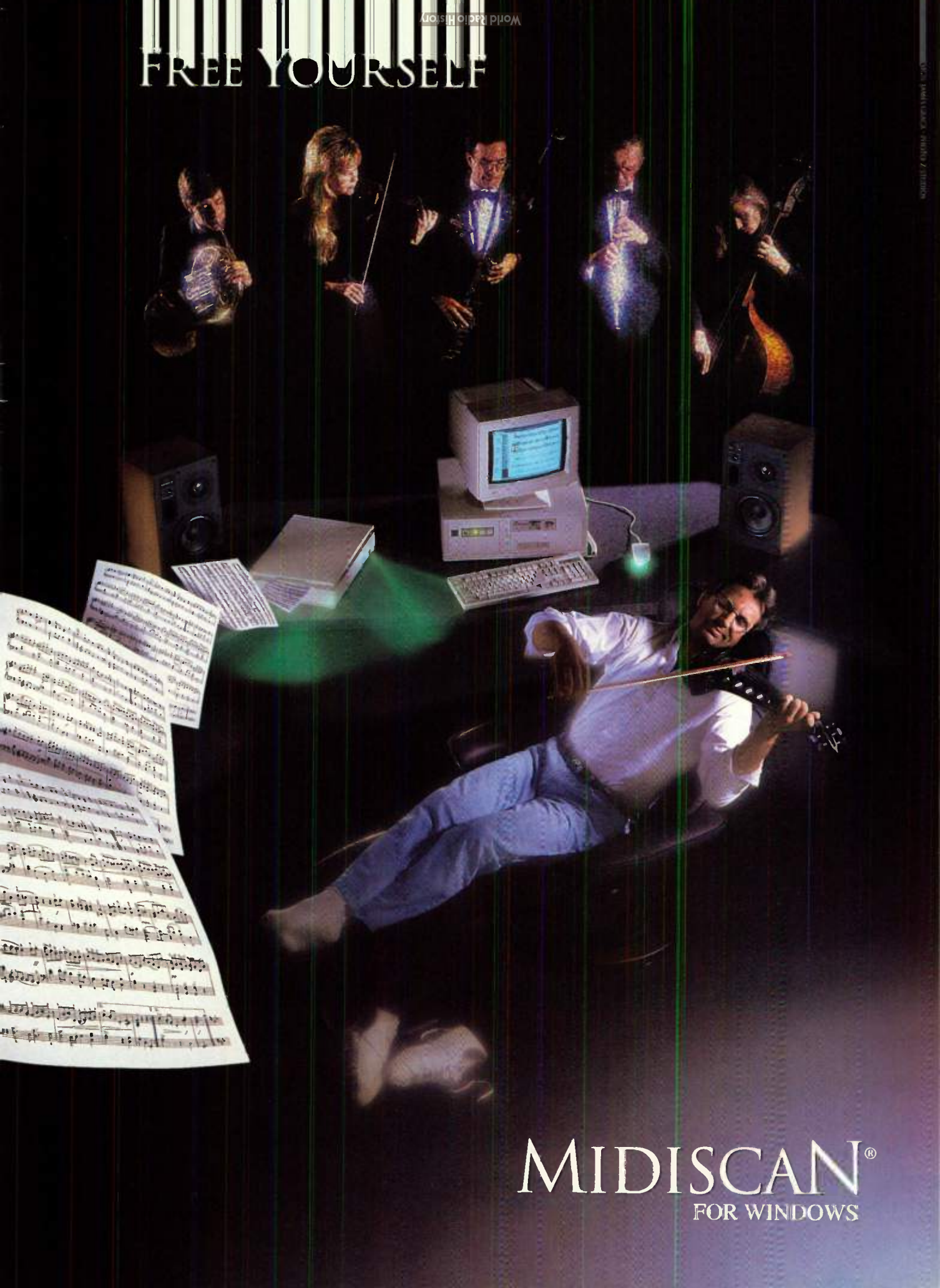

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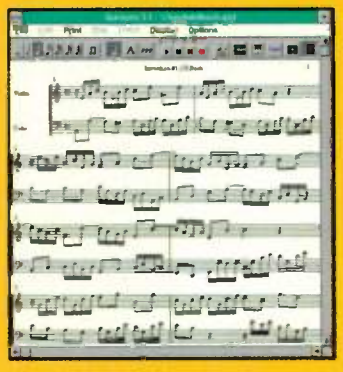

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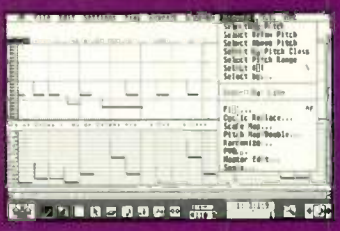

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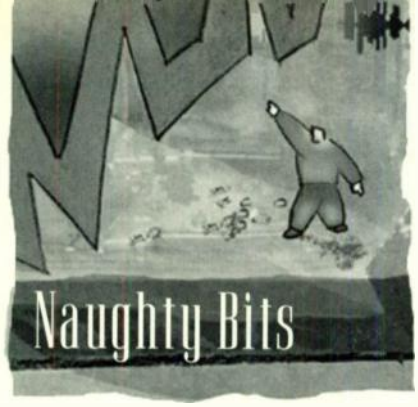

To facilitate cueing, I edited white leader tape at the beginning and end of the background vocal performance. A bit of dexterity was required to run the ADAT, then engage the 2-track and simultaneously punch in the vocal mix during the choruses. It also took a few tries to anticipate the time it took the 2-track's transport to engage, but after that, the choruses synched up perfectly. (If you try this method but still end up with the punched-in tracks slightly ahead of time, you can also use the ADAT's or DA-88's ability to delay individual tracks and dial up a perfect match.) Since then, I've used this system to punch in distorted organ riffs from sample-based synths to emulate cranky old Hammonds.

Here's a bonus bonehead tip: To digitally record the tortured vocal timbres often preferred in industrial and techno productions, I've had great luck with those toy, Mr. Microphone-type wireless mics. I use a small boom box as the receiver (the microphone transmits over a FM signal), place it in a tiled bathroom or hallway, and mike the box with a Shure SM57. The vocal tone is horrible! And I haven't met a digital recorder yet that could clean up Mr. Microphone's cacophony of crackles, pops, and squelches.

> An analog loyalist's mixing mantra might be: "Track on digital, mix on analog."

DIRTY MIXING TRICKS

My recording partner, Neal Brighton, has a personal mixing mantra: "Track on digital, mix on analog." Mixing to analog tape is a simple cure for sterile DAT masters, especially now that high-output tapes such as 3M's 996 and Ampex's 499 allow you to really slam down hot levels. On most productions, the signal-to-noise ratio of these high-output tapes can almost match the specifications for digital.

I mixed my latest single to analog (¼-inch format at 15 ips), then transfered the mix to DAT for duplication. I found the master to be much warmer than if I had mixed directly to DAT. Bass tones sounded more full, and the mix had an overall punchier quality. Of course, being the paranoid type, I auditioned both the digital and analog versions for some trusted friends. In a blind listening test (I didn't tell them which version was being played), everyone picked the analog mix. Most of

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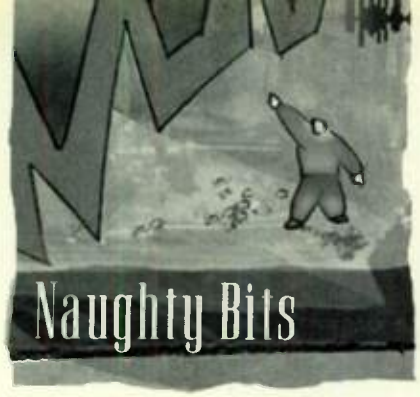

the "taste testers" even swore that the analog version was a different and better mix!

If you don't have access to an analog 2-track, try inserting a tube preamp, compressor, or EQ into the stereo bus before mixing. Hitting the tube device before your DAT recorder should help warm up the master. Another trick is to use a single-ended noise-reduction device to tame the high-end grittiness inherent in some digital multitrack recorders. (The anomaly seems more pronounced on tape-based systems.)

Insert a dual (or stereo) noise-reduction device into the stereo bus, and adjust the parameters to slightly diminish the high-end sizzle. Take care not to invoke too much processing; overdoing it can produce dull-sounding masters. Careful use, however, definitely calms the digital bite and helps mixes sound warmer and more full-bodied.

THE MAJESTY OF MUCK

Digital's promise of clean, transcendant audio quality is a boon to the recording industry. It would be foolish to argue otherwise. However, some tracks aren't *supposed* to sound dapper and well-scrubbed. In these instances, decisions regarding sonic quality should be made by the artist, producer, and/or engineer, not the recording medium. In short, digital's pristine bearing shouldn't inhibit recordists from making blissfully rotten noises. Sometimes a little audio mutiny is necessary to get the right sound. So I say, raise the Jolly Roger and burn those prissy digital bits until they scream for mercy!

EM *managing editor* **Michael Molenda** *edited the new (indispensable)* **EM** *book,* Making the Ultimate Demo. *This gold mine of valuable tips and techniques is published by Hal Leonard and is available through* Mix Bookshelf [tel. (800) 233-9604 or (510) 653-3307].

Dolby AC-2 Audio Compression System

A pioneer of film sound tackles pro-quality digital-audio compression.

By Scott Wilkinson

Digital audio is certain to become central to our lives. Sound files are large, however, so recording, processing, and storing audio data requires considerable memory, bandwidth, and storage capacity. As with other forms of computer data, compression can reduce the size of audio files, relieving the bandwidth and storage crunch.

Audio data compression is currently used for broadcast applications and relatively low-fidelity products, such as PC sound cards. Other potential applications include professional digital-audio recording, home theater, and cinema. Unfortunately, most compression schemes fail to completely retain the sound file's sonic and timbral qualities, which are paramount considerations in professional audio-recording environments.

Because high-quality compression is needed for a variety of applications, Dolby Laboratories has developed three different data-compression techniques. AC-1 is designed for direct-to-listener broadcasting, such as digital cable radio. AC-3 encodes six channels of audio into a low bit-rate data stream for commercial-cinema and home-theater applications.

That leaves AC-2, which is intended for professional audio applications, such as digital-audio recording and wide-area phone-line networks. Among the primary design criteria are a low bit-rate and minimal delay. (As we'll see, these parameters are mutually exclusive, so a compromise must be reached.) Other design criteria include high audio quality and low complexity, which translates to low cost.

In order to understand how AC-2 and most other audio-compression schemes work, it's important to understand how human hearing works. The human auditory system acts much like a multichannel, real-time audio analyzer, dividing the audio frequency range into several *critical bands*. One researcher has established 24 critical bands from 20 Hz to 15 kHz.

Another fundamental concept is called *masking*. If two tones with different amplitudes and frequencies sound at the same time, the louder tone may "mask" the softer one, rendering it inaudible. The relationship between the tones' frequencies and levels is critical to achieve this effect, which drops off sharply below the frequency of the louder tone (called the "masker") and more gradually above the masker's frequency.

Masking is crucial to reducing the amount of data needed to represent an audio signal. High-level frequencies can mask the audible noise that inevitably results from any compression process by shifting the noise to frequencies near that of the primary signal. In addition, low-level frequencies in the program material needn't be reproduced at all if they occur near a high-level frequency component.

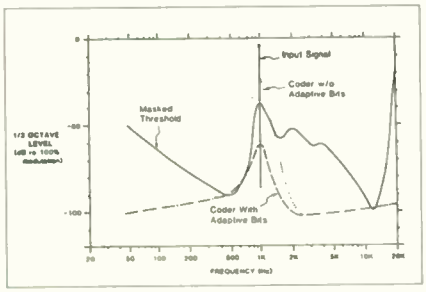

FIG. 1: The error-noise distribution for a 1 kHz sine wave fed into a moderate-delay AC-2 encoder. When adaptive bits are used, the noise falls below the masking threshold, except in one small section, and thus is generally inaudible. The noise is clearly audible when adaptive bits are not used. (Graph courtesy of AES.)

An AC-2 encoder, such as the Dolby DP501, accepts an analog audio input, converts it to linear digital form, and processes the resulting 16-bit samples in small groups called *frames*. Each frame is divided into critical bands, which are analyzed for their harmonic content, masking characteristics, and relative importance with respect to the entire signal. In addition to some fixed bits, a number of *adaptive* bits are then dynamically allocated to represent each band according to this analysis. This distributes the error noise to frequencies that are well-masked (see **Fig. 1**). At the other end of the line, an AC-2 decoder, such as the DP502, restores the signal to 16-bit, linear resolution and converts it back to the analog domain.

Currently, AC-2 includes two encoding schemes: low delay and moderate delay. The low-delay process sacrifices bit rate and compression ratio to minimize processing delay, offering an 8 ms delay with 4:1 compression and a data rate of 192 kb/s for each channel. The moderate-delay scheme takes the opposite side of the compromise with a delay of 45 ms, 6:1 compression ratio, and a data rate of 128 kb/s for each channel. The DP501 and DP502 follow the latter approach.

This technology holds great promise for the future of digital audio. AC-2 and other digital-audio compression techniques greatly increase the utility of available bandwidths and storage media, which is music to our ears.

(For more information, contact Dolby Laboratories; 100 Potrero Ave., San Francisco, CA 94103; tel. [415] 558-0200.)

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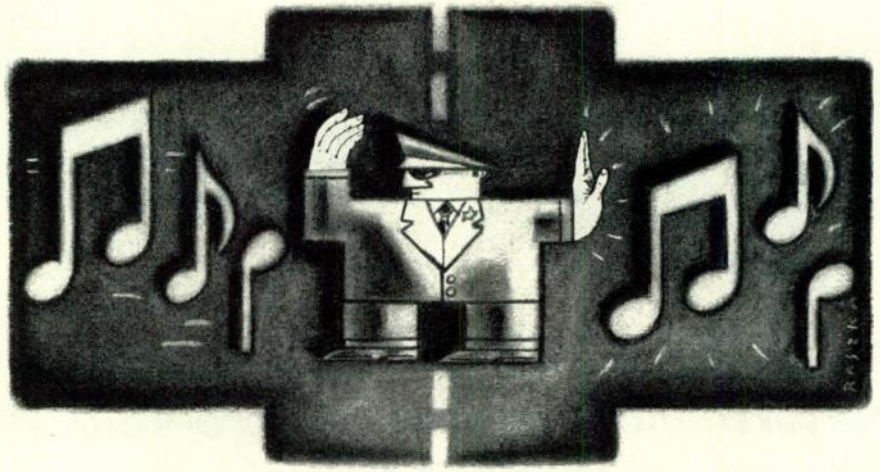
Basic Audio Connections, Part 2

By Scott Wilkinson

Making connections between the mixer, tape deck, and sound system.



Audio connections form the circulatory system of any recording studio; cables and connectors carry the audio signals from one device to another, each of which processes the signal before sending it on to the next device. Without the correct connections, the signal can be distorted beyond recognition, choking the life out of your studio.

Last month, we examined the audio connections between sound sources, mixers, and signal processors. Now it's time to discuss the connections between mixers, tape decks, power amplifiers, and speakers. If you haven't read last month's installment, you should before reading further.

ALL HANDS ON DECK

Like mixers and signal processors, tape decks have both inputs and outputs. Consumer stereo cassette decks operate at -10 dbV line level and use unbalanced RCA connectors. Semi-pro stereo decks and multitracks also operate at -10, using RCA or ¼-inch connectors. Professional tape decks operate at +4 dBm line level and typically use balanced XLR connectors.

To get a signal from the mixer to the multitrack tape deck for recording, connect the mixer's *subgroup* outputs to the inputs on the multitrack (see **Fig. 1**). Many mixers have four, eight, or more of these subgroups, which act as mixers within the mixer. Any channel input can be routed to any subgroup, letting you combine several channels and send them to a single output (hence the name "subgroup"). Ideally, there are equal numbers of subgroups and tape tracks. In many home studios, an 8-subgroup mixer is connected to an 8-track multitrack.

In some cases, there are fewer subgroups than tape tracks; typically, there might be four subgroups and eight tape tracks. The most common solution to this problem is to use an audio *patch bay*, which consists of a series of connectors called *patch points* on the front and back of a rack-mount housing. In home-studio patch bays, these connectors are usually RCA, ¼-inch, or a combination of both, and they typically work with line-level signals only (see "Recording Musician: The Patch Bay" in the May 1992 **EM**).

Connect the subgroup outputs from the mixer to one set of patch points and the tape inputs to another set of patch points (see **Fig. 2**). These connections should be made on the back of the patch bay. You can then connect any subgroup to any tape input by connecting the patch points on the front of the patch bay with short *patch cords*. For example, if you want to record mixer inputs 1 and 3 on tape track 5, route inputs 1 and 3 to a subgroup, then connect that subgroup to tape input 5 with a patch cord on the front of the patch bay.

If your mixer lacks subgroups, you have several alternatives. The simplest is to connect the main L/R outputs to a patch bay, where you can route the signals to the required tape tracks. In addition, if the mixer channels have either *direct outputs*, or *insert points*, you can route the signal for each mixer channel directly to a patch bay or tape-deck input.

The multitrack's outputs are connected to the *tape returns* on the mixer, which are nothing more than additional line-level inputs (see **Fig. 1**). Many mixers have at least eight tape returns. In some mixers, these tape returns are additional inputs along with the primary channel inputs; such mixers are called

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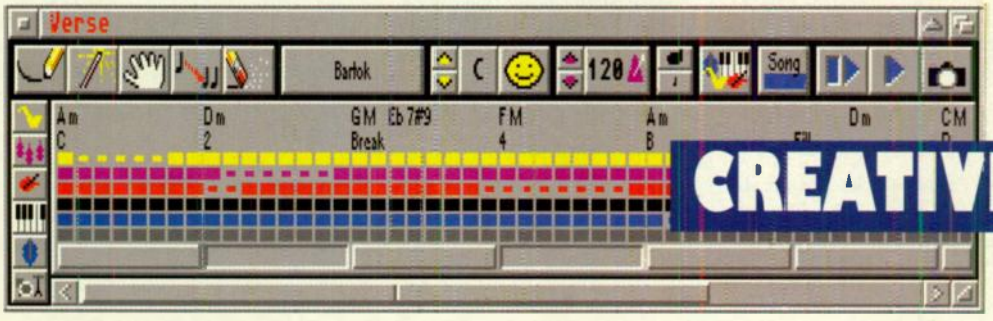

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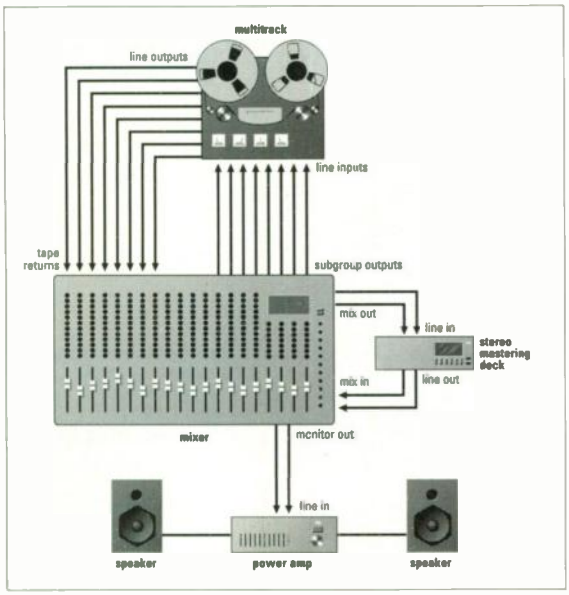

FIG. 1: In this system, the mixer's subgroup outputs are connected to the multitrack's line inputs, and the multitrack's line outputs are connected to the mixer's tape return inputs. The mix outputs are connected to the stereo mastering deck's inputs, and the mastering deck's outputs are connected to the mix inputs. The monitor output from the mixer is connected to the inputs of the power amp, whose outputs are connected to the speakers.

in-line mixers. Other designs include the tape returns in a separate section of the mixer; these are called *split* mixers. Most home-studio mixers use the in-line design.

If your mixer has no tape returns, you must use the primary channel inputs to hear and mix the tape tracks. To avoid unplugging and replugging the microphones and instruments that must also be connected to the channel inputs, use a patch bay. Connect the outputs from the multitrack to one set of patch points; the outputs from your instruments, guitar DI boxes, and microphone preamps to another set of patch points; and the mixer inputs to a third set of patch points (see **Fig. 2**).

You can then connect instruments and/or tape-track outputs to the mixer inputs with simple patch-cord connections. For example, connect the instruments to the mixer inputs during initial recording; during overdubbing and mixdown, connect some or all of the tape tracks to the mixer inputs. (For more on these processes, see "From The Top: Overdubbing" and "Mixdown Basics" in the June and July 1993 **EM**.)

As you can see, the connections between a mixer and multitrack use a lot of cables. To prevent a rat's nest of techno-spaghetti, use cable *snakes*, which are groups of cables bundled together in a large plastic sheath. The individual cables and connectors extend a couple of feet beyond the sheath at each end. Snakes are also useful between a rack of closely mounted audio gear and the mixer.

Of course, none of this matters if you are using a ministudio that incorporates a mixer and multitrack tape deck in one device. Tape-to-mixer connections are made internally, so all you need to do is connect your sound sources to the mixer inputs. Almost all ministudios include a switch on each channel input that selects between the line input and a tape track, almost like an internal patch bay.

Whether you use a separate tape recorder and mixer, or an integrated ministudio, you must be careful about feedback, which will occur if you route the output from a tape track through the mixer and back to itself. Howling feedback can damage your ears and equipment.

Ultimately, the tape tracks (and any MIDI sequenced tracks) are mixed down to a stereo master tape for duplication or listening on a sound system. Home studios typically use a stereo cassette or DAT machine for this mixdown. Most mixers include a *mix output*, which is connected to the inputs on the mastering deck (see **Fig. 1**). The output from the mastering deck is then connected to the *mix input* on the mixer; a switch on the mixer determines whether you are listening to the channel inputs and tape returns or the mastering deck.

If your mixer lacks a mix input, connect the output from the mastering deck to two channel inputs, tape returns, or aux returns. If you have no unused inputs, use a patch bay as previously described.

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POWER PLAY

Once the signals are mixed and processed through the mixer, and perhaps recorded on tape, you'll want to listen to the music over speakers or headphones. Virtually all mixers include a headphone output, which requires no additional connections. However, the signal from a mixer is at line level, which must be amplified before it reaches the speakers.

This is the job of a power amplifier. Connect the two *main* or *monitor* outputs to the line inputs on the amplifier (see **Fig. 1**). Some amps include balanced and unbalanced inputs; use the balanced inputs, if possible. Mixers intended for live-sound applications sometimes include an internal power amp, which makes these connections unnecessary. However, it's better to use a separate power amp in the studio, because it can be treated separately if something goes wrong.

The output of a power amp is said to be at *speaker level* (like line level and mic level, this refers to a range of levels rather than a specific level value). The output is measured in units of power called *watts*. There are many different ways to measure power; most manufacturers use a method called *root mean square* (RMS) to measure the average power output. Sometimes, you see a *peak power* rating in an amp's specs. (For more on power amps, see "The Power and the Glory" in the August 1993 **EM** and "Basic Studio Series, Part 2" in the December 1989 **EM**.)

SPEAKERS OF THE HOUSE

Finally, connect the outputs from the amp to the inputs of your monitor speakers. This connection is made with speaker cable, which is typically unbalanced and unshielded. It carries a high-level signal, so it must be quite hefty as well; 12-gauge or better is usually recommended. (Interestingly, lower gauge numbers translate to thicker, heavier wire.) Speaker cables usually have no connector on either end; they terminate in bare wires

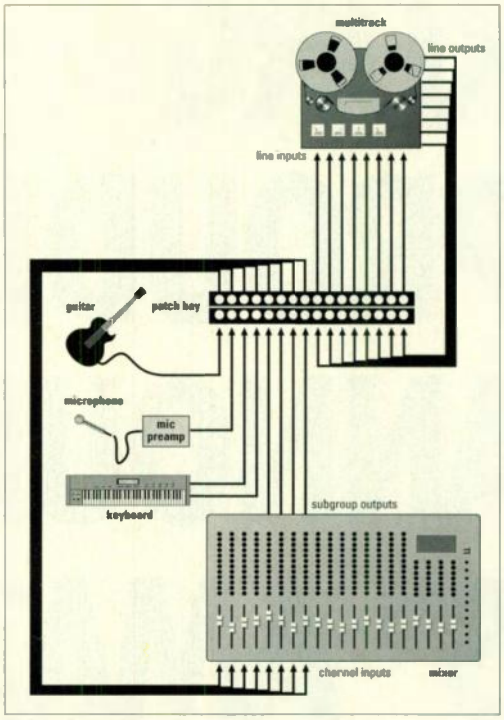


FIG. 2: The patch bay acts as a "traffic cop" for the audio signals being sent between the multitrack tape recorder and the mixer. Feedback is likely in this diagram if a "normalled" patch bay is used.

that are screwed down to terminals on the amp and speaker. Sometimes, speaker cables are terminated with spade lugs or *banana plugs*, which make the connections easier to establish.

Keep the speaker cables as short as possible. Because they are unbalanced and unshielded, they can pick up induced noise, although the high signal level reduces this risk. Place the power amp as close as possible to the speakers, and run long, balanced cables to it from the mixer's main outputs.

Speakers are rated as to the amount of power they can handle from the amp. Like power amps, most speakers include a continuous and peak power rating. Make sure that your amp and speakers have similar power ratings. If the amp is radically overpowered with respect to the speakers, it might destroy the speakers entirely. If the amp is underpowered, it will easily distort the signal; paradoxically, this can damage the speakers, as well.

Another important speaker spec is called *impedance*, which is measured in units called *ohms* (abbreviated with the Greek letter omega, Ω). This is the electrical resistance the speaker presents to the current from the amp. The higher the speaker's impedance, the less current it draws from the amp. As the impedance drops, the amp must work harder to supply the current drawn by the speaker. In addition, the power output of an amp drops when it's connected to high-impedance speakers. For example, a particular power amp might deliver 100W to an 8Ω speaker, or 200W to a 4Ω speaker.

Many speakers have an impedance of 8Ω, while others have an impedance of 6Ω, 4Ω, or even 2Ω. Most amplifiers can handle speakers with different impedances, but if the impedance drops too low, the amp could be damaged, so you should make sure that the impedance levels of your amp and speakers match.

CONCLUSION

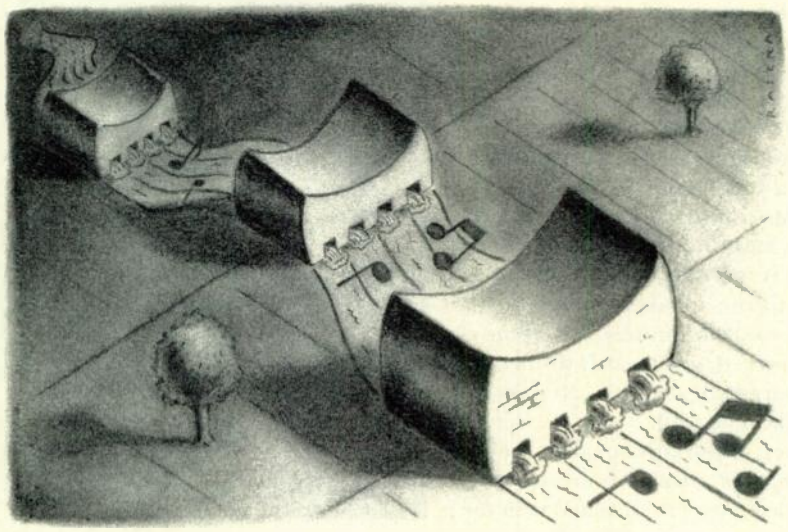
Well, there you have the basics of audio connections. The information presented here over the last two months should put you well on your way to connecting the audio in your studio with a minimum of trouble.

Scott Wilkinson, EM's *technical editor, loves to make connections between anything and everything.*

Gain Stages

By Larry "the O" Oppenheimer

If you want optimum recording levels, gain structuring takes center stage.



How is it that an important audio-engineering technique is not only one of the least understood, but one of the least discussed? Gain structuring takes the stage as soon as the cables are run and stays there throughout the entire recording process. It's not necessarily a benign presence. Poor gain structuring can ruin the sonic quality of your tracks.

Ah, I can almost see your head nodding in agreement about the importance of this seldom-mentioned technique. However, I can also hear that tiny little voice deep inside your brain, asking nagging questions about whether you *really* understand what gain structuring is.

THE GAIN MAZE

The answer is fairly simple: Gain structuring refers to how you choose to set the myriad level controls populating most signal paths. The active circuit element governed by a level control is called a *gain stage.*

To illustrate how gain stages can affect the purity of a signal, let's consider the recording of a simple keyboard part (**Fig. 1**). Modern electronic instruments typically have several internal software-controlled level parameters and a master volume slider on the front panel. Fortunately, the output levels of most digital musical instruments can be set quite hot, though I wouldn't crank them up all the way. Analog synths are another matter, as their output stages are easily clipped. On the other hand, a little bit of analog output distortion can be desirable, fattening the sound (as with tube distortion). Start with the synth's volume fader at a moderately hot level and experiment.

When the instrument is connected to the mixing console, it passes through the channel input trim and channel fader. From there, the signal—if it isn't routed directly to tape—is assigned to a subgroup fader and then to the master fader.

Aside from the main signal path, the keyboard may also be fed through one or more effects-send pots. These sends are routed to effects-send masters and then out to signal-processing devices. Of course, signal processors have their *own* input-level controls and usually have several internal level parameters.

The processed signal may then travel into an effects return or channel input, where it may go through a group on its path to the stereo master. Whew! Obviously, there are many opportunities for setting improper levels that can compromise the sound quality of your keyboard patch. And even all these gain stages don't take into account the passive, non-user adjustable levels inherent in some equipment and the calibrated input (level) reference on the recorder. How can you possibly determine optimum settings for all these highly interactive level controls?

LEVEL WITH ME!

As with most audio techniques, the answer is a combination of theory and application. Professional engineers often use rules of thumb modified by experience and individual circumstance. The critical issue is that each level control exists within the context of an entire signal path. This path is affected by both "upstream" and "downstream" level controls. And here's another little detail: The signal-to-noise ratio can only get worse after a signal leaves the instrument.

A practical guideline for achieving optimum levels is to avoid setting any gain stage at its highest or lowest extreme,

BRIAN RASZKA

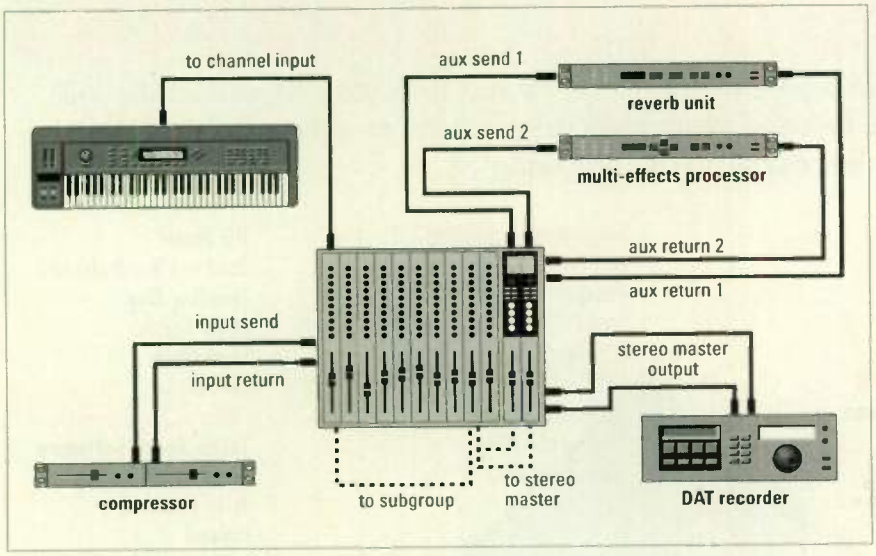


FIG. 1: Recording a simple keyboard part isn't simple when you consider the number of gain stages the signal must travel through. An improper level adjustment at any point along these signal tendrils can compromise audio quality.

with the possible exception of the original source's output level. Typically, an ideal level for each gain stage is 75 percent of its maximum output. This setting allows "room to move" and ensures that no stage is operating at the fringe of its functionality.

My usual approach is to start at the beginning of the signal chain and work through each gain stage. I raise each level control from zero up to an optimal level—as determined by metering, listening, and experience—with an eye toward keeping all levels within the maximum output margin (75 percent). This process often involves readjusting controls until an ideal balance is achieved. (It is wise to keep monitoring levels moderate until the gain structure is established in order to protect speakers and ears against rude surprises.) Don't be slavish, however, there are times when uneven or extreme settings produce the best sound and noise performance. Use your judgment.

Also, remember that each gain stage has a behavior pattern that must be considered when setting levels. Every stage has a maximum input level (before distortion rears it ugly head), a noise floor that varies with the amount of gain, and a definitive breaking point regarding how well it can handle signal transients and extreme high or low frequencies.

Fortunately, knowledge of all the different "gain personalities" can help you tame even the wildest signal paths. Let's get acquainted with some of the more typical gain stages.

PREAMPS

The preamp trim level is the most important level setting in the signal chain. Preamps—especially microphone preamps—apply more gain to a signal than anything else in the signal path. Once again, experimentation and critical listening are your main tools in learning preamp characteristics. (Some mixers help you out by providing pre-fader metering and/or preamp clip indicators.)

If the incoming signal is very low, the preamp must be cranked high to compensate. Unfortunately, extreme level settings not only add noise, they leave you vulnerable if the signal has a huge dynamic range. A topped-out trim has little or no headroom to accommodate louder passages, and soft passages will require additional gain to be made up elsewhere.

In most cases, it is best to set the preamp trim in a position that leaves some flexibility and avoid changing it. Any adjustments affect not only the main signal path, but also the pre-fader sends and insert points. Also, remember when setting a preamp trim that an EQ boost will add more gain to the signal. If you don't allow sufficient headroom, the signal may clip.

In sound reinforcement, the level of the preamp trim often determines the amount of signal gain before feedback occurs. In my experience, feedback problems increase when preamp trims are turned up, instead of when volume is increased at the channel or group faders.

FADERS

From the preamp, the signal continues to the channel fader. Again, the setting of this fader affects a number of things, especially headroom. If a fader is set very high, there isn't much room to increase levels should the signal volume dip. Conversely, if the fader is set too low, it is difficult to perform a smooth fade-out. The channel fader also determines the maximum level available to post-fader sends and direct outputs, as well as any subgroups it feeds.

This brings up a perfect illustration of the interactive nature of gain structuring. Let's say that while all other levels are fine, the signal processor isn't getting sufficient level. Unfortunately, the channel post-fader send pot and the post-fader master are set at maximum. In this case, changing the preamp trim or channel fader would upset several other balances, so the best move is to simply turn up the signal processor's input-level pot.

If *that* pot is already at maximum, you have a tougher decision to make. It may not do too much damage if you change the preamp or fader level, but the wisest choice may be to live with the low input level. You can make up the insufficient level at the processor's output stage, or at the mixer's effects-return pot. Many processors have rear-panel switches that alter the input sensitivity and/or output levels. These present more alternatives, but you shouldn't assume the added gain will be clean and quiet. The best choice

Even relatively uncomplicated mixers such as Soundcraft's Spirit Folio 4 have multiple gain stages that can butcher audio signals.

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can only be made through an evaluation (usually a quick one) of the individual circumstances, knowledge of your equipment, and the priorities of the given situation.

The considerations for the group fader are usually less complex than those for the previous gain stages. The primary parameters are headroom within the group, the send level to the recorder (if you are recording from groups instead of direct outputs), and the balance with other groups within the stereo mix.

STEREO MASTER

Finally, there is the stereo mix. Recording consoles have monitoring pots to provide independent level control between the stereo master and the listening volume in the room. One would not want to have the meters for the stereo mix very low "in the grass" in order to make the listening volume comfortable. But gain structuring is a consideration even with the monitoring

▼

The preamp trim level is the most important level setting in the signal chain.

pot. A relationship must be established between the monitor-level pot and the power-amplifier level pots to allow the range of monitoring levels you need.

LEVELED OUT

Most people don't have the time or resources to analyze why their mixer preamps or synthesizers sound better at one level setting than another. But if you work with the same equipment for awhile, you'll discover audible differences between level settings. Obviously, the sonic quality of your masters can only improve if you take the trouble to find optimum gain ranges. I recommend taking time outside of a pressured session environment to devise systematic listening tests of different combinations of settings. Don't be afraid to experiment. The bottom line with gain structuring, as in all audio matters, is do whatever sounds good. ❦

Interactive CD Audio

By Paul Potyen

The making of Paramount Interactive's Rock, Rap 'n Roll.

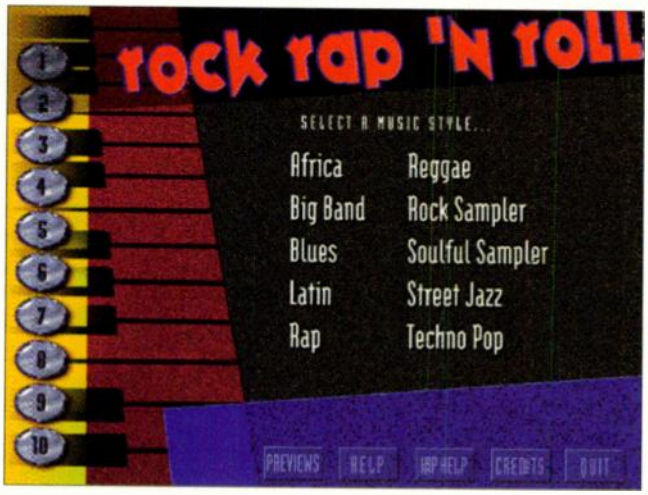

FIG. 1: The opening screen from *Rock, Rap 'n Roll* shows the ten different music styles in which you can create songs.

Most of today's successful CD-ROM titles are entertainment products that take advantage of the whiz-bang motion-video technology Apple's *QuickTime* and Microsoft's *Video for Windows* have wrought. But recently, a pair of San Francisco Bay Area companies teamed up to produce an award-winning title that uses interactive audio. *Rock, Rap 'n Roll Digital Music Machine,* developed for the Macintosh and now available for *Windows*-equipped PCs, is a fun music-construction set that lets you piece together tunes from prerecorded audio clips. Produced jointly by Interactive Audio and Medior, Inc., and published by Paramount Interactive (700 Hansen Way, Palo Alto, CA 94304; tel. [800] 821-1177), *Rock, Rap 'n Roll* appeals not only to those who know nothing about music theory or production, but also to professional musicians.

When you double-click on the *Rock, Rap 'n Roll* icon, you encounter the opening screen (see **Fig. 1**), which provides a gateway to one of ten musical styles (or "rooms"), such as Latin, Rap, Street Jazz, and Techno Pop. Choose a room by clicking on its name, and you are presented with a screen like the one shown in **Fig. 2**. Once in a room, you drag any combination of ten prerecorded music loops, which are from two to eight bars long, into one of the "Song-a-lizer" slots at the bottom of the screen. Clicking the Start button causes your sequence of loops to cycle repeatedly until you stop it.

You can then jam along with your customized sequence by clicking on hot buttons on the screen and/or pressing keys on the computer keyboard. An independent key map is available for each music style. Available sounds include horn phrases, vocalizations, and percussion sounds.

Rock, Rap 'n Roll is noteworthy in several respects: First, it's *very* cleanly recorded, with 8-bit, 22 kHz resolution; and second, all the loops work together from a musical standpoint, no matter how you arrange them, which is no trivial task. Another intriguing aspect of this well-conceived package is that it lets you record, save, and play back your own performances to and from your hard disk. And if you have 8-bit audio-digitizing hardware, you can record your voice or any other performance along with your sequence.

PROTOTYPE DEMOS

Interactive Audio president Gary Levenberg describes how the concept for *Rock, Rap 'n Roll* came about. "Last summer [composer] Nic tenBroek and I started thinking about what kind of product we could come up with that would make it fun for people to make music. Ideally, we wanted to make a hand-held toy to do this, but we opted to create something on the computer using the keyboard and mouse as input devices. And since we know how to create music that's used in an interactive way, it seemed like a logical thing to do."

They used MacroMedia *Director* to create a prototype of their concept. "*Director* lets you play two tracks of audio," says Levenberg, "so we could play this 'beat' track and click on some buttons to make other sounds play at the same time." They took the idea to Barry Schuler at Medior, Inc., a multimedia developer they work with frequently.

Levenberg then created a *HyperCard* interface that demonstrated the main ideas for the product, such as selecting audio loops so they would play consecutively. The final program was written in C. "We're doing very sophisticated things with the Macintosh *Sound*

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Manager," emphasizes Levenberg, "and at the same time reading input from the keyboard and the mouse. Neither *HyperCard* nor *Director* is capable of responding quickly enough to that kind of processing."

AUDIO RESOLUTION

Levenberg says he would have liked to have used 8-bit stereo for playing back the digitized music, but he was limited by space considerations and other problems they encountered on the Mac. "Using audio on the Mac forces you to deal with Apple's *Sound Manager* and its limitations. For example, at the time of this product's release, it supported only 8-bit, 22 kHz files." The new *Sound Manager* (3.0) supports 16-bit files, although it plays them at 8-bit resolution if you don't have a 16-bit sound card installed in your computer.

"A bigger problem," continues Levenberg, "happens when you mix sounds with the version of *Sound Manager* [2.0] we used while developing the project. The number of audio tracks you can play at one time is dependent on processor speed, but each time you add a track, *Sound Manager* cuts the level down by 6 dB.

"We designed the *Rock, Rap 'n Roll* so that when you play back a sequence you recorded to hard disk, you hear the samples mixed together appropriately; that is, without the 6 dB cuts. Thankfully, *Sound Manager* 3.0, which is now shipped with the program, solves this problem."

MUSIC PRODUCTION

Producing the music was one of the most interesting parts of the project. As Levenberg explains, "Our basic idea was to create songs in modules you can put together in a variety of ways. Our first approach is manifested in the first two rooms we created: Rock Sampler and Soulful Sampler. If you listen to all the loops in the Rock Sampler room, you'll notice the second five have the same drum beats as the first five, but with additional rhythm parts on top.

"The other eight songs were written more like typical pop or rock songs in which the loops correspond to the chorus, bridge, verse, break, etc. We used *Vision* to create the rhythm beds. *Vision* is perfect for this because you can assemble your song from little pieces called sub-sequences. We digitized each sub-sequence, played it three times into our 8-track Pro Tools system, and kept the middle section, which was easier to loop because of reverb tails, etc. Then we mixed it to stereo and created a final loop."

Levenberg and tenBroek decided to use MIDI for the drum and bass tracks. All the other parts were written out for

FIG. 2. Once you've entered a particular music "room," you can create tunes by putting together the ten preprogrammed groove loops in any order. You can also jam along by hitting other keys on the computer keyboard.

THE PRODUCT

For anyone interested in the audio possibilities of interactive CD-ROM, *Rock, Rap 'n Roll* is an excellent example of how to do it right. It works on any *QuickTime*-capable Macintosh with a minimum of 4 MB of RAM, a CD-ROM drive, and System 6.0.7 or later. The new *Windows* version requires 4 MB of RAM, *Windows* 3.1, an 8-bit sound card, and a CD-ROM drive. The company also offers floppy-disk versions of the program with three music "rooms" instead of ten. Suggested retail price for the CD versions is $79.95; it's $59.95 for the floppy-disk versions. You can get *Rock, Rap 'n Roll* at your local retail store under the Paramount Interactive label.

musicians and singers. Each musician (or horn section) was recorded separately into Pro Tools while listening to the rhythm-bed loops. "The musicians never got to listen to or play the whole song all the way through," Levenberg says. The final step in recording was to let the musicians and singers improvise to a repeated loop. "Afterwards, we saved the best takes written for each rhythm-bed loop, and then picked out the best solo and lead phrases to be used for overlaying by the user."

Levenberg feels it was no different from producing an album in terms of audio production. "We had to produce all ten songs. One difference was that we didn't really do any mixing in a traditional sense; all we did was set volumes for each element. This was easy in Pro Tools." Another difference from

▼

"I think interactive audio is going to be an important part of the music industry."

—Gary Levenberg

album production and typical CD-ROM production was the breakdown of the budget. "On a typical production, you should figure on spending 20 to 30 percent on music and audio. In this case it's really more like 50 percent music and audio, 30 to 40 percent programming, and 10 to 20 percent art."

At the end of the music recording and editing process, they had a 16-bit, 44.1 kHz stereo mix of all the elements. Converting it downward was a bit tricky. "We had to convert the files into 8-bit resolution and make sure all the loops were intact. If you use *Sound Designer* to convert to 8-bit, 22 kHz files, two bad things happen: First, to make sure you don't get weird artifacts from filtering, the software adds samples at the beginning and end. That means the loops have extra samples in them, so the rhythm is not quite right.

"Second, sample-rate conversion from 44.1 kHz to 22 kHz does not achieve the best results. It's better to transfer digitally to DAT, then resample at 8-bit resolution with compression using the

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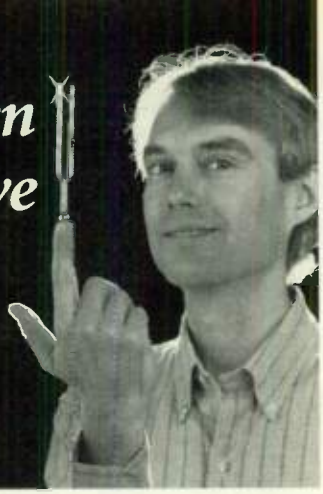
"They laughed when I said they could have Perfect Pitch ...until I showed them the secret!"



The TRUE STORY by David L. Burge

IT ALL STARTED in ninth grade as a sort of teenage rivalry.

I would practice the piano for five hours daily. Linda didn't practice anywhere near that amount. But somehow she always seemed to have an edge which made *her* the star performer of our school. It was frustrating.

What does she have that I don't? I would wonder.

Linda's best friend, Sheryl, sensed my growing 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. "*Linda's* got *Perfect Pitch*."

"What's Perfect Pitch?" I asked.

Sheryl gloated over a few of Linda's uncanny musical abilities: how she could name any tone or chord—*just by ear*; how she could sing any pitch she wanted—*from mere memory*; and how she could even 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 someone a mastery of the entire musical language!

It bothered me. Did Linda *really* have Perfect Pitch? I finally got up the nerve and point-blank asked Linda if the rumors were 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 couldn't wait to make her eat her words...

My plan was ingeniously simple:

I picked a moment when Linda least suspected it. Then I boldly challenged her to name tones for me—*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 other classmates could not help her. I got everything just right so I could expose Linda's Perfect Pitch claims as a ridiculous joke.

Nervously, I plotted my testing strategy. Linda appeared serene.

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 quickly played another tone. She didn't even stop to think. *Instantly* she announced the correct pitch.

Frantically, I played more and more tones, here and there on the keyboard, but each time she knew the pitch—without effort. She was SO amazing—she could identify tones as easily as *colors!*

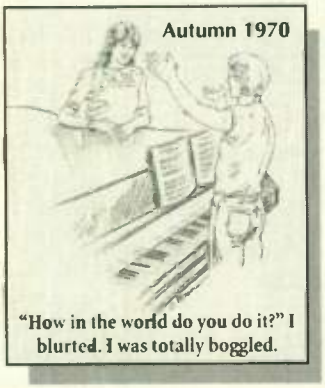
"Sing an E♭," I demanded, determined to mess her up.

Without hesitation she sang the proper pitch. I had her sing more tones (trying hard to make them increasingly difficult), but still she sang each one 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 great dismay, that was as much as I could get out of her!

The dazzle of Perfect Pitch hit me hard. My head was dizzy with disbelief, yet from that moment on I knew that Perfect Pitch is real.



"How in the world do you do it?" I blurted. I was totally boggled.

I couldn't figure it out...

"*How does she DO it?*" I kept asking myself. On the other hand, why doesn't *everyone* know musical tones by ear?

Then it dawned on me that *most musicians* can't tell C from C♯, or A major from F major—like artists who brush painting after painting without ever knowing green from turquoise. It all seemed so odd and contradictory. I found myself even more mystified than before.

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 brothers and sisters into playing tones for me so I could guess each pitch by ear. My many attempts were dismal failures.

So 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. I simply could *not* recognize the pitches by ear.

After weeks in vain, I finally gave in. Linda's gift was indeed extraordinary. But for me, it was out of reach.

Then came the realization:

It was like a miracle. A turn 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 "let go"—and *listened*—to discover these subtle differences within the musical tones.

Soon I too could recognize the tones by ear! It was simple. I could hear how F♯ sounds one way, while B♭ has a different pitch color sound—sort of like "hearing" red and blue!

The realization hit me: THIS IS PERFECT PITCH! This is how Bach, Beethoven and Mozart could mentally envision their masterpieces—and identify tones, chords and keys just by ear—*by tuning in to these subtle pitch colors within the tones.*

It was almost childish—I felt that *anyone* could unlock their own Perfect Pitch by learning this simple secret of "color hearing."

So I told my best friend Ann (a flutist) that *she* could have Perfect Pitch too. She *laughed* at me.

"You have to be *born* with Perfect Pitch," she asserted.

"You don't understand how Perfect Pitch *works*," I explained. "It's *easy!*"

I showed her how to listen. Timidly, she confessed that she too could hear the pitch colors. Soon Ann had *also* acquired Perfect Pitch! We became instant school celebrities. Students tested us in great amazement. Everyone was awed by our virtuoso ears.

Back then I would not have dreamed I would later explain my discovery to college music professors. When I did, many of them *laughed* at me at first. You may have guessed it—they told me you had to be *born* with Perfect Pitch.

But once I revealed the *secret* to Perfect Pitch—*and they heard for themselves*—you'd be surprised how fast they'd change their tune!

As I continued with my own music studies, my Perfect Pitch allowed me to progress far faster than I ever thought possible. I even *skipped over* two required college courses. Perfect Pitch made *everything* much easier—performing, composing, arranging, sight-reading, transposing, improvising—and it skyrocketed my *enjoyment* as well. I learned that music is definitely a HEARING art.

And as for Linda?

...Oh yes—well, time found us at the end of our senior year of high school. I was nearly 18, and it was now my *final chance* to outdo her.

Our local university sponsored a high school music festival each spring. That last year, I scored an A+ in the most advanced performance category. Linda only got an A.

Sweet victory was mine at last!

TODAY, thousands of musicians and *two university studies* have confirmed the effectiveness of my Perfect Pitch method. *Now I'd like to show YOU how to discover your own Perfect Pitch—whatever your age!*

I hope *you* won't laugh as you picture yourself with various Perfect Pitch skills—like naming tones and chords by ear with laser-like precision! Of course, you might be surprised at how *simple*—and how *very valuable*—Perfect Pitch really is!

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Quadra and reloop the sample with *SoundEdit Pro.*"

Levenberg adds that there is yet another problem with sample-rate conversion: *Sound Manager* is designed for audio sampled at 22.255 kHz, not 22.050 kHz. Although you can specify a conversion rate in *Sound Designer* that yields 22.255 kHz, the converted files sound grainier because of quantizing errors. On the other hand, if you downsample at a ratio of exactly 2:1, the file plays back slightly off-speed in the application, and the pitch is slightly off.

"Eight-bit audio is difficult," maintains Levenberg, "because it takes more time to get it right than 'real' audio. If you use a spectrum analyzer to compare an 8-bit file that was converted using *Sound Designer* with one that used the Macintosh hardware, the *Sound Designer* file wins hands down. But if you listen to the two sounds, the Macintosh wins, because it seems to have much less quantization noise. I think it's because Apple really did a good job optimizing the hardware for 8-bit audio."

LOOKING TO THE FUTURE

The folks at Interactive Audio are not content to rest on their laurels. What's next? "*Celebrity Rock, Rap 'n Roll.* Ideally, we'd like to record name artists, but we could also use existing material." And concerning the current content of *Rock, Rap 'n Roll,* Levenberg comments, "We have the master files, and we're looking at other platforms that support better audio quality."

Levenberg also had some comments on the philosophy behind what they've done. "People like Todd Rundgren are getting involved in multimedia, but his vision is different from ours. Todd argues that listening to music is passive, but I'm not sure I agree. I think this concept is going to be a part of the music industry, just like remixed songs are now. In a sense, that's what we're doing, plus a lot more. We're letting you be your own remixer."

With the continued growth of interactive audio products like *Rock, Rap 'n Roll,* the move toward more active listeners could, in fact, have a profound influence on how the music industry works and how musicians create music. The future isn't clear, but it sure looks interesting.

Paul Potyen *is an associate editor at* Mix *and a freelance composer/producer.*

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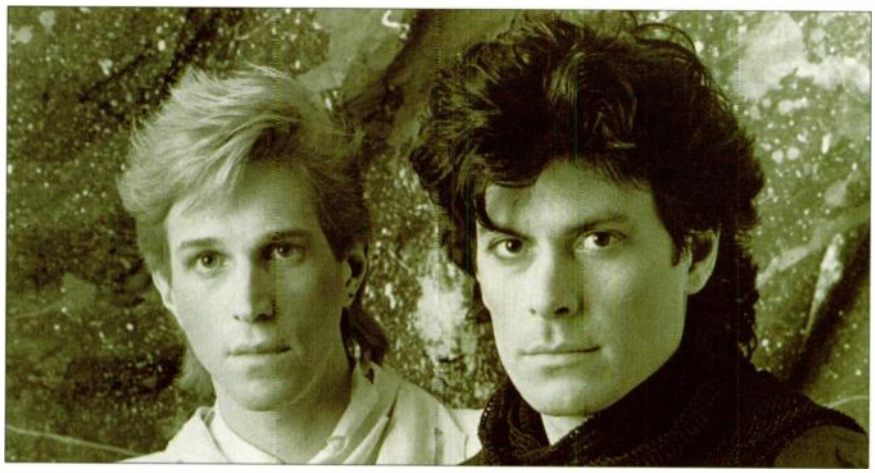
Contract Warfare

By Michael A. Aczon

Surviving the trench combat for record deals requires taking the offensive.



The Hooters (pictured are frontpersons Rob Hyman, left, and Eric Bazilian) proved they deserved a major deal by releasing a self-produced cassette album that ultimately sold 100,000 copies.

You don't have to follow music-business trade magazines to be aware of the mega-million-dollar record deals signed by artists who have found the intersection of art and commerce. And as multimedia and cable entertainment conglomerates scramble for programming, future deals will most likely expand territories, formats, and the pool of money available for artist contracts.

Unfortunately, this is not great news for struggling musicians. Although opportunities often increase with programming needs, the stakes are so high—especially considering the enormous investment required to break a new act in today's market—that the chances of signing a major-label recording contract run about even with winning the Super Jumbo Tri-State Lottery. And if you want even *that* microscopic chance at a label deal, you'll need to plan a near-military offensive based on field intelligence, intensive planning, and decisive strategy.

BASIC TRAINING

Prior to inking the deal that promises fame and fortune, it's essential to possess a clear artistic vision. It's also critical that this vision can be clearly articulated to a label executive. Are you selling a solo act or a band? Are you emphasizing songwriting or production chops? Can you play live? Are there so-called "key members" in your act that are irreplaceable? Do you want complete artistic control over your project and image? All of these factors contribute to the packaging of your presentation.

For example, if you want to prove you can handle the responsibility of producing a commercial master tape, your demos should sound like hit records. If you wish to showcase only your songwriting chops, a "less-produced" demo is appropriate. And if charisma is your major strength, make sure that a video or live showcase is part of the shopping package.

The bottom line is this: Most label executives give you just one chance to explain your act. It is critical that you tailor a presentation to emphasize your good points. Unfortunately, accomplishing this goal sometimes means keeping your day job so you can spend more time (and money) tightening up your demo tape, or fine-tuning your live show. If you make a great first impression, your new communication line with industry heavies is worth all the extra struggle. Bad impressions can sever the industry lifelines for months.

THE DEAL COMMANDOS

The greatest act on earth will not find its public unless a record executive finds the act first. Let's face it, if not for the tenacity of Brian Epstein, the Beatles might still be playing cover tunes at The Cavern Club. Just about every label in Europe turned down the Fab Four before Epstein wrangled the fateful meeting with George Martin at EMI Records.

Finding someone to shop an act is not easy. Not only must the individual be dedicated and tireless and charismatic and convincing, but he or she must have enough industry clout to approach the right people. Lawyers are tried and true links to record contracts, as are independent producers (who often act as "pre-screeners" and artist-development agents for the labels) and personal managers.

The price of opening the door of opportunity varies, depending upon the representative. Powermongers with

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proven track records charge big bucks for "leasing" their clout and contacts. They may get you a deal, but you'll probably be locked into a long-term contract that devours a substantial percentage of your royalties and other music-related earnings. Of course, these deal-makers are at the high end of the music-industry food chain. Less-expensive, and less-experienced, representatives are legion.

However, joining forces with a "weak" dealmaker doesn't necessarily ensure defeat. (Remember David and Goliath?) Fruitful deals are cut by all types of people for all types of artists. Let's look at three fictional deals to illustrate how an assault on the music industry can pay off.

THE DUAL OFFENSIVE

Sally Singer can't write a song or play an instrument, but she is an above-average jingle and session singer. Singer's demo tape—which showcased her vocals on songs written and recorded by local songwriters—came to the attention of Peter Producer, an aspiring independent songwriter and producer. Producer decides to provide Singer with original material and produce master tapes for the express purpose of signing the vocalist to a major-label contract. Because Producer has successfully exploited his songwriting talents for years, he can get his phone calls answered by high-level people in the industry.

Singer and Producer enter into an exclusive artist agreement, which commits them to each other for the duration of any third-party deals they sign. During the recording of Singer's master tapes, Producer uses mostly his own tunes to increase the possibility of generating songwriting and publishing revenues. However, Producer also introduces Singer to the art of songwriting, and they collaborate on several tracks.

When four masters are completed, Producer prepares a photo package to showcase Singer's carefully developed visual image. (Singer's "look" is another collaborative effort.) Then, Producer calls all of his contacts: fellow songwriters, his lawyer, BMI and ASCAP representatives, major and independent record-label A&R executives, and music publishers.

A bite finally comes from an independent label that commits to releasing one of the songs as a single, provided they get the rights to the subsequent album *and* a specific number of future records. As an added bonus, the little indie has a distribution agreement with a major label. *Voila!* Singer is now a recording artist on a small label with major distribution. Producer—because of the clout garnered by a major label distributing a record that contains a number of his songs—negotiates a separate publishing deal for his complete songwriting catalog.

GROUP TACTICS

The Major Labels, an alternative band with a unique and startling live show,

Robert Berry (far right) started out as just another small studio owner in California's booming Santa Clara county. However, Berry's talent as a multi-instrumentalist earned him quite a reputation as a demo producer and one-man-band for hire. When the music industry came calling in the late 1980s, Berry ended up in the band 3 with legends Keith Emerson (far left) and Carl Palmer (center).

SUCCESS STORIES

Anyone seeking a career in music should adhere to the cliché "knowledge is power." The more you know, the less chance you'll get blitzkrieged in a raw deal. (Although, I've seen some real brainiacs get slaughtered on the bargaining table.) On the brighter side, an understanding of the music industry's superstructure can help you develop a viable career plan. **EM** has published numerous articles on the business of music. The following back issues can be ordered from Mix Bookshelf at (800) 233-9604, or (510) 653-3307. A 24-hour fax is also available for placing orders: (510) 653-5142.

In addition, some of these career-oriented articles are reprinted in the new **EM** book, "Making The Ultimate Demo" (Hal Leonard Publishing). You also get scores of tips for recording killer demos. Call Mix Bookshelf to order.—*Michael Molenda*

Article	Issue
"Working Musician: Lyric Writing"	6/93
"Working Musician: Record Promotion"	5/93
"Working Musician: The Management Dossier"	4/93
"Working Musician: The Business of Music Production"	3/93
"Working Musician: Shopping Your Demo Tape"	2/93
"Working Musician: Getting Down To Business"	1/93
"Working Musician: Mirrored Images"	12/92
"Working Musician: Performing Rights Societies"	10/92
"Working Musician: Making Contact"	9/92
"Working Musician: Music Contracts"	6/92
"Working Musician: Comprehending Copyright"	2/92
"The Architecture of Pop"	5/92
"Production Tips"	12/91
"Legal Beagles: How Lawyers Hunt Record Deals"	11/91
"Making Your Album"	6/91
"Marketing Your Demo"	6/91

Sometimes luck plays as much a part in getting a deal as talent. Michelle Shocked recorded herself strumming a few tunes at a campfire, and the demo just happened to meet the right ears. The funny thing was, she not only got a record deal, the label released her Walkman-recorded "live" tape as her first album!

becomes the darling of the artsy community. They also attract significant attention from the local press.

A regional A&R representative from a major label catches the buzz, attends a club date, and likes what she sees. In an effort to get other label representatives to see the band, the regional rep works with the group's manager to get the Major Labels a showcase gig. After the show, the regional rep finds an ally in a "higher-up," who agrees to fund a demo project in exchange for the first shot at signing the band.

If all goes well, the Major Labels will get a deal based on the strength of the regional rep's ears and commitment to the project. In return, the regional rep will have a signed act under her belt, which increases her chances for advancement at the label.

DIVIDE AND CONQUER

The Urgents start out as a cover band, but soon tire of Holiday Inns and weddings. The band decides to seek a record deal. To make important industry contacts, each member uses their regionally renown musical chops to get road gigs with major recording artists. Between tours, the members get together to write, rehearse, and record demos. Feeling they're ready, the Urgents pool their money, release a self-produced single, and hire an independent record promoter to generate radio airplay.

Soon, the Urgents generate enough of a buzz that the band's fledgling personal manager is able to set up a private audition with a label executive. The showcase is successful—it didn't hurt that the single was receiving tons of college airplay and that major artists

▼

The greatest act on earth will not find its public unless a record executive finds the act first.

vouched for the musicianship of each member. Soon the Urgents are recording their first album for a major record label.

GRACE UNDER FIRE

The ticket to success for all these groups was that they made their contacts happen. It's easy to moan that powerful lawyers and label execs plot to keep talented people out of the music industry by inundating the media with cute, no-talent marketing pawns. Contrary this belief, the record industry needs new artists to survive.

However, the warriors who enter (and exit) the industry arena with the fewest scars are usually smart tacticians who are practical about fame, money, and business realities. The opportunities for victory are out there. Go get that deal!

Michael A. Aczon *is a San Francisco-based entertainment lawyer.*

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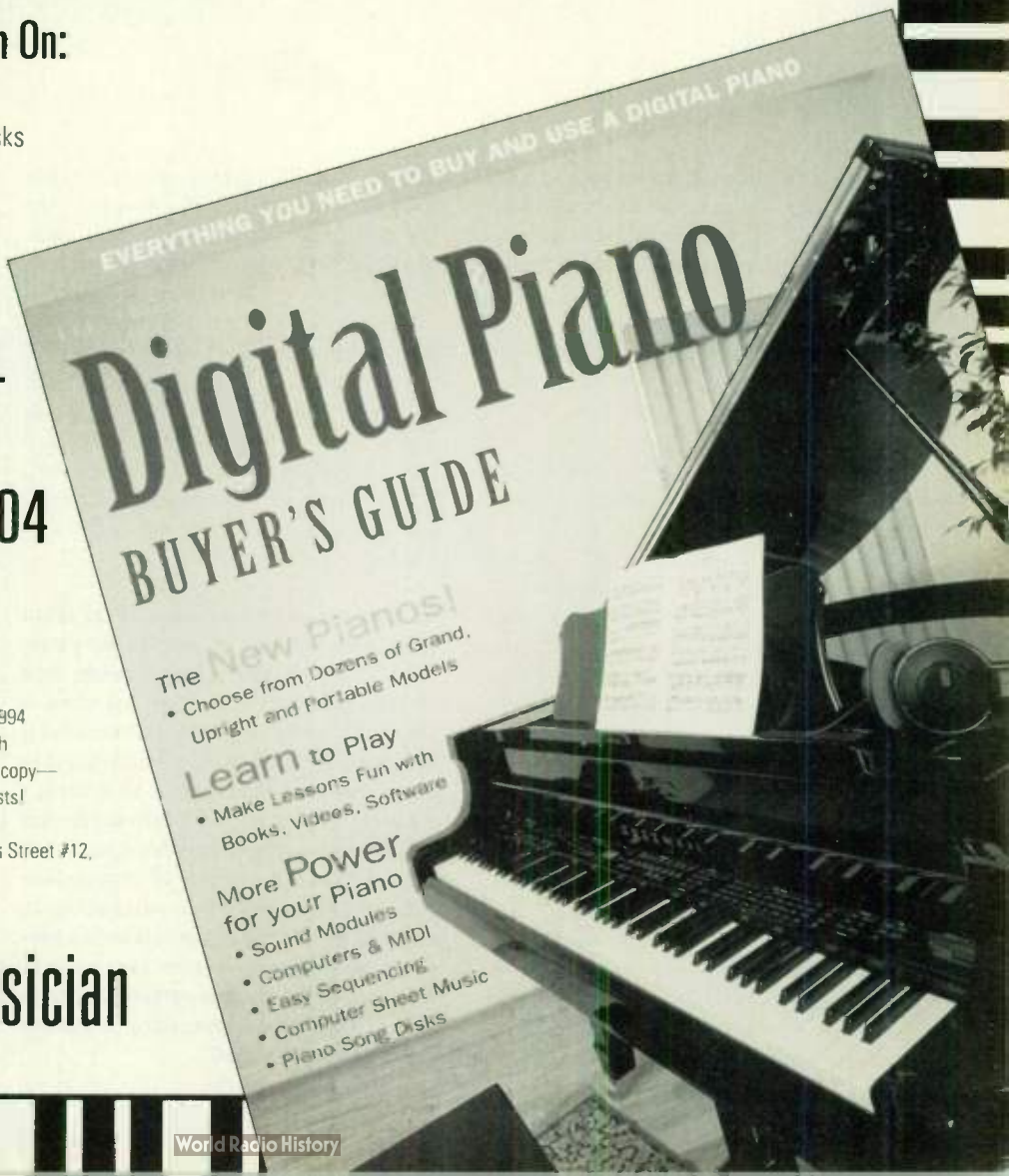

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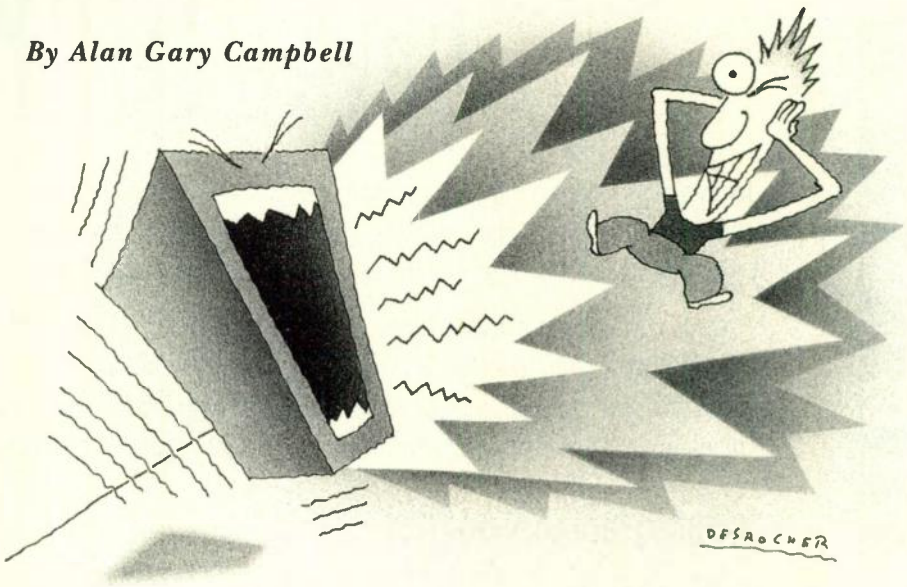
Electronic Musician

Questions & Answers

By Alan Gary Campbell

The Sultan of Service discusses SPL meters, JV-30 output failure, and screw extraction.



Q. How loud does sound have to be to cause hearing damage? Is there an inexpensive way to measure dB levels on the gig?

A. Reference sources vary somewhat with regard to numbers, but a commonly quoted figure (EPA) for safe, continuous exposure is 75 dB SPL. Try to find a music work environment that quiet! In comparison, consider that many auto sound systems produce such dangerous levels that, transported to an industrial setting, legal exposure limits would be reached in seconds.

The value of measuring sound-pressure levels to identify and avoid potential sources of hearing damage can hardly be overestimated. Everyone who works with amplified sound is a potential victim of hearing loss; moreover, noise-damage effects are largely *cumulative*. When the damage is acute, the results can be devastating: deafness, tinnitus (ringing), intermodulation effects, and extraneous noises. Aside from precluding work in music, such symptoms can severely degrade quality of life. (For more on hearing preservation, see "Hear Today, Gone Tomorrow," in the June 1993 EM.)

Radio Shack carries an affordable ($31.95), analog sound-level meter (catalog number 33-2050) that measures up to 126 dB SPL in eight ranges, with fast or slow damping and A or C weighting, and operates on a 9-volt battery. This device does not offer the ultimate in precision or accuracy, but it is useful, well-made, and attractive. It can be tripod- or bracket-mounted and looks cool on a camera mini-tripod. When you first use a sound-level meter, you may be shocked to discover how loud many ambient sounds are.

Q. The output of my Roland JV-30 synth "goes away" at times. It doesn't make static, it just goes silent for a few seconds, then comes back on. It doesn't do this often. It seems to happen only under MIDI control (I don't play from the keyboard much) but is in sync when the sound returns, so it doesn't appear to be MIDI-related. I've tried different cables, mixers, amps, etc. What could it be?

A. Since the audio failure is intermittent, and you don't play from the keyboard very often, it seems likely that the symptom does, in fact, affect the output under keyboard control, but this effect has not been observed. The JV-30, like many current synths, uses transistor-based, output-muting circuits to disconnect the outputs during power-up, reset, etc. Given that the problem affects both line outs (and, presumably, the headphone out), the source is most likely the output-muting control subcircuit, comprised of transistors Q5, Q7, Q9, zener diode D2, and associated components (refer to a JV-30 service manual). It may be difficult to catch this intermittent fault with a 'scope and quicker simply to rebuild the subcircuit. The muting subcircuit is driven directly from the system Reset line, so it seems unlikely that the problem lies elsewhere. Note that this is definitely a job for an authorized service center.

Q. I recently took apart my JV-30 to replace a broken keytop, and when reassembling it, I noticed that the right-front hole for the keyboard mounting screw is not even close to lining up. I left the screw out temporarily. Is it possible that I bent something, or that the instrument is damaged?

The frame of the JV-30 keyboard assembly is used in several products, and the rightmost section (as viewed from above) constitutes an unused key position. In the JV-30, the mounting screw passes through a section of the frame,

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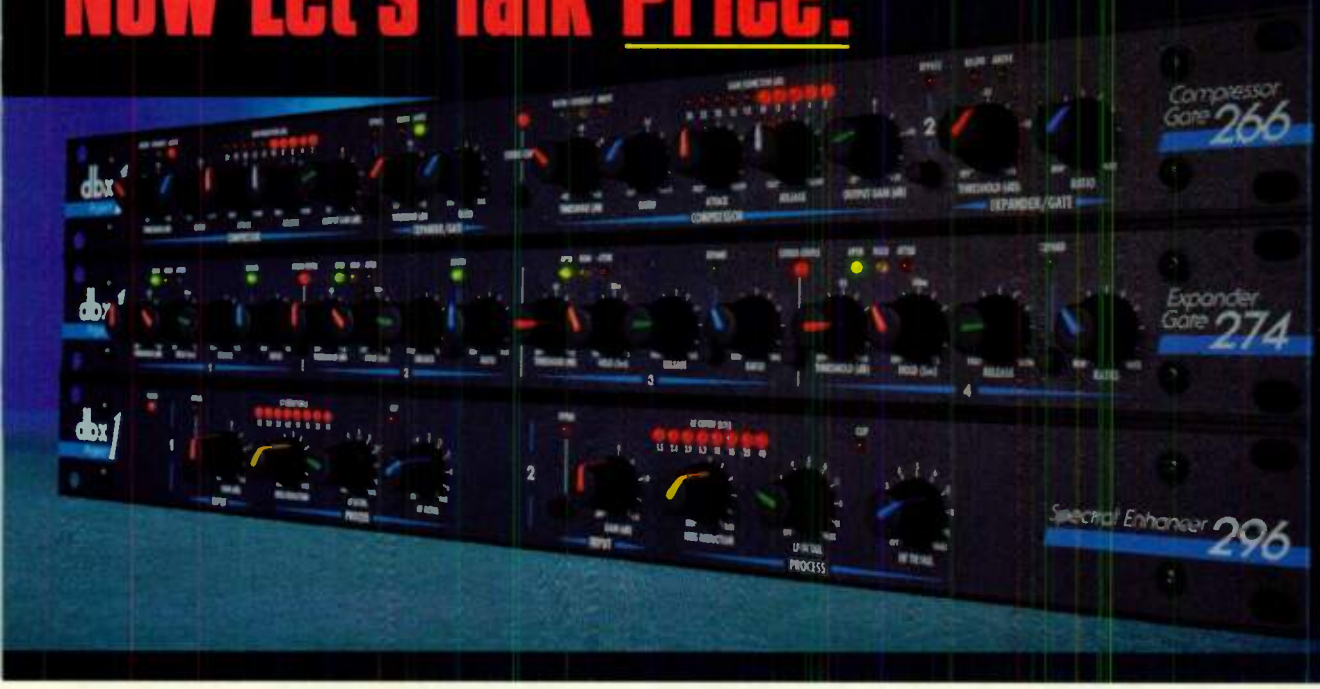

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not through either of the forward holes. If you look carefully, in bright light, you can probably detect telltale "swirl marks" from the screw. This is a secure mounting position, and the screw should be reinstalled to ensure mechanical stability.

JV-30 service tip: While replacing a damaged JV-30 top panel, I needed a substitute for the self-adhesive strips (which were temporarily out of stock) that hold the switch buttons in position. Roland's service department suggested the use of generic, double-sided carpet tape, *à la* K-Mart. This can be substituted, but it is slightly thicker and tedious to cut to fit, as it is many times too wide. It is probably best reserved for minor repairs, rather than instrument overhaul.

Q. I use a flexible-neck, halogen lamp—the kind with a wall wart—to illuminate my mixing console. The fixture holding the bulb appears to have provision for a cover or screen beneath the bulb, and I've heard that it is unsafe to operate the lamp without one. Is this true?

A. Halogen bulbs sometimes shatter when they fail, and the use of a protective glass cover over the bulb is essential. The lamp should be removed from service until this cover is installed. Contact the manufacturer for a replacement; if none is available, discard the lamp. When replacing halogen bulbs, do not let the bulb contact the skin. Contamination with body oil aggravates the bulb's tendency to shatter. If inadvertent contact is suspected, wipe the bulb with a tissue moistened with alcohol.

For such applications, where the bulb may be positioned close to the face and eyes, a conventional, "high-intensity" fixture is somewhat safer, and an old-fashioned incandescent, more so. Nonetheless, even conventional bulbs can shatter, and it is wise to cover the open end of the lamp fixture with wire mesh and keep the fixture oriented away from the body.

Q. In replacing a broken reed on my old Wurlitzer EP-200A electric piano, I sheared off the screw that holds it in place. How can I get it out?

A. Sometimes there is just enough shaft protruding at the top of a sheared screw to grab onto it with a pair of mini Vise Grips and back it out. If not, take the instrument to an auto repair shop or a machine shop. The sheared screw can be removed with a device called a

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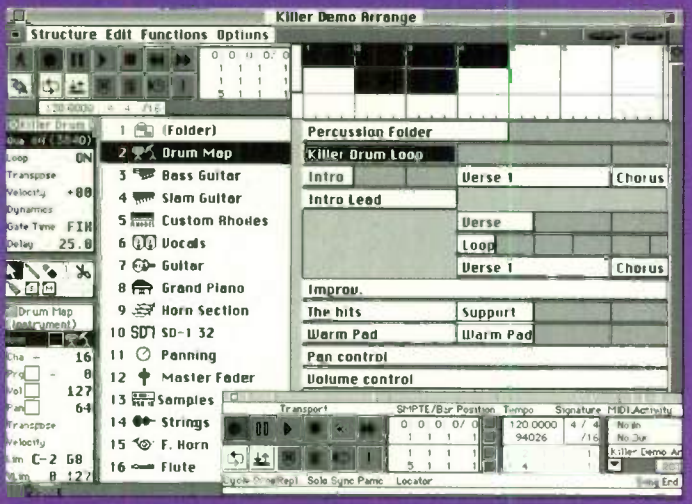

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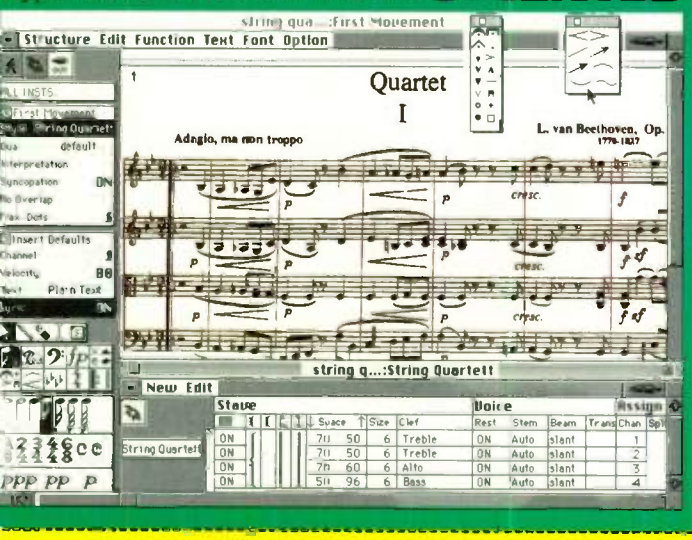

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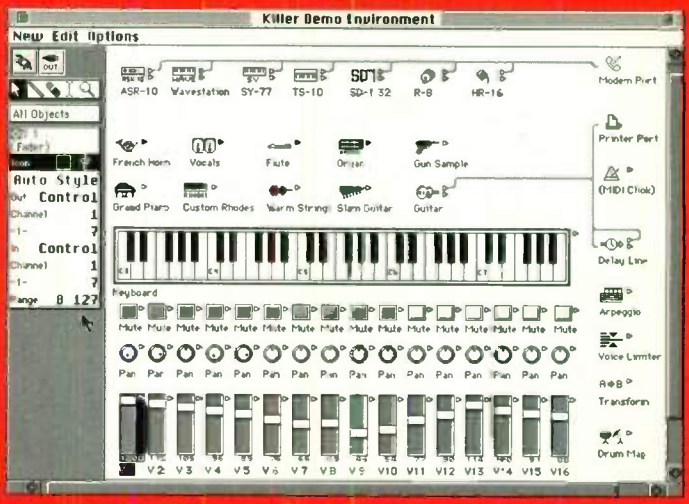

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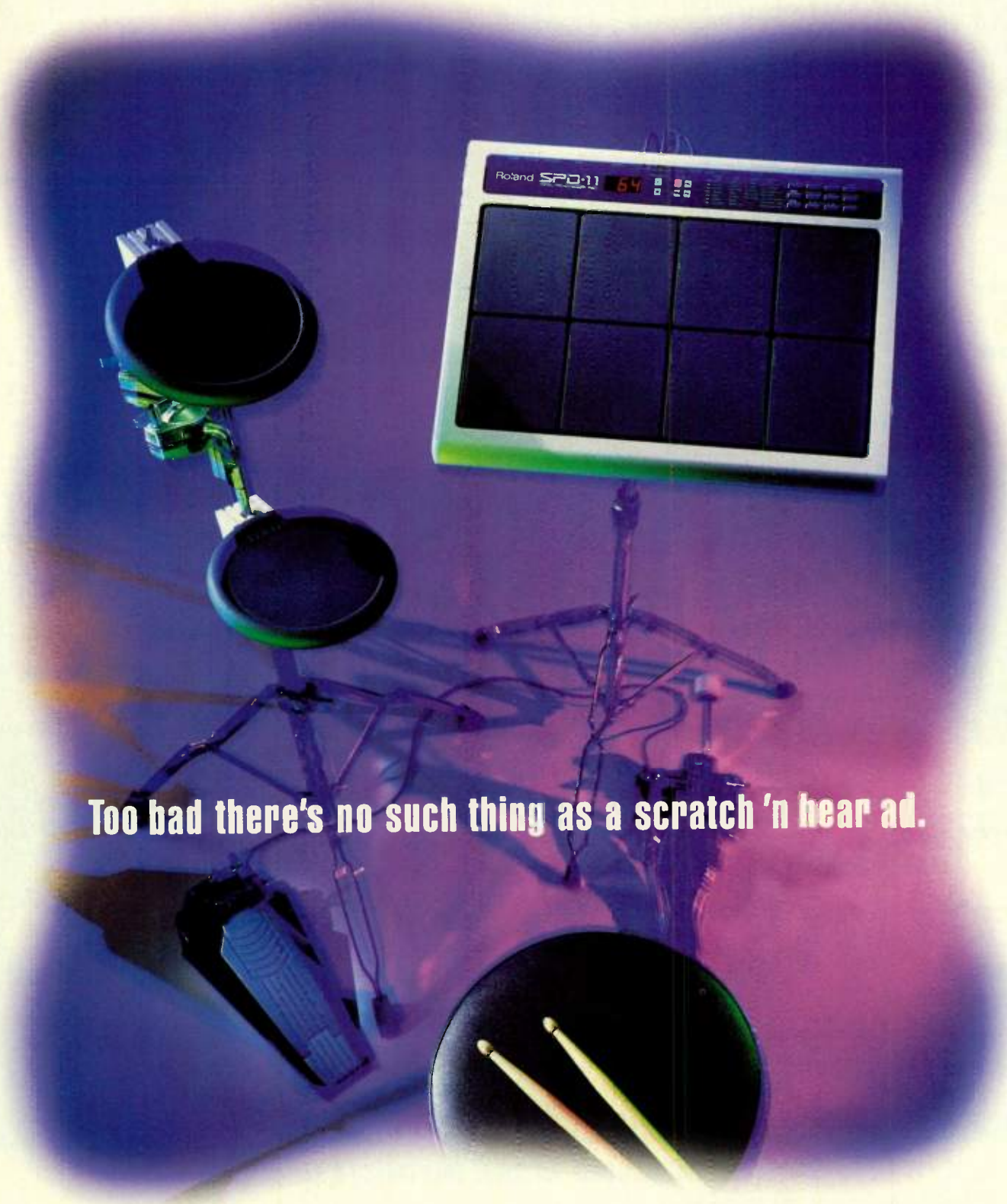

control plus cymbal chokes and rim shots. To try out the new SPD-11 Total Percussion Pad, visit your local Roland dealer. We'd say that in an ad this short, we can only begin to scratch the surface. But then we'd have to end with a pun. And that would be unfortunate.

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screw extractor. This requires drilling a pilot hole in the sheared screw, which obviously works best with larger screws, and damage to the threads may require that the hole be retapped for a slightly larger screw once the old one is removed. Attempting to drill a small screw may even fortuitously cause it to break up and clear the hole with minimal damage, so wear your lucky socks.

Q. I'm a guitar player with a cover trio, and I use an Ensoniq ESQ-1, played via its internal sequencer, for accompaniments on some songs. I would like to send a metronome or click to the drummer, but if I turn on the ESQ's internal metronome, it shows up in the main output. I'm pretty sure I saw a modification that relates to this in a friend's hobby magazine some years ago. Does this sound familiar?

A. The mod you're looking for is "DeWitt's Click," which reroutes the click signal to an external jack. (The mod also works with the SQ-80.) This was detailed in "Modifying the ESQ-1/SQ-80: 'DeWitt's Click,'" in the December 1988 **EM**. Back issues are available from the Mix Bookshelf; tel. (800) 233-9604 or (510) 653-3307; fax (510) 653-3609.

Some tips if you elect to install this mod: On the original, type 4001 ESQ mainboard, the signal is tapped from capacitor C54, which is located approximately 1¼ inches behind the CV Pedal jack. On the later-model, type 7001 ESQ mainboard (and on the SQ-80), the capacitor is designated C74; it is located next to 40-pin IC U27 and approximately four inches behind the Tape Input jack. Vacant rear-panel space is scant on these instruments, and the rear panels on later ESQs and the SQ-80 are difficult to drill. You can adapt an existing jack by lifting the circuit side of each ferrite bead associated with the selected jack and tack-soldering jumpers to the appropriate circuit points. The Sequencer Footswitch jack would be an ideal choice, because it uses the system digital ground, as does the click, so that only the signal side of the jack need be rewired. Further, containing the mod in the digital area of the board eliminates the need for shielded cable.

EM *contributing editor* **Alan Gary Campbell** *is owner of Musitech, a consulting firm specializing in electronic-music product design, service, and modification.*

The Open Music System

By Bob Lindstrom

Mac system software tools have matured, but will confusion reign?

There's a truism about technology: The more you have, the more you need. When Apple introduced *MIDI Manager* into the Macintosh system software several years ago, its port-sharing and interapplication communication capabilities promised a breakthrough in MIDI convenience. However, Mac MIDI hardware grew more elaborate, MIDI applications became more complex, and the performance limitations of *MIDI Manager* became more apparent, rendering Apple's solution inadequate.

Earlier this year, Opcode Systems announced plans to resolve the MIDI shortcomings in Apple system software by encouraging industry-wide support for their *Open Music System* version 2.0 (*OMS* 2.0). *OMS* 2.0 is an upgrade to the company's *Opcode MIDI System* (*OMS*), which adds extensive MIDI support to the Mac operating system (see "Computer Musician: OMS" in the March 1992 **EM**).

Just prior to the introduction of *OMS* 2.0, Mark of the Unicorn (MOTU) announced their own MIDI system software: the *FreeMIDI System*. Offering similar features, *FreeMIDI* represents an alternative effort to resolve many of the same Mac problems.

Mac music-software developers were forced to create their own MIDI system software because Apple offers no comprehensive, professional MIDI system solution. Unfortunately for users, software developers have each gone their own way with simple or complex solutions to port-sharing, timing services, and MIDI-system management. This means increased development costs for the developer (to provide software support that many believe Apple should have provided) and a babble of setup and control interfaces for the Mac musician.

THE BACKGROUND

To date, the most comprehensive approach to the problem has been the *OMS* software, already implemented in Opcode's *Vision* and *Studio Vision* sequencers, *MAX* software-construction kit, *Galaxy* librarian, and *Galaxy Plus Editors* and *Edit One* editor/librarians. *OMS* places an extra software interface between hardware and application software that manages ports and MIDI channels and maintains an ongoing record of your entire system configuration (see **Fig. 1**). Effectively, this additional level of integration between software and hardware turns your entire setup into one giant MIDI instrument. Rather than worrying about ports and channels, you just send MIDI data to the instrument you want to play.

OMS 2.0 enhances *OMS* with a number of additional features. If *OMS* 2.0 is to become a standard, one of the most important features is interapplication communication (IAC). With IAC, MIDI data can be piped directly from one application to another. This capability has far-reaching implications and uses, including, for example, the ability to link separate sequencers for simultaneous MIDI processing. You could move between Sequencer A and Sequencer B, using their unique features to process the same MIDI data. In addition, support for new technologies such as MIDI Machine Control could be added by using IAC to link a machine-control utility to your other MIDI software.

OMS 2.0 also promises to end 2-port tyranny. No longer will the Macintosh's modem and printer ports reign supreme. With serial-port independence in *OMS* 2.0, Macintosh musicians

DMITRY PANICH

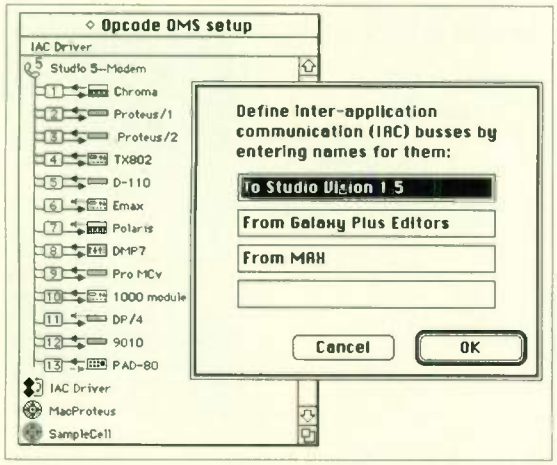

FIG. 1: Opcode's *Open Music System* 2.0 offers all of the studio description and routing capabilities of *OMS* 1.2 and adds inter-application communication, for sharing real-time MIDI data between programs, and system-level timing services.

could benefit from industry-wide support for devices such as multiport MIDI interfaces and Applied Engineering's QuadraLink 4-port NuBus card.

Finally, there are plans to add efficient, low-level timing services to *OMS* 2.0 that will outperform the less-than-satisfactory timing services in *MIDI Manager*. Although Opcode has made no official announcement at the time of this writing, the company is currently working with Steinberg/Jones to add those timing services to the *OMS* 2.0 release.

Of course, the *OMS* 2.0 upgrade will retain all the current features, including the system-setup description and real-time multitasking.

POWERBOOK PROBLEMS

Owners of Mac PowerBooks have already benefited from the *OMS* interface between software and hardware. Early PowerBook users experienced MIDI data-flow problems when using their portables. By modifying the *OMS* code, Opcode was the first vendor to make their products work with the PowerBook. Apple's solution came many months later. With *OMS* 2.0 as an industry standard, MIDI problems related to changes in the Mac system software would require only a modification to the *OMS* 2.0 code to maintain compatibility with all your MIDI software.

To move toward standardization, Opcode makes the source code available to participating developers at no charge (apparently in partial response to MOTU's similar offer with *FreeMIDI*) and plans to incorporate source code from other companies into the *OMS* 2.0 software. Since the National Association of Music Merchants (NAMM) show in January 1993, more than 25 software developers have announced support for *OMS* 2.0, including Steinberg/Jones, Passport Designs, Emagic, PG Music, Lone Wolf, Roland, and Digidesign.

"We're planning to support [*OMS* 2.0] with all of our Mac products," says Lowell "Banana" Levinger, director of development at Passport. "We'll use the timing and routing services that are being written for [*OMS* 2.0]. Much of the rest of it applies to universal editor/librarians like *Galaxy* and [MOTU's] *Unisyn*, which we don't do. I think it could all come together by NAMM next year."

"Almost all of our software is now *OMS*-compatible, or will be by the next update," explains Evan Brooks of Digidesign. "We deal peripherally with MIDI; our main focus is digital-audio editing. I look at *OMS* as a big performance improvement that will probably become a *de facto* standard."

MOTU'S FREEMIDI SYSTEM

In February of this year, Mark of the Unicorn (MOTU) published an overview of its *FreeMIDI System*, a specification for adding high-level MIDI capabilities to the Macintosh operating system.

From the beginning, MOTU conceived *FreeMIDI* as an industry-wide standard that would be available to other vendors at no charge. "We will make the source code available to any user, any developer at no charge," says Daniel Rose of MOTU. Any user? "Yes, if a user wants to become a registered *FreeMIDI* developer in order to create something within the system, all they have to do is write to us. The only legal requirement is that no one can charge for *FreeMIDI*."

Although Rose is reluctant to speculate on what the final release of *FreeMIDI* will contain, the 30-page overview (released at the January 1993 NAMM show) describes a system that goes beyond the current abilities of

OMS but is now matched in Opcode's *OMS* 2.0.

Like *OMS*, *FreeMIDI* permits the user to create a description of their MIDI setup in a Routing Manager, which is then accessible to all *FreeMIDI*-compatible software (see Fig. 2). *FreeMIDI* also eliminates the need for users to keep track of ports and channels. If you assign a track to the D-50 Slow Rotor, *FreeMIDI*'s Routing Manager communicates with the Port Manager and automatically sends the data to the correct port and channel.

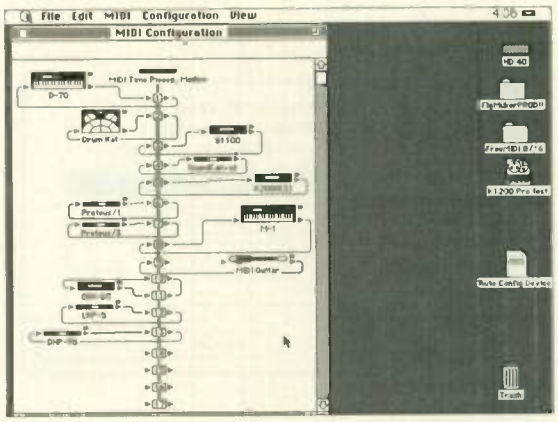

FIG. 2: Mark of the Unicorn's *FreeMIDI* offers an alternative to *OMS* 2.0 for sharing system information between MIDI programs and for universal timing services.

MOTU also plans to incorporate port sharing, IAC, and synchronization; extensive timing services through a Timebase Manager; and a Database Manager that allows editor/librarian software to provide information such as patch names to other *FreeMIDI*-equipped applications. Finally, a System Dispatcher will act as traffic cop for all data requests and data-flow through the *FreeMIDI System*.

Like *OMS* 2.0, *FreeMIDI* should improve software developers' ability to support a wide variety of hardware with a standard Application Programming Interface (API). Bugs related to hardware and system software can generally be fixed by altering the *FreeMIDI* code, instead of rewriting pieces of the application code.

Although the company is planning to put intricate technology into *FreeMIDI*, Rose believes that the "elegance of the system" will distinguish their approach. "We are planning a very elegant, intuitive system that will be attractive to musicians," Rose says. "We don't want a system that is computer-oriented and too technical."

FreeMIDI and *OMS* 2.0 are seen as rival technologies in some quarters, but Rose indicates that the systems might not be mutually exclusive. "We want to maintain as much compatibility as possible. That's part of what makes an open system. Anyone can do what they need to in order to use it." *FreeMIDI* is currently in development; a release date has not yet been announced.

APPLE'S MIDI MANAGER

Apple Computer's decision in the late 1980s to integrate MIDI functions into the Macintosh operating system was highly praised and anticipated. Once again, Apple was setting a precedent in system software that would later be imitated when Microsoft *Windows* and IBM's OS/2 received system-level MIDI support.

Unfortunately, when Apple's solution appeared in the form of *MIDI Manager*, the results were somewhat less thrilling than the expectation. *MIDI Manager* did provide *PatchBay* for IAC (within those programs that supported *MIDI Manager*) and integral timing services (see Fig. 3). However, serial-port support was limited, and, most troublesome of all, the processing overhead made

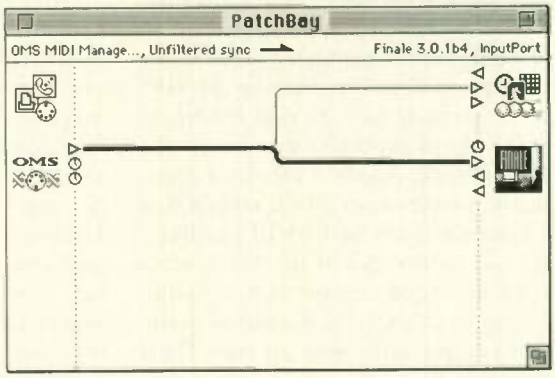

FIG. 3: The *PatchBay* program, which is part of *MIDI Manager* 2.02, lets you make simple routing connections between active MIDI applications, but *MIDI Manager* requires a great deal of processing time and lacks built-in support for multiport MIDI interfaces.

The Top 500

Play the D4 with its onboard trigger inputs.

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2	NEW ▶		STUDIO TOM / BRAND NEW D4	16" MAPLE TOM w/VERB
3	NEW ▶		BIG "O" / BRAND NEW D4	DOUBLE HEAD KICK w/VERB
4	5	52	RIM SHOT ROOM / SR-16	BRASS PICCOLO w/VERB
5	10	156	BIG FOOT / HR-16:B	SINGLE HEAD 26" MAPLE
6	NEW ▶		SLAM / BRAND NEW D4	POWER TOM w/VERB
7	23	156	COMBO SNARE / HR-16:B	PICCOLO PLUS WOOD
8	NEW ▶		BIG BALLAD / BRAND NEW D4	WOOD SNARE w/BIG VERB
9	NEW ▶		FAT CITY / BRAND NEW D4	SUPER FAT SNARE
				ARTICULATED HI HAT

THIS WEEK	LAST WEEK	WKS. ON CHART	TITLE / ARTIST	
302	NEW ▶		RAW HIDE / BRAND NEW D4	CUSTOM SNA
303	450	52	HI ROOM TOM / SR-16	10" MAPLE
304	NEW ▶		WET HALF / BRAND NEW D4	HALF OPEN HAT
305	327	52	RIM 2 CENTER / SR-16	ARTICU
306	123	208	DOUBLE HEAD	DOUBLE HEAD KICK
307	223	156		
308	401	5		
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the software ungainly on lower-powered Macs.

"*MIDI Manager* performance and architecture were not satisfactory," notes Digidesign's Evan Brooks. "The architecture was the inverse of what Opcode has created: *MIDI Manager* fixed the number of ports available and left it up to the user to make the connection. *OMS* 2.0 tries to make much of that invisible."

As the Apple product line expanded, and MIDI incompatibilities developed between old software and new hardware, *MIDI Manager* was not updated rapidly enough to address newfound problems. "PowerBooks could send MIDI data to *MIDI Manager*, and *MIDI Manager* could send it out with no problem," explains Aaron Hyde, Apple's MIDI evangelist. "But there was a problem receiving MIDI data from an external source."

According to Hyde, that problem has been resolved in a new version of *MIDI Manager* (2.02) that is now available. "Those are the only plans we currently have for *MIDI Manager*," explains Hyde, "but we are always looking at ways to improve the software."

▼

With inter-application communication, MIDI data can be piped directly from one application to another.

In discussing the future of *MIDI Manager* with *OMS* 2.0 supporters, however, I found little optimism regarding Apple's MIDI support. "We just don't think about something that's never going to happen," says Brooks about the possibility of a major overhaul of *MIDI Manager*.

An unofficial source from within Apple suggests that the size of the MIDI market does not justify the development costs that would be required for Apple to pursue *MIDI Manager* support beyond addressing bug fixes and compatibility problems.

At present, *MIDI Manager* appears destined to be the domain of entry- and mid-level MIDI sequencers on the Mac, with *OMS* 2.0 and *FreeMIDI* providing the growth path for high-level applications.

THE PARTING SHOT

"Our intention is not to replace *MIDI Manager*," says Paul de Benedictis of Opcode. "We believe that many low-level or entry-level programs will still use *MIDI Manager* exclusively. But we feel that the adoption of the *Open Music System* is a really important step for high-level Macintosh MIDI products."

Of course, MOTU feels similarly with regard to *FreeMIDI*. The real question is how the users feel. Let's hope complete confusion doesn't reign supreme.

Bob Lindstrom *is a freelance writer who owns one of just about every major computer type available.*

Reviews

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Ensoniq TS-10 Synthesizer

By Geary Yelton

An American synthesizer with world-class sound.

When a major synth-maker announces a "flagship" keyboard workstation, it had better be a serious, up-to-date, stage and studio tool. It should do just about anything any modern synth can do. Armed with hundreds of sounds you may have heard at a nightclub or on the radio, the Ensoniq TS-10 competes successfully with other synthesizers in its price range.

Much about the TS-10 resembles other Ensoniq instruments, but with enhancements. The 32-voice, 12-part multitimbral, wavetable synth includes onboard effects; a 24-track sequencer; and a 3.5-inch, high-density, floppy-disk drive. If you've ever used a VFX or EPS, you'll feel right at home with the front-panel layout. The 61-key, Velocity- and Poly Aftertouch-sensitive keyboard, the two Patch Select buttons for instantly accessing four variations within each Program, the recessed disk well—they're all there. The large, fluorescent display goes all the way back to the ESQ-1. The 30,000-note (expandable to 97,000) sequencer is similar to the sequencer in Ensoniq's previous top-of-the-line synthesizer, the SD-1. The onboard stereo effects, no longer a luxury on any serious synthesizer, are better than ever.

Though related to the SD-1, the TS-10 offers a few new twists. The most significant is that although it can't sample, the TS-10 lets you use and edit samples from Ensoniq's ASR-10, EPS, and EPS-16 Plus samplers. Two megabytes of sample RAM come standard, which allows you to load up to ten RAM-based sounds. You can expand the memory to 8 MB using easily installed, 30-pin, Macintosh SIMMs (though installing them yourself voids the warranty), which lets you load up to twenty sounds.

To create the TS-10 synthesizer, Ensoniq combined and enhanced the synthesis architecture and sequencer of the SD-1 and the effects processing of the ASR-10, then added a new Hyper-Wave wave-sequencing feature and the ability to import Ensoniq samples.

ARCHITECTURE

Each TS-10 Voice consists of a single oscillator, two multimode filters, three hardware envelopes, an LFO, and modulation routings (see **Fig. 1**). Each Program is made up of as many as six independently editable Voices (including samples) and an effect. There are five BankSets, each of which contain ten Banks of six Programs, for a total of 300 available programs. Of these, 180 are in ROM and 120 are user-programmable. In addition, there are 180 Performance Presets in ROM and 120 user-programmable Presets.

The two digital filters are not resonant, which may be a problem for some die-hard synthesists. (The effects processor provides a resonant filter with an envelope as part of a multi-effects algorithm, but it's not part of the main synthesis architecture, so it's not as useful as a regular resonant filter.) The first filter is always lowpass, but the second filter can be highpass, allowing

you to configure the two filters as a single bandpass filter. The 6-stage envelope generators can track the keyboard, and the attack and release times and levels can respond to Velocity.

Interestingly, if you use samples for the oscillator, the voice architecture changes to one that's identical to that found in the company's ASR-10 sampler. The envelope generators for sampled sounds allow you to define two envelopes and interpolate between them depending on Velocity. Unfortunately, you can't use these types of envelopes with the instrument's 6 MB of ROM waveforms. On the other hand, the 254 16-bit ROM waveforms can be controlled through a modulation mixer/shaper which can't be applied to samples.

A delightfully huge assortment (32) of alternate pitch tables is built into ROM, including historical temperaments and tunings from Arabia, China, India, and Java, plus three invented by Wendy Carlos, and Harry Partch's 43-tone-per-octave scale. Each Program can have a user-defined pitch table. You can instantly switch between pitch tables in performance without changing programs, making it possible to really "play between the cracks" without using pitch bend.

TRACKS AND PRESETS

A Track consists of a sound Program (which may use ROM waveforms, samples, or a combination of the two), a MIDI-channel assignment, and various performance parameters, such as controller settings, mix volume, key zone, and transposition. Up to three Programs may be instantly layered by double-clicking the Programs you wish to add. The MIDI channel, send/receive status, and other parameters are defined for each Track. A Performance Preset is a combination of up to three Tracks, which may be layered or split, along with an effect (which defeats the effects that are assigned to individual Tracks) and a host of Track-specific parameters.

In Multi mode, you can play Programs and Presets from the keyboard, independent of Tracks played by an external source. This means you can actually hear up to fifteen Programs simultaneously. If you assign Tracks to receive external MIDI messages, you can also use an external controller to overdub or perform with TS-10 sounds while the sequencer is playing.

This complex hierarchy of sounds and Tracks is typical of high-end Ensoniq instruments. It may seem confusing, but it's ingenious once you get used to it. Unfortunately, getting used to it may take awhile.

SEQUENCING

The 96 ppqn sequencer is so similar to the one in the SD-1 that, rather than describe it in detail, I'll refer you to the SD-1 review (November 1991 EM). Suffice to say that it's one of the best sequencers I've seen on a keyboard workstation—much better than the Ensoniq ASR-10 sampler's sequencer—and includes such features as swing, percentage, and random quantize. TS-10 features that aren't found on the SD-1 include the ability to send external controllers to other instruments; brightness and attack parameters; fine detune; LFO rate offset; Velocity splits; and a global tempo track.

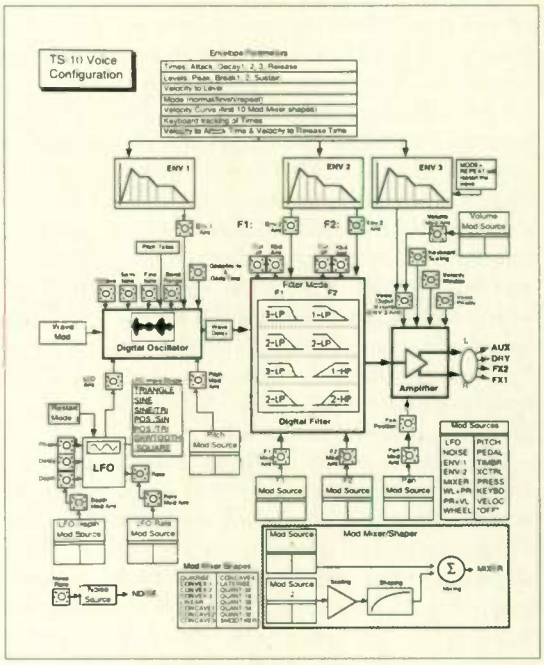

FIG. 1: The Ensoniq TS-10's sophisticated voice architecture is an expanded version of the SD-1's structure. Note the Mod Mixer/Shaper (bottom), which lets you combine, scale, and shape modulation-source waveforms.

In Multi mode, sequencer Tracks respond to Program Changes. In all other modes, you select a BankSet with a Bank Select message, and Program Changes 0 to 59 change Programs, while Program Changes 60 to 119 change Performance Presets. With a little planning, this allows a good bit of on-the-fly flexibility.

EXPLORING THE SOUNDS

When I first received the instrument, I spent more than a week just checking out its sounds. At first, I didn't hear much that really blew me away. There are some cool sounds, to be sure, but I'd heard variations on most of them before. With a little more exploration, I began to appreciate more fully the complexity of the instrument's sonic architecture and the completeness of its timbral palette.

For my first project, I decided to rearrange a song I'd sequenced on my computer over a year ago. I harnessed the TS-10 as the only sound source, modifying its programs as necessary. I kept it simple, using ROM-based sounds and the default effects settings. All the sounds I needed were there, even the D-50's Fantasia patch and a DX7 bass. I discovered that switching a Track's effects algorithm in the middle of a sequence isn't a good idea, as this causes the audio output to pause for a moment. But other than that, everything went smoothly, with no unpleasant surprises.

For my next project, I wanted to create new sounds with custom effects. This meant really jumping into the belly of the beast and learning to program from scratch. The Wave button revealed the 254 waveforms, arranged into sixteen categories, called "Wave Classes." These include sampled keyboards, strings, horns/brass, winds/reeds, vocals, bass, drums, cymbals, percussion, tuned percussion, and sound effects, plus single-cycle waveforms, inharmonic loops, and Transwaves (tables of up to 100 single-cycle waveforms of varying harmonic spectra, which can be swept with a modulation source).

The keyboard sounds feature Rhodes and Wurlitzer electric pianos, organs, celeste, harpsichord, three very good analog (perhaps Moog or Sequential) synthesizers, and a mediocre grand piano. Strings include guitars, harp, string ensemble, solo violin, and cello.

Ethnic stringed instruments include biwa, goto, shamisen, and dulcimer. There are also eight sound effects, none of them especially useful. In the basic waveform category, there are 38 waves, ranging from sine waves and sawtooth to tuned bells and synthesized organ. Poorly looped noise is one of the inharmonic waves.

The majority of sound Programs are good to excellent. I was especially impressed with the third-world instruments; the sitar is probably the best I've ever heard on a synthesizer. Other standouts include the steel-string guitar and most of the drums. On listening to the raw waves, I noticed a few with loops that created strange harmonics. Fortunately, the programming and effects processing does a good job of hiding the occasional blemishes.

The TS-10 supports Bank Select (MIDI Controllers 0 and 32) commands, allowing you to select any ROM Programs, sampled sound, or Preset, with or without effects, via MIDI Program Change commands. Without Bank Select, no more than 128 Programs could be selected by MIDI messages, so this is a welcome feature.

HYPER-WAVES

Hyper-Wave is Ensoniq's version of the Wave Sequencing found on the Korg Wavestation. A "Wave List" plays a series of steps in the order you specify. Each step has its own waveform, duration, tuning, relative volume, and crossfade time. Steps can be played forward or backward, and keyboard tracking can be turned on or off. Step duration and crossfade time can be up to a minute long. Because each step can be tuned, you can set up a list that plays a melodic sequence when you hold down a key. The Hyper-Wave feature is most often used to create percussion patterns or sweep through sounds to invoke exotic, timbre-shifting pads.

You can define how many steps are in the list; which step is the first to play; and the Loopstart and End points. When the list has been played to the end, it returns to the Loopstart point and repeatedly plays through the steps to the End point as long as the note is sustained. Modulation can be used to change the start step, the Loopstart step, or the End step. If "Traveler" is selected, the modulation source directly controls which individual step plays, allowing you to simulate vector synthesis. You can manually change the wave list tempo in the Track Parameters section, but not the durations of the individual steps. Unlike the Wavestation, you can't sync tempo to an external clock.

The maximum number of steps is sixteen, compared to the Wavestation's 256, but unlike the Wavestation, each program can store its own Hyper-Wave. In addition, each Voice in a program can play any portion of the entire wave list with different Start Steps, loop points, and modulation assignments. By layering voices, you can get differing polyphonic parts to play from the same list. This allows you to do things like have a single wave list play both bass and drums. Each cross-faded step requires two Voices to allow for the step faded in between, but the polyphony for other sounds is not affected, so polyphony is significantly reduced only if you use a lot of Hyper-Waves with crossfaded steps.

When I tried to create my own Hyper-Wave list, I found the manual's explanation vague and incomplete. I called Ensoniq and was told they were working on a separate programmer's manual for the TS-10.

I wish there were more Hyper-Wave-based programs among the TS-10's onboard sounds. There are some really nice ones, such as the percussion in "Genesis," but Ensoniq programmers would have done well to concentrate more on the possibilities inherent in Hyper-Waves. My greatest disappointment, however, is that you can't include sampled sounds in wave lists, which might have opened up some incredible horizons.

SAMPLES

The most radical feature of the TS-10 is its ability to read EPS, EPS-16 Plus, and ASR-10 sample disks. All the parameters are retained (loops, key maps, envelopes, etc.). Because the effects processors are of slightly different designs, effects parameters are converted to the most similar algorithm. If a sample has no effects, mild reverb is added, which can be annoying. Although samples can't be saved in the TS-10's internal memory, the synth can store edited parameter information to disk. It's also possible to assign a substitute sound to be used in a Track when the sampled sound is absent or muted. Using sampled sounds limits the polyphony to 31 voices, however.

A big limitation is you can only use samples from Ensoniq machines. True, Ensoniq's sample library is extensive and of good quality. But it's a most unfortunate limitation for the many musicians who are committed to non-Ensoniq samplers. I'd like to see the TS-10 import, say, Akai S1000, Roland S-770, Peavey SX, and/or Kurzweil K2000 samples. If the TS-10 supported the SMDI protocol, Macintosh users with other samplers could pull in samples from Passport's *Alchemy*. Ensoniq says they plan to support other sampler formats in the future.

For $250 more, you can buy the optional SP-4 SCSI interface to read sampled sounds from CD-ROMs and hard disks. The SCSI interface is read-only, however, so you still have to save your parameter edits to floppy disk.

EFFECTS PROCESSOR

Being a signal-processing kind of guy, I had an abundance of fun playing with the effects. It would be a better world if every synthesizer had an effects processor like this one. The reverbs are sparkling and smooth. Its flexibility and sound rival many outboard effects processors, and most of the effects are clean and quiet.

The TS-10's processor uses the same DSP chip as Ensoniq's DP/4 (which has four of them) and ASR-10, but with all-new algorithms. Effects parameters are part of every Program, Preset, sequence, and song, even if none of the voices are routed through the effect. There are 73 well-chosen effects algorithms, including delay, various reverbs,

Product Summary

PRODUCT:
TS-10 Performance/Composition Synthesizer
PRICE:
$2,595
MANUFACTURER:
Ensoniq Corp.
155 Great Valley Pkwy.
Malvern, PA 19355
tel. (215) 647-3930
fax (215) 647-8908

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ●
SOUND QUALITY	● ● ● ●
VALUE	● ● ●

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chorus, flanger, rotary speaker, parametric EQ, distortion, pitch shifter, compressors, and guitar amp simulation. Many algorithms include three simultaneous effects, such as delay+flanger+reverb, but you can't alter the order. So you don't have to spend time tailoring the effects to your needs, there are 692 pre-programmed parameter variations on the 73 algorithms, including some that change the effects routings (e.g., parallel vs. serial).

Any two effects parameters can be modulated simultaneously, in real time, using the mod wheel, Velocity, Aftertouch, a CV pedal, a footswitch, the data-entry slider, or any MIDI Control Change message. Which parameters are available depends on the algorithm. Most algorithms are extensively programmable; Large Plate Reverb 2, for example, offers control over decay time, pre-delay time, high-frequency damping, bandwidth, stereo balance, definition (echo density over time), two types of diffusion in series, early reflection levels, spread (time intervals between seven discrete echoes), and shape (relative volumes of these echoes). Even the guitar-amp simulations have nineteen adjustable parameters. You should be able to adapt any effect to your liking.

FINAL JUDGMENT

If you're looking for a synthesizer that plays samples from disk *and* lets you sequence ROM-based waveforms, the TS-10 is the only game in town. These aren't new ideas, but they're well implemented, and this is the first time they've been combined on one instrument. If only you could use your own samples in Wave Lists! Wave Sequencing on the Wavestation has had an undeniable impact on the sound of electronic music, and I'm sure Ensoniq users will find new ways to create fresh textures with Hyper-Waves.

As an instrument for live performance, the TS-10 shares the advantages of other Ensoniq keyboards. Once you've grown accustomed to its operation, using it onstage becomes second nature. The Patch Select buttons and Poly Aftertouch offer expressiveness lacking on most keyboard instruments. The ample selection of preprogrammed effects variations and the ability to modulate any two effects make the effects a seamless element in every sound's architecture.

No instrument is perfect. At the least, I wish the TS-10 had resonant filters and more than 6 MB of waveforms in ROM. (There is a slot for 512 Kword of additional waveform ROM, which can be added later.) If there were more waveform memory, the sampling rate could be higher, the loops could be longer, and there could be more multisampling, resulting in better sound. On the other hand, most of its competitors don't read sampler disks. Those that do offer less sample RAM standard and don't keep all the sampler parameters intact.

There's a lot to recommend the TS-10. With its flexible sequencer and high-quality effects, it excels as a standalone workstation. It makes a few unique sounds, and except for its piano programs and a few others, its imitative voices are pretty much top-notch. (As much as I hate sounding like everyone else, I admit most of us need to work with traditional sounds.) If you're trying to pick from among the current crop of wavetable synthesizers with tried-and-true presets and exten-

sive programmability, the TS-10 may be just what you want.

Geary Yelton *is beginning to show signs of sensory overload from the constant parade of exciting products that his life revolves around. Nonetheless, he may be happier than he's been in years.*

Circle # 437 on Reader Service Card

Tascam Porta 07 Cassette Ministudio

By Jim Pierson-Perry

If you ever wanted to get into multitrack recording, do it now.

Few companies have contributed as much as Tascam to the democratization of multitrack recording. Their extensive line of cassette ministudios—offering varied features at prices that fit practically any budget—allow musicians to quickly sketch out ideas or record complete arrangements with a minimum of fuss.

Tascam's latest entry-level ministudio is the Porta 07, which pairs a 4-channel mixer with a 4-track cassette recorder. The Porta 07 sports a clean design with logically placed, color-coded controls. First-timers can dive right in, and there's plenty of capability for more experienced users. Its low weight (approximately three pounds), slim design, and sturdy construction make it well-suited in cramped home studios or on the go. This is a serious notebook studio.

The Tascam Porta 07 includes a 4-track cassette deck, 4 x 2 mixer, dbx II noise reduction, and excellent sound.

MIXING

The Porta 07 mixer is a 4 × 2 configuration: Four input channels and two stereo outputs. Each of the four mixer channels includes an unbalanced, ¼-inch, mic/line input on the front panel. A pair of RCA Sub In jacks mix an external stereo signal onto the left and right buses, a handy way to get more inputs into your recording. Channels 1 and 2 accept mic or line inputs and have input-level trim sliders. Channels 3 and 4 are for line-level use only. Separate pairs of RCA jacks provide independent Monitor and Line Out signals, both at -10 dBV. There's also a ¼-inch, stereo headphone jack.

Each channel has a dedicated fader with a 4.5-inch throw, 3-position signal-select switch, and rotary knobs for EQ, pan, and effects-send level. A separate fader handles the master level. The signal-select switches determine the input to each channel: mic/line jack, tape track, or off. When using tape tracks as the signal source, tracks 1 through 4 play through the corresponding channel numbers. EQ consists of shelving filters fixed at 100 Hz and 10 kHz, both ±10 dB. The audio paths are clean, and I heard minimal crosstalk.

Mixer channels are routed to tape tracks through their respective pan pots and two tape-track select switches. One switch selects tape track 1 or 3 (on the left bus), and the other selects track 2 or 4 (on the right bus). The pan pots assign signals between the two selected tracks. Both of these switches also include a "safe" position to prevent signals from getting accidently recorded onto either track. With this design, up to two tracks can be recorded simultaneously: one from the left bus and one from the right bus. Any mixer channel can be routed to any tape track. This is a flexible system that accommodates recording single channels to individual tracks, multiple channels to a track (e.g., bouncing several tape tracks to one), or routing all inputs to a stereo mix.

Four, 10-segment LEDs provide bar-graph meters of the audio levels. These are scaled from -10 to +6 dBV and can be set to display the signal levels of the tape tracks or the L/R

buses and effects send. Activating a track for recording lets you monitor levels before rolling tape. Unfortunately, there are no clipping indicators, so you'll have to trust your ears to avoid distortion.

The Porta 07 offers good monitoring capabilities, letting you switch between Cue, Remix, and Effects modes. Cue mode monitors the current channel inputs and any active tape tracks, with individual rotary dials to adjust tape-track volumes in the mix. Remix mode presents the left and right bus signals as they feed into the Line Out jacks. Effects mode lets you hear the combined effects-send signal from each channel going to the Effects Send jack. A separate rotary dial controls the signal level going into the headphone and Monitor Out jacks.

The effects loop provides a mono send and stereo return, all on ¼-inch jacks. The channel effects send is post-fader and post-EQ. The effects-return jacks can be used in a pinch as a pair of line-level inputs. These feed into the left and right buses, before the master fader. A single rotary control regulates the volume of both return lines.

RECORDING

The Porta 07 is designed to use high-bias, chrome (Type II) tape and cannot be adjusted to work with standard or metal tape. Tascam recommends using the shortest tape length necessary for any given project. C-120 cassettes cannot be used, as the tape is too thin for multitrack use. These are basic caveats for most of today's cassette ministudios, not a specific limitation of the Porta 07.

The transport operates at 3¾ ips, which is twice the speed of normal stereo cassette players. As a result, you get only one-quarter of the nominal cassette time; for example, C-60 tapes provide fifteen minutes of multitracking. The unit records all four tracks in one direction. (You can't record on two sides of the tape, as with a standard cassette deck.) A variable pitch control allows you to adjust the transport speed by ±12%.

The transport controls are comfortably thumb-size, and the transport worked smoothly. The unit has a mechanical 3-digit tape counter with manual reset and Return-To-Zero. In addition to the LED meters, each track has a dedicated Record Ready light.

The Porta 07 supports punch in/out either manually from the transport controls, or via a remote, ¼-inch footswitch. (Those of us with two or fewer hands will prefer the footswitch.) In practice, punching in worked without a hitch; it was clean, with no detectable gap in the recording. Punching out usually caused a gap of about one second, which is typical for ministudios. Tascam sells a remote footswitch, but my synth footswitch worked just fine.

The Porta 07 offers built-in dbx II noise reduction. You can't defeat the dbx on the sync track, which is usually recommended when recording a sync tone. Instead, the Porta 07 offers a bandpass filter on track 4 that is designed to compensate for the dbx processing. The Porta 07 provides an RCA Sync Out jack that is hard-wired to the output of track 4. There is no dedicated Sync In port; you must record the sync signal through one of the mixer channels. I successfully recorded a SMPTE sync tone onto track 4 from a MIDI Time Piece, then used it to lock my software sequencer.

Overall playback quality is impressive, although high-end sounds such as cymbals lost a bit of sparkle. Tascam claims a frequency response of 40 Hz to 16 kHz and a signal-to-noise ratio of 85 dB (using noise reduction). Overall, the Porta 07 sounds great and more than holds its own against competitors in its class.

BOTTOM LINE

During my evaluation of the Porta 07, I came to appreciate its overall design and ease of use. It was very simple to

Product Summary

PRODUCT:
Porta 07 cassette ministudio
PRICE:
$449
MANUFACTURER:
Tascam
7733 Telegraph Rd.
Montebello, CA 90640
tel. (213) 726-0303
fax (213) 727-7635

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ◖
EASE OF USE	● ● ● ● ◖
SOUND QUALITY	● ● ● ●
VALUE	● ● ● ●

jump into a recording session without worrying about a lot of setup. The controls are logically laid out and easy to operate, even by "sausage-thumbed" users. It competes well on features against similar ministudios and is priced to move.

I tend to anthropomorphize gear, especially when I use it for creative pursuits. To me, the Porta 07 has a peppy "can-do" attitude and lives up to it. Anyone looking to get started in multitrack recording or needing an inexpensive recording sketchpad should definitely give it a try.

Jim Pierson-Perry *is a non-profit organization dedicated to the care and feeding of a Macintosh computer system, a music studio, and seven cats.*

Circle # 438 on Reader Service Card

Alesis BRC Master Remote Control

By Neal Brighton and Buddy Saleman

The ADAT's big "brain" is a comprehensive control center.

Serious recording engineers will attest that an autolocator is critical for maintaining one's sanity when working on a number of musical selections and cues. Basic autolocators set up location points for songs and song sections (verse 1, chorus 1, bridge, etc.) and help the engineer keep track of overdubs and punch-in points.

Alesis' autolocator for its ADAT system, however, is a mammoth, multifaceted nerve center. Perhaps that's why it's called the BRC, or Big Remote Control. The BRC offers transport control of up to sixteen ADATs; record enable/disable for up to 128 tracks; chase-lock and sync capabilities for sequencers, video decks, and analog recorders; and routing control for the ADAT's digital busing system.

In essence, the BRC can replace two or three boxes in your studio, while giving you all of the features expected from a high-end autolocator. On top of that, all of the information loaded into the BRC—song titles, cue locations, MIDI information, etc.—can be stored in the data field at the beginning of your ADAT tape. As an excited Hollywood agent might say, "This BRC is big, big, big!"

START UP

Before you can take advantage of the BRC's myriad features, your ADAT must have operating-system version 3.03 or higher. ADATs with earlier versions will not communicate properly with the BRC. To check your ADAT's operating system, push the Set Locate and Fast Forward buttons simultaneously; the LED screen will display the currently installed OS version. If you don't have the required system, call Alesis' service department for more information. Depending on the version, you will be sent a user-installable chip, or be instructed to return your ADAT to the factory for a hardware and software upgrade.

THE WAY OF HOW

The BRC's rear-panel connections

reveal a lot about the unit's versatility. For interfacing with the video-production world, there are standard BNC connectors for Video Sync In, 48 kHz (Sample Clock) In, and 48 kHz (Sample Clock) Out. A Remote Out To ADAT, 9-pin D-Sub connection links the BRC to its slave ADATs. Footswitch connections for Punch In/Out and Locate/Play are handled with ¼-inch jacks. SMPTE In and SMPTE Out connections are also on ¼-inch jacks. Finally, there are MIDI In and MIDI Out connections, the receptacle for the 90 to 250 VAC power cord, and the power on/off switch.

The remote controller is laid out for quick and intuitive operation. All controls are clearly labeled with function descriptions such as Copy Song, Save Setup To Tape, Auto-Punch, and so on. The function buttons are so well-documented that you can get pretty cozy with the BRC's features by simply reading the front-panel controls.

The left half of the BRC controls Record and Input monitor functions for up to sixteen ADATs. LED status lights inform you which tracks are armed. The large, rugged transport controls are positioned at the bottom right of the unit. An amber, 32-character LCD screen and alphanumeric keypad are just above the transport controls, surrounded by more function buttons. The console's plastic sides are removable so the BRC can be mounted in a standard, 19-inch rack.

The Alesis BRC offers ultimate autolocation and sync control over an ADAT system.

The BRC is not exactly a petite device. If you rack-mount this beauty, you'll need a big rack, because the unit eats up six spaces. Its size may impose a few logistical problems in a cramped studio. We have a pretty big control room and still had a hard time figuring out where to put the unit.

The BRC doesn't override the LRC (Little Remote Control) included with the ADAT, which is a great advantage. Up to two LRCs can work in tandem with the BRC. This means that you can establish a main control station near your mixer with the BRC, and two satellite stations—perhaps one by your computer and another next to your keyboard rig—with the LRCs. Of course, the LRC has limited functions compared to the BRC, and its transport buttons are far from great, but the ability to control basic functions from three important creative stations is an obvious boon to the personal recordist.

FUNCTIONAL REALITIES

One of our main concerns was whether the BRC meets the needs of the average personal or project studio. One valuable feature we found immediately was the BRC's ability to store twenty songs and twenty location points within a designated song. Using the alphanumeric keypad, you can enter short names for each song and brief descriptions of specific location points. The BRC also provides preset song-structure names—verse, chorus, bridge, etc.—that can be used to mark location points.

Obviously, it takes a substantial amount of time to enter all this information, but the toil is worthwhile in the long run. Once everything is entered, you can tool around an entire album's worth of songs with a couple of button pushes and know *exactly* where you are. In addition, you need to enter location points to use some other BRC features such as Loop and Auto-Punch. Saving all your carefully input data is as easy as pushing the Save Setup To Tape button. Your S-VHS master is immediately rewound to its beginning, and all location information is recorded onto the tape's data section.

RECORDING TRICKS

The BRC can make its slave ADATs do some nice tricks. One of these is Continuous Play, which allows one ADAT to

start recording (or playing) just before another stops. Multiple ADATs can be staggered to offer almost unlimited recording time. For live recording, where the 40-minute limit of the ADAT's S-VHS tape can be problematic, the advantages of Continuous Play are obvious. You also can record opera or theatrical productions in a studio environment *non-stop*. In addition, the function allows premastering for CD duplication—a major service our studio offers—when the audio material exceeds 40 minutes.

The automatic looping function also proved invaluable. This feature allows you to loop between two location points to rehearse or develop a part. However, the value of other functions depends on your situation. For example, the Auto Punch In/Out is superfluous in a project studio where an engineer is running the deck. To the solo recordist, however, this feature can be a lifesaver. Likewise, Rehearsal mode is of little value in pro studios, but less-adept recordists might enjoy the ability to audition a tight punch.

During recording sessions, we found setting pre- and post-rolls a waste of time. It's quicker to simply set your locate points a bar or so before each song section. After all, while overdubbing, you don't want the tape to start right on a chorus or verse. (In sections requiring multiple punch-ins, it proved easier to abandon locate points altogether and manually rewind a few seconds past the desired punch-in point.) Pre-roll is necessary only during digital editing, when the locate point must be *exactly* where the punch-in is to be executed.

Track delay—which allows you to set up to a 170 ms delay on individual tracks—is one of those "neat" tricks that sounds fun, but has limited practical application to everyone except tweakoids. (Super-nerds like Donald Fagen, who seem dedicated to turning the recording process into an exercise in nuclear physics, will love this feature.) Track delay *is* beneficial for adjusting the rhythmic feel of a track or correcting microphone phasing. If your masters consistently suffer from these problems, however, we advise you to lay down the right groove in the first place and practice your mic placement.

Perhaps the BRC's coolest feature is its digital bus. By employing the optical link between ADATs, the BRC can initiate track bounces within the digital domain and do limited, random access-style editing. For example, you can copy-and-paste song sections between two ADATs, do an assembly edit of several individual vocal performances to create a stunning composite track, or drop a perfect background vocal performance into every chorus. With the optional Alesis AI-1 AES/EBU and S/PDIF interface, you can transfer two tracks at a time to compatible digital equipment.

Unfortunately, the digital bus has a limitation: You can't adjust individual levels on the tracks being combined, because the digital bus has no DSP functions. The Alesis rep I talked to explained that the feature should be called a "multichannel, optical digital interface." It was designed to facilitate routing cloned tracks to other locations, such as when copying one perfect chorus take to every chorus in a song. It still makes more sense to bounce tracks in the analog domain through your mixing board. If you own a quality mixer, the sound quality should be minimally compromised.

SYNCHING UP

Another key feature of the BRC is its versatility with SMPTE. The BRC will read, write, generate, chase, and lock your ADATs to just about anything. (And because sync data is maintained in sub-codes, you don't lose an audio track to the time code.) This opens up vast new worlds for personal and project studios.

One of these worlds is audio post-production for video. The BRC's video-sync function recognizes NTSC, PAL, and SECAM formats, as well as 48 kHz Clock. The BRC generates and reads SMPTE (24 fps, 25 fps, 29.97 fps drop-frame, 29.97 fps non-drop frame, 30 fps, and 30 fps drop-frame), so you can stripe time code onto one of the audio tracks of a VCR and chase-lock to video.

We tested the sync capability by locking up two ADATs to a JVC 6400 professional video recorder. The sync was right on, although it took a frustrating amount of time—about ten seconds—for the ADATs to lock to the video deck. However, we're used to rapid punch-ins, and because the ADATs won't go into Record mode until they're time-locked, we kept trying to punch in when the machine wouldn't let us. (We adjusted our impatience to allow for the chase-lock time.)

We also locked up our Tascam MS16 analog multitrack to the two ADATs. One track of the analog 16-track was surrendered to SMPTE, a fair exchange considering we could now gain up to 128 ADAT tracks. If you purchase an ADAT system, we recommend you keep your analog deck. The ability to combine analog and digital tracks is a huge benefit and renders moot any complaints of digital sterility. Imagine: You could invoke all that wonderful analog-tape coloration for drums and guitars and use the pristine digital tracks for vocals and synth pads.

The synchronization of analog and digital tracks, after the chase-lock process was completed, was right on. However, the machines took between eight and fifteen seconds to lock up, which was pretty frustrating. According to Alesis, the slow chase-lock times will be fixed in future software updates.

Lastly, we tested how the BRC—which outputs MIDI Clock and MIDI Time Code (MTC)—synched with computer sequencers. When we locked a *Performer* sequence to the ADATs, using MTC, the sync was flawless. Chase-lock time was still between six and ten seconds, but by now we were accustomed to the lags.

We had less luck when a client brought in a keyboard workstation that used MIDI Clock; the sync proved to be inconsistent.

CONCLUSION

The BRC is a well-conceived, sturdy unit that is packed with features. Some features may be superfluous or unnecessary to your working style, but the comprehensive autolocation and sync functions, which should be of value to *all* recordists, are well worth the $1,995 price tag. Just consider functions that don't interest you as value-added bonuses.

For the most part, the BRC worked brilliantly. Occasionally, the unit would go momentarily brain-dead, or confuse us with alien messages and shut down the system. But nothing bad happened that couldn't be cured by restarting the unit. (Or giving it a short break.) The only thing that couldn't be fixed or worked around was the owner's manual: It's far from a comprehensive or well-written reference.

For example, we didn't appreciate that the BRC's error messages had nothing to do with any malfunctions on the slave ADATs, a fact that was not fully clarified in the manual. Besides, why aren't the ADAT error messages defined? It is rather disconcerting that Alesis, in calling their MDM "professional," have denied audio professionals high-level documentation. (Alesis is allegedly planning a more comprehensive applications manual and/or

Product Summary

PRODUCT:
BRC
PRICE:
$1,995
MANUFACTURER:
Alesis Corporation
3630 Holdrege Ave.
Los Angeles, CA 90016
tel. (800) 525-3747
or (310) 558-4530
fax (310) 836-9192

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	●●●●◖
EASE OF USE	●●●●
DOCUMENTATION	●●
VALUE	●●●●

video guide for the BRC.)

Of course, we're only complaining because the BRC is such a powerful machine that we want *more* out of it. The BRC was a joy to work with and helped make some difficult sessions run much easier.

And in the tense madness of the recording studio, anything that makes our lives easier earns our everlasting and undying devotion.

(Many thanks to Clark Cable for providing us with the additional Elco cable needed to complete our review.)

Neal Brighton *is an independent producer/engineer and co-owner of Sound & Vision studios in San Francisco.* **Buddy Saleman** *is head staff engineer at Sound & Vision.*

Jupiter Systems Infinity (Mac)

By Paul D. Lehrman

Innovative, automated sample-looping and processing.

Are you a sampling addict? Do you stay up all night trying to get the perfect loop in your string sound, your vocal samples, and even your snare drum? Have your ears become so sensitive you can hear periodicity—that "repeating" feeling—in a square wave? Do you long for a flute sound you can loop 'til next Tuesday without anyone catching on that it's not a real player with gigantic lungs? In that case, *Infinity DSP Looping Tools for the Macintosh*, the first product from Jupiter Systems, is for you.

Infinity does one thing, but does it extremely well. It creates sample loops, always good ones and occasionally great ones. Most of the time, it creates them a heck of a lot faster and more easily than anything else you've used.

The program loads and saves samples in *Sound Designer* (I or II) or AIFF format. It lets you audition the samples through Digidesign's Audiomedia and SampleCell cards (any of the various models), the Sound Accelerator II and Pro Tools systems, or—if you really must—the Mac speaker. It requires, nay *demands*, System 7, running in 32-bit mode. Four megabytes of RAM is essential, 8 MB is recommended, and more isn't a bad idea. A 68030 or 68040 processor is strongly recommended, but the program will work on a 68020 machine running *Mode 32* (a 32-bit addressing extension available from Apple).

FEATURES

Infinity has a traditional set of editing tools: region selection, vertical and horizontal zoom, Cut, Copy, Paste, and Mix. All come with "blending" options for crossfading, as well as Normalize and Gain Change. When you select a region, an Auto-Zero option restricts the boundaries of the selected areas to positive-going zero crossings. Using this option ensures that when you move regions around, you won't get any pops or clicks.

However, the heart of the program is its looping functions. You can insert start and end points and move them around while the sound plays, a unique feature among sample-editing programs. The cursor moves with the sound, so you can see where it loops while you listen.

Once you set up the loop points, you have a choice of four processing algorithms to make them smoother. The Crossfade Looper (see **Fig. 1**) will be familiar to most sampling fans, with its graphic juxtaposition of the start and end points and onscreen arrow buttons for adjusting the points one sample at a time. Controls are provided for determining the length of the crossfade (in percentage of the loop length) and the volume taper, which is continuously variable between linear (for simple, periodic sounds) and equal-power (for more complex ones).

As with other sample-editing tools, there is a function that automatically searches for appropriate loop points, but *Infinity*'s is more elaborate than most. According to the manual, in addition to scanning for zero-crossings and matching slopes, *Infinity*'s Smart Auto-Scan feature "compares the amplitude and phase of the major harmonics at the loop points" and matches them. The number of points tested is adjustable from ten to 1,000. Also adjustable is how closely two points must

match to be considered acceptable, on a scale from zero (anything goes) to 100 (perfect match).

Another algorithm is the Freeze Looper. Used for simple instrumental sounds, this creates a perfectly uniform, short loop, with a length based on an FFT analysis of the sound and the MIDI note number assigned to the sample. The idea is that the loop will always sound at the correct pitch relative to the rest of the sample, even in samples with poor pitch stability. An adjustable crossfade parameter determines how much blending takes place between the loop and the attack segment that precedes it. The program also has a Crossfade Taper control and a Pre-condition control, which minimize noise in the loop and adjust its timbre.

Now things start to get interesting. According to the manual, the Rotated Sums Looper "rotates and adds the sound in the [computer's] accumulator," which takes the idea of bidirectional looping one step further than usual. This process "randomly distributes sonic irregularities throughout the loop." The result is that highly complex sounds, such as string or brass ensembles, can be looped without sacrificing their complexity and without sounding repetitive or artificial.

Several control parameters are available here. The number of rotation operations can be specified from one to 100, with more rotations giving better results (up to a point), but taking longer. As in the Freeze Looper, you can specify whether the loop and attack segments are crossfaded, and if so, the crossfade's length, its taper, and how many test points are used to determine the crossfade point. This function has the added feature of sounding different each time you use it, so if it doesn't work the first time around, you can try, try again.

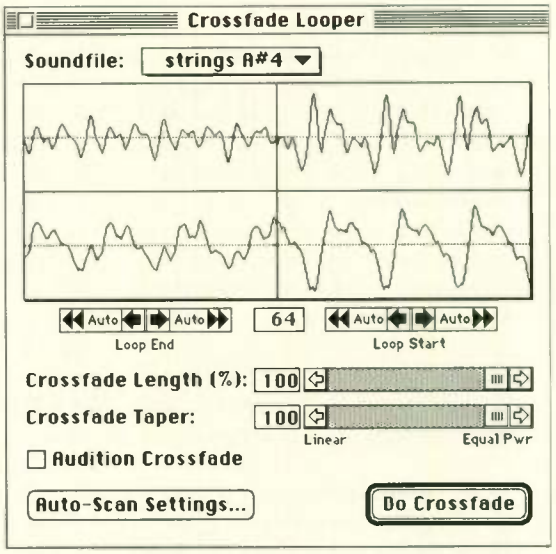


FIG. 1: *Infinity*'s Crossfade Looper provides a graphic view of the start and end points and arrow buttons for adjusting the loop points. The Smart Auto-Scan feature finds zero-crossings and matching slopes, but also matches the amplitude and phase of the major harmonics at the loop points.

Finally, there's the SPR Looper (see Fig. 2). SPR stands for Spectral Phase Randomization; the mechanism is similar to the Rotated Sums Looper, but it independently processes every frequency in the loop. It's more thorough

than the other algorithm, but it takes longer. The manual claims it is particularly suited for stereo samples, in that ambience and dimensionality are maintained. The controls are the same as in the Rotated Sums Looper, and it also comes out differently each time you run it.

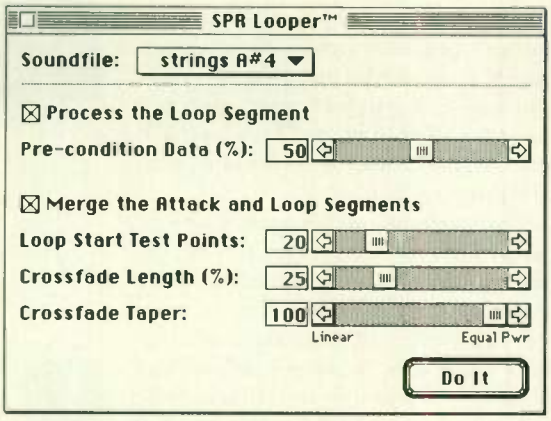


FIG. 2: *Infinity*'s Spectral Phase Randomization (SPR) Looper independently processes every frequency in the loop. It is well-suited for stereo samples, as ambience and dimensionality are maintained. The results are different each time you run it.

EVALUATION

For the most part, the program works remarkably well. As you might expect, relatively simple sounds, such as solo winds or guitar, are no problem at all. But the real strength of the program becomes apparent when you use it with complex waveforms, such as instrumental ensembles, percussion, and industrial sounds. After trying out a few orchestral samples, I wished I could process the ROM samples in my Proteus and Kurzweil 1000 modules through *Infinity* to eliminate some of the more annoying loops.

Even short samples you'd never consider looping give useful results as long as the sound has some "body" the program can grab. I got some fascinating results with timpani, bell tree, and a deep snare drum. Overall, the program lives up to its billing handsomely.

LIMITATIONS

It's hardly Jupiter Systems' fault that Digidesign pulled external-sampler support from *Sound Designer II*, or that Passport has stopped development on *Alchemy*. When they were developing *Infinity*, the Jupiter Systems staff considered it an adjunct to the other programs and had no idea their product would end up being the only game in town. But that's what it is, and when viewed in this light, the program has significant gaps. Forgive me if it seems I'm criticizing a horse for not being a duck, but Mac-based sampling fans happen to be in dire need of a new duck.

If *Infinity* is to fill this enlarged role, several minor points need improvement. Although the user interface is neat, professional-looking, and easy to use, it's somewhat incomplete. You can adjust the loop points while a sample is playing, but you can't change the screen resolution, or even scroll it. If you happen to be zoomed in on the loop start, there's no way to adjust, or even look at, the loop end without stopping playback, resetting the screen, and starting playback again. Each of these operations takes a few seconds, so the program seems sluggish at those times, even though it certainly can't be accused of running slowly. True, *Alchemy* and *Sound Designer* fare no better in this regard. However, *Infinity* is designed to be a looping specialty program, and on-the-fly zoom and scrolling are important to achieve excellence in looping.

In addition, the manual isn't particularly forthcoming about what's *really*

Product Summary

PRODUCT:
Infinity DSP Looping Tools
PRICE:
$495
SYSTEM REQUIREMENTS:
Mac II or better; 32-bit addressing; 4 MB RAM (8 MB recommended); Digidesign audio card highly recommended
MANUFACTURER:
Jupiter Systems
59 Crother, PO Box 697
Applegate, CA 95703
tel. (800) 446-2356
or (916) 878-6666
fax (916) 878-8577

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ●
EASE OF USE	● ● ● ● ●
DOCUMENTATION	● ● ● ◖
VALUE	● ● ● ●

going on in the more esoteric algorithms. Undoubtedly, much information was omitted so the manual would be a reasonable size, but without better explanations, or (preferably) some comprehensive tutorials, adjusting the loop points and parameters seems like shooting in the dark.

The loop-processing operations don't always give good results for every sample. There may be some phasing problems, or the periodicity may still be audible. You can (and should) try different algorithms, as well as adjusting the Test Points parameter and Attack/Loop Crossfade factor. A spokesperson for Jupiter Systems suggests making the loop longer and performing several processing operations on it, which helps randomize anomalies in the loop and make the periodicity less obvious.

▼

Macintosh-owning sampling fans happen to be in dire need of a new duck.

Occasionally, some annoying pops turn up when a processing function causes digital overload. Fortunately, judicious use of the Gain Change control can prevent this.

One limitation may not be easy to overcome (*Alchemy* never did): The program only plays and processes sounds that fit into RAM. It *loads* sounds from disk that are bigger than the available RAM, but you can only play portions of them. Granted, most instrument samples are only a few seconds long, so this is not a big problem under normal circumstances. But if you get ambitious and want to work with very long samples (gongs and church bells come to mind), you may need to allocate more megabytes to the program than you think. A Preview From Disk function should not be difficult to implement, and it would be welcome, especially to those with large sample libraries.

The editing tools are rudimentary. There is no "pencil" waveform-editing, fading, EQ, compression, resampling, or time-stretching. The program doesn't provide pitch-shifting, either, which is especially unfortunate. Without real-time pitch change, you can't hear the sample played back over several MIDI notes; in fact, there's no MIDI interaction at all.

The program only recognizes a single loop; there are no provisions for release loops, which many users consider important. Furthermore, after you process a sample, all the data that originally came after the loop is deleted and replaced by enough iterations of the looped section to fill out the original sample length. So if your sample had a nice, tapered release, kiss it goodbye, and pray you can reconstruct it using your sampler's envelope control!

The most important drawback is that *Infinity* gives you no way to send the processed samples to an external sampler other than SampleCell. To use it with anything else, whether it takes MIDI Sample Dump Standard, SMDI, or E-mu's old RS-232 protocol, requires *Sound Designer* (an older version!) or *Alchemy*.

Fortunately, the company is aware of all of these problems and is working to solve them. Sampler support and multiple loops are likely to show up in future versions of the program, and another program is under development that will have a more comprehensive set of editing tools.

CONCLUSIONS

Although it's not (yet) the mother of all sample-editors, *Infinity* deserves an enthusiastic reception by serious sampling junkies. If you depend heavily on instrumental samples for compositional work, it is an extremely valuable tool. It won't solve all of your problems overnight, but it accomplishes many tasks no other program can touch, and it does them quickly, painlessly, and often very impressively. Although the loop-processing algorithms are presented rather cryptically, they are clever indeed, providing a unique and welcome addition to the sound-designer's bag of tricks.

Paul D. Lehrman *hopes you will buy* MIDI for the Professional, *published by Music Sales Corp., which he co-authored with Tim Tully. He has this nasty sampling habit to support.*

Circle # 440 on Reader Service Card

Etymotic Research ER-4 Canal Phones

By Don Labriola

Put the music directly in your ear.

Let's face it: Most conventional headphones are far from perfect. They're great for some applications, but they usually have wimpy bass, let in outside noise (or provide isolation but weigh a ton), and tend to fall off your head at the wrong time. Intra-aural (in-the-ear) headphones, a relatively recent development, address these problems by fitting into, rather than over, your ears.

Intra-aural phones look and act like conventional earplugs, fitting snugly in your ear canal to shut out external sounds and eliminate the bass roll-off that characterizes open-air models. Most intra-aural phones are quite expensive, but Etymotic Research, a manufacturer of audiological earphones and binaural recording equipment, has released a relatively inexpensive model called the ER-4 Canal Phone. The ER-4 features a 20 Hz to 16 kHz (±4 dB) frequency response, 115 dB maximum output levels, and a hefty 24 dB of isolation, comparable to that of many earplugs.

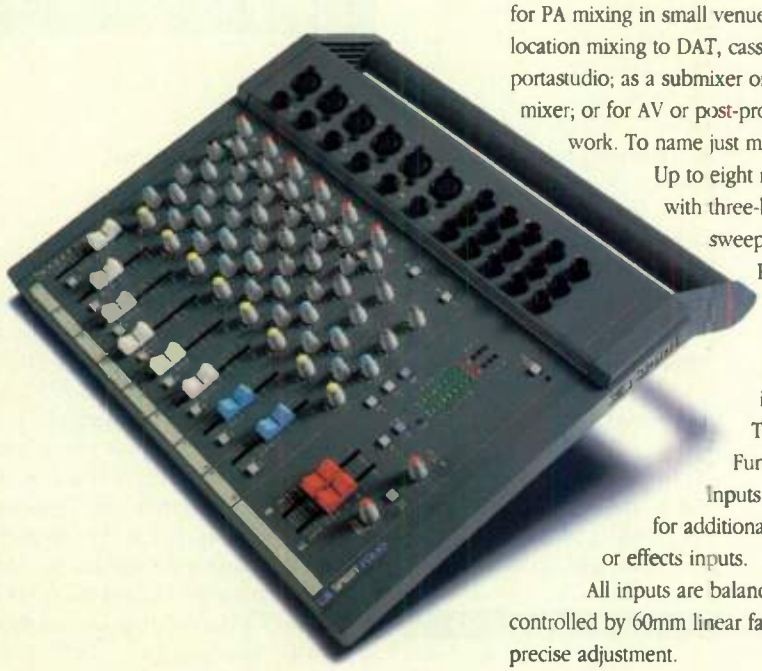
The ER-4 comes in three flavors. The ER-4S is the standard, stereo model. The ER-4L is a low-impedance version with increased sensitivity for use with devices such as wireless sound systems and low-output headphone jacks. The ER-4B is optimized for binaural signals. All three have molded cables that look like they'll stand up to lots of abuse. The ER-4s are packaged with both foam and rubber eartips that are about as comfortable as standard earplugs, and custom-molded tips are available from third-party suppliers for about $100.

I tested the ER-4s on a home stereo, in the recording studio, and as a live-performance monitor. In all cases, sound quality was very good. The test units produced impressive bass and a detailed, natural-sounding high end. The 4L's bass response is the same as that of the other models. Because the 4Ls are more sensitive and filter out

FreeSpirit

Here it is. The mix set free. Spirit Folio may be small, and light enough to carry with ease; but it's no lightweight when it comes to big professional features - it's simply packed with them. And it gives you quality that's even better than DAT.

You can use Spirit Folio anywhere - for PA mixing in small venues; home or location mixing to DAT, cassette or portastudio; as a submixer or keyboard mixer; or for AV or post-production work. To name just many.



Up to eight mono inputs with three-band mid-sweep EQ, High Pass Filter and two Auxiliary sends are included. Two Full Function Stereo Inputs are perfect for additional keyboard or effects inputs.

All inputs are balanced and are controlled by 60mm linear faders for precise adjustment.

A separate Stereo Effects Return doubles as a Two Track Return for monitoring in recording applications. Spirit Folio comes in 3 frame configurations: 10/2, 12/2 and 12/2 rackmount.

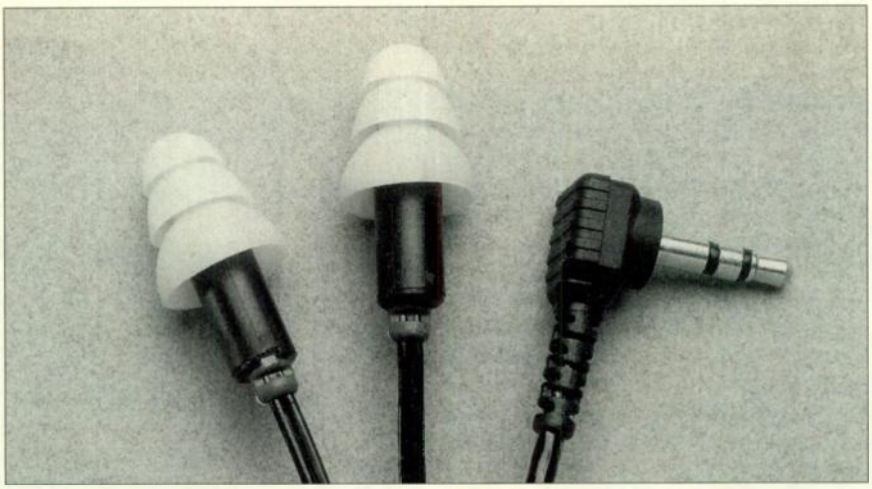
And here's the real beauty of Folio. It's totally affordable – starting at under $500.

Folio will set your spirit free.

Soundcraft/JBL Professional,
P.O. Box 2200, 8500 Balboa Boulevard,
Northridge, CA 91329. U.S.A.
Tel: 818-893 4351. Fax: 818-893 0358.

SPIRIT
FOLIO

A Harman International Company



Based on a hearing aid-type transducer, Etymotic Research's ER-4 Canal Phone earphone delivers quality sound with a broad frequency response and excellent isolation.

some of the highs, however, they appear to provide a bit more bass when driven with equal power.

Onstage, the ER-4s' isolation characteristics, lightweight, non-slip design, and discrete profile made them a good alternative to conventional stage monitors. My only complaint is that the 4-foot audio cable is too short and has an ⅛-inch miniplug; don't plan on using these earphones without an extension cable. They'd be an even better choice if used with a small wireless system.

Before you run off and buy a pair, be aware that, like other headphones, the ER-4 can produce dangerous sound-pressure levels that can damage your hearing. This is especially true if you're using them live, where popping connectors, cable shorts, and feedback (not always caused by conventional monitor/mic interaction) can fire dangerous transients into the mix. Unlike conventional monitors, however, you can't just duck out of the way to escape sonic mayhem; the sound source is within your ear, and if the worst occurs, your hearing could be history. Therefore, *never* use such devices without putting a peak limiter on the line.

With a limiter and an extension cable in-line, I liked the ER-4 a lot. It's an innovative product and a good value.

Don Labriola *is a frequently published author and consultant to the multimedia and telecommunications industries.*

Product Summary

PRODUCT:
ER-4 Canal Phones
PRICE:
$330 (all models)
MANUFACTURER:
Etymotic Research, Inc.
61 Martin Lane
Elk Grove Village, IL 60007
tel. (708) 228-0006
fax (708) 228-6836

EM METERS	RATING PRODUCTS FROM 1 TO 5
AUDIO QUALITY	● ● ● ● ◖
VALUE	● ● ● ● ◖

Digital Designs Magma M6 Monitors

By Neal Brighton

Hot sound reproduction for a cool price.

If you revel in choosing from a wealth of possibilities, you probably love shopping for monitor speakers. More than 30 companies offered near-field monitors the last time I checked, and the number increases all the time. If you are lucky enough to have a champagne budget, you can easily spend a cool $2,000 to $3,000 for high-end speakers. Fortunately, those of us on beer budgets can get some surprisingly good monitors at rock-bottom prices.

An fine example is Digital Designs'

DIGITAL DESIGNS MAGMA SPECIFICATIONS

Model	Impedance	Power Handling	Woofer	Tweeter	Frequency Response	Dimensions
M6	4Ω	60W RMS	6.5-inch ported	20 mm	45 Hz-20 kHz (±3 dB)	13.5 x 9 x11 11 inches
M26	8Ω	120W RMS	6.5-inch ported (2)	20 mm	30 Hz-20 kHz (±3 dB)	18.5 x 10 x 15 inches

impressive Magma M6, which retail for $219 *a pair* (finished). I drove the speakers with an AB Precedent-series power amp and auditioned a variety of music, from rap to rock to classical. I engineered most of the recordings used in my tests, so I'm in a position to assert that the M6s reproduced my work with reasonable accuracy. (They weren't absolutely right on, but I'll discuss that shortly.)

DESIGN

The M6 cabinet is made from good-quality, high-density fiberboard. Two models are available: the M6-OU (unfinished) and M6-OB (black oak-veneer finish). The veneer work on the M6-OB cabinet is well-crafted, and all the components are solidly mounted. Grilles are optional ($5 each). The speaker terminals accept stripped wire and banana plugs.

The speakers are designed to lay on their sides, so you can easily see over them. This may not seem like a big deal, but for those of below-average height, a good line of sight to the sound booth is a plus.

THE SOUND

The M6s have excellent frequency response, which is of supreme importance when mixing. Considering they have a single 6.5-inch woofer, they reproduced the entire bandwidth with excellent clarity. I found the M6s compared quite favorably to similar speaker systems with 6.5-inch woofers (e.g., JBL, Yamaha, and Peavey).

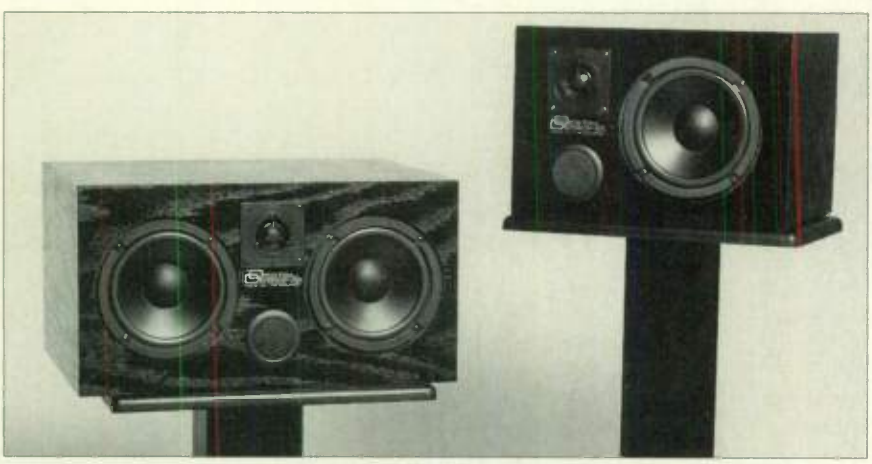
The speakers' front port has a convenient, removable, foam plug that provides an inexpensive, yet effective, way to change the bass response. Removing the plug adds approximately 1 or 2 dB at the bottom end. I didn't notice much difference with classical music, but it made an unmistakable difference for bass-heavy music (especially rock).

The stereo imaging is good, though as with many small speaker systems, the sweet spot is relatively narrow. However, the speakers don't provide the front-to-rear depth I want. I listened to several mixes of different tunes, and even when I added plenty of reverb, the soundfield still seemed 2-dimensional.

A big plus is that the speakers are very efficient in their power consumption. My AB power amp is rated at 125 watts per side, and that was far more than I needed.

CONCLUSIONS

I like the M6; for the most part, it compares quite favorably with the Yamaha NS-10. (The NS-10 is considered the



Digital Designs' low-cost Magma M6 near-field monitors offer good frequency response and efficient power-handling.

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d. "Naughty Bits: Warming Up Digital Audio," p. 64	713	714	715	716
e. "From the Top: Basic Audio Connections, Part 2," p. 76	717	718	719	720
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510	529	548	567	586	605	625	644
511	530	549	568	587	607	626	645
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513	532	551	570	589	609	628	647
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standard by which small monitors are judged.) If you can afford a few extra bucks and have a bit more space, the larger M26 ($331) provides twin woofers, an extra 7 Hz of low-end response, and a bit wider soundstage. You might not really need that extra "woof," though. And if you're an independent producer and haul your favorite speakers from studio to studio, the 15-pound M6 might be a better choice than the 30-pound M26.

Besides, it's tough to argue with the price. For $218 a pair, the veneer-finished M6s are great little monitors. You'll be impressed at the quality a beer budget can buy.

Neal Brighton *is co-owner of Sound & Vision studios in San Francisco.*

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Product Summary

PRODUCT:
Magma M6 monitors
PRICE:
M6-OU $199/pr.
M6-OB $218/pr.
MANUFACTURER:
Digital Designs
100 N. Quapah, Suite K
Oklahoma City, OK 73107
tel. (405) 946-4500
fax (405) 946-4544

EM METERS	RATING PRODUCTS FROM 1 TO 5
SOUND QUALITY	● ● ● ● ◖
VALUE	● ● ● ● ◖

Roland TDE-7K Compact Drum System

By Michael McFall

An all-in-one MIDI drum system.

Long known for their innovative drum machines, Roland has brought electronic drumming into the 1990s with the TDE-7K. This system combines the sounds of the company's best drum machines with a trigger-to-MIDI converter and dual-trigger pads. I know it has been done before, but never so elegantly or conveniently.

The Roland TDE-7K Compact Drum System consists of several units: a TD-7 drum brain, eight PD-7 pads, an FD-7 hi-hat controller pedal, a KD-7 kick-drum trigger pedal, and a lightweight, compact rack system that holds it all in place (including the brain and boom stands for the cymbal pads). All cables are provided, and once set up, the whole unit (except the pedals) can be picked up easily and moved from place to place on stage or in your living room. Just don your headphones, crank up the volume, and you're off into a world of sound so realistic you'll think you're in a recording studio tracking with your favorite band. Which, incidentally, you can do, because there is an auxiliary input for an external sound source (such as a CD player) that can be mixed with the TD-7's output.

HEART OF THE MATTER

The heart of the system is the tiny (not much larger than one of Stephen King's hardbound volumes) TD-7 percussion sound module. Small it may be, but it sure packs a wallop in terms of performance: 512 programmable sounds, onboard sequencer, 32 patch memories, nine trigger inputs, and four audio outputs (one stereo pair and two individual outs).

The sounds are 16-bit, of course, and all are uniformly excellent in quality. The selection of sounds is impressive, as well, encompassing 43 kick drums, 59 snare drums, 36 toms, 8 hi-hats, 8 cymbals, 52 additional percussion instruments (including eleven mallet instruments such as marimba and vibes), 34 sound effects (reverse drums, gun shots, crashes, etc.), and 5 bass guitars.

Although the sounds are superb as they are, you can alter them to your own discriminating taste by changing parameters such as pitch (over an 8-octave range), decay, nuance (which lets you crossfade between two samples, e.g., the edge and near the bell of a ride cymbal), brilliance, attack damp, dynamic pitch bend, panning, velocity curve, and volume.

In addition, the TD-7 has two built-in effects processors. One provides three reverbs (hall, room, and plate) and delay, and the other provides chorus and flanging. The high-quality effects are simple, but programmable. However, you can only use one effect at a time from each processor. For example, you can have reverb plus chorus, delay plus flange, or delay plus chorus, but unfortunately you can't use reverb and delay at the same time.

In addition to playing its own sounds, the TD-7 can trigger external sound modules via MIDI. You can also connect the TD-7 to a sequencer and use it as a drum machine, or 14-note polyphonic, 4-part multitimbral sound module.

PADS 'N' CONTROLLERS

Each PD-7 pad has triggers mounted in the center and outer edge, letting you play two sounds per pad. Strike the center of the pad, and you get one sound; strike the rim of the pad (or the rim and head simultaneously, as in a rimshot), and you get a different sound. You can also layer two sounds on each trigger and switch, mix, or crossfade between them with Velocity. With this arrangement, you can play 35 different sounds in all: eight pads with four sounds each, plus kick-drum and hi-hat pedals with one sound each. The pads possess a nice, non-rubbery feel, providing excellent playability. And although they are small by acoustic-drum standards, I was able to adapt to their size quickly.

The PD-7 pads are preset with minimum and maximum trigger levels, but it's easy to configure the system for other manufacturers' pads: Simply strike each pad hard three times followed by three light strikes. The TD-7 sets corresponding minimum and maximum trigger levels. A Cross-Talk Cancel function effectively eliminates false triggering from closely mounted pads, although it reduces the pads' sensitivity. You can assign each pad to one of four groups; the Cross-Talk Cancel function is applied separately to each group.

The pads also offer a "choke" function that cuts off a sound when you grab the rim of the pad, precisely as a drummer chokes cymbals on an acoustic drum set. However, you can choke *any* sound with the PD-7 pads. This is one more feature that makes playing the TDE-7K electronic drum kit like playing an acoustic kit and then some.

The hi-hat is arguably the most expressive instrument in the drummer's arsenal; to date, electronic percussion manufacturers have had varying

The Roland TDE-7K includes everything you need to assemble a MIDI drum system, including PD-7 pads, FD-7 hi-hat pedal, KD-7 kick-drum pedal, TD-7 drum brain/sound module, cables, and rack hardware to hold it all.

degrees of success emulating it. Roland has come very close with the FD-7 hi-hat controller pedal, which turns any electronic pad into a hi-hat by letting you switch between open and closed hi-hat sounds (or any two sounds).

Being continuous, rather than on/off, the FD-7 can also control the pitch, pitch bend (great for timpani), decay, and nuance. By pressing the pedal—to trigger a hi-hat foot-close sound, for example—it can act as a trigger in its own right. You can use the rimshot and choke functions on the pad assigned to the FD-7 at the same time, making it extremely versatile.

How does it compare with a real hi-hat pedal? Very favorably. After adjusting the volume and spring tension, I was totally comfortable with the FD-7.

Rounding out the controllers is the KD-7 kick-drum trigger. Simply attach your regular acoustic kick-drum pedal to the KD-7 trigger unit, replace your acoustic beater with the reverse beater provided by Roland, and you're ready to go.

SEQUENCING

The onboard sequencer is intended only as an "idea-tester" with which you can quickly record a pattern and jam. It does not have the capability to combine patterns into songs as with most drum machines. Nevertheless, it was quite useful in a number of situations.

You can record and play back patterns of up to sixteen measures each, with memory for 24 user patterns and 24 preset patterns. You start and stop the sequencer by striking an assigned pad and set the tempo by tapping on a pad. The metronome taps the meter as a click, cowbell, woodblock, or maracas sound, or you can choose to hear a sampled human voice counting "one, two, three, four." This last option is a useful practice tool; it's nice to know where "one" is when you're attempting a particularly difficult over-the-barline fill.

After specifying the tempo and length of the pattern, start the metronome and begin playing the pads. The TD-7 has a resolution of 96 pulses per quarter note and quantizes from eighth notes up to 64th notes. (It will record 96th notes with Quantize turned off.) Sadly, there is no swing function; you must provide your own swing feel in real time.

Patterns can be assigned to, and triggered by, any of the pads, rims, or pedals. In addition, there are several pattern-playback modes. In Loop mode, hitting a pad starts playback, and the pattern plays repeatedly until you strike the pad again. In Once mode, hitting the pad starts playback, and the pattern plays once. Hitting the same pad again while the pattern is playing retriggers it from the beginning.

In Tap mode, the pattern advances one note every time the pad is hit. There are four versions of Tap mode; each resets the pattern to the beginning after a preset period of time has passed since the pad was last hit. Tap 1 resets almost immediately, while Tap 4 waits indefinitely to advance the pattern to the next note.

These sequence-playback modes are powerful performance tools. For example, you could assign a bass-guitar pattern to the kick-drum pad in Tap mode and step through the bass line as you play the kick drum. Talk about a tight bass and drum groove! Additionally, you could assign a guitar-stab pattern to the snare drum, so every backbeat would outline the chord structure of the song—great for those 12/8 blues numbers. Or you could assign a fast roll pattern to each of the tom rims in Once mode; each time you hit a tom, you get a flurry of notes—great if you'd rather be twirling your sticks while seemingly rolling with one hand.

Another performance feature of the TD-7 is Sound On Sound, which lets you record a pattern (up to 28 measures in 4/4 time), and then play it

back immediately while adding a new part on top of it. You need a couple of normally open on/off footswitches to make it work, but it is quite useful for building progressively more complex drum solos in a live situation.

MIDI

Because the TD-7 is a trigger-to-MIDI converter, you can use it to play sounds on other sound modules instead of (or in addition to) its own sounds. It also responds to incoming MIDI messages on up to four channels at once.

Each Patch is organized into four Sections. The Instrument Section assigns different sounds to each pad. This lets you assemble drum kits and assign up to 58 different sounds to individual MIDI note numbers, which expands your sonic palette when playing from an external MIDI controller. All 58 sounds can also be recorded in a pattern from an external controller.

In addition, there are three Performance Sections that assign the same sound to all pads and note numbers, which play the sound at different pitches. This lets you play melodies, bass parts, and chord parts from the pads and/or an external MIDI controller or sequencer. All four sections can be assigned to different MIDI channels, which gives you a 4-part multitimbral sound module. You can also assign the Sections to the same MIDI channel for layering purposes.

You can save kits and sequencer data via SysEx bulk dumps. The hi-hat controller pedal can transmit various Control Change messages, including Modulation and Hold. You can also use the mod wheel of a MIDI keyboard in place of the hi-hat controller pedal.

CONCLUSION

There's not much wrong with the TDE-7K, but I will try to point out a couple of items on my electronic wish list. One drawback is that a pattern can only access the sounds in the current patch, which admittedly includes all four sections. This has always been one of my pet peeves. I want access to *all* the sounds at any time.

Another drawback is the lack of pattern-editing and song-building capabilities found in most drum machines. You should be able to tweak the feel of a pattern without having to re-record it from scratch every time. The TDE-7K could have been the best of both worlds: a full-featured drum machine/multitimbral sound module and

Product Summary

PRODUCT:
TDE-7K Compact Drum System
PRICE:
$2,640
MANUFACTURER:
Roland Corporation US
7200 Dominion Circle
Los Angeles, CA 90040
tel. (213) 685-5141
fax (213) 722-0911

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ◖
EASE OF USE	● ● ● ● ◖
SOUND QUALITY	● ● ● ● ◖
VALUE	● ● ● ●

electronic drum kit.

As it stands, however, the TDE-7K gets high marks. Excellent sounds, coupled with effortless playability, make this the one to beat (so to speak). I strongly suggest that you make tracks to your local music store and test bang a TDE-7K. I'm sure you'll be impressed. Or you can get a copy of the CPP Media/Roland video in which Roland product representative Steve Fisher takes the kit through its paces with a band. The video is available from any music store that sells CPP/Media videos or directly from Roland. The TDE-7K Compact Drum System is a serious piece of gear. My only problem is when—not if—I'm going to buy one.

Michael McFall *is the former editor of* Rhythm *magazine. Currently, he heads VDO Productions, a video production company specializing in music-instruction videos.*

Circle # 443 on Reader Service Card

Beyerdynamic M 54 Microphone

By Michael Molenda

A mini mic clamps down on drums.

Drum miking is no picnic, so beyerdynamic's new M 54 mic simplifies the process with a "clip-on" design that renders mic stands unnecessary. Developed primarily for drum and percussion miking, the M 54 employs a small-diaphragm, dynamic mic capsule, mounted on a gooseneck and heavy-duty clamp. Sound-reinforcement and recording engineers can simply clip the mic to a drum rim or percussion rack.

To diminish signal-bleed, the M 54 has a hypercardioid polar pattern. Frequency response is rated by the manufacturer at 40 Hz to 12 kHz, and impedance is 200 ohms. A balanced XLR cable is permanently connected to the microphones.

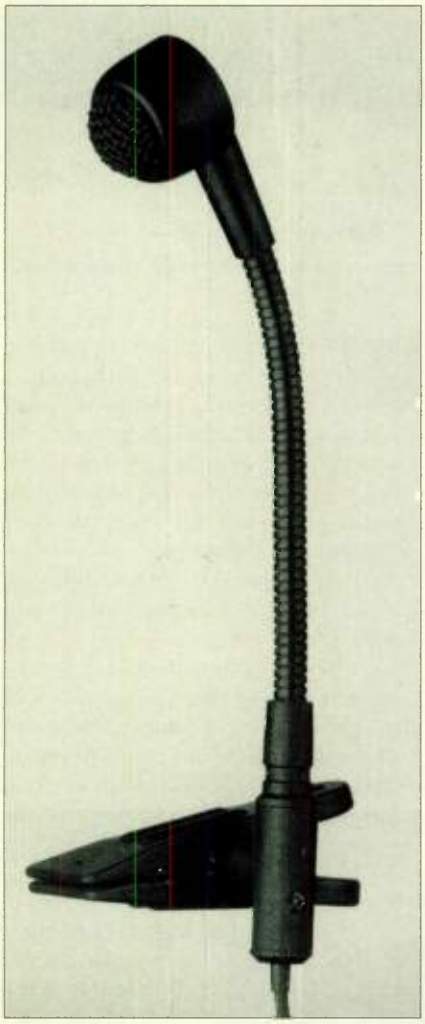
MIKING THE KIT

Most drummers configure their kits more densely than a rain forest, so I'm constantly aggravated by squeezing mic stands into close quarters. I was initially encouraged by the M 54's clip-on construction, but unfortunately, I hit a snag when miking the snare drum: I couldn't get the M 54 into a suitable position. The fixed length of the gooseneck caused the mic capsule to extend into the middle of the drum. Any mic positioned in this danger zone is sure to

Product Summary

PRODUCT:
M 54 Dynamic Microphone
PRICE:
$189
MANUFACTURER:
beyerdynamic
56 Central Ave.
Farmingdale, NY 11735
tel. (516) 293-3200
fax (516) 293-3288

EM METERS	RATING PRODUCTS FROM 1 TO 5
SOUND QUALITY	● ● ●
VALUE	● ● ●



The beyerdynamic M 54 is a clip-on dynamic mic designed for percussion miking.

get shattered by a drum stick. But if I moved the mic away from "ground zero," it hovered approximately five inches above the drum. My favorite snare-miking position is approximately two inches over the rim and an inch above the drum skin. I couldn't get near it with the M 54, so I bailed.

I had much more success miking rack toms. Because it's typically advantageous to mic toms further back (to catch drum harmonics and room ambience), the M 54's fixed gooseneck length allowed reasonable positioning. Sonically, the impact of stick to drum head was clearly defined, as was the deeper resonance of the drum tone and harmonics. Each tom on a 4-tom kit was miked with the M 54, and each tonality—from high to mid-low—was reproduced well. The mic sounded so good that I recorded the toms flat (no EQ). On floor toms, however, the M 54 sounded a bit papery. I couldn't get

a full-bodied wallop. (I typically use a large-diaphragm dynamic or condenser mic on floor toms to reproduce a suitable punch.)

I also discovered the M 54 sounds great on percussion racks bearing chimes, bells, cowbells, and other sparkly delights. The mic's frequency range tops out at 12 kHz, so you don't get shimmering highs, but the overall timbre was still bright and articulate.

Just to be stupid, I clamped the M 54 to a cymbal stand and miked a ride cymbal from *below* (an "under-head" mic?). I discovered a cool effect: The cymbal harmonics are too obnoxious for conventional applications when miked from below, but conceptualized as an effect, the harmonics produce a beautiful wash of ringing tones. I recorded a separate pass of a ride-cymbal performance, compressed the signal to accentuate the harmonics, and used the shimmering sonic waves as a song introduction.

During aggressive drum performances the M 54's clamp held tight. I never had to worry about it slipping off the drum rim. I have some concerns about the toughness of the mic's attached XLR cord; it's pretty slim, and rough handling might sever the connection. I recommend taking more care than normal when connecting and routing the cable.

CONCLUSION

For miking rack toms with a minimum of fuss, the M 54 is a champ. Because the mic clips to the rim of the drum, you can say goodbye to mic-stand traffic jams when placing mics for multiple tom configurations. Also, the M 54 sounds great on ancillary percussion, such as chimes and bells.

However, the M 54 is not an all-purpose drum mic. (To be fair, I'm not sure there *is* such a thing.) The fixed gooseneck makes it difficult to find an optimum position for snare miking, and I wasn't blown away by the M 54's reproduction of floor toms. But as a "team player," the M 54 is definitely an all-star rack-tom mic. ●

Circle # 444 on Reader Service Card

Creative Labs Sound Blaster 16 ASP

By Bob Lindstrom

The new version of this PC sound card is better than ever.

Multimedia computing got a big boost from the combination of General MIDI (GM) and Microsoft's Multimedia Personal Computer (MPC) standard. For the first time, a common ground was defined for the instrumentation and playback of MIDI music. However, every synthesizer that claims GM compatibility interprets the GM-sanctioned sounds in a unique way.

One hedge against this predicament is to compose for the most popular GM devices. The leader among outboard GM synths is generally considered to be Roland's Sound Canvas, but that represents a small percentage of the

PC audience. In that marketplace, the majority leader is Creative Labs' Sound Blaster plug-in sound card.

Creative Labs' newest incarnation of the Sound Blaster is the Sound Blaster 16 with Advanced Signal Processing (SB16ASP). This powerful audio card offers complete Sound Blaster compatibility; 4-operator, 8-waveform, TX81Z-style FM synthesis (compared to the original Blaster's 2-op); and 16-bit, stereo, direct-to-disk recording at sampling rates up to 44.1 kHz.

Creative Labs also sells an add-on board, the Wave Blaster, that puts GM samples from E-mu under SB16ASP control. Together, the combo is a potent GM solution that puts high-quality sound in your PC and maximizes your GM compatibility.

THE HARDWARE

The tinny, 2-op synthesis in the original Sound Blaster is greatly improved in the 4-op SB16ASP, which offers 11-voice polyphony (six melody, five percussion) using Yamaha's OPL3 chip. (The chip produces twenty voices in 2-op mode.) Digital recording with 8- and 16-bit resolution can also incorporate up to 4:1 compression and decompression *in real time*, thanks to the proprietary Advanced Signal Processing (ASP) chip.

The card includes a proprietary connector for Creative's internal CD-ROM drive (other internal and external drives can be used only with a separate controller card), along with microphone and line inputs that can be

Product Summary

PRODUCT:
Sound Blaster 16 ASP
Wave Blaster
PRICE:
Sound Blaster 16 ASP: $349.95
Wave Blaster: $249.95
MANUFACTURER:
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1901 McCarthy Blvd.
Milpitas, CA 95035
tel. (408) 428-6600
fax (408) 428-6611

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EASE OF USE	● ● ● ●
SOUND QUALITY	● ● ●
VALUE	● ● ● ●

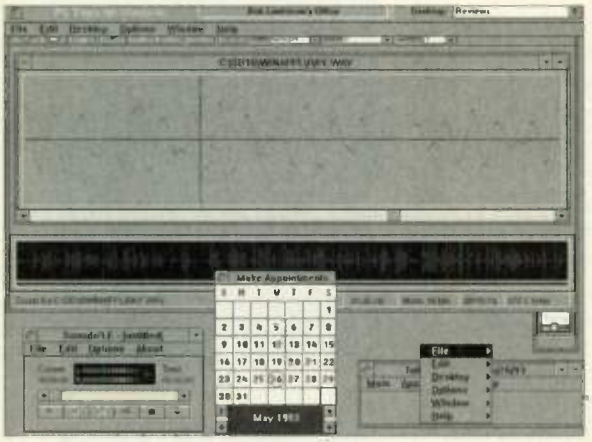

FIG. 1: Software bundled with the SB16ASP includes a sample editor, OLE server for *Windows*, and a talking scheduler/appointment program.

used simultaneously. A 4-watt amplifier drives a stereo headphone/speaker output. Finally, the joystick port doubles as a Roland MPU-401 UART-compatible MIDI port when used with an optional MIDI interface kit.

If 4-op synthesis doesn't ring your digital bell, the Wave Blaster (WB) option adds 4 MB of 16-bit, Proteus-based GM samples: 128 instruments, 18 drum kits, and 50 sound effects. This provides up to 32 Wave Blaster voices in addition to the SB16ASP. As a bonus, the WB offers an MT-32 emulation mode that provides access to the thousands of MIDI files composed for that Roland synth.

If you already have several boards in your PC, you'll face the usual problem finding a free interrupt (IRQ) and I/O base addresses when installing the SB16ASP. However, the card does provide two conveniences to ease the process. First, it is configured through software, so there's no need to crack open the PC case. Second, all settings are automatically tested during the setup process. Still, it took a couple hours of pulling boards and removing cables to find settings that worked with all my software.

THE SOFTWARE

The SB16ASP comes with five disks full of bundled software, including a moderately capable sample editor, an OLE (Object Linking and Embedding) audio server, a text-to-speech utility that uses the card to generate computer speech, and a multimedia scripting language and player (see Fig. 1). For applications, there's *HSC InterActive SE*, a graphical, multimedia authoring program; *PC Animate Plus*, an animation program; and *Monologue for Windows*, which adds text-to-speech features to almost any *Windows* application.

The WB's software bundle isn't so copious, but it's no less useful (see Fig. 2). The big bonus is *Cakewalk Apprentice for Windows*, a junior version of *Cakewalk Pro* from Twelve Tone Systems. This version of the sequencer is programmed to work only with Creative Labs devices, which means it can't access another MPU-compatible MIDI card. Fortunately, it does work with the Blaster's optional MIDI port. (According to Creative Labs, this is being updated to work more generically.)

THE SOUND

Both the SB16ASP and WB power up in GM mode, so it was easy to compare their sound quality by switching between them. I loaded *Windows' Media Player* and compared playback of the omnipresent CANYON.MID, the MIDI

FIG. 2: A specialized mixer and configuration panel for the Wave Blaster works in conjunction with the sound-source mixer for the Sound Blaster 16 ASP.

file that comes with *Windows*. The 4-op FM synth in the SB16ASP was what you would expect: good but not great.

The Wave Blaster, however, went beyond my expectations. The GM samples took CANYON.MID into the world of real instruments with lively digitized percussion sounds, a fairly rich piano sound, and tonal subtleties I'm just not used to hearing in a PC sound card. At the consumer level, WB is just about as good as PC sound cards get.

Repeated exposure reveals some of the compromises that led to the WB price point. Mainly, the support circuitry of the Proteus is missing in the WB. As a result, you get Proteus samples and that's all; there's no envelope manipulation or other subtleties. If you're used to hearing the expansive tones of the Proteus/1, the WB will sound a little drab, but when you compare the WB to its PC sound-card competition, it shines.

BITS OF AUDIO

Because you can sample at up to 44.1 kHz with 8- or 16-bit resolution, there are many quality-versus-storage choices. There are three types of real-time compression: Creative Labs' ADPCM, CCITT A-Law, and CCITT μ-Law. Without compression, I used over 2.7 MB on a 44.1 kHz, 16-bit stereo sample. Creative's ADPCM compression took that sample down to just under 700 KB.

I was not entirely pleased with the digital-recording performance. In my 80386/33 MHz setup, the mic input was noisy. Background hiss was audible, and the signal "breathed" in files that went through real-time data compression and decompression. I tried several mics, including the inexpensive dynamic that comes with the card, a somewhat better Sony dynamic, and a Sony condenser. When sampling directly from a CD into the line input, the sound was clear and free of hiss.

But let's put this in perspective. At $350, the SB16ASP is great for the home consumer and in many semi-pro contexts. It simply does not achieve a level of quality required to create the highest-quality professional multimedia audio.

THE OUTCOME

For home and office, the SB16ASP with the Wave Blaster is among a small handful of exceptional PC sound cards. Toss in the remarkable library of bundled software, and you have an incredible software/hardware value.

For professional musical production, it's a different story. With its noisy mic input, this combo comes up a little short for pros. For the same price, you can get the superior direct-to-hard-disk digital recording of Turtle Beach's MultiSound card, which includes a Motorola-based DSP and a full-fledged Proteus/1 with all the support circuitry. However, that system does not include a CD-ROM interface and Sound Blaster compatibility.

When you're "mixing for the masses," you need something in common use to evaluate how the average person will hear your sounds. For that purpose, I can recommend the versatility of the Sound Blaster 16 ASP and Wave Blaster as a mix of good-quality sound and widespread popularity.

Bob Lindstrom *is a freelance writer, composer, and conductor who has created music for several computer games.*

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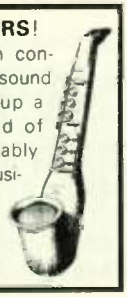

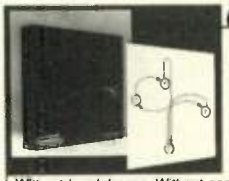

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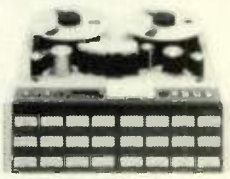

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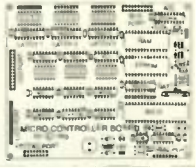

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PRO FILE

Excitement at Your Feet

George Martin takes on The Who's Tommy.

By Daniel Levitin

Producing the cast album for Broadway's version of a celebrated rock opera is difficult enough, but capturing the excitement of a show as visually compelling as *The Who's Tommy* is even more daunting.

"When you don't *see* anything, you've got to make the sound work for you," says legendary producer George Martin.

Having a producer of Martin's stature oversee the recording of *The Who's Tommy* is an event in itself. Few producers are better equipped to translate the excitement of this hit Broadway show to tape.

"The role of the record producer," Martin muses, "is to get inside the mind of the performers and encourage them to give their best. You've got to know how far to push them. Raising the level of an artist's performance often requires asking them to do something difficult when they'd rather take a cautious, easy way."

The show was recorded at New York's Hit Factory rather than live on stage, for both acoustical and logistical reasons. "With a show going on every night," Martin realized, "you'd have to constantly clean up and clear out, which would be too constraining."

Because the soundtrack was recorded in the relative sterility of a studio, maximizing the performers' chemistry was a more important concern than pristine sonics. Martin decided against segregating the musicians and vocalists in isolation booths. "The whole idea was to create a *live* performance. Isolation often inhibits interplay between the performers and can result in a pretty dull recording. We put the drums right bang in the middle of the studio; the guitars and keyboards around that; then the strings, horns, and percussion; and on the extreme perimeter of the studio, in cubicles around the edge, we put the vocalists."

The New York sessions were recorded in less than 30 hours on a Sony digital 48-track and mixed in one week at Air Studios in England. The biggest recording challenge was the show's ending, in which the entire cast slowly assembles onstage, singing "Listening to You." "I wanted to build up the sound to convey the excitement of the show's climax," explains Martin. "We have a very large room at Air, and it's got a long reverberation decay. So I fed a mix [of the track] back into the room through a couple of really good speakers placed at the far end; then I recorded that sound and added it to the final mix. It sounded like a giant echo chamber."

The room's acoustics were crucial to the finale's sound, as Martin is famously reluctant to use digital reverb. "I somewhat distrust electronic devices, because I think nothing beats a really good echo chamber and a bit of tape delay. I am always a little scared about synthesizing things; I like to have as natural a sound as possible."

An original cast recording was a departure for the studio wizard. "I was kind of a jobbing contractor. My job was to convey the show's feeling as faithfully as possible, as well as the excitement it presents in the theater. It isn't like *Sgt. Pepper* where I was involved creatively. Here we have a fait accompli: A very good piece of music that's been finely orchestrated and well-performed."

Daniel Levitin *is a researcher in cognitive science and psychoacoustics at the University of Oregon.*

Producer George Martin, Michael Ceveris (who plays Tommy in the production), and composer/librettist Peter Townshend.

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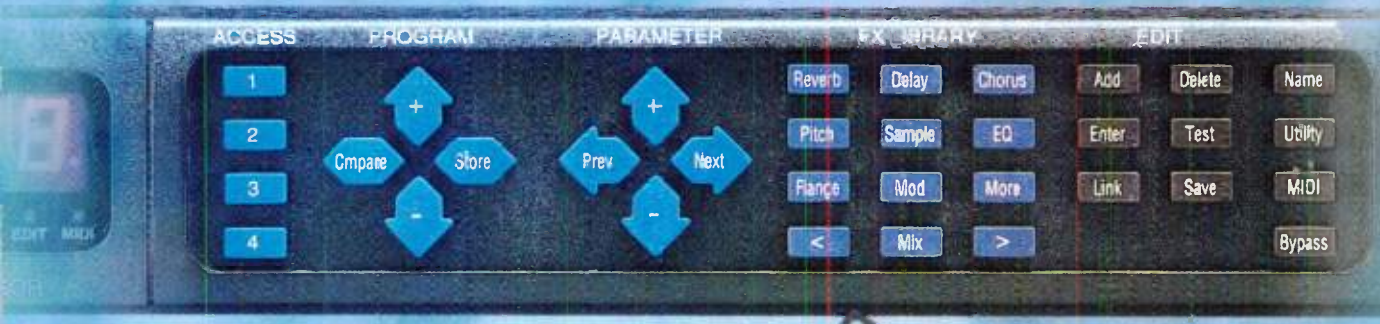

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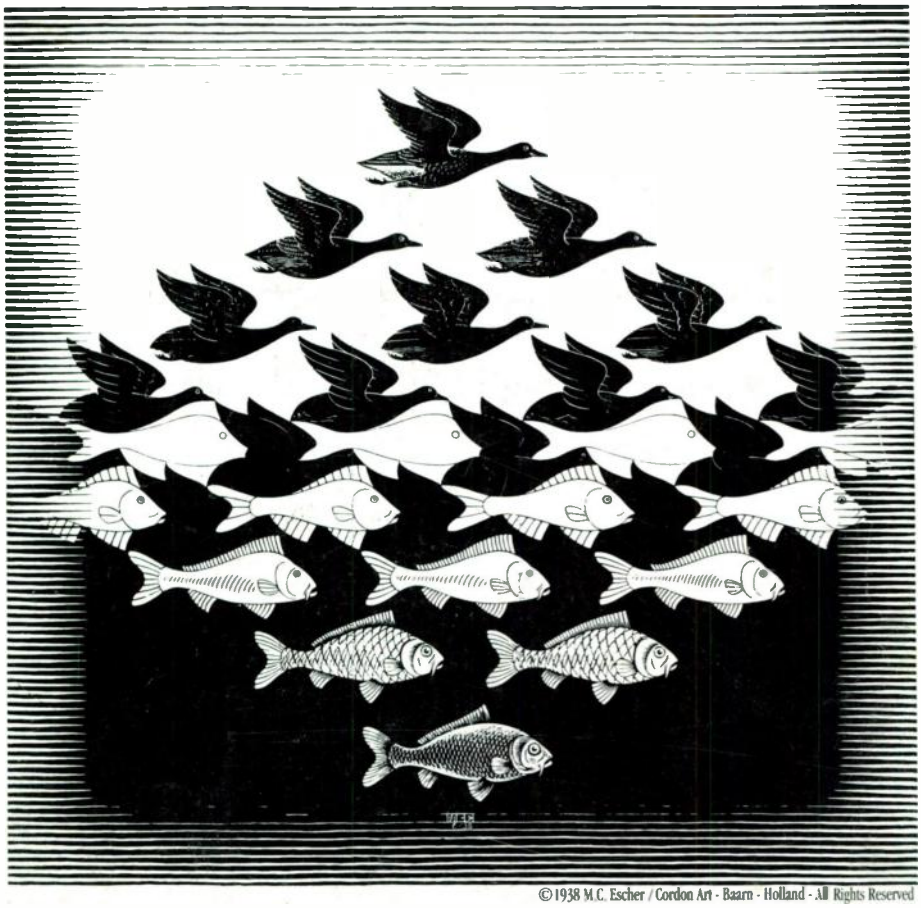

Study the illustration. Are the geese becoming fish, the fish becoming geese, or perhaps both? Seasoned recording engineers will agree that your eyes *and* your ears can play tricks on you. In the studio, sometimes what you think you hear isn't there. Other times, things you don't hear at all end up on tape. And the longer you spend listening, the more likely these aural illusions will occur.

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