Electronic Musician

December 1994

ALL THAT JAZZ

Make Your MIDI Tracks Swing

Play Live and Survive!

Top Tweaks for Ensoniq TS Series Synths

Smoking in the Boy's Room: Women Producers Storm the Charts

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Cover: Photo by Leslie Hirsch.
*Special thanks to Roger Lewis of the Dirty Dozen Brass Band,
courtesy Rosebud Agency.*

Holiday Wishes

Time to get teary-eyed and blubbery
about the good things in life.

I'm not really in the holiday mood
yet—probably because the bizarre timeline of
the publishing world has me writing December's
"Front Page" in October—but I certainly have a
lot to be thankful for. **EM** is bigger, better, and
way cooler than ever. This joyous little miracle
was made possible by the support and counsel
of our publisher, Peter Hirschfeld; our never-
say-die advertising sales staff; the enormous talents of our art and graphics crew;
our scrupulously precise production department; the intrepid newsstand explor-
ers in our circulation department; and, of course, the blisteringly hot writers and
editors who construct columns upon columns of words every month.

We were actually quite surprised by the level of our success earlier in the year,
which necessitated adding between sixteen to twenty-four pages each issue. This
meant a lot more work for everyone: more writing, more editing, more design-
ing and planning, more story assignments, more art, and more s-t-r-e-s-s. Luck-
ily, no one went berserk, got sick, or quit to seek a less stressful job on the local
bomb squad.

I believe that the main reason the staff surrendered pieces of their nights and
weekends—and in some cases, even their vacations—to ensure we published a
class magazine is PRIDE. Now, I don't want to sound like a Marine Corps recruit-
ing officer, but these crazy **EM** people are *committed* to excellence. On days where
practically everyone was flying perilously close to burn-out, staffers would still be
struggling to make stories more comprehensive and redesigning graphics to
deliver more "wham." Unbelieveable!

The sustenance for our success, of course, is all you readers out there. I'm
blown away by the incredible support we have received from everyone. Your
kind words and constructive criticisms have been like cool, soothing compress-
es amidst the workload panic. I thank you all.

I think the best way to honor your enthusiastic support is to make **EM** even bet-
ter next year! We're planning to beef up the presentation of our popular Reviews
section with a new look. Even as this column is being written, products wizard
Steve Oppenheimer is thinking up slicker ways to disseminate critical data with
more informational charts and graphics. Also, in response to popular demand,
there *will* be more DIY projects in 1995.

Personal recording issues and applications will continue to be a major focus,
and we'll be getting more pros to reveal their production and engineering
secrets. Our multimedia reporting will be similiarly enhanced to give electron-
ic musicians all the data they need to make music for this infant, but vital, indus-
try. In short, "more better" of everything you (hopefully) already dig about **EM**.

Have you read enough sentimental mush and impassioned declarations yet? But
wait, there's more! I'd like to offer my personal wish that everyone has a joyous
holiday season and that bucketfuls of bliss await you in the new year. Let's hope
some of the only things that *don't* have a good year in 1995 are sickness, ignorance,
violent crime, and environmental plundering. Happy Holidays!

Michael Molenda

Four Steps To

1 Take The Test

✓ Hear the new sounds in the Alesis QuadraSynth™ Master Keyboard and S4™ QuadraSynth 64 Voice Sound Module.

☐ Stay home and mope.

2 Learn The Technology

True 64 voice polyphony for super-fat live playing or complex multitimbral sequencing. 16 MB of sample ROM featuring classic synth textures and incredibly realistic instrument samples. Superior synthesis engine from the same team of synth designers who turned modulation routing into high art. 256 Programs and 200 multitimbral Mixes from Alesis and InVision - the leading sound development lab.
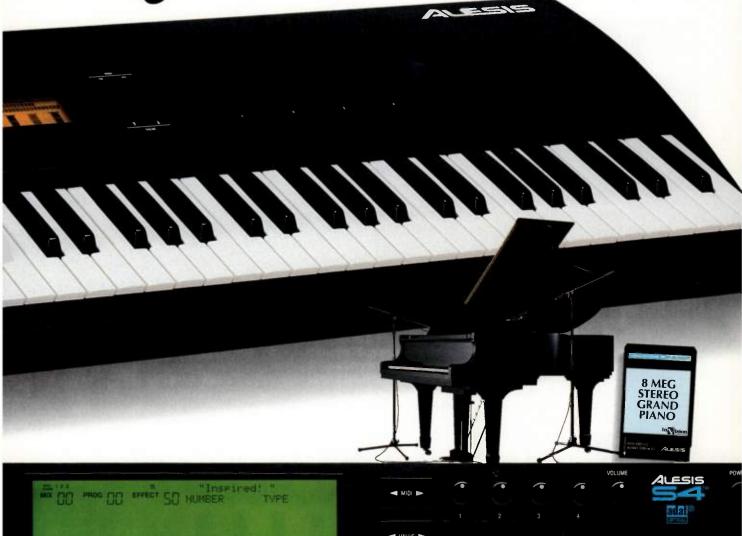
Open architecture PCMCIA RAM/ROM cards, featuring an 8 Meg Stereo Grand Piano and a stack of other great cards. ADAT Optical Interface for direct digital recording. Now you're an expert.

GENERAL MIDI

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GLOBAL   SELECT   EDIT 4   EDIT 1   ◀ PAGE

Keyboard Bliss

8 MEG STEREO GRAND PIANO

ALESIS
S4

VOLUME

MIDI ►

◄ VALUE ►

SOUND CARD

3 Do The Math

From November 15 '94 to January 15 '95, buy a QuadraSynth or an S4 and get a free QuadraCard: The Stereo Grand Piano or any of the other six titles. Your choice. Start adding.

	Value	Your Cost
S4 Sound Module	$999	$999
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For details, see the Hands On display at your Alesis dealer.

4 Play It

ALESIS
STUDIO ELECTRONICS

HARD DRIVEN

I read "Diving into Digital" (October 1994) with great interest. Great job! However, some points about hard drives need clarification.

You state that IDE drives top out at about 1.1 to 1.4 MB/second and that SCSI-2 drives can reach 6.7 MB/s or more. I don't know which drives were reviewed for the article, but most current IDE drives will move at least 3 MB/s, while SCSI-2 can handle up to 10 MB/s. Some of the newest IDE drives can transfer 11.1 MB/s, even faster than SCSI-2!

You also state that IDE drives are limited to 540 MB. This is a DOS limitation, not an IDE limitation. There are several ways around this barrier, from installing a new BIOS (expensive) to installing software drivers, such as On Track's *Disk Manager* (inexpensive, takes up a little memory).

Lastly, you describe a new Enhanced IDE standard that will remove this limitation. The only people who make Enhanced IDE drives are Western Digital. They own the name and the specification, and no other company manufactures Enhanced IDE drives.
Bruce Bullis
Maxtor Tech Support
Ben Lomond, CA

Bruce—You're right, we do need to clarify some of the information that we published in that article. In drafting our response, we consulted with Dal Allan, chair of the SFF Committee, a storage industry group.

First we should explain that we are speaking of the IDE/ATA drives installed in IBM PC-compatible computers running MS-DOS

and Windows. (IDE/ATA stands for Integrated Device Electronics/AT Attachment.) Although the maximum sustained data-transfer rates for an IDE/ATA drive can be higher, Dal tells me the typical sustained data-transfer rate for a machine with an ISA (Industry Standard Architecture) bus and a run-of-the-mill IDE adapter is about 3 MB/second.

The drive size limitation is indeed not set by the IDE/ATA specification, but by a combination of DOS and the computer's BIOS. The work-arounds that you describe will work, but they're not simple solutions.

You're also correct about Enhanced IDE. It is not a standard, but a marketing term used by Western Digital to describe their particular implementation of the SFF 8011 and SFF 8039 specifications and other advanced features. Other disk-drive manufacturers offer similar feature sets and describe them using different labels.

However, in order to benefit from the faster performance that drives and controllers built to these specifications can offer, you must have a PC equipped with a local bus (VESA or PCI). Older computers equipped with only ISA bus slots don't possess sufficient bandwidth to handle the 13+ MB/s sustained data-transfer rates these new devices can put out.—Michael B.

'NETTING THE TRUTH

Your Magical Mystery Tour of Cyber-Space is a little mysterious! When I received my copy of the October 1994 issue and read "Magical Musical Tours," I immediately sat down at my computer to add my name to a few of the listserv resources that you mentioned. You may have "infobots" scurrying through the Internet, but some of them are gathering misinformation. Specifically, I had trouble finding the Internet Underground Music Archive (IUMA) at the address you supplied.

I enjoyed the article very much, but I wish your research also included verification of 'net addresses and lists. Keep up the excellent articles on all aspects of electronic music. I always anxiously await my next issue.
Richard Rayer
Chagrin Falls, OH

Author Scot Gresham-Lancaster responds: Because the Internet is such a dynamic environment, some of the information we published was bound to change between the time we researched it and when it was published. We made every effort to verify that all the addresses we published were accurate, but 'net addresses change, sometimes without advance notice. Listservs even dry up and disappear now and again. For the record, the most current address I have for IUMA is majordomo@iuma.com. You can also get an automated response by mailing to info@iuma.com.

JV-90 JIVE

I just read the Roland JV-90 review (September 1994). Unless I am mistaken, you left out some information about the product. I was intrigued by the expansion capabilities at first, so I rented a JV-90. This product sounded like the answer to a gigging musician's dreams: 76 keys; lightweight; control of eight zones; 28-voice polyphony expandable to 56; and so on.

Expansion is a good idea, of course, but the "gotcha" is that the expanded polyphony is not available from the keyboard, only via MIDI. For instance, the VE-JV1 expansion board has the same sounds as the JV-90 itself (which you failed to mention). If you think you can put in a VE-JV1 and then call up a piano patch with 56-note polyphony, forget it. I suppose you could set up a 2-way keyboard split and put a MIDI cable from the MIDI Out to the MIDI In, but what a kludge!

Also, as a master controller, the JV-90 only takes you part of the way. Sure it has eight zones, but there are only two controller inputs, so you can have a sustain footswitch and a continuous-control pedal. Or maybe you get a sustain switch and two pedals. At any rate, the pedals are global, which is really stupid on something offered as a master controller. To be called a controller in my book, the pedals have to be assignable to zones, can be turned on or off, can be assigned different controller numbers in different performances, etc. In that respect, the JV-90 fails.

BUD PEEN

Also, when you start layering sounds in the JV-90, the processor gets bogged down, and you can actually hear the delay as the poor old processor tries to sound all the notes. Pretty crummy performance for a modern synth, I would say. How you guys can call this a "synthesist's synthesizer" is beyond me. I would say "a good synth for a first-time buyer" might be a better description.

Richard Whitehouse
rwhitehouse@pan.com

Author Geary Yelton responds: *How right you are. I spoke with a product specialist at Roland who confirmed your discoveries. Even if you route the MIDI Out to the MIDI In, you wouldn't really increase the number of notes you could play simultaneously. Instead, you'd have twice as many voices to layer. And yes, the pedals are global and can't be reassigned for each performance. For live use, this is a definite limitation. As for the audible delays, Roland admits that when you layer multiple sounds with four tones each, the processor is indeed overtaxed. Of course, then your polyphony is minimal, and most people aren't going to be layering quite so many tones. I apologize for omitting these details. Good work; maybe you should consider writing product reviews!*

SOUND ON SOUND

My compliments on your September 1994 issue. Heck, I have been a reader of your magazine since *Polyphony* days, and this is probably the first time you have covered English mixers in detail ("The British Invasion").

Most professional audio people are aware of the fact that all audio equipment distorts (i.e., "colors") the sound in subtle ways. Guitarists kill for tube warmth and its luscious tone, and most analog purists will not touch 16-bit digital audio with a barge pole. Like Marshall, Hi-Watt, and Vox Class A AC-30 amplifiers, only the British know how to deliver their sound in ways that boggle the mind. To back this up, watch the proliferation of English consoles in studios around the world: Neve, SSL, Trident, AMEK, DDA, Soundcraft, CADAC, etc. Most important, look at the resale values of these classics.

Not long ago, American sound ruled the recording world: MCI, API, Quad-8, Sphere, Langevin, etc. Sadly, most of these companies are now in foreign hands. I have a lot of respect for Greg

Mackie, since the Tapco and EV days. However, I feel he should spend more time perfecting and innovating the good ol' American sound, as opposed to copying "Ye Olde English Sounde."

Most major-name studios in Japan won't buy or use the same "project-priced" equipment they delight in exporting to the rest of the world. I bought the Yamaha DMP-7 digital mixer a few years back for $5,000 (Canadian). I recently disposed of it for about a tenth of its original price. Next time, I will seriously consider buying high-end analog preamps and equalizers.

For now, Trident and Studer analog with Dolby SR is the only thing for me. When modern audio designers and manufacturers get rid of the term "planned obsolescence," then they will be welcome to my dollars and ears.

Simon King
Leo Project Productions
St. Albert, AB, Canada

KEEP THEM COMING

I am glad I didn't miss the August 1994 issue! "Square One: In the Hall of the Reverb King" was so helpful to me. Several months ago, I bought my first sequencer workstation, a Yamaha SY-85, and knew absolutely nothing about how the effects system worked. The system is modeled after mixing-board effects systems, and I wasn't at all familiar with them. Thanks to your article, which I put side-by-side with my keyboard manual (for cross-referencing), I now have a working knowledge of that part of my system. A friend who works with sound boards told me he wished he had that article back when *he* was learning about effects. Don't by any means stop publishing your more technical articles, but the basic stuff does help some of us. Thanks so much!

Michael Jones
Shreveport, LA

EM EXPANSION?

Having subscribed to **EM** for the last three years, I'd like to let you know how much I enjoy and have profited from your publication. I especially enjoy the warmth, wit, and humor, as well as the solid information and helpful advice that characterizes your articles. "Unabashed flattery," you may

be thinking, but I was raised with the maxim, "Whatever you say, be sure you really mean it," so my conscience is clear.

I would like to suggest that you seriously consider producing a Spanish-language edition of **EM**. After the usual lag of a few years needed for the transfer and assimilation of new technology, there is now a rapidly growing market for electronic musical gear in Venezuela and other Latin American countries, including Mexico, Argentina, and Chile. This means that the receptivity for a magazine like **EM** can only go from good to better in such countries, not to mention the Spanish-speaking population of the U.S.

Ludwig Tuman
Puerto Ordaz, Venezuela

Publisher Peter Hirschfeld responds: *Thanks for your kind comments. We always welcome "unabashed flattery." Your idea of producing a Spanish-language EM is a good one. As you are aware (in an unpublished portion of the letter), our sister magazine* Mix *produces a quarterly version in Spanish, which is sent directly to many recording studios in Latin America.*

However, producing an EM version in Spanish is trickier than it appears. The biggest criteria for viability and success of a magazine is getting good distribution. Musicians are located all over a country, not just in identifiable locations, such as pro studios. Therefore, a Spanish EM would have to be available on the newsstand. The Latin American newsstand market can be fragmented and unreliable. When the distribution becomes more economically feasible, we will happily offer EM in other languages.
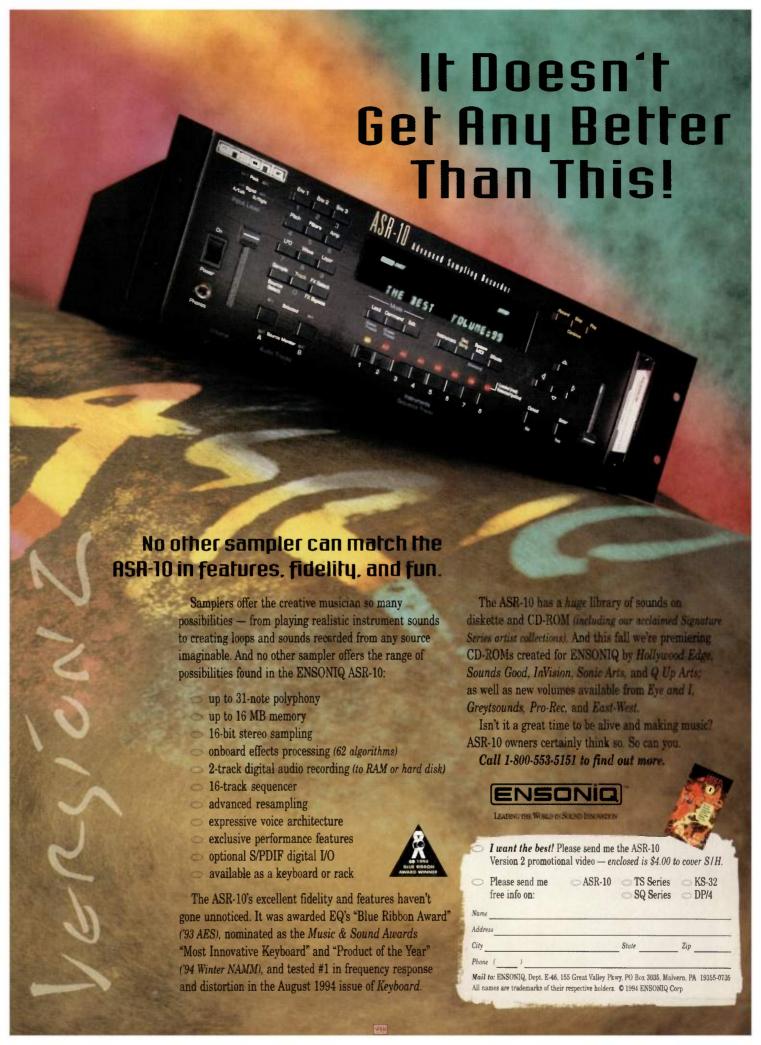
ERROR LOG

October 1994, "What's New," p. 28: The operating system upgrade to Peavey's C8p and C8 is version 2.0, not 3.0.

October 1994, "Diving Into Digital," p. 66: The minimum CPU needed for compliance with the MPC Level 2 specification is a 25MHz 80486SX.

It Doesn't Get Any Better Than This!

No other sampler can match the ASR-10 in features, fidelity, and fun.

Samplers offer the creative musician so many possibilities — from playing realistic instrument sounds to creating loops and sounds recorded from any source imaginable. And no other sampler offers the range of possibilities found in the ENSONIQ ASR-10:

- up to 31-note polyphony
- up to 16 MB memory
- 16-bit stereo sampling
- onboard effects processing *(62 algorithms)*
- 2-track digital audio recording *(to RAM or hard disk)*
- 16-track sequencer
- advanced resampling
- expressive voice architecture
- exclusive performance features
- optional S/PDIF digital I/O
- available as a keyboard or rack

The ASR-10's excellent fidelity and features haven't gone unnoticed. It was awarded EQ's "Blue Ribbon Award" *('93 AES)*, nominated as the *Music & Sound Awards* "Most Innovative Keyboard" and "Product of the Year" *('94 Winter NAMM)*, and tested #1 in frequency response and distortion in the August 1994 issue of *Keyboard*.

The ASR-10 has a *huge* library of sounds on diskette and CD-ROM *(including our acclaimed Signature Series artist collections)*. And this fall we're premiering CD-ROMs created for ENSONIQ by *Hollywood Edge, Sounds Good, InVision, Sonic Arts,* and *Q Up Arts;* as well as new volumes available from *Eye and I, Greytsounds, Pro-Rec,* and *East-West.*

Isn't it a great time to be alive and making music? ASR-10 owners certainly think so. So can you.
Call 1-800-553-5151 to find out more.

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G I N E

Score Display: Lyrics + chords + melody, chords, lyrics (Zoom)

Video Output: Direct connection with Computer Monitors or Televisions

Display: Large Neon Backlit graphic display (240 x 64 pixels)

Sequencer: 16 track, 1/192 Resolution, 250,000 Events, 8 Songs, Background Sound Loading, Realtime, Step by Step, Overdub, Quantize, Microscope Editing

Styles: 96 (64 ROM + Variation / 32 RAM + Variation User-programmable)

Sections/Split: 16 sections available real-time/16 Split/Layers

Keyboard: Lightly-weighted Action with Velocity and Aftertouch

Basically, this whole ad is trying to tell you that Generalmusic's **SX** keyboard is much better than the Korg i3. In case you still don't get it, here are the precise reasons why.

	Generalmusic SX2	Korg i3
SOUNDS	376 ROM+1672 RAM	256
SEQUENCER	250,000	40,000
SEQUENCER RESOLUTION	192 ppq	96 ppq
BATTERY-BACKED SEQ. RAM	Yes	No
RECORD TEMPO CHANGES	Yes	No
LOAD WHILE PLAYING	Yes	No
STYLES	96	48
STORE USER STYLES	32 + 32 Variations	4
USER PROFILE DRUMKITS	Unlimited	2
LYRIC / SCORE	Yes	No
VIDEO OUTPUT	Yes (option)	No
LOAD NEW SAMPLES	Yes (2MB option)	No
PRICE	Less	More

Generalmusic's New SX Multimedia Workstation
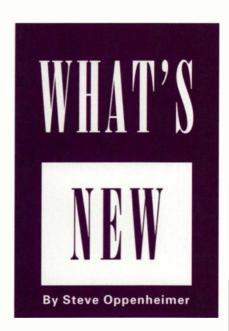
Sure, that would be enough - but wait, there's more!
Imagine, without using any complicated computer software, pressing a single button and seeing any sequencer track instantly displayed as a musical score. Plus, the unique **SX** "preload" feature allows you to load new sounds and sequences while playing. The **SX** can be expanded with 2MB of optional sample RAM, which allows you to load in your favorite digital samples from choirs, to guitars, to applause. And, the optional video interface provides the connection to most televisions or computer monitors, which allows you to share your lyrics, music, or both with other people, turning your every performance into a multimedia event. For even more versatility the **SX** is available in two keyboard versions.

The **SX2** is 61-note keyboard, while the **SX3** has the expanded 76-note keyboard, both with lightly-weighted actions.

The new **SX** from Generalmusic is possibly more keyboard than you ever imagined. Way better, way cool, lots of fun, and for a lot less money than you might think. Visit your Generalmusic dealer today or contact Generalmusic Corporation for more information and the dealer closest to you.

GENERALMUSIC CORPORATION
1164 TOWER LANE
BENSENVILLE, IL. 60106
1•800•323•0280

WHAT'S NEW

By Steve Oppenheimer

▲ SOUNDCRAFT SPIRIT FOLIO LITE

Soundcraft has announced the Spirit Folio Lite mixer ($369.95). This small board features four mono inputs and four stereo inputs, each of which has two aux sends; 2-band, fixed EQ; pan; input trim; PFL solo; and a rotary level pot. Aux 1 is switchable pre/post-fader, and aux 2 is post-fader. The 48V phantom power is globally switched.

The mono channels have both balanced XLR and balanced ¼-inch inputs, as well as TRS channel insert points. The stereo channels have left and right ¼-inch inputs only.

The master section includes one stereo aux return, a L/R mix level control, and a monitor control, all on rotary pots. A 2-track tape input, with level pot, can be routed to the stereo mix bus. A pair of 10-LED ladders indicate the mix, 2-track input, or PFL levels. Spirit/DOD; tel. (800) 255-4363 or (801) 568-7600; fax (801) 568-7662.

Circle #401 on Reader Service Card

▶ E-MU EMULATOR IV

E-mu Systems has announced the Emulator IV, its new top-of-the-line, stereo, digital sampler ($5,995). The 3U rack-mount module comes with 8 MB of RAM, expandable to 128 MB using standard SIMMs. The E-IV is 16-part multitimbral and 128-voice polyphonic, and it includes 128 sixth-order, resonant filters.

The user interface features a new, icon-based, 240 × 64 graphic display and supports an optional ASCII alphanumeric keyboard. The operating system is stored in Flash memory for quick boot-up and easy software upgrades. An onboard library-search function is also included.

In addition to retaining all the features of the company's E-IIIx, the E-IV offers matrix modulation, resample-while-play, load-while-play, and the ability to audition samples directly from disk. Pitch transposition is said to be virtually distortionless over a 10-octave range.

The unit samples at 22.05, 24, 44.1, and 48 kHz, using 16-bit, 128× oversampling Sigma-Delta A/D converters. Playback is at 44.1 or 48 kHz using separate, 18-bit DACs for each of the eight polyphonic

outputs. The main L/R outputs are on XLR and balanced, TRS, ¼-inch connectors, while the six sub-outputs are on balanced, TRS, ¼-inch jacks. Both AES/EBU and S/PDIF digital I/O are provided.

The sampler can read E-III, E-IIIx, Emax II, and Akai S1000/1100 samples and programs from any SCSI device, complete with keymaps and all program parameters that have equivalents in the E-IV.

Planned options include Macintosh remote-control software, up to 32 MB of Flash RAM sample memory, an Alesis-compatible, optical, 8-channel interface, internal digital effects processing, and additional MIDI In and Thru ports for 32-part multitimbral operation. E-mu Systems; tel. (408) 438-1921; fax (408) 438-8612.

Circle #402 on Reader Service Card

▶ MACKIE DESIGNS 24•E

Mackie Designs has announced the 24•E ($2,995), a 24-channel expander board for the company's 24•8 and 32•8 mixers. (The model 16•8 mixer cannot be expanded.) The 24•E is almost the same as the regular 24-channel board, but without the master section. It provides 24 full-featured input channels (with mic preamps, phantom power, 100 mm faders, switchable EQ, etc.) and 24 tape returns. The expander board connects to the master console with a proprietary multipin cable, and multiple expanders can be daisy-chained.

The expander can be used as a tabletop unit

or mounted on an optional floor stand like the master unit. The optional MB•E meter bridge ($695) provides 24 12-segment LED ladders, shielded by a Lexan cover. The bridge tilts up to 90 degrees and folds flat for transportation. Mackie Designs; tel. (206) 488-6483; fax (206) 487-4337.

Circle #403 on Reader Service Card

BETA Bio

SHURE BETA 87

PROFILE:

A new arrival that sets the world's highest standard for condenser mic performance. Sensitive. Tough. Extremely reliable. A perfectionist. Very smooth in any situation. Shrugs off abuse that would disable an ordinary condenser microphone.

GOALS:

Looking for opportunities to provide outstanding, studio-quality sound for serious musicians in live performance situations. Eager to travel.

SPECIAL STRENGTHS:

A hard worker with an extraordinarily tight supercardioid polar pattern consistent at all usable frequencies. Produces a balanced, natural and detailed sound in both wired and wireless versions. User friendly — rejects irritating feedback and cymbal leakage.

REFERENCES:

M. Jackson, P. Gabriel, D. Bowie, M.C. Carpenter, L. Lovett, B. Idol, T. Tritt, Sade, C. Glover, et al.

AVAILABILITY:

To arrange a personal audition, call 1-800-25-SHURE.

SHURE BETA

▼ ZOOM 1202

Zoom's 1U rack-mount 1202 Zoom Studio reverb ($249.99) offers true stereo or dual mono operation, a 44.1 kHz sampling rate, 16-bit converters, and 24-bit internal processing. It contains 512 editable programs (two banks of sixteen effects, each with sixteen variations), many derived from the company's high-end Model 9200 reverb.

One bank of presets includes reverbs, delays, and gated effects, including reverse reverbs and effects designed especially for vocal, orchestral, and percussion applications. The other bank produces modulation and other effects, including 2-octave pitch shifting, flanging, chorusing, and noise reduction. A 2-band EQ can be applied after any effect.

You can use one stereo effect at a time, or two discrete mono effects (one per channel), but some of the modulation effects also include echo or reverb. Two independent sets of editing controls let you adjust different parameters for each effect in dual-mono mode. Dedicated input, output, and wet/dry mix controls also are provided. Samson Technologies; tel. (516) 932-3810; fax (516) 932-3815.

Circle #404 on Reader Service Card

▶ SYMETRIX 602

Symetrix has introduced the 602 Stereo Digital Processor ($1,995). Designed for in-line studio and sound-reinforcement applications, the 602 is closely related to the company's model 601, but with a stereo line input. It provides simultaneous digital-domain dynamics processing (compression, limiting, AGC/leveling, de-essing, dynamic noise reduction, and downward expansion), stereo delay with modulation and feedback, and parametric EQ. All three EQ bands can be swept from 31 Hz to 21 kHz, with up to +18 dB boost and -50 dB cut.
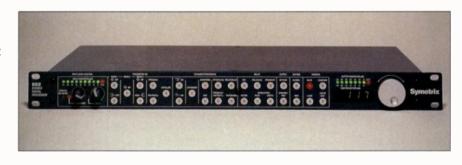
The dynamics processor's attack and release times range from 100 microseconds to ten seconds. The unit has a sidechain with an adjustable-frequency, lowpass/highpass filter for frequency-dependent processing, but there is no external Key Input.

In addition to its stereo analog inputs and outputs, the 602 has AES/EBU I/O on XLR jacks and S/PDIF I/O on RCA jacks. The unit always operates in Stereo mode—only the delay time has separate left and right controls—but you can digitally mix the two inputs or assign one input to feed both outputs. An output Pan control is also provided.
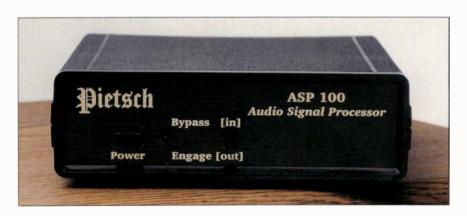
The 602 is fully MIDI programmable, with 128 ROM presets, 128 RAM presets, and SysEx dump and load capabilities. According to the manufacturer, parameter and program changes are glitch-free, with no interruption of the audio. The user interface offers 1-button access to most parameters. Symetrix; tel. (800) 288-8855 or (206) 787-3222; fax (206) 787-3211.

Circle #405 on Reader Service Card

▼ PIETSCH ASP 100

Pietsch Electronics has introduced the ASP 100 ($229.95), a signal processor designed to remove hum and enhance line-level signals for recording, sound-reinforcement, home theater, and post-production. The stereo device provides fixed notch filters at 60, 120, and 180 Hz designed to reduce hum by 95%. Adaptive logic is used to pass most music at the notch frequencies, while rejecting hum.

A harmonic enhancer is also included, which operates by modifying the transfer function between the input and output signals. According to the manufacturer, by selectively making loud sounds louder in a nonlinear fashion, the ASP 100 emphasizes lead vocals, drums, bass, etc., without coloring the sound or adding hiss or pumping.

All connectors are RCA jacks. Power is supplied by an external adapter (included). The ASP 100 measures $5 \times 5 \times 2$ inches and weighs just two pounds. Frequency response is rated at 10 Hz to 100 kHz and THD <0.3%. Pietsch Electronics; tel. (301) 540-2471; fax (301) 540-4197.

Circle #406 on Reader Service Card
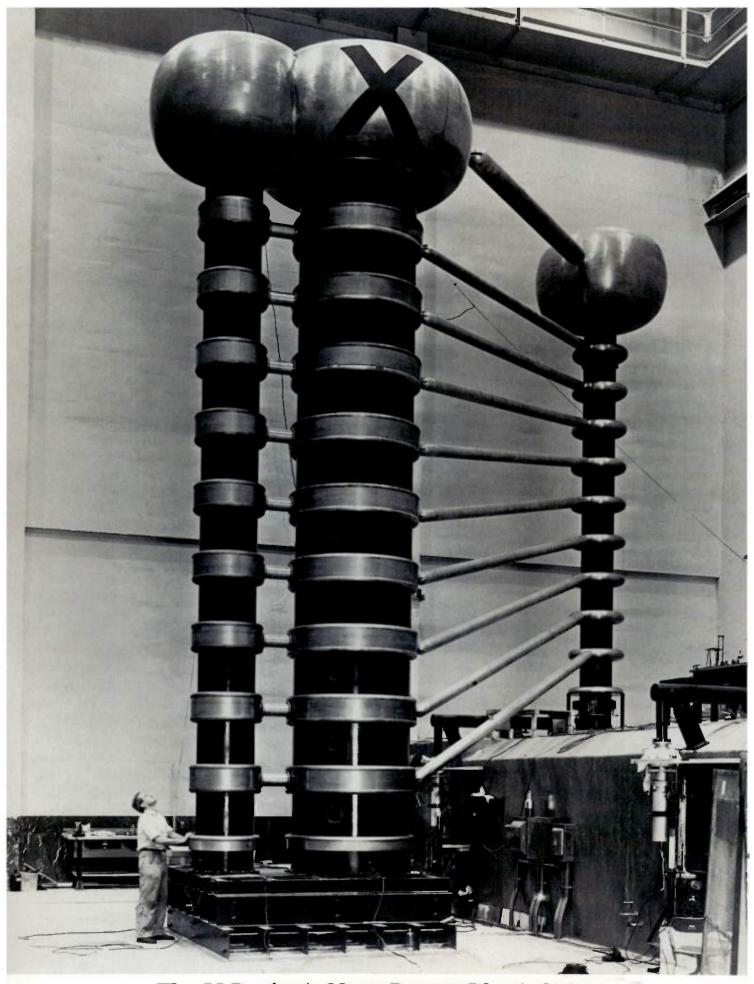
The X Project, Korg Power Plant, 1994.

REV UP ▲ ▲ ▲ ▲ ▲

▼ DISSIDENTS

dissidents is offering version 2.2 of its *Sample Wrench* 16-bit audio-file editor for the Commodore Amiga ($299; upgrades $20). Many DSP functions have been optimized to use less memory and run faster. Waveform drawing speed has been increased by up to 5,000% for large sound files, with no loss in accuracy. Edit Smoothing helps to create a soft transition into and out of spot edits, and support has also been added for 8- and 16-bit, mono or stereo WAV files.

Other new features include default save/load paths, easy loop creation via mouse, and the ability to interrupt many functions in midstream. dissidents; tel. (315) 797-0343.

Circle #407 on Reader Service Card

▲ OSC

OSC has announced *Deck II* 2.2 ($399; free upgrades from v. 2.1; upgrades from v. 2.0 $50; upgrades from *Deck* 1.x $150). The new version supports Apple's *Sound Manager,* which means you can use the software with any Mac that is capable of 16-bit audio, such as the Power Macintosh and PowerBook 500 series. No additional hardware is required.

Deck II 2.2 adds an open plug-in architecture, allowing it to run with compatible third-party DSP software. Finally, the new version adds new key commands, editing shortcuts, and support for 22 and 24 kHz sampling rates. OSC; tel. (415) 252-0460; fax (415) 252-0560.

Circle #408 on Reader Service Card

▼ OPCODE SYSTEMS

Opcode announced *Studio Vision Pro* 2.0.8 ($995; upgrades from v. 1.4 $149.95; from v. 1.5 $49.95; from v. 2.0 $15). The new version supports the latest version of *Digidesign Audio Engine (DAE;* included). This allows Opcode's integrated digital-audio recorder/MIDI sequencer program to use any Digidesign hardware, including Session 8. Opcode Systems; tel. (415) 856-3333; fax (415) 856-3332.

Circle #409 on Reader Service Card

BINARY SOUNDS

Binary Sounds has taken over marketing, distribution, and support of former Hybrid Arts programs *SmpteTrack, EditTrack, GenEdit,* and *EZ Score+* for the Atari ST. Binary Sounds; tel. and fax (713) 776-9118.

Circle #410 on Reader Service Card

▶ DIGITECH STUDIO VOCALIST

The Studio Vocalist ($995.95) is DigiTech's latest and most advanced vocal-harmony processor. The 2U rack-mount device generates up to 4-part vocal harmony (plus the original signal) from a single vocal input. It recognizes the singer's pitches and generates musically correct, chromatic, diatonic, or chordal harmonies based on standard or user-defined scales. It can also generate vocoder effects.

The Studio Vocalist comes with 99 factory programs in ROM, which include a variety of harmony styles, and it has 99 user program locations in static RAM. Editing parameters for each of the four individual voices include Detune, Vibrato, Scoop, and Timing. You

can adjust the volume of each harmony and the speed, depth, and attack of the vibrato.

Key changes and editing are performed with eight soft keys and a 1-octave, front-panel keyboard. You can step through preset harmonies in real time with a footswitch, and harmonies and individual notes can be triggered via MIDI.

The unit features 48 kHz sampling

and uses 18-bit, Sigma-Delta A/D converters. It has two parallel mic inputs, each with 48V phantom power and an aux send and return. Each harmony part has a separate, balanced, XLR audio output. A digital I/O expansion card is targeted for May 1995. DigiTech; tel. (800) 777-3637 or (801) 568-7600; fax (801) 568-7662.

Circle #411 on Reader Service Card
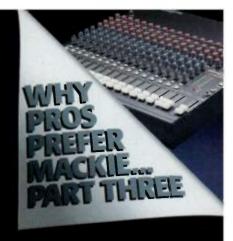
CHARLES THE MIXOLOGIST ROANE

D.C.'s hot urban contemporary station WPGC got its current #1 market standing by doing risky, innovative things. Like hiring Charles "The Mixologist" Roane & his trusty CR-1604. Roane's remix of "Whomp (There it is)" helped spur total sales of the single to over 6 million — making it the all time best selling rap record. Now record labels seek out The Mixologist for his "flava" and clean mixes.

Though he recently bought a Mackie 32•8 8•Bus console, Charles won't part with the CR-1604 that got him started. He calls it his "lucky mixer."

WHY PROS PREFER MACKIE... PART THREE

MACKIE REMIX MASTERS

BONZAI JIM CARUSO

Ever since he engineered five remixes on Madonna's smash "Erotica" album, Bonzai Jim Caruso's engineering career has been on fast forward. As he did on the Madonna sessions, he usually brings along his personal CR-1604 mixers.

The rest of Bonazi's resume speaks for itself: top-charted dance remixes for George Michael, Luther Vandross, Jon Secada, Gloria Estefan, The Basement Boyz, Natalie Cole, Class-X with Michelle Weeks and Jose Feliciano. All mixed on two Mackie CR-1604s.

Now let Bonazi speak: "I went through four other small mixers* before finding the Mackie. The others didn't even come close. The CR-1604 is the only mixer that can handle really huge drum sounds, monstrously fat bass and a ton of synths simultaneously."

Recently, while mixing Crystal (La-Da-Di La-Di-Da) Waters at the Basement Boyz' Baltimore studio, the SSL console didn't have enough inputs. So Bonzai rented a CR-1604 to help mix two cuts, "Ghetto Days" and "What I Need." The studio was so impressed with the Mackie CR-1604 that they immediately bought two!

Bonazi then used those two CR-1604s to mix Martha Wash's new remix album release "Leave A Lite On" — sans help from the SSL board.

MACKIE.

All of our mixers are made at 20205 144th Ave NE, Woodinville, WA, 98072, USA • For more info, call 800/268-6883 M-F, 9 to 5 PM PT • FAX 206/487-4337 Outside the US, call 206/487-4333 • Internet address: technical.support@mackie.wa.com • On CompuServe: GO MACKIE • Represented in Canada by S.F. Marketing 800/363-8855

*Although Bonzai specifically listed the other brands he'd tried and rejected, we don't mention direct competitors in our ads. If you want a hint, though, see who else is advertising compact mixers in this issue.

Announcing a much

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

Not just another pretty face, our new Anastasia font is loaded with symbols and is easily the most powerful and flexible anywhere.

Playback of dynamics and repeats gives students a sound lesson in the nuances and subtleties of music.
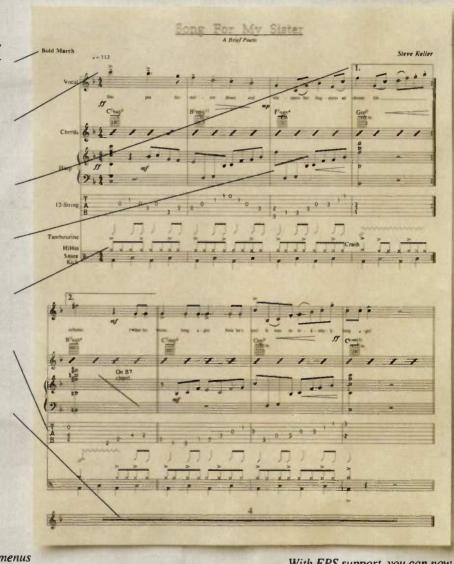
Avant garde musicians take note. Encore now supports cross staff beaming.

Talk about an unbeatable new feature. Our percussion staff is perfect for marching bands.

Here's something to please even the pickiest guitarist: Any music can be instantly turned into guitar tablature.

Finally there's a simple way to include compressed rests for groups playing from one piece of music.

Encore has the cleanest interface and the tastiest menus in the industry.

With EPS support, you can now place your compositions into other applications.

You can now have up to eight voices per staff.

If you can operate a tapedeck, you already have all the necessary expertise to operate our playback and record.

REV UP ▲ ▲ ▲ ▲ ▲

▶ ENSONIQ

Ensoniq is shipping version 2.51 of the operating system for its ASR-10 sampler on a free floppy disk. The new OS lets you backup files (in the form of a complete disk image) from SCSI drives to audio DAT via the optional DI-10 digital I/O interface ($399). The DAT can be verified to ensure the integrity of the backup. The files then can be restored from DAT to disk. This feature is compatible with most DAT recorders that have S/PDIF digital I/O. Ensoniq Corp.; tel. (610) 647-3930; fax (610) 647-8908.

Circle #412 on Reader Service Card

▼ EVENTIDE

Eventide has introduced two upgrade packages for the H3000SE Ultra Harmonizer. The H3000-D/SE ($2,695; upgrades from $340) adds 3-D spatial imaging programs,

Mod Factory dynamic algorithms with 100 presets, and 100 new studio presets created by well-known industry pros. The 3-D, speaker-based, spatial imaging effects were created specifically for 3-D positioning, rather than adding a dimension to existing presets. The Mod Factory algorithms, introduced in the H3500, let you patch together delays, filters, pitch shifters, envelopes, modulators, and mixers to create custom, dynamic effects.

The H3000-D/SX ($1,995; upgrades from $340) also adds Mod Factory algorithms with 100 presets plus 90 new guitar-oriented presets written

by top guitarists. But instead of 3-D processing, it adds a 6-octave Multi-Shift pitch shifter and a 19-waveform (plus pink and white noise) LFO function generator, which is assignable to almost any parameter in the unit. Eventide; tel. (201) 641-1200; fax (201) 641-1640.

Circle #413 on Reader Service Card

▼ AKAI

Akai Professional has announced a software upgrade and a modification kit for the CD3000 sampler. The KIT-CD3000 modification ($300, factory-installed) adds stereo analog sampling inputs, a rotary input-level control knob, and new screen functions for accessing the new analog inputs.

Operating system version 1.6 ($52) lets the CD3000 use the SB3001P Piano Sound Board option ($250). The sound board provides 8 MB of ROM-based keyboard samples, such as acoustic and electric piano, harpsichord, and clavinet. The option board is installed into one of the four memory-expansion slots.
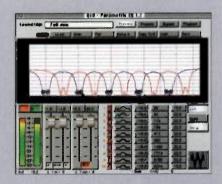
Version 1.6 also adds portamento, which can be set for each program. Rate portamento lets you vary the length of the glide depending on the distance between notes. Time portamento lets you vary the length of the glide, but the selected length stays

the same regardless of the distance between notes.

Akai simultaneously announced operating systems 1.5 and 2.0 for the S2800, S3000, and S3200 samplers. OS 1.5 ($52) supports the SB3001P Piano Sound Board option and adds portamento. OS 2.0 (price tba) lets the samplers read Roland and E-mu sound-library disks from floppy or SCSI disk. Akai/IMC; tel. (817) 336-5114; fax (817) 870-1271.

Circle #414 on Reader Service Card

▲ WAVES

Waves has announced version 1.1 of its *Q10* ($450; upgrades $50), a 10-band EQ software plug-in for Digidesign's TDM and *Sound Designer II*. The new version features a new EQ Setup Library with more than 80 preset Setups optimized for specific applications, such as multimedia, mastering, and post-production.

Waves has also added its proprietary Increased Digital Resolution (IDR) process, which combines noise-shaping and dithering to improve digital signal resolution. Also new are Phase Inversion, Channel Swap, and AutoTrim, which provides gain normalization during processing.

Finally, version 1.1 adds editing and direct keyboard entry of all parameters. The left and right channels have also been color-coded. Waves; tel. (615) 588-9307; (615) 588-9472. ●

Circle #415 on Reader Service Card

Write to MIDI, no keyboards.
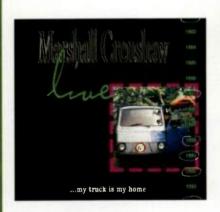
Finally. No more running back and forth between fretboard and keyboard every time you compose with MIDI.
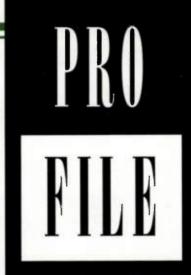
Because with G-VOX Bridge™ software, you can go direct from your guitar, via the G-VOX Pickup and Belt Pack, to top Windows-based sequencing and notation software.* Meaning you can write, record, edit and save to MIDI with a package designed expressly for guitarists.

Bridge is fast, accurate and easily conforms to your playing style. What's more, Bridge comes with Midisoft's Recording Session™ and Howling Dog's Power Chords™ for G-VOX. Absolutely free.

So get started. See your local dealer or just call (215)922-0880 for more information. Then go write to MIDI. With your guitar.

...my truck is my home

PRO FILE

It's Alive!

Marshall Crenshaw's concert confessions.

By Michael Molenda

To musicians, part of the fun of a live concert album is arguing with the band's fans about how much of the record was actually recorded "live." You see, it's common practice to sneak live recordings into the studio and redo bungled guitar parts, off-pitch vocals, and other less-than-sterling examples of an artist's performance prowess. Some venerated live albums are rumored to be complete studio re-recordings, with only the audience's cheers saved from the actual concert.

So, depending on your point of view, Marshall Crenshaw is either a savior or a spoiler. The veteran singer/songwriter is spilling the beans about his new live album, *My Truck Is My Home* (Razor & Tie Records). For starters, half of the live album's fourteen songs were actually recorded direct to cassette (or DAT) right off the stereo bus of the house mixing console by the coproducer of *My Truck Is My Home,* Will Schillinger.

For the curious, the live-to-cassette songs are: "Tonight," "Have You Seen Her Face," "There She Goes Again," "Girls," and "Twine Time." ("If you listen to 'Twine Time,' you'll notice that the kick drum is missing in the beginning of the tune," reveals Crenshaw. "The mic didn't come on until midway through the song, but the overall track was really good, so we kept it.") "Cynical Girl" was also recorded direct-to-cassette, but Crenshaw overdubbed a tambourine part while he was transferring the track to a DAT master. The live-to-DAT numbers are "Vague Memory" and "Julie."

But a few songs on *My Truck Is My Home* are far from honest documentations of a concert performance. Crenshaw readily confesses that every trick in the book was used to improve some tracks.
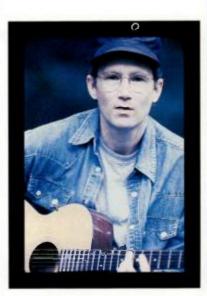
"Fantastic Planet of Love," for example, was recorded live to an Alesis ADAT. However, some pitch problems compromised the background vocals, and Crenshaw wasn't entirely happy with his lead vocal. To facilitate overdubs, the ADAT tracks were transferred to analog 24-track at Pilot Recording Studios in New York (which is owned by Schillinger). The background vocals were completely recut, and Crenshaw recorded another lead vocal. The vocal you hear on the CD is a composite of approximately half the original live version and half the studio track.

"I also asked Graham [Maby] to redo the bass, because I wanted a part that swung a little more than what he played at the gig," says Crenshaw.
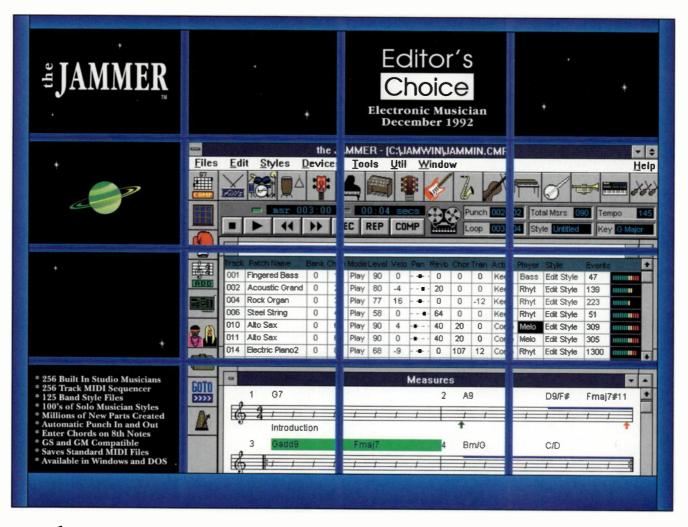
Two of the songs on the record are completely bogus concert tracks. (Esther's East, the club credited on the album, doesn't exist.) "Calling Out For Love (At Crying Time)" and "Knowing Me, Knowing You" were recorded at Pilot studios, because Crenshaw wasn't happy with any of the actual concert versions.

"We're just one step removed from reality here," says Crenshaw. "I played through a P.A. system set up in the studio with my full band. There were no punch-ins or overdubs, so they're still live tracks, they just weren't performed in front of audiences." ●

Marshall Crenshaw

The

By Teri Danz and Michael Brown

Interactive Concert Stage

Multimedia changes the face of live performance.
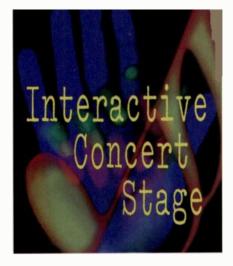
Imagine the scene from *The Sorcerer's Apprentice* where Mickey dons his master's magical cloak and cap and commands an army of brooms to do his bidding. Now imagine yourself in live performance. Donning the cloak and cap of new technology, you dim the stage lights with a nod of your head, trigger the playback of three-dimensional computer animations with a flick of your wrist, and switch between live camera feeds by slapping on drums and other percussion. You are commanding a symphony of sound *and* vivid imagery.

Welcome to the new frontier: interactive live performance. Except for the work of a few cutting-edge performance artists, and a few well-heeled mega-bands, this area remains largely unexplored. But get ready. The cost of the technology that makes inter-active live performance possible is becoming less and less of a barrier. At the same time, new products are coming to market that are taking the concept of interactive beyond anything that's been possible.

ILLUSTRATION BY ERIC ADEGARD

Interactive Concert Stage

SAME HAT, NEW TRICKS

Using prerecorded MIDI sequences to augment live performance is nothing new; musicians having been doing it for years. Using MIDI to trigger devices other than drum machines, synthesizers, and sound modules isn't new, either. MIDI lighting rigs have been around for some time, and theater stages in several major cities have been completely automated using MIDI Show Control (see "The Show Must Go On" in the September 1992 **EM**).

Over the course of the last few years, however, several other critical elements have fallen into place. Among them are the addition of robust MIDI support to high-level multimedia authoring systems, significant drops in the prices of microprocessors and other electronic components, and the introduction of more powerful versions of MIDI sequencer software. These developments are bringing interactive live performance out of the realm of fantasy and into the mainstream.

LIVE-ACTION MAGIC

Audiences at the 1993 WOMAD (World of Music, Arts, and Dance) tour were treated to state-of-the-art multimedia systems on and off the stage. One of the most interesting aspects of both systems is that musical instruments were used to control the computers, instead of using computers to control musical instruments. And in the midst of all the high-tech automation, human beings were the most important elements of these systems.

The San Francisco leg of the tour featured an elaborate, onstage, interactive video system, created by the Bay Area multimedia production company Magnum Design. The system featured a drumKAT MIDI percussion controller that tour sponsor and star performer Peter Gabriel used to control the display of video and computer-graphic images on a 15 × 15-foot video wall.

"Drums—in this case, a drumKAT—were the obvious choice for a trigger device," says Lawrence Doyle, an independent designer who consulted for Magnum on the system's design. "You can see the performer raise his or her arms to play them. They're much more dramatic than keyboards."

The video wall, an array of 25 37-inch video monitors, was driven by a Truevision NuVista+ graphics engine installed in the NuBus slot of a Macintosh Quadra 800. Three video sources—the Cyberlab Swarmcam (more on this later) and two laserdisc players—fed the NuVista+. The card functioned as a video switcher; selecting and routing one of the three video signals to its three video-out ports (see **Fig. 1**).

The NuVista+ can overlay computer-generated text and graphics onto live video, a process known as digital linear keying. It also can overlay live video over computer-generated graphics, a process known as digital chroma keying. Typically, a user sitting at the Mac would switch between the three video sources using Truevision's *VIDI/O* control panel software and access the two keying modes through its *ChromaKey* and *Blender* control panel software.

In this system, however, the band controlled the NuVista+'s display modes by tapping on the drumKAT's pads. Empowering the band to interact with and control the display on the video wall, instead of leaving it under the command of an off-stage operator, made for a more dynamic and exciting performance.

THE X FILES

Doyle decided to take advantage of the XCMD (X-Command, for External Command) and XOBJ (X-Object, for External Object) support offered by the NuVista+ card and Macromedia's *Director* multimedia authoring system when designing the performance system. XCMDs and XOBJs are small, compiled programs that extend the functionality of the programs that support them.

In this case, Doyle took the NuVista+ XCMDs and XOBJs and embedded them in *Director*. Once he did that, the same commands that were in Truevision's *VIDI/O*, *ChromaKey,* and *Blender* control panel software appeared as commands in Lingo, *Director*'s scripting language.

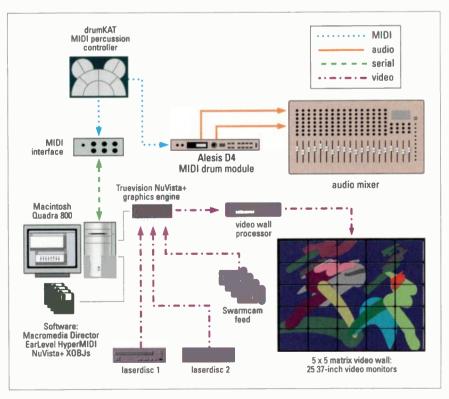Doyle wrote his Lingo script so that a MIDI Note On message generated by

FIG. 1: With this elaborate interactive multimedia system on the stage, performers can control the images displayed on a video wall by tapping on a drumKAT MIDI percussion controller.
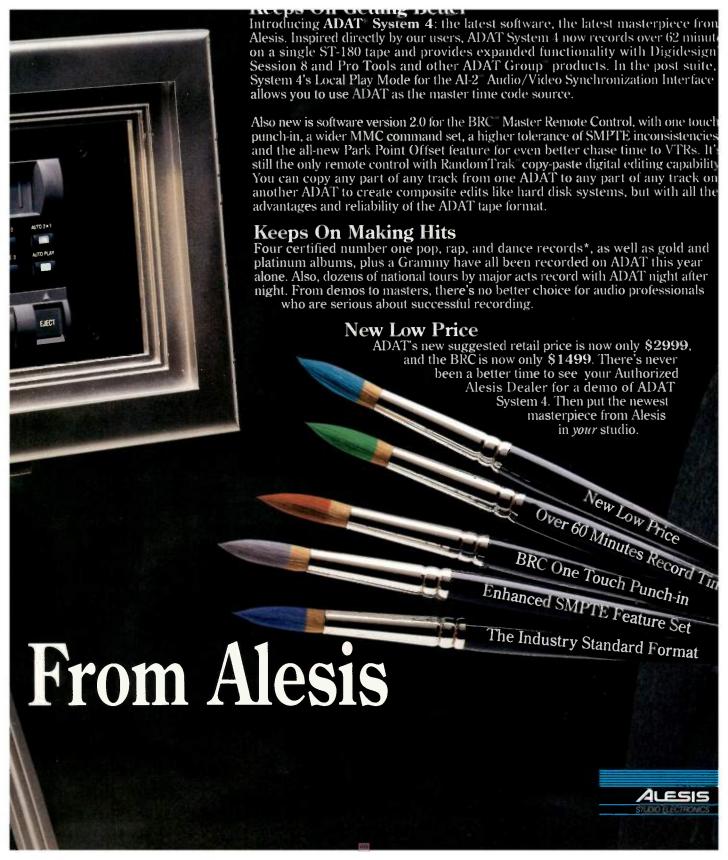
ADAT Sy

The Latest Masterpiece

Interactive Concert Stage

the drumKAT sent a command to the NuVista+ card or controlled some other aspect of what the Quadra was doing. Each tap on Pad 1, for example, caused the NuVista+ to cycle through its three video inputs: The first tap routed the video output from Laserdisc 1 to the video wall display; the next switched the display to the output of Laserdisc 2; the next switched to the Swarmcam feed; and the next switched back to Laserdisc 1.

Each of the drumKAT's nine other pads were similarly mapped. Hitting Pad 5, for example, played back an animation that was stored on the Quadra's hard drive. A tap on Pad 6 called up a series of photographs that were taken and digitized during the day. A slap on Pad 9 overlaid computer-generated graphics on top of certain areas of the video, and so on.

TRANSLATE, PLEASE

Because *Director* does not understand MIDI, Doyle used EarLevel Engineering's *HyperMIDI* program to interpret the MIDI messages generated by the drumKAT.

"*HyperMIDI* also enabled us to make the system respond faster," says Doyle. "The only thing we needed from the drumKAT was the Key Number segment of the MIDI Note On message, so we used *HyperMIDI* to filter out everything else. Then we set *Director* to respond to the ten Key Numbers that the drumKAT was sending."

THE SWARMCAM

The onstage performers weren't the only ones jamming with multimedia technology at WOMAD. Thanks to the

Swarmcam system, designed by Dan Mapes and his Cyberlab 7 crew, the crowds grooved with a multimedia melange of their own making.

Unlike the Magnum Designs system, which was featured only at the San Francisco event, the more elaborate Swarmcam was featured at each of the ten stops on the WOMAD tour.

The Swarmcam consisted of eight Sharp Viewcam Hi-8 camcorders and a complex network of video switchers, laserdisc players, personal computers,
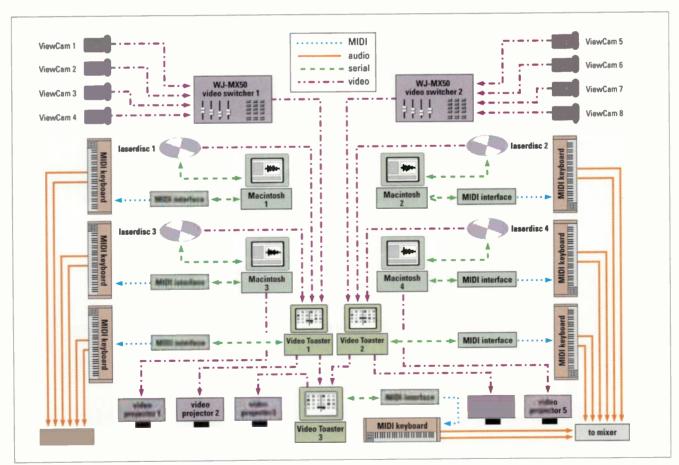

FIG. 2: Thanks to the Swarmcam, audiences at the 1993 WOMAD festivals became performers, too. In this system, which used three different computer platforms (Amiga, Atari, and Macintosh), as many as seventeen people could collaborate to produce their own music videos in real time.

Studiomaster

Interactive Concert Stage

MIDI keyboards, sequencers, and video projectors. (See **Fig. 2**; because of the complexity of the Swarmcam system, it was not possible to display every one of its elements.) Instead of having artists onstage performing for a relatively passive audience, the Swarmcam represented a massive collaborative effort between artists and audience. The entire system was controlled by members of the audience, assisted at each station by people who Mapes describes as "knowledgeable facilitators."

"It was an audience-mediated, artist-supported environment," says Mapes. "People weren't just coming in and facing raw technology. We built everything so that they could play with the tools and have something cool come out."

THE COMPONENTS

The Viewcams that formed the front-end of the Swarmcam system don't use the tiny, black and white viewfinders found on most camcorders. Instead, they use 4-inch color flat-panel LCDs. Shooting video with a conventional camcorder means holding the device directly in front of your face and blocking your view of everything else around you. With the Viewcam, you can hold the camera away from your body at almost any angle. That specific feature made it the perfect choice for the performance-art-oriented Swarmcam.

The video signals from the eight Viewcams (divided into two banks of four) were routed through two Panasonic WJ-MX50 video switchers. The camera operators could watch what they were shooting on their own viewfinder, but the operators of the WJ-MX50s determined which video signals would be passed on to the next station in the chain (one of two NewTek Video Toaster 4000s) and what that video would look like when it got there. (The WJ-MX50 can perform a variety of visual effects on a video sig-

nal, including colorizing and keying.)

Four audience members, each playing a customized membrane-switch MIDI keyboard, controlled one of four Macintosh Quadra 800s. In turn, each of the Quadras controlled a laserdisc player plugged into one of its serial ports. A custom program developed using Opcode's *Max*, an interactive graphical programming environment, triggered the laserdiscs based on the MIDI Note On messages generated by the keyboards.

"Our main intention was to eliminate QWERTY keyboards," says Mapes. "All of the computers were controlled solely by MIDI keyboards."

The video outputs from the four laserdisc players were routed to the next stations in the chain (one of the same two Video Toasters described above). The video outputs from two of the four Quadras were routed directly to one of five video projectors.

Each of the two Video Toasters routed one of their video outputs to a third Video Toaster and an identical second output to a video projector. The third Toaster output its video to a fifth video projector. All three of the Video Toasters were controlled by audience members playing MIDI keyboards. Mapes turned to The Blue Ribbon SoundWorks' *Bars & Pipes Professional* MIDI sequencer for the Amiga for this part of the system.

"Using *Bars & Pipes Professional*," says Mapes, "we were able to trigger various Video Toaster effects and switch between the various video inputs right from the MIDI keyboards. Depending on which notes were hit on the keyboards, the images projected on the screen might be any combination of the Viewcams, the laserdiscs, or the Macintoshes."

WHO'S IN CHARGE?

Because no single device exercised complete control over the entire system, no one ever knew what would ultimately appear on the five video screens after all the visuals had finished cascading through the system.

A single Macintosh running Opcode's *Studio Vision Pro* digital-audio sequencer played a sequenced groove that accompanied the solo work being generated by the seven MIDI keyboard players. Of course, having a constant stream of nonmusicians banging away on all those keyboards would have resulted in cacophony. Mapes' solution was to route the second MIDI Out from each of the seven multiport MIDI interfaces to a multiport MIDI interface plugged into an Atari Falcon030.

The Falcon030, running Hotz Instrument Technology's *Hotz MIDI Translator*, remapped the MIDI data from all seven MIDI keyboards in real time, so that all the notes played fit
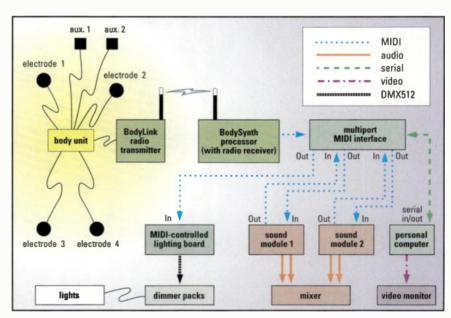
FIG. 3: Wearing a BodySynth, a dancer becomes a human, alternate MIDI controller. By converting the electrical signals generated by muscle contractions into either MIDI or RS-232 datastreams, the BodySynth enables the dancer to use his or her body to control MIDI sound modules, lighting rigs, laserdisc players, or even software running on a personal computer.

Interactive Concert Stage

within the key and chord structure of the sequenced groove *Studio Vision* was playing. The soloists couldn't hit a wrong note.

"It was like an interactive, collaborative, multimedia jazz ensemble," says Mapes. "We had seventeen people playing visual music together."

CARE TO DANCE?

Like many musicians, the CyberBeat Brothers, (dancer/sound designer Chris Van Raalte and dancer/singer John Zane-Cheong) use their muscles to create music, but they don't use them to slap drums, pound keyboards,

FIG. 4: Chris Van Raalte of the CyberBeat Brothers demonstrates the BodySynth, an alternate MIDI controller that picks up the electrical signals generated by muscle tissue contractions.

or pluck strings. These performers *wear* their instruments, and they generate MIDI messages with the electrical energy generated by the contraction of

their muscles.

Van Raalte's and Zane-Cheong's instrument is the BodySynth (see **Fig. 3** and **Fig. 4**). Developed over a six-year

period by Van Raalte and engineer Ed Severinghaus, the BodySynth is a modular instrument that can be configured in different ways. The most common setup is for the performer to wear the Body Unit, BodyLink radio transmitter, and four electrodes. The BodySynth processor (and its radio receiver) are placed within a cable-length of the devices the processor will be controlling.

To play the BodySynth, the performer attaches the instrument's electrodes directly to the skin over different major muscles. Each electrode picks up the tiny electromyographic (EMG) signals generated by the contraction of muscle tissue.

Each of the signals from the four electrodes are transmitted to the Body Unit, which amplifies them and sends them to the BodyLink (see **Fig. 3**). From there, the signals are transmitted over the airwaves to the BodySynth processor. The signals from each of the four electrodes—as well as each of the signals from as many as eight auxiliary devices—are multiplexed into a single channel for transmission. The signals are de-multiplexed by the BodySynth
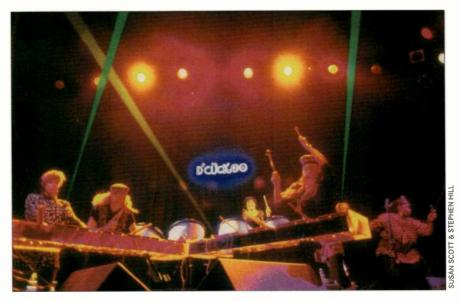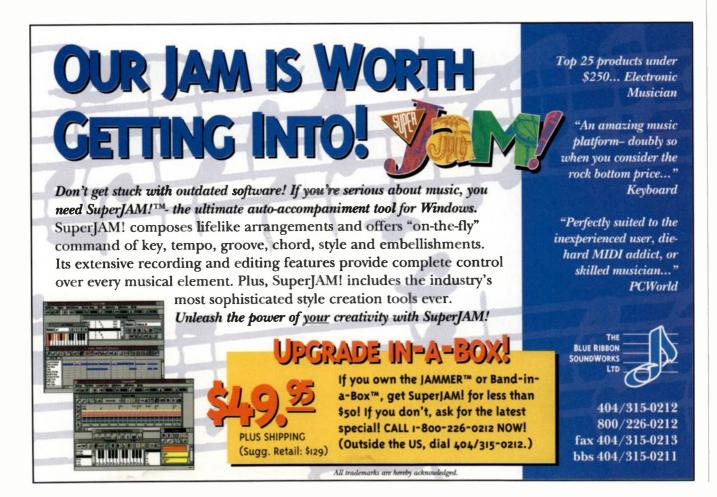
FIG. 5: Candice Pacheco, Jennifer Hruska, Janelle Burdell, and Tina "Bean" Blaine of D'Cückoo designed and built their own custom MIDI instruments. (Also pictured: Terri Brown.)
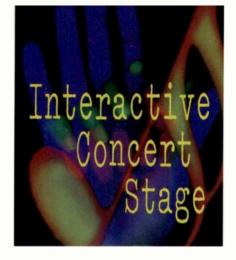
processor at the receiving end.

"We once put a pair of force-sensitive resistors into a tap dancer's shoes," says Severinghaus, explaining one use for the auxiliary Body Unit inputs. "By using the inputs that way, he could trigger MIDI devices while he danced."

Van Raalte typically wears one electrode on each his calves and one on each of his forearms. He uses his legs to establish a rhythm track and his upper body to trigger other instruments and sounds.

Interactive Concert Stage

"The BodySynth is like any new instrument," says Van Raalte. "There's a learning curve you go through in terms of mastering your timing."

DIVISION OF LABOR

As far as the equipment is concerned, the BodySynth processor is the device that performs most of the work, according to Severinghaus. Powered by a Motorola 68HC11 embedded controller (a microprocessor more commonly used in automobile engine monitors), the BodySynth processor converts the EMG signals picked up by the electrodes and the electrical impulses generated by the auxillary devices into MIDI or other protocols. The device has two ports: a MIDI Out and a RS-232 serial port for controlling devices such as professional VCRs.

Like many other multimedia artists who work with the Macintosh, Severinghaus and Van Raalte have used Opcode's *Max* to develop customized programs for their productions.

"We've taken MIDI messages from the BodySynth," says Severinghaus, "processed them through a program we developed using *Max,* and used them to control a laserdisc player. Chris also performed in a production at San Francisco's New Music Theater in which he used the BodySynth to control the playback of a *QuickTime* movie."

He was able to control the direction in which the movie played and even the speed at which it was played just by raising and lowering his arms."

HAVE A (MIDI) BALL

Never content to sit back and wait for someone else's cue, Candice Pacheco, Jennifer Hruska, Janelle Burdell, and Tina "Bean" Blaine of D'Cŭckoo are pioneers in interactive live performance (see **Fig. 5**). The band has been including the audience in the creative process during live performances for several years.

In addition to designing and building their own custom instruments, including MIDI marimbas and 6-foot bamboo MIDI "trigger sticks," the band created the MIDI Ball, a huge, helium-filled sphere studded with MIDI triggers and a wireless transmitter. During their performances, the band throws the MIDI Ball out into the crowd for the audience to play. As the band lays down a groove, the MIDI Ball bounces across the audience like a giant, electronic volleyball, and every hand that touches it triggers new sounds and video images.

ENDLESS LOOP

Imagine yourself a visionary back in 1983, when the MIDI spec was first defined. "One day," you prophesy, "the human body will become an alternate MIDI controller. MIDI instruments will be used to drive video wall displays. And nonmusicians will be able to solo expertly over prerecorded MIDI sequences, using keyboards that have no keys. A computer will remap in real time all the notes that they play, so they will never be out of key."

You probably would have been dismissed as a crackpot. ("Yeah, sure Nostradamus—and one day Jimmy Carter and Richard Nixon will be respected elder statesmen!") But those things and more have come to pass. What do these developments mean to the average musician? First and foremost, they represent a brand new set of tools (or toys, depending on your perspective) for musicians who perform live. And interactive live performance opens up broad new avenues for creative expression. Now, just imagine what the next ten years will bring.

Teri Danz is a San Francisco-based singer/songwriter.

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Bars & Pipes Professional 2.5 (Amiga)
List price: $429
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or (404) 315-0212
fax (404) 315-0213

BodySynth
(Price depends on configuration)
Ed Severinghaus
tel. (415) 387-1142
e-mail ed.s@sfnet.com

Director 4.0 (Mac)
List price: $1,195
Macromedia, Inc.
tel. (415) 252-2000
fax (415) 626-6554

drumKAT 3.5
List price: $1,149
KAT MIDI Devices
tel. (516) 481-3004
fax (516) 481-8467

Hotz MIDI Translator (Atari)
List price: $299.95
Hotz Instrument Technology
tel. (805) 492-5553
fax (805) 493-4650

HyperMIDI 3.0 (Mac)
List price: $195
EarLevel Engineering
tel. (310) 316-2939
fax. (310) 540-8459

Max 2.5 (Mac)
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Studio Vision Pro 2.08 (Mac)
List price: $995
Opcode Systems, Inc.
tel. (415) 856-3333
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NuVista+ graphics engine (Mac)
List price: $2,995
Truevision, Inc.
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Video Toaster 4000 (Amiga)
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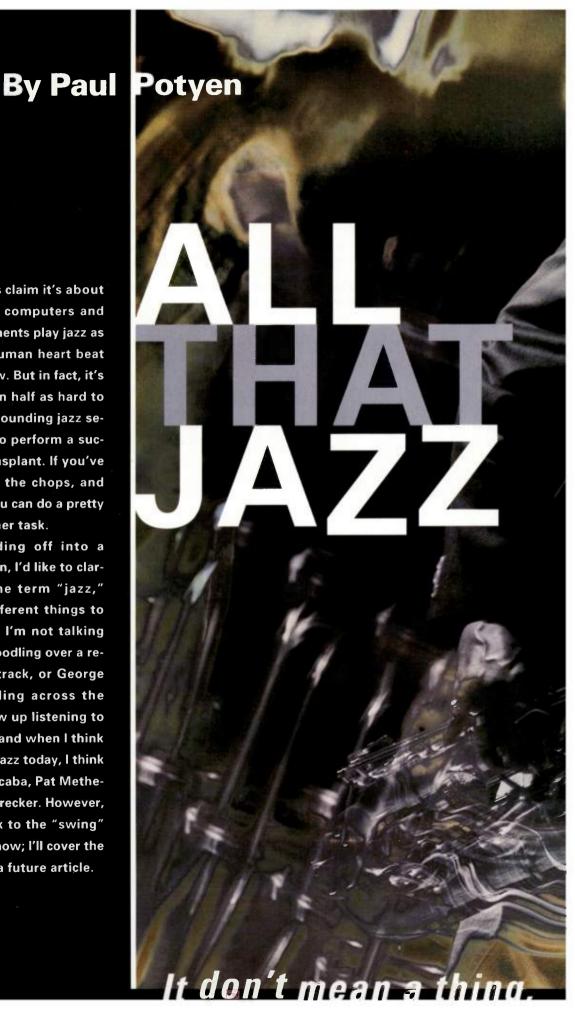
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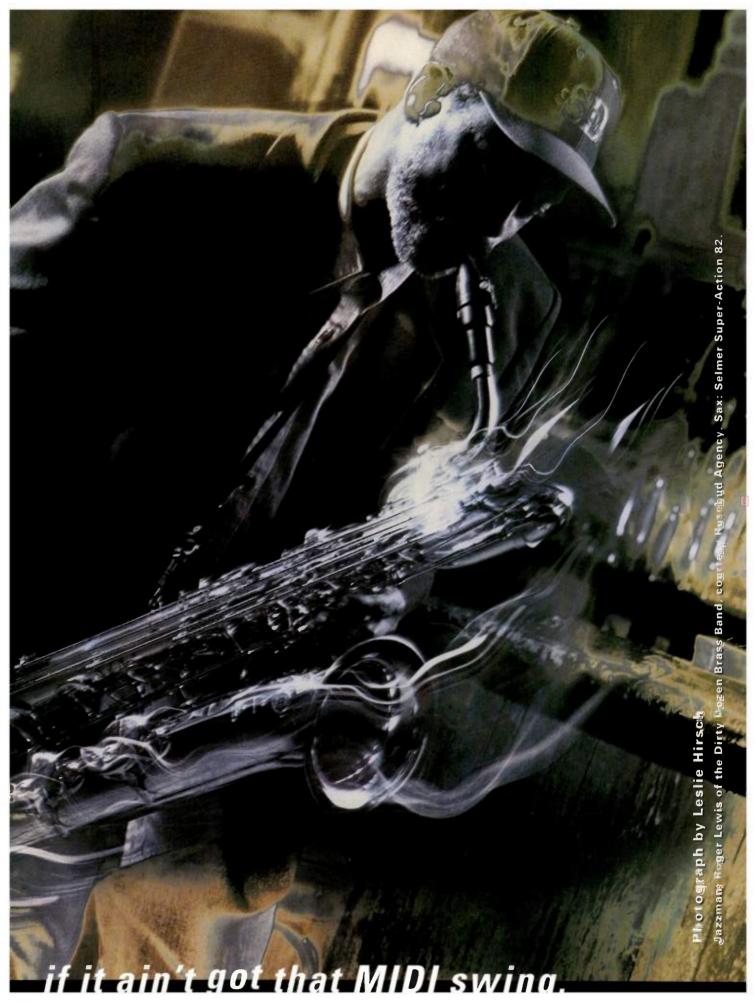
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By Paul Potyen

ALL THAT JAZZ

ome jazz purists claim it's about s easy to make computers and lectronic instruments play jazz as is to make a human heart beat side a scarecrow. But in fact, it's robably not even half as hard to ake authentic-sounding jazz se- uences as it is to perform a suc- essful heart transplant. If you've ot the interest, the chops, and e right tools, you can do a pretty ecent job of either task.

Before bounding off into a ngthy discussion, I'd like to clar- y my use of the term "jazz," hich means different things to ifferent people. I'm not talking out Kenny G noodling over a re- eating rhythm track, or George inston stumbling across the hite keys. I grew up listening to iles and Monk, and when I think f contemporary jazz today, I think f Gonzalo Rubalcaba, Pat Methe- y, and Michael Brecker. However, m going to stick to the "swing" tyles of jazz for now; I'll cover the tin varieties in a future article.

It don't mean a thing,

if it ain't got that MIDI swing.

ALL THAT JAZZ

BEGIN THE BEGUINE

Probably more than any other musical genre, jazz is about improvisation. But how do you capture the improvisational essence of jazz in a sequencer? The answer is *not* "very slowly." Don't even think about entering "Dolphin Dance" in step time. MIDI keyboard and other controllers are the most practical way to render realistic-sounding, sequenced performances. The more accomplished you are at playing your controller, the better your results will be. Not only are the timing aspects difficult to achieve in step time, but the *dynamic* aspects of a jazz performance can be realistically captured only in real time.

Even if your technique rivals that of Keith Jarrett, an awareness of jazz that comes from decades (okay, years) of analytical listening is critical. (A good friend, with whom I recently completed production on a third jazz album, tells her aspiring voice students, "First, you have to dedicate your life to this music.") The same idea applies here, but there are some additional tips and tricks that can help enhance your music. Jazz is not easy to teach; some of the most renowned musicians were great because they ignored or weren't aware of the rules. So here are some rules. Who knows? If you break them, you may become as rich and famous as Richard Twardzik.

Most people agree that music consists of melody, harmony, and rhythm. Of the three, rhythm is near the top of the list in the jazz idiom, so I like to start by recording a bass part. There are several reasons for this approach. First, it's easier and faster to record a complete-sounding bass part than a complete-sounding drum part. Second, like drums, the bass usually plays a fairly regular, steady pattern throughout a tune. Third, *unlike* drums and percussion, the bass part provides a much better way to keep track of where you are in a tune, because it gives you a harmonic *and* rhythmic foundation. Occasionally, I put down a piano ref-erence track to help establish the form of a piece and create an environment for playing the bass part, but I don't spend a lot of time on it.

I find it helpful (but not necessary) to use a lead sheet when I put my tracks together. I prefer not to copy and paste similar choruses, except as a scratch track to define the overall structure and length of the tune. I also try to perform the track at the right tempo. If I play a clam, I make a judgment call on the fly about whether to stop and punch in or go back and correct an offensive or missing note later. If I absolutely can't cut the part at the intended tempo, I slow it down in record mode and check the "feel" at the right tempo.

There is a parallel to this approach in the professional recording world. The best jazz producers get the right musicians together with the right music in the right environment and use a minimum of overdubbing and high-tech tricks, a much different scenario than the world of pop.

QUANTIZE WITH CARE

When the bass part is done—from top to bottom—I listen repeatedly to the track with the sequencer's metronome and quantize only those notes/passages that don't feel right. Most sequencers allow you to quantize to a specified "strength." *Never* quantize to 100%; if you can avoid quantizing entirely, that's even better for the feel of the tune.

For a common, walking bass pattern, I usually quantize only those quarter notes that fall behind the beat, using a quantization value of 50%. If that's not enough, I can always quantize it again. If a whole phrase sounds late, I look at the notes in graphic-editing mode, grab them with the mouse, and manually move them closer to the grid. That way, at least I haven't destroyed the integrity of the phrase.

If you just can't get the right feel by playing in real (or almost real) time, you can try quantizing all notes. On a swing tune, the bass rarely plays anything faster than eighth notes, so quantize to the nearest eighth note with a certain "swing" value. But here's the rub: The amount of swing varies according to the tempo. At slower tempos, use a triplet feel for the bass (see Fig. 1), quantizing to eighth-note triplets at a value of 50%. (You'll run into trouble if your part includes triplets and you quantize to eighth notes with a swing value of 66%.)

A bass player plays differently (read "not as many notes per bar") at a faster tempo, so it's a good idea to avoid changing the tempo and quantization values. For faster tempos (above approximately 200 bpm), where bass triplets sound unnatural, quantize to eighth notes, with a swing value under 66%. The faster the tempo, the less swing you should use. Again, I prefer to use as little quantization as possible for this type of music.

The bass part is literally the rhythmic and harmonic foundation of the tune, so perfecting the bass track is time well spent. Today's synths usually include at least one good electric bass sound, and even the acoustic bass sounds are generally respectable. With
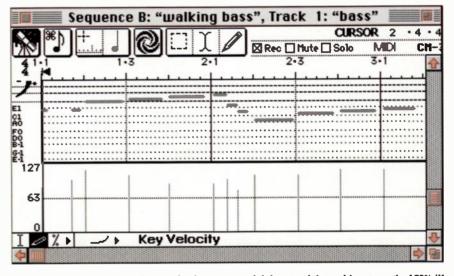

FIG. 1: At a slow tempo, quantize a swing bass part to eighth-note triplets with a strength of 50% (if you absolutely must quantize the entire part).

ALL THAT JAZZ

the proper knowledge of what this instrument can do—its range, when and how to play the "open" strings, and what patterns can be played naturally—you should be able to create a good bass track. Unfortunately, you'll never get the sound of snapping acoustic bass strings, or that growling E string you hear with the genuine item.

HARMONY

Now that you have the bass "mortar" finished, it's time to start adding the bricks. I generally choose a polyphonic instrument, such as piano, guitar, or vibes. However, sometimes I first put down a basic hi-hat, ride cymbal, or conga drum to provide more energy and momentum to the track, especially if I intend to record a solo. But for now, let's concentrate on the harmonic aspect of the track.

If I recorded a reference piano track, I usually erase it, although sometimes it pays to keep the parts you like and build around them. (After all, it's only a sequence, so you can keep it and mute it.) I start at the top and comp at the intended tempo through the whole piece, stopping only if I have a complete train wreck. The essence of this music is spontaneity, and the more you piece the parts together, the more disconnected and awkward they sound.

Modern jazz comping is the subject

of a different article (if not an entire book!), but my rule of thumb is this: Unless you're playing a Freddy Green-type guitar comp, you probably shouldn't quantize. If you must, use the suggestions outlined for the bass part. However, a truly creative jazz comp defies quantization. Sometimes, you may want to create a hard groove, while other tunes might call for pushing and pulling on the time, like a rubber band.

If you use a MIDI keyboard controller and record in real time, it should sound like you're playing a keyboard. This is great if you're using a piano or organ patch. On the other hand, if you're using a guitar or vibraphone patch, think like a guitarist or vibraphonist, not a keyboard player. Guitars have six strings, which are basically tuned in fourths, so voice your chords

accordingly. For example, keyboards can easily play clusters of seconds or extremely wide voicings, but guitars can't. Stick to chords of mostly thirds and fourths, and you'll be fine.

Vibes are usually played two notes at a time, although some players (such as Gary Burton) are adept with four mallets. They also use a lot of tremolo, striking the same note many times in rapid succession. Here are a couple of tricks to simulate this effect. If you have two keyboard controllers, play the same note on both controllers, alternating strikes. This is easier than playing the same note on a single controller. If you have only one controller, play the two notes an octave apart. Then go back and transpose one set of notes to the same pitch level as the other set. For even better results, record one hand to one MIDI track and the other to a different track with the same patch. I've found this provides the most realistic effect.

Real pianos, organs, vibes, and (to a lesser extent) jazz guitar make a sound when you hit them. You don't have a whole lot of control over what happens after you hit a note. As a result, synths can emulate their performance fairly well, as long as the basic patch isn't too cheesy. Unfortunately, this is not so true with some of the other members of the jazz family.

RHYTHM

At first glance, it seems odd that drums in pop music are easily simulated using
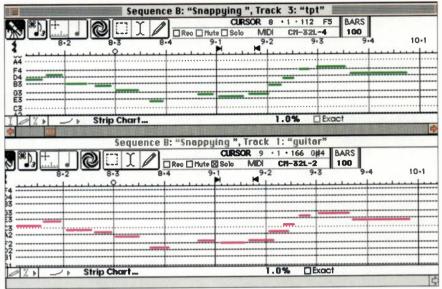

FIG. 3: The trumpet and guitar are playing the same melody line here, but the guitar part includes characteristic overlapping notes in bar 9, beat 2, while the trumpet doesn't.
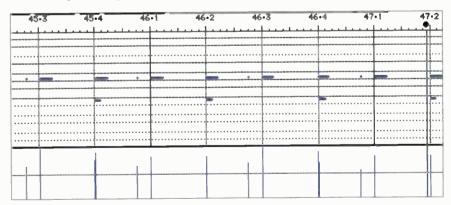
FIG. 2: Drum dynamics are critical to effective jazz. Notice that the secondary cymbal strikes have lower Velocity (and therefore a lower dynamic level) than the primary strikes, as indicated by the vertical lines under each note.

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ALL THAT JAZZ

MIDI, but they are more problematic with jazz. However, anyone who has played jazz drums knows that a wide variety of different techniques and patterns are used in this music, many of which are only rarely used in pop and rock music. The same basic instrument is used for both genres, but the sounds are quite different. I usually begin the drums by working with the hi-hat.

Jazz drummers normally use their left foot to play the hi-hat on beats 2 and 4, or variations thereof, which helps give this music a distinctive color. Unfortunately, few synthesizers include anything approximating this sound. If you lack a real, foot-closed, hi-hat sound in your aural arsenal, try sampling one. If you use a sequencer that incorporates digital audio, you can place, copy, and paste together an acceptable hi-hat pattern using a stored hi-hat sound on your hard disk. As a last resort, use a struck, closed, hi-hat sound.

In any event, you'll still miss the variety of subtle things a drummer can do with a foot-controlled hi-hat, such as closing it and letting the cymbals ring, or slowly opening the hi-hat while rolling on the top cymbal with a pair of sticks or brushes. I create simple hi-hat parts, and, if necessary, I quantize to the nearest quarter note with a strength of 50%.

Next, I work on the ride and crash cymbals. It's fairly easy to put down a good track if you already have a good

bass and hi-hat part. Quantizing the ride cymbal is a lot like quantizing the bass part. However, although the bass part might sound good quantized to eighth-note triplets at 112 bpm, this doesn't necessarily apply to the ride cymbal. A swing value closer to 60% works well at around 360 bpm, but a value of 80% sounds more natural at 112 bpm. Having said that, I hasten to reiterate that overuse of quantization in jazz tends to drain a performance of its soul.

The next problem area is the snare drum, which, like the hi-hat, produces a much wider variety of sounds than you find in pop music. Buzz rolls, soft hits, and cross-sticks are just a few of the snare sounds not found on your basic synth. In the jazz idiom, it's a much more dynamic instrument than the bass or piano, and it is often used to create a busy, nonrepeating background of accents and fills behind a soloist. This is difficult to re-create using a sequencer. But you can come close if you know how jazz drums are played. Use two different acoustic snare sounds—a soft and loud hit (please, no gated reverbs!)—and a drumKAT or other MIDI drum controller. Simply assign the soft hit to low Velocity values and the loud hit to high Velocity values from the drum controller.

Lacking such a controller, I use the double-keyboard technique described previously to create snare fills and rolls, often combining these sounds with a kick drum in a single performance. Alternatively, you can just map the snare sound, or two almost identical snare sounds, to adjacent keys. To play a roll, you simply trill. It takes a bit of work, and after many years of doing this, I've learned to resist the temptation to overplay. It's not realistic for me to create a Tony Williams drum part, so I opt for

the minimalist approach. It saves time and gives the other instruments more space to realize their potential.

I often try to play the snare and the kick drum at the same time because the parts are tightly interwoven in a good jazz performance. I have found some decent jazz kick-drum sounds on a variety of synthesizer modules. I prefer a somewhat roomy, acoustic sound, with a soft attack. The same is true for tom-toms, which I also try to use sparingly.

Dynamics are important in jazz phrasing, and they're *critically* important for the drums. The difference between a smoking groove and a lame one is largely due to dynamics, i.e., which snare or cymbal hits are loud and which ones are soft (see **Fig. 2**).

So far, I've assumed that the "MIDI drummer" is holding a pair of sticks, a spurious assumption indeed. If you have access to a sampler or hard-disk

THE BASS PART IS LITERALLY THE RHYTHMIC AND HARMONIC FOUNDATION OF THE TUNE.

recorder, you can create an entirely new palette of snare, cymbal, hi-hat, and tom sounds played with brushes. Using brushes requires a whole new set of techniques, but they produce a new vocabulary of sounds that help give jazz its distinctive flavor, particularly on ballads and quieter tunes.

Did I mention that quantization in jazz is not a great idea?

MELODY

Last, but not least, are the lead instruments. Saxophone, flute, trumpet, and trombone are among the most common in this genre. Keyboards and guitars are hybrid instruments, because they perform a lead function in some contexts. For example, if I'm creating a piano/bass/drum trio piece, I may have already recorded the piano solo as part of my basic performance. So at this point in the process, I might want to edit my solos or record additional solo tracks. Once again, my approach is to record whole choruses, if not whole takes, in real time.
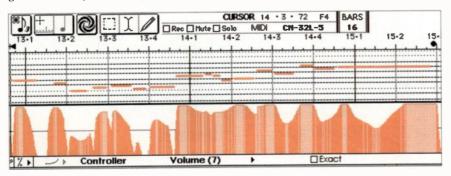

FIG. 4: The shaded curves below the graphic note-editing window depicts Breath Controller information after it has been transformed into Volume data, which helps add realism to wind-instrument parts.

On the other hand, if I've decided on a "front line" of, say, trumpet and guitar, with a piano/bass/drum trio behind them, now is the time to record the theme and solos with those instruments. Even if a theme is played in unison by two different instruments, *don't* copy and paste one track into another. A guitar is a polyphonic, plucked instrument. You play it differently than a trumpet, which is a monophonic, wind instrument. This is particularly important with MIDI, because you must play a variety of instruments on your controller (usually a keyboard).

To this end, be aware of the way you finger your parts. In **Fig. 3**, note that the same lead line is played by the trumpet and guitar. The triplet figure in bar 9, beat 2 has overlapping notes in the guitar track, which is a natural result of playing that instrument. The same performance would sound artificial if played with a trumpet patch.

Let's look at how to create a good trumpet track. First, a breath controller lets you produce wind-like articulations, even if you don't have a MIDI device that recognizes Breath Controller messages. In addition to note data from my master keyboard controller, I use the breath controller on my old Yamaha DX7 to send Breath Controller messages to an Opcode Studio 5 MIDI interface, which translates the Breath Controller information into Volume information (see **Fig. 4**).

If you don't have a Studio 5, do the translation in your sequencer. Most professional MIDI sequencers let you manually transform the data from one type to another after it's recorded. If you don't have a DX7, you can get an Anatek Wind Machine, which accepts input from a Yamaha BC2 breath controller and outputs continuous MIDI messages of your choice. With a heavy touch (high MIDI Velocity) on the keyboard, you get more dramatic results from the breath controller.

If you don't have a breath-controller setup, you can also get good results using key Pressure (aftertouch). For example, a common wind technique is *szforzando-piano-crescendo,* in which the player hits the note hard, immediately backs down to a soft level, then slowly increases the volume over the course of the note. This is relatively easy to simulate with Pressure.

Finally, jazz and blues lead lines make ample use of pitch bend. Almost all brass and wind instruments, as well as guitar and bass, can sound more authentic with judicious use of pitch

The Wind Machine, from Anatek, accepts input from a breath controller and outputs continuous MIDI messages of your choice.

ALL THAT JAZZ

bend. I usually set my pitch-bend range to a whole step. A possible exception is the Dixieland trombone, which is capable of sliding as much as an augmented fourth up or down. As with other controller data, I prefer to use pitch bend in live performance, rather than manually drawing it in later.

MIXDOWN

Once I've finished the solos, I listen to the whole piece from start to finish. If necessary, I might add a drum fill or two, or a percussion part. Then it's time to mix tracks. If you use a General MIDI sound module, such as the Roland Sound Canvas, you need to use MIDI Volume and Reverb Depth messages to mix the parts. (A MIDI fader box, such as the Peavey PC 1600, helps immensely here.) If you have separate MIDI devices and/or audio outputs, you have more flexibility in terms of level, EQ, and reverb.
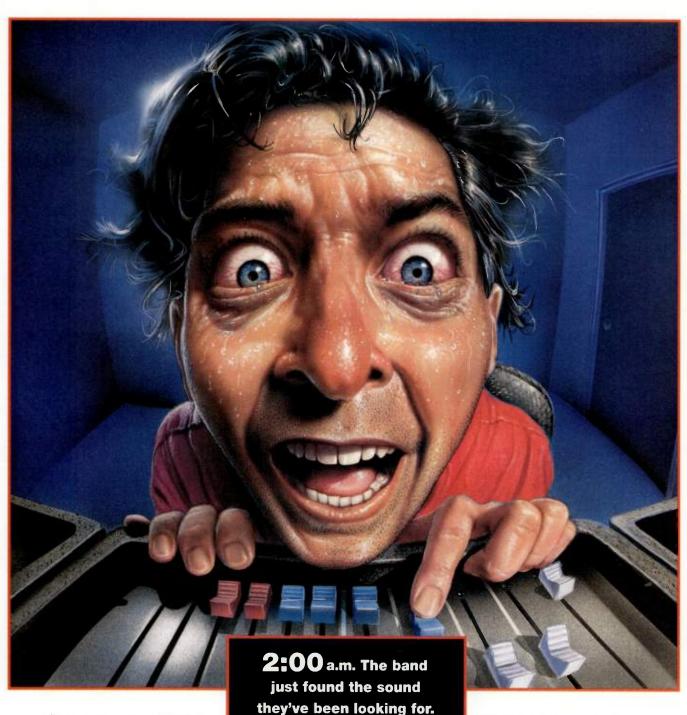
The hardest part about mixing MIDI jazz is making it sound like several musicians actually playing together. Once again, listen to the great jazz recordings made by people like Bob Thiele, Teo Macero, and Rudy Van Gelder. I usually mix the drums—particularly the snare and kick drum—fairly low, with a moderate amount of natural, small-room reverb. Judicious use of the same reverb on the other instruments can help create a feeling of ensemble.

If you want to use MIDI technology to manifest your jazz ideas, you might want to start by transcribing a favorite tune from a CD, then re-create it using the steps outlined here. When in doubt, go back and listen closely to what each player is doing, then refine your parts, one at a time. Repeated listening and emulating is the fastest way to learn about this uniquely American musical form.

Paul Potyen is, among other things, a jazz musician. He finds it comforting to know that he is never more than a half-step away from the right note.

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SONIC
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Conjuring the sonic spirit of Ensoniq's TS-series instruments.

SYNTH VETERANS often criticize popular wavetable synthesizers such as the E-mu Proteus and Korg M1, because their waveforms quickly fall into short loops due to limited sample memory. Old timers contrast such instruments with two mighty wavetable synths of yore: the PPG Wave and Sequential Prophet VS. These pioneer instruments sequentially scanned through waveform lookup tables, more or less defining the term "wavetable synthesis."

Contemporary synth manufacturers have addressed the problem of static loops in several ways. Korg's Wavestation (reviewed in the November 1990 **EM**) extends the concept of the Prophet VS, providing timbral variety by stringing waveforms together in a sequence. You can also pan between waveforms assigned to terminal points on two axes, or vectors, using a joystick or other two-dimensional MIDI controller.

Waldorf's MicroWave (reviewed in the April 1990 issue) revived PPG-style wavetable scanning. More recently, E-mu took another tack by introducing filter morphing in the Morpheus (reviewed in the May 1994 issue) and UltraProteus.

At the forefront of this flurry of development is Ensoniq, which introduced a wavetable-scanning construct called *Transwaves* in its 1989-vintage VFX. (Transwaves have been included in every Ensoniq synth since then, including the TS series.) Taking this idea another step, Ensoniq added considerably more control features when it introduced *Hyper-Waves* in its top-of-the-line TS-10 and TS-12 synth workstations. (The TS-10 was reviewed in the November 1993 **EM**.)

By Clark Salisbury
photograph by Carol Drobek

SONIC SORCERY

The TS-10 and TS-12 are fraternal twins; the only difference is the TS-12's 76-key, weighted keyboard versus the TS-10's 61-key, unweighted action. They have a powerful synthesis engine, spectacular effects processing, a user-friendly sequencer, and the ability to play and edit samples from the extensive EPS/ASR-10 library. But it is Hyper-Waves that give the TS series its special magic.

Hyper-Waves are based on a simple idea. They let you string together disparate pieces of the synth's waveform ROM, with control over the amount of time each piece sustains, the way it crossfades to the next piece, and so on (see Fig. 1). However, this simple idea can produce a nearly limitless number of dynamic timbres, ranging from the subtle to the exotic, with occasional forays into the bizarre.

Hyper-Waves are also a potential source of some confusion. If you feel lost among the Hyper-Waves, read on. You'll learn the sorcerer's secrets that make Hyper-Waves jump through hyper-hoops.

THE FIRST SPELL

To get a handle on this sonic resource, we must start with the basics. TS Hyper-Waves, or *Wave-Lists* (these two terms are synonymous), are created in a special programming area accessed via the Program Control button (more about this in a moment). Once they are created, you assign a Hyper-Wave to a Voice, as with any other waveform. (Although there can be up to four Voices in a Program, the basic playable entity, all Voices in a Program have access to only one Hyper-Wave at a time. However, each Voice can access different parts of the Hyper-Wave, as explained later.) A Hyper-Wave uses two voices of polyphony if it includes any crossfades, so a 4-note chord played with a single, crossfading Hyper-Wave uses eight of the instrument's 32 polyphonic voices.

Each Hyper-Wave contains up to sixteen *Steps*. Each Step tells the TS which wave to play at a given time, at what

pitch, for how long, and how to cross-fade to the next Step. In addition, you can control volume, the direction the wave plays (forward or reverse), crossfade depth and time, and whether or not the wave's pitch tracks the keyboard.

The incantation for creating a Hyper-Wave is simple, if not completely transparent. First, of course, you need a Program to use as a starting point for Hyper-Wave experimentation. For our purposes, the ROM Program OLD-FRIEND (located in BankSet R2-8) should do nicely. Select this Program and enter Program Edit mode by hitting the Select Voice button.

Mute all but the first Voice. To mute a Voice, press its associated soft button to select it, then press the same soft button a second time; parentheses should appear around the Voice name. Next, to provide a relatively generic starting point for our Hyper-Wave experimentation, initialize Voice 1 by selecting it (so it's underlined), pressing the Copy button, then pressing the soft button under DEFAULT.

Next, create a Hyper-Wave: Press the Program Control button and then select OPTION=*-NONE-*. Next, press the Up Arrow button; the TS presents you with the option of replacing Voices 5 and 6 with a pitch table, Wave-List, or drum map. Select Wave-List by pressing the appropriate soft button.

This returns you to the Select Voice page. In place of Voices 5 and 6, you

see the message "EDIT WAVE-LIST." To hear the Wave-List, though, you must first assign it to a Voice. This is accomplished by selecting the Voice you wish to use—in this case, Voice 1—and pressing the Wave button. Select the Waveform Class parameter, displayed in the middle of the top row; in this case, it's showing WAVEFORM. Grab the data slider and move it all the way up to change this to WAVE-LIST.

Return to the Select Voice page. The Wave-List has been assigned to Voice 1, and if you play the keyboard, you'll hear the default Wave-List. It's not too exciting, but we're just getting started. The next step is to make the Wave-List do something musical.

APPLICATION INCANTATIONS

There are four basic applications for Hyper-Waves. You can make rhythmic "groove loops," create a sound that smoothly crossfades between two or more ROM waves, and produce automatic articulations that might be physically difficult or impossible to play. In addition, you can "splice" waves by attaching the attack portion of one sound to the sustain or decay portion of another. You can also combine these techniques to create some truly wonderful sounds. We'll spend a bit of time on each application.

The key to getting the most out of Hyper-Waves is understanding the special relationship between the Hyper-Wave and the Voice. Although it's true

Ensoniq's twin top-of-the-line keyboard workstations offer a powerful synth, elaborate sequencer, high-quality effects processor, and the ability to play and edit EPS series and ASR-10 samples. The TS-10 (bottom) has a 61-key, unweighted action, while the TS-12 has a 76-key, weighted action.

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SONIC Sorcery

that only one Hyper-Wave can be used in a Program, clever programming at the Voice level can provide, in effect, up to four independent Wave-Lists. By specifying different Wave-List Start, End, and Loopstart points for each Voice, different parts of the same Wave-List can be accessed by different Voices (see **Fig. 2**).

When you consider that a Wave-List can also be processed through the instrument's filters, amplifiers, effects, and so on, it becomes clear that a single Hyper-Wave can be used to generate several completely different sounds. These different sounds can be split and/or layered on the keyboard in whatever arrangement you like, and they can be controlled with the Patch Select buttons.

ALCHEMICAL CROSSFADES

Perhaps the simplest Hyper-Wave application is to crossfade from the attack sound of one instrument to the sustain portion of another. At its most basic level, we need to use only two Steps of the Hyper-Wave: the first Step

A FREE INCANTATION

Before engaging rival TS programming wizards in battle, study this sample Program, which illustrates several of the techniques we've discussed. I've named the Program "Broadway" for no particular reason.

The lower half of the keyboard triggers a drum and bass groove, while a Transwave for solo and rhythm work is played from the upper half. The Voices assigned to the Wave-List are in Mono mode, and using just one Wave-List, you get three layered parts. The tambourine groove uses the last two Steps of the Wave-List. While all other Steps of the Wave-List produce a precise rhythm, the two Steps that control the tambourine produce an imprecise rhythm for a natural, human feel.

The mod wheel bends the Hyper-Wave's pitch up quite a bit; moving the wheel produces a record-scratching effect. On the right, pressure controls the LFO rate and depth, which in turn modulate the pitch, amplitude, and filter, providing tremolo, vibrato, and wah.

WAVES	1	2	3	4
WAVE	WAVE-LIST	WAVE-LIST	RESONANCE	WAVE-LIST
WAVE CLASS	WAVE-LIST	WAVE-LIST	TRANSWAVE	WAVE-LIST
DELAY	0000	0000	0000	0000
INDEX	n/a	n/a	75	n/a
MODSRC	n/a	n/a	ENV-01	n/a
MODAMT	n/a	n/a	-099	n/a
SAMPLE START	n/a	n/a	n/a	n/a
START MODSRC	n/a	n/a	n/a	n/a
MODAMT	n/a	n/a	n/a	n/a
START-STEP	1	15	n/a	10
LOOPSTART	1	15	n/a	10
END	9	16	n/a	14
MODDEST	none	none	n/a	none
MODSRC	off	off	n/a	off
MODAMT	+00	+00	n/a	+00

MOD MIXER				
SRC-1	n/a	n/a	LFO	n/a
SRC-2	n/a	n/a	PRESS	n/a
SRC-2 SCALE	n/a	n/a	2.00	n/a
SHAPE	n/a	n/a	SMOOTHER	n/a

PITCH				
OCTAVE	+00	-01	-02	+00
SEMITONE	+00	+00	+00	+00
FINE	+00	+00	+00	+00
GLIDEMODE	MINIMODE	MINIMODE	none	MINIMODE
GLIDETIME	+00	+00	+00	+00

PITCH MODS				
MODSRC	WHEEL	WHEEL	off	WHEEL
MODAMT	+99	+99	+00	+99
BEND	SYS	SYS	SYS	SYS
PITCHTBL	SYSTEM	SYSTEM	SYSTEM	SYSTEM
ENV1	+00	+00	+00	+00
LFO	+00	+00	+04	+00

FILTER 1				
MODE	LO-PASS/3	LO-PASS/3	LO-PASS/3	LO-PASS/3
CUTOFF	127	127	006	127
KBD	+00	+00	+12	+00
MODSRC	off	off	MIXER	off
MODAMT	+00	+00	30	+00
ENV2	+00	+00	99	+00

FILTER 2				
MODE	LO-PASS/1	LO-PASS/1	LO-PASS/1	LO-PASS/1
CUTOFF	127	127	127	060
KBD	+00	+00	+00	+15
MODSRC	off	off	off	VELOC
MODAMT	+00	+00	+00	+50
ENV2	+00	+00	+00	+00

OUTPUT				
KBD SCALE	ZON	ZON	ZON	ZON
LO/HI KEY	A0-0C4	A0-0C4	C4+-0C8	A0-0C4
VOL (DB)	+04	+01	+00	+00
MODSRC	off	off	off	off
MODAMT	+00	+00	+00	+00
DEST BUS	FX1	FX1	FX2	DRY
PAN	+00	+30	+00	+00
MODSRC	off	off	off	off
MODAMT	+00	+00	+00	+00
VOICE PRIOR	MED	MED	HIGH	MED
VEL-WINDOW				
LO	000	000	000	000
HI	127	127	127	127

LFO				
RATE	n/a	n/a	60	n/a
MODSRC	n/a	n/a	PRESS	n/a
MODAMT	n/a	n/a	+30	n/a
DEPTH	n/a	n/a	+00	n/a
MODSRC	n/a	n/a	PRESS	n/a

LFO (continued)

MODAMT	n/a	n/a	+99	n/a
WAVESHAPE	n/a	n/a	TRIANGLE	n/a
RESTR MODE	n/a	n/a	ON	n/a
PHASE	n/a	n/a	000	n/a
DELAY	n/a	n/a	00	n/a
NOISE RAT	n/a	n/a	70	n/a

SELECT VOICE

00	on	on	on	on
0*	on	on	on	on
*0	off	off	on	off
**	on	on	off	on

ENV1

ATTACK	n/a	n/a	50	n/a
DECAY	n/a	n/a	50	n/a
DECAY 2	n/a	n/a	50	n/a
DECAY 3	n/a	n/a	50	n/a
RELEASE	n/a	n/a	50	n/a
PEAK	n/a	n/a	99	n/a
BREAK 1	n/a	n/a	00	n/a
BREAK 2	n/a	n/a	99	n/a
SUSTAIN	n/a	n/a	00	n/a
VEL-LEVEL	n/a	n/a	00	n/a
MODE	n/a	n/a	REPEAT	n/a
VEL CURVE	n/a	n/a	CONVEX-01	n/a
KBD TRACK	n/a	n/a	+00	n/a
VEL-ATTACK	n/a	n/a	00	n/a
VEL-RELS	n/a	n/a	+00	n/a

ENV2

ATTACK	n/a	n/a	00	n/a
DECAY	n/a	n/a	60	n/a
DECAY 2	n/a	n/a	60	n/a
DECAY 3	n/a	n/a	50	n/a
RELEASE	n/a	n/a	26	n/a
PEAK	n/a	n/a	99	n/a
BREAK 1	n/a	n/a	75	n/a
BREAK 2	n/a	n/a	50	n/a
SUSTAIN	n/a	n/a	00	n/a
VEL-LEVEL	n/a	n/a	70	n/a
MODE	n/a	n/a	NORMAL	n/a

ENV2 (continued)

VEL CURVE	n/a	n/a	CONVEX-01	n/a
KBD TRACK	n/a	n/a	+00	n/a
VEL-ATTACK	n/a	n/a	00	n/a
VEL-RELS	n/a	n/a	+00	n/a

ENV3

ATTACK	00	00	00	00
DECAY	00	00	32	00
DECAY 2	00	00	50	00
DECAY 3	00	00	50	00
RELEASE	00	00	05	00
PEAK	99	99	99	99
BREAK 1	99	99	95	99
BREAK 2	99	99	88	99
SUSTAIN	99	99	80	99
VEL-LEVEL	10	10	25	10
MODE	NORMAL	NORMAL	NORMAL	NORMAL
VEL CURVE	CONVEX-01	CONVEX-01	CONVEX-01	CONVEX-01
KBD TRACK	+00	+00	+00	+00
VEL-ATTACK	00	00	00	00
VEL-RELS	+00	+00	+00	+00

PGM CONTROL (Page 1)

TYPE	HYPER-WV
OPTION	WAVELIST
PRESS	KEY
PATCH	LIVE
RESTRIK3	10

PGM CONTROL (Pages 2 & 3)

ATCK	+00
BRIGHT	+00
RATE	+00
RELS	+00
TIMBRE	+00
XCTRL	+00

EFFECTS

EFFECT # 37 NONLINEAR REVERB 3
VARIATION   2 KICK NONLIN

WAVE-LIST

STEP	WAVE	WAVE-CLASS	START	DUR	VOL	X-FADE TIME	DEPTH	DIR	PITCH-KBD-TRK	XPOS	DETUNE
01	R+B-KICK	DRUM-SOUND	+00	288	+03	+00	+00	FORWARD	off	+00	+00
02	SYNHAT-OP	CYMBALS	+00	288	-08	+00	+00	FORWARD	off	+00	+00
03	SYNSNAR-03	DRUM-SOUND	+00	288	+02	+00	+00	FORWARD	off	+00	+00
04	CLO-HAT-01	CYMBALS	+00	192	-02	+00	+00	FORWARD	off	+00	+00
05	LOOP-KICK	DRUM-SOUND	+00	96	-03	+00	+00	FORWARD	off	+00	+00
06	SYNKICK-03	DRUM-SOUND	+00	288	+02	+00	+00	FORWARD	off	-12	+00
07	CLO-HAT-01	CYMBALS	+00	288	+00	+00	+00	FORWARD	off	+00	+00
08	SYNSNAR-03	DRUM-SOUND	+00	288	+04	+00	+00	FORWARD	off	-03	+00
09	LOOP-KICK	DRUM-SOUND	+00	288	+02	+00	+00	FORWARD	off	-03	+00
10	ANABASS-01	BASS-SOUND	+00	288	+03	+00	+00	FORWARD	on	+00	+00
11	ANABASS-01	BASS-SOUND	+00	288	-01	+00	+00	FORWARD	on	-02	+00
12	ANABASS-01	BASS-SOUND	+00	288	-02	+00	+00	FORWARD	on	-03	+00
13	ANABASS-01	BASS-SOUND	+00	192	-01	+00	+00	FORWARD	on	-02	+00
14	ANABASS-01	BASS-SOUND	+00	96	+02	+00	+00	FORWARD	on	-05	+00
15	TAMBORINE	PERCUSSION	+00	180	+02	+00	+00	FORWARD	off	+00	+00
16	TAMB-LOOP	PERCUSSION	+00	108	-05	+00	+00	FORWARD	off	+00	+00

for the sound's attack, and the second for its sustain. Let's give it a shot.

Make sure Voice 1 is still selected for editing. Press the WAVE button to move to the WAVE page and change the setting for END=16 to END=2. This specifies that the Wave-List assigned to this Voice includes only two Steps. Note that you can also control the Wave-List Start-Step and Loopstart from this page. We will work more with these parameters in a moment; right now, we are going to operate directly on the Wave-List.

Press the Select Voice button, then hit either of the soft buttons under EDIT WAVE-LIST. From here, we can assign the ROM waves we wish to use and determine settings for several other Wave-List parameters. Let's try fading the attack from the piccolo-trumpet wave into the body of a flugelhorn wave. This will produce a composite sound with a brighter attack than the flugelhorn and a richer sustain than the piccolo trumpet.

Select the Wave Class parameter, displayed at the upper right; it should show WIND+REEDS. Hit the Down

FIG. 1: Anatomy of a Wave-List, which includes up to sixteen Steps, each of which can play a different ROM waveform. You can independently set the tuning, volume, crossfade times, and several other parameters for each Step in the Wave-List.

Arrow button once to change this to BRASS+HORNS. Next, select WAVE-TRUMPET and scroll up to change this to PICO-TRMP. This assigns the piccolo-trumpet wave to the first Step of our Hyper-Wave. Next, select STEP=01 and change it to STEP=02. (This is how you select any of the Steps in the Wave-List for editing). Assign the FLUGELHRN wave (in the BRASS-HORNS Wave Class) to this Step.

At this point, the sound should continuously fade back and forth between the trumpet and flugelhorn as you hold a note. We want to hear the trumpet sound only during the attack of the note, so we must change the loop points so that only the flugelhorn sound loops. Press the Wave button, and set LOOPSTART=01 to 02. This causes the Wave-List to play through Step 01 (its start point), continue on through Step 02 (its end point), and then jump back to the Loopstart point, which in this case is the start of Step 02. Step 02 now repeats indefinitely, which is what we're after. The attack of the trumpet fades into the flugelhorn sound, which sustains for as long as a note is held (see **Fig. 3**).

The crossfades from trumpet to flugelhorn and from the end of the flugelhorn wave back to its beginning (as Step 02 loops) are still rather crude at this point. Press the Select Voice button and select EDIT WAVE-LIST to return to Wave-List Edit mode. Select Step 02 (if it isn't already selected), then select *TIME* to move to the menu in charge of crossfade times and shapes. We'll improve the sustain portion of this sound before we tackle the crossfade from Step 01 to Step 02.

We can easily smooth out the sustain portion of this loop simply by increasing the value for crossfade time. Select XFADE-TIME and move the data slider up a bit. Any number above 1,000 or so produces perfectly smooth fades.

Now move back to Step 01 of the Wave-List. Because we're using this wave only for the attack portion of the sound,

you may want to shorten the DUR (duration) parameter, which controls how long a particular Step sustains before beginning its fade into the next Step. Set DUR to 00000, the minimum time available.

Next, adjust the XFADE-TIME parameter; this controls how long it takes for the current Step to fade into the next. Short fade times—say, between 12 and 108—produce very tight attacks. If you want a hint of the characteristic "blip" at the beginning of a trumpet note, try values in the range of 24 to 72. If you want a bit more "meat" in the attack, values from 72 on up will produce the desired results.

SPIRITED ANIMATION

Another, closely related Hyper-Wave application is to continuously crossfade between two or more waves. Using dissimilar waves as Hyper-Wave material can produce some striking results. This type of crossfading can also be used with similar waves to impart a more subtle animation and sense of movement to sounds that might otherwise be somewhat static. We can check this out using a different portion of the same Hyper-Wave we've already started working with.

Hit the Select Voice button and mute Voice 1. Select and unmute Voice 2, and assign the Wave-List to it, just as you did with Voice 1. Press the Wave button, select the Wave Class parameter, and move the data slider all the way up. While you're still on the Wave page, set START-STEP to 03, LOOPSTART to 04, and END to 06. Obviously, we'll be looping three Steps in this portion of the Wave-List: Steps 04, 05, and 06.

What may not be obvious is why we're starting this portion of the Hyper-Wave at Step 03. The reason is pretty simple, actually. We're using Steps 01 and 02 of our Wave-List to create the piccoloflugeltrumpethorn sound used in Voice 1. By using a different portion of the Hyper-Wave to create our next sound, we can leave the first sound intact, providing two (or more) Wave-Lists for the price of one.

Next, we'll assign the waves we want to use to the appropriate Steps in the Wave-List. Hit the Select Voice button, then select EDIT WAVE-LIST. Select STEP-, and set it to 03. For this Step, let's use the TRUMPET wave. Now, move to Step 04, and set it to TRUMP-SEC. (This grafts the attack of a solo trumpet onto the sound of a full trumpet section, which can provide some additional bite in a brass-section type of Program.) Similarly, set Step 05 to
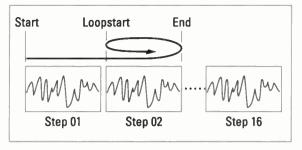
FIG. 3: At the Voice level, Wave-List Start, End, and Loopstart points can be set. In the above example, Step 01 is the Wave-List start Step, with Step 02 looping continuously for the duration of the note.

FRHRN-SEC and Step 06 to ANA-WAVE1 (in the WAVEFORMS Wave Class).

If you listen to what we've done so far, the first thing you might notice is that not all the Steps in our Hyper-Wave play in the same octave. This can be a cool effect, but for our immediate purposes, it makes more sense to have all waves playing in the same octave. Press the soft button under the word *PITCH* to call up the Pitch Control menu. Set the XPOS (transpose) value for Steps 05 and 06 to +12, raising each of these Steps an octave. If it isn't immediately obvious, you can select the Step you wish to edit by pressing the soft button above STEP- and using the data slider and/or the Up/Down Arrow buttons.

Now that the various waves and transpositions have been assigned to the appropriate Steps, you can use the process previously described (hitting the Select Voice button and choosing EDIT WAVE-LIST and *TIME*) to specify the desired crossfade times. Try using a DUR of 00000 and XFADE-TIME of 00144 for the first Step. (Remember, the first Step for this Voice is Step 03.) Use fairly long crossfade times—in the range of, say, 01000 to 01500—for the remaining Steps. Durations for these Steps are not particularly critical, as the crossfade times are so long. Still, keep the durations on the short side—00000 to 00192—if you want to keep the Steps of the Hyper-Wave continuously fading from one into the next.

Here's a neat little trick to fatten up the sound a bit. Use the Copy function to copy the current Voice (Voice 2, in this example) to another location (e.g., Voice 3). Select the copied Voice, hit the Wave button, and set the END parameter to a value one less than the
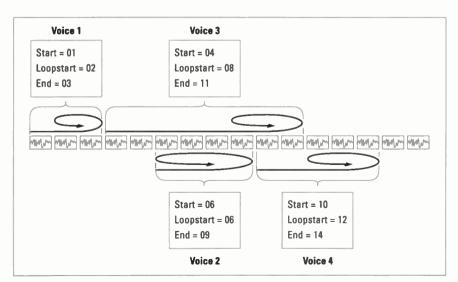

FIG. 2: Each of the four Voices in a Program can play different, contiguous segments of a Wave-List. As you can see, segments can overlap, and loops can be set independently for each Voice.

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current value. In our example, change END from 06 to 05. This causes the two Wave-Lists to loop at different points, producing a less-predictable cycle. You may want to detune the two Voices slightly and pan them in opposite directions, as well.

HYPER-ARTICULATION
Another cool application for Hyper-Waves is to create articulations that might otherwise be difficult to play. For example, you can create an unplayable trumpet riff to go along with the rest of the brass sounds we've created by assigning the same sound to successive Steps of the Wave-List, but at different pitches. By setting appropriate crossfade times and durations, you can program nearly any short phrase at nearly any tempo.

For example, initialize the fourth Voice in the current Program by selecting Voice 4, hitting Copy, then selecting DEFAULT. Hit the Wave button and assign the Wave-List to this Voice. Set the START-STEP to 07, and set both LOOPSTART and END to Step 11. Hit the Select Voice button and select EDIT WAVE-LIST.

Assign the TRUMPET wave to Steps 07 through 11. Select *PITCH*, set XPOS for Step 08 to +02, and set XPOS for Step 10 to -02. Hit *EXIT*, then select *TIME*. Set the duration for Step 07 to 00144 and the crossfade time to 00024. For Steps 08, 09, and 10, set DUR to 00048 and XFADE-TIME to 00024. For Step 11, set DUR and XFADE-TIME to their maximum value. Play a chord, and *voila!* Instant fanfare!

MYSTICAL GROOVISM
Groove loops are another excellent application for Hyper-Waves. You can easily set up drum or percussion parts with one section of a Wave-List, while using another section to supply a bass line or other melodic part. You can then layer the two (or more) parts to generate play-with-one-finger grooves like those found in several of the TS user Program locations. If you've been paying attention, you shouldn't have too much trouble stringing together sounds to create such a Wave-List, but there are several things that might be helpful to keep in mind.

First, use a Hyper-Wave to create a drum pattern by assigning various drum and percussion sounds to several consecutive Steps in the Wave-List. Obviously, you want to keep crossfades at

their minimum value to retain the percussive quality of the drum, percussion, and cymbal waves. Therefore, you should rely on the DUR (duration) parameter to control the timing of the Steps in the Wave-List.

This is a bit easier if you understand that the numbers used for setting duration and crossfade time represent milliseconds (1 ms = 0.001 seconds). These time values are varied in 12 millisecond increments. So when you're designing a groove, it's handy to use some sort of beat calculator, such as those found in recording studios, to calculate duration and crossfade times. If you don't have such a device handy, you can calculate the duration of a single beat in milliseconds using the following equation:

60,000/beats per minute = duration of one beat in milliseconds.

From there, it should be simple to subdivide the beat to determine the durations needed to produce eighth notes, sixteenth notes, and so on.

You might not want the percussive Voices in groove loops to transpose when played in different areas of the keyboard. If your groove loop consists of electric bass and drums, for example, you'll probably trigger the bass part with different keys, but it will sound hokey if the drums transpose, too. This is easily remedied by setting PITCH-KBD-TRK to OFF (located on the *PITCH* page of the EDIT WAVE-LIST menu) for any Steps of the Wave-List that you don't want transposed when different notes are played.

With some groove loops, you might want to play more than one note of the melodic instrument sound at the same time. This can pose a problem for the percussion part, though: If you simultaneously trigger two or more notes of a Voice that doesn't track the keyboard, you risk unwanted flanging and phasing. The solution? Put the Voice that's playing the percussion part in one of the various mono modes (called "GLIDEMODE" and accessed via the Pitch button). MINIMODE is probably the best selection if you want to maintain a constant tempo, but other GLIDEMODEs work, too.

THE FINAL WAVE
If you've followed along and have a handle on all the techniques we've explored, you can consider yourself a Master of Hyper-Wave Magic, with a minor degree in Tech Talk Tolerance.

There are several other ideas you may wish to explore if you're going for your doctorate in Hyper-Wave Wizardry. Using various TS modulators to control Hyper-Wave start, end, and loop points springs immediately to mind. But if you can use the techniques described here, and especially if you can combine two or more of these techniques in the same Program, you definitely qualify as a Hyper-Wave sorcerer.

In addition to composing, technical writing, and owning Virtual Music (a project studio), **Clark Salisbury** *works with Music & Sound Associates, which specializes in sound design, sequencing, and consultation. He really wants to play guitar, loudly, in a blues band.*

I'm usually pretty good at keeping tabs on pop culture. I know which movies are "must sees," which bands are zooming up the record charts, and which novels are getting trashed at cocktail parties. But every once in a while, something falls through the widening chasms of my consciousness. So tell me, when did *Rolling Stone* become a humor magazine?

I cracked up big time when I saw their October special issue on "Women in Rock." There was star-of-the-moment Liz Phair on the cover, crouching barefoot and wearing a cheap negligee. The representation of this popular female rocker as a young, hip, MTV-friendly sex doll was totally hilarious. Ha-ha. Unfortunately, the Lolita schtick has one small problem: It's not funny.

For one thing, Phair is no boy toy. She is one of the few mainstream women artists that exercises a good deal of control over her career. Phair coproduces her records, directs her videos, and has full approval of her album cover artwork. Because women producers and engineers are almost invisible in the

THE GENDER

GAP

WHY ARE WOMEN PRODUCERS SUCH A RARE BREED?

male-dominated music industry, it's no joke when a woman who has attained a position of power is sabotaged by tired old sexual stereotyping.

Although they are often hard to find, women producers *do* exist, and they are making great records. I interviewed five women producers about their musical concepts, styles, and backgrounds and let them do the talking. Some have engineering backgrounds, and some are artists themselves. Most have won industry awards, gold records, and Grammy nominations and have produced tracks you've

probably heard on radio, MTV, VH-1, or in the movies. And as far as I know, none of them have massaged the fragile ego of the music industry's "boy's club" by showing up for work in their nighties.

BELINDA CARLISLE

When punk and new wave music started pogoing onto mainstream radio in the early 1980s, primal girl groups such as the Slits were still too raw to dent the pop charts. Enter the Go-Go's. The bubbly, radio-friendly melodic rock of the band's 1981 debut album *Beauty and the*

BY MICHAEL MOLENDA

Beat smothered the airwaves, ultimately selling more than two million units. And despite almost constant derision from critics and "authentic" new wave musicians, the Go-Go's were the scene's most successful all-female group until the act disbanded in 1985.

As the band's vocalist and frontperson, Belinda Carlisle enjoyed an extremely successful post-Go-Go's solo career and, in 1993, decided to self-produce her next album. The result was *Real,* a tough little jewel of a record that was largely—and unfairly—ignored by the public. Carlisle is currently on tour with the re-formed Go-Go's promoting the band's two-CD retrospective, *Return to the Valley of the Go-Go's.*

On Taking Control. I decided to produce *Real* because I wanted to take more responsibility for my career. I was already recording some 8-track home studio demos with [former Go-Go's guitarist] Charlotte Caffey, and actually, it was Charlotte who suggested I produce the album. She said, "Hey, we're doing it in your house, so why can't we do it with a great engineer in a big studio?" It made sense. We were having a blast working on the demos, *and* we were getting a lot done. Taking sole responsibility for the record actually made it easier—and more comfortable—for me to really focus in on exactly how I wanted each song to sound. I ended up pro-

ducing more than half of *Real,* and making that record was definitely the most fun I've ever had in the studio.

Inspiration. I don't think that I fully understood the importance of record production until I heard U2's *Achtung Baby.* Here was an incredibly raw record that redefined the band's sound but also stayed true to the group's roots and identity. As a creative and commercial breakthrough, *Achtung Baby* offered other artists a lot of creative options. You see, a large part of my musical style is that raw rock 'n' roll sound, but it was always difficult for industry people to see me in that light. But U2 proved an artist could push the envelope without necessarily tossing record sales out the window.

Production Style. There's a certain perception that a major-label release for a mainstream artist has to sound full and "produced." But you really don't need a bunch of production tricks if you have a good song. For *Real,* I wanted a stripped-down, live-sounding record, so I worked hard to ensure that the songs stood up with minimal instrumentation. Although my previous solo records used a lot of studio musicians, I knew I'd never create a raw, live vibe for *Real* using hired guns. I ended up putting together a temporary band made up of some trusted friends and played live in the studio. Most of the tracks you hear on the record are first or second takes, and I actually sang live with the band. I didn't do a lot of retakes or vocal comping, because getting nit-picky often destroys the immediacy of the performance. I would rather have a few rough spots in a passionate track, than a dull but perfect vocal recording.

However, it took a while to get comfortable with the fact that less can be more. Some songs worked with sparse arrangements, and some didn't. We even did a conceptual somersault on a few tunes and used drum loops because

AMY YAMAMOTO

Cookie Marenco

The re-formed Go-Go's (Belinda Carlisle, second from right).

VICKI BERNDT
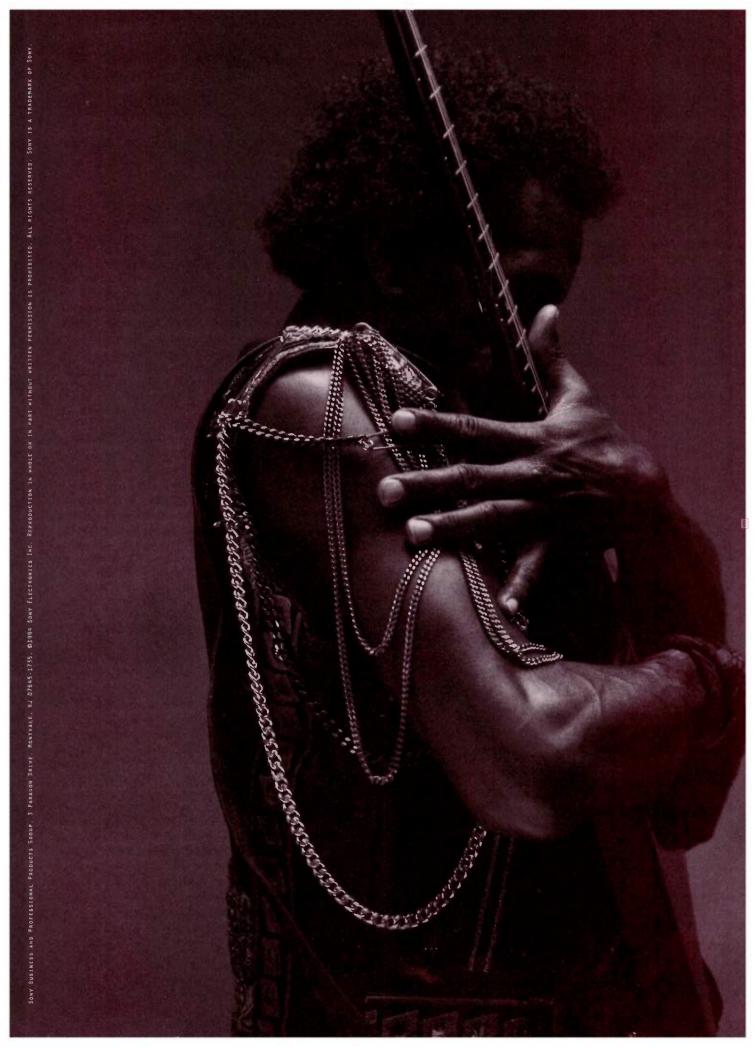
the live versions sounded stilted. Basically, the key was to use the [home] demos as guidelines and trust our instincts. If the album version didn't stand up to the demo—in the sense that the new track sounded too slick or downright boring—we knew we were heading in the wrong direction.
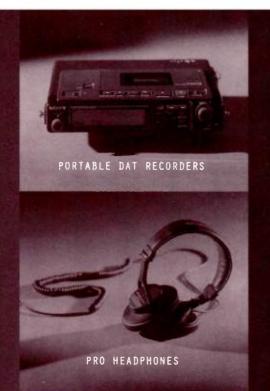
COOKIE MARENCO

A former staff producer and A&R person for Windham Hill Records, Cookie Marenco owns her own commercial 24-track studio in Belmont, California. However, she isn't there very much. The renowned engineer and producer is in constant demand, often commuting to studios in New York, Los Angeles, Colorado, and London. The "short list" of artists Marenco has worked with include Ladysmith Black Mambazo, Oregon, Alex de Grassi, Mark Isham, Praxis, Chanticleer, Peter Apfelbaum & the Hieroglyphics, and Diamanda Galas. She earned a gold record for her production of Windham Hill's *Winter Solstice 2* and, in 1993, was nominated for a "Best Jazz Recording" Grammy for engineering Charlie Haden and Quartet West's *Haunted Heart.*

As if Marenco isn't busy enough, her production company, Cojema Music, recently jumped into the record business and is releasing albums by Ben Demerath, Sugarbeat, and Pieces (featuring bizarro guitar hero Buckethead). To date, all of Cojema's releases have been produced and engineered by Marenco.

Production Style. I strive to capture the moment of an artist's highest performance. Unfortunately, there are no guidelines for documenting blistering performances, so I trust my instincts.
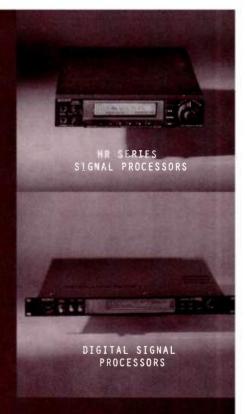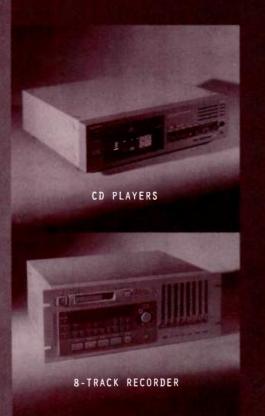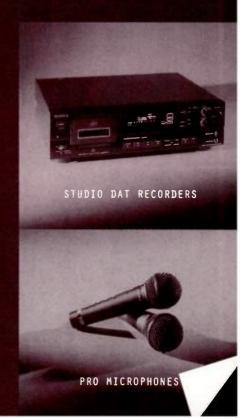
Sometimes that means figuring out if the artist is going to burn out after one or two takes, or if they can keep getting better and better. The funny thing is, many musicians—even the great ones—don't know when they've achieved their best performance. I find that I often capture exceptional moments when the artist least expects it.

To get these great moments on tape, I focus on emotion and toss technique out the window. I mean, there has to be a certain level of technical perfection or I can't live with myself, but technique means less to me than it did a few years ago. The goal is simply to get the artist's best stuff on tape. Luckily, because I was a musician before I started engineering or producing, I can usually tell the difference between a passionate track and a so-so performance.

As far as developing my production chops, working at Windham Hill proved to be a good experience, because I had the opportunity to compare my work with other engineers and producers. That exposure—and discovering that I wasn't so bad at what I did—gave me the confidence to grow. It also helped that I was lucky enough to work with a very high caliber of musicians who could execute what I asked them to do. Someone like [multi-instrumentalist] Paul McCandless can be pushed technically to improve their tuning, timing, or feel. Now, I'm confident enough to push artists toward higher *emotional* levels of performance.

The Gender Thing. Nobody wants to get on a soapbox, but behind closed doors women do talk about the issues of being female in the music industry. It's kind of an "in the closet" thing, but the dialog exists, and we take stock of what's going on. The sad truth is that young women and men often never see women in engineering or producing roles, and they grow up thinking women can't achieve these positions.

I *really* notice the lack of women in the industry at awards shows when my date is asked what he was nominated for, or when I go to a music store and never get hassled by a salesperson. It also hurts when I read interviews with women artists who want to work with women producers and engineers but can't find them.

This situation is troubling but understandable. There's a simple lack of female role models. When I was at Windham Hill, I'd get twenty demo tapes from men for every one tape I'd get from a woman. And even today, when it's a good time for female performers—the industry is actually looking for them—women artists don't put themselves out there like men do. I certainly never had any women I could look up to—either because they didn't exist or because no one pointed them out to me—so I never aspired to be a producer or engineer. It just so happened that I started falling into these roles and was good enough to get paid for them. Now I just do the best job I can and hope that I inspire other artists to work with women.

SYLVIA MASSY

Sylvia Massy followed the treacherous course from making noise in crummy clubs to making gold records in the studio. Few musicians survive (or complete) the journey, but Massy has flourished. As an engineer or producer she has worked with the former Prince, Aerosmith, Paula Abdul, Julee Cruise, the Sea Hags, Babes in Toyland, and Mojo Nixon, among others. She turned a joke band, Green Jellÿ, into a pop fave and helped propel alternative rock band Tool toward mainstream sales and airplay.

In August of this year, Massy established Third Hole Records, an independent record label dedicated to releasing alternative and underground artists. (The label accepts unsolicited demos at Box 481175, Los Angeles, CA 90048; tel. [213] 655-6307; fax [213] 655-6303.) Currently, Massy is in London producing Skunk Anansie's debut album for Epic Records.

The Breaks. I started out as a radio deejay, but got frustrated when I discovered that radio is more about advertising than music. So I began working at a studio in the San Francisco Bay Area, making radio commercials. Because I was also a musician,

there was a natural progression toward recording music and making demos for my own band and then making demos for other people. Becoming a producer just kind of happened; it wasn't even on purpose. I never thought I'd be doing what I'm doing.

My big break—although I had a lot of little breakthroughs—was mixing Aerosmith's 12-inch single version of "The Other Side" when the regular engineer couldn't make the session. Producing Green Jellÿ's *Cereal Killer* was another boost, although I never expected that record to be the huge commercial success it became.

Production Style. When I'm asked to do a record, I listen critically to the artist's material. If there's nothing there, I'll pass on the project. The most important aspect of making a great record is finding a band with great songs. A lot of today's rock music is a bunch of riffs, rather than well-constructed songs, which is why I think a lot of great sounding but totally empty records are being made. A great vocalist is important, too. It's a big problem to create a vocal track in the studio with comping and harmonizing and other electronic gadgets. I hate it.

What I like to do is make it possible for a band to express their own ideas, while I throw in things to spur their imagination. The records that I do are

Sylvia Massy

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THE GENDER GAP

usually representations of what the band is about. To achieve this, I'll see the band play live and try to recapture on tape the energy they put out in performance. Usually that live energy gets lost in the recording studio. Then, I may throw weird ideas at them to get the band thinking about developing their sound further.

For example, during the Skunk Anansie sessions, I wanted to experiment with the cheapest second-hand store electric guitar they could find. By a fluke, I happened to hear the drums through the guitar's pickups. The drummer was warming up and the guitar was just sitting on the floor facing the kit. It made the weirdest noise because the strings resonated with the kick drum and produced a sitar-like sound. So for one song, we set the guitar on a stand directly in front of the kick drum, tuned all the strings to the song's key, and recorded the drums through the guitar pickups.

The Gender Thing. The music business is hard for everyone. I don't think I have it any tougher because I'm a woman. Luck is really everything. You can be incredibly talented, but it doesn't mean much unless you get a break. Women who want to get into producing and engineering just have to stick with it and keep doing it. You have to work all the time. Beyond that, it's a matter of being serious about what you're doing and trying to be in the right place at the right time.

SUSAN ROGERS

One of the few women producers represented by a professional management firm (World's End in Los Angeles), Susan Rogers' first engineering gig was working on Prince's studio staff—not a bad way to break into the business. She stayed with the Purple One from 1983 to 1988, and since going independent she has engineered projects for Michael Penn, Edie Brickell, the Jacksons, and Tevin Campbell. Rogers coproduced David Byrne's latest album (titled simply *David Byrne*) with Byrne and Arto

Lindsay and is currently producing the Odds in Vancouver. She always travels with a trusty "second engineer," her wonder dog, Gina.

The Breaks. Obviously, getting the gig as one of Prince's engineers was a major career boost. He does an incredible amount of work, however, and there came a time when I decided that I just couldn't offer him anything anymore. But even with his name on my résumé, I was scared to leave. I had been living in Minnesota and was out of the engineering scene. I had no contact with other engineers or producers, no clue about personal managers or agents, and no idea how I would get work. Luckily, I got a call from the Jacksons, which brought me to California and introduced me to the recording industry there. That exposure enabled me to get a manager and forge out a career as an independent.

Production Style. I'm not a songwriter or a musician, so I became an engineer/producer by default. I usually work with artists who are quite capable of producing themselves. They don't need as much musical input as they need technical help with the recording medium. I often become a facilitator; the artists have an idea of what they want to do, and they need an experienced engineer to get that idea on tape. Of course, I also put my own ideas into the project.

Sometimes my production methods are determined by geographical location. Studios on the east coast are generally smaller and very expensive, so you have to work faster and put in longer days. On the west coast, however, everyone has home studios, so the pace is slower and a bit more open to experimentation. For example, when I did the Geggy Tah album for David Byrne's Luaka Bop label, I recorded it in the band's house using a Mackie CR-1604 and a Stephens 2-inch 16-track deck. We didn't have a lot of money, but we did have a lot of time to experiment and get things to sound exactly the way we wanted. As a consequence, it's probably the record I'm proudest of.

Going from big commercial studios to home studios takes some getting used to, but I find working in home

studios to be a wonderful experience. It changes your working habits. For the Geggy Tah record I had just three mics to choose from. I also had to plan out each song ahead of time, because only having sixteen tracks to work with meant that I had to be pretty facile at combining and submixing tracks. But going through the home studio process has made me more efficient when I work in a large studio.

I don't find working in alternative recording environments to be repressive, because I've found that many artists these days are actually calling for less technology. They want to concentrate on documenting great performances, so the main thing is to record wherever, and however, they're comfortable. I don't mind. If an artist wants to record in his or her bathroom, I'll just say, "Okay, let's go!"

The current marketplace is very accepting of this "lo-fi" approach. Ten years ago, records had to sound more polished. But I think the lo-fi approach is a great thing, because it allows the artist with less money to be just as competitive as the artist with a big budget.

The Gender Thing. I know that some artists and A&R people might see my name on a list and automatically disregard me just because they don't want to work with a woman. In general, I believe the industry does not encourage women to be engineers and producers. I can say this with some authority because so many women have assistant engineer positions and never move up. This lack of advancement is not because these women are getting married or pregnant and leaving the industry. It is because they are not being offered the jobs.

Susan Rogers with her "trusty second engineer" Gina.

THE GENDER GAP

PATRICE RUSHEN

It's difficult to find something that the multitalented Patrice Rushen *hasn't* done in the music business. She is an international recording artist, composer, session musician, engineer, and producer. She has won an ASCAP Songwriter's Award and has been nominated for Grammy Awards as both a vocalist and instrumentalist. She produced Sheena Easton's *No Strings* album, which included the hit "The Nearness of You" from the movie *Indecent Proposal,* and has scored numerous movies and television shows including *Hollywood Shuffle, Without You I'm Nothing,* "The Women of Brewster Place," and "Comic Relief V." Last year, she was musical director for the mammoth Janet Jackson World Tour. No wonder her fans call Rushen "Ms. All That."

Rushen has also stomped down a few sexual barriers in the industry. She was the first woman in 43 years to serve as

Patrice Rushen

musical director for TV's Emmy Awards, the first woman musical director of the NAACP Image Awards, and the first and only woman musical director for the People's Choice Awards. Rushen's new solo album, *Anything But Ordinary,* was just released last month by Sin-Drome Records.

Production Style. The producer's role is to ensure that the artist's cre-

ative goals are met. Part of the art of producing is being able to read the artist's creative needs and determine how much to push them to get the best performance. It's an interesting balancing act. There is an incredible amount of logistics to deal with, and you can't let any of them interfere with achieving the best artistic outcome. I've even discovered that *where* you record becomes part of the production process. It's essential to find a studio where the artist feels comfortable. Of course, you can't control everything. Sometimes you hire the musicians and just hope everyone is having a good day and that the chairs don't squeak.

In commercial, mainstream projects, the weight is often on the producer to minimize the artist's weaknesses. Many of today's artists are signed because their delivery of a song may generate a hit, and a lot of nonmusical factors are involved: the artist's fashion sense, what community they represent, how charismatic they are on video, and so on. Often, the production comes down to how the vocalist can deliver five notes, and suddenly a career is based on those five notes. The artistic depth of the artist is not the primary focus anymore.

I think this is why sound is everything these days. Many producers dress up marginal performers in these amazing sonic productions. But as a contemporary producer, it has been very difficult to suppress the part of me that believes music should have integrity. The *performance* should be the best it can be. When I get less than that out of a performance—as an artist, producer, or music consumer— I feel cheated. I'm constantly disappointed at what gets released. It's just tape, why not record it again until it's great? I believe that if someone is going to take the time to listen, you owe it to them to give them the best you've got.

The Gender Thing. The industry still sees female producers as being a novelty. There certainly aren't very many of us around! And unfortunately, the

perception is because you're female, you're less effective. Sure, a female's point of view regarding an artist is usually different than a male's, but if you queried five male producers about the same artist, they'd have different views, too. The female perspective should not carry less weight. Of course, there is still a double standard. But my stance is that if women have to be better than men to break in, fine, let's do it. It does no good to sit around and whine, "Why can't there be parity; why do I have to be better than my male peers?" Just get over it and *be* better. I'm not afraid of being the best I can be. If that's the challenge, so be it."

THE SVENGALI STIGMA

Unfortunately, even though many women are out there making marvelous records, boorish musicians continue to devalue their efforts. It's not uncommon to hear that a woman producer had a man lurking behind the scenes, secretly doing all the work. Even an artist as talented as Kate Bush falls victim to these rumors. Besides being an evocative singer and songwriter, Bush is perhaps one of the most fearlessly creative producers working today. But the sonic majesty of her recordings doesn't stop misguided fans from assuming that her longtime engineer Del Palmer is the man behind the curtain.

"People will come up to me occasionally and say, 'Well—wink, wink—I'm sure that you're the one who *really* does the producing,'" admits Palmer. "Of course, I always tell them that Kate is in complete and total control of her work. I'm probably a little too sensitive about it, but no matter what I say to one of these doubters, I always feel like a hint of skepticism remains."

Music should be a joyous noise, where chops and ideas are paramount, and sex and race disappear in a roar of artistry and emotion. But until these insidious "hints of skepticism" about the work of women producers and engineers disappear, we really can't call ourselves a music *community*. Let's grow up, give proper credit where it's due, and encourage more women to move up the recording hierarchy. Until that day... .

EM Editor Michael Molenda *has spent almost twenty deliriously happy years abusing guitars, eating microphones, and sitting behind mixing consoles.*

TONY FRIEDKIN

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Dynamic Duos, Part 1

Compressors and limiters tame signals that are too hot to handle.

By Scott Wilkinson

Suppose you've just recorded a killer lead vocal on your new modular digital multitrack (MDM), the best you've ever sung. But when you listen back, you hear some ugly distortion in the loud passages. You were so caught up in the performance, you didn't notice that the level meter went above 0 dB. This inevitably results in clipping, a nasty form of digital distortion. How could you have avoided this tragedy?

The most common method is to use a *compressor/limiter*, a particular type of *dynamics processor*. Dynamics processors serve a multitude of purposes by altering the *dynamic range* of an audio signal in various ways.

Dynamic range is simply the difference, measured in decibels (dB), between the lowest and highest levels in a given signal or audio device. (The lowest level is generally the noise floor.)

The term can also be used to describe the maximum range of signal levels that a given audio device or recording medium can accommodate without distortion.

For example, Ampex 499 open-reel analog tape has a dynamic range of about 80 dB, because it cannot cleanly handle signals below its noise floor of -71 dB and becomes saturated when signals exceed +9 dB. Cassette tape has less dynamic range than high-quality, open-reel tape; Type II cassettes generally offer a dynamic range of about 60 to 70 dB.

Digital-audio systems—MDMs such as the ADAT and DA-88, DAT recorders, and hard-disk recorders—feature a theoretical dynamic range of 96 dB, although the practical range is more like 90 dB. This is much better than most audio recording media, but unlike analog tape, you must not exceed the upper limit by any amount, or you will hear the dreaded digital clipping.

COMPRESSION AND LIMITING

Compressors are probably the best-known dynamics processors. If the level of an input signal rises above a user-specified *threshold*, the compressor controls the amount by which the signal's level can continue to rise. If the input signal stays below the threshold, it is not affected.

The *ratio* parameter determines the extent to which the level of a signal

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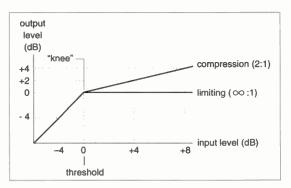
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Digital Magic.

FIG. 1: Below the threshold, an input signal at -4 dB is output at -4 dB. If the compression ratio is 2:1, a +4 dB input signal is output at +2 dB. If the limiter is set to a threshold of 0 dB and a ratio of ∞:1, all signals above the threshold are output at 0 dB, no matter what their input level.

above the threshold is compressed. For example, if the ratio is 2:1, and the input signal is above the threshold, a 2 dB increase in the input level results in a 1 dB increase in the output level. If the ratio is 4:1, a 4 dB increase in the input level results in a 1 dB increase in the output level.

Other important compression parameters include *input gain* and *output gain*. The input-gain control is used to adjust the overall input level to compensate for a particularly strong or weak incoming signal without clipping the compressor's input circuitry. Output gain adjusts the final output level, which is especially important because it lets you compensate for the way a compressor attenuates signals. However, raising the output level also raises low-level signals, including noise. This can be corrected using a different type of dynamics processor, which we'll discuss next month.

The operation of a compressor can be illustrated in a graph of input level versus output level. As you can see in **Fig. 1**, if the input level is below the threshold, it is equal to the output level; what goes in comes out unscathed. Above the threshold, the rate of increase in the input level is controlled according to the ratio. This is indicated graphically by the slope of the line above the threshold; the closer that line is to being horizontal, the greater the compression ratio.

The angle in the graph at the threshold point is called the *knee*. In some compressors, the knee is rounded rather than forming a sharp angle. This is called *soft-knee* compression. Soft-knee compression tends to smooth out the transition at the threshold setting, making it less noticeable when the compressor kicks in.

Some compressors offer *attack* and/or *release* parameters. Once the input signal rises above the threshold, the attack parameter determines how long it takes before the compressor starts attenuating. This may seem a bit strange. Why would anyone want to delay the compressor's operation? Sometimes, you want to pass high-level peak transients, such as drum hits and instrumental attacks, through the compressor to retain the bite or sizzle of the original sound. By setting the attack parameter to 100 milliseconds or more, quick peaks get through before the compressor can process them.

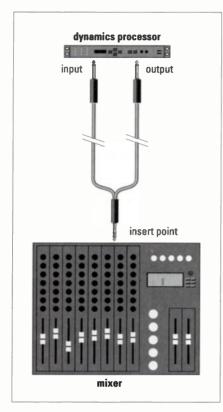As the level of the input signal drops below the threshold, the release

FIG. 2: Dynamics processors are often connected to a mixer's insert point, which routes the entire signal out of the mixer, through the processor, and back into the mixer.

parameter determines how long it takes for the compressor to stop working. Slower release times smooth out the signal and, in some applications, help increase sustain (discussed shortly).

When the ratio is high (10:1 and above), compression becomes *limiting*. This effectively prevents the output level from rising much beyond the threshold level. At a ratio of ∞:1, the limiter clamps a lid on the signal level, allowing it to rise no higher than the threshold (see **Fig. 1**). A compressor/ limiter is a dynamics processor with a wide variety of compression ratios to accommodate both applications. Many compressor/limiters can even perform both tasks simultaneously.

THE RIGHT CONNECTIONS

Next, let's take a look at how compressors are connected to a sound system. Unlike most other effects units, which split the input signal into two parts and mix the processed and unprocessed copies, all of the signal is routed through a compressor/limiter. Otherwise, a copy of the unprocessed signal would get through, which is exactly what we're trying to prevent.

In the studio, compressor/limiters are connected to the mixer's *insert points*, rather than the effects or aux send/return loops (see **Fig. 2**). This routes the entire signal out of the mixer, through the processor, and back into the mixer. (In contrast, an aux send/return loop splits the signal, sends one copy through the processor, and mixes the result with the unprocessed signal; see "Square One: In the Hall of the Reverb King" in the August 1994 **EM**.)

This connection usually requires a ¼-inch, TRS (tip-ring-sleeve), stereo-to-mono, "Y" cable (i.e., the stereo plug is attached to two 2-conductor cables, which terminate in standard ¼-inch or XLR plugs; see **Fig. 3**). The ring on the stereo plug goes to one mono cable, which carries the signal from the mixer to the processor. The other mono cable in the "Y" returns the signal from the processor to the mixer via the stereo plug's tip. (This is reversed in some mixers, such as the Alesis 1622; check your mixer's specs to be sure.)

Sometimes, you need to process the signal from only one mixer channel, in which case you connect the processor to that channel's insert point. If you want to process several signals from different input channels with the same device (e.g., if you have several mics on a horn section or choir), just route those channels to one of the mixer's subgroups (see "From The Top: Mixers and Mixing" in the May 1991 **EM** and "Recording Musician: Maximizing Your Mixer" in the April 1992 issue) and connect the processor to that subgroup's insert point.

Finally, many engineers process the entire mix during mixdown. You need a stereo compressor for this job. Most stereo compressors include a switch that links the two channels to operate together, with one channel as the master. They can also be unlinked to operate independently. Simply connect the processor to the insert points for the mixer's main L/R buses, using two "Y" cables.

APPLICATIONS

One of the primary applications of compression is smoothing a vocalist's performance dynamics. If a singer's mic technique or vocal control is less than optimal, a compressor can clean up the volume surges and fades. This can be a real gig-saver onstage, as well

as in the studio. When recording vocals to tape, I use relatively mild compression with a threshold of -10 dB and a ratio of about 3:1.

If you want to increase the impact and punch of virtually any recorded sound, try compressing it slightly. This is particularly true when recording to digital media, which lack analog tape's coloration and slight, natural compression.

A compressor is also useful for preventing unwanted or excessive tape saturation when you are pushing the limits of the recording medium's dynamic range. For example, if you are planning to release your music on cassette, compression is vital to accommodate its limited dynamic range.

If you're recording on a digital machine, where even the slightest amount of clipping is unacceptable, you need to go beyond simple compression and limit the signal. Place the limiter between the signal source and the tape
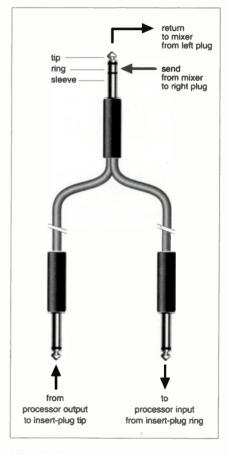

FIG. 3: An insert "Y" cable consists of a TRS plug at one end and two 2-conductor, ¼-inch or XLR plugs at the other end. In most mixers, the TRS ring sends the signal to the processor, and the tip returns the signal to the mixer.

deck, and set the threshold to 0 dB and the ratio to ∞:1.

As mentioned earlier, compression is also commonly applied to an entire mix during mixdown. This produces a tighter, more professional sound, with more impact and kick. It is especially important for radio airplay because radio transmitters have a very narrow dynamic range. As a result, radio stations routinely compress the heck out of their signal.

Another application of compression is to artificially sustain each note. Guitar and bass players often use compressors for this purpose. To increase the sustain of a sound, set a relatively low threshold (-20 to -30 dB or lower), high ratio (5:1 to 10:1 or higher), very fast attack time, and slow release time. Adjust the input and output gain as high as possible without distortion in your system.

Under these conditions, the input level can vary over a wide range, while the output level remains relatively constant. For example, as a vibrating guitar string dies away, the instrument's level may fall by 10 dB or more. If the ratio is set to 10:1 or higher, the output from the compressor will fall by only 1 dB, which sustains the sound and makes it more punchy and even. This works very well on lead and rhythm guitar, as well as bass.

Limiting absolutely prevents a signal from clipping in digital recording. It can also protect speakers and eardrums from being blown out by levels that are too hot. You must be especially careful how you set the limiter's threshold in sound-reinforcement applications. If the threshold is too high, you're unprotected; if it's too low, your amplifiers must work extremely hard, but you won't get enough sound.

TAKE IT TO THE LIMIT
Compressors and limiters are essential tools in the studio and onstage, offering a variety of useful functions. Using them properly requires a little practice, however, so it's a good idea to spend some time experimenting.

Next month, we'll take a look at the other types of dynamics processors. Until then, keep your ears on the music.

EM Technical Editor **Scott Wilkinson** *finds that a touch of compression limits his audio woes.*

Green Jellÿ Oozes Out

A terrible band expands into video games and comic books.

By Michael Brown

After being gonged on *The Gong Show* in 1987 and then practically swindling a major record company in a 1991 deal that nets you $60,000 in video equipment, what do you do for an encore? If you're the gonzo-band Green Jellÿ, you launch yourself on a multimedia production juggernaut, beginning with the soundtrack for the Spider-Man video game, *Maximum Carnage.*

Of course, Green Jellÿ accomplished a few things in between those two momentous events. They sold more than two million copies of their whacked-out soundtrack *Cereal Killer,* for instance. Then they took the MTV audience by storm with their stop-frame animation, fractured fairy-tale video, "Three Little Pigs."

How did that lead them into producing a video-game soundtrack? As Green Jellÿ's Moronic Dicktator might say, "Why don't you sit right back, and I may tell you a tale."

IN THE BEGINNING...

"The band was just a roving party for a long time," says singer Gary Helsinger (aka Hōtsy Menshot). "None of us can really sing—much less play—so we decided we would be the world's worst band. We took on stage names and wore elaborate costumes to divert the audience's attention away from the fact that we were terrible musicians and singers."

Prior to the 1993 releases of the *Cereal Killer* long-form video and CD soundtrack, the Green Jellÿ lineup was in a state of constant flux. Seventy-four people have been members of the band during its 13-year existence. Now that the band has tasted a measure of commercial success, the lineup has "jelled" with a cast of ten. Helsinger has been with the band for eight years.

"At our early live shows, the band would usually outnumber the audience," explains Helsinger. "Some nights

Gamers jam to a Green Jellÿ soundtrack while they pummel the villians in Acclaim's *Maximum Carnage* video game.

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we would have as many as twenty people on stage. There might be three guitar players, three bass players, and five percussionists. Somebody might show up and bang on a car hood all night long."

Green Jellÿ songs freely steal from the musical styles of the rock, punk, metal, and pop bands the band targets for skewering. Victims of the band's twisted sense of humor have included the Red Hot Chili Peppers, Metallica, and XTC.

Unlike acts that perform send-ups of other bands, Green Jellÿ doesn't replicate the melodies of their targets. "None of us can play, so we can't lift [musical] parts off records and do cover versions," admits Helsinger. "We have to write our own songs."

SCAMMING A DEAL
The story of how the self-proclaimed world's worst band secured a lucrative record contract is as strange as some of Green Jellÿ's songs.

"I was doing A&R for Chrysalis Music," says Helsinger, "and [band manager] Kevin Coogan was an A&R assistant at Zoo Entertainment. By this time, we were drawing about a thousand people to our shows. We even took little road trips to San Francisco and San Diego. Kevin, [lead singer] Bill Manspeaker, and I wrote our own record and publishing deals. Then, we went to Zoo Entertainment and said, 'Hi. We're the world's first video band. We're going to make you a video album for—uh—$60,000. Wanna do it?' And they basically said, 'Sure, here's a check.'"

Helsinger sticks to his story, no matter how incredulous the listener may be. On the other hand, Green Jellÿ's story demonstrates that unconventional

FIG. 1: Members of Green Jellÿ have developed more than 25 larger-than-life characters. Here (from left to right) are Super Elastic (Joe Cannizzaro), Orange Krunch (Gary Helsinger), Piñatahed (also Gary Helsinger), and T.O.A.H. (Bob Wieder).

methods can yield results.

"We were totally shocked," he maintains, "because we had lied. We had never made videos, and we had never done an album. We totally lied, and they believed us. Then we had to fake it, because we were afraid we'd get our legs broken if we didn't come up with something."

Green Jellÿ used their advance to buy video cameras and editing equipment, in turn producing eleven music videos for *Cereal Killer,* none of which MTV was willing to broadcast. That situation changed soon after "Three Little Pigs" became a hit. A radio hit, that is.

"It blew our minds that 'Three Little Pigs' became popular on the radio first," says Helsinger. "We had concentrated on the video much more than the songs. We recorded all the music for the video in just three days; we didn't even release the CD until six months after the video was out."

FRESH JUNK
This time around, Green Jellÿ released the CD-version of its new production, *333,* in advance of the long-form video. And according to Helsinger, the band exhibits some serious musical chops on the new album. "You can ac-

tually listen to it without the video," he says. "It really rips."

Nonetheless, each of the new songs (except for "Maximum Carnage") is built around one of the band's cartoon-puppet characters. "We don't write a song until we've already developed the character and built his or her costume," says Helsinger. "If the costume is funny, we'll spend an hour or so writing a song about it. We consider ourselves to be more of a rock-and-roll puppet show than a band."

One of the new characters that Helsinger plays is called Orange Krunch. The foam-rubber and latex costume of the huge, bald, tattooed, and pierced crunch-rocker is 8-feet tall and weighs a whopping 150 pounds (see **Fig. 1**).

"It's giant and heavy and smelly," Helsinger says of the costume, "and I seem to be the guy for that job. Besides, Bill [Manspeaker] is the president of the company, and you just can't make the president wear smelly rubber."

The band recorded *333* at Ooz Jellÿ, their new Hollywood production facility, which is housed in the late Frank Zappa's former rehearsal studio. ("Yeah, we've got a little of the Frank vibe happening," says Helsinger.)

A joint venture between the band and Zoo Entertainment, the Ooz Jellÿ facility features a television sound stage, a stop-motion animation system, 3-D computer-animation systems, a digital

The Green Jellÿ Lineup

Performer	Alter Ego	Role
Bill Manspeaker	Moronic Dicktator	vocals
Gary Helsinger	Hötsy Menshot	vocals
Kym O'Donnell	Sadistica	vocals
Joe Cannizzaro	Dunderhed	vocals
Bob Wieder	Bad Water Bob	vocals
Steven Shenar	Sven Seven	guitar
CJ Buscaglia	Jesus Quisp	guitar, vocals
Mike Bloomquist	Rootin	bass
Roy Staley	Coy Roy	drums
Rosemarie Mattrey	Rosebud	photographer
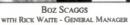
video-editing bay, and an 8-track demo studio. The band recorded their new CD at Ooz Jellÿ, but they brought in a rented 24-track mobile truck for tracking and mixing. The soundtrack for *Maximum Carnage*, however, was recorded using the equipment in their demo studio.

Because they were going to digitize it into video-game format, the quality of the sound wasn't as critical, according to Green Jellÿ guitar player Steven Shenar (aka Sven Seven). "We didn't have to go into a big studio or do anything heavy. We recorded to an 8-track reel-to-reel deck and then mastered to DAT. Acclaim's programmers digitized the music from that."

COMIC CAPERS
Avid comic book fans, Green Jellÿ established a link with the comic-book industry by renting a booth at the San Diego Comic Book Convention for several years.

"It started out as a way to promote ourselves," says Helsinger. "We convinced a lot of the comic-book shops to stock our video and our CD."

That association with the comic-book industry brought the band to the attention of Marvel Comics, and it became a factor when Marvel licensed Acclaim to produce the *Maximum Carnage* video game.

"There were several record companies eager to join the project," says Dan Harnett, public relations manager at Acclaim. "Naturally, we worked very closely with Marvel on the game, and they knew that Green Jellÿ were big fans of their comic books, especially the Spider-Man series."

Although Acclaim provided the band with early video footage from the game, Helsinger and Shenar say the footage arrived too late for them to make use of it. The band, therefore, wrote and recorded the soundtrack without knowing how the game would actually look. (Although they spent a lot of time talking to the game designers on the telephone.) The soundtrack—more than 30 minutes of music—was recorded over a two-week period.

"After we gave them the theme music," says Shenar, "they would tell us, 'Can you do something that's like the theme, but that's more *down*, for this part when we introduce the bad guy?' None of the game developers

were musicians, so they told us what they wanted using their own lingo."

The band was motivated to do this project by a love for the comic book genre. "Yeah, we're big fans of comic books," says Helsinger. "It was an honor to write a soundtrack for Spider-Man. I worship that stuff."

"We did it for fun," adds Shenar. "Acclaim agreed to pay the production costs—if they ever send us a *check*—but we essentially did it for free. It was a promotional vehicle for both of us."

REFRIGERATE AFTER OPENING

Green Jelly's production schedule for 1995 includes developing an animated television series, an interactive CD-ROM, a comic book series that will be distributed by DC Comics, and licensing a line of toys.

"In the comic book," explains Helsinger, "Dunderhed receives a distress signal from Earth, because Moronic Dicktator has enslaved all the children through television and his biological computer chip called 333. Ooh! That's the name of the new CD, isn't it? So Dunderhed gathers together all the 'stuporheroes' from around the universe, and a big battle ensues."

The television show was being developed for MTV, according to Helsinger, but that deal fell through. "With MTV, the show got turned around into something that it didn't start out to be," he says. "MTV insisted on owning the rights to all the characters, which we weren't willing to sell. I mean, our whole production company is about developing new characters and exploiting them."

Whether you consider Green Jelly to be a talented comedy troupe, a bad parody band, or just a bunch of musical con artists, the band's history proves that truth is indeed stranger than fiction, and good things really can happen when you follow your bliss.

"We're stunned by our success," says Helsinger. "We know it will be over, sooner or later. We'll just do it as long as we can and as long as people like it. We don't even really know how to do this stuff, we just know how to make each other laugh."

EM Associate Editor Michael Brown *justifies his video-game purchases by claiming they're for his children.*

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Fear of Stereo

Where have all the intrepid explorers of the sonic spectrum gone?

By Michael Molenda

Record productions are becoming as monochromatic as the fashion victims in that television ad where the hip designer screams, "Everything black!" There's no *dimension* anymore. Everything sits in the middle of a static universe, where stereo sound modules, signal processors, and guitar preamps dole equal shares of sonic sparkle to each channel. It's not the fault of the gear. It's that few musicians seem willing to tweak their stereo boxes to break the yoke of left/right conformity.

Just because a piece of gear is labeled "stereo" doesn't mean that it automatically exploits the full potential of the stereo spectrum. That's the user's domain. And think about this: If you mix with a collection of untweaked stereo processors and sound modules that simply output similar levels and timbres through each channel, what have you got? Enhanced *mono?*
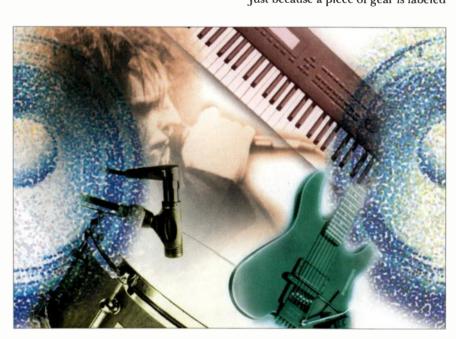
Timid musicians are missing the creative boat when they ignore the possibilities of outlandish stereo placement. Instruments should rage all over the sonic spectrum, and mixes should come alive with movement. It's time to stop being sonic scaredy-cats and mine the stereo field for all its ample treasures. I say, let's get psychedelized!

SPLIT SIGNALS

In the old days, when digital technology was just a military secret, most vinyl pop records were available in monaural and stereo formats. (Stereo records usually cost a dollar more than the mono versions, which was a major budget buster in my allowance days.) Many pop producers really went to town with stereo. Suddenly, the Beatles were singing on one side of the living room, while their instruments rock 'n' rolled from the other. Wow!

Monaural records always sounded like mysterious explosions of indecipherable instruments with a voice rising above the din. But stereo introduced me to the individual timbres of

particular instruments. I couldn't help but notice acoustic guitars, organs, and saxophones when all of my Monkees and Paul Revere and the Raiders records were constantly tossing things exclusively in one speaker. Now, however, when I play "I Had a Dream" for someone, they just giggle at all the silliness. Lead vocals in one speaker, indeed!

Okay, some of that stuff *is* pretty dumb. However, I consider it rather closed-minded and snobbish to pooh-pooh these conspicuous exploitations of the stereo spectrum. The best examples of hard-stereo music productions were expansive, creative, and evocative. Oh and by the way, they were also tons of fun.

LEFT/RIGHT RATIONALE
At this point some of you are probably still saying, "Lame gimmickry aside, is there any sensible purpose to exploiting the stereo spectrum?" Actually, there are some very practical reasons for "nonambiguous" placement of elements within the stereo field. For example, the alternative music market often champions a return to stripped-down, live-sounding record produc-

The production of Sam Phillips' *Martinis & Bikinis* album (Virgin Records) showcases how subtle, almost psychoacoustic use of signal processing and hard stereo panning can stretch the sonic spectrum into the wide open spaces.

During his tenure as leader of Electric Light Orchestra, Jeff Lynne's cinematic record productions were often punctuated with bizarre flanging and phasing effects that panned from side-to-side.

tion. What better way to emulate live performance than to document the sonic image of the concert stage?

By using monaural source signals during a mix, you could pan all the drum tracks slightly to the left, and position the bass a touch to the right. The rhythm guitar and keyboard could sit at the far left, while the lead guitar and background vocal appear far right. Only the lead vocal should command center stage. A concert hall ambience can be simulated by assigning all tracks to a stereo reverb unit (set to a Large Hall or Chamber algorithm with approximately two seconds of decay time) and subtly positioning the effect behind the more prominent source sounds. Now, you not only have a performance that sounds live, you also have an aural *presentation* that enhances the performance. In my book, wrapping an artist in a nurturing, supportive sonic environment is what audio production is all about.

The simulated live stage is a starting point for other practical uses of hard stereo. In the previous mixing example, the lead guitar really stands out, because it is virtually alone in the right-channel sound space.

● RECORDING MUSICIAN

It does not have to fight other instruments and tonal registers to break out of the mix. Hard stereo positioning can also clarify source sounds that are buried in a bloated sonic environment. If an important musical phrase isn't cutting through your mix, pan the instrument (or instruments) performing the phrase to the far left or right. The phrase should now jump right out at you. If the phrase still sounds muddled, make sure that you're not gumming things up by drowning the track in reverb or other effects. In a thick mix, drier, less-effected elements often sound clearer than tracks saturated with processing.

And speaking of clarity, a predominantly dry mix with monaural elements split into hard stereo positions can sound amazingly transparent. Listen to *Wood Face* by Crowded House, *Martinis & Bikinis* by Sam Phillips, or *Jollification* by the Lightning Seeds for some good examples of wide, crystalline sonic vistas. All of these productions are drier than a desert sirocco and employ monaural elements panned for hard stereo.

DIVIDE AND CONQUER

Unsubtle stereo placement can help solve other nagging mixdown problems. If you can shake off your addiction to stacked, stereo tracks and conventional stereo signal processing, you can beat the aural demons that turn mixes into mush. Here are some common roadblocks that can be avoided by twisting your mixer's pan pots.

Muddy bottom. If your punch is puny and you've tried every EQ tweak your mixer can muster, try turning the kick drum channel's pan pot toward 10 o'clock, and pan the bass track toward 2 o'clock. The slightly off-kilter positioning of these low-end elements diminishes bass glut in the center space, which cleans up the mud and clarifies the groove. You could also solve this problem by buying or renting a spectral enhancement processor, but turning a pot is definitely a cheaper solution.

Washy cymbals. Got a chorus of annoying sizzles in your high-end and no de-esser to calm down cymbal ring? No sweat. Just move both of your stereo

On several Utopia albums, Todd Rundgren's guitar solos jumped out of the mix by moving somewhat randomly between the right and left speakers.

overhead tracks full left or right. A hard-panned, monaural overhead track can stop cymbal wash from saturating the stereo perspective. (There will still be a bit of cymbal bleed on the tom and snare tracks, however.) It's also kind of cool to hear ride- and crash-cymbal figures explode from one side of the mix.

Soupy rhythm guitars. This is a bad one. Unless you're in some gothic grunge band, murky guitars are totally lame. Start by taking all the reverb and other effects off the guitar tracks. If you have a thick Les Paul-style track and a thin Strat-like track, try panning the two distinct timbres hard right and hard left. The same approach should work if one guitar is playing full chords and the other is doing single-note rhythm licks. If you only have one guitar track, it still helps to move it slightly to one side or the other—say, 3 o'clock or 9 o'clock. When you start bringing the effects back in, go easy. Don't submerge the tracks in signal processing that will only muddy things up again.

Paunchy pads. Stacked keyboard tracks can be deadly. If you lay the sounds on too thick, the production can smother itself. Try panning staccato counter melodies opposite chordal pads. Listen to the timbres of your patches (organs, horns, etc.) and try not to have similar sounds sitting right on top of each other. Also, consider using well-positioned monaural tracks, rather than a wash of stereo tracks panned hard left and right. Stacking stereo preset patches from a variety of keyboards can really get gooey if the signals output from the module's left and right outputs sound similar. Of

course, the audio glut can be avoided if you program extremely different tonalities for the right and left outputs of each patch.

STUPID STEREO TRICKS

Of course, you *can* get really down and dumb with stereo manipulation. Sometimes, it can be thematically appropriate to toss a few sounds from speaker to speaker, but usually any bizarre stereo tricks are produced purely for the wow factor. Want to have some fun? Steal these wacky ideas—if you dare.

Merry-go-round solos. When the soloist is cranking out his or her solo, grab the pan pot and twist it to the left and right. You can time your pans to the music, or go completely out to lunch. (The more random, the better.) In either case, the lead break becomes extremely animated, as the solo flies around the stereo field. This effect is a real kick when you listen through headphones.

Ghost vocals. On ballads, it can be fun to put a completely dry lead vocal in the left or right channel and assign the vocal reverb to the opposite channel. If you're not brave enough to try this trick on the lead vocal, it works with background vocals, too.

Fractured fade-ups. Got the hankering to start a song with a really weird introduction? Assign your entire mix to a stereo subgroup and slowly fade the stereo mix up while a friend turns the pan pots from full right to full left and then to the conventional left/right configuration.

Messing with you. You can drive listeners crazy by subtly changing the stereo position of support instruments such as rhythm guitars and keyboards. For example, in verse one, leave the guitar hard right, then move it slowly to the left during the first chorus, and leave it hard left for verse two. Reverse the process for the next chorus and verse.

Double your fun. Who says that doubled vocal tracks have to be mono? Put the original lead vocal on the left side and pan the doubled (overdub) vocal to the right.

Hello, Tom. During a tom fill, pan the appropriate tracks from left to right (or vice-versa) so that the fill sounds like a train rolling through a railroad crossing. You can get stranger by panning any reverb you've assigned to the drums in the *opposite* direction.

Early Beatles albums showcased the wonders of stereo by enforcing strict segregation: Instruments rocked in the left speaker, while vocals blared from the right.

Rolling Thunder. You can create a storm front of cascading guitars by running a center-panned, single-note guitar line into a stereo delay processor. Simply program the left side of the delay to produce eighth notes with multiple repeats, and program the right side to spit out multiple repeats in sixteenth notes.
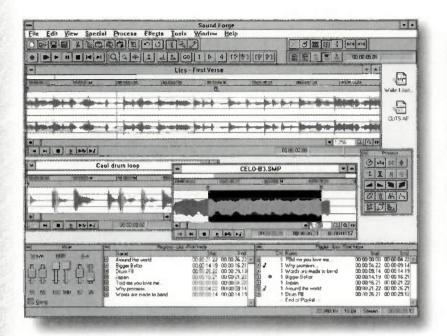
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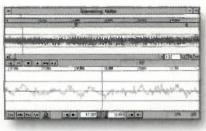
Perhaps many musicians avoid unconventional (or downright wacky) stereo treatments, because they fear their work will be disregarded as gimmicky. Well, there's always that risk. Once you put your music out in the world, it's fair game for cruel pot shots. But what would have happened if Jimi Hendrix was afraid to expand beyond his traditional R&B chops? Or if Herbie Hancock and Jan Hammer decided they'd better not rock the boat with this electronic music thing?

You can always produce conventional work. The road map is clear. But the next time you're mixing a project, try doing a good, traditional mix *and* a completely nutzo stereo extravaganza. The only thing you're risking is a few minutes of mastering tape, and the experimentation might open your mind to a few truly revolutionary mixing ideas. So what *are* you afraid of? ●

Surviving Club Gigs

Serious planning can make your live performances more enjoyable.

By Michael Molenda

Let's talk about lust. On the Rolling Stones' _Voodoo Lounge_ tour, Mick Jagger is jumping around a stage larger than most municipal airports. This summer, Pink Floyd's stage was so impressive that it received better reviews than the band. And many U2 fans still haven't forgotten the surrealistic empire of media machinery that dwarfed Bono and company during the Zoo TV tour. The chance of bashing out a few chords in such wondrous performance environments can be a powerful incentive to hit the live concert circuit.

But, whoa! Unless you can sell out football stadiums, don't plan on renting the _Voodoo Lounge_ stage when the Stones are done with it. Most musicians perform in clubs that are smaller than Barbra Streisand's portable dressing room. You can forget about the luxuries of well-tuned sound systems, light shows, video projections, crystal-clear stage monitors, and sound checks. The reality of many small club gigs is coping with abused P.A. systems, indifferent house sound crews, and stages so tiny that you'll spend entire shows with the back of your legs rubbing against a kick drum and your left earlobe being rhythmically pulverized by the headstock of a Fender Precision. It's a sobering experience to realize, that for most of us, the concert stage is not a tool to enhance one's performance, but a wicked patch of hell on earth.

Fortunately, there are ways to tame the savage club environment. If you just employ your wonderful, creative mind to think about more than chord changes, you _can_ survive the travails of bringing your music out in the open.

GET ORGANIZED

The best protection against the frustrating anarchy of small clubs is scrupulous planning. Rehearse your act until it's second nature. Nothing is more annoying to an audience (or house soundperson) than a band that diddles away onstage, apparently unaware of

Pink Floyd's stage explodes with a thousand points of light and thunderous sound, while club musicians are usually crammed into dark corners and must wrestle with battered P.A. systems.

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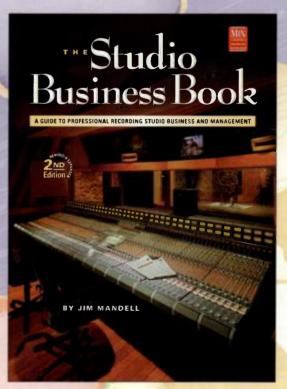©1995, 475 pp. (Spiral) **1950A) $74.95**

Announcement: Beginning with the 1995 edition (available in January), Recording Industry Sourcebook has joined Mix, Electronic Musician, Mix Bookshelf and Mix Directories as part of the CBM Music and Entertainment Group. This massive industry guide is a top-quality, spiral-bound, tab-divided directory which has been critically acclaimed for its comprehensive scope of information and design characteristics. With over 10,000 updated listings in 75 categories, *The Recording Industry Sourcebook* offers comprehensive directories of record labels, producers, managers, distributors, attorneys, equipment suppliers, music video companies, media contacts and much more. Entries list contact names, titles, phone and fax numbers, styles of music preferred and information on whether or not they'll accept unsolicited demos. The entire Sourcebook database is also available on disk, packaged with contact management software; please call for details.

THE STUDIO BUSINESS BOOK, Revised Edition

Jim Mandell ©1994, 288 pp. (P) **1319A) $34.95**

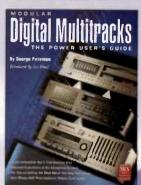MixBooks announces the updated and expanded edition of *The Studio Business Book*, by Jim Mandell. This informative, wide-ranging guide is a necessity for anyone running or owning a studio, whether it's a project setup in your basement, a midrange commercial studio or a world-class room hosting the stars! You'll get in-depth, detailed, *specific* information on all aspects of putting together and managing a recording studio, including writing a business plan; getting funding and finding partners; choosing and buying equipment; determining your monthly expenses; setting hourly rates and bidding on projects; developing new income sources and making contacts; doing advertising and PR; scheduling employee shifts and sessions; studio politics and psychology; managing, hiring and firing personnel; and more. *The Studio Business Book* also includes information on the legal debate over home studios and how to put together an employee handbook, plus a reference guide to studio terms and technology, a list of industry organizations and a recommended reading list.

MIX & EM

Books

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what to do next. Although many San Francisco bands made a ceremony of tuning up during the city's psychedelic heyday, few of today's music fans—who have developed their attention spans experiencing the slam-bam presentation of MTV—have the patience to watch musicians struggle with ineptitude. You don't have to plan your show as rigorously as a space shuttle launch, but you should at least write out a set list to minimize onstage "down time."

Written documentation is extremely helpful when you're dealing with a house soundperson. These much-maligned professionals are bombarded with inexperienced and unprofessional acts, and they often adopt devil-may-care attitudes as a survival tactic. Most of the time, by necessity, they just drop mics down and hope for the best. However, you can usually improve their demeanor by giving them two helpful sheets of paper.

A stage plot (see **Fig. 1**) and a set list are invaluable tools if the soundperson has never seen you perform. The stage plot instantly informs the soundperson of the player's names, the instruments they play, and how they are arranged on stage. This guide is a tremendous help when something has to be tweaked in the middle of a set. For example, if a background vocal is buried in the mix, the soundperson simply checks the stage plot for the singer's location and moves the appropriate fader. A house soundperson will never make adjustments as quickly as someone who has rehearsed with the band, but the stage plot can certainly save an important harmony vocal from being lost for an entire song.

The set list is a professional courtesy because it gives the soundperson a definitive schedule of events. If a band starts to run over its allotted performance time, he or she can instantly see how many songs are left in the set and signal the act to cut one or two numbers. Although it's always a drag to shorten your set, a fair warning is preferrable to the soundperson's panic tactic of pulling the plug in the middle of a song.

TOMMY CAN YOU HEAR ME?
The stages of many clubs are architectural afterthoughts. For example, a bar owner who decides to offer live music might simply remove a few tables and cram a P.A. system near the restrooms.
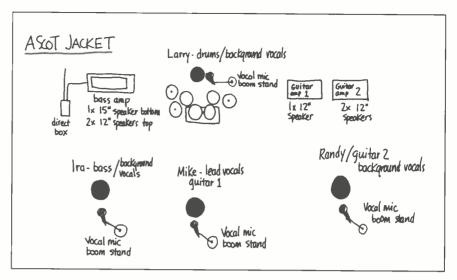
FIG. 1: Stage plots provide house soundpersons instant familiarity with your act's setup.

But even club owners savvy to the requirements of live music can be hampered by the physical limitations of their venues.

A good percentage of small clubs are housed in former restaurants, warehouses, and storefronts. These spaces were not designed to accommodate large stages, lighting rigs, and concert sound systems. As a club owner wrestles to optimize audience sight lines and bar access, the bandstand often loses out to economic and space considerations. In other words, don't expect the classic rectangle shape that allows a well-positioned back line of amps facing the performers.

The necessity of fitting amps, drums, keyboard rigs, and band members into bizarre geometric configurations can make it difficult for performers to hear each other. (I've played on stages shaped like diamonds, triangles, L's, and circles, and I have avoided support beams, low-flying chandeliers, and onstage refrigerators.) In tight situations, don't even try to set up a conventional back line. Position amps on each side of the drum kit and angle them toward the center of the stage. This configuration immerses the drummer in the roar of the instruments and throws sound across the stage to the other band members. Obviously, the sound is not projected directly at the audience, but in a small club the rumble should punch through just fine. If you can raise the amps up on chairs or road cases, do so. Getting the speakers off the floor improves sound dispersal.

Hearing vocals onstage is another problem, because club monitor systems are typically spartan and underpowered. Most of the time, you'll get two monitor wedges at the lip of the stage, but no monitor for the drummer (which is a big problem if your drummer sings). The only consistently workable way to fight a weak monitor system is to keep your instrument volumes down. If your stage volume is blasting, the house soundperson will be unable to get the monitors loud enough to cut through the stage din. Period. Remember, there are live microphones up there, so feedback can be a problem if the monitor levels are too high. It doesn't pay to be stubborn, because you can't win. If you want to hear yourself sing, turn down those amps! (And it's not a sin to ask your drummer to play softer, either.)
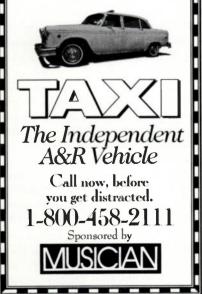
PEACE BY PIECE
Self-sufficiency is a good thing when you're dealing with the great unknown of club sound systems. Never assume that the club can accommodate your needs: When in doubt, bring it yourself. Here's a few helpful hints I've learned from watching unprepared musicians self-destruct at gigs. (And, yes, I've developed many of these habits through personal experience!)

Vocalists. To ensure consistent sound quality, bring your own mic to each gig. Many clubs use Shure SM58s—which are great if your voice sounds good through one—but I've played a few places where consumer-level microphones were the only option. However, even if you adore SM58s, club mics can sound a bit ragged because they are sometimes dropped and

abused by performers and inexperienced sound crews. Believe me, you'll be happier if you're singing through a known quantity.

The selection of mic stands in most clubs is pretty limited, so I also recommend lugging your own boom stands to the show. If you play guitar or bass, straight stands make it difficult to "eat" the mic without banging your instrument against the stand. Obviously, keyboardists and drummers require a boom to extend the mic past their equipment.

If you really want your vocals to shim-

strument, so I can readily grab a backup if a string breaks midsong. Extension cords and multi-outlet boxes (with internal fuses) are also essential accessories, because most stages never seem to have enough power outlets. Bass players should bring their own direct boxes.

Keyboardists. Frequent power fluctuations are an unfortunate fact of life in many clubs. If you want to protect your valuable gear, plug everything into a good AC line regulator to ensure stable AC. Bringing your own direct boxes can save frustration *and* fighting with

Vocalists can beat the often poor selection of signal processors available in small clubs by bringing their own gear to the gig. An inexpensive, easy-to-transport vocal "lifesaver" is dbx's Model 286 Microphone Processor. The 286 packs a mic preamp, phantom power, compression, de-essing, spectral enhancing, and noise gating/expansion into a single unit.

mer, load up a small road rack with a compressor and multi-effects unit. Preprogram the signal processors to your favorite settings—although, you may have to tweak them when you start working with the club's acoustics—and carry the rack to the mixing station. Make sure that you have all the necessary connecting cables and that each cable is labeled ("compressor input," "compressor output," etc.). The soundperson can simply patch your gear into the house mixing console to provide you with customized signal processing. Few sound crews will argue with having additional effects at their disposal, but it never hurts to call the club in advance of the show and let them know that you're bringing in an effects rack.

Guitarists/Bassists. For fretted performers, the operative word is "extra." If you don't bring extras of every single thing that can break, blow up, or get lost, you're inviting tragedy. Make sure that your gig bag includes an additional power cord and fuses for your amp; spare guitar cables of assorted lengths; a handful of fresh 9-Volt batteries for your stomp boxes and tuner; a backup guitar or bass; and a massive, almost inexhaustible supply of extra strings.

I also bring guitar stands for each in-

your bass player over who gets the single direct box provided by the club. A cheap spare sound module should be standing by for emergencies. Also, label every input and output cable (audio *and* MIDI) to facilitate panic repatches if a module fails. (For more complete information on live keyboard systems, see "Racking Your Brain" in the November 1994 **EM**.)

Drummers. Acoustic drummers should bring extra drum heads and a spare snare drum. If you break a snare, you can just reach for the spare and quickly exchange drums. Punctured tom heads are problematic, because you can't take the time to remove and replace the head during a performance. However, covering the hole with a strip of gaffer's tape can keep you going until the break or end of the show. Like keyboardists, electronic drummers should bring spare cables and plug everything into a power conditioner. If you can afford a spare sound module, it never hurts to have a backup.

STRESS FIGHTERS
Performance jitters can also be minimized if you try to keep your mind engaged throughout the often numbingly

boring process of setting up for the show. There's a lot of down time between loading and unloading gear, doing a sound check, and hitting the stage. Bring a book or some magazines to read during these numerous "hurry up and wait" periods. Also, you can diminish the nerve-wracking background noise of musicians chattering, bars being stocked, and other bands' sound checks by wearing earplugs or listening to a portable cassette player through headphones.

Getting food is sometimes an issue at clubs that are off the beaten path, or in a city's industrial area. Even if a store or restaurant is within walking distance, a late sound check may not give you enough time to eat *and* prepare for the show. (I still shudder at the times I've had just five minutes to tune my guitars, slap some water on my face, find my drummer, and jump onstage.) To avoid collapsing midset, always carry some crackers or another favorite snack to stave off hunger until after the show. It's also smart to pack a bottle of mineral water. Quenching thirst with booze or soft drinks can make you a bit woozy, especially on an empty stomach.

Unless you enjoy walking around drenched in sweat after a performance, a complete change of clothes, or at least an extra shirt or sweater, is advisable. Don't expect your "dressing room" (which is typically the club's bathroom) to have clean towels. Either bring your own, or plan to wipe your brow, neck, and hands with toilet paper. And speaking of hands, a pair of work gloves will make carrying your gear on and off stage a touch easier on those invaluable musician's fingers.

ENCORE!

To be completely honest, I must admit that I've played shows where careful planning *didn't* save the day. When Fate tosses out one of her deadly curve balls, there's often nothing you can do except close your eyes and swing. But even if the stage is falling down around you, one psychic lifeboat is always available: your sense of humor. If, in the middle of a climatic solo, your keyboard rack explodes like the gluttonous diner in *Monty Python's The Meaning of Life,* brush the shrapnel from your hair and scat sing the final bars. A shrug and a chuckle always wins more audience support than visible panic. ☕

Questions and Answers

The Service Lord meets the Phantom of the MIDI Line.

By Alan Gary Campbell

Q Is there any way to modify my strap-on master keyboard controller so I can phantom-power it via a 5-pin MIDI cable? I'm tired of wasting batteries.

A. We continue to receive requests for a MIDI phantom-power mod, so here it is. Strap-on synths and controllers, and remote devices such as MIDI footswitches, often require batteries or external, "wall wart" power supplies. The former are expensive, heavy, and environmentally unsound, and the latter are inconvenient and unreliable. Many devices can be modified to draw "phantom" power via the MIDI cable.
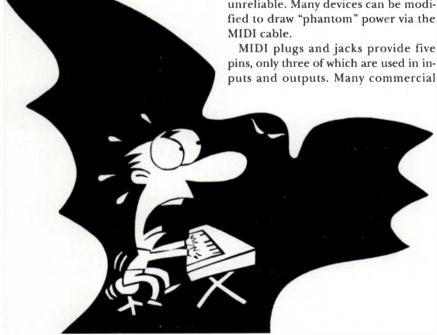
MIDI plugs and jacks provide five pins, only three of which are used in inputs and outputs. Many commercial MIDI cables, however, have all five pins connected. With the use of a do-it-yourself interface, these "extra" connections can carry power from an AC adapter to a modified controller or other device.

The interface is straightforward. A coaxial DC power jack is selected to match the plug of the appropriate AC adapter. (Alternatively, use one of the handy, Radio Shack panel-mount DC jacks, and if the adapter plug doesn't match, cut it off and add one that does.) The DC jack, a diode, and two standard, panel-mount DIN jacks are wired as shown in **Fig. 1.**

All the components fit in a small project box, wired point-to-point using small-diameter hookup wire. (The diode can "hang" from the DIN jack terminal.) A metal box is preferable if the interface will be placed near audio cables or components. The 1N4001 diode provides reverse-polarity protection for load currents up to one amp. Substitute a 1N5400 for currents up to three amps.

Getting the power from the cable to the device is less straightforward. The device must be disassembled and leads tack-soldered from the unused pins of its MIDI Out jack to its DC power nodes. This is not a job for the inexperienced do-it-yourselfer. If, after reading the instructions, this doesn't seem "old hat," refer the mod to a technician.

JACK DESROCHER

World Harmony Achieved!

Industry professionals praise Studio 5000 harmony processor and share their secrets

STUDIO 5000 FEATURES

Affordable Quality Pitch Shift

Our patented pitch recognition technology provides extremely accurate sample splicing so pitch-shifted instruments retain their natural character.

Instant Multitrack

Add up to 4 musically correct instrument harmonies to horns, guitars and more. It's like instant multitracking in a box!

The Creative Fast Track

- Data Wheel and 10 jump-to-function keys for faster editing
- Full MIDI Control
- MIDI Chord recognition

Feature Packed and Ready

- True stereo
- Balanced I/O
- 180+ combinations of harmony, reverb, delays, chorus and other effects
- Stereo sampler for flying in parts
- Convincing amp and speaker emulation
- EQ and compression
- Digital strobe tuner

The Studio 5000 takes its predecessor, the amazing DHP-55, a step further. With new features and programs crafted by studio pros, the Studio 5000 is sure to stir your creative urges. Here's what the people who make their living in the studio have to say about it.

Excellent Pitch Shifting

John Ross, President of LA's Digital Sound and Picture needs quality pitch shifting for Foley without the artifacts of competitor's products: "... the 5000 sounds natural, just like 2 inch tape at half speed. We require high quality products for simple applications. The Studio 5000 fits in very well indeed." Digital Sound and Picture is an advanced facility using a Lan-based digital audio network and multiple digital audio workstations.

"I immediately started doing things I never was able to do with my other processors"
KIM BULLARD

Sweeten Mixes, Add Depth

Five time Emmy award winning composer and producer Dominic Messinger says the Studio 5000 "makes the normal exotic". He uses it to sweeten mixes by using the regenerative pitch shifting Mix Imager programs. On drums he adds depth with detuning and dynamic filtering effects.

"It's like getting a whole new set of presets for your instrument"
DOMINIC MESSINGER, 5 TIME EMMY AWARD-WINNER, COMPOSER & PRODUCER

Creative Control

Clif Brigden, engineer for Thomas Dolby uses the Studio 5000 on Dolby's upcoming album and interactive video projects. "It sounds wicked right out of the box but lets you get to and change every parameter to make your own unique noises."

Leading Edge Special Effects

The Studio 5000 adds a futuristic edge to new animated series 'Phantom 2040'. Producer/composer Gerald O'Brien explains: "The 5000's Lush Chord Shifting algorithm is used on the main theme vocals

"It will be an indispensible product for musicians and engineers."
DHP-55 review
SOUND AND RECORDING MAGAZINE, JAPAN

while guitar and bass tracks get cabinet emulation and special effects. The robot voices are created using combinations of pitch shifting, flange and chorus." Gerald has scored over 22 shows including the successful Cobra series. His songwriting clients include Manhatten Transfer, Hall and Oates and Deborah Harry.

"Imagine what music will sound like in 40 years - that's what we're working to create."
GERALD O'BRIEN

Guitar Effects

"They asked for a Jimi Hendrix type cue to go with a 60's acid experience... I quickly called up the Voodoo Wah program... and they loved it!" Larry Brown, TV composer, producer and Emmy winner uses the Studio 5000 to closely emulate guitars with his synthesizers.

Fast

"I have to write a lot of music in a short period of time and the Studio 5000 really helps me to achieve that" says Kim Bullard, writer, producer and session musician. He's using the Studio 5000 to help create the soundtrack for the upcoming film, 'Taxandria.'

Record Quality

The Studio 5000 gives Kim Bullard the detailed sound he demands. "A soundtrack album is also part of the Taxandria project so the cues have to be record quality" he says.

"Sounds so deep, I'm still looking for the bottom."
CLIF BRIGDEN, ENGINEER FOR THOMAS DOLBY

DigiTech
HARMONY PROCESSORS

H A Harman International Company
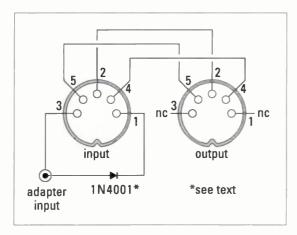
FIG. 1: Many devices can be modified to draw "phantom" power via the MIDI cable by supplying power from an AC adapter via the normally unused pins 1 and 3. A coaxial DC power jack, a diode, and two panel-mount DIN jacks are wired as shown.

Make sure to wear a properly grounded, antistatic wrist strap while working inside the device. Use a low-wattage, grounded-tip soldering iron to make the tack-solder connections. Take care not to overheat the joints and damage the jacks.

Generally, the unused pins of MIDI Out jacks "float," i.e., are unconnected. Occasionally, the pins are connected to one of the circuit grounds, and the respective traces must be cut. Use an X-Acto knife and make each cut with several light strokes to avoid cutting nearby traces.

Rarely, manufacturers use proprietary, PC-mount DIN jacks that, to save money, don't have pins 1 and 3! The pin spacing of such jacks is often non-standard, which makes substitution problematic. Drilling extra holes through the PC board, where traces below may be severed, is inadvisable. A panel-mount jack can supplant the PC-mount jack, but leads must be added between jacks.

The leads from the MIDI Out jack can be tack-soldered directly to the terminals of the DC-input jack or battery holder of the modified device. This greatly simplifies connection, and correct polarity can be discerned from the indicated polarity of the jack or the physical orientation of the battery or batteries. At the MIDI Out, pin 1 is positive, and pin 3 is negative, as indicated on the schematic. But this applies only to adapters with a positive tip and negative sleeve. Some are the reverse. *It cannot be overemphasized that correct polarity is critical.* If you wire the mod incorrectly, you may damage the modified device severely! Triple-check all wiring before you close up.

Note that the diode polarity should be reversed for use with negative-tip/positive-sleeve plugs. If the diode polarity is incorrect, no damage will result, but no current will flow.

To use the phantom-power interface, connect the MIDI Out of the modified device to the MIDI In of the interface, and connect the Out of the interface to the In of a switcher, keyboard, module, etc., as usual. Plug in the adapter, and you're ready to go. If the interface fails to operate, check for wiring errors, a reversed or bad diode, or a bad cable. Remember that the interface must connect to the modified device via a MIDI cable that connects all five pins (for example, Conquest or Hosa cables). Pin connection can be verified with the ohms function of a multimeter.

The photographs illustrate a typical mod on a Casio CZ-101 synth. Fig. 2 shows the tack-soldered connections at the MIDI Out jack. Fig. 3 shows the positive-lead connection at the positive battery terminal. (In this instrument, a reverse-polarity protection diode, not relevant to the present mod, has been added at the terminal.) Fig. 4 shows the negative-lead connection, tack-soldered to the shield-ground terminal, which in the CZ connects to the negative battery terminal and is easier to reach.
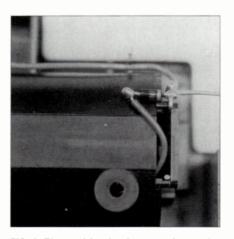
FIG. 3: The positive-lead connection at the positive battery terminal. In this instrument, a reverse-polarity protection diode, not relevant to the present mod, has been added at the terminal.

It is possible that some future MIDI spec may use pins 1 and 3, but the interface design precludes incompatibility, as both pins are unconnected at the MIDI Out. Furthermore, the modified device may be used without the interface simply by connecting a standard, 3-wire MIDI cable, which will effectively "disconnect" the mod.

Q. How do "phantom-power" MIDI devices, such as the Anatek boxes, work?

A. In a normal MIDI Out, pin 5 connects to the driver IC through a 220Ω, current-limiting resistor, and pin 4 connects to the 5 VDC supply, also through a 220Ω resistor. Phantom-power devices draw current from pin 4, through the current-limiting resistor, and back to the shield ground at pin 2. It's very "out-of-spec," but useful.

Assuming a MIDI loop current of 5.3 milliamperes, when the output driver is "low" (taking into account the series
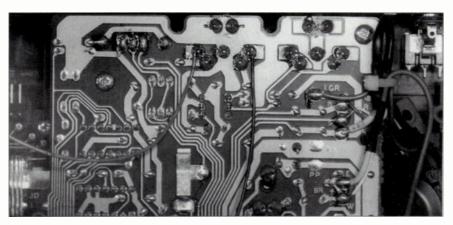
FIG. 2: The leads from the modified CZ-101's MIDI Out jack have been tack-soldered.
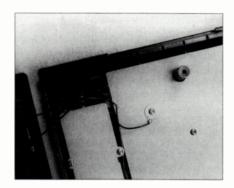
FIG. 4: The negative-lead connection has been tack-soldered to the shield-ground terminal, which connects to the CZ-101's negative battery terminal.
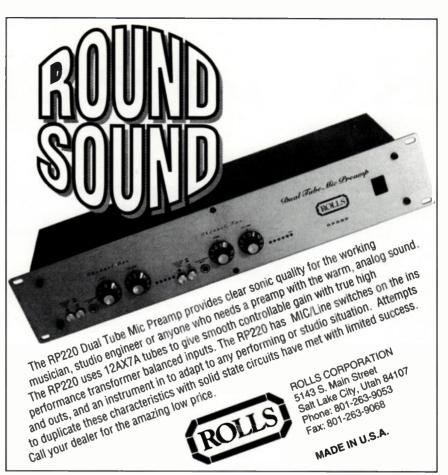
resistor and LED at the associated MIDI In) the voltage drop across the resistor is 1.2 volts, which leaves only 3.8 volts to power the device. The more current it draws, the less voltage is available. Therefore, phantom-power devices must use very low-power components.

To minimize power drain, some phantom-power devices "fudge" and provide lower-than-spec MIDI Out current. Most MIDI Ins are sensitive enough to accommodate this, but long cable runs should be avoided. Also, phantom-power devices cannot be cascaded.

Q. I replaced a contact strip on a Korg Concert 600 digital piano, but after reassembly, the top eight keys, C to G, were dead. I noted that a pin had broken off the keyboard-cable connector, but that pin is apparently not used, and running a jumper had no effect. From the schematic, if one of the scan lines was open, every eighth key or the top four keys should be out, shouldn't they? This symptom makes no sense.

A. Similar cable connectors and mainboards are used on the 76- and 88-key Concert pianos. For proper operation on some 76-key units, the CN8A connector must be deliberately plugged in with one pin offset. The seemingly "broken" pin is removed at the factory to allow this. Offset the associated plug by one pin position toward the rear of the instrument, so that the position of the missing pin remains uncovered and the rearmost cable lead connects to nothing.

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

Reviews

Kurzweil PC88 Controller

• • • • • • • • • • • • • • • •

By Steve Oppenheimer

One weighted controller plus one onboard sound source equals one good buy.

Hauling an extensive MIDI rig to a casual gig is like moving one kitchen chair in a semi-truck. It's a huge hassle for small results. Most of the time, you probably need at least a dozen good sounds: acoustic piano, electric piano, an organ or two, some strings, bass, and so on. If serious piano playing is a major part of the act, an 88-key, weighted controller is almost indispensable. However, you're not making enough dough

Kurzweil's PC88 combines a smooth, 88-key weighted keyboard; extensive controller functions; and 64 onboard sounds in a portable package. A optional expansion board adds a full Bank of GM sounds, supplemental sounds, and more effects.

to break your back for this gig. Wouldn't you like a nice, moderate-sized instrument that covers your main requirements?

Relief has arrived in the form of Kurzweil's PC88, which combines a weighted, 88-key MIDI master keyboard and 32-voice polyphonic, 16-part multitimbral sound module with effects. An optional expansion board adds a full palette of General MIDI sounds. Plug this baby's two ¼-inch audio outputs into an amp, attach a few pedals and footswitches, and you're ready to cover most of the basics. If you need a studio controller or more sophisticated stage setup, the PC88 provides an impressive, if not quite state-of-the-art, all-purpose master controller. And it weighs just 50 pounds.

Power is supplied by an oversized wall-wart. If you simply add a short, heavy-duty extension cord, the wall-wart becomes a lump-in-the-line supply. So quit griping and just deal with it.

USER INTERFACE

One of the PC88's best features is its friendly user interface. The sounds and parameters are well organized and accessed from clearly labeled front-panel buttons. Each set of parameters (e.g., Transpose, Controllers, MIDI Transmit, and so on) has its own button. There are no multilevel menus to wade through.

You step through the various parameters in each category using the left and right arrow buttons, which scroll at high speed when held. Values can be set with the -/+ keys, alphanumeric pad, and data-entry wheel.

A handy feature, which I used constantly, allows you to set values for any parameter (e.g., Key Range and controller values) by holding the Enter button and using any controller, including the MIDI keyboard, as a data-entry device. Kurzweil calls this feature *intuitive entry*.

Of course, the PC88 is easy to use because its programming features are mainly limited to control functions and a few effects parameters. But it's also easier to access these functions on the PC88 than it is on many master keyboards that don't even offer onboard sounds. If you want to play the internal sounds, but you need to create a

split, tweak the reverb, and adjust a few controller values, the PC88 is as close to punch-and-play as you'd want.

One critical part of the user interface is the keyboard action. After all, it's hard to play your best when your instrument doesn't feel right. The PC88 uses a new Fatar weighted action, which is firmer than earlier models (including the 76-key weighted Fatar action on the Ensoniq TS-12), but not overly tight. I have strong fingers, but I use a light touch and lots of dynamic variation, so the PC88 action suited me well.

The action feels especially good for piano parts, which I greatly appreciated while cooking up PC88 and Micro-Piano parts for singer/songwriter Joan-

na Cazden's new album (coproduced and engineered by **EM** Technical Editor Scott Wilkinson). Of course, when I played Cazden's acoustic grand piano, I was reminded that no MIDI keyboard can match the real thing.

INTERNAL VOICES

There are many 88-key, weighted master keyboards around, some of which are easy to use. What makes the PC88 different are the onboard sounds. A stock unit offers 64 ROM-based, preset sounds, called *Internal Voices*, and 128 user RAM locations, called *MIDI Setups* (discussed shortly). Additional sounds for the PC88 are available with the addition of the VGM expansion

board (see sidebar, "VGM Voice/GM Expansion Board"). Although you can assign layers, splits, and controllers and edit the effects somewhat, the synth Voices themselves can't be edited.

The PC88's display can indicate the program numbers as 0 to 127 or 1 to 128 to match different program numbering schemes. You can also change MIDI Setups or Internal Voices via MIDI Bank Select and Program Change.

The Bank of Internal Voices is divided into four Groups of sixteen sounds. To call up an Internal Voice within the current Group, you simply hit one of sixteen front-panel Sound Select buttons and start playing. The buttons are clearly labeled with the names of the basic Internal Voice programs: Classical Piano, Strings, Acoustic Guitar, etc.

A pair of buttons steps between related Groups. For example, the Jazz Organ button in Group 1 calls up Jazz Organ 1, while in Group 2, the same button gives you Jazz Organ 2. In Group 3, it calls up Ballad Organ 2, and in Group 4, you get Chiffy Pipes. Most Internal Voices are similarly related across Groups, although Kurzweil throws you a slight curveball once in awhile. For instance, Stereo Strings is in with the Synth Pads, an understandable exception.

MIDI SETUPS

In Internal Voices mode, everything is set in read-only silicon. MIDI Setup mode lets you create custom setups in RAM, using the Internal Voices and external MIDI instruments. Half of the 128 MIDI Setup locations contain factory programs, and the remainder are empty.

The PC88 offers four independent Zones, each of which can be used for layers or splits. Each Zone can address the Internal Voices, an external MIDI device, or both. You can mute or solo any Zone, which is great when you're working with splits and multilayer sounds.

Each Zone can be set to its own MIDI channel, and it can transmit Bank Select as well as Program Change messages. There's an undocumented "gotcha" here, though. If you set two Zones for the same MIDI channel (e.g., if you want one Zone to control an Internal Voice and the other Zone to control an external sound module on the same channel), they must send the same Program Change. The tricky part

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VGM VOICE/GM EXPANSION BOARD

Although the stock PC88 is a very good value, the product is greatly improved by the addition of the VGM Voice/General MIDI Expansion board. This $495 option provides 32 notes of additional polyphony, a second effects processor, and two supplemental sound Banks containing approximately 6 MB of sounds. In 64-voice mode, notes are assigned alternately to the Internal Voices and the identical set of VGM Bank 3 Voices, combining their polyphony.

The factory-installed VGM board integrates seamlessly into the PC88. Its sounds work pretty much the same way as the stock sounds, except that not all VGM sounds respond to Key Bend. The effects processor is identical to the stock PC88 effects processor, and the two operate in parallel. (The extra processor can only be applied to VGM sounds.)

The General MIDI Bank isn't perfect, but it's one of the better GM sets I've heard. The basses are particularly excellent, and most of the ethnic and orchestral percussion is well done. The clean guitars and chromatic percussion (e.g., marimba, vibes, celesta) mostly range from passable to good.

On the down side, the Overdrive Guitar and Distorted Guitar are a waste of ROM, and the banjo and bagpipe are disgraceful. The brass and woodwinds are mostly disappointing; they're better than those on

some GM synths, but that's faint praise. Most of the special effects get the job done. Don't even think about using the Gun Shot, though; you can get more realistic firearms sounds by clapping your hands.

Because the sounds are voiced for General MIDI, their volume levels top out at 100, a convention used to varying degrees throughout the GM world to ensure consistent General MIDI file playback.

In addition to the regular GM sounds, the VGM board includes 67 Expansion voices. This Bank includes several horn and string sounds that are definitely superior to those in the GM set. The guitars also get the job done; in fact, if you put the Strat through some tasty outside processing, you'll be downright pleased with the results.

Finally, the Expansion Bank includes ten drum kits, seven of which use the GM map. The drums are probably the most consistently top-notch section in the VGM board. You get a standard kit, a kit with room ambience, a power kit, brushes, orchestral, synth drums, and more. They're all extremely good.

It's true that $495 isn't chicken feed, but you get your money's worth, especially if you want the PC88 to be a one-stop performance machine. If you can't spend the extra cash right away, start saving up to add the VGM board later.

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is that you *appear* to have the option of setting up independent Program Changes on the same channel, but when you go to save the Setup, it won't save the second Zone's Program Change setting. This is by no means a disaster, but the more I used the PC88, the less I liked this design decision.

Some Zone parameters, such as Pitch Bend Range (in semitones and cents) and Transpose (±127 semitones), are pretty straightforward. But Kurzweil has implemented some nifty extras. For instance, you can set inverse Key Ranges by setting the Zone's high limit below the low limit. In this case, the sound is only active above and below the limits, not in between.

A Zone's Velocity response can be customized in several ways. You can set minimum and maximum Velocities and scale Velocity by ±300%, with 100% producing a 1:1 response. The Offset parameter adds or subtracts a constant value to the Velocity. To top it off, you can apply any of three Velocity curves: linear, positive sine (the first ¼-cycle of a sine wave), or positive cosine (see **Fig. 1**). Many master keyboards provide

far more curves, so this could be a limitation, but I suspect few people use the extra curves anyway.

IN CONTROL

One of the most promising aspects of the PC88 is its many fully programmable controllers. Wheel 1 is comfortably spring-loaded and can have separate up and down functions. Wheel 2 is a modulation-type, unidirectional controller. Four sliders and three buttons are designated with letters A through G. Rear-panel jacks accept four continuous pedals and two switch pedals. And of course, the keyboard senses Channel Pressure (aftertouch).

Each continuous controller can send any MIDI Control Change message or a variety of special functions (see the table, "Continuous Controllers and Switch Functions"). An extremely useful controller parameter sets entry and exit values that take effect whenever you change Setups.

I found exit values useful for making sure any changes made in real time (e.g., changing the reverb time) were undone when I changed the Setup.

The obvious application for entry values is calling up a patch with some controllers already active—perhaps a pad with a bit of LFO tremolo already happening, controlled from Wheel 2.

This feature also solves a common problem found in synths such as Roland's JD-800: As soon as you change patches, the sliders instantly jump to their new values. In the PC88, a slider doesn't become active until you move it past the entry value. This allows many kinds of smooth transitions. For instance, you can create custom-balanced multisounds with smooth volume crossfades by assigning one fader to two Zones, programmed with opposite entry values (i.e., one positive and the other negative). When you call up this MIDI Setup, the levels remain at their respective entry values until you want them to change.

As with Velocity, you can scale each controller, program an offset, and apply a curve. A shortcut jumps to the same parameter for each controller in succession (e.g., from the Offset for Slider A to the Offsets for Sliders B, C, and D, then Pedal 1's Offset, and so

● **PC88**

on). There's no shortcut to jump backward, but the scrolling is plenty fast. This is just another example of the PC88's musician-friendly user interface.

The Pitch Bend has a cool twist that Kurzweil has implemented in all their keyboards since the K250. When used with the internal sounds, Pitch Bend only affects the keys you manually hold down, regardless of what you sustain with a switch pedal. So if you sustain a chord with a pedal, but only keep your finger on one key, only the notes triggered by that key are affected by the Pitch Bend messages. This feature, which Kurzweil calls "Key Bend" (key-based Pitch Bend), is great for pedal steel-type bending, especially if you work on holding just the right notes to bend smoothly between chords.

The three programmable control buttons have separate on and off functions, with the "on" position indicated by an LED. Each button can operate as a momentary or toggle switch. As with the other controllers, all button functions are independently programmable for each Zone. You can even program a button to send a specific Program Change, or to increment/decrement programs. If you make the button a toggle and send different Program Changes for on and off, each button can switch between two programs per Zone. This is nicely demonstrated in the "E Piano Trio" and "A Piano Trio" MIDI Setups.

For the most part, the controllers are quite flexible and well implemented, albeit not as powerful as those on the Peavey C8. But I was greatly disap-

Continuous Controller and Switch Functions

Channel Pressure
Control Change 0-127
Go to a specific MIDI Setup Go to a specific Program
Pitch Bend Up (64-128) or Down (63-0)
Program Increment or Decrement
Sequence Stop, Start, or Continue
Setup Increment or Decrement
Tempo (for the arpeggiator or an external device)
Transpose Up/Down
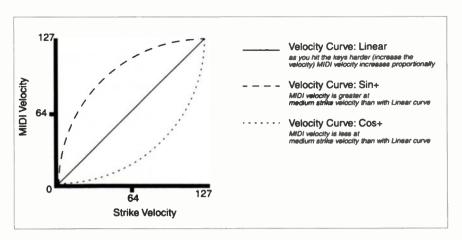
FIG. 1: The PC88 offers three Velocity curves. The default setting for all internal sounds is "Linear." (Courtesy of Kurzweil/Young Chang America.)

pointed to discover that when the programmable sliders are set for Control Change 7 (Volume), they cannot fade an Internal Voice all the way off. As long as the master volume fader is up, you always hear each Zoned internal sound at least a little bit. The sliders work fine on external synths, though. You can still fade the internal sounds from a continuous pedal, but I hope Kurzweil changes the slider implementation in a future software upgrade.

My review unit was one of the first in the initial production run, and I experienced some flakiness. Wheel 2 often sent data after being pulled all the way back, and I managed to crash the PC88 several times when making unremarkable edits. Sometimes, Program Changes were sent to external synths from a Zone assigned to control internal sounds only (set to Local in the MIDI Transmit menu).

These problems did not appear in the second instrument I examined. However, this unit occasionally flipped out when turned on. The sound was extremely distorted, the volume was very low, and some keys didn't work. Upon rebooting, everything was fine. I would be concerned about this except that I had no problems with the PC88 I used for the Cazden album, and I've seen several problem-free PC88 demos. I think I just had reviewer's luck.

I had no problem with hung notes when programming, but in case such a misfortune were to occur, the front-panel Panic button sends Reset All Controllers, All Notes Off, and Note Off 0 to 127 messages on all channels. The Panic button doesn't mute the effects, so reverb tails don't get cut off.

UP THE DOWN STAIRCASE
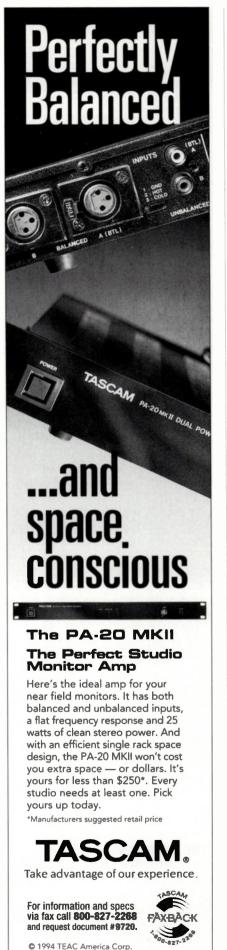
MIDI arpeggiators were fashionable years ago, but you only occasionally see them now. To be honest, I have never used arpeggiators in my work. I'm not against them; they just don't suit my musical style. But I had fun goofing around with the PC88's version. Kurzweil hasn't merely revived the global upward, downward, and up-plus-down arpeggiators of old. The PC88 arpeggiator includes everything but the kitchen sink, and if you have a MIDI-controlled sink, it will play that, too.

An arpeggiator is available for each Zone and can capture (latch) and arpeggiate any notes you play within the Zone. There are lots of variations. In one mode, the device latches the keys you are holding when you switch the arpeggiator on and keeps playing just those notes after you release the keys. In another mode, you can add notes to the arpeggio even after hitting the "on" switch. The machine can keep playing arpeggios after the feature is turned off, or stop playing when turned off, as you choose. It can play notes in the order entered, in ascending or descending order, play them at random, and more.

In fact, there's a lot more to the arpeggiator than I have space to detail. Without a doubt it's the coolest arpeggiator I can remember. Just figure that what you want is what you get.

GLOBAL ISSUES

In addition to the usual global controls (Local On/Off, Clock Transmit On/Off, etc.), the PC88 offers a few extras. There are seven levels of keyboard touch sensitivity. You can instruct the PC88 to ignore All Notes Off messages,

PC88 Effects

Algorithms*	Reverb, Chorus, Chorus + Reverb, Delay, Delay + Reverb
Reverb Types	Room, Stage, Hall
Reverb Variations	Bright, Normal, Warm
Other Effects	Choruses 1 and 2; 1-tap Delay
Parameters	Effects (chorus) wet/dry mix, reverb wet/dry mix, reverb time, reverb depth, delay time (up to 350 ms)

* = Selectable via MIDI Control Change 83, values 66 to 112.

and you can send a General MIDI On message to external synths and the optional VGM board. You can also check the available memory, a handy function I wish every instrument and controller offered. MIDI Setups can be saved via SysEx, which is especially important because the PC88 doesn't include a disk drive or card slot.

A version of Kurzweil's well-known MIDIScope data viewer is included, too. This feature lets you see incoming or outgoing MIDI data in good old decimal numbers, which is useful for troubleshooting. (For instance, MIDIScope confirmed Wheel 2 was flaking on me.) Unfortunately, given the PC88's small display, you can only see one message at a time, and you can't scroll back to see the previous data that blew by.

You can tell that musicians were involved in the PC88 design process. For instance, in Keys Up mode, a newly selected Setup won't become active until you release all the notes played by the previous Setup, which helps produce slick onstage sound switching. MIDI In Remap mode splits incoming MIDI notes and their associated controllers across the four Zones—notes are sent out all Zones that are programmed for the appropriate Key Ranges—regardless of their original MIDI channels. Virtually everything can be transmitted over MIDI in real time, with buttons and effects settings going out as System Exclusive.

SIDE EFFECTS

The PC88's effects are few, but usable (see the table, "PC88 Effects"). You get Room, Stage, and Hall reverbs, with Bright, Normal, and Warm variations. You can set the wet/dry mix and reverb time. There are two choruses, with wet/dry mix, and a single-tap delay with

a delay-time parameter. You can combine reverb with either chorus or delay.

The effects sound clean enough, but they're nothing to write home about. There isn't a lot of difference between the choruses, and the reverb reflections are simple. The delay brings very little to the party; I love delay processors, but I didn't have much use for this one.

Essentially, what you get is the minimal GM effects collection. A flanger would have been nice, but the manufacturer had to keep costs down somewhere, and this was a good place to do it. You can always add more effects in the studio or for big gigs, and the PC88 effects are adequate for simple casuals.

Basic effects combinations for Internal Voices are automatically remembered, and you can save effects settings with MIDI Setups. You can change the effects with MIDI messages, from the front panel, as part of a Setup change, and as part of an Internal Voice change. Four Global modes allow various combinations of these options. For example, in Panel mode, effects can be changed from the front panel and MIDI, but not when Setups or Voices change. The sixteen panel buttons can remember any Bank, Program, and Effects change you set, which is especially handy onstage. This is yet another example of the PC88's flexibility.

SOUND QUALITY

Let's cut to the chase. The PC88 must stand or fall on the combination of solid control and the strength of its sounds. The control is there; fortunately, the sounds provide a solid foundation, too.

The acoustic pianos are mostly drawn from the MicroPiano module (reviewed in the June 1994 **EM**). The Classical and

Stage pianos are darn good for an instrument with limited ROM (as opposed to a RAM-based sampler), and they respond well to Velocity changes. There are few differences between them, though. The Suitcase EPiano is my favorite of the electric pianos; it doesn't sound like you're about to break a tine when you smack it, but it gives the general impression of a Fender Rhodes nonetheless. The rest of the electrics sound good, too, if distinctly digital. If you want nasty, you won't find it here.

The electric grands sound very close to the acoustics, but a bit brighter. They don't resemble Yamaha or Kawai electric grands. The Digital EGrand is the only distinctive member of this group, with its FM-like harmonics.

The strings are a strong suit. Sure, you can tell they're coming from a synth, but they're lush, and the Touch Strings and AttVel Strings offer excellent Velocity response. As mentioned earlier, the Stereo Strings are located in the Synth Pads category. I guess Kurzweil felt a fifth string patch was more important than a fourth pad.

It's almost impossible to find an acoustic-sounding, synthesized harpsichord, and the PC88 doesn't break with that tradition. On the other hand, it uses Release Velocity to trigger the characteristic plectrum-damping "click" of a harpsichord's release. (You can disable this.) In addition, experimenting with the keyboard's touch response and Velocity settings can help produce a convincing performance. The Clav is

Product Summary

PRODUCT:
PC88 Performance Controller
PRICE:
PC88 $2,295
VGM board $495
MANUFACTURER:
Kurzweil Music Systems
13336 Alondra Blvd.
Cerritos, CA 90701-2205
tel. (800) 421-9846
or (310) 926-3200
fax (310) 404-0748

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

of similar or slightly better quality than the harpsichord. It's good enough to get the idea across, complete with a release "thump." As with the harpsichord, you can toggle the release sound with a button.

The acoustic guitar is good, but its volume is too low. I still found myself going back to my K2000's wonderful Steel String Guitar preset instead. The 12-string sounds like a doubled and transposed version of the acoustic guitar, rather than a real 12-string. Fluid Guitar makes good use of chorusing. Nowhere is there a convincing electric lead-guitar sound, though.

Vibes are among the easiest sounds to produce with a synth, and the job is well done here. What really caught my ear was the regular and metal marimbas. I love Caribbean music, and I had a field day with these. (Okay, sometimes I cheated and substituted marimba for steel drum on a few calypso tunes, but it sounded great.)

The rock and jazz organs are quite nice and include a fair rotary-speaker effect that accelerates with Modulation. It won't stack up against a Hammond with a good Leslie emulator, but it's better than you'll find in many synths. The Full Pipes patch is just the thing for church.

Aside from the pianos, bass is the place for the PC88. To begin with, you get a small assortment of acoustic and electric basses, including legato and sustain variations. Several basses are layered with ride cymbal for jazzy walking parts, and you can pull the ride level back with a slider. It gets even better when you add the superior basses in the VGM board. But aside from the actual bass sounds, the entire instrument puts out lots of clean, strong bass response.

The 64 factory MIDI Setups build on the Internal Voices to create some very nice layered sounds. Aside from their practical uses, the unit's factory Setups provide a nice tutorial in getting the most out of the instrument, including innovative uses for the arpeggiator and buttons.

The main downside is that Modulation and Pressure are not used often or well in Internal Voices, and you can't do anything about it. You can adjust the output of the controllers, but you can't edit their effect on a given sound. Although this limitation detracts from the expressive possibilities

● **PC88**

of the Internal Voices, it doesn't offset their overall quality.

PORTABLE BAND

Whether you're starting a home studio, or equipping yourself for live gigs, the PC88 is an excellent first piece. You get a very nice, weighted action and a fairly complete group of controllers (unless you want something exotic like a ribbon, touchpad, or joystick). The stock sounds are enough to lay down the kinds of sounds most bands expect of a keyboardist.

If you add the VGM board—and you really should—you can cover all the basic food groups. This makes the PC88 a one-ax-does-all gig machine, although it doesn't cover all the bases equally well.

The documentation is excellent, with few oversights and plenty of detail. The best part is, despite the PC88's depth, if you know your way around MIDI controllers, you will only occasionally need the manual. Kurzweil deserves lots of praise for designing this instrument with both the average musician and the serious MIDI user in mind.

If you need the ultimate MIDI master controller, this isn't quite it. It has only one MIDI In and one Out, it doesn't store external SysEx, and it has only three controller curves. I could go on about what it doesn't do, but you get the point.

I don't think these limitations will matter to most people, anyway. If your studio features a high-quality software sequencer and a MIDI patch bay, you probably don't need any more MIDI control than the PC88 supplies. Gigging musicians are well advised to keep things simple anyway. Besides, none of the master controllers that have more MIDI power include onboard sounds. Even if you still have to haul another sound module and effects processor or two to fill things out, the PC88 starts you out ahead of the game.

The combination of good sounds, better-than-average control, and a quality action make the PC88 pretty hard to beat, especially if you play piano parts. Say, would you like to buy my old 88-key weighted controller?

*As **EM**'s resident product analyst, Senior Editor **Steve O.** checks out lots of great music gear he can't afford to keep.*

Circle #437 on Reader Service Card

Sound Quest MIDI Quest 4.0 for Windows

By Allan Metts

A potent, but ponderous, universal patch editor/librarian.

I have a nasty habit of collecting things. I collect hundreds of sounds for my synthesizers and spend more time tweaking and organizing them than I'll ever spend using them. And when I do need that perfect sound, I often settle for a factory preset to avoid a lengthy search. Sound familiar? If so, Sound Quest's *MIDI Quest* can help.

MIDI Quest is a comprehensive suite of synth-programming and patch-organizing tools rolled into one *Windows* application. You can extract patch data from your MIDI devices and use the software's powerful editing and auditioning tools to get your sounds just the way you want them. Data can be organized in several different ways and saved to a Standard MIDI File or *MIDI Quest, MIDIEX,* or *Cakewalk* PATCHES.INI file. Just a few clicks of the mouse reconfigures your entire MIDI setup.

DRIVE ON!

MIDI Quest's centerpiece is its Driver List (see **Fig. 1**). A MIDI channel-specific driver is provided for every type of data in every supported synth in your setup. A typical synth has a patch driv-

FIG. 1: The Driver List is the center of all activity in *MIDI Quest*. The program builds a Driver List based on the instruments you identify during installation.

er, a Patch Bank driver for the list of patches, a Multi/Combi driver, a Multi/Combi Bank driver, and drivers for miscellaneous settings such as effects, microtuning tables, and system parameters.

MIDI Quest builds a Driver List based on the instruments you identify during installation. If you get another synthesizer, simply rerun the installation routine to add the appropriate drivers. You can remove drivers from the list without erasing the associated files, and you can maintain multiple Driver Lists. If you never edit the microtuning tables in your synth, remove them from the Driver List and forget about them. I made a Driver List with only Patch Banks for easy sound auditioning.

At first, I didn't like *MIDI Quest*'s Driver List. I wanted a pictorial view of my setup, where I could click on each instrument to see the data inside. But I soon realized that the "modularized" driver concept is quite flexible. I especially like the ability to incorporate a patch-bay Program Change with every driver. Once I set up the patch bay, my JLCooper MSB+ obediently pointed to whatever synth I was editing at that moment.

Once your drivers are set up (which can be a slow process), you're ready to start moving MIDI bytes. Click on one of your instrument's Patch Bank drivers, click on the edit button, and up pops a window containing a list of the instrument's patches. Now pick a patch, hit the Bank window's Edit button, and you get a graphic editing window containing all of that patch's parameters.

IN THE BANK

Over the years, I have managed to collect over 2,000 sounds for my Yamaha TX81Z synths. I have been editing and moving them around using Voyetra's *Sideman 81Z* editor, which stores the entire memory contents of the TX81Z in one file. Because *Sideman*'s file structure is so rigid, I ended up
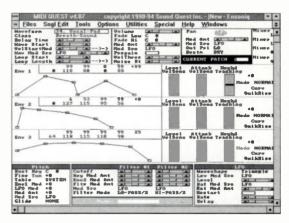
FIG. 2: *MIDI Quest*'s patch editors are complete, accurate, and easy to use, and they make heavy use of color. Note the use of graphs for envelope generators. Many parameters turn red when changed and go back to black when returned to their original value.

with quite a bit of redundant or unused sound data.

Organizing these sounds was a good test for *MIDI Quest,* so I started the process of converting the files. The program comes with a file-conversion utility that, with a little digging into the bits and bytes of your files, lets you write routines to move your old data into *MIDI Quest* files. After taking a look at my old files with the program's built-in hexadecimal viewer, I was able to write a short macro that extracted the *Sideman* patch data and saved it as *MIDI Quest* files.

In the Bank Editors, you can cut, copy, and paste your patches between Banks. You can also edit, sort, audition, and rename them, or move the patches to other parts of the program. Anything that can be done with Patch Banks can also be done with Multi/Combi Banks, as well.

The Bank Editors can also create new Banks of sounds automatically. Want to mate a Steinway with a buzz saw? Simply select some patches, press a button, and *MIDI Quest* creates an entire Bank of sounds based on random combinations of the patches you selected.

To *MIDI Quest,* anything that isn't a Bank is a Single. This includes individual patches, combis, system parameters, effects, and microtuning tables. The Single Editors are complete, accurate, and easy to use, and they make heavy use of color (see **Fig. 2**). Envelopes are displayed as graphs, and a generous supply of sliders and knobs control other parameters. Where graphic devices aren't used, parameters

are edited by typing in a number, sliding the mouse, or tapping the increment/decrement keys. Many parameters turn red when changed and go back to black when returned to the original value. That's a nice touch.

Cut, copy, and paste operations are available in the Single Editors, as well. This feature lets you move data between editing windows and to other parts of the same patch. Any number of parameters can be randomized within a narrow or wide range, and the results can be saved to one of four temporary buffers. If you don't like your work, you can restore the original patch.

Sounds can be auditioned in many different ways. In some windows, the right mouse button sends a note to your synth. The pitch is determined by the position of the mouse on the screen. You can also set up a chord or short sequence of notes and play it with a menu selection. My favorite auditioning tool, however, is the built-in support for Standard MIDI Files. I could load my latest work-in-progress and edit the sounds while the song played. Very cool!

MIDI Quest doesn't keep track of which sounds are in the instrument. If you open a Bank from disk and receive a patch map from a synth, the patch map displays the patch names from the disk file, even though you never sent those sounds to the instrument. *MIDI Quest* also doesn't care if you send a Bank of sounds to an instrument when memory protection is activated.

IN THE LIBRARY

After playing around with my TX81Z Banks, it was time to get them organized with *MIDI Quest*'s Library. To be honest, I resented having to build a Library with my newly converted Patch Banks. The Bank files were sitting on my hard disk, and I wanted *MIDI Quest* to tell me which Banks had what patches and find all the duplicates for me. Instead, I had to open each file with the Bank Editor, copy the Banks to the Library using the Move To button, and answer an "OK or Cancel" dialog box every time *MIDI Quest* noticed a duplicate patch name. Because many of my files had factory presets and patches named "INIT," I saw that dialog box several hundred times.

Once the Library is built, though, you have powerful searching capabili-

Instruments Supported

Sound Quest supplies drivers for most popular MIDI devices. Instruments with *MIDI Quest* editing support appear in bold. Those in lightface text can only send and receive bulk memory dumps.

Alesis	**D-4**, HR-16, SR-16
A.R.T.	DR1
BOSS	SE-50
Casio	**CZ-101, VZ-1, VZ-10**
Digital Music Corp.	**MX-8**
DigiTech	DSP128, DSP256
E-mu	**MPS, Procussion, Proteus/1, Proteus/2, Proteus/3, SoundEngine, Vintage Keys**
Ensoniq	EPS, **ESQ, ESQ-1, ESQ-M, KS-32**, Mirage, **SD-1, SQ-1, SQ-2, SQ-80, SQ-R, TS-10/12, VFX**
Eventide	H3000 Ultra Harmonizer
JLCooper	**FaderMaster, MSB-1620, MSB+**, PPS-100
Kawai	**GMega**, K1, **K3, K4, K5, K11, R-50, R-100, Spectra, XD-5**
Korg	**01/W, 03R/W, 05R/W**, 707, DDD-5, DS-8, **DVP-1, DW6000, DW8000**, EX800, **EX8000, i2, i3, M1, M1R, M1ex, M3R**, POLY800, S3, SDD-3300, **T1-2-3, WaveStation, X-3**
Lexicon	LXP-1, **LXP-5**, LXP-15
Oberheim	**Matrix-1000, Matrix-6/6R**
Peavey	**DPM 3/V3**
Rhodes	Model 660, Model 760
Roland	A-50, A-80, **CM-32L, CM-32P, CM-64**, D-5, **D-10, D-20, D-50, D-70, D-110, E-20, E-660, GP-16, GR-50, GS (all), JD-800, JD-990, JUNO-106, JV-80/880, JV-90, JV-1000, JX-8P, MKS-80/Super Jupiter, MT-32**, P-330, Pro-E, **R-5, R-8, R-8II, R-8m, R-70, RA-50, SC-55, SCC-1, U-110, U-20, U-220**
Sequential	Drumtraks, MultiTrack, Prophet 10, **Prophet 5**, Prophet 600, Prophet T8, Six-Trak, Tom
Voce	DMI-64
Waldorf	**MicroWave**
Yamaha	DMP7, DX, **DX5**,, **DX7, DXII, DX7IID, DX7IIFD, DX11, DX21, DX27, DX100, FB01**, RX11, RX17, RX21L, RX7, **SPX90, SY22, SY35, SY55, SY77, SY85, TF01, TG33, TG55, TG77, TG100, TG500, TX7, TX802, TX812, TX816, TX81Z, V50**

ties at your fingertips (see **Fig. 3**). *MIDI Quest* can find duplicate sounds, even if the names are different, and it can search for similar sounds with a certain percentage of identical bytes. It can find sounds with a particular value in a particular byte and search by name or keyword. Unfortunately, the method for entering the keywords themselves is cumbersome for large libraries.

You can open patch-editing windows, send individual sounds to your synths, and build Bank files from within the Library. Unfortunately, the Library contains a separate copy of what was

in the Patch Banks and doesn't stay up to date once it is built. Any changes made in a Bank must also be made in the Library.

MIDI Quest's Database feature is very flexible, allowing you to combine many different types of data elements. Simply click on several drivers in the Driver List, hit the Database button, and *MIDI Quest* collects the corresponding data from your synths. What you put in these databases is entirely up to you. You can send out a few new program maps, or reconfigure your entire setup in one pass. I made a different database

for each song I was working on.

The MIDI Monitor, MIDI Controller, and Driver Creator windows round out *MIDI Quest*'s bag of goodies. The MIDI Monitor lets you see data arriving at your computer, and the MIDI Controller lets you send out several of the more common types of MIDI messages.

The Driver Creator lets you create your own Bank Editors for unsupported instruments. But to do this, you must be willing to dive deep into your instrument's MIDI implementation and use *MIDI Quest*'s technical macro language. If you want to create your own Single Editors, you must buy Sound Quest's $100 *Tech Quest* package.

THE USER INTERFACE

MIDI Quest is *not* a well-structured, wonderfully organized *Windows* program. For example, most of the drop-down menus are only one level deep. But some of them have as many as 22 entries (see **Fig. 4**). Although several well-known *Windows* business programs (such as Microsoft *Works*) use a similar approach, I would prefer to see nested submenus in a program with this many menu items. In addition, many menu items are not where you would expect to find them.

Examples of programmer oversight

Product Summary
PRODUCT:
MIDI Quest for *Windows*
4.07
PRICE:
$319
SYSTEM REQUIREMENTS:
80286 or better PC-compatible with 2 MB RAM (4 MB recommended); Microsoft *Windows* 3.1; *Windows* compatible MIDI interface; mouse
MANUFACTURER:
Sound Quest, Inc.
131 W. 13th Ave., Suite 2
Vancouver, BC V5Y 1V8
Canada
tel. (604) 874-9499
fax (604) 874-8971

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

● **MIDI QUEST**

FIG. 3: The Library's search options are quite powerful. The program can find duplicate sounds, even if the names are different.

are rather easy to find. Some windows lack scroll bars where they are needed. For example, the Driver Creator window is bigger than a 640 × 480 monitor screen, yet it lacks a scroll bar. In addition, certain fields don't allow direct entry; some buttons don't turn gray when their use is inappropriate; and in places, the cursor keys don't let you move around the window. Some icon buttons don't match their corresponding menu entries, and the Settings button performs the same function as the Update selection in the menus. Comments in one part of the program are the same as Notes in another.

When you move between windows, only one menu changes to account for window-specific functions. Sometimes this is fine, but on occasion, related activities are scattered throughout the menus. For example, let's say you want to open a Bank of patches from a file on disk and then open a Bank from your synth. To open a Bank from disk, you use a Tools menu selection, pick the file you want, and end up with your Bank open in the Bank Edit window. (The Tools menu is global and always available.)

But to import a Bank from your synth, you use the Drivers menu, which is only available when the Drivers window is active. Once the program has fetched the patches from the synth, you also end up with a Bank open in the Bank Edit window. Why couldn't both functions have been provided in the same menu? I found this to

be counterintuitive.

Other aspects of the program seem to be more for the programmer's convenience rather than the user's. If you tell the installation routine you have a Yamaha TX81Z, *MIDI Quest* gives you 38 drivers for the TX81Z, DX21, DX11, DX27, and V50 synths. I have four TX81Zs, so I had to delete 116 of the 152 drivers *MIDI Quest* installed. Sound Quest told me they bundled their drivers to allow *MIDI Quest* and their single-instrument editors to use the same files.

Users are responsible for coming up with their own naming conventions. Any type of file (Driver List, Library, Patch Bank, etc.) can be saved to any directory using any file name and any extension. I need more structure than this. This problem is particularly apparent when adding a driver. The user must wade through a directory full of 11-character file names in the hope of identifying the desired file.

MIDI Quest's documentation is only marginally helpful. The manual describes the program in adequate detail, but it is written for implementations of *MIDI Quest* on five different computer platforms. Because all computer platforms have their differences, the manual can be hard to read. Online help for the program includes only ten short text windows. (*MIDI Quest* does not use *Windows*' help system.) I like *MIDI Quest*'s Fast Tips, though. These online blurbs describe the idiosyncrasies of each instrument *MIDI Quest* supports.
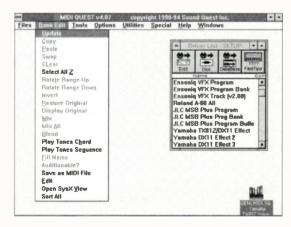
FIG. 4: *MIDI Quest*'s menus can be long and disorganized. This Bank Edit menu has 22 items.

Customer support is available by phone from 11 a.m. to 4 p.m. PST on weekdays. Sound Quest also maintains a bulletin board with updated drivers and utilities. The support staff was quite helpful each time I called and was even able to fix some problems while I was on the phone with them.

Most of the bugs I found are minor, although the one that kept changing

▼

**Once
a Library is built,
you have
powerful searching
capabilities
at your
fingertips.**

my synth's master tuning proved particularly annoying. (According to the manufacturer, the offending control file has been fixed.) *MIDI Quest* is a relatively stable program, although I did experience one program crash and several nonfatal *Windows* errors. (Sound Quest is still trying to pin these down.)

FINAL QUEST-ION
Is *MIDI Quest* for you? If you aren't the technical type who enjoys playing with the guts of your instrument's data, you may need to look elsewhere. The program's complexity and lack of intuitive organization can make your life more difficult.

If you are the tech type, however, and you can live with some of the program's warts, you'll appreciate the ability to do just about anything you want with MIDI patch data. There is an awful lot of power in *MIDI Quest*, but you must be prepared to tackle a learning curve in order to use it.

(Special thanks to Chick Cusick at Atlanta Discount Music.)

Allan Metts *is an Atlanta-based MIDI consultant, musician, and systems designer.*

Circle #438 on Reader Service Card

Audix PH25 Powered Monitors

By Erik Hawkins

**Reference
speakers for multitracking
and multimedia.**

Audix's PH25 powered speakers provide impressive sound quality in a compact cabinet. Although they were originally conceived for desktop multimedia applications, they also are a good choice for the small studio.

F inding midsized, powered, shielded speakers with studio reference-quality sound is a bigger task than it appears. Most midsized, self-powered speakers are not up to pro sound quality and aren't shielded. So I was pleasantly surprised to find that Audix's PH25 powered monitors rival studio reference speakers costing twice as much.

The PH25's speaker cabinet is made of ½-inch wood composite lined with acoustic fiberglass. Inside is a 0.75-inch, ferrofluid, polycarbonate dome tweeter sandwiched between two 5.25-inch woofers. The power amp is housed in one cabinet, which is dramatically heavier than its mate.

The left and right inputs are RCA jacks, which indicate the speakers were designed with consumer multimedia electronics in mind. However, Audix is planning a version with balanced XLR inputs, which is what one would expect for a powered studio reference monitor. A ¼-inch jack allows you to connect the powered cabinet to its unpowered mate. (The PH25s come in both shielded and unshielded versions.)

MAKE A WISH

Audix missed a few basic amenities I expected to see. For example, an on/off switch and a power-indicator light on the powered speaker would be greatly appreciated. According to an Audix representative, these features will be added to an upcoming model.

In addition, I'd like to protect my investment with an easily accessible fuse like those on the back of professional amplifiers. Perhaps Audix will consider this improvement for next year's model, as well.

USES AND ABUSES

I began my tests with a typical multimedia production composed of some music beds and narration. From speakers this size, I expected a harsh upper midrange bump that would place the project's narrator painfully up front. Instead, I got a smooth, accurate representation of the human voice, with only a slight bump in the lower midrange of the male voice. Moreover, the PH25s did an excellent job of reproducing the music beds, which had very little bass but a lot of highs. (The music

beds where composed mostly of strings, bells, cymbals, and piano.)

The highs were very pleasant, exhibiting crisp resolution, but not harsh like my Yamaha NS-10s. Upon completion of the project, I listened to the final mix on both the NS-10s and the PH25s and found the PH25s had a smoother overall response in the mid and high ranges, without sacrificing accuracy. The PH25s were much warmer, nicer sounding speakers.

Impressed by the results of the PH25s' first outing, I felt it was time to take the PH25s on a really tough sonic expedition. To do this, I tackled an acid jazz remix with a barrage of bass frequencies, including a TR-808 kick drum boosted at 60 Hz, and an upright fretless bass.

The upright fretless bass was played live so that, unlike a MIDI bass sound, it contained all the bass harmonics, pops, and squeaks of the real instrument, making it a real low-end acid test. I anticipated the bass would sound either thin or muddy, and the TR-808 kick drum would make the PH25s shudder in their boots.

Amazingly, the PH25s produce an incredible amount of bass considering the size of their woofers, and the bass was clear, full, and rock-solid. I compared the PH25s and a pair of Meyer

PH25 Specifications

Frequency response	50 Hz to 18 kHz (±3 dB)
THD	0.01%
Impedance	4Ω
Sensitivity	89 dB (1W @ 1m)
Maximum power output	60W peak/channel
Weight (both cabinets combined)	26 lbs.
Dimensions	15.5 x 7.5 x 10 inches

HD-1s (powered speakers that cost several times as much) and, as expected, heard a flatter overall sound with the HD-1s. But the Audix speakers gave the Meyers a real run for their money.

SPEAK OUT!

Having tried the PH25s in a variety of situations, from multimedia applications to hardcore studio mixing, I've come to appreciate the incredible versatility of these speakers. Because the speakers are self-powered, they are incredibly space-efficient, making them ideal for desktop applications where space is at a premium. The amplifier is extraordinarily quiet and operates at an incredibly low temperature. With their studio reference-quality sound, I would not hesitate to mix a CD on the PH25s.

Features such as these make the PH25s remarkably versatile. They're great for desktop multimedia production and do a good job in studio reference applications. But you could also plug them into an Ensoniq ASR-10 sampling workstation with operating system 2.0 or higher (which offers two tracks of hard-disk recording) and create a compact, full-featured production system. You could even use them as the ultimate Walkman speakers. The applications are limited only by your imagination.

Erik Hawkins *is a producer and musician in the San Francisco Bay Area. He is the owner of Sound Proof Productions and a partner in Digisonic Recording Studios.*

Circle #439 on Reader Service Card

Product Summary

PRODUCT:
Audix PH25 Powered Monitor
PRICE:
Unshielded $589
Shielded $609
MANUFACTURER:
Audix Corporation
PO Box 248
Lake Forest, CA 92630
tel. (800) 966-8261
or (714) 588-8072
fax (714) 588-8172

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

GeneralMusic S2R Music Processor

By Scott Wilkinson

Enhancements galore make the rack-mount S-series instrument a serious contender.

As a music-equipment reviewer, I've grown accustomed to manufacturers being mad at me for pointing out flaws in their products. So it comes as a nice surprise when a company actually accepts my criticisms as useful feedback. For example, in my review of the General-Music S2/S3 keyboards (see the November 1992 **EM**), I mentioned several serious shortcomings that hobbled an otherwise excellent instrument.

Instead of huffing and puffing their indignation, the company representatives actually thanked me. Then they went back to work and implemented many of my suggestions in the Turbokit retrofit upgrade for the keyboards. These changes also have been incorporated into the new S2R rack-mount version.
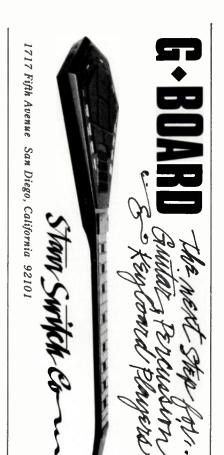
I delved deeply into the instrument last time, so I won't repeat myself much here. Instead, I'll concentrate on the new features, with particular emphasis on GeneralMusic's answers to my previous complaints. However, I will briefly summarize the basic features as we go. Please refer to the original review for detailed background information.

FIRST GLANCE

Dubbed a "Music Processor," the S2R includes a multitimbral sound module, sequencer, dual effects processor, and floppy-disk drive in a 3U rack-mount package. Like its keyboard predecessors, it features a large, graphic LCD measuring 240 × 64 pixels, with the brightest backlighting I've ever seen. The front panel is festooned with 64 buttons, which might seem a bit intimidating at first. However, they are organized into logical groups, and triangular, red LEDs indicate which buttons are active.

The LCD is still framed by seven small, triangular soft buttons on both sides, which are difficult to see (being black on black) and do not line up with
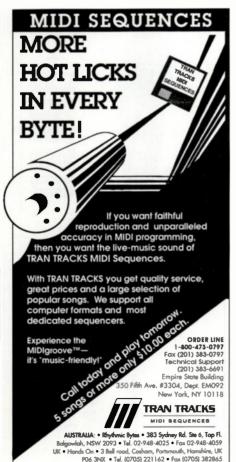
• S2R

their labels in the display. This makes it hard to find the desired button at a glance. Of course, it requires a physical redesign to change hardware items, such as the size of the LCD, which costs a lot more than changing the software. Nevertheless, this misalignment is quite annoying to me.

The infinitely rotating, detented data wheel and oddly shaped Enter and Exit buttons are still there, and the wheel still doesn't "wrap" from one end of a list to the other. Instead of seven assignable sliders as there are on the keyboard, the wheel can send assignable messages from seven virtual sliders in the display. This is an elegant way to incorporate that feature in a rackmount version of the unit.

The rear panel is little changed from the keyboards, with six audio outputs (two main stereo outputs and four separate outputs) and two sets of MIDI In/Out/Thru ports, which offer 32-channel operation and MIDI merging. This is a great feature I wish more manufacturers would adopt. A dedicated volume-pedal input is joined by two assignable footswitch inputs that cannot accommodate continuous foot controllers.

In addition to loading files from the floppy disk while playing a sequence, the multitasking operating system now lets you edit a sound while that sound is being played in a sequence. This is a wonderful capability; I was able to edit sounds in their musical context, which made editing much easier.

MAKING TRACKS

The polyphony has been doubled to 32 voices, thanks to the addition of a single-oscillator mode (discussed shortly). The unit is still 16-part multitimbral, and each part is called a Track, whether or not the sequencer is used.

At the bottom of the sonic hierarchy, there are 208 Waveforms (sample roots), many of which are multisamples, in 6 MB of ROM. Among these Waveforms are 61 drum and percussion samples. Unlike the previous instrument, you can now import various types of samples into 2 MB of sample RAM with a feature called Sample Translator 2.0. Earlier versions of this feature were available as a disk-based option, but it has now been incorporated into the Turbokit ROM.

The Sample Translator reads samples from DOS and Atari disks in *Avalon, Sound Designer, Sample Vision,* and WAV formats. It also reads samples and programs (multisamples) from Akai S1000 disks. However, it reads only the first sixteen samples in an Akai program. It can also receive samples via the MIDI Sample Dump Standard. (I'll examine the Sample Translator in more detail shortly.)

Waveforms are assigned to Sounds, of which there are over 500 in ROM. These include over 150 new Sounds in addition to the original 350, which retain their original Bank Select and Program Change numbers. The Sounds, along with their associated Waveforms, are organized into instrumental groups within a Sound Library. These groups, or Families, follow the General MIDI format and include Piano, Chromatic Percussion, Organ, Guitar, Bass, Strings, Ensemble, Brass, Pipes (flutes), Synth Lead, Synth Pad, Effects, Ethnic, Percussive, and Drum Kits. You can immediately jump to any Family in the Library, which is great for finding the Sound you want.

However, although the Sound Library includes some changes, the manufacturer didn't alter its unusual numbering convention and response to Bank Select and Program Change. Each Family consists of eight primary Sounds, with up to fifteen variations of each. To call up a particular Sound, you must first send a Bank Select message (Control Change 00, value 0 to 15) to specify the variation, followed by a Program Change to select the primary Sound with which that variation is associated. I realize this makes it easy to support General MIDI (which the S2R does), but it seems misguided in a professional instrument such as this. I would prefer to see Bank Select specify the Family and to have Program Change select the specific Sound within that Family.
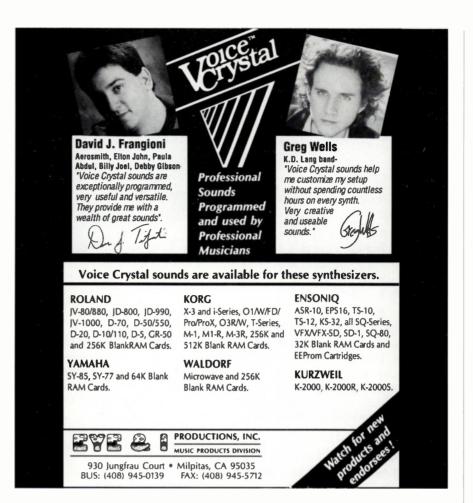
The number of multitimbral Performances in ROM has been increased from 10 to 100, organized in ten Banks of ten. This is a big improvement over the original instrument. The unit boots up in Performance Play, the normal playing mode. Sounds are assigned to Tracks in a Performance. Each Track has its own channel and In/Out port assignments and includes its own note and Velocity range, as well as one of seven Velocity curves (which still can't be edited).

Whenever a Performance is called up, each Track sends several messages on its assigned channel and MIDI Out port. These messages include Bank Select, Program Change, Volume (Control Change 07), and Pan (CC 10), as well as four S2R-specific messages: Attack (CC 12, Effect Control 1), Release (CC 13, Effect Control 2), Filter 1 Cutoff (CC 14, Undefined), and Filter 2 Cutoff (CC 15, Undefined). You can also create "dummy" Tracks to send any values you want for these messages to external gear by disabling the Track from controlling the internal sound generator. Each Track can send any incoming MIDI data on its assigned channel and MIDI Out, as well. In addition, the maximum pitch-bend range of each Track is now ±12 semitones instead of ±3. (Thank you!)

The Sound assigned to each Track can be changed by sending a Bank Select and Program Change to that Track's assigned MIDI In port and channel. In addition, the Performance can be changed with a Program Change (0 to 99) on a user-defined Common In port and channel. To avoid potential conflicts between the Common channel and a Track's channel, it's a good idea to specify the second MIDI In port for the Common channel. Although the display update is relatively slow when changing Performances, the new Performance is ready to play almost immediately, which is much better than the slow response of many instruments.

Each Track can be controlled by several sources simultaneously. The Local source is equivalent to an internal keyboard, accepting MIDI data on the Common channel, while the MIDI source accepts data from any and all channels and from both MIDI In ports for multitimbral applications. The Song source lets the internal sequencer control the Tracks, and the Option source keeps the door open for future upgrades. There are currently no Option sources available. Each Track can be enabled or muted for each source independently. Overall, this assignment scheme is extremely flexible. It can be a bit confusing at first, but I grew to like it a lot.

Finally, you can save up to ten Songs (sequences) in RAM. Each Song is associated with a Bank of ten Performances. Together, they form what GeneralMusic calls a Bank/Song.

• **S2R**

The GeneralMusic S2R offers 32-voice polyphony and a host of new features and improvements, such as user sample importing and editing, enhanced synthesis capabilities, extensive MIDI routing and control functions, and a better multitasking operating system.

MEMORY LOST AND FOUND

In addition to lots of ROM, the S2R includes 4 MB of user RAM, which is partitioned into 2 MB for user samples and 2 MB for user Bank/Songs and Setup files. The Setup files consist of the Sound Library, which includes any imported Waveforms; Effects Library, which stores Effects settings; Clipboard, an extremely cool feature that works like the Macintosh clipboard; and General, which includes things like tuning tables, footswitch assignments, and global MIDI parameters. As before, this RAM can't be expanded, and the partition can't be altered, which is too bad; 2 MB of sample RAM is pretty skimpy these days. In addition, none of this RAM is battery-backed, so all is lost when you power down if you haven't saved your work.

Happily, there are now two ways to save your work. In addition to the floppy-disk drive, which can now format disks in DOS (1.44 MB) and Atari ST (720 KB) formats, you can install up to 2 MB of static RAM chips to form a battery-backed RAM disk. (These are relatively standard chips, but they are not as easily installed as SIMMs. You can get them from GeneralMusic at a cost of about $500 for a full 2 MB.)

Both the RAM disk and a floppy disk can hold up to ten groups of ten Bank/Songs (that's 1,000 Performances and 100 Songs) in addition to one each of the four Setup files. Anything in the RAM disk is automatically loaded into user RAM on power-up. This is a creative solution to the origi-nal problem, but a RAM disk of maximum capacity is only half the size of the entire user RAM. Besides, it seems like a bit of a kludge; why couldn't they simply install battery-backed user RAM? Although there is still no SCSI interface, you can now perform SysEx bulk transfers.

SYNTHESIS

The S2R employs straight-ahead, sample-based, subtractive synthesis with resonant filters, but the new version is much improved over the original. Instead of always using both oscillators to produce the same Waveform, there are now three oscillator modes. Single mode uses one oscillator and affords 32-voice polyphony. Dual Crossfade mode uses the same Waveform for both oscillators and continuously crossfades between them, reading different portions of each one. These Waveforms are offset in time so the loop points don't coincide, which produces a fuller sound than Single mode.

Dual Oscillator mode lets you assign different Waveforms to each oscillator. You can also assign the Pan or Pitch envelope to crossfade between oscillators in this mode, although the selected envelope's primary function is disabled. I heartily applaud these significant improvements.

Sound Edit mode is somewhat unusual in that it retains an edited Sound in the Edit buffer. If you exit Edit mode, select a different Sound or Performance, and enter Edit mode again, the previously edited Sound is still

there. This is somewhat unexpected, as most instruments automatically replace the sound in the buffer under these conditions. You must consciously abort an edit to clear the buffer for another Sound. In the original version, the contents of the edit buffer were retained even if you made no changes to the Sound, but the new version preserves the buffer only if you edit the Sound. This is a nice safety feature once you get used to it.

The Sample Translator lets you edit imported samples, but its capabilities are pretty rudimentary by current standards. The waveform is displayed in the LCD, and you can zoom in and out to see more or less detail. You can also change the gain of the sample and set loop start and end points. The loop can cycle in the forward direction only, alternately cycle forward and backward, or can be turned off. The S2R can implement only one loop in a sample, but you can adjust the loop start and end points and loop types for up to eight loops in an Akai sample. (This seems completely superfluous. Why would you want to adjust these loop points if you can't use them in the S2R?) Finally, you can assemble up to sixteen samples into a Waveform (multisample), and send samples to an external device via MIDI.

As before, the quality of the ROM Sounds varies over a wide range. Some are great, while others are not so hot. In particular, I was disappointed to find nothing but weird synth-based sounds in the Ethnic Family. On the other hand, the new ROM Performances are often excellent, sounding rich and full and tending toward the lush synth-y

Product Summary

PRODUCT:
S2R Music Workstation
PRICE:
$1,995
MANUFACTURER:
GeneralMusic Corp.
1164 Tower Lane
Bensenville, IL 60106
tel. (708) 766-8230
fax (708) 766-8281

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

side. Many are multiple layers and splits intended to be played from a simple keyboard controller on the Common channel. Of course, you can build a huge, wonderful wall of sound in a Performance, but this seriously cuts into your polyphony.

SEQUENCER

The onboard sequencer is among the best I've seen on any synth "workstation." Along with 250,000-event capacity, sixteen tracks, and 192 ppqn resolution, it offers complete editing capabilities.

New features include user-specified note ranges for the Erase, Dynamics (Velocity scaling), Quantize Note-On, and Copy functions. Double notes can be eliminated with Erase Events. A track can now be copied to another Song, and Song Position Pointer is generated or recognized in Record and Play modes. In addition, a footswitch can now be used to start and stop the sequencer. These additions enhance an already excellent sequencer.

MIDI IMPLEMENTATION

The S2R includes some very powerful MIDI control features. As mentioned earlier, each Track is assigned its own MIDI channel, In/Out port, Velocity curve, note and Velocity ranges, Volume, Pan, etc. These assignments turn a simple, single-channel keyboard controller into a sophisticated master controller for an extensive MIDI setup.

Among the new MIDI functions is Proportional Volume, which was inspired by Don Goldstein and Jay Winding after using an S3 keyboard on Madonna's *Girlie Show* tour. They wanted to use a continuous foot controller as a master volume control for their MIDI rig. However, sending MIDI Volume set all the synths to the same level, destroying the mix. The S2R (or Turbo-kit-equipped S2 or S3) can now calculate proportional Volume values for all sixteen Tracks on the fly and send them in response to the movement of a single volume pedal. This is way cool for live performance.

A new General MIDI function configures the S2R as a GM device. In particular, it translates Program Changes on channel 10 to correspond to the GM drum map, and it accommodates Roland's GS superset of GM. This is great for GM applications, but as I said

earlier, it's less important for a professional instrument.

Three new Control Change messages have also been implemented: Balance (CC 08), which adjusts the balance between the oscillators in either Dual oscillator mode; Sostenuto (CC 66); and Soft Pedal (CC 67). Not only that, the Panic Button on the front panel now sends individual Note Off messages for all notes on all channels, as well as Reset All Controllers, All Notes Off, Channel Pressure 0, and Damper Pedal 0. It also sends a Pitch Bend value of 00 64, instead of 00 00, as it did before. This is important because Pitch Bend is bidirectional; 00 64 represents the center point, where there is no pitch alteration, while 00 00 causes full downward bend. Hallelujah!

EFFECTS

The effects haven't changed. They consist of two processors in a fixed, serial configuration. The first processor offers 23 reverb algorithms, while the second processor offers 24 modulation algorithms, including delay, flanging, chorusing, phasing, and rotary-speaker emulation, among others. User effects settings are stored in Effects Libraries, one for each processor. These are similar to the Sound Library.

As I stated in the previous review, the reverbs sound fine at low levels, but they get a little harsh and grainy at high levels. The delays seem to diffuse the attacks a bit. I hear little difference between the various modulation effects, although they sound good in general. I wish the routing were more flexible.

CONCLUSION

The S2R represents a significant improvement over GeneralMusic's previous offering. Many of my original concerns were addressed, and many new features were added. I especially like the new oscillator modes, the ability to edit Sounds while they are playing in a sequence, the extensive MIDI routing and control functions (such as Proportional Volume); the sequencer; and the Sample Translator (although it is rudimentary).

On the other hand, I still wish the Bank/Song RAM was battery-backed and the sample RAM could be expanded to accommodate larger samples. SIMMs would be great for the sample RAM. The RAM disk is nice, but it should have a maximum capacity

of 4 MB with the current user RAM limitations, even though that would be quite expensive. The misaligned soft buttons still bug me, and I don't like the Bank Select/Program Change organization of the Sound Library.

Nevertheless, I'm quite impressed with the work that GeneralMusic has obviously put into the S2R. The rackmount synth is worthy of serious consideration, especially at this price, which is $1,000 less than the S2 keyboard. I'm also more than a little pleased that they took my suggestions to heart. With that kind of attitude, I have no doubt that each product generation will be better than the one before. ☻

Circle #440 on Reader Service Card

Q Up Arts David Torn/Tonal Textures

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By Geary Yelton

Compositional snippets from this out-there guitarist offer musical nirvana.

C D-ROMs of drum loops and instrumental grooves are not unusual, but ambient guitar loops? *Tonal Textures,* David Torn's CD-ROM collection of ambient guitar loops, is unique. None of the tracks even sound like guitar, but David Torn isn't your typical guitarist. He specializes in coaxing unique timbres from his Klein guitar and signal-processing rig in collaboration with such well-known artists as Bill Bruford, Patrick O'Hearn, and David Sylvian. If Brian Eno were a master of the electric guitar, he might sound like this.
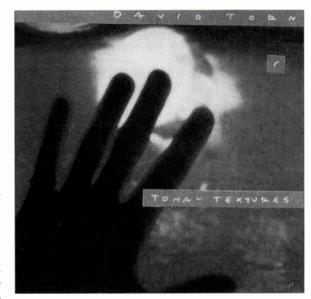
Tonal Textures is a mixed-mode, multiplatform disc. As a CD-ROM, it can be read by the Akai S-series, E-mu EIIIx, Kurzweil K2000, Roland SP-700, and

Roland S-760. It also plays like an audio CD, with almost an hour of absolutely lovely timbres. In fact, when I'm in the mood to play ambient electronic music on my stereo, it's usually a toss-up between *Tonal Textures* and Eno's *The Shutov Assembly.* Because of the sampler data, however, it won't play on my Macintosh's CD-ROM player like other audio CDs. (According to a Q Up Arts representative, Optical Media International is distributing a driver for the Macintosh that ignores the data tracks on Akai mixed-mode CD-ROMs, allowing you to play the audio directly from your drive.)

There are 33 loops, most of them over a minute long and many over two minutes. None of these samples are static sounds. Rather, they are musical atmospheres that move through time. Many include constant harmonic changes, as well as tonal modulation. Most sounds evoke visual metaphors that would be perfect for background movie music.

One cut, "Singing Arch," sounds like Wendy Carlos meets George Ligeti on the soundtrack of *2001: A Space Odyssey.* "Voice from a Tower" would have been right at home in Peter Weir's *The Last Wave.* "A Map of Dusk" actually sounds like a guitar and reminds me of King Crimson's "Matte Kudasai."

If you're looking for something similar to the drone of an orchestra tuning up, "Jangled Night" is better organized

David Torn's *Tonal Textures* sampling CD-ROM presents lovely, extended, ambient electronic music in Red Book audio, Akai S-series, E-mu EIIIx, Kurzweil K2000, Roland SP-700, and Roland S-760 formats. The collection also is available on audio CD.

than the real thing. "Slow & Buzzy Bubblebirds" invokes aural visions of Syd Barrett fumbling in the jungle at the far end of a large shopping mall. "The Aperiodic Benares Shuffle" is a throbbing mess with backwards guitar, which might make a good soundtrack for a bad acid experience.

"Heavy Barrels" is a solid block of shimmering texture that sounds a bit like Fripp and Eno's "An Index of Metals." There's good use of feedback on "Dinner Call for Baby Whales," a title that would be most appropriate if whales were at home in the vacuum of outer space. "Exotic Vox in Technobabble" includes the heavily processed sounds of unintelligible human speech, and at 33 seconds, it's the shortest track on the disc.

David Torn's liner notes explain how some of these sounds were created and offers suggestions for their use. Among other applications, he suggests layering them on top of one another or stringing them one after another.

When you use these sounds in your music, David Torn becomes, in effect, your musical collaborator. According to the license agreement that accompanies the disc, you may not use the samples as stand-alone music for broadcast or synchronization without altering it in some way unless you first obtain permission. As I understand it, that means all you have to do is change the pitch, process it through effects, or play a few notes of your own on top to legally use it anyway you please. Of course, the best use for these tracks is to add new textures to your original compositions. Even so, unless you plan to base your compositional style on these

Product Summary

PRODUCT:
David Torn/*Tonal Textures*
PRICE:
CD-ROM $399
Audio CD $129
MANUFACTURER:
Q Up Arts
PO Box 1078
Aptos, CA 95001-1078
tel. (408) 688-9524
fax (408) 662-8172

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

tracks, it's unlikely you will use all the sounds found here.

As great as it is, *Tonal Textures* is on the pricey side, even for a sampler CD-ROM. When you consider that you're paying not just for sounds, but for the right to legally use David Torn's compositions as part of your own compositions, maybe it's worth four hundred clams. Unless your time is worth its weight in gold, however, I think you should probably buy the audio CD and get your hands dirty sampling the tracks yourself. Sorry, David.

Geary Yelton *spends too much time sampling and sequencing and not enough time pursuing the finer things in life.*

Circle #441 on Reader Service Card

BBE 362NR and SPL Vitalizer

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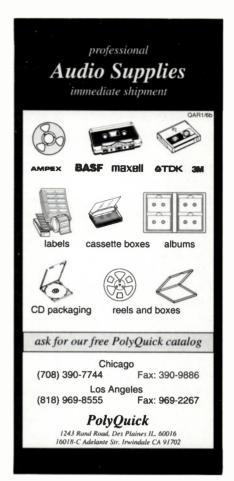
By Buddy Saleman

Psychoacoustics come to your sonic rescue.

Psychoacoustics processors seem to be the Next Big Thing on the signal-processing front. Sure, various models of the Aphex Aural Exciter and BBE processors have been around for years, but these units usually found their way into home studios by accident. However, new products are increasing musicians' awareness of this somewhat mystical side of audio production.

For example, various types of expanded-soundstage ("3-D") processing are popping up in music production and consumer electronics. And now, BBE offers its 362NR, while Germany's Sound Performance Laboratory has released the Vitalizer Psychoacoustic Equalizer.

These psychoacoustic processors do not produce 3-D sound, however. Both units reorient audio signals so that the ear hears the information the way nature intended. Each processor also performs other chores that can spruce up dull tracks and punch up your master tapes. Let's take a look at what these wonder boxes can do.

At $1,299, SPL's Vitalizer won't be an impulse buy, but its control over the sonic spectrum is astounding.

BBE 362NR

The 362NR Sonic Maximizer and Noise Reduction unit is not actually a psychoacoustic *processor,* because it introduces nothing artificial into the audio signal. Instead, it compensates for the phase and amplitude distortion inherent in all loudspeakers.

These speaker problems are frequency-dependent. For instance, a speaker's impedance increases as a function of frequency input, so higher frequencies are reproduced at lower amplitudes. Similarly, time delay is greatest on high frequencies, so these phase problems have to be compensated.

The 362NR adds factory-set phase correction to the high frequencies present in an input signal. Phase alignment is accomplished by breaking the audio signal into three bands. Low frequencies (20 Hz to 150 Hz) are delayed approximately 2.5 ms with a passive lowpass filter, while an active bandpass filter delays midrange frequencies up to 0.5 ms.

To compensate for frequency-dependent amplitude variations, an RMS average loudness detector continuously monitors the midrange band (150 Hz to 1.2 kHz) and high frequencies (1.2 kHz to 20 kHz) to compare the two bands' relative harmonic content. When the program material contains enough high frequencies, it applies a control voltage to a voltage-controlled amp, passing the high-frequency content. All of this delaying and adding of high frequencies is aimed at restoring the time and amplitude relationships that are distorted when an acoustic sound is reproduced through a speaker system.

The 362NR also includes a single-ended noise-reduction system that can diminish—or even eradicate—audible hiss from guitar amps, microphone preamps, noisy sound modules and effects processors, and poorly recorded analog master tapes. These features make the 362NR a marvelous mastering tool. You can add life to dull tracks without tweaking EQ and fatten up final mixes for that professional gloss.

The 1U rack-mount box itself, however, offers no frills. The front panel fits four rotary function knobs, an LED level indicator for each of the 362NR's two channels, a Process in/out button, a noise-reduction in/out button, and a power switch. Although the dynamics processor has two channels, one set of sonic maximizing and noise-reduction controls are shared between each channel. You cannot, for instance, add more "Lo Contour" to one channel and not the other. The nominal input level is set at -10 dBu.

The level indicators light up when the 362NR is 3 dB below actual clipping, which is a pretty cool feature. The early warning allows you to adjust input levels *before* you hear any nasty distortion.

Sonic maximizing control is simple. The unit has a Process knob for tweaking the phase-corrected treble frequencies (minimum to maximum) and a Lo Contour knob for dialing in the amount of phase-corrected bass frequencies (also minimum to maximum). The handy BBE Process in/out button allows you to instantly compare the processed signal to the unprocessed signal.

In the noise-reduction section, you get a Threshold knob (minimum to maximum) that determines the input signal level at which noise reduction begins. The implementation took a little getting used to, as a minimum setting is actually invoking *maximum* hiss reduction. A Release knob (fast to slow) determines the amount of time that processing stays active. The noise-reduction section also includes an in/out button.

The simple back panel offers ¼-inch input and output jacks for each channel *and* a set of RCA input/output jacks for each channel. The input jacks are high-impedance and unbalanced, while the output jacks are low-impedance, unbalanced. The unit has an internal power supply.

SONIC TEST

In pro studio applications, the 362NR was extremely clean. The unit did not add noise or other unpleasant artifacts to processed signals. The 362NR's Sonic Maximizing feature was pretty impressive. It always pulled more definition out of poorly miked or bad-sounding instruments. I used it on a messy (stereo) drum overhead track—the wash from the cymbals obliterated everything—and was able to clarify the elements enough to produce an adequate "room" sound.

On digital synth tracks, the Lo Contour added a fat timbre that evoked that analog warmth. I also tested the

Product Summary
PRODUCT:
SPL Vitalizer Psycho-acoustic Equalizer
PRICE:
$1,299
DISTRIBUTOR:
beyerdynamic
56 Central Ave.
Farmingdale, NY 11735
tel. (516) 293-3200
fax (516) 293-3288

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

Product Summary
PRODUCT:
362NR Sonic Maximizer
PRICE:
$349
MANUFACTURER:
BBE Sound Inc.
5381 Production Dr.
Huntington Beach, CA 92649
tel. (714) 897-6766
fax (714) 896-0736

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

unit on miked guitar and bass cabinets and found that the 362NR reduced the amp noise, while still leaving the high-end clear and distinct. For cassette duplication, the 362NR enhanced and clarified the top end and kept tape hiss to a minimum.

I wasn't as thrilled with the unit during professional mastering sessions. I couldn't tune in a "ready for CD duplication" sheen. In fact, the 362NR tended to add a slight harshness to mixes. But overall I was very impressed with the 362NR. It was easy to use and provided good, solid sound in most applications.

The 362NR certainly saved its share of butchered signals. For a price tag of just $349, the 362NR offers an inexpensive way to improve the sound of your recordings. (If you want a more professional-quality version of the 362NR, BBE offers a sonic maximizer device without noise reduction, the model 862.)

VITALIZER

The Sound Performance Laboratory Vitalizer (distributed in the U.S. by beyerdynamic) combines equalization, phase-shift manipulation, and harmonic enhancement in a 1U, rackmount box. The Vitalizer employs a network of computer-controlled, interactive filters to respond to and process the dynamics and frequency content of an input signal. Also, while some spectral enhancers only deal with high-frequency information, the Vitalizer uses amplitude-controlled phase shifting to accentuate bass frequencies.

Like the 362NR, the Vitalizer can be used to process an entire mix through your mixer's stereo bus, or individual instruments through a channel insert. You can also improve the sound of cassettes during tape duplication and enhance audio post-production and live sound reinforcement.

The Vitalizer front panel is split into two separate channels, each with its own dedicated controls. An Active button lets you audition processed and unprocessed signals. The Process Solo button lets you hear what the Vitalizer is adding to the original signal. This button is an excellent feature that can help you identify and tweak problem sonics, or just study exactly what this magic box does.
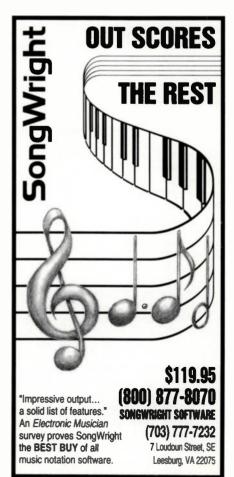
If you leave the Solo button active, only the processed signal is output. This means you can route the processed signal to a mixer's effects return (or additional input channel) and have both a dry signal and a processed signal available.

A peak LED alerts you when the input signal is overloading the unit, and a dual-concentric, rotary control dials in the output level and Process Depth. Process Depth regulates the amount of equalization added to the parameters selected on the Sub Bass (soft to tight) and Mid-Hi Tune (1 kHz to 22 kHz) knobs. The Mid-Hi Tune can be further adjusted by a recessed "Q" preset control, which can only be accessed using a small screwdriver. Higher "Q" settings enhance high-frequency material, such as hi-hats and cymbals.

The Vitalizer's Deep button spreads out the frequency range controlled by the Sub Bass feature and adds +4 dB at a center frequency of 50 Hz. Upper harmonics can be restored, clarifying high-end signals, by tweaking the Harmonics knob (0 to 100%). However, care should be taken when increasing the Harmonics processing, as any noise present in the original signal will be accentuated.

A separate Surround Processor, or Stereo Width control, is independent of the Vitalizer's other functions. One knob controls the effect (off to 100%) for both channels. Basically, this control detects the off-center elements of a stereo signal and "widens" the left/right relationships by feeding back these elements to the opposite channel, phase inverted. The resulting effect clarifies stereo imaging somewhat, but you should not confuse it with the

More affordable at $349, the BBE 362NR adds a clean sparkle to dull tracks.

ADVERTISER INDEX

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c. "Sonic Sorcery," p. 56	709	710	711	712
d. "The Gender Gap," p. 74	713	714	715	716
e. "Recording Musician: Fear of Stereo," p. 105	717	718	719	720
f. "Working Musician: Surviving Club Gigs," p. 112	721	722	723	724

signal enhancement found in movie theaters and high-end home-entertainment systems.

The back-panel connections are balanced XLR input/output jacks for each channel. There is also a ground-lift switch and a rotary selector knob for setting the Vitalizer's input level to -10 dB (consumer/home recording systems), 0 dB (consumer/home recording systems), +4 dB (professional recording systems), +6 dB (professional recording systems), +8 dB (television and radio broadcast equipment), or High (high-level insert signals). An internal power supply completes the back-panel offerings.

VITAL TESTING

I loved this box! I used it for professional mastering and found that the Vitalizer restored sonic punch and life to every mix. (And I thought the mixes were pretty good to start with.) Individual instruments, such as guitars, basses, and synths, sounded lush and full when routed through the Vitalizer. Processed drum tracks sounded amazing; the ping of the ride cymbals was right in my face. All levels of processing, from discreet to intense, were clean and artifact-free.

The Stereo Width feature really helped some old Sony F1 stereo masters that suffered from poor imaging. The Vitalizer spread out the stereo field, making the previously narrow-sounding F1 masters appear almost 3-D. I also corrected some stereo sub-mixes that were not bounced together with optimum separation.

During processing, low-end articulation was clear and warm, and the high-end was crystalline without exhibiting harshness. And even with all the spectral control available, the Vitalizer was easy to use.

The only knock I can give the SPL Vitalizer is that it isn't cheap. The list price of $1,299 puts the Vitalizer out of reach for most home recordists, but project and professional studios should gobble them up. For those who can afford it, the Vitalizer is well worth the price.

Buddy Saleman *is head engineer at Sound & Vision studios.*

For Reader Service information:
Circle #442 for the SPL Vitalizer;
Circle #443 for the BBE 362NR

Tech 21 TRI-O.D. Guitar Preamp

· · · · · · · · · · · · · ·

By Peter Freeman

Popular amp sounds stuffed into a stomp box.

The previous Tech 21 product review in **EM** cost editor Michael Molenda a milkshake. Mr. Molenda, it seems, couldn't distinguish between a recording made with the SansAmp PSA-1 tube-amplifier emulator and a recording made with a real Marshall amp (see the October 1994 **EM**). He bet on someone else's game, and he lost.

While unwilling to make such a foolish bet, I was eager to check out the TRI-O.D., the latest Tech 21 stomp-box. This simplified distillation of the most popular SansAmp sounds provides three preset amp simulations: British, Tweed, and California. These are intended to sound like Marshall, Fender, and MESA/Boogie amps, respectively.

LIVE AND KICKING

Tech 21's new stomp box is so easy to use that it requires little explanation. A single footswitch steps between the three available channels of emulated sounds. LED indicators show which channel is currently in use, and each channel has its own independent level and drive controls.

Each channel can be switched off, transforming its selection from an amp simulation into a bypass mode. For example, if you turn off channel 2, the footswitch steps through channels 1 and 3, then the bypass. You're never more than two steps away from where you want to be. If all three channels are switched off, the TRI-O.D. routes input signals directly to its Speaker Simulator, a clean, non-overdriven setting that provides tonal shaping similar to a multiple-

miked speaker cabinet. (The Speaker Simulator is normally active after the selected amp-distortion circuit.) This feature lets you run a signal through a fuzz box or preamp into the TRI-O.D. and then directly to a mixer. In this mode, the footswitch selects between Speaker Simulation and Bypass.

Master volume, bass, and treble controls complete the sound-shaping circuitry. The unit is powered by a 9-volt battery or Tech 21's optional, 200 mA, 9 VDC adapter ($12.95).

IS IT REAL?

As you would expect from the success of the SansAmp Classic pedal, the TRI-O.D. provides a variety of tones that are indeed reminiscent of the various simulated amplifiers. I found plenty of practical uses for the three basic sounds, as they gave a wide enough range of tones for many different situations. The active bass and treble controls, and the inclusion of independent Drive and Level settings for each sound, make the unit especially good for stage use.

In studio situations, the TRI-O.D. proved useful when driving an amplifier, or when recorded direct. However, the lack of refined tonal controls (such as those in the SansAmp Classic) limits your ability to fine-tune a sound.

I used the unit primarily with guitar (a simple ESP Strat copy), but I also experimented with bass and synthesizer processing. Generally, the TRI-O.D. sounded best when processing guitars and keyboards. With guitars, in particular, the essence of this device quickly became apparent. In contrast, bass sounds lacked depth and fullness.

Tech 21's TRI-O.D. simulates the sound of Marshall, Fender, and MESA/Boogie amplifiers. Although the timbral controls are limited, the simulations are impressive.

Like an amplifier, the TRI-O.D. reacts differently to assorted combinations of input levels and pickup selection. Although I was able to get pleasing sounds with the Strat on each of the TRI-O.D.'s channels, I preferred the sound of the Tweed position (Fender simulation) with that instrument. When I used synthesizers with the box, the scenario changed quite a bit. All three positions provided interesting musical tones, which depended entirely on the synth sound I had up at the moment.

CONCLUSIONS

I won't claim that the TRI-O.D. provides perfectly accurate amp simulation. I was more interested in whether the TRI-O.D. sounded good enough to do the job. A critical listener might be able to tell the difference, but I wouldn't bet any milkshakes on it.

If you're after a good selection of usable amplifier and distortion sounds, and don't want to haul three amps around, the TRI-O.D. is a good solution. True, you won't have the precise control you get with conventional amps or the SansAmp Classic, but I don't think great flexibility was Tech 21's main aim with the TRI-O.D. The box was designed to be good-sounding, easy-to-use, and practical, and it clearly succeeds at all three.

Peter Freeman is a freelance bassist/synthesist and composer. He has worked with John Cale, Jon Hassell, Chris Spedding, L. Shankar, Sussan Deihim, and Richard Horowitz.

Circle #444 on Reader Service Card

Product Summary

PRODUCT:
TRI-O.D. Guitar Preamp
PRICE:
$245
MANUFACTURER:
Tech 21
1600 Broadway
New York, NY 10019
tel. (212) 315-1116
fax (212) 315-0825

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

Aquila Systems MR2 Wireless MIDI System

• • • • • • • • • • • • • •

By Edward Tywoniak

Make MIDI fly through the air with the greatest of ease.

E ver since Italian entrepreneur Guglielmo Marconi pioneered wireless radio transmission at the turn of the century, we have attempted to free ourselves from electronic tethers. In this age of wireless microphones, infrared remote controls, cordless headphones, and cellular telephones, it seems only fitting that we should add MIDI to the list of wireless wonders.

Long sought by electronic musicians, wireless MIDI has been difficult to attain. Several MIDI wireless systems have been marketed, the most popular of which is a low-end system from MIDI-MAN that translates MIDI into an analog signal and sends it through the air with a conventional Nady wireless transceiver. But this system does not solve all the typical problems that face any wireless transmission system.

The most common problems include spurious radio frequency (RF) interference, cross-channel interference, and multipath cancellation (signal dropout due to phase cancellation from multiple reflections). While signal interference and unwanted noise are merely annoying in an audio signal, such interference wreaks havoc on a MIDI datastream, in which every bit is essential.

Aquila Systems has addressed the problems that beleaguer the developers of wireless MIDI systems by using *spread-spectrum* technology (explained shortly). Aquila is not the first company to attempt spread-spectrum wireless MIDI transmission; this technology was first applied to MIDI signals in the Gambatte! MidiStar Pro system (reviewed in the June 1989 **EM**). But according to Aquila Systems, its MR2 multichannel transceiver system employs recent advances in this area that were not available to its predecessor. Perhaps more important, the MR2 wireless system costs half as much as the MidiStar.

LOOK AND FEEL

The look and feel of the MR2 system is similar to that of conventional wireless audio systems. The MR2 has two components: a battery-powered, belt-pack transmitter (model MT2) and a dual-antenna, diversity receiver (model MR2A). The MT2 transmitter is extremely lightweight, even with the battery installed, but it is slightly larger (approximately 4.5 × 2.5 × 1.5 inches) than the transmitter modules found on typical wireless lavaliere microphones.

At the top of the transmitter unit is a MIDI In jack and a power switch with an associated LED. A recessed rotary-selector switch selects the desired RF channel. A 3.5-inch, rubber-coated transmitter antenna hangs from the bottom of the case, and a durable aluminum belt clip is attached to the unit's rear. Power can be supplied to the transmitter from a 9-volt alkaline battery, which provides up to five hours of continuous use; a rechargeable, nickel-cadmium battery that gives you only 2.5 hours; or an Ultralife lithium battery, which features up to ten hours of continuous use. Ultralife batteries are available through many retailers, including most Radio Shack outlets.

The MR2A receiving unit is housed in a black, metal box seven inches wide and approximately one rackspace high. An optional rack-mount kit for one MR2A is available. The front of the receiver includes two rows of LEDs; each row includes one green and ten red LEDs that indicate RF signal strength for each of the two diversity channels. The signal-strength meters are also good for detecting other unwanted interference that may be present in the operating area.

The presence of a MIDI signal is indicated by another front-panel LED. The front panel also includes a low-battery indicator for the transmitter unit. The main power switch, with an associated LED, is located to the right of the faceplate.

The rear-panel layout is simple, with one MIDI Out jack; a 3-position switch for selecting Antenna A, Antenna B, or diversity; and a 3-position switch for selecting between the different All Notes Off modes. A recessed, rotary selector for choosing one of the eight available broadcast frequencies is located on the receiver's rear panel, as is the input jack for the 12 VDC wall-wart power supply.

STRENGTH IN DIVERSITY

The MR2 is a true diversity system, which means it uses a dual-antenna receiving array and continuously monitors and noiselessly switches between antennas, depending upon which one is receiving the strongest signal. This makes the system less susceptible to interference, improving reception quality.

Sometimes, reception can be further improved by setting up the two antennas in a "near-far" array. The MR2 provides capabilities for such a setup by offering an Antenna Extension Kit (catalog number 1005) that lets the user stagger the antennas to achieve optimal reception. Another nice feature lets you defeat either antenna, so that each one can be individually adjusted for optimal placement.
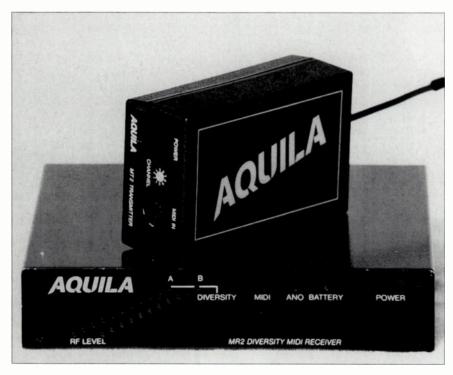
The MR2 system operates in the UHF high band between 902 and 928 MHz. This frequency range is typically the best for wireless audio applications. Aquila uses eight separate channels within this frequency range. Aquila recommends that no more than four wireless MIDI systems should be used simultaneously. For setups using four frequencies, the manufacturer also recommends assigning each instrument to every other channel to minimize interference problems.

SPREAD SPECTRUM
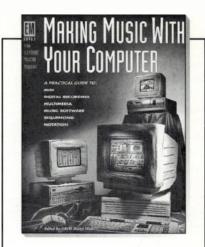
Spread-spectrum wireless transmission technology is generally used in RF data communications. Although it was developed to ensure secure military communications, spread spectrum is now commercially available in products such as the Aquila MR2. One advantage of spread-spectrum technology over conventional wireless technology is greater resistance to RF interference. It is also more difficult for unauthorized users to access a spread-spectrum system. As a result, a spread-spectrum wireless system provides more secure data transmission.

Conventional FM wireless systems transmit their RF signals over a relatively narrow bandwidth around a specified frequency, typically 50 kHz. A narrow-band receiver, tuned to this frequency, detects the signal and extracts the audio or data information. Radio interference is typically caused by other transmitters on the same or adjacent frequencies, or by dropouts created when the transmitted signal reflects off multiple objects, causing waveform cancellation.

Spread spectrum comes in two flavors. One technique, *frequency hopping* (FH), involves transmitting a signal while changing the frequencies or

Aquila Systems' MR2 MIDI wireless system utilizes the latest advances in spread-spectrum technology to provide reliable transmission at up to 350 feet.

channels several times a second. This method is not useful for music applications due to the relatively long time required to switch frequencies.

The second technique, *direct sequence* (DS), takes digital information, such as MIDI data, and combines it with a special error-checking code before transmission. This special code, which is used as a security key in industrial and military applications, is a unique and repeating sequence with a much quicker response factor than that of FH, thus making it useful for musical applications. The downside is that it requires a greater bandwidth. (The MR2 typically uses a 3 MHz bandwidth.)

Three frequency bands are legally available for spread-spectrum broadcasts: 902 to 928 MHz, 2.4 to 2.4835 GHz, and 5.725 to 5.85 GHz. However, in the bands above 2.4 GHz, the signals are difficult to manage, making the technology cost-prohibitive for music applications. These bands are generally used for scientific and medical purposes. The MR2 stays within the more manageable 902 to 928 MHz range.

Spread spectrum not only helps reduce the effects of signal interference, but it also is less likely to create interference in other radio systems, because it has limited energy at any given frequency. Multipath cancellation occurs at one frequency at a time, so broadband signals are somewhat resistant to this problem.

Another reason for using spread spectrum is that the Federal Communications Commission allows up to one watt of power for spread-spectrum transmission, while conventional UHF and VHF wireless systems can only transmit 50 milliwatts of power. As a result, spread-spectrum systems have much greater range and less need for line-of-sight broadcasting, which is usually required for conventional wireless transmission. Among other things, this greatly reduces onstage "dead" spots that often occur with conventional wireless systems.

IN THE AIR

Setting up the MR2 is quite easy, thanks to logical, straightforward connections. However, site-specific tuning of the antenna array is another story. Because other wireless services are also licensed by the FCC to broadcast in the UHF high band (including police and ham radio), there is always the potential for unwanted interference. The design, shape, size, and materials of the installation space can also contribute to the various types of transmission problems.

Fortunately, the MR2 uses a strong signal. The manufacturer claims up to 350 feet of operating range, thanks to the robust, 75 mW RF amplifier. In one of my favorite RF torture chambers, a small metal-and-cement-walled auditorium located in an area of concentrated RF traffic, I didn't notice any MIDI signal dropout until I was well away from the base station and behind several large physical obstructions. Even then, I encountered only sporadic problems. I tested the Aquila system simultaneously with two wireless microphone setups to check for intersystem crosstalk and interference. The MR2 worked flawlessly under these conditions.

Nonetheless, as with any wireless technology, sooner or later something could interrupt or cause errors in the MIDI datastream, resulting in hung notes. Aquila has provided two user-selectable features to deal with this lurking specter. Active Sense (AS) mode takes advantage of Active Sensing, a MIDI System Real Time message. With Active Sensing, if the datastream is interrupted, all connected MIDI instruments cease and desist their noise-making activities. In AS mode, if the MR2 detects MIDI transmission errors, the receiver cuts off all data for 0.3 seconds, which mutes the MIDI sound source.

However, some instruments (such as Ensoniq and E-mu synths) don't support Active Sensing, so they don't understand that an interruption of the datastream is a hint to chill out. (Many musicians' parents believe their children have a similar limitation.) If you have such ignoramuses in your rig, simply switch to NO123 mode, which has a more comprehensive error-detection structure. When data errors are detected in NO123 mode, the MR2 sends an All Notes Off command on the channel that was most recently used. (All Notes Off is MIDI Channel Mode message number 123, hence the name "NO123.") My review unit operated only on channels 1 and 2, but a recent upgrade extended support to all sixteen channels. An LED labeled "ANO" indicates an All Notes Off command is being sent.

If your synth doesn't recognize either Active Sensing or ANO (the Alesis QuadraSynth is such a beast), and you really feel the need, you may have to improvise. One possibility is to hang a small gadget such as the Eye and I Voice Crystal MIDI Merger between the controller and the MR2. This pint-sized, 2 × 1 merger has a well-implemented panic button that sends Note Off 0 to 127 on all channels, as well as All Notes Off. To be fair, though, the MR2 is so reliable that these panic features probably won't be needed.

ALL NOTES OFF

Wireless control of MIDI devices is an idea whose time is long overdue. The obvious audience for such a system are performing keyboardists and guitarists with a MIDI guitar controller. However, other alternative controllers, such as MIDI wind controllers, can also benefit from a cableless MIDI environment. Although I didn't have the opportunity to test the MR2 with MIDI-controlled visual effects, the potential for wireless MIDI Show Control applications is inviting.

It appears the engineering design team at Aquila has managed to tame the temperamental problems of wireless MIDI transmission. The MR2 paves the way for a previously unavailable degree of performing freedom.

Educator, author, composer, and performer **Ed Tywoniak**'s *busy schedule often prevents him from pursuing his true passion: watching and playing basketball. At 6-feet small, he is still waiting for his first NBA contract.*

Circle #445 on Reader Service Card

Product Summary

PRODUCT:
MR2 Wireless MIDI System
PRICE:
$1,499
MANUFACTURER:
Aquila Systems, Inc.
PO Box 572
Hatboro, PA 19040
tel. (800) 386-4554
or (215) 957-5450
fax (215) 957-5450

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FEATURES	●	●	●	●	◖
EASE OF USE	●	●	●	●	
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Tech Page

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Neurotic Synthesis

Neural networks bring new sounds to synthesis.

By Scott Wilkinson

When you consider all the different types of synthesis available today—additive, subtractive analog, sample-based, FM, physical modeling, etc.—you might think that the potential pool of basic synthesis techniques has been exhausted. However, at least one more technique for generating sound electronically is now being explored: artificial neural networks.

Among the researchers in this area is Forrest Warthman, who conceived the idea of a synthesizer based on the Intel 80170NX Electrically Trainable Analog Neural Network (ETANN) chip. Working in collaboration with Mark Thorson, who designed the hardware, and Mark Holler, Intel program manager for neural-network products, Warthman developed the first prototype using a single 80170NX chip. The latest version uses three chips.

The 80170NX includes 64 artificial neurons, each with 128 inputs and one output. These artificial neurons emulate the behavior of biological neurons, which accept inputs from many neighboring neurons and produce a single output.

Each input is connected to all neurons on the chip with a "synapse" that corresponds to the interface between biological neurons. The strength of each synaptic connection is specified with a weighting factor. These inputs can accept signals from the neurons in their own chip, neurons

in other chips, and external sources.

If the sum of the inputs to a specific neuron is well below a user-specified threshold, called the *sigmoid gain,* the neuron doesn't "fire" (i.e., it produces no output). If the sum of inputs is well above the threshold, the neuron fires, resulting in a strong, steady output. If the sum of inputs is at or near the threshold, the output exhibits a linear response. There is one sigmoid-gain value for all neurons on a chip, but you can vary each neuron's response by manipulating its synaptic weighting factors.

In the current version of the synth, a single input signal passes through a multitap analog delay. The input signal can be from an external source, such as a microphone, or from the synth's own final output. The signals from the delay taps are fed into the first chip's inputs. This chip performs a Fast Fourier Transform (FFT), which determines the outputs from the chip.

These outputs are fed into the second 80170NX, which simulates the behavior of biological neuron bundles called *cortical columns* (see "Tech Page: Trion Music" in the June 1994 **EM**). The cortical-column chip
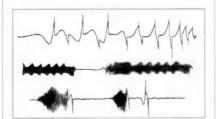

FIG. 1: These are some of the simpler waveforms generated by the neural-network synth. These waveforms are sometimes similar to bioelectric waveforms in the brain.

includes several external inputs, as well. This chip adds further complexity to the signals, which are then fed into the third 80170NX. This chip, which also includes several external inputs, behaves like a set of oscillators to produce the final output signals. These signals can be directed to a sound system, a visual display such as an oscilloscope (see **Fig. 1**), and/or back to the external inputs of any or all of the three ETANNs.

The instrument has no keyboard and does not respond to MIDI. In fact, it doesn't behave like a "normal" synth at all. Instead, the performer adjusts various dials, makes connections between outputs and inputs with patch cords, and varies the synaptic weighting factors with a computer. These activities control the complexity of the sound and the speed of transition between one sound and another, but the specific waveforms are not under user control, and you don't play notes in the traditional manner.

The whole point is to sonically model neurological processes, which are somewhat random in nature. For example, the Merce Cunningham Dance Company's repertoire includes a piece in which the original prototype produces sounds like a haunting and beautiful whale song. The output of the current version can be likened to the sound of a rain forest as it pulses with life. In addition, this device provides a unique insight (both audible and visual) into the behavior of neural networks, which should help neurobiologists and neurotic musicians alike. ●