EQ

THE RECORDING & SOUND MAGAZINE

AUG
1992

HIP HOP AT HOME

Hammer
DJ Jazzy Jeff
Marley Marl
& More

RAP TECHNOLOGY SPECIAL ISSUE

IN A BAND VAN

IT DON'T MEAN A THING (If It Don't New Jack Swing)

by Teddy Riley

EXCLUSIVE EQ REVIEWS

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- Kurzweil K2000
- SansAmp Rackmount

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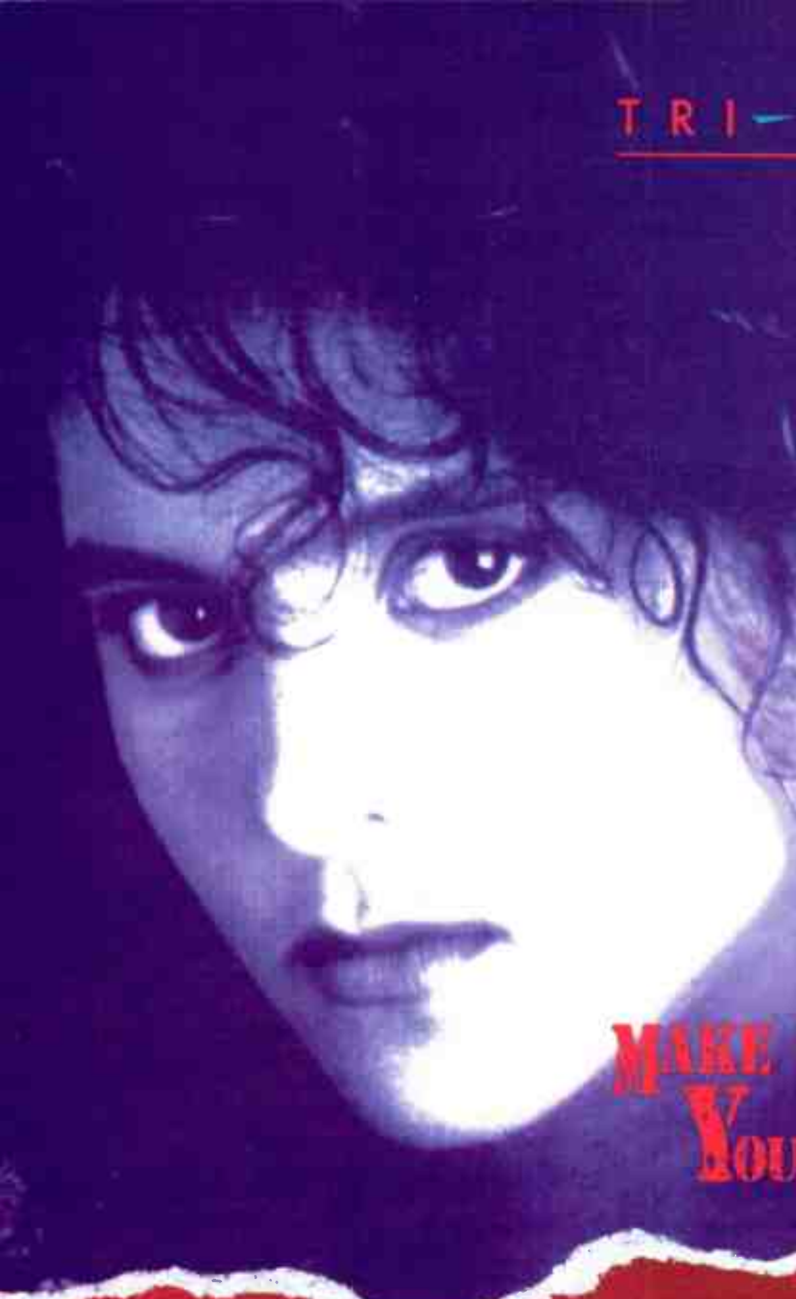

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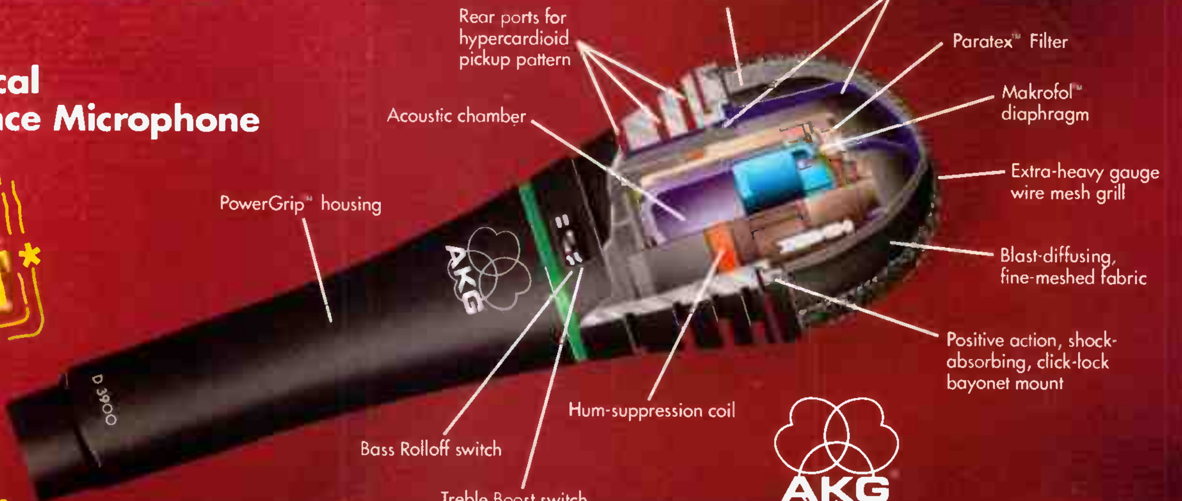

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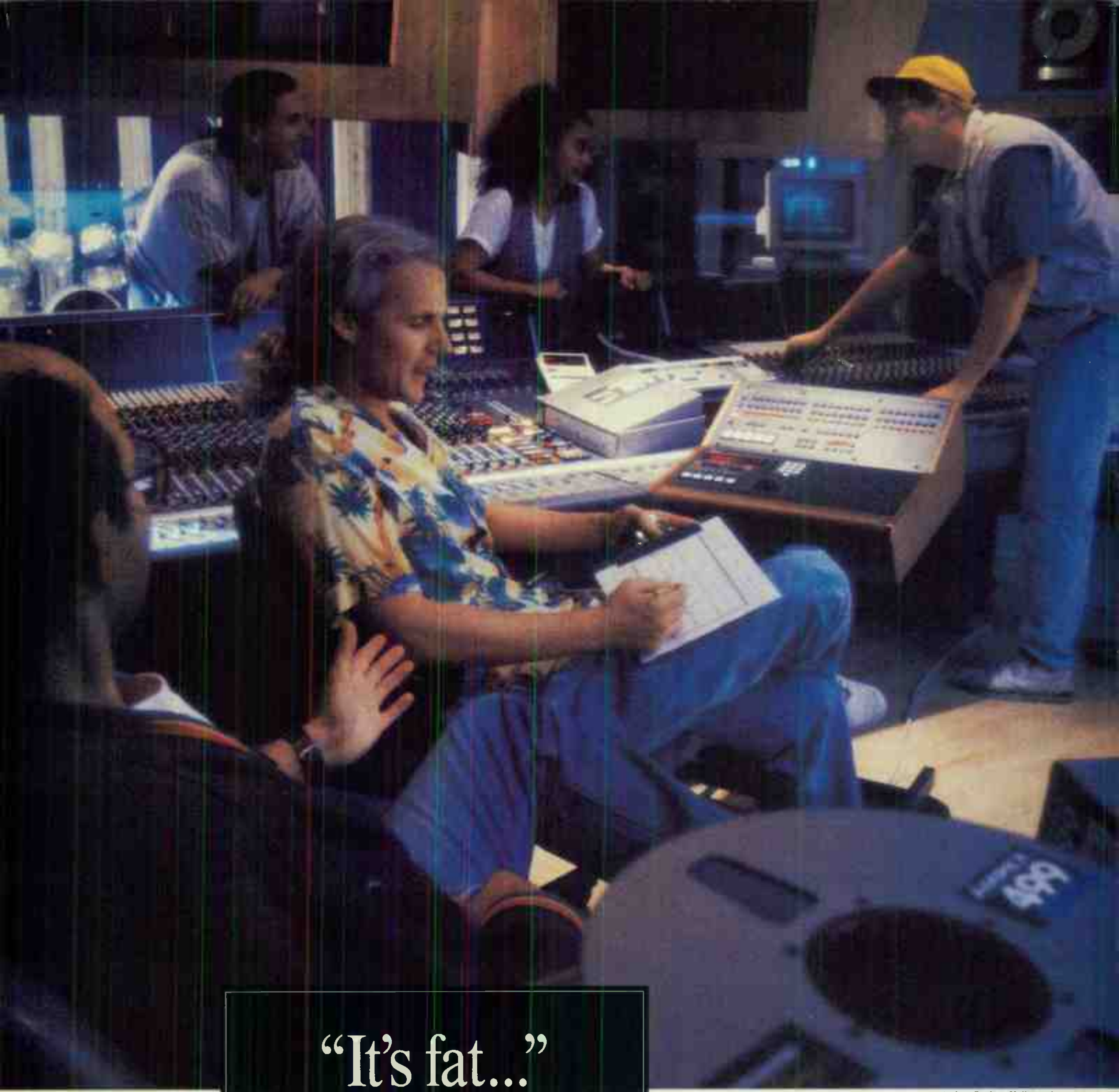

Andora Studios, Hollywood

We're printing at 9 over 250 and it still doesn't sog up.
I know, the harder you hit it, the better it gets.
Listen to that high hat.
It's stiff.
Yeah, and it's not squashed.
No, it's not compressing at all.
That's pretty amazing.

What's amazing is the noise floor.
What noise floor? I don't hear any noise.
Right!
Hey, this sound's got everything I need.
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I'd say it was audibly superior.
I'd say it just *sounds* better.

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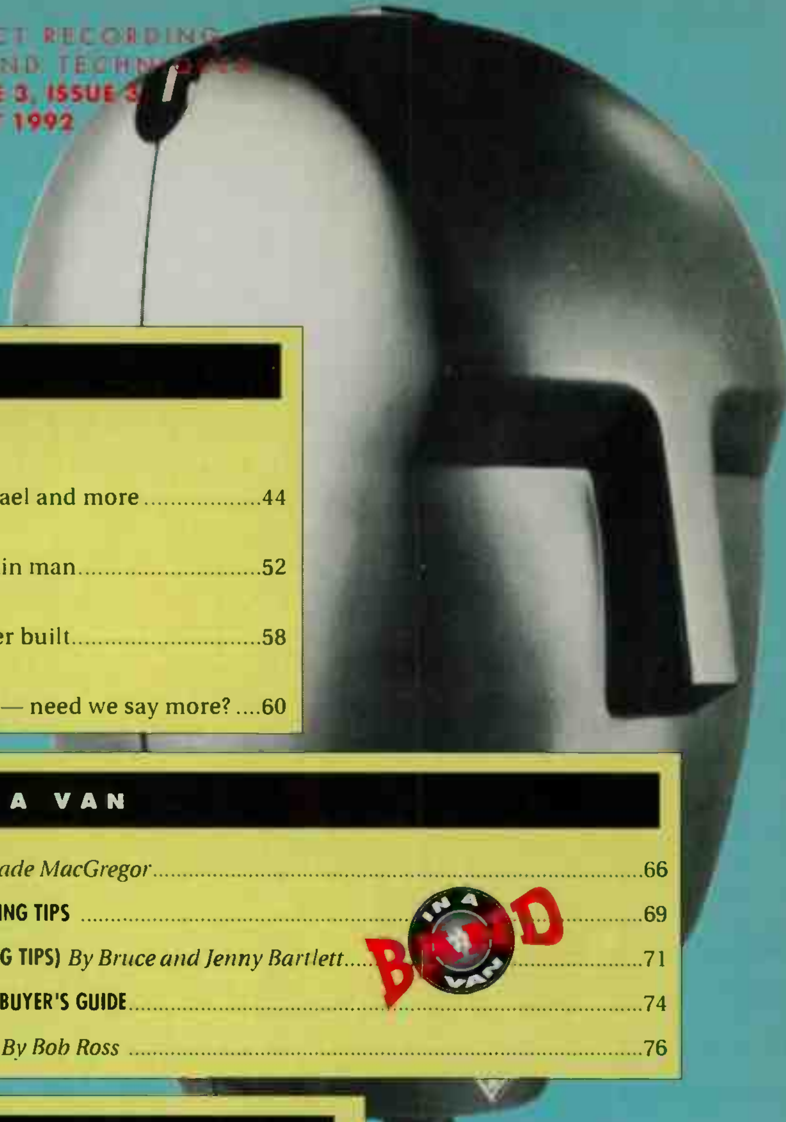
PROJECT RECORDING
& SOUND TECHNIQUES
VOLUME 3, ISSUE 3
AUGUST 1992

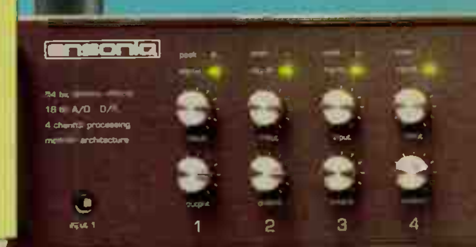
EQ

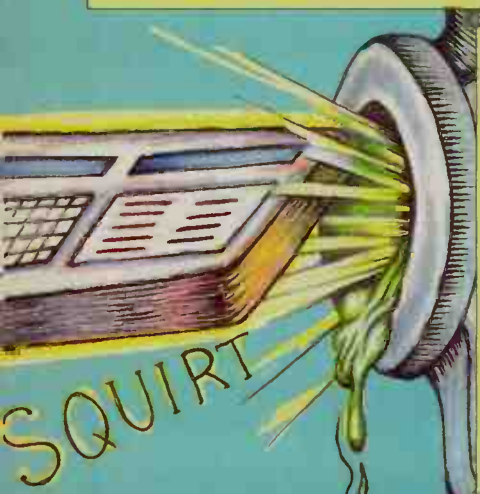


FEATURES

BAND IN A VAN

TECHNIQUES / WORKSHOPS





COLUMNS/DEPARTMENTS

On the cover: Riley by Christopher B. Reddick; DJ Jazzy Jeff by Steven M. Falk; Band In A Van logo by Mark Alhadeff.

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ALESIS
STUDIO ELECTRONICS

CIRCLE 06 ON FREE INFO CARD


Delay

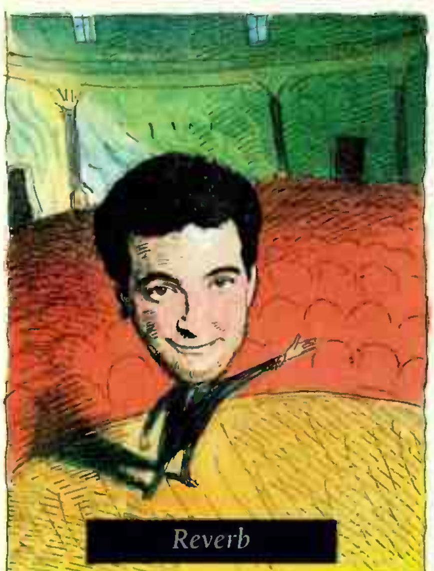

Stereo Flanger


Ambience


Rotary


Multi Band Chorus


Phase Shifter


Reverb


Vocoder


Stereo Pitch Shifter

Now, just about anyone can become a schizophrenic.

Yes, that's right. You too can become a schizophrenic.

All you have to do is buy an RSP-550 Stereo Signal Processor and follow these instructions. The possibilities are limitless—you can become 10, 20, even 30 different personalities—so read on and read carefully.

You can choose from as many as 39 personalities (algorithms). The somewhat disturbed illustrations to your left are examples of but 9 of them.

Technically speaking, these algorithms are as follows: (Please excuse us as we lapse into our decidedly multifaceted personas here. It can't be helped.)

1. The Delay algorithms range from simple single-line to genuine stereo and multi-tapped delays featuring up to eight independent delay lines, with up to 2700 ms of delay time each. With the RSP-550's Tempo Delay function, you can automatically assign the delay time according to, believe it or not, tempo. Or, if you'd rather, you can simply tap in the delay time.

2. The Stereo Flanger can be used for bi-flanging effects or independent left/right flanging.

3. Ambience is an effect that simulates the pickup from an ambience microphone and may be further modified with the Edge Expander function to emphasize the attack of a sound. It lets you create a realistic "presence," for instance, with the ambience of a recording studio or small club.

4. The Rotary algorithm delivers a detailed simulation of the distinctive rotary speaker sound—complete with independent rise/fall times for the horn and rotor. An Overdrive parameter reproduces warm tube amp-like distortion.

5. When combined with Roland's pioneering high-definition chorus effect, the chorus algorithms sport innovative effects such as Multi Band Chorus. This particular effect features two separate stereo or four separate mono bands, each with its own adjustable parameters. With the Penta Chorus algorithm, the input signal is divided into different frequency ranges, with each range independently processed so that you'll experience the most subtle or radical sound.

6. The Phase Shifter also has independent left and right channels and provides a 12-stage phaser per channel.

7. The RSP-550's Reverb has the high-density spaciousness that acoustic environments create as well as a smooth and natural release. The Hall/Room/Plate algorithms feature options for a wide range of reverb time settings—0.1 to 480 seconds—with a frequency response of 15Hz to 21kHz.

Parameters such as Pre Delay Time and Early Reflection enable you to set the apparent "length" of the room while HF Damp simulates reverberation from different wall materials.

By the way, all of the reverb algorithms also include three-band EQ for tonal adjustment of effected sounds.

8. Only the RSP-550 has a Vocoder algorithm which superimposes your voice onto other sounds, such as brass or a jet taking off, to give your voice characteristics of that sound. Incidentally, brass makes you sound like a robot.

9. The Stereo Pitch Shifter allows an independent pitch shift per channel because it features independent left and right channels. Or you can use up to four pitch shifters simultaneously each with a four octave range.

There are 30 more algorithms where these came from, but more on this later.

One of the truly cool things about the RSP-550 is the true stereo ins and outs that both create spacious-sounding stereo effects and retain the integrity and panning of the input signal.

No doubt that by now you've already guessed that this machine is for serious users only. This is because only serious users will quite know what to do with a dynamic range of 95dB coupled with a frequency response of 15Hz-21kHz and a THD of 0.02 or less. Not to mention signal processing conducted at a CD-quality sampling rate of 48kHz, with fully independent 16-bit A/D and D/A converters for each channel.

Beyond all of these qualities, the gonzo-in-straightjacket effects, the commensurate professional sound quality and the ability to control effects via foot-switches,the RSP-550 has tremendous MIDI capabilities. With MIDI, you can control up to four parameters simultaneously from controllers, aftertouch, velocity, note range or pitch bender.

Now, as we promised, here's more on the 30 additional algorithms. To hear them, you need to visit a Roland dealer, who, in this case, can be thought of as a kind of reverse psychologist. If that makes any sense. It does to us, but then we're already schizophrenic.

No we're not. Yes, we are.

Roland®

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CIRCLE 51 ON FREE INFO CARD

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EQ wins award for editorial excellence

Defining Recording & Sound in the Nineties

Two years ago this month, *EQ* was launched with the mission to define recording and sound for the decade ahead. By all indications we have succeeded beyond our wildest dreams. History proves that *Modern Recording* was the magazine for the creative recordist of the 1970s. *MIX* was the pro studio magazine for the 1980s. And *EQ* is the magazine for the 1990s, dedicated to the merger of creativity and professionalism.

Besides the significant advertising growth that is obvious to anyone who reads us on a regular basis, *EQ* is now also recognized for its editorial leadership. What's more, *EQ* recently won this year's MagazineWeek Award for Editorial Excellence, for delivering "hands-on, how-to information for today's project recording and sound market."

We are honored by this award. We are equally honored by the fact that *EQ* was the first magazine to identify this rapidly growing technology trend, where advanced recording and sound capabilities are being made available to an ever-broadening number of creative recordists, musicians, producers, songwriters, and audio-for-video specialists — those of you who now consider your home or project studios your primary creative instrument. We also recognized that practical, expert advice was what you wanted to read most.

Similarly, we became the only magazine to concentrate on the sound reinforcement counterpart to this trend with our "Band in a Van" section, which serves the live sound information needs of the gigging musician and his band. Many of you work in your project rooms by day and then make a living at night, taking your bands to regional clubs and concert venues. Once again, we supplied you with your only source of practical, how-to, expert advice.

We will continue to nurture and grow *EQ* in the years ahead with the type of editorial leadership you, our readers, require, exciting your creativity and advancing your technique. Our editors, Hector La Torre, Martin Porter and Craig Anderton, are the most experienced and creative team in the business and I commend them on creating the fastest growing, most influential magazine available. Two years ago we set out with the mission statement that "useful information is what *EQ* is all about." We thank you for helping us determine what type of articles are really "useful" and we look forward to your input in the years ahead — a decade when affordable technology and personal creativity will help us all pursue new horizons.

EQ is your magazine. We have pointed out the direction. You will ultimately determine how far and how fast we go.

Paul Gallo

Paul G. Gallo
Publisher

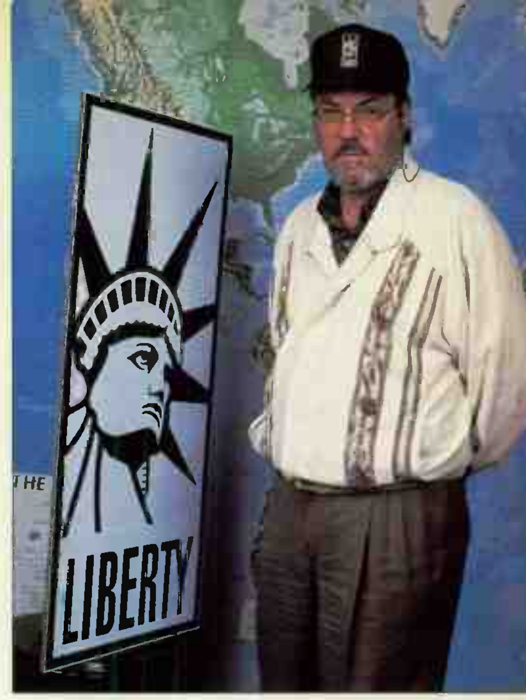

Photo by Beth Gwinn

AN IN-VIG-NANT RESPONSE

First off, let me thank you for providing a much needed magazine that provides *useful* information for those of us in the audio industry.

My point is short. I thought it odd that you interviewed Butch Vig about the Nirvana LP. Granted, he played a big part in the sound of that LP, but look closely at the liner notes — "mixed by Andy Wallace," who is really the one responsible. Listen to other Butch Vig recordings. They are not as good, nor as commercial.

Scott Colburn
GravelVoice Records
Hollywood, CA

BOWEN BASHING

I am writing in regard to the article written by Jimmy Bowen in the April issue of *EQ*.

The title, "Liberating Nashville," was a good opener. Offensive, but good. Where did Mr. Bowen get the idea that Nashville needed liberation? Liberation from what?

Mr. Bowen rode into town in 1976. By anyone's measure, that makes him a Jimmy-Come-Lately to the Nashville recording scene. As I recall, Nashville producers and musicians had been creating good music and smash hits many years prior to his arrival. Surely he's heard of Chet Atkins, Owen Bradley, Jerry Kennedy, Jack Clement, Billy Sherrill and Glen Sutton. There are many other great record producers that could be mentioned, but to do so would only further bury Mr. Bowen's thinly-veiled contention that no one in town knew how to produce until he arrived.

He mentions that no one used drums (or if they did, only with brushes) until he came along. This statement only reveals the shallowness of his knowledge about the city he now calls home. Surely he's heard a little Roy Orbison ditty called "Pretty Woman." Drums, drums, drums.

How about Elvis' recording of "Hound Dog?" Those were drums D.J. Fontana was playing, and he was playing with sticks, not brushes. Again, to

WRITE TO US

EQ wants to dialogue with you. Write to: Letters to the Editor, EQ, 939 Port Washington Blvd., Port Washington, NY 11050. Letters must be signed, and may be edited for clarity and space.

mention more examples would be overkill.

In Nashville, the usual recording procedure is to use the instrumentation the song needs, not what the producer might personally prefer. The procedure is also to be prepared, then go into the studio and get to work.

Before Mr. Bowen came to town, album budgets were usually somewhere between $25,000 and $50,000. There were a few exceptions on both ends, but most albums fell within this range. The cost was reasonable and it was easy to understand why. Nashville has some of the best pickers in the country. They go into the studio, hear a song a time or two, write their charts, record it, and go on to the next song, or go home.

Mr. Bowen comes along and budgets hit the ceiling. A $50,000 album now costs $150,000 or more. Mr. Bowen is a competent record producer and has been for some time. But I've heard most of the records he's done lately, and I have a question: Where did the production money go?

Perhaps that's what this liberation thing is all about. Maybe the Bowen Liberation Plan is related to the federal government plan: Spend more money but deliver the same old goods.

Joe Keene
Kennet, MO

EMULATING GROOVE

Your April '92 issue carried an article entitled "The (<$50) Speaker Emulator" by John Catheline. Mr. Catheline says that the Speaker Emulator is "one of the most useful accessories for guitarists since tube guitar amps were first introduced over 50 years ago." We surely do agree. Mr. Catheline's comments included details on the construction of a relatively simple example of the device, and some application "ideas." Unfortunately, he left out something rather important: namely, that we have a patent on the device (U.S. patent # 4,937,874 granted to R. Aspen Pittman and Dr. Marshall Buck, filed July 2, 1986) and that we also have trademark rights to the name Speaker Emulator.

Because of my patent and trademark, I must formally inform you and your readers that the Speaker Emulator was invented by a team of researchers employed by my company. This team included Richard Heyser, Dr. Marshall Buck, Bill Isenberg, Dean Jensen, Dick Rosmini, and myself. I thought of it, we invented it, patented it, trademarked the name, first sold it, and have licensed the technology to several well-known manufacturers — most notably Marshall Amplification of the UK who now produces (under license) a version of the device.

I mean, we really don't care and won't sue any of your more interested readers who want to mock up a version of Mr. Catheline's circuit *for their own private use*. Heck, we went public and published our patent so the world could know about the Speaker Emulator and the music could get better. However, we will strongly object to any person or company who would build such a device for sale or commercial use without first taking out the appropriate license from us. That is what U.S. patent law is all about — protecting intellectual property and the invention process, so that scientists and companies can recover their investment in research and development.

The first Speaker Emulator offered for sale was our Studio Tube Preamp for Guitar, or STP-G, and it went on sale in 1986. The STP-G has a complete tube preamp section (5-GT-7025s) and a traditional power section (matched duet of GT6V6) that produces about 35 watts RMS @ 8 ohms. The STP-G power section drives the built-in Speaker Emulator for direct recording with real power tube distortion feel and sound. The STP-G is still in production and has been a very successful product for us. It is owned and used by hundreds of recording studios and top players including Chet Atkins, Lee Ritenour, Jeff "Skunk" Baxter, Dave Gilmore and Billy Gibbons, just to name a few.

The simple circuit diagram of a reactive load device drawn by Mr. Catheline, which he labels a Speaker Emulator, doesn't have the same parts count or parts values, nor is it as complex. It will not sound nearly as good as our device. Of course, ours sells for considerably more than $50. The fact that you've labeled the article "The (<$50) Speaker Emulator" really makes the point: you get what you pay for. It costs me a whole lot more than $50 to do it right. Companies like Groove Tubes and Jim Marshall are not ripping people off with Speaker Emulators that sell for around $500. Just having the raw parts and knowing how to make a basic speaker or guitar pickup, even with the circuit diagram, doesn't mean you can start making JBLs or Seymour Duncan pickups next week. I mean there's probably only a few dollars worth of wire and magnets in any guitar pickup, but making the pickups sound great takes years of experience and refinement. That's where the "art" comes in, and the artist should get paid for his knowledge.

Our Speaker Emulator is really well known, has been around for quite a while, and is protected by a U.S. patent and trademark.

Aspen Pittman
President
Groove Tubes

DIP SH*T!

Controlling the Fostex R8 using the MTC-1 and C-Labs Notator is much easier than Phil Shackleton's article "Now It's a Sync" would lead one to believe. Our goal at Fostex was to provide a "thread it and forget it" multitrack tape recorder, and through cooperation with Atari, C-Lab and Steinberg-Jones, we have achieved our goal. The fact, simply stated, is this: thread a tape that has been striped with SMPTE timecode on the R-8, connect the R8 to either C-Labs Unitor or Steinberg-Jones Cubase, using the Fostex MTC-1, and you never have to touch the R8 again until you take the tape off at the end of the session — period. All transport functions can be controlled via Cubase or Notator, including punch in/out recording and auto-locate, using the same transport controls that run the sequencer. Phil Shackleton devoted much of his article to DIP switch settings on the MTC-1, and while it is important to set your DIPs correctly, once they are set you can forget about them. Writing about DIPs clutters and confuses the real message — that this is a dramatic breakthrough, making the entire recording process easier to manage, and more FUN!

Glenn S. Fisher
Product Specialist
Fostex Corporation

more letters on page 105

The Peavey DPM® SP/SX Sampling Combination

"The Peavey DPM SP has enough sound-processing power to generate incredible sounds.... Overall, the SP represents tremendous value for the money....The engineers at Peavey are to be commended for building a highly capable sound module into a cost-effective, upgradable package."

Electronic Musician
May 1992 Issue

"The SP offers ambitious programmers the potential for creating new signature sounds. Particularly considering its low price, expandability and first-rate storage and loading capabilities, the SP gives a musician more than just an introduction to sampling. With the SP, Peavey moves the flexible-architecture philosophy to new frontiers."

EQ Magazine
February 1992 Issue

The DPM® SP/SX sampling system is a phenomenal value. Costing thousands less than comparable units from our competitors, and hundreds less than most low end systems, the SP/SX combination represents the most powerful yet affordable, full-featured 16-bit sampling system on the market today!

The DPM® SP rack-mount sample playback module offers 16-bit resolution and 44.1 kHz stereo sample playback rate for industry standard sonic quality that is without equal.

The SP is capable of handling up to 32 megabytes of internal sample memory. The sample RAM is expandable with low-cost industry standard SIMMs expansion boards.

The DPM® SX Sampling Xpander module allows you to digitally record your own 16-bit samples and send them over SCSI to the SP or in the standard SDS format to your DPM 3 or other compatible instrument.

Up until now, high-quality sampling has been something that was out of reach for most people. Not only because of the expense, but because of the tedious time and effort required to create good samples. The union of the SP/SX finally brings together high-end full-featured sampling with ultra affordable pricing for the working musician.

Sample the new DPM SP and DPM SX sampling system today! Be sure to ask about the new DPM SP sample library available now at your nearest Peavey dealer!

CIRCLE 46 ON FREE INFO CARD

The Monitor [illegible] magazine from Peavey is a publication filled with [illegible] info musicians want to know. Included are [illegible] players. You also get the latest news on Peavey equipment. To receive [illegible] for only [illegible] check or money order to Monitor magazine, Peavey Electronics, 711 A Street, Meridian, MS 39302 2898 *Price good in U.S. only

PEAVEY

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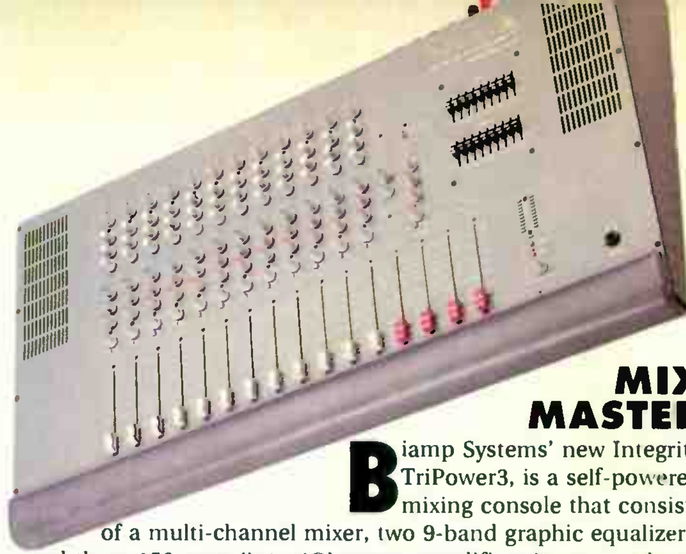
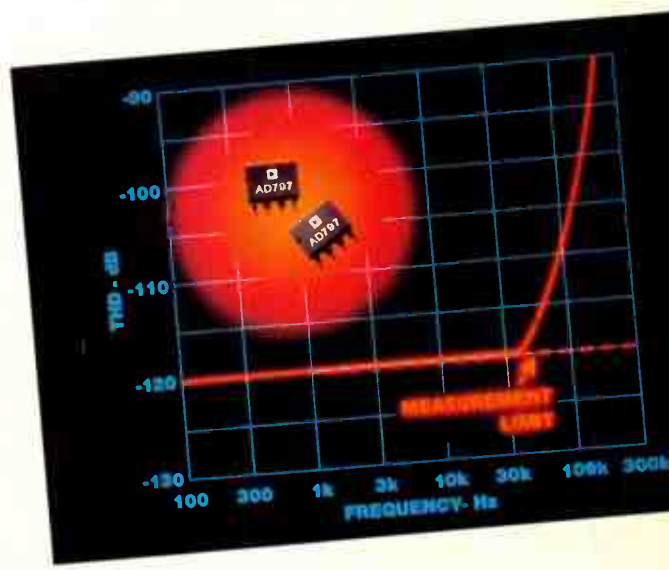
EQ PRODUCT VIEWS



MIX MASTER

Biamp Systems' new Integrity TriPower3, is a self-powered mixing console that consists of a multi-channel mixer, two 9-band graphic equalizers, and three 150-watt (into 4Ω) power amplifiers in one package. These system components can be assigned in various ways to handle most any application. Discrete transistor mic/line preamps and MOSFET power amplifiers are incorporated to provide quality sonic performance. Individual circuit boards and an all-metal chassis help to provide long-term reliability. This unit features 12 or 16 mic/line level input channels and 3-band equalization. Other features include a complete channel solo system, stereo and mono main outputs from the mixer, stereo tape output from the mixer, two stereo returns to mains and aux 3 (for effects or tape) and more. For more details, contact: Biamp Systems, 14270 N.W. Science Park, Portland, OR 97229; 503-641-7287 or 1-800-826-1457. Circle EQ free lit. #101.



HEAD STRONG

Neumann has introduced the third generation of the binaural "artificial head" microphone system. The KU 100 "Fritz III" provides improved acoustic performance and better specs than its predecessor. It has transformerless circuitry and a built-in battery supply. Single-ended BNC connectors as well as standard XLR-type connectors are supplied. In addition, Neumann has simultaneously introduced the KFM 100, a new stereo mic. It is a wooden sphere, 20 cm. in diameter, with two pressure microphones flush mounted on it. The two mics are diametrically opposed from one another. Low frequency response goes down to 10 Hz. Arrangement and distance of the capsules result in a nearly constant directivity factor and a smooth diffraction of the soundwaves around the sphere, according to the manufacturer. For more information on these mics contact Neumann/USA, 6 Vista Drive, P.O. Box 987, Old Lyme, CT 06371; 203-434-5220. Circle EQ free lit. #102.

SHURE BET

Shure Brothers has introduced the BetaGreen line of mics that provides high quality performance at affordable prices ($60 to $220). Three dynamic and two condenser models are offered to handle a wide range of recording and live applications. Some models feature neodymium magnets and superior shock isolation. All offer an on/off switch and unbreakable stand adapter. Four of the five models come outfitted with a deluxe foam-padded nylon carrying case. Find out more by contacting Shure Customer Service at 1-800-25-SHURE. Circle EQ free lit. #103.

PEST CONTROL

The FBX 900 Feedback Exterminator from Sabine is an automatic, digital feedback controller (AFC) that finds and eliminates feedback in any sound system — and it does it quicker and with less tonal degeneration than other methods, according to the manufacturer. Employing Digital Signal Processing (DSP) technology, the FBX analyzes the entire audio signal, quickly senses any feedback and immediately determines its pitch. Once feedback occurs, the FBX assigns one of nine notch filters to the resonating frequency and eliminates the feedback within a half of a second. The single RU unit is built for roadworthiness and it can be placed anywhere in a sound system that a graphic equalizer might be used — the most common configuration being between a mixer output and input of a power amp. Another useful placement is between a microphone and mixer or preamp. It lists for about $600. For the whole picture, contact Sabine Musical Manufacturing Company, 4637 N.W. 6th Street, Gainesville, FL 32609; 800-626-7394. Circle EQ free lit. #105.

OP (AMP) ART

The AD797 operational amplifier from Analog Designs boasts voltage noise of only 0.9 nV/√Hz at 1 kHz while total harmonic distortion is just -120 dB at 20 kHz. The AD797 voltage noise remains flat over the full 8 MHz bandwidth (Gain=10). Settling time is guaranteed at 1.2 µs (800 ns typical) to 16 bits. Input offset current is typically 200 nA (300 nA maximum over temperature). The AD797 is specified for operation from supplies ranging between ±5 to ±15. The AD797 features a slew rate of 20-V/µs and a gain-bandwidth product of 110 MHz (Gain=1,000). Full power bandwidth is 280 kHz at 20 Vp-p. Output current drive is typically 50 mA (30 mA minimum), permitting small resistor values to be used for gain setting and thus keeping resistor noise below the noise level generated internally by the AD797. For the full story contact Analog Devices, 181 Ballardvale Street, Wilmington, MA 01887; 617-937-1428. Circle EQ free lit. #104.

BOTTOMS UP

The Punch-10™ Bass Enhancement System from Furman is a subharmonic processor designed for augmenting the bass content of recorded music. Not a mere equalizer, the Punch-10 synthesizes bass where none existed before by generating a bass component one octave below the existing fundamental. A subharmonic level lets you control the amount of synthesized bass and it can be used with or without a subwoofer system. A Low Frequency Assign button routes the bass content either to the subwoofer output only, or to both the subwoofer output plus the main stereo outputs. If no subwoofer is being used, the latter position allows the enhanced bass to be heard through the available speakers. The subwoofer output contains frequencies of 85 Hz and below; higher frequencies are rolled off at 18 dB per octave. The bandwidth appearing at the main outputs is set with the Main Speakers High Pass control, adjustable from 15 to 135 Hz. To protect the speakers, the Punch-10 includes a hard limiter acting on the processed subharmonic signal, with a threshold adjustable from -20 to +20 dBu. For more details contact: Furman Sound, Inc., 30 Rich Street, Greenbrae, CA 94904; 415-927-1225 Ext. 23. Circle EQ free lit. #106.

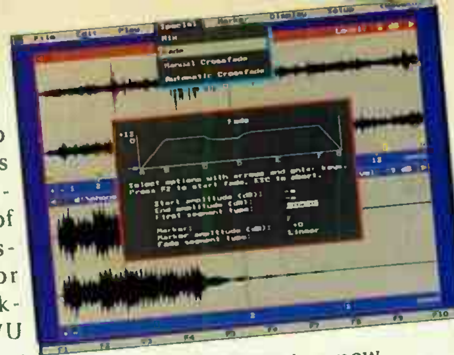
HEAVY SET

The N-Series FB™ "Fatboy" is an electronically-controlled nearfield speaker system from Community Professional Sound Systems that is designed for any application that requires strong bass response. The Fatboy is operable between 45 Hz and 18 kHz, has an impedance of 8 ohms, and produces 124 dB of maximum continuous SPL at one meter. Housed in a black-carpeted trapezoidal enclosure, measuring 26 1/2 inches tall by 18 1/4 inches wide by 14 1/32 inches deep with a 22 1/2 degree cabinet pitch, the loudspeaker features steel edges with integral rigging points and a built-in stand adaptor. A proprietary 12-inch long-excursion woofer is ferro-fluid cooled and is supported by a triple-spider and cast frame. It receives frequencies up to 200 Hz. Above this frequency, signals are routed to dual 6 1/2-inch horn-loaded midrange drivers until 1,800 Hz, at which point a single high frequency driver with a 1-inch exit and a titanium diaphragm takes over. Give it a listen by calling Community Professional Sound Systems, 333 E. Fifth Street, Chester, PA 19013; 708-367-8187. Circle EQ free lit. #107.

ENCORE ONCE AGAIN

Passport Designs has begun shipping Encore version 2.5, an upgrade to the popular composing and notation program for the PC and Macintosh. Version 2.5 retains the features of its predecessors while adding new fonts at no additional cost, plus a proprietary "Frets" font for guitar compositions. This font enables you to place guitar chord diagrams anywhere in the score. Encore for Windows is also fully compatible with Multimedia PC specifications. Notation can be displayed and edited right on the computer screen, and played back as an entire score or as individual parts. Encore supports 64 staves, and allows you to work on 16 files simultaneously. A score can contain multiple clefs, tempo changes, key and time signatures. Text and lyrics can be entered directly into the program. Take note by calling: Passport Designs, 100 Stone Pine Rd., Half Moon Bay, CA. 94019; 415-726-0280. Circle EQ free lit. #108.

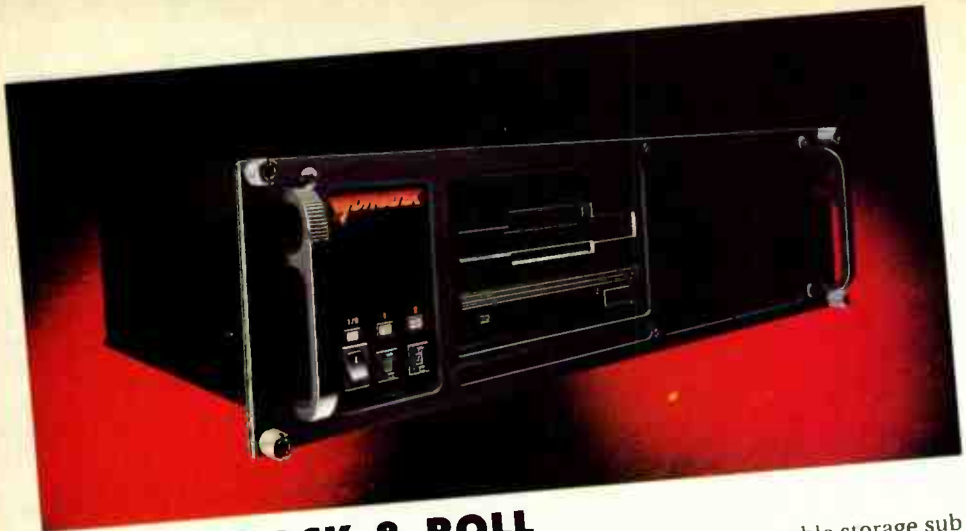
CUTTING EDDGE



Digital Audio Labs has begun shipping Version 2 of its CardD System's EdDitor program. Peak-reading VU meters have been added to the Recording screen and there is a new elapsed recording time counter as well as a time-remaining-on-the-disk counter. Fade and Cross-fade have been added to the editing functions and simple fade-ins and fade-outs are quick and easy to perform. More complex customized fades can be drawn, specifying the gain at each fade point. Linear and logarithmic fades are available. Cross-fades can be customized to create a smooth transition between any two recordings and automatic crossfades are used on cut-and-paste editing to provide click-free splice points. Markers can be placed anywhere on the edit screen and may also be dropped on the fly during recording and play-back. The Edit History feature lets you undo up to 10 edit operations and can be used to compare different ways of performing an edit. Accelerator keys have been added to reduce the number of keystrokes required to do an editing job. For more cutting-edge remarks contact Digital Audio Labs, 14505 21st Avenue, Plymouth, MN; 612-473-7626. Circle EQ free lit. #110.

PIECE OF MIND

Mark of the Unicorn's MIDI Time Piece II is a networkable MIDI processor that functions as an 8x8 stand-alone MIDI merger/router/filter and a SMPTE synchronizer. Four rotary encoders provide fast access to all programming parameters and they also double as MIDI control knobs. It performs many simultaneous functions and provides an 8-cable interface for the Macintosh with 128 discrete MIDI channels. Contact: Mark of the Unicorn, 222 Third Street, Cambridge, MA 02142; 617-576-2760. Circle EQ free lit. #109.



TRACK, RACK & ROLL

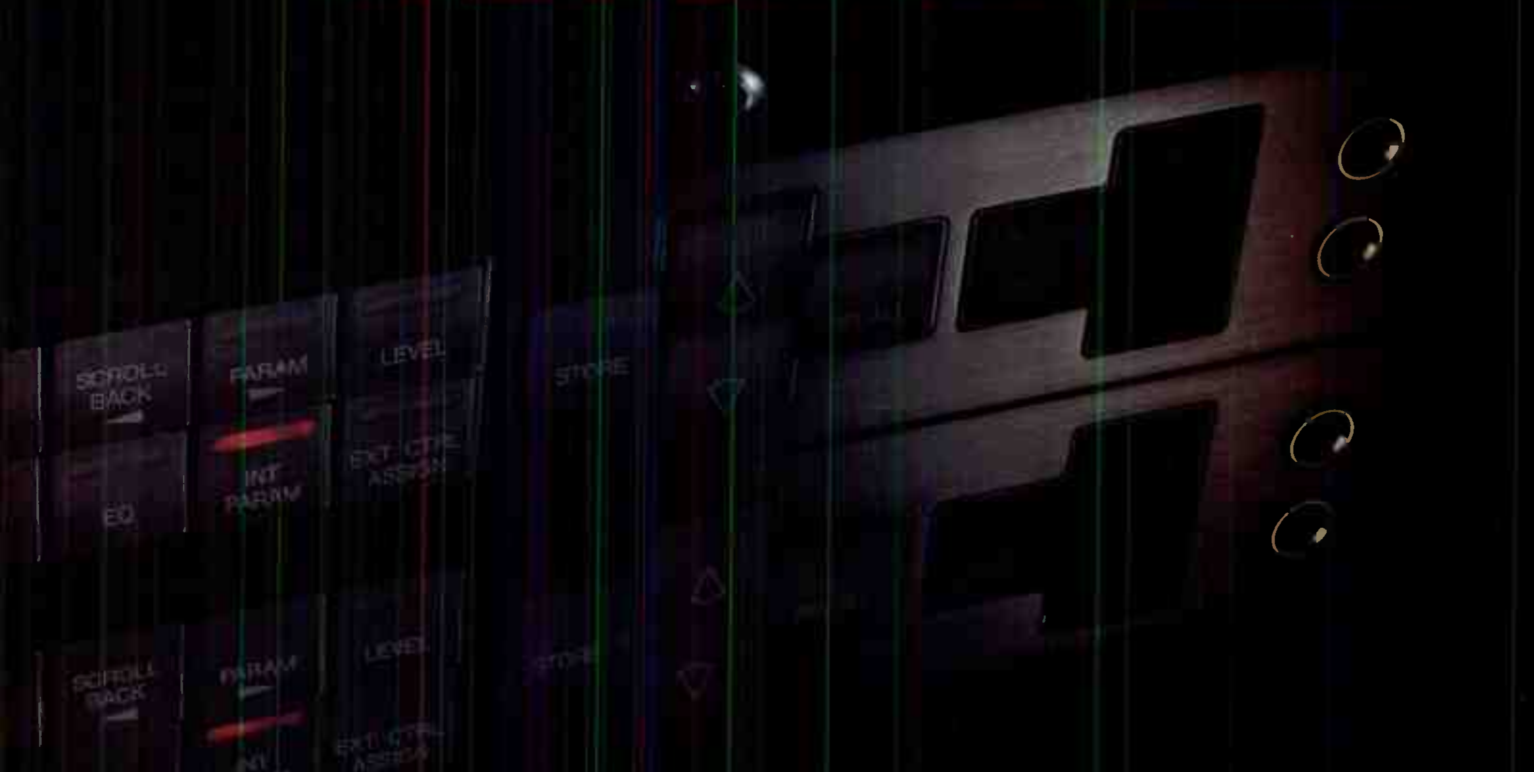
DynaTek Automation Systems' new TRACK series of rack mountable storage sub systems occupy 2 or 3 standard rack spaces and can contain either single, dual or triple storage devices. Multiple SCSI connectors allow the flexibility of a SCSI daisy chain configuration or independent operation of each device. Every system features "in use" indicator lights and analog SCSI ID display selectors for each device, all located on the front panel. Steel enclosures and steel rack supports distribute the weight of the system evenly and enhance durability. A unique mounting system provides full internal shock resistance and front-mounted handles, and quick release rack screws provide easy installation, removal and transport. Independent heavy-duty, high capacity, low noise fans and power supplies are incorporated to provide constant operating temperature and voltage. Only Hewlett-Packard, IBM, Fujitsu, ExaByte, SyQuest and Sony mechanisms are used in TRACK series construction. The TRACK series is compatible with Macintosh, IBM, Atari and Amiga computers. For more data (sic) contact DynaTek Automation Systems Inc., 15 Tangiers Road, Toronto, Ontario, Canada M3J 2B1; 416-636-3000. Circle EQ free lit. #111.

MINI MIDI MIXI

The Minimix mixing console from Ross Systems is a 19-inch rack mountable unit that offers the flexibility of a larger console. It has been designed to provide a large number of inputs in its compact frame. Sixteen selectable mic/line inputs, 2-band shaped shelving EQ, 4 auxiliary sends (2 pre and 2 post fader, internally selectable), +48VdC phantom power on all mic inputs, PFL headphone cue, channel muting, peak +10 dB headroom indicator and smooth 100 mm channel faders make up the incoming section of the board. The master section includes 4 auxiliary master sends, 12 segment bargraph display for monitoring, headphone level control, output mute switch and two high quality 100 mm faders. The 16x2 Minimix can be used for live sound reinforcement, project recording applications, keyboard or sub-mixing applications. Suggested retail is $995. For more information contact Ross Systems, 1316 E. Lancaster Avenue, Fort Worth, TX 76102; 817-336-5114. Circle EQ free lit. #112.

THEN THERE WERE THREE

E-mu's Proteus line now includes the "3," which emphasizes esoteric musical instruments and sounds from around the world. Along with ambiences such as "Netherworld" and "Undersea Life" is an array of effects and instruments, from nylon-string guitar, the Australian didgeridoo to panpipes, to a variety of percussion choices, various flutes from around the world, and oceanscapes — to name but a few. There aren't many standard Western instruments among the over-150 sampled voices on the 3. (More and truer string sounds and brasses are found on the "1" and the "2".) But if you're looking to enhance your rack with sounds of the world, or require ambient atmospheres for soundtracks for film or video, this unit will serve your needs. For more information contact E-mu Systems, Inc., 1600 Green Hills Road, P.O. Box 660015, Scotts Valley, CA 95067-0016; 408-438-1921. Circle EQ free lit. #113.



EQ PRODUCT SHORTS

Technical Audio Devices (TAD), a division of **Pioneer Electronics**, has installed a custom **TAD Studio Monitor (TSM)** that — this is really true — is housed in 40 cubic feet of solid, poured concrete at Bad Animals/Seattle's Studio X.These monitors can handle up to 135 dB SPL with high-quality, near-field listenability ◆ **Ac-cetera** has been granted a registered trademark for **RUBBER-NECK®**, a black no-creak, no-rust, no-glare, flexible extension (similar to a gooseneck). Four styles are offered. "S" series is solid throughout, ""H" series is hollow for internal wiring, "L" series includes a locking **XLR** and is ready for internal wiring, and the "X" series is internally wired from male to female **XLR** ◆ **The Linear 85 Absorber** is a frequency linear absorber panel of a proprietary design from **ARcoustics**. According to the manufacturer, a unique configuration solves the problem of too much high frequency absorbtion that is associated with most absorption panels ◆ A line of steel rack panels with a professional black finish have been introduced by **J.H. Sessions & Son**. They are available in 1,2,3 and 4 space blank panels, single space panels prepunched with XLR male, XLR female, 1/4-inch jack or any combination. Also available are 3 space panels to accommodate 1 and 2 rack space vent panels and 3 space panels to accommodate 1,2 or 3 fans ◆ Due to increased demand, **S&S Industries** has stepped up production of its **Stealth Trigger 7000** bass drum trigger. The manufacturer says that the ST7000's streamlined wing-like design makes it the most compact unit on the market ◆ **Applied Research & Technology (ART)** has begun shipping its **Phantom Series Mixing Consoles**. The 1608, 2408 and 3208 offer a wide array of pro features at an affordable price and they are designed to be at home in the studio as well as in high headroom live sound applications ◆ **Passport Designs'** new **Producer™**, is a software-based media integration tool that serves as a central "scheduler", integrating and providing time based synchronization of **QuickTime** from **Apple Computer, Inc.**, computer animation, graphics, titling, video decks, sound cards, MIDI instruments, digital audio, music, and multimedia software applications on the Macintosh ◆ **Community Light & Sound** has introduced the **RS Jr. MicroArray™** 2-way fully horn-loaded loudspeaker system that is designed to provide natural-sounding and intelligible voice reinforcement in flying arrays and to handle high SPL levels — all in a small package ◆ **Adamson Acoustic** has released the **F Series** loudspeaker enclosures, all of which feature passive crossovers built-in and are designed for use in small clubs and similar venues ◆ The new **BX Series** 12- and 15-inch cone drivers from **Celestion** utilize the latest materials and design technology and can handle up to 600 watts (average continuous pink noise). They feature three leg cast aluminum frames, fiber composite cones, and 2.5-, 3-, and 4-inch edgewound reinforced Kapton voice coils ◆ **Tube Works** has answered the prayers of many guitarists with the release of a genuine tube-enhanced reverb that is designed to fit in a rack-mount rig ◆ **DiscMakers** is now offering a seven-day express cassette package that includes free graphic design, test cassettes, art proofs, printing of one and two-color inserts, and retail ready cassettes for as little as $550.

"Intelligent design, dependable... flawless performance"

Frank Serafine, President Serafine Inc.

Frank Serafine exclusively used DynaTek Track Series Rack Mountable Data Storage in the production of

THE LAWNMOWER MAN™

Hollywood knows it can depend on Frank Serafine for creative sound design.

For digital data storage, Frank knows he can depend on DynaTek.

For years, Frank has used digital technology to produce sound effects for such innovative projects as "Tron," "The Day After," "Star Trek" and most recently, "The Lawnmower Man."

Low noise, ergonomic design and its vast selection of media has made DynaTek's "TRACK" series Frank's choice for rack-mountable data storage.

Whether for hard disk recording, sampling, or archive, DynaTek's "TRACK" series provides quiet, dependable performance, time after time.

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CIRCLE 30 ON FREE INFO CARD

EQ TIPS

EDITED BY DAVE BRODY

INSTANT INTONATION

The problem: you're sick and tired of guitar and bass players pulling their cords out (when you least expect it, of course) in order to plug their axes into those little tuning meters. But you don't want to leave the tuner hanging in the signal path while you're recording.

The fix: drive the tuning device off of a buss or an unused Auxiliary send. It can even be driven off of the output of the tape machine or the monitor mix. Most intonation meter boxes will accept a pretty wide latitude of level, but make sure the level isn't too hot (or you could end up "tuning" the unit's own harmonic distortion). By using a separate buss or Aux send you can tweak the input level for the most accurate reading.

From Tom Chianti,
Mix Engineer, Flying "Z"
Recording Studios, NY

SEND AND YE SHALL RECEIVE

Wanna turn an unused group of Auxiliary post faders into a pre-fader group in the mix without paying a fortune to have an internal jumper installed? With our Yamaha 1602 mixer, I simply use a standard 1/4 inch patch cord to come out of the Aux 2 (or Aux 3) post fader master output on the back of the board and patch back into the Aux Return input (of the Aux Send, Aux Return sub-group), also located on the back of the mixer. This allows me to send only what I want to tape via the Aux 1 output.

For example, I can record from a choir mic to cassette using (the pre-fader) Aux 1 output. Then, when a Rector speaks with his wireless mic, I bring up his fader, which will feed the house sound while I simultaneously keep the Aux 2 up on that channel. That sends the signal to the Aux 2 out and, presto, it's back to the Aux 1 mix. When I pull the wireless mic fader back down, the Rector is not only out of the house mix, but he has no fear of being recorded to tape while he converses with others, since Aux 2 on that channel is still post fader.

From John Stevens,
Audio Engineer,
Northern Kentucky University
Media Services

SCOPIN' THE SWEET SPOT

Many minutes of precious session time can be wasted in moving a mic around trying to find the desired placement for a given acoustic instrument. A short length of cardboard tube held up to your ear and pointed towards the sound source can help you narrow down the possibilities quickly. [Oh, the things we do to get the sound !] Think of this rig as a sort of audio telescope. The tube will, of course, impart considerable "honk factor", but you'll soon learn to compensate for this. A special variation on this theme for string instruments (especially guitars and acoustic basses) is the use of an unsharpened pencil with the eraser touching the body of the instrument — ask the performer's permission first — and the flat graphite end in light contact with your outer ear. Carefully fold over the pinna of your ear (that little flap of skin that sticks out from your head) to seal off your ear canal and touch the flat end of the pencil thereto. DO NOT stick the pencil (or anything else smaller than a basketball) in your ear!! Now gently hunt around the instrument with the eraser end. You'll be amazed at how a shift of a few centimeters can play hob with the frequency spectrum. Modern mics — "pencil" condensers particularly — can really spotlight an instrument. It's nice to know where to start pointing. EQ

Keep On Tippin':
EQ Tips EQ Editorial Offices,
939 Port Washington Blvd.,
Port Washington, NY 11050
Fax: 516-767-1745

Positively Brilliant

The new H3500 *Dynamic* Ultra-Harmonizer: It puts digital effects in a very different light.

Cutting-edge innovation is a long-standing Eventide tradition. Flawless pitch shifters, dense reverbs, lush stereo delays and rich choruses have made the Ultra-Harmonizer an essential audio production tool in recording studios, post suites and broadcast facilities around the world. More on the same line would have been enough for most people—but not for the restless minds of Eventide engineers. They've endowed the H3500 with revolutionary new DFX Dynamic Effects algorithms and presets. Plus 16 bit, 44.1 kHz sampling, and all the goodies that made the Ultra-Harmonizer famous.

DFX Dynamic Effects— A new direction in digital audio processing

The H3500 has 22 effects algorithms: Many have set industry standards for quality and versatility in areas like pitch shifting, reverb, delay effects and sampling. Now we're setting a dramatically different course for audio processing by adding two mod factory DFX algorithms. Each one includes a full set of independent processing modules, from delays and filters or pitch shifters to envelopes, modulators and mixers. The modules can be patched together in any combination. Factory presets range from "ducked" delays (echoes that only appear between vocal or instrumental phrases) to choruses, flanges, even reverbs that respond instantly to a musician's touch. The H3500's DFX processing takes a dramatic new step beyond static effects, one that can bring tracks, performances and mixes to brilliant life.

Full Sampling Capabilities

For looping rhythms, flying in backup vocals or replacing snares, nothing's faster, cleaner or easier than an Ultra-Harmonizer: Ask the leading engineers and producers who use one every day. The H3500 gives you all the power of Ultra-Sampling right out of the box: It has an internal sampling board, our most advanced sampling software, and either 23.8 seconds mono (11.7 seconds stereo) or 95 seconds mono (47.5 seconds stereo) of memory. Eventide's world-renowned pitch change technology gives you freedom no other sampler can match. Change playback time on the fly—without changing pitch. Or retune the sample from the front panel or MIDI, without changing playback length. Even access sample memory from mod factory algorithms for long delay loops with beat-per-minute timing. Of course, you can also use the H3500 as an ordinary sampler.

Limited Availability

Clearly, not everyone can handle this much Ultra-Harmonizer power. That's why the new H3500 *Dynamic* Ultra-Harmonizer will only be available from a select group of Eventide dealers. Before you can explore the new direction in world-class multi-effects, you may have to do some traveling. But we assure you, it's worth the trip.

Sample responsibly: Credit and compensate your sources.

One Alsan Way • Little Ferry
NJ 07643
Tel: 201-641-1200
Fax: 201-641-1640

Eventide
the next step

I currently have nine CD-ROMs of samples, which basically means that if I need a sound, it's probably in there somewhere.

will threaten your piece of the pie. Getting more women into audio should increase the overall size of the pie; if we have more women artists, more women engineers running more studios, and more women executives at record companies, we'll have a growing business instead of a stagnant one. That should mean more opportunities for all.

THOSE NEW-FANGLED RAM RECORDERS

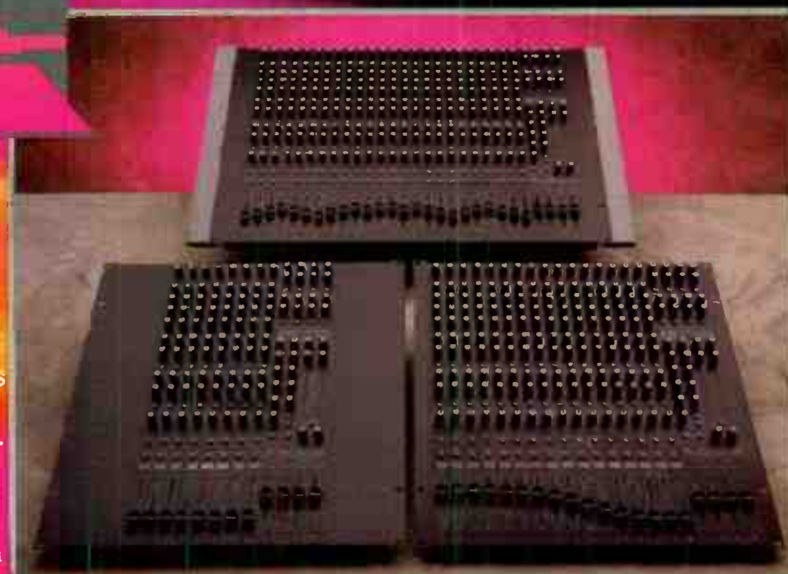
First there was tape, then hard disk recording. The next logical step, RAM recording, would eliminate moving parts altogether. The bottleneck has been the high price of RAM, which is why the more cost-effective hard disk is still king of the hill.

You may already own a RAM recorder — only thing is, it's disguised as a sampler. For example, the Peavey SP is expandable to 32 megabytes of RAM with 4-megabyte SIMMS (this provides about 6 minutes mono/3 minutes stereo recording), and the Kurzweil 2000 is expandable to 64 megabytes with 16-megabyte SIMMS. That's enough memory to do some serious digital audio. Even samplers with 2, 4, or 8 megabytes of memory can do a lot.

Samplers give a good preview of what's to come when RAM recording comes on-line for real: lots of tracks, each with variable length and processing; instant playback because no head has to pull information from moving platters; and simple sample editing by shuttling these files to a computer editing program.

Of course, there's still the issue of backup. Digital tape backup or optical-storage removable cartridges are the preferred way to go, but you can squeeze by if you play the sampler output into a DAT. Then if anything happens to your primary storage medium, you can always sample off the DAT. This will degrade the fidelity a bit, but at least you won't have to recreate the sound from scratch. EQ

A.R.T. SHATTERS THE PRICE OF PROFESSIONAL MIXING TECHNOLOGY!!!



THE PHANTOM

PROFESSIONAL SERIES CONSOLE

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

- 16/24/32 CHANNEL VERSIONS
- ULTRA LOW NOISE LINE AND XLR BALANCED MIC INPUTS
- 8 AUXES
- 8 DIRECT OUTPUTS
- 16 PATCH POINTS
- 4 SUBGROUPS
- OVER 20 MIX OUTPUTS
- COMPREHENSIVE 4-BAND EQUALIZATION
- PHANTOM POWER
- MAIN-SUB SELECTOR SWITCHES
- 8 CHANNEL DEDICATED TAPE RETURN SECTION
- BOTH PRE AND POST-FADE MONITOR AND AUX SENDS
- CLIPPING INDICATOR LIGHTS EACH CHANNEL
- FULL CHANNEL SOLOING
- FULL CHANNEL MUTING
- LONG TRAVEL SHIELDED FADERS
- SWITCHABLE OUTPUT METERING
- SWITCHABLE MONITOR METERING
- ISOLATED 2 TRACK MONITOR TAPE RETURN
- SEPARATE STEREO CONTROL ROOM OUTPUT
- ASSIGNABLE TALKBACK-MAIN, MONITOR, AUX.
- INDEPENDANT SOLO LEVEL
- SEPARATE HIGH POWERED HEADPHONE OUTPUT
- RUGGED ALL STEEL HEAVY DUTY CHASSIS
- SLEEK,WOOD END PANELS (OPTIONAL)
- LINEAR CAPACITIVE ULTRA LOW NOISE CIRCUITRY
- HIGH-HEADROOM INPUT PROCESSING
- ADVANCED INPUT/OUTPUT COMPARATIVE DESIGN PROVIDES EXTREMELY WIDE DYNAMIC RANGE
- SEALED BODY LOW NOISE POTS
- 100% R.F. SHIELDING VIA METAL ENCLOSURE
- HEAVY DUTY HARDCONTACT SWITCHES THROUGHOUT
- COMPUTER CONTROLLED PRECISION SOLDERING
- PRECISION AUTOMATED ASSEMBLY INSURES CONSISTENCY
- MIL-SPEC GLASS EPOXY CIRCUIT BOARDS
- HIGH FIDELITY EQUALIZATION CIRCUITS
- NEW DESIGN AND ASSEMBLY TECHNIQUES PROVIDE THE TRANSPARENCY OF CONSOLES COSTING THOUSANDS OF DOLLARS MORE.
- AUTOMATED ASSEMBLY AND COMPUTERIZED TESTING MAKE THE PRICE UNBELIEVABLY LOW.

ART

THE SOUND OF PERFECTION

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CIRCLE 66 ON FREE INFO CARD

I Had a Secret

Out in Woodland Hills, Ventura Boulevard gets as loud and tawdry as a twenty-dollar manicure. But just a few blocks south, the terrain changes abruptly. All of a sudden, you're in a genteel, tree-lined neighborhood. Down one of these shady suburban lanes stands a picturesque, rambling ranch house, well appointed, with fieldstone arches, antique gas lamps, leaded Victorian windows — the works. It's hard to imagine this idyllic spot swarming with inspectors, police officers pounding on the front door, search warrant in hand. But that's just what happened on April 3, according to the house's owner, Charles Sanford. See, this piece of prime residential real estate is the location of Secret Sound — the focal point of L.A.'s home recording controversy.

Photos by Ed Colver

GIVE YOURSELVES UP!

Sanford and his wife were out of town when L.A. police and inspectors from the Department of Buildings and Safety came to his door: "I had a housesitter in to take care of the pets," Sanford says. "When she came to the door, they pushed it open, shoved the warrant in her face and barged into the house. The same thing happened at Hank Sanicola's place [O'Henry, another residential studio in town]. The police came up and blocked both entrances to his driveway. Guys ran up to his house with a search warrant. They raided Eddie Van Halen, Steve Vai and Jay Graden's home studios too."

Everyone who reads the pro audio press knows something of Sanford's ongoing conflict with H.A.R.P. Formed in 1989, the Hollywood Association of Recording Professionals is a consortium of people who own the top recording studios in town. Sanford is a songwriter, solo artist and producer/engineer. His credits

As in easy to use. As in no restrictions to your creative flow. As in the Alesis MMT-8® MIDI Recorder, the world's most popular hardware sequencer three years running.

If you're like most songwriters we talk to, a powerful software sequencer can sometimes be intimidating to work with, and can actually get in the way of your music. Songwriting is hard enough. Your sequencer should make it easier.

We designed the MMT-8 so just a hit on the record and play buttons puts you in songland. Intuitive, quick and easy.

The MMT-8 is the hardware sequencer for everybody. Whether you're a beginner or an advanced power user with a 400 meg hard drive.

Use the MMT-8 as a powerful notebook sequencer. It's got all the features and editing you need for your music. And now with our new MMT-8 Music Link Software™*, power users can Sysex their work over to their software sequencer for full screen editing. Stand alone or in conjunction with a computer system, the options are inspiring.

Still the best. Still only $299.** Buy an MMT-8 at your Alesis dealer today. It's that simple.

Now in basic black.

** Call 1-800-5-ALESIS for details*
**Slightly higher in Canada ®Macintosh is a registered trademark of Apple Computer, Inc.*

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CIRCLE 07 ON FREE INFO CARD

ALESIS
STUDIO ELECTRONICS

The Art of Drum Replacement

The grass is always greener on the other side of the fence. The woman is always cuter on the other side of the room. The snare drum on the other guy's record always sounds better than the one you just recorded. The never-ending quest for that extra ten percent.

Of electronics, acoustics and the search for the ultimate drum sound

BY ROGER NICHOLS

Drums and other percussion sounds add to the basic feel of a song, providing the fundamental rhythm on which all the other instruments rely. The three important factors here are the pattern, placement (in the pattern) and sound of the percussion instrument. All are interrelated to the point where any change in one greatly affects the others. A perfect scenario for endlessly chasing your tail.

In the MIDI studio, we are blessed (or cursed) with having an unlimited number of percussion sounds available to us. If we don't have just the right one, we can record it ourselves, and add it to the collection. I have 400 megabytes of edited drum samples, and at least four hours of DAT tapes that haven't yet been edited. It now takes longer to audition all of the drum permutations than it does to do everything else combined.

Usually, when you start to write a song in your MIDI studio you use a drum set that already exists or create one that has only the bare essentials needed to get you started. The initial groove seems fine. As you progress, you keep telling yourself, "This is OK for now, but I'm gonna fix that snare and kick as soon as I get a chance." Well, you've put it off as long as possible.

FACE THE MUSIC & DANCE

If you are still working in the sequencer domain, you have an advantage: all you have to do is substitute different sounds while the sequence is playing — almost. Since we know that the sound has a lot to do with the feel of the pattern, we have to choose a drum sound that appears to come in the same place, rhythmically, as the drum we are replacing. You can put two sounds side-by-side on an edit or screen and it may appear that the peaks are in the same place, but when you listen to the sounds it is a whole other story.

One of the snare drums could sound early because of its length. Where the sound ends makes you think that it starts in a different place. If the sound is too short, it tends to sound early. If the snares ring longer, it makes it sound as though the snare is more laid back. This effect is more pronounced with sounds that are processed. Processed snares and other drum sounds use the decaying part of the sound to accentuate the effect.

Illustration by Normal Rockhard

CR-1604 FANATICAL ENGINEERING PART 1

Before you buy any brand of 16-channel mixer, consider why the CR-1604's become the rack-mount mixer of choice for top pros (request our literature packet and we'll include an astonishing list of touring and session players, studio producers, engineers, songwriters, sound track composers and late-night talk show bands who rely on Mackie).

Exclusive design features. One reason we specialize in mixing boards is so we can be maniacally meticulous about details others overlook. Some details help make the CR-1604 rugged and roadable. Others contribute to the mixer's excellent sound quality. And some, like our unique gain architechture and mix amp topology aren't really details at all, but rather represent a fundamentally better way to mix music.

Fanatical engineering means phenomenal sound. Compare our specs and features with anyone. But remember that the bottom line is what your ears tell you. Put a CR-1604 through a hands-on work-out and you'll understand why several major-label CD releases have been recorded and mixed solely on Mackie mixers.

OUTPUT-O-RAMA: Main stereo outputs with enough oomph to drive any balanced or unbalanced input. In fact, all outputs have sufficient level to drive any input. Naturally all outputs/inputs are fully protected.

TRIM matches any signal from instrument levels to −10dBm semi-pro to +4dBu pro gear.

7 SENDS via 4 knobs with plenty of gain above unity for special effects and center detent at unity gain. **4 STEREO AUX RETURNS** with super-high headroom, low noise and enough gain to work with all levels.

MUSICALLY-USEFUL EQ. We're amazed at how many users tell us that our 80Hz, 2.5kHz and 12.5kHz turnover points and EQ circuit design not only sounds sweet but in some cases have replaced outboard equalization.

STUDIO-GRADE MIC PREAMPS. Only the CR-1604 incorporates discrete, large-emitter geometry transistor preamplifiers instead of just simple ICs. That's why only our mic preamps deliver −129dBm E.I.N., 0.005% THD yet can handle +14dBu max inputs without need for a pad. Whether you're miking a 22" bass drum or a nose flute, you'll get all the punch and delicacy of expensive studio board preamps.

LARGE INTERNAL POWER SUPPLY provides power for +48V phantom mike power, 12V BNC lamp socket, and bi-polar power for main circuitry, hi-output headphone amp and sophisticated enhancements such as our XLR10 mic preamp expander & internal full-fader/muting MIDI automation board (late summer availability). Our supply not only eliminates inconvenient "wall warts & line lumps" but gets rid of their hum-inducing magnetic fields, too. They're typically driven into non-linearity (1500 Gauss or more) causing stray 35-350V magnetic fields that are easily picked up by shielded audio cables. Our own ideal, instrumentation-grade transformer loafs at 1000 Gauss, reducing stray fields to under 1V for quiet operation.

COLD-ROLLED STEEL CASE

BUS 3&4 ASSIGN

GNARLY CIRCUIT BOARDS. Unlike brittle phenolic, our horizontally mounted, double-sided, through-hole plated fiberglass boards and rigid multi-point suspension prevent damage from external impact.

MIX AMPS WITH TWICE THE HEADROOM. Signals from all inputs simultaneously converge here. Conventional mix amps can overload and distort when you start pouring on the channels. Our gain structure solves the problem for audibly better sound. Cram the CR-1604 with 16 **hot** signals and it still has more headroom than other mixers running 8 inputs. No wonder ours is the mixer of choice among top percussionists like Pablo Face (C+C Music Factory) and Bashiri Johnson (Whitney Houston).

STEREO IN-PLACE SOLO maintains stereo perspective for all soloed channels and returns.

UNITY PLUS special-taper faders with 0dB gain at center detent minimize noise, maximize headroom, provide 20dB more gain above unity to reduce constant trim re-adjustment as input levels change.

BETTER-THAN-DIGITAL. −90dB S/N ratio with 108dB dynamic range (vs. digital with −90dB S/N and 90dB dynamic range). Incidentally, when comparing specs, make sure they reflect real world conditions as ours do. We're so fanatical that we include a short White Paper on Mixer Specifications with our lit packet. Read it to appreciate the fanatical engineering that went into the CR-1604's specs. And why we differ vastly from mixers that merely masquerade as a Mackie.

Silky smooth **SEALED ROTARY POTENTIOMETERS** prevent contamination from dust, smoke, liquids and are CO-MOLDED without the use of easily-shattered phenolic mechanical parts (a major reliability problem in many conventional mixer designs). Plus, our exclusive **ENERGY-ABSORBING KNOB DESIGN** transfers vertical impact away from pots and main circuit board, greatly reducing possible damage. Three more reasons we offer a generous 3-Year Limited Warranty.

ULTRA-DENSE CIRCUIT TOPOLOGY. For example, unlike budget mixers which cut corners with passive circuitry, we added input and output buffers on the CR-1604 is compatible with a much wider range of devices. This ultra-miniaturization was an engineering and manufacturing challenge. But whaddaya expect from fanatics?

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This makes the timing of the sound's decay as important as the sound's attack.

Placement of the new sound is critical. If you want to make the sound happen earlier, you can make the sequencer shift it ahead by a slight amount. This amount usually has the resolution of the smallest increment available to that particular sequencer. If the resolution of the sequencer is 480 ticks per quarter note, then moving the snare one tick earlier at a tempo of 120 beats per minute will advance the snare by 1.04 milliseconds. This is, of course, the best case scenario. Most sequencing software won't have this high resolution and a lot of dedicated hardware sequencers don't allow adjustments of one tick. Moving the snare one increment earlier on an Akai MPC60 will make the snare about 13 milliseconds ahead of where it was.

The way Donald Fagen works is to run the audio from each drum and sequenced instrument through individual digital delays. He can then move any instrument by any amount. Most of the delays he uses are made by Roland and allow increments of .1 milliseconds up to 10 milliseconds, where the resolution drops to one millisecond increments. If he wants to have available .1 millisecond increments above 10 milliseconds, then he puts two delays in series. There are delays available that allow increments of around 20 microseconds (which is 1/50 of a millisecond). Donald has a stack of these, as well.

I am not saying that these are necessary devices for everyone. If you put together a sequence and everything sounds right, then that is just fine. If you bust your hump trying to get a machine groove to feel good and you aren't having much success, then try moving a few things around until it locks into the groove you're looking for. Put every instrument through its own delay and crank them all to 15 milliseconds. Now you can make some sounds earlier and others later.

The biggest problem with using all of these delays is that they do degrade the quality of the sound passing through them. They also add little whirs and wheezes of their own to the sound. If the sound the delays add bothers you, then there is a solution.

When you are ready to print the sequence on multitrack tape, stripe SMPTE on the tape and then use the SMPTE to drive the sequencer. Make sure you set the start offset so that the sequence starts where you want. Never start the sequence at 0:00:00:00. Always let at least 10 or 15 seconds of SMPTE go by before the sequence starts.

Now, let's pretend for this example that we have a 5.7 millisecond delay on the snare drum and a 2.1 millisecond delay on the kick drum. Take the snare drum audio out of the delay and run it as you normally would to the tape machine. Insert the delay on the SMPTE signal between the tape machine and the SMPTE reading device that is driving the sequencer. Make sure that the SMPTE is not clipping the delay. About half way up the meter should be fine. Make sure that there is no regeneration or any other effects active on the delay. We want straight, no frills, delay. In the case of the snare drum, make sure that the delay is set at 5.7 milliseconds. Start the tape, punch up RECORD on the snare track and let the sequencer print the snare drum. If you have other instruments that were set for the same amount of delay, you can print them, too. After the snare is printed, do the same thing with the kick drum, but change the SMPTE delay to, in our example, 2.1 milliseconds. Continue with this method until all of the instruments have been printed to tape.

See how easy this is? Maybe some day we will have Super-MIDI and sequencers with high enough resolution so that we won't have to do it this way, but, let me tell you, we've come a long way since 1978. There were no sequencers to speak of then, especially universal ones that you could use with any synthesizer. Everything we sequenced was based on the timing of the one-eighth-note click on tape. I guess the resolution was two ticks per quarter note. We had to build our own delays to get small increments. It worked, sort of.

SNARE DRUM REPLACEMENT

OK sports fans, what if you already put everything on tape and your ex-lover got your sequencer in the palimony settlement and you need to replace that ugly snare drum with one that doesn't rip the top of your head off every time you play it back? Well, that's just what we're going to cover next.

There are basically two ways to replace or add to an existing snare drum:

1. Build a tempo map and play it back to trigger the new snare.

2. Trigger the new snare in real time from the existing snare.

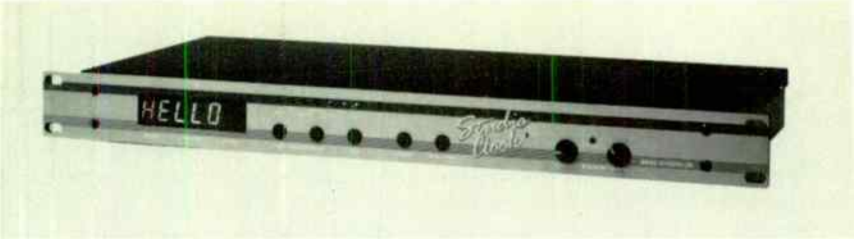
Right here, let me say that if the drums on tape were played by a real live human-type drummer, I would prefer to add the sound of the new snare to the sound of the existing snare, rather than to replace it entirely. This greatly improves the chances of the finished track sounding as though a real person played it.

TEMPO MAPS

If the snare drum you are replacing plays a regular pattern, such as two and four, all the way through the song, then you can use a wide variety of devices to create your tempo map. These include, but are not necessarily limited to, the Roland SBX-80, the Roland SBX-1000, the Aphex Studio Clock, Syncman Pro and Opcode's Studio 5. Each unit operates a little differently, but basically they read SMPTE to figure out where they are, and you feed them audio from the snare drum so that they can determine where the beat fell. Some units have the ability to mask out unwanted sounds that occur between the snare beats. If the unit that you are using does not have this capability, then you may have to gate the snare drum or otherwise process it to mask out the junk.

Next, write a simple pattern into your sequencer that will play the new snare drum on every quarter note. This is because the tempo mapper thought it was hearing quarter notes when the snare was fed to it and that the first snare it heard was the downbeat of the song. Make sure it is quantized to play exactly on the beats. Hook up the device used to make the tempo map so that it reads SMPTE from the tape machine and sends MIDI song pointer to your sequencer. Set the sequencer to "External MIDI" and play the tape.

In reality what is happening is that the sequence is being played back at a variable tempo. Let's say that the tempo of the original song was 120 beats per minute. The tempo displayed by the sequencer when playing back should read 60 beats per minute (because of the half note snare part). When we get to a bar where the snare drum was rushed, the tempo will jump up to 60.13 bpm or 61 bpm. If the next snare beat was right on the money, the tempo will drop down to 59 bpm so that the average comes out



DOES ANYBODY KNOW WHAT TIME IT IS?: The Aphex Model 800 Studio Clock

to be 60 bpm for the duration of the song.

Now that we have the new snare playing along with the old snare, we have to determine if the new snare is in the right place. If you listen to both snares at once and you like where the new one lands without hearing any flams, then cool. Print it. If you are not exactly sure whether the new one is right, then there are a few different things you can try to see how they line up:

One method is to make the new and old snare drum be at the same apparent level. It doesn't matter what the meters say, just make them sound as though they are both at the same volume. Now pan them to opposite speakers. If the snare appears to come from the right speaker, then the snare panned to the right is earlier than the one panned to the left. If the image seems to come from between the speakers, then they are probably happening at the same time.

I have seen a few people use the Russian Dragon to determine which snare is early. The only problem is that it is not calibrated, so you can't tell how early or late one input is to the other. The Bee Gees used to record the new snare on the multitrack tape and then, by scrubbing the tape across the heads (like finding an edit point), mark both snare drums and measure the difference with a ruler. At 30 inches per second, 1/4 inch would be about seven milliseconds. I use a dual-trace oscilloscope to see the difference between the events.

If the new snare is late, because of MIDI slop or the difference in sound quality or whatever, the same methods used earlier to shift sounds around can be employed. Shift it to earlier by using either the sequencer or the offset in the tempo mapping device, and then run it through a delay to line it up. Then move the delay over to the SMPTE line, and print it to tape.

If the snare part is not as straightforward as in our first example, we can resort to another method of mapping where the snares lie. (Where the snare lies? If only chickens lay, should we call it *lieback* instead of *layback*? Never mind.)

There are a few devices around that will produce a MIDI event when given an audio trigger. The ones that I have used over the years are the Simmons MTM (MIDI-Trigger-MIDI), the Akai ME35T and, most recently, the Alesis D4 drum module. Some of them have internal processing to mask out the junk, while others make you do it yourself. Whenever you get the masking straightened out so that as few stray triggers as possible exist, play the tape, record the MIDI events into your sequencer and delete any of the bogus events from the sequence. An added advantage to using this method of mapping the snares is that you can record the dynamics of the live snare. The MIDI devices I just mentioned follow dynamics rather well.

All audio trigger-to-MIDI devices have delay associated with this

I was shocked when I got the call from Hafler to do an ad!

Usually manufacturers want some big name producer or **mixing** engineer with a lot of big name credits. Rarely do they want a **technical** engineer to endorse a product. Well, this ad proves I'm wrong. I've been using Hafler amplifiers since first cutting my teeth in the recording industry. Over ten years ago I started using the 200's at the Record Plant as headphone amplifiers. I was quite surprised at how good they performed and sounded. I moved to Capitol Records and started using the 500's to drive the studio monitors in Studio B and Studio C. We put the 200's on the nearfield monitors which most engineers reference to. When it came time to rebuild the world famous echo chambers at Capitol Records, naturally my choice was Hafler amplifiers. Then I designed MCA recording studios and the Uni Manufacturing Plant. I chose Hafler amplifiers exclusively to drive their speaker systems as critical listening is a must for final QC product.

One might ask why I chose Hafler, when with the budgets I've had I could have spent thousands more on esoteric amplifiers. The answer is simple. ***I think for the money spent, these are the finest amplifiers obtainable.*** End of story."

Pat Weber, Technical Engineer
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TECHNIQUES

SET THE TEMPO (MAP): Opcode's Studio 5

process. The MTM has about a five millisecond delay between the time the audio event happens and the MIDI event is transmitted (all three bytes, channel, note and velocity). The Akai ME35T is pretty quick, with only two milliseconds delay. The Alesis D4 delay depends on which trigger type is selected. If you use one of the low trigger types, intended for triggering from pads, the delay is between three and four milliseconds. If you use types 21 through 25, the ones that Alesis recommends for triggering from analog tape, the delay is between six and seven milliseconds.

You have to remember, though, that the delay amount doesn't really matter, so long as you figure out what the delay is for your unit. You are going to compensate for it when you print the new snare, anyway. Once you have your map cleaned up, shift it earlier by whatever the delay is. If your sequencer shifts the sequence earlier by 13 milliseconds, and the delay in the D4 is six milliseconds, then crank seven milliseconds into the delay fed by the new snare. Everything should line right up. Do a little delay shuffling to compensate for the new sound, move the delay over to the SMPTE line and print it.

REAL TIME TRIGGERING

This means that you don't want to build a tempo map into a sequencer, you just want to play the tape, feed the snare drum into something, and come out with the perfect snare sound. OK, but if you figure it out I want to be the first to know how you did it.

All trigger units have delay, some just have more than others. If you are going to use the original snare in addition to the new snare, then some delay in the triggering may not matter. If the initial attack is used from the original snare and the added snare is used for the meat of the sound and doesn't really have a sharp attack of its own, then a delay of a few

continued on page 108


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Chick Corea's Keys Unmasked

When Chick Corea called and asked me to design a Macintosh-based keyboard setup, I was faced with several obstacles. The first obstacle was to determine Chick's exact needs. He knew what he wanted, but felt it would be very difficult to accomplish. In the past, he would have had at least three different setups or variations thereof: one for live, one for studio, and one for writing.

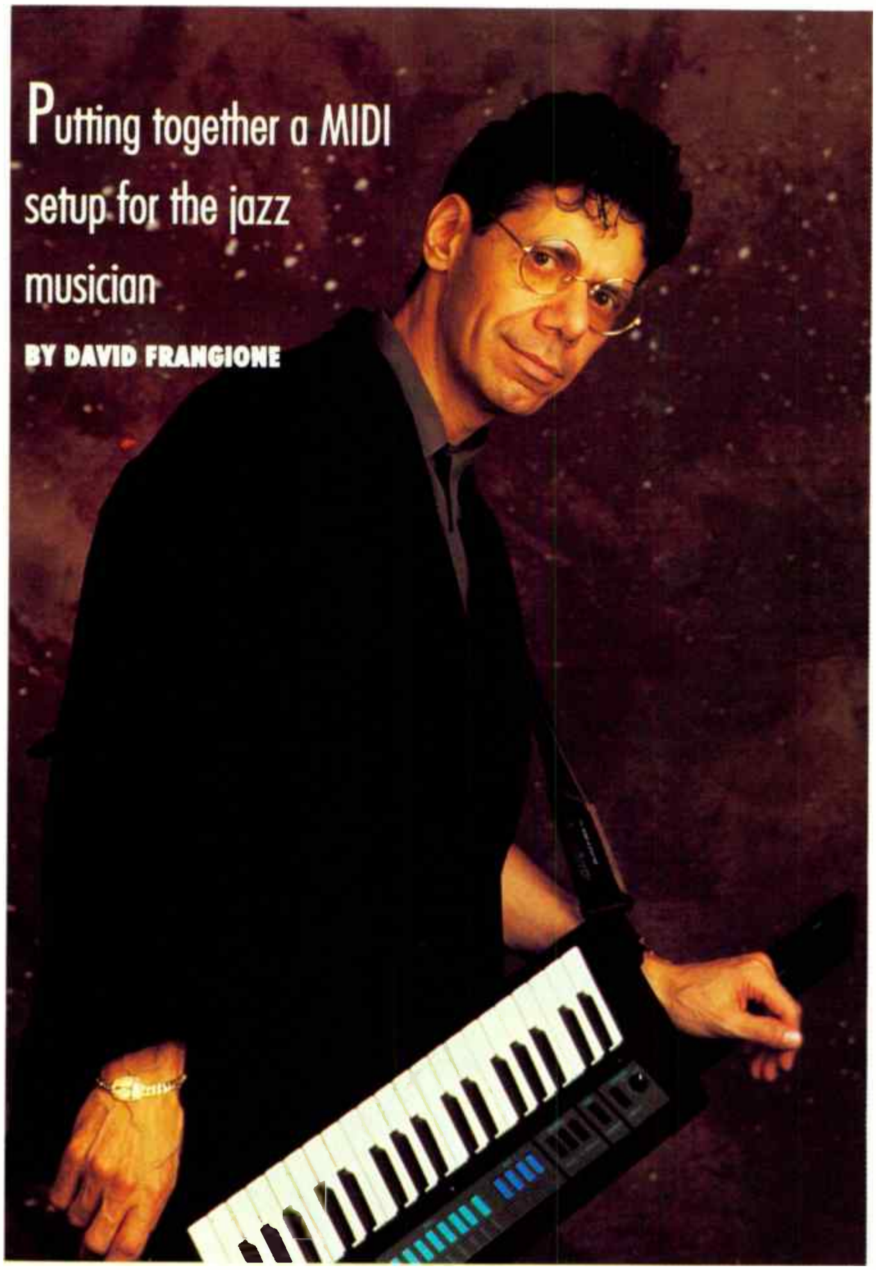

Photo courtesy of Chick Corea Productions

Although Chick and his crew had done a great job streamlining the multiple setups into one, it was still inefficient to repatch, move pieces of gear and rethink the entire rig for each musical application. I needed to come up with something very flexible that would be able to accomplish anything Chick wanted, with little, if any, modification.

My goal was to design one setup that could be used live, at home, in the studio, or in any combination. After sorting out Chick's needs, I contacted Mick Thompson, Chick's keyboard technician, to discuss further requirements. He explained that because the timing of this new setup coincided with a very tight touring schedule, major programming and wiring changes would have to wait until the end of the tour. We had to move fast and efficiently to ensure that the new system would be powerful, yet easy to use. Therefore it was important to be able to continue using the MIDI Mentor processor. The Mentor is a 6 x 6 MIDI patchbay/ processor with many unique and powerful features, including the ability to set up 32 synths at the press of a footswitch and to scale incoming MIDI volume. Chick had been using one for several years and wanted to continue using it in his current live setup. In the essence of time, Chick, Mick, and I agreed that, because of all of the potential reprogramming, the Mentor would stay, for the present.

INS 'N OUTS

Looking over my notes, I figured that 14 or more MIDI inputs and outputs would be required to be able to quickly route any device. Along with that, as I stated above, it would be essential that Chick be able to effortlessly sequence, edit, and change between different environments such as live sequencing or studio sequencing (which could have different synths in each setup).

STUDIO 5 SAVES THE DAY

Because the Macintosh was the front end of Chick's system, Opcode's new Studio 5 Macintosh interface offered

The other guys missed the bus. Actually, 2 of them.

Also available as 822XL and 1222XL table top versions at the same prices.

Why settle for a simple 16 channel stereo mixer, when you can have the 1642 4 bus mixer for $1099.

Or if 8 or 12 channels are enough, you'll find most of the same features on the 822 and 1222 stereo mixers, starting as low as $429.

With four similar discrete outputs (Sub1, Sub 2, Left and Right), the 1642 may be used as a 16x4 for recording, or, by assigning the subs to the L/R outputs, as a 16x2x2 for sound reinforcement. Each channel of the 1642 has a 3-band EQ with Adaptive-Q circuits for a contoured response that adds clarity to the highs and warmth without "boomy-ness" to the lows.

The 822 and 1222 models are offered in two versions, designated XL (table top) or RM (for rack mounting). Each comes with phantom power, 3 bands of EQ, 1 monitor and 2 effects sends, and monitor & headphone outputs. We're so confident that you'll love our new mixers that we'll back them with a two year warranty. So if you want some great mixers with low fares, don't miss the bus. Check out the 1642 and the rest of the DOD lineup at your favorite Dealer now.

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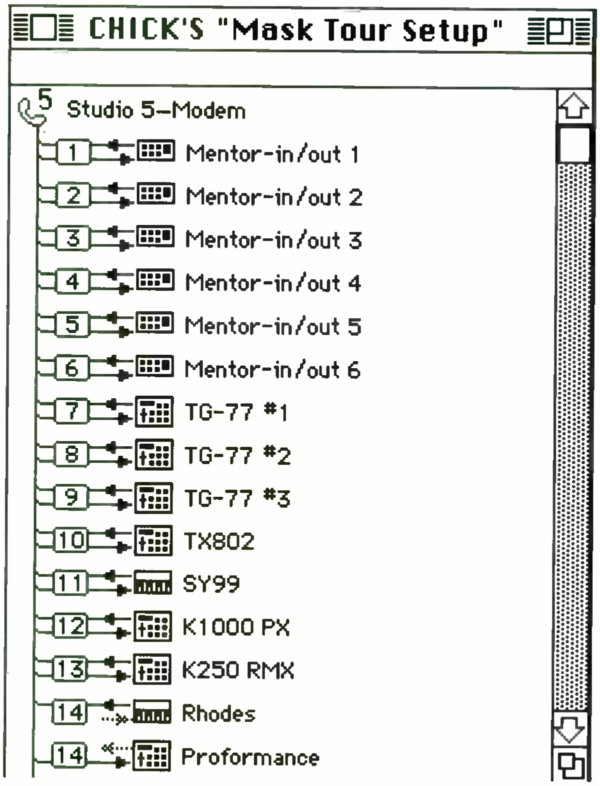


Figure 1: The Opcode Studio 5's menu lets Chick choose whichever keyboard he wishes to use.

the answer to all of our needs. Deciding on the keyboards to be used (displayed in Fig. 1) was very straightforward, as Chick already had a synth setup with which he pretty much wanted to stay. This new system, however, would easily allow him to add or subtract synths at will. The Studio 5 is a 15 x 15 Mac-to-MIDI interface with very powerful MIDI processing, SMPTE to MTC, and a built-in MIDI patchbay. Each of the Studio 5's 15 MIDI ports are completely independent of one another, providing 240 MIDI channels. It also has an emulation mode where it can invisibly act as an MTP (MIDI Time Piece) or a standard interface (like the Studio 3) in programs that do not yet directly support the Studio 5.

The Studio 5 works along with Opcode's OMS (Opcode MIDI System) to form a fully integrated working environment. OMS learns which devices are in your setup and tells the Studio 5 and all OMS-compatible software which devices are controllers, multitimbral, sending or receiving sync, etc. OMS also instructs each program as to where the different devices are located, freeing the user from any concern about where and how devices are routed to one another.

OMS served as the central "brain" of Chick's rig. The Macintosh would be configured with a Studio Setup (that's OMS lingo for your device routing) and a patch list corresponding to the proper Studio Setup (see Fig. 2 to understand how OMS configures the routing). Each patch could reconfigure the entire routing for the setup. To conform with Chick's previous live setup, we routed the six ins and outs of the Mentor into the first six ins and outs of the Studio 5. We then hooked up all of his other synths and controllers to the remaining ins and outs. The OMS Setup was named "Mask Tour 1" to designate that this matrix corresponded to the "Beneath the Mask" tour.

Our sequencer of choice was Vision and our Editor/Librarian was Galaxy Plus. Both Vision and Galaxy Plus created an instrument list based on our OMS Setup. This allowed Chick to choose any device, sequence with it (from any controller), edit it, store any changes, and have access to 240 MIDI channels (far more than were needed, but who's counting?). Because the Studio 5 can store 128 patches, I programmed a patch for Sequencing/Editing (patch 1) and a patch that routed the Mentor to all devices (patch 2), just as it had been before this change. Therefore, all Chick had to do was choose Preset 1 to sequence or edit and choose Preset 2 for the live show (Mentor controlling the show). We still have 126 patches available for use when Chick gets off the road and switches over from the Mentor.

As far as MIDI processing goes, the Studio 5 does almost everything that the Mentor can do (and more), including scaling, mapping, and sending multiple program changes. The elegance of this system made the change completely painless (which excited Mick) and, because all of Chick's programs support OMS, they worked very smoothly together, leaving Chick to think about only one thing: music.

CHICK'S SCHTICK

In the past, I would have needed a MIDI patchbay (with lots of ins and outs), several different unrelated software programs, and printouts of which devices were going where. Not only would this have been cumbersome, but it would have interfered with the creative process of making music.

This new system has been out for about four months and working perfectly night after night. Chick can now walk on stage, sequence a tune (patch 1), do the soundcheck (patch 2), edit a synth (patch 1), and then instantly get ready for the show (patch 2). The only requirement for doing this is to switch between the two patches!

After the tour I'm going to write QuicKeys for each of Chick's applications so that at the touch of one Mac key, an entire setup will be configured (including the sequencer, the necessary Studio Setup, patching, etc.). When he switches to a new Studio Setup, OMS tells all of the software that there has been a change. At that point, all that is additionally needed is to plug in whichever module Chick needs!

It should be noted that the successful design of any MIDI setup requires the time and energy of many dedicated individuals, both in and out of the organization involved. Special thanks to everyone at Opcode Systems.

David Frangione, owner of David Frangione Enterprises in Arlington, MA, is a MIDI specialist and audio designer. He has worked for a host of biggies including: Aerosmith, the Boston Symphony, Reggie Lucas (Madonna), Elton John, and three recent feature films. His sampling CD has just been released by East West Communications, Inc.

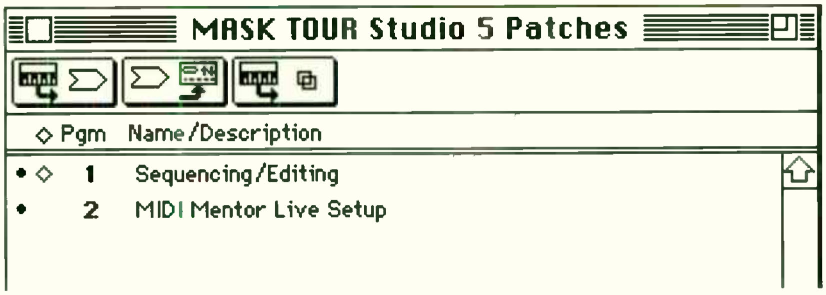


Figure 2: Opcode MIDI System (OMS) routing configurations.

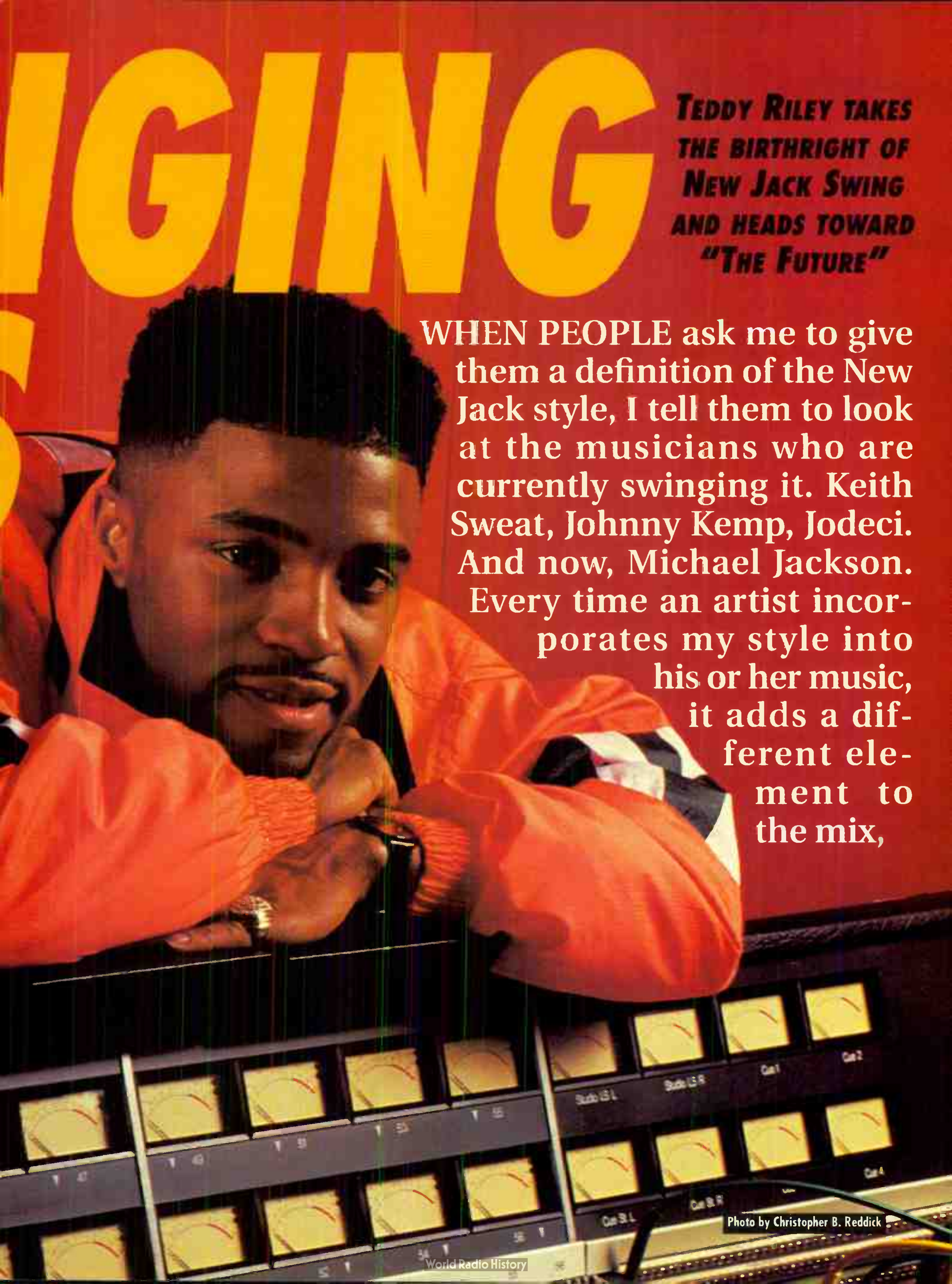
RAP TECHNOLOGY

SWIN LESSONS

BY TEDDY RILEY

IGING

TEDDY RILEY TAKES THE BIRTHRIGHT OF NEW JACK SWING AND HEADS TOWARD "THE FUTURE"



WHEN PEOPLE ask me to give them a definition of the New Jack style, I tell them to look at the musicians who are currently swinging it. Keith Sweat, Johnny Kemp, Jodeci. And now, Michael Jackson. Every time an artist incorporates my style into his or her music, it adds a different element to the mix,

Photo by Christopher B. Reddick

I LOVE TO BLEND, BIG RICH SOUNDS WITH DIRTY SOUNDS IN ORDER TO GIVE MY MUSIC A NATURAL, YET HARD-EDGED FEEL.

changing the people's perception of New Jack Swing once again. Already, the New Jack movement has felt the influence of a wide variety of musical genres, ranging from dance to R&B to funk to pop. However, no matter what styles come together to create my vibe, there's one factor that will always remain constant: The roots of this new music lie in the sights and sounds of the town in which I was born, Harlem, New York — otherwise known as New Jack City.

Basically, anyone who was born in Harlem is going to have some sort of rap or gospel background. That's why, at the heart of New Jack Swing, there lies a strong commitment to the sounds of street music and the rhythms of gospel. I love to blend big, rich sounds with dirty sounds in order to give my music a natural, yet hard-edged feel. Plus, I only record in analog so that none of the warmth is lost during the swing. If I didn't go analog, I wouldn't receive the same fat texture that keeps the bass drums boomin' and the snare drums slappin'.

Instead of using stock drum sounds, I prefer to sample live acoustic drums and then loop them into a natural beat. I don't sequence the drums in a quantized fashion because it's not natural — it doesn't produce any grooves. The drums must have an authentic feel to them if they're going to have a positive impact on the listener. I also like to go for the natural sound when I'm recording vocals, lead and background. Some of the best examples of my work, like "Remember The Time" from *Dangerous*, were recorded naturally, with minimum use of effects. A lot of people will record their vocals with a barrage of delays and echoes, but I prefer to use the microphone plain, so that the voice sounds clear.

REGARDING MICHAEL

When we recorded Michael Jackson's voice we didn't EQ very much. There wasn't any use in overloading him with too many effects when his voice sounds so much better dry. The real challenge was to create an effect that brings Michael right to the front of the speaker, in stereo. Even though he's the closest thing to a legend that we have living today, there's a very intimate side to Michael — a side that I wanted to bring out and into the listener's room. Besides, I think that the lead and background vocals sound best when they're smack-dab in your face.

Most of the vocal tracks that were taken from Michael Jackson, and Bobby Brown for that matter, were completed on the first or second try. I like to keep it that way because if you're going to make a mistake in the studio, you're going to make that mistake on stage. And when it comes to recording, we like to remain as true to the original sound as possible. If you can't come into the studio and sing a song the way it's supposed to be sung,

LARRABEE'S FUTURE ENTERPRISE

When Teddy Riley decided he wanted his dream studio, he chose the sky as the limit.

When the King of Swing, Teddy Riley, was selected to write and produce new material for the "King Of Pop" (guess who), it came as no surprise that Larrabee Studios, Universal City, CA, won the coveted coup of housing the dynamic duo during the recording of *Dangerous*, the latest album offered by a very New-Jack Michael Jackson. While many producers and musicians alike consider Larrabee to be one of the best studios this side of the Atlantic (Prince and Paula Abdul are on the A-list), Riley refers to the recording facility as *the* studio for maximum comfort and high-quality sound. As a matter of fact, the young producer's experience at Larrabee was so positive that when it came time for him to build his own music studio, The Future, he turned to the premier recording powerhouse for advice and inspiration. What began as words of wisdom from the horse's mouth, however, quickly escalated into a full-scale construction project, with Larrabee turning their words into action, and eventually heralding the enormous task of building Riley's hi-tech dream studio.

"Teddy was very specific about what he wanted in a recording studio," says Larrabee's owner, Kevin Mills. "His first priority was building a facility that would be capable of handling any challenge, problem or piece of equipment — pronto. The quest for maximum efficiency was the key here." One way Mills went about building Riley's super-studio was to wire everything up so that all of The Future's major racks could be plugged in with just one connector. Thanks to some Larrabee handiwork, long snakes with large connectors were designed so that Riley's MIDI gear could be plugged right into the back of outboard racks and spontaneously show up under the patchbay without any hassles.

Other Riley objectives included creating an environment that strongly resembled that of Larrabee's elaborate recording haven. Everything from the acoustics design to the lighting system was built with the Larrabee mind-set in tow. Even their custom George Augspurger speaker system was incorporated into the framework of The Future, enabling the modern-day maestro to pump up the bass to thunderous levels without worrying about speaker blow-out. Other pieces of equipment that came readily approved by Mills' production staff included Pultec equalizers, AMS reverbs and delays, Lexicon 480Ls, UREI 1176 compressors, Neumann U 47 and U 87 microphones and a Studer 827 multitrack. Not to be outdone by Larrabee's mind-blowing arsenal of audio consoles (they have four SSLs), Riley himself purchased a 60-input SSL with plans to one day install a 100-input board. "I want to create the most technologically advanced studio around," says the man who recorded Michael Jackson.

At only 25, it looks as if Teddy Riley is more than halfway there.

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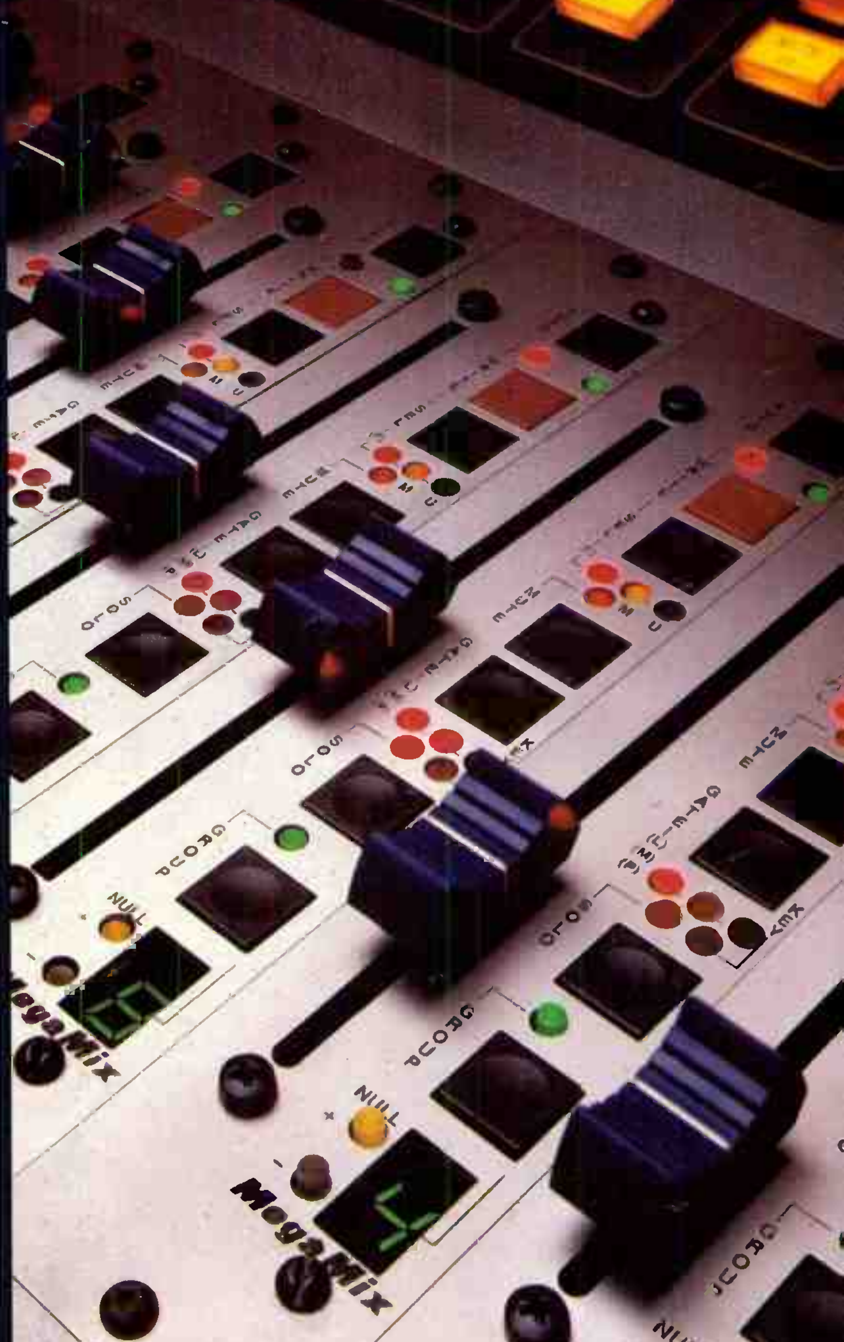

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RAP TECHNOLOGY

WITH THE RECORDING OF A MICHAEL JACKSON ALBUM COMES THE FUN JOB OF CREATING NEW SOUND EFFECTS.

then you don't need to be working with me. Jackson tunes such as "Remember The Time" and "In The Closet" were done in only two takes. But a song like "Jam," which requires a strong attitude (and an even stronger voice, which Michael nearly lost) was completed on the second or third take. If the singer feels the music, and you've got the melody recorded beforehand, you're going to get the vocals down cold.

Along with the recording of a Michael Jackson album comes the fun job of creating new sound effects for each song. On *Bad* (Quincy Jones, Bruce Swedien, prod.), there were a lot of unique sounds recorded with high-tech finesse, including the live beat of Michael's heart. When it came to recording *Dangerous*, I went out and got live car sounds which I weaved into the beat of "She Drives Me Wild." For another track, "In The Closet," we increased the song's intensity by incorporating door slams and videogame-like effects throughout the melody. I've been sampling and recording effects for some time now, even during my days with Guy, when I used Performer software to sample bomb sounds and loop beats.

BACK TO THE FUTURE

Even though technology has greatly expanded our recording horizons, I find myself going back to the old style that existed in the 70s, when real instruments were the norm and live recording was *it*. Equipment such as the Mini Moog and the old Juno 106 are coming back, while rap vocals are being recorded right in front of the console, closed in by gobos, rather than windows. As far as sound inventions go, however, I'll use whatever stock material I can find and put it through the technological wringer to come up with something new and exciting. I don't spend too much time analyzing the techniques of my heroes — Stevie Wonder, Prince, Quincy Jones — because I'm always looking for a different way of doing things. For example, if you give somebody a Vocoder or a Vocalizer, he might come up with some great stock sounds, but if I get my hands on it, we'll put it through any given piece of equipment and create some sounds that you've never heard before.

We have a whole slew of sound-altering techniques that I like to keep relatively secret because I want to be able to use them for the next three years without anybody capitalizing on

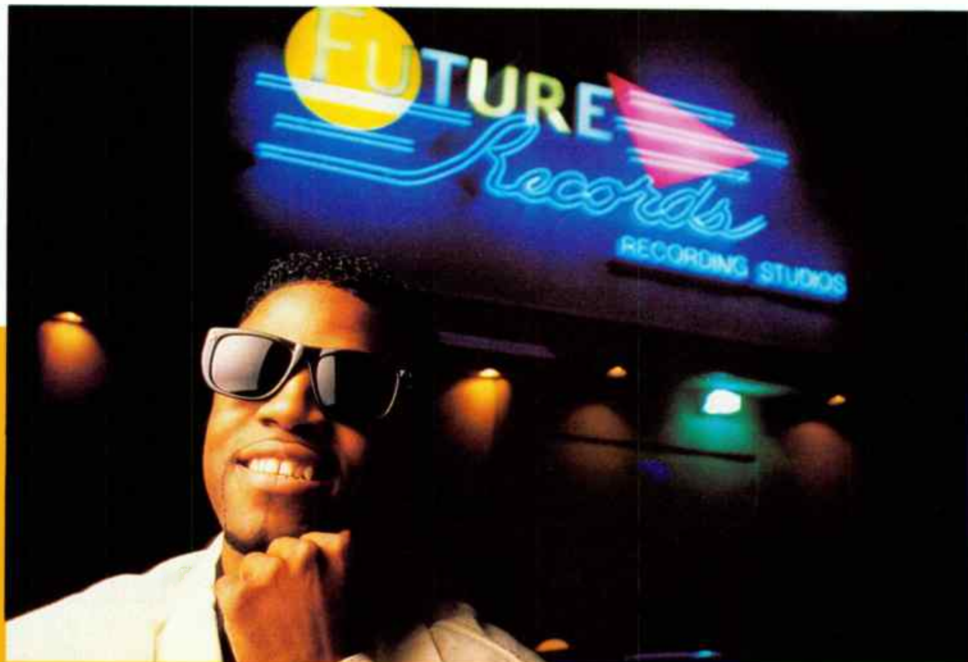
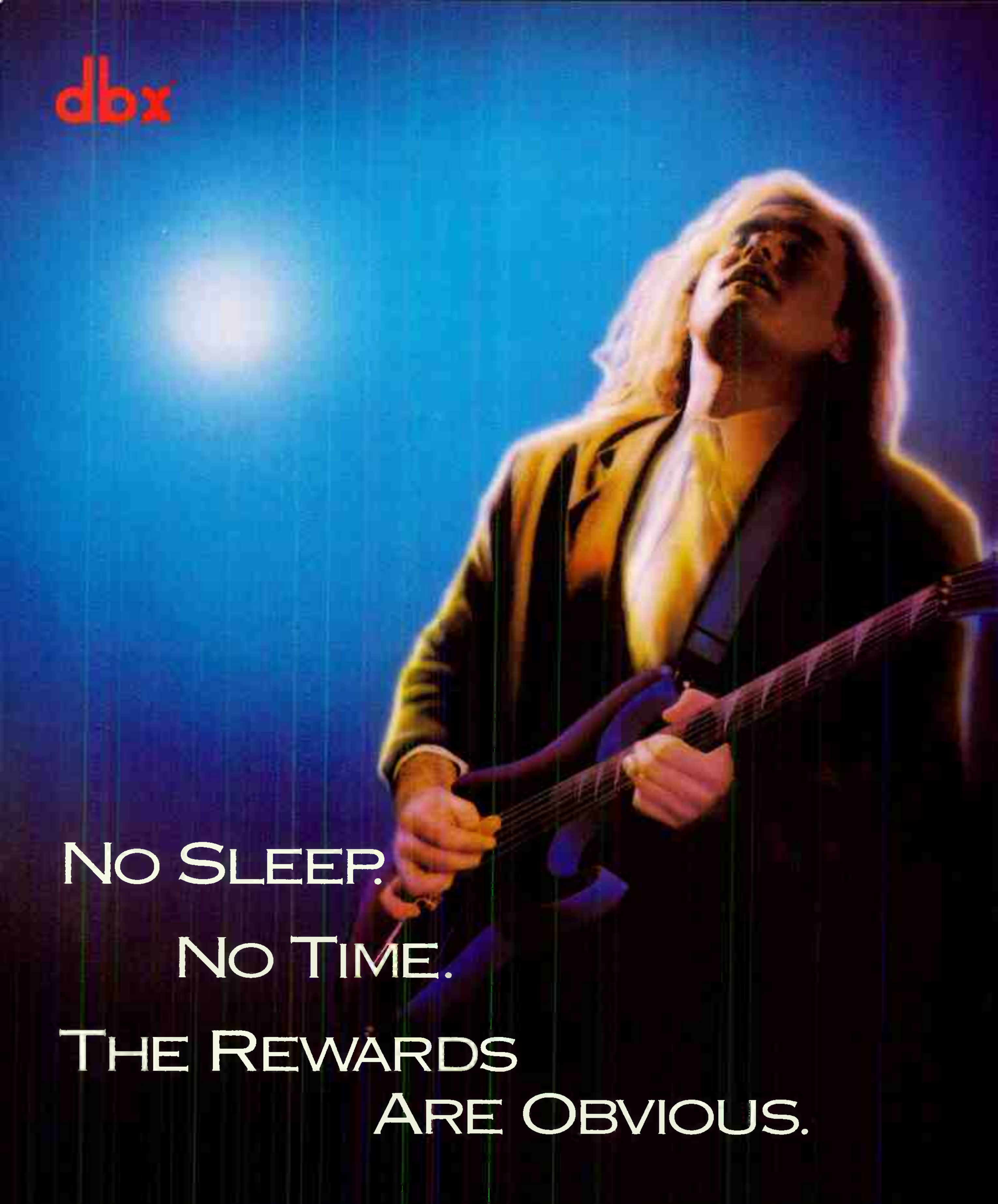


Photo by Christopher B. Reddick

LIFE OF RILEY

In the parking lot of The Future studio complex there sits a row of black and red Porsches, one of which bears the license plate NEWJAK1. There's only one person who can lay claim to this title and that's Teddy Riley, the 25-year-old wunderkind who has been revered as the founder of New Jack Swing, a smart mix of brash, jacked-up street sounds and fresh gospel rhythms that is currently ruling the radio airwaves. Riley has been revered as top dog ever since he delivered double-whammy productions of "I Want Her," by Keith Sweat (1987), and "Just Got Paid," by Johnny Kemp (1988). His own band, Guy, also hit megaplatinum status, further laying down the groundwork for his billing as a major player in the music biz. It wasn't long before Riley's New Jack hybrid of snappy drums and hardcore horns caught fire, opening the door to a variety of artists seeking his vibrant new sound. Bobby Brown, Stevie Wonder, Kool Moe Dee, Boy George and Heavy D. are just a few of the majors that Riley has worked with since his induction as the newest whiz-kid on the block. But it wasn't until his friend, comedian Eddie Murphy, introduced him to pop-phenom Michael Jackson, that Riley entered the realm reserved for only a chosen few. In true New Jack style, Riley stood up to public scrutiny and created a unique compilation of Jackson songs (heavy on strings and dry on vocals) while still presenting him as the king of cutting-edge pop. More recently, Riley has finished working on Bobby Brown's latest album, hoping to recapture the glory heaped upon him after producing the super-star's hit album, *Don't Be Cruel*.

— Jon Varman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NO TIME.
THE REWARDS
ARE OBVIOUS.

THERE ARE JUST TOO MANY PEOPLE IN THIS INDUSTRY WHO TREAT MUSIC LIKE A COMPETITION.

them. There are just too many people in this industry who treat music like a competition, and that makes it hard for me to pass down a bit of information without someone going out and saying that *he* discovered it. That's just the downside of the industry and it comes with the territory.

The upside is my studio, The Future, which I've settled down in Virginia Beach. I like the fact that I can work here in quiet, away from the hype of the big cities, and do what it is I have to do without any interruptions. I never leave this place unless I absolutely have to, and even then I won't go unless I've completed the essentials. I'm not looking to be a gigantic star or the big guy on campus. I just want the freedom to make music that people can appreciate, and it's here in Virginia Beach, in the relative calm, that I can best achieve my primary goals.

Word to the wise: *Studios are not built overnight.* I realized that after I was told that I would have my studio completed in three months and it ended up taking a year. Now that I have it, I'm ready to take it all the way. My plans are to have the biggest studio with the biggest sound. I don't think any other studio besides Larrabee, and a handful of others, can get the bottom that The Future gets. I used to go into studios and complain about their speakers because they used to bust off of my music. Now, when I go into my own studio, I couldn't bust them if I tried.

The Future is made up of two studios, an SSL room and a smaller CAD room, both of which I use exclusively. A MIDI room has also been set up to house computers, sequencers and keyboards. We're currently in the process of building a new room that is going to feature every technical and vintage element known to man. Like Larrabee, I want my studio to have the capacity to create any sound and manage any production technique. My dream would be to have a little bit of everything — SSL, Neve, CAD, Focusrite, UREI — not a console of each brand, but a whole soundboard made up of the best of the best. Right now we're trying to get an 80-100 channel SSL board in here so that we can go all out.

TAKE IT TO THE CHARTS

Working on the Michael Jackson album was an all-around great experience. Not only did I get the chance to bring the New Jack sound to Michael's music, but I also received the opportunity to work with Bruce Swedien, one of the masters of modern recording. [See *EQ* June 1991 cover story.] Most recently, we worked on a remix of the new single "Jam," with some help from my coengi-

continued on page 107

TEDDY'S CADILLAC RANCH

Riley's SSL room may be the main attraction at The Future, but as business continues to grow, more and more New Jack Swingers are being drawn into the CAD room — a compact, versatile studio that features Tannoy and Yamaha NS-10 monitors, Mac and Atari computers and the E-frame 84-input automated Maxcon Mixing System.

Franklyn Grant, a Future engineer/producer, said "The Future's Maxcon room is primarily used as a writing room. It features a crisp, clear sound and provides inhabitants with an explosive experience."

"When we first heard this console," says Grant, "it really hit us. Teddy and I really love the frequency response of the CAD. And the fat bass on this board is incredible!"

The Maxcon Mixing System features the MegaMix Inboard Series M-400 automation, which includes a full hardware-based dynamics package. Because the system utilizes Class A VCAs, it's known for its transparent sound and it enables the operator to accomplish rapid-muting, fader level and real-time control of its full parameter gates. All functions are displayed on a video monitor.

Currently, The Future is laying down tracks for Wrecks N' Effects and Big Ty and the Hoods in the CAD room. "We're looking for that hardcore, New Jack sound," says Grant. "And in this room, we can do more than simply achieve our goals."

Photo by Christopher B. Reddick

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STO

RAP TECHNOLOGY

KIN IT UP

Getting Street-Smart in the Rap Studio

By Stoker

Rap Music has come a long way since the days of turntables and the vibes of The Furious Five. MIDI dumping, striping SMPTE and CD special-effects sampling have become the norm for rap engineering, while favorite pastimes, such as record scratching, have been relegated to the backburner. Even though the face of rap and rap mixing has changed over the years, it is still a music that remains deeply entrenched in its original roots. Vintage equipment and age-old beats, combined with the perks of modern technology, have helped rap become a high-tech force that has managed to maintain its familiar, streetwise sound.

I WAS LOOKING FOR A REALLY DEAD SOUND AND THROUGH SOME SIMPLE EXPERIMENTATION, I FOUND IT.

Before actually mixing a rap single, the first thing I do is set up the drum machines, mic the rapper and equip the DJ so that he has everything he needs at his fingertips. The rapper will usually preprogram his drum beats, but the DJ must be able to sample spontaneously with nothing standing in the way of his creative mind-flow. Some drum machines I like to work with are the SP1200 by E-mu and the MPC60 by Akai. The SP1200 has a 10- or 12-second sampling time, which gives you just enough time to put all your beats, loops and bass lines together, and carry them around in one compact box. As far as miking goes, an AKG 414 is my best bet. While working on Ice Cube's album *Death Certificate*, I miked Cube with a 414 and put him in a U-shaped alcove with a massive acoustic blanket draped over one side. I was looking for a really dead sound and through some simple experimentation, I found it.

GOING TO THE DUMP

After I'm done with preproduction, I'll stripe the SMPTE timecode and prepare for what will eventually be the biggest part of the production process: the MIDI-dump. The drum machine receives SMPTE from tape and is told exactly when to start its playing pattern. This enables you to play back a second pass of loops and drums in the same exact time as the original pass. When I start the MIDI-dump, I'll EQ the drums and prepare

SAMPLES SELL

Sampling clearance houses clear the air for rap recordists

In the light of recent suits against a slew of rappers and rap producers, sampling clearance houses have been doing gangbusters business, helping record companies to safeguard against ugly legal battles while carving out a substantial niche for themselves in the rap recording community.

Since their recent inception into the music industry, companies with names such as Clearance 13'-8", Diamond Time and Sample Doctor have found themselves in constant demand for their sample tracing services, hired to explore brave new worlds (and ancient lands) in the etymological quest for the lost sample. These days, rarely will you find a record company or entertainment entity bold enough to buck the system by releasing sampled material without the benefit of a security blanket, courtesy of a sampling clearance house.

"Our purpose is to catch the problems in preproduction before they arise in post-production," says Cathy Carapella, V.P. of Operations at Diamond Time, Ltd. "Using a sample on a record is like booking a studio. You don't use the studio unless you know how much you're paying an hour." Although clearance payments could sometimes rise to painful proportions (up to $2,000 advances on masters), ignoring sampling costs altogether may mean giving up 100 percent of a song's copyright, as was the case when Kraftwerk sued Afrika Bambatta in the early '80s for using a sampled drum beat. "Copyrights are commodities," continues Carapella, who cites George Clinton and Ray Charles as recent beneficiaries of the sampling phenomenon. "And the fact of the matter is — you don't get paid for your song, you get paid for your copyright."

As record companies continue to seek out sampling clearance services (Warner Bros. has made it a must for their rap artists), clearance "doctors" such as Hope Carr, president of Clearance 13'-8", continue to find themselves in high demand for their labor. "In the beginning, business was not booming because people would only clear large sections of songs," says Carr, "but now we're getting smaller samples to clear and everyone wants to claim a publishing interest, especially when it comes to rap." One of Hope Carr's clients, ex-3rd Basser Pete Nice, has coped with the sampling witchhunt simply by recording live music jams over old records, and then sampling snippets of his own live-band overdubs. "Sampling has grown from the garage into big money," says the man behind such rap hits as "Gas Face" (a gibe at Hammer) and "Pop Goes To Weasel" (a successful attempt to melt Vanilla Ice). "Unfortunately, there's a certain amount of greed out there."

For all its negative hype, however, the sampling phenomenon has proven to be a boon to musicologists, publishers and clearance houses alike, all of whom have tapped into previously unforeseen sources of revenue. In recent months, even record companies have begun to get into the action, sending their executives and attorneys back to school to sharpen their musicological skills. "No matter what happens, there will always be a place for people like me," says the owner of Sample Doctor, Rod Moskowitz. "Clearing is a long, tedious process and we have the contacts to get it done quicker and sooner than anybody else."

While enthusiastic about rap's current status, Moskowitz has a less optimistic outlook when it comes to the future of sampling. "In my opinion, sampling is not going to last for more than five years," asserts Moskowitz, "because most people see that it may be cheaper to get a live musician to replay a song rather than sampling it. In this case, you're not dealing with the master rights and there's no up-front money involved." So, if live sound is the future of rap, where does that leave sampling clearance houses? "When live musicians come in to do a song, the artist will still have to seek publishing clearance," says the Sample Doctor himself, "And that's where we come in."

— Jon Varman

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them so that they'll sound like a record is streaming right out of the drum machine. To me, this is insurance guaranteeing that what I'm putting on tape is going to sound great when we actually mix the song.

PUMP UP THE VOLUME

I record at really loud levels (+9). You're dealing with a lot of noise on these grungy samples, some of which are 10-15 years old, and basically, the louder you can get a sample on tape, the less hiss there will be. Sometimes a band (yes, a band) will come in to liven up the samples or lay down a few tracks, yet I always make sure to retro-engineer the sound so that the songs will have an earthy feel to them. You don't want to bury the music under too much technology. This is where vintage equipment such as a Moog keyboard or a classic console can really enhance your recording. *Death Certificate* was mixed on an old 6068 Neve, which used to make its home at Abbey Road Studios. Another old-fashioned goodie is the UREI 1176 compressor; however, it's wise not to use too much compression when you're

Photo by Peter Monroe

SAMPLING: A BAD BIZNESS PRACTICE?

Last year, *EQ* featured BMV Digital Studios in it's "Room With A VU" section and unwittingly photographed what would soon be the birthplace of scandal. BMV was the facility used to record rapper Biz Markie's song "Alone Again," which turned out to be the most controversial rap tune put on vinyl since "Fuck The Police" by N.W.A.

Unfortunately for The Biz, the government didn't just give the rotund rapper a slap on the wrist for bad language (as was the case with N.W.A.), but instead threw the entire judicial book at him, citing him for violation of U.S. copyright laws and condemning his use of an 8-bar sample from the song "Alone Again (Naturally)," which was originally recorded by Raymond "Gilbert" O'Sullivan in 1972. In a decision that was considered "stern" by some and "fitting" by others, Judge Kevin T. Duffy of Federal District Court in Manhattan barred Biz Markie's album *I Need A Haircut* from being sold at retail stores, leading to Warner Bros.' subsequent removal of the record from shelves and display racks across the nation.

With this court decision, Biz Markie joins the ever-growing legion of rappers (De La Soul, Rob Base, Beastie Boys) who have been reprimanded in one way or another for using an unwarranted sample. But what the court calls an "8-bar sample," Biz Markie calls a remake, one which O'Sullivan should be happy was resurrected in the first place. "It's a good song," says The Biz, "but they're trying to say that all my wealth came from that one record, which is crazy. I was out there rapping for eight years, man. Eight years!" The album in question was out for approximately six months before its removal.

Although some industry insiders say that this will accelerate the trend towards more live music and less record sampling, Biz Markie's producer, Cutmaster Cool V, says that sampling is here to stay. "You can use whatever sample you want," says Cool V, "but you need the right people to give you permission for it. In our case, there was a mix-up between the lawyers and the record company and we were left on the outside. If we had known that clearance wasn't granted, we would have never used the record in the first place."

By facing up to the various responsibilities that go along with the sampling process, it may seem that rappers and rap producers may finally be one step ahead of the legal powers that be. However, even as rap insiders look towards modern technology to disguise and alter unauthorized samples, new problems seem to be looming over the musical horizon. Excessive royalty costs, combined with the time-consuming process of locating copyright owners, contribute to the shroud of infeasibility surrounding the sampling practice. Plus, to make matters worse, a whole new crop of sampling skirmishes have found their way into the courtroom.

Marley Marl and LL Cool J are currently being challenged for sampling — rapper's beware — an age-old drum track, while the self-proclaimed King of Pop himself, Michael Jackson, is being zapped for sampling 67 seconds of the Cleveland Orchestra's rendition of Beethoven's Ninth Symphony.

Meanwhile, The Biz is back to work on a new album which will still retain its humorous edge and feature sampled goodies from the sound vault at BMV Digital Studios. And what of the fate of the scourged record, *I Need A Haircut*? Cutmaster Cool V says that the album has virtually disappeared from the face of the earth. But if you did happen to get your hands on a copy you're in luck. Can you say — collector's item?

— *Jon Varman*

working with bass drums or thick beats. If your Roland 808 sample (the bass drum of rap) isn't rumbling the speakers, then you haven't captured the full essence of the rap experience.

SAMPLE CITY

Sampling records and television programs continues to play a large role in rap music, albeit a different one. Since royalty fees are now enormous, you'll find many rappers acting out bits or making up their own TV sounds. On Cube's album we worked with the George Lucas Sound Effects Library, utilizing their CDs for gunshots, screams, and a whole variety of scenarios.

That doesn't mean rappers have stopped sampling. They've just gotten more cautious about what and how they sample something. In order to keep track of all the samples we used on Death Certificate, we had someone sit in the studio with a pen and pad and take inventory of every sample, right down to the last James Brown screech. As the engineer, this took a lot of the onus away from me and put it on the legality shelf, where it belongs.

SAMPLING GEAR

As far as equipment goes, I like using the Akai S900 sampler, which came out in 1985 or 1986 and is still one of the best samplers around. You can spend somewhere around five grand on an Akai S1000 and get a great machine, or spend $300 on Macintosh software for the Akai S900 and you'll have everything you'll ever need in this aging machine. Basically, obsolescence rules.

And always remember, the thing to know about rap mixing is that you're generally aiming your sound at your average B-boy with a serious bottom-end box.

Stoker is a freelance rap producer. In addition to engineering tracks for DJ Jazzy Jeff & The Fresh Prince, Ton-Loc, Vanilla Ice and Queen Latifah, he recently produced the songs on Death Certificate, *the latest album from rap star Ice Cube.*

VINTAGE GOODS

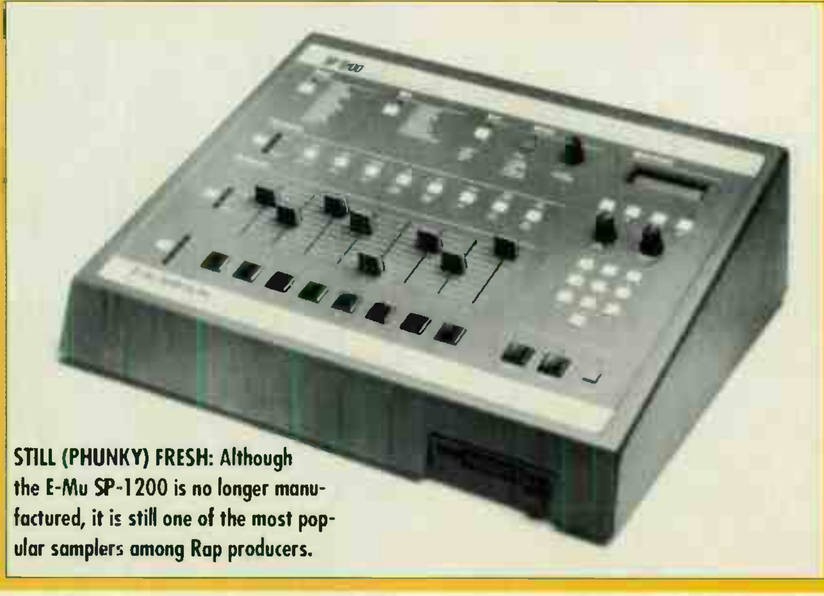
E-mu's SP-1200 can easily be found in most rap studios. The trick is getting your hands on one.

What do Jeff Townes (a.k.a. Jazzy Jeff), Mario Caldato, Jr., Eric Sadler and Marley Marl have in common? Besides mixing for some of the biggest names in the rap community, these rap-mavens swear by the SP-1200 sampling/drum machine from E-mu Systems. This highly-prized unit can often be heard banging out beats across the rap-ready airwaves, providing the snare and bass backdrops for a bevy of artists including The Beastie Boys, LL Cool J and Salt 'N' Pepa, just to name a few.

Eric Sadler, owner of Street Element, calls the SP-1200 "the starting point of all jams," while Teddy Riley [see cover story] attributes E-mu's invention with incredibly warm sounds. However, for all its incredible kudos, the fact remains that the SP-1200 is a rare-breed — a drum/sampling machine that is magnanimously adored, yet painstakingly absent from E-mu's current production line.

"People who currently own SP-1200s do not want to let them go," says Rob Faulkner, marketing communications manager at E-mu Systems. "The demand is so high that people have taken out ads in the paper looking for them, willing to pay as high as $3,000 for one." Part of the machine's allure is due to its unique user interface, which is particularly suited for the rap project studio owner/engineer as well as producers and songwriters. The Tune/Decay mode enables the user to record snappy rhythms while waggling a favorite sound (cowbell, agogos, etc.) up and down at random pitches. Other features that make the SP-1200 one of the premier rapmasters include great SMPTE chasing abilities, easy-loading disc set-up and stellar overall performance.

So what options exist for those who want to get their hands on one of these vintage works-of-art? Well, the prospect of the SP-1200 being reintroduced into the industry is currently under consideration, reason being that while project studios continue to gain in popularity, the demand for the SP-1200 continues to grow as well, making it impossible for E-mu to ignore the public at large. Meanwhile, interested parties can look into purchasing E-mu's Emax II Digital Sound System, the combination keyboard/sampler that offers the best in new and old school sounds.



STILL (PHUNKY) FRESH: Although the E-Mu SP-1200 is no longer manufactured, it is still one of the most popular samplers among Rap producers.

Hammer's House

From deep within the heart of Fremont, CA, come the rumbling sounds of a house being torn down. No, it's not an earthquake. It's Bust-It Studios, the music recording facility whose president happens to be one-man conglomerate and international rap star, Hammer.

Some of the biggest names in rap and R&B come here year round to "tear it up," while Steve Young, Bust-It's chief engineer, works double-time to record the tracks for platinum posterity. Included on the studio's in-house A-list are rapper Doug E. Fresh, ex-Prince group member Sheila E., 90's-crooner Ralph Tresvant and, of course, Bust-It's illustrious owner, Hammer (MC not included).

What began as a studio bus for on-the-road recording has evolved into a multiroom facility complete with MIDI capabilities, a 56-input Amek Mozart console and nearly every brand of outboard and effects gear available. Young, the main man who designed Bust-It Studios, along with general manager, Herbert Curtis, says that versatility is the key to the studio's success. "You must be able to adapt to any production style," says Young, "and that includes having the ability to change your configuration at any given moment." To cater to this sound philosophy, Bust-It Studios has been equipped with a flexible MIDI configuration that features a Macintosh Quadra 900 9 MB RAM computer system and Opcode Studio 5 and Mark of the Unicorn MIDI interfaces. "Just the fact that you can set up your own patches with the Studio 5, and adjust them from the front panel rather than the back, is a strong plus," says Young.

When it comes to sampling, Bust-It Studios boasts an inventory comprised of two Akai S1100s and a Roland S-770. The S-770 comes in handy when remixing songs such as "Too Legit To Quit" and "Burn It Up" (Felton Pilate, producer), which call for heavy background chants to answer Hammer's high-energy rallies. The S770 enables Young to truncate

Busting out the hits at Hammer's rap/R&B heaven

BY JON VARMAN

Chief engineer Steve Young swings with an MC-less Hammer at Bust-It Studios. Photo by David Weintraub.

vocal samples, while still saving the drama of their last natural breath.

For miking rap artists, Young uses a Neumann U 89, simply because it's a tough microphone that'll take the "P"s and "B"s gracefully. If the day's session calls for a smooth vocal performance, however, Young will break out an AKG tube mic, which he also credits with having a fine tone.

As far as drums go, Young likes his skins tight, snappy and, above all, live. A room dedicated entirely to the live experience has been built into Bust-It's framework for recording everything from drums to horns to an old Moog bass. But on those occasions where drum machines are absolutely essential, Bust-It Studios offers a cornucopia of sounds from the drum-banks of Yamaha, Akai and Roland. "Be creative with your sounds," suggests Young, who comes from the "mix and match" school of music production. "It's better to have a lot of drum sounds from a variety of manufacturers, than to get stuck into the stasis of one kit."

As is the case with many rap/R&B studios, spontaneity remains job one. EQ

STUDIO NAME: Bust-It Studios/Touring, Fremont, CA.
MAIN MEN: Hammer — President; Steve Young — Chief Engineer; Herbert Curtis — General Manager.
PRODUCER CREDITS: Felton Pilate, Demetrius Shipp, George Cauthen
ARTIST CREDITS: Hammer, Troop, Doug E. Fresh, Sheila E., Ralph Tresvant, B. Angie B., Ho Frat Ho, Oaktown's 3.5.7.
CONSOLE: Amek Mozart 56 x 32.
AUTOMATION: Amek Supertrue Mute/Fader (VCA)/Switches.
SAMPLERS: (2) Akai S1100 (32 Meg), Roland S-770 16 Meg
DRUM MACHINES: (2) Akai MPC60, Yamaha RY30 and RX5, (2) Roland R-8.
KEYBOARDS: Korg Wavestation, Moog Source, Mini Moog, Roland W-30, (2) Korg T-3.
KEYBOARD MODULES: Proteus w/Invasion Update, Korg M-3R, M1R and O1/W.
MONITORS & AMPS: 1 pair Meyer HD-1, 1 pair Westlake BBSM 15, 1 Crown Micro Reference.
EFFECTS: Lexicon 480L, (2) Eventide H300-SE, TC Electronics 2290, Quantec QRS, (2) Lexicon PCM 70, (2) Yamaha SPX900, Yamaha SPX 1000.
COMPRESSORS/LIMITERS: Tube Tech, (8) Aphex Expressors, (2) Aphex Compellors, (2) UREI LA-4, (8) dbx 166. Gates: (8) Aphex 9611, (8) dbx 363.
MIDI COMPUTER SYSTEM: Quadra 900-9 Meg Ram, 40 Meg HD with external Eltekon 80 Meg; Opcode Studio 5 MIDI Interface, (4) Mark of the Unicorn MIDI Time Piece MIDI Interface.
SOFTWARE: Performer 3.61, Sound Designer, Studio Vision, Galaxy Plus, Archie.
EQ: Tube Tech Program EQ.
OTHER: BBE 702 Sonic Maximizer.

DJ Jazzy's Joint

STUDIO NAME: Touch of Jazz Productions, Gladwyne, PA.
MAIN MAN: Jeff Townes (a.k.a. DJ Jazzy Jeff)
PRODUCTION CREDITS: D.J. Jazzy Jeff and the Fresh Prince, The Simpsons Sing the Blues, El Sid
CONSOLE: DDA DMR-12 (56 inputs with MIDI Muting)
TAPE RECORDERS: Otari MTR-90 MK II
SAMPLERS/DRUM MACHINES/KEYBOARDS: (2) Akai S1000, Forat F16, Akai MPC60, E-mu SP 1200, E-mu Proteus 1XR, Roland JD-800, Yamaha TX816, Yamaha TX81Z, Alesis D4, Korg M-3R, Korg M-1R, Roland U-220, Roland D-110, Roland D550, Yamaha TG-77, Oberheim Matrix 1000, Korg 03R/W, Roland R8M, Roland S-10, Yamaha TG33
MONITORS: Westlake BBSM-12, Yamaha NS-10M, Tannoy PBM 8, Bryston 4B, Hafler 1200
DATS: Sony PCM-2500, Sony TCD-10 PRO
MICS: Neumann U 87
TAPES: Ampex 456
COMPUTERS: Apple Macintosh IIcx with Digidesign SoundTools and Opcode Galaxy, plus Atari 1040ST with C-Lab Notator
OUTBOARD GEAR: (2) Eventide H3000-SE/B with Sampling Board, Lexicon LXP-5, TC Electronics 2290, Orban 642, Tube Tech C1-1a, Yamaha SPX900, Yamaha SPX90, (12) dbx 166, (4) Drawmer DS-201,dbx 120.
TURNTABLES: (2) Technics SL-L20
EQUIPMENT NOTES: The DDA console is great! It is very logically laid out. By having the same EQ in the monitor section of the board, the production flows a lot faster. I do all my writing on my Atari computer using C-Lab Notator software. My Akai S1000s are hooked up to a couple of 600 MB hard drives and to a 44 MB removable hard drive. These samplers are the workhorses of the studio. The Westlakes give me the bottom that I need to feel.
PRODUCTION NOTES: The studio was put together from necessity. I was being approached constantly to do production for other artists. My technical support guy (George Hajioannou) helped me make all the right decisions on what to get and how best to use it. EQ

Photo by Steven M. Folk

Hitz-Maker

STUDIO NAME: Marley Marl's House of Hitz, Chestnut Ridge, N.Y.
MAIN MEN: Marley Marl, Frank Heller
PRODUCTION CREDITS: LL Cool J, Monie Love, Tragedy The Intelligent Hoodlum, BBD, Heavy D, Marley Marl
CONSOLE: SSL SL 4000 G Series w/Ultimation
TAPE RECORDERS: Tascam 24-track ATR-80 (2)
SAMPLERS & DRUM MACHINES: Akai MPC60II, SP1200, Roland 808, Yamaha RY30, Akai S1000, Akai S950, Eventide H3000SE
SAMPLER/SEQUENCERS: M-1R (2), Proteus 1, and D-550 Roland
KEYBOARDS: Kurzweil Midiboard
MONITORS AND AMPS: Tannoy FSMU Double 15-inch, Yamaha NS10M, Hafler Amp, Perreaux 9000B
DATS: Sony 2300
MICROPHONES: Neumann U 87
EQUIPMENT NOTES: We're the first home studio with the 48-input SSL SL4000G w/Ultimation. We also have the Digidesign SoundTools system.
PRODUCTION NOTES: When we make our sounds, we stay to the streets and look for a raw edge. We stick to analog because with digital there's too much clarity, and we need a dirty sound. Our main priority is to make music for people of the underground.

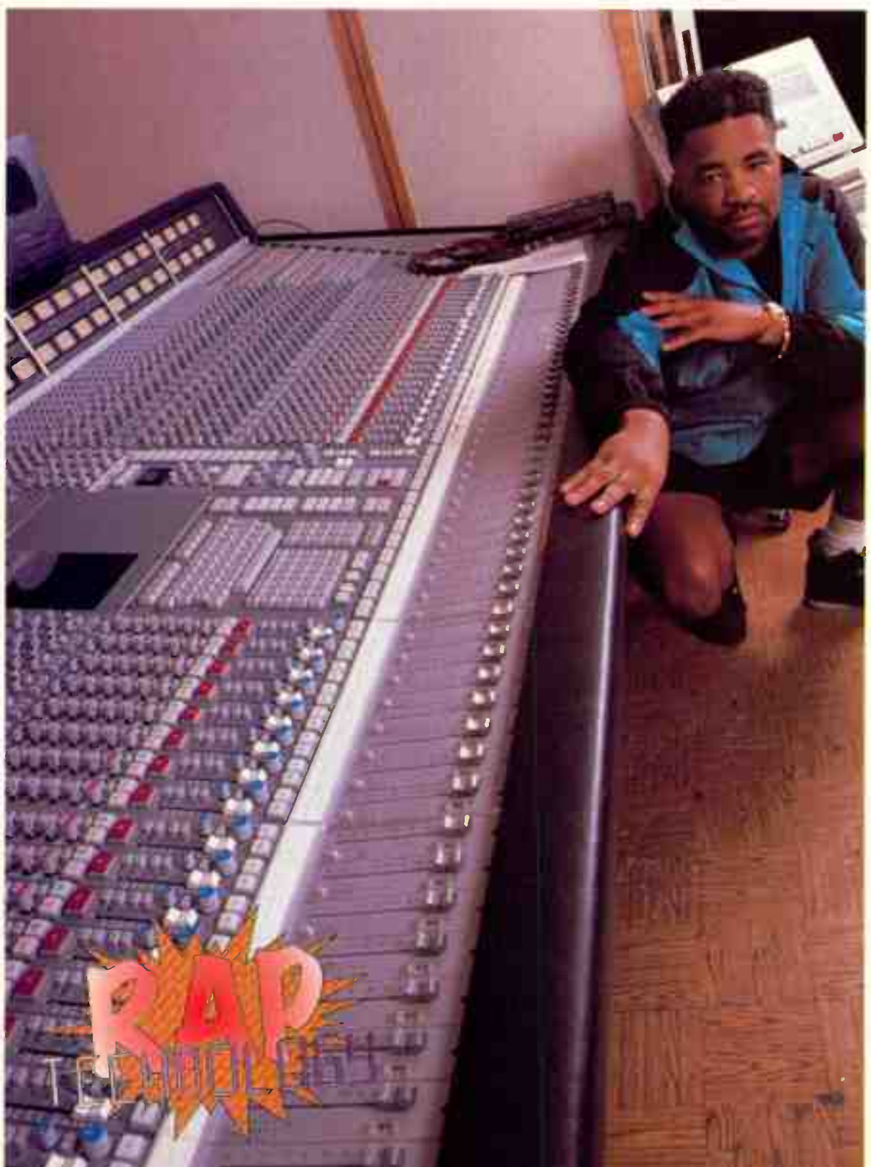


Photo courtesy of Solid State Logic

One-On-One

STUDIO NAME: One Little Indian Music, El Cerrito, CA.
MAIN MAN: Al "BJ" Eaton.
PRODUCTION CREDITS: Too Short, Paris, Kid Sensation, Biscuit, DJ Flash.
CONSOLE: Soundtracs Quartz 48 x 24 inline console.
TAPE RECORDERS: (2) Tascam ATR-80, Fostex B-16D, Digidesigns "Soundtools," Sony PCM-601 Digital Audio Processor.
DRUM MACHINES: (2) Alesis SR-16, E-mu Procussion, Roland R8M, Roland TR-909, 808, 707/Rhythm Composer.
SAMPLERS: Digidesign Sample Cell and Sound Tools, Akai S950, S1000HD, (2) Roland S-550, Ensoniq Mirage DMS, Sequential Circuits Prophet 2002, Yamaha TX16W.
KEYBOARDS & SYNTH MODULES: Roland JX-8P, MKS-50 and SVC-350, (2) Juno-106, (2) Yamaha TX816, Vocoder, U-220, E-mu Proteus 1 & 2, Korg M-1REX, Oberheim Matrix-1000.
MONITORS & AMPLIFIERS: UREI 813, JBL 4412, Yamaha NS10M, Auratone 5c Super Sound Cubes.
DATS: Panasonic SV-3500, SV-3700.
MICS: (3) AKG 414 B/ULS, Neumann U 87A, Countryman Associates D.I. boxes.

Tales From The Cribb

STUDIO NAME: Howie's Cribb, Brooklyn, NY.
MAIN MAN: Howie Tee.
PRODUCTION CREDITS: Chubb Rock, Color Me Badd, Heavy D and The Boyz.
CONSOLE: Tascam M3700.
TAPE RECORDER: Tascam MSR-16 and 32-2 track, Aiwa double cassette deck
SAMPLERS & KEYBOARDS: Akai S950 and S900, Roland JD-800, U-220 and D-50, Korg M-1REX and M-3R, Yamaha TX812, Proteus 1XR.
DRUM MACHINES: Yamaha RY30, Roland TR-808.
MONITORS: Yamaha NS10M, UREI 809.
AMPLIFIERS: QSC MX 700, Crown Power Base-2.
DATS: Tascam DA-30.
MICS: AKG C414 B/ULS, Sennheiser.
TAPES: Ampex Grandmaster Gold 499, 456, 472-C-10 and 472 DAT C30.
EQUIPMENT NOTES: For sampling, I like to use the Akai machines because they do the job when it comes to arranging beats and bass lines. If I need something done and there's a piece of equipment that can do it, I won't hesitate to check it out.

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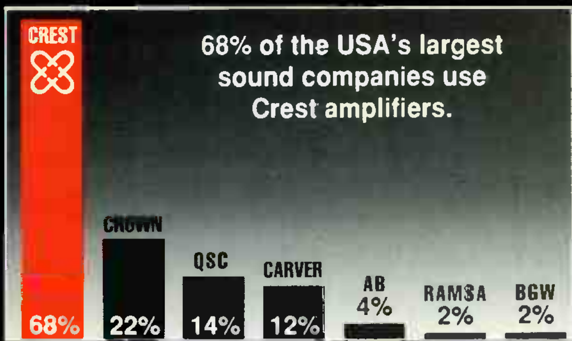

Totals are over 100%; some respondants mentioned more than one brand.

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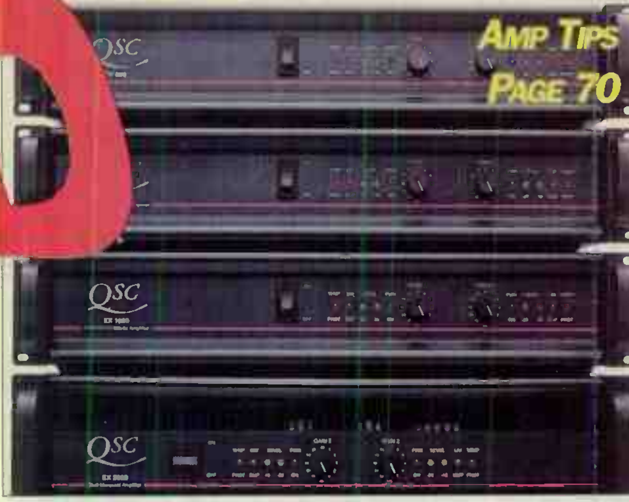

AMP TIPS
PAGE 70

A SPECIAL SUPPLEMENT TO EQ MAGAZINE

HI-HO, HI-HO, IT'S UP ON-STAGE WE GO

MIDI ELVES GO TO WORK TO CONSTRUCT DIGITAL STUDIO SOUNDS IN CONCERT

PAGE 88

INSIDE: WIRELESS MICS BUYER'S GUIDE, THEATER SOUND & MORE

Hi-Ho, Hi-Ho, It's Up On-Stage We Go

MIDI ELVES GO TO WORK TO CONSTRUCT DIGITAL STUDIO SOUNDS IN CONCERT

BY WADE MACGREGOR

THE DEMAND for studio sound quality (and complexity) on stage has never been greater. Audiences at concerts expect performers to deliver the same, layered, fat and intricate compositions that they get on CDs. What's a MIDI homebody to do?

Thanks to today's technology, performers often achieve this effect by using MIDI control of the sequencing, sampling and synthesis. The same type of MIDI control also can help the house mixer maintain more consistency from one show to the next. For example, the use of MIDI control allows the reverb time to change for each chorus of a song, the delay to regenerate through the half-note rest in the bridge (and only there), and the whole effects rack to change to the next preset between songs without the house mixer ever having to look away from the stage.

Devices from a number of manufacturers help the house mixer manage all the effects. MIDI-controlled VCA mixers allow the level of mics, submixes or effects returns to be automated either dynamically or as presets (snapshots). Some of these rack-mount mixers even have auxiliary sends and are usually patched into the inserts of the house console's inputs or subgroups. MIDI gates or mutes, designed for channel-mute automation, can be inserted in a similar way. There are also full-scale mixing consoles that incorporate MIDI muting and even MIDI-controlled VCAs into their design. These consoles can help to manage complex mixing and control external effects devices by mapping the MIDI data to the console sends.

MAPPING MIDI

Mapping is simply changing a MIDI message of one type to a MIDI message of another type. For instance, a MIDI Note On message can be mapped to a MIDI Program Change message (i.e., making a reverb unit change to Preset 34 [program change 34, channel 12] when the mixer [input channel number 38] MIDI Mute is switched off and sends a MIDI Note On [C4, channel 16]). A mixer's MIDI VCA faders can be mapped to control reverb time, output balance, or many other functions requiring continuous control. The mapping can be done by dedicated MIDI mapping devices, or preset within some MIDI-controlled effects units.

There are also MIDI-controlled patchbays available that will remap MIDI messages as well as change the patching of MIDI control to and from devices. Some of these patchbays will also control the audio patching to allow the sends and returns of the effects to be changed even during the songs. This would allow the use of one auxiliary send to route the snare drum to a digital delay unit for the bridge, a reverb unit for the chorus, and having those same units fed from another auxiliary in the verse to add effects to the lead vocals. This can save having to dedicate an effects unit to one instrument or change auxiliary send levels throughout the show.

TAKING CONTROL

The various MIDI settings of the MIDI system can be assigned to a MIDI control unit, which can range from a personal computer to a book-sized dedicated unit with sliders, and even to smaller control units that simply produce MIDI program changes from a keypad. Program change commands can be mapped to control all the effects units in a processing rack. Any processor that has a MIDI IN connector and accepts program changes can be accessed — including reverb units, digital delay units, multieffects units, or MIDI-controlled equalizers. When looking for a MIDI controller that suits your needs, try it out with your equipment. A controller must be straightforward to program and operate or you'll end up hating it. Assuming you've chosen your controller wisely, once the cues are set up, a simple keypress can make dramatic changes in the sound and still allow the person mixing to concentrate on what the audience hears instead of having to stare at the front panel of the effects rack.

COMMON GROUND

Of course, MIDI control doesn't have to stop at just controlling the effects rack. MIDI Show Control will allow manufacturers to use a common protocol for addressing a wide range of theatrical systems, sound, lighting, video projection, stage machinery and special effects. This should allow

Illustrations by Normal Rockhard

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even more integration of remote-controlled audio sources into the expanding MIDI network.

To incorporate MIDI control into large scale productions, a number of manufacturers are producing boxes that translate MIDI data into a form that can be sent over 1,000 feet without degradation. This allows the MIDI devices to be physically positioned where it is most practical while still communicating with a central control system; or decentralizing control systems and allowing any number of people involved in the production to access any of the MIDI devices. For example, a concert's opening sequence, before the band comes onstage, can be started from the house mix position before it reverts to the musicians' control after they're onstage. Furthermore, the guitarist's effects rack settings can be adjusted from the house mix position; synchronization signals can be distributed to dedicated sequencers; and the more routine, or even the physically impossible tasks (like keeping the light cues on beat), can be automated.

Automation in live performance requires a reliable system that includes a back-up of all the crucial data. It is also necessary to retain sufficient real-time control so that the operator can compensate for the unexpected. This is not easy to do, and the more restricted the budget is, the more selective the use of automation must become. Live-sound reinforcement is the last area of audio production to accept computers and automation at every budget level. The increase in low-cost MIDI control devices and standards for the control of theatrical equipment will inevitably put MIDI control in the hands of most live-sound system operators. If you haven't been using those MIDI ports on your effects rack, perhaps now is the time to cable them up. The sooner you are familiar with how MIDI control works, the easier it will be to deal with the increasingly complex demands that are due to come.

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7 Power Amp Buying Tips

1. REMEMBER, a *professional* power amplifier is different from one made for home hi-fi use. It should be designed to be roadworthy and built to handle rigorous use (and abuse).

2. While specs are important, they must be read carefully in order to make sense. Power ratings will be the most important to you and your speakers. Specs are normally given for different loads (in ohms). The higher the ohms, the greater the load. If your speakers are rated at 8 ohms (the most common), that is the power spec that applies to you. For example, an amp's power specs will read as follows: 200 watts into 4 ohms or 100 watts into 8 ohms. Thus, while the manufacturer is truthful in stating that his amp is rated at 200 watts, this applies only to users who have 4-ohm speakers.

3. When comparing specs between manufacturers be sure that all the parameters are the same. For example:

Amp A: 8 ohms, 20 Hz to 20 kHz, 0.1% THD — 150 watts

Amp B: 8 ohms, 1 kHz, 1% THD — 170 watts

On the surface, Amp B has more output power. But if you look closely, you'll see that Amp B's power-rating tests were conducted using a 1 kHz tone only. It is easier to reproduce the 1 kHz midrange frequencies, which in turn gives Amp B a higher power rating. Amp A, on the other hand, was tested over the entire hearing range. It is quite likely that Amp B's measured power at full range will be less than that of Amp A.

4. Your amplifier should be designed with proper built-in protection. There must be adequate *short-circuit protection* to ensure that the circuits are protected from accidental overloads. *Thermal protection*, such as heavy heat sinks (internal parts that dissipate excess heat), and built-in fans can provide peace of mind. *DC fault protection* will prevent your speakers from getting fried should a power transistor fail, and *On/Off muting* can protect your system from transient AC power spikes that cause loud, annoying, and sometimes damaging, pops.

5. Know how much power you need. What size rooms will the amp be used in? How loud do you want to be? How much power can your speakers handle? How many speakers will you power?

6. Make sure your amplifier can fit into your rack. Professional power amplifiers are 19 inches wide — the standard rack width. The depth can vary, so be sure that the amp you choose will fit within its allotted space. Standard rack-mounted equipment's height is measured in "rack spaces": 1.75".

7. The power supply is the "engine" of the amp. This is the section that draws electricity from the wall and converts it into useful power for the amplifier section. A robust power supply gives a significantly higher 4-ohm power rating than the 8-ohm power rating. There are three types of power supplies found in amplifiers: common, split, and dual mono. Split and dual mono configurations provide a high degree of isolation and reliability. Plus, if a channel should fail, the other will continue to perform.

These tips were adapted from the booklet *Not All Power Amps R Created Equal*, published by QSC. For a free copy or further information, contact: QSC Audio Products, 1926 Placentia Ave., Costa Mesa, CA 92627. Tel: 714-645-2540.

IN A BAND VAN

Hang Ten (Stage Miking Tips)

Here's how to ride those stage sound waves without taking a spill

By Bruce Bartlett and Jenny Bartlett

MIC YOU LIKE A HURRICANE: Klaus Meine of the Scorpions

Photo courtesy of Sennheiser

MIC CLOSE W/DIRECTIONALS

Close miking increases the sound level at the microphone and makes the sound system louder without causing feedback. Use unidirectional mics to reduce feedback and leakage. Most of these mics boost the bass when in close. At low frequencies, this provides extra volume without feedback. Rolling off this extra bass with EQ reduces low-frequency leakage.

GET CLOSE TO THE SOURCE

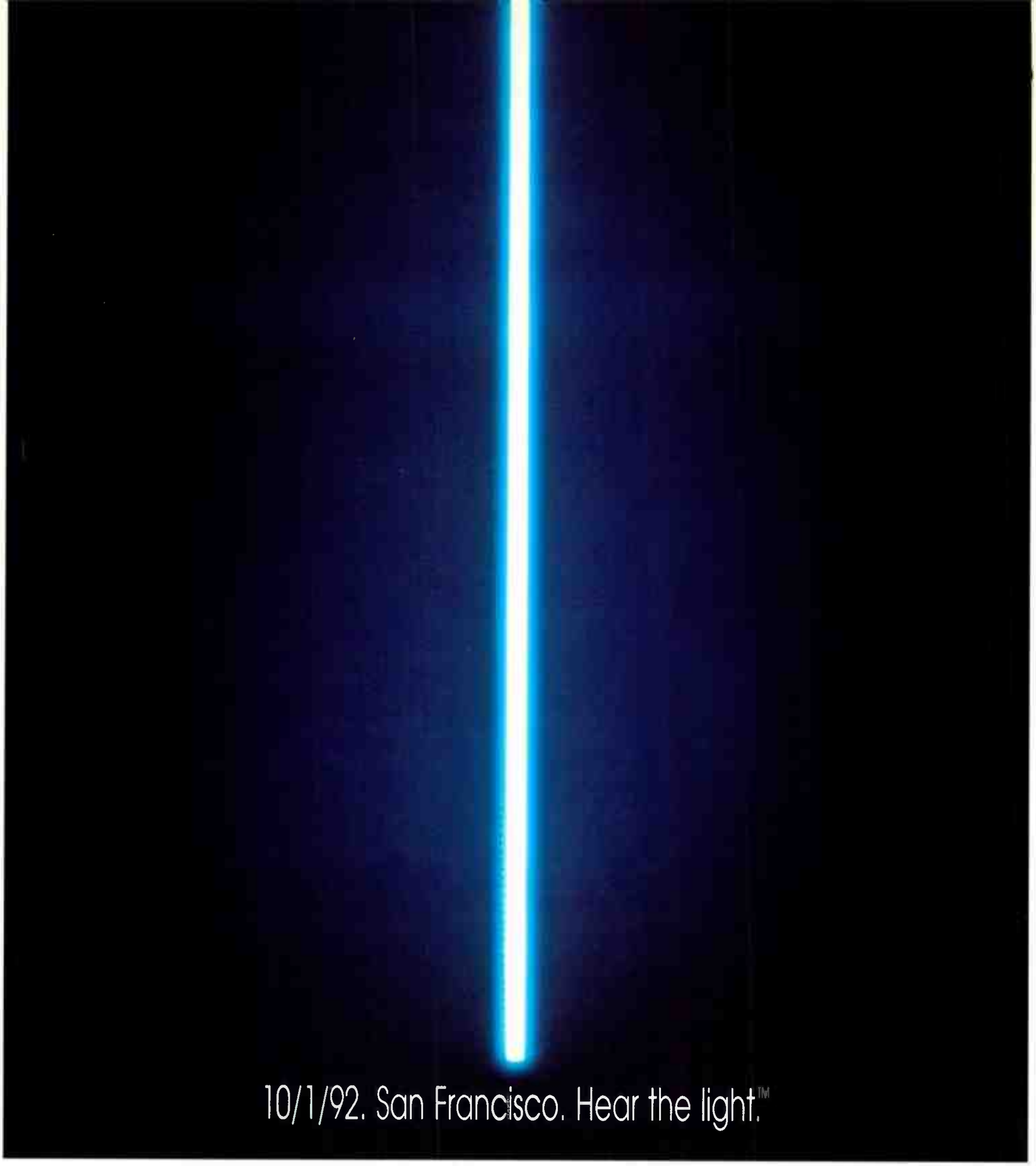
To increase gain-before-feedback, place the mic near the loudest part of the musical instrument — such as near the sound hole of an acoustic guitar, in the bell of a sax, or inside the shell of a tom-tom. Close miking can color tone quality, so use EQ to restore a more natural sound.

MIX CONTACT PICK-UPS

A contact pickup can solve feedback problems because it is sensitive to mechanical vibrations and not sound waves. A pick-up makes an acoustic guitar sound electric. To take advantage of the natural string sound, mix in a clip-on mini-mic. Try C-Ducer tape on an acoustic piano's soundboard.

USE FEWER MICS

The more mics you have on, the more likely you are to run into feedback. So turn down any mics that are not in use at the moment. On electric guitar and bass, try using direct boxes instead of mics. Rather than using a multitude of mics on a drum set, try using a single miniature omni mic in the center of the set. It will pick up the toms and cymbals all around it. You'd be amazed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IN A BAND IN A VAN

how good that single mic can sound. Boost the bass to add fullness. If the cymbals are too weak, lower them a few inches. You can hang another mini omni in the kick drum, as well.

USE NOISE-CANCELING MICS

A noise-cancelling or differential mic is designed to cancel sounds at a distance, such as instruments on stage or the monitor signals. Therefore, it can be an excellent choice for vocals. Such a mic can provide outstanding gain-before-feedback and almost total isolation.

USE WIRELESS CORRECTLY

Position the receiver on stage where it will pick up a strong signal, with less static and fewer dropouts. Inside most wireless mics is a trim pot that adjusts the output level from the mic head into the transmitter. Starting with the trim pot wide open, have the artists sing into the mic at the loudest level they will be performing. Gradually trim the pot until any distortion disappears.

PREVENT HUMS AND BUZZES

Keep mic cables well separated from lighting and power cables. If the cables must cross, keep them at right angles to reduce coupling between them, and separate them vertically. For best hum rejection, use twisted-pair mic cable with a braided shield. Float the connector-shell ground lug to prevent ground loops.

PREVENT SHOCKS

At times, electric guitar players can receive a shock when they simultaneously touch their guitar and a microphone that's on-line. This occurs when the guitar amp and mixer are plugged into widely separated outlets at different ground potentials. It helps to power all instrument amps and audio gear from the same AC outlets. That is, run a heavy extension cord from a stage outlet back to the mixing console (or vice versa). Plug all the power-cord pins into grounded outlets. An alternative is to plug the instrument amps and the mixer into local outlets, and place a foam windscreen on each vocal mic to insulate the guitarists from the potential shocks. As a bonus, a foam windscreen suppresses breath pops better than a metal-grille screen does. If you're running the electric guitar direct, use a transformer-isolated direct box and set the ground-lift switch to the position with the least hum.

TRY MINI MICS

Mini mics clipped onto musical instruments reduce clutter on stage by eliminating boom stands. Plus, the performer can move freely across the stage. Mini mics work well on drums, horns, acoustic guitars, flutes and grand pianos (usually taped to the underside of the lid). Get application notes from the manufacturers to be sure that you're using a particular mic in an application for which it is well suited. You can also reduce clutter when using regular-size mics by mounting them on drum rims and stands via clips.

REDUCE MONITOR LEVELS AND EQ THE MONITORS

Stage monitor speakers are the main cause of feedback. Try to keep the monitor levels down as far as possible. You can use a 1/3 octave (or narrower) graphic equalizer to notch out feedback from the monitors. Finally, check out the new in-the-ear monitors. Sound mixers love them because there's no leakage into the mics.

WIRELESS MICROPHONE BUYER'S GUIDE

There you are, singing your heart out on a set of songs that has done nothing but rock this smokey, sweaty club for the past two hours. The only problem is the microphone cord, which has found its way around your leg and is threatening to send you toppling over the center of the stage.

Some rock 'n' roll nightmare, huh? Well, no need to fret. Through the benefits of semiconductor technology, you can now obtain a smaller and sweeter sounding wireless mic at an affordable price. Plus, features such as true diversity, companding and squelch control have made wireless mics more attractive than ever to the average "band in a van" buyer.

First off, when you're in

Brand	Model	Price	Diversity/ Non-Diversity	VHF/ UHF	# of channels/ switchable?	Noise Reduction Type	Freq. Band of Channels (MHz)	Squelch	Comments
Audio-Technica *Free Lit #117*	ATW-1031	$650	Diversity	VHF	15/No		169-216	•	Guitar/instrument
	ATW-1032	$750	Diversity	VHF	15/No		169-216	•	Hand-held
	ATW-1032HE	$950	Diversity	VHF	15/No		169-216	•	Hand-held neodymium
	ATW-1031-M35	$775	Diversity	VHF	15/No		169-216	•	Sax/horn/instrument
	ATW-1031-M73	$790	Diversity	VHF	15/No		169-216	•	Head-worn
	ATW-1031-M71	$740	Diversity	VHF	15/No		169-216	•	Head-worn
beyerdynamic *Free Lit #118*	S170H	$1500	Diversity	VHF	1-12/No	patented companding	169-216	•	Includes TGX 480 Mic head, transmitter and rack mountable receiver & antennas.
	S186H	$4500*	Diversity	VHF	1-60/No	patented companding	169-216	•	
	SDM700	$6000**	Diversity	UHF	12 & Up/No	patented companding	450-950	•	
Nady *Free Lit #119*	RW 1	$530	Diversity	VHF	1	companding	170-218	•	Balanced XLR and unbalanced 1/4" output; rack mount.
	2000 VHF	$1600	Diversity	VHF	1	companding	151-216	•	Exclusive hiss mute circuity; bass boost.
	301 UHF	$800	Diversity	UHF	4/switchable	companding	920.3-927.5	•	
	RW-3	TBA	Diversity	UHF	4/switchable	companding	920.3-927.5	•	Rack mount; balanced XLR and unbalanced 1/4" output.
	950 UHF	$2500	Diversity	UHF	10/switchable	companding	490, 800 and 950 ranges	•	Exclusive hiss mute circuity; bass boost.

the music dealer's lair, don't be afraid to try out the microphone at a healthy distance, even if it means leaving the store to do so. If you're testing a mic for true diversity, you should go at least 100 feet away from the receiver, 200 if possible, checking for any "buzz zones" or "drop-outs" (interrupted output) along the way. True diversity mics will protect your signal from dropping out, even if you're in a building sheathed with many metal surfaces. Non-diversity mics, on the other hand, can be used effectively in most small club situations.

Audio quality can be tested by setting levels as high as possible and belting a high-pitched "ah" sound into the microphone. If you hear excessive noise or unwanted breathing, you may want to take a look at a different model. Be aware that even the best wireless mics will betray a slight bit of breathing in their output.

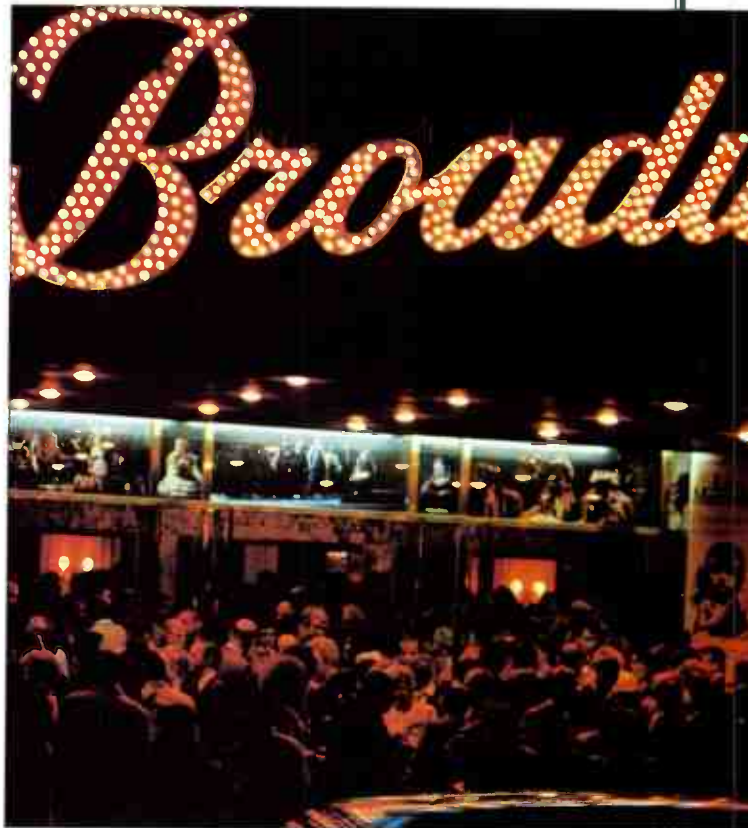
Finally, after you've purchased your wireless mic system, if your wireless is adjusted too low and the PA is too high, all you're going to get is a lot of unwanted noise. Conversely, if your wireless is too high and the PA is too low, you'll get too much distortion. *—Jon Varman*

Brand	Model	Price	Diversity/ Non-Diversity	VHF/ UHF	# of channels/ switchable?	Noise Reduction Type	Freq. Band of Channels (MHz)	Squelch	Comments
Sennheiser *Free Lit #120*	SKM 4031-90/ EM 2003-90	$3425	Diversity	VHF	1	Hi Dyn	168-216	•	All metal transmitter design; DC to DC converters in transmitters (regulates battery supply); helical filters allow the use of 20+ channels.
	SKM 4031-TVH/ EM 2003-TVH	$5835	Diversity	UHF	2/switchable	Hi Dyn	450-960	•	
Shure *Free Lit #121*	LS23/58	$532	Non-diversity	VHF	1	Shure	169-216	•	Handheld system; Also available in a diversity version featuring Shure's exclusive MARCAD reception
	LS24/Beta 58	$614	Diversity	VHF	1	Shure	169-216	•	Handheld system; also available in non-diversity version
	LS113/16	$635	Non-diversity	VHF	1	Shure	169-216	•	Headworn system (with WCM16 headset); also available in diversity version featuring Shure's exclusive MARCAD reception.
Samson *Free Lit #122*	Super TD	$1100	Diversity	VHF	16/fixed	dbx	170.245	•	Guitar wireless; dual power supply; detaches on the body pac
	Concert Series II	$600	Diversity	VHF	16/ fixed	dbx	To 213.20	•	Fully rack mountable; removable antennas
	UR-4 UHF Wireless System	$1749	Diversity	UHF	5/fixed	dbx	947.125-951.875	•	
Sony	WRT-830A/ WRR-820A	$2500	Diversity	UHF	up to 282/ switchable	companding	794-806	•	Phase Locked Loop; Dual antenna inputs; microprocessor controlled w/LCD display.
Telex *Free Lit #123*	FMR-70/WT-55	$595	Diversity	VHF	1	Pos-i-squelch™	165-216	•	Internal Squelch control
	FMR-100/WT-60	$900	Diversity	VHF	1	Pos-i-squelch™	150-216	•	
Vega *Free Lit #124*	R662/T677	$3749/ $1248	Diversity	UHF	4 /switchable	Dynex II	494-690	•	High-End Touring System
	R42A/T77	$3570/ $1086	Diversity	VHF	1	Dynex II	169-216	•	

*S186H Systems start at $4500

**SDM 700 Systems, minimum of 12 channels, start at $6000 per channel

Theater-In-The-Sound



Theatrical sound design offers insights for anyone dealing with sound on-stage

By Bob Ross

IF THERE'S NO BUSINESS like show business, where does that leave sound business for shows? At a very interesting point in its development, it appears, according to recent innovations in theatrical sound. In fact, sound design for theater is a flourishing industry, and can provide engineers with opportunities that rival the club and concert scene for challenge and excitement.

Theatrical sound systems have requirements different from those of concert systems, however. The objective for concert sound systems could be summed up in one word: BIG. In theater, the goals of the sound designer are more indeterminate. The first priority is intelligibility; wide bandwidth or high SPL's are of little use if a system cannot provide clearly comprehensible dialog without sounding conspicuously amplified. This last condition is crucial to traditional theater sound design. The subtlety with which vocal reinforcement is applied directly determines the extent to which the theatergoer suspends disbelief; if the audience becomes aware of the sound system, the designer has failed.

Setting the Stage

When designing a sound system for theater, one must consider the requirements of the script. Will the show need dialog reinforcement exclusively? Or does it require multiple SFX cues? Another important consideration is the size and layout of the venue. 50 seats, or 5,000? Is there a balcony? Is the theater in the round?

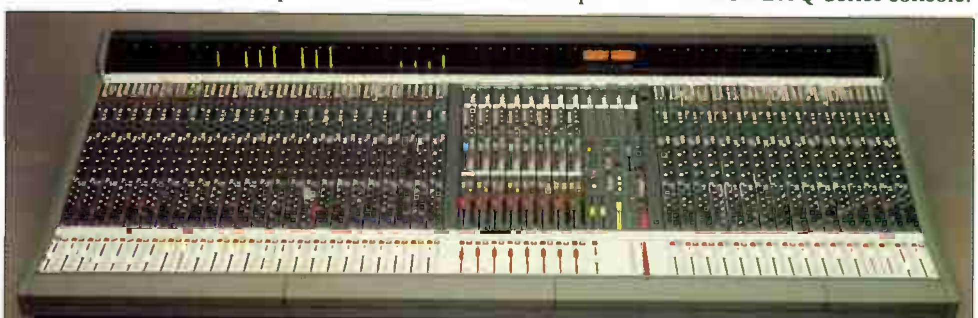
Must the system be transportable? Is it a touring show, requiring setup and teardown every month, every week, or every night? Or will the show run in one venue for its duration (which is about as permanent as theater gets)?

And, of course, there's the budget. Sound equipment for theatrical productions is almost always rented, and therefore shows up as overhead throughout the run of a show. When specifying the equipment complement for a show, be prepared to see your initial request brutally slashed by bottom-line conscious producers. Hey, that's part of the fun of sound design.

Showing Off

Equipment requirements for theater differ from those for concert sound. When mixing a live band, you are often struggling to limit yourself to the console's available input channels. In theater, where situations require (and the budget allows) numerous input sources, a sound designer may divide roles between several separate consoles, each with an attendant console operator. For small productions, it is unlikely that 24 or 36 inputs will even be used. However, numerous output channels are an absolute necessity; theatrical reinforcement consoles must be equipped with an output matrix, allowing the designer to send autonomous mixes to several separate outputs. These output channels will be utilized to feed the separate banks of speakers used to cover the main house, the under-balcony area, the upper balcony, etc., as well as any specialty speakers located onstage that may be used for specific effects or ambiences.

The Yamaha PM-3000 (or the smaller PM-2000) is a commonly-used console for theater productions. The Soundcraft Europa series is also gaining popularity; its individual noise gates and sweepable mid-EQ on each input channel are desirable features. Touring productions, concerned with ease of portability, have shown a fondness for the DDA Q-Series console.



BROADWAY BOUND: The Soundcraft Europa console is getting rave reviews from theater sound designers.

IN A BAND VAN

Active (processed) loudspeaker systems have become prevalent in theatrical sound. These systems utilize a separate electronics package containing crossovers, EQ, and protection circuits. The Apogee AE-5nc is one such system; its flat response and even coverage make it ideal for vocal reinforcement and music/SFX playback.

Actors in musical productions are often fitted with wireless microphones. These yield clean sound because of their proximity to the individuals speaking or singing; there is little room resonance or footfall from these body-mounted units. Cost, however, is a significant factor; one mic/receiver per performer doesn't come cheap. And that doesn't include the batteries, which have to be replaced after every performance.

Effective results can also be achieved by miking the stage in zones. For downstage applications, the Crown PCC-160 is the mic of choice. This half-hemispherical PZM, when arrayed along the lip of the stage, captures the bulk of the dialog with quality sound and low susceptibility to feedback. When specific upstage areas require special attention, shotgun mics are utilized, mounted above the proscenium in the lighting electrics or hidden in the scenery. Condensors such as AKG 451's or Sennheiser's MKH series, fitted with a shotgun capsule, are a common choice.

Those are the basics. As an evolving sound reinforcement specialty, theatrical sound design can teach even that "band in a van" some valuable lessons about getting a clear, clean sound on the concert stage — without breaking a leg.

[Thanks to NYC-based sound designers John Gromada and Timothy J. Anderson for their help in preparing this article.]

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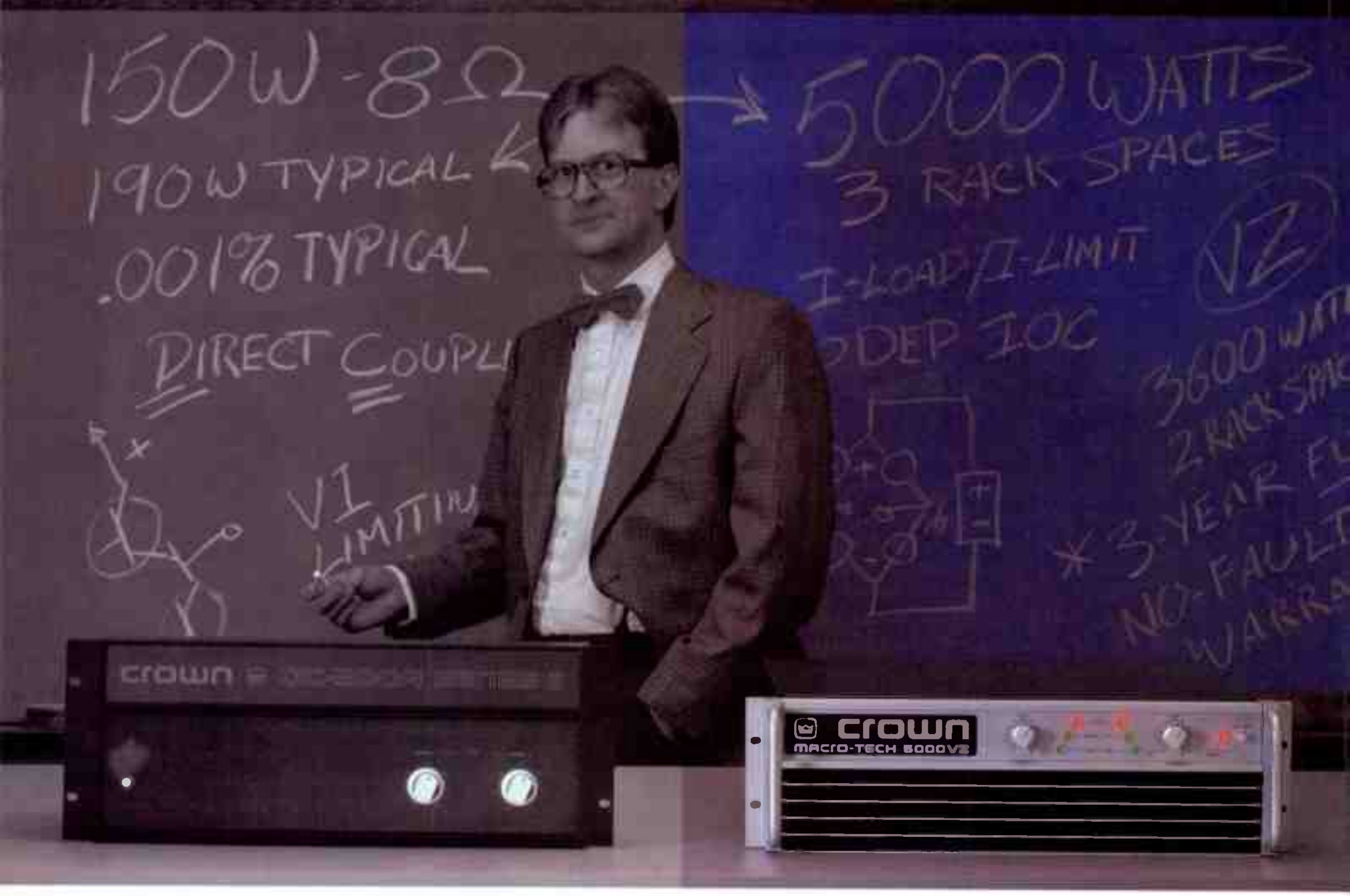

DC-300A II. Current version of the DC-300 introduced in 1967.

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Giving and Sharing

You don't always have to spend a lot for Macintosh music software. In fact, some of it can be absolutely free

BY TED GREENWALD

In the Land of Macintosh, music software companies take such good care of their customers that it's easy to fall into a complacent slumber of MIDI-dreams come true. With such a bounty of full-featured applications — sequencers, patch editors, sample processors, video scoring utilities, autocomposers, even object-oriented musical programming languages — users tend to forget that there's more to the world of software than commercial products. If you have a modem and membership in a local computer bulletin board, or one of the national networks (CompuServe, Prodigy, the Source, America OnLine, etc.), there's a universe of software at your fingertips that you won't find in any store.

This sort of software falls into two classes: freeware and shareware. Programmers of freeware rescind their copyright and donate their product to the public domain. There's no charge for the software, and it can be distributed by anyone so long as he or she doesn't charge for it. Shareware can also be distributed freely, but the programmer retains the copyright and expects to be paid by the users of the program. If you keep a shareware program, you are asked to pay a fee — usually $30 or less — directly to the programmer.

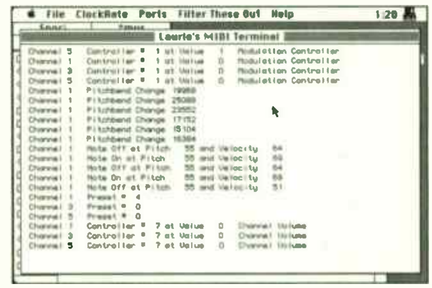
The nominal charge for shareware makes it possible for programmers to fix bugs, add features, and keep their work compatible with new Mac system versions. It also gives them a return on their time, expertise, and creativity, making it likely that they'll offer more shareware in the future. If you use any shareware program more than once, do the right thing and pay for it. Fee and contact information can usually be found in the program's "About..." box under the Apple menu.

Let's take a look at a few programs floating around in the Grid that are of particular interest to Mac-minded musicians, producers and engineers.

MIDI TERMINAL

Nobody works in a MIDI studio without witnessing more than his or her share of MIDI mysteries — you know, the kind that crop up as if on cue to ruin a perfectly good session. When that happens, only two things will get you out of the woods: a solid knowledge of your system, and a MIDI data analyzer. MIDI Terminal 1.2, a shareware utility programmed by Laurie Spiegel fills that latter need to a tee.

Spiegel calls her program a "MIDI-to-English translator," and that's exactly what it is. It translates incoming MIDI data into English so that you can see what you're sending into the computer. It's ideal for sussing out problems with a MIDI controller, figuring out what codes a footpedal or slider is sending out, and discovering which knobs on a given device send out sys-ex data. Moreover, the program can store incoming MIDI data for later viewing and analysis.



MIDI Terminal 1.2: A MIDI-to-English translator.

MIDI Terminal allows for selective filtering of various MIDI messages, so you can view the messages you're interested in in isolation. In all, 60 MIDI message types are supported — that's every command defined by the MIDI spec except for MIDI time-code and sample dumps. Shareware payment ($25) and inquiries can be addressed to Laurie Spiegel, 175 Duane St., New York, NY 10013.

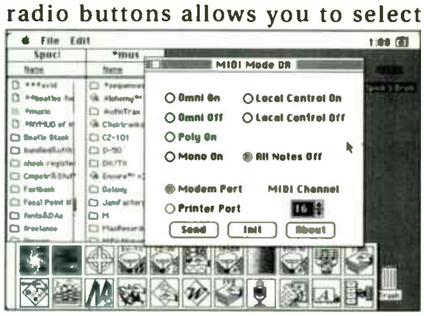
MIDI MODE DA

In the middle of a session, scouring the hierarchical menus of your Maxi-Whizbang 1000 for the Local On/Off switch is the last thing you want to do. The MIDI Mode DA, a freeware product from Austin Development (227 Main Street, San Raphael, CA 94901), helps make this task, and a few others,

In the middle of a session, scouring the hierarchical menus of your MaxiWhizbang 1000 for the Local On/Off switch is the last thing you want to do.

easy as pie.

The DA presents a box offering radio buttons for omni on, local on, local off, poly on, mono on, and all-notes-off commands. Another pair of radio buttons allows you to select either your modem or printer port, and you can set a transmission channel. Once you've enabled the desired buttons, simply click on the "send" button and the Mac squeezes out the corresponding code. MIDIots will find the MIDI Mode DA particularly handy for sending out an all-notes-off command to silence pesky stuck notes.



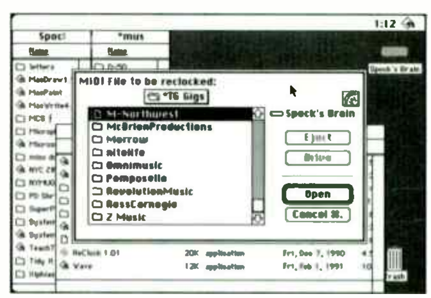
FREE & EASY: The MIDI Mode DA

RECLOCK

One of the limiting factors on sequencer-produced music is the difficulty of working on pieces that don't adhere to a strict tempo. While it's easy to record in "real time," playing as freely as you like, the sequencer's internal metronome always ticks away at a steady tempo. When you look at what you've recorded, the note data doesn't line up with the measure and beat structure you had in mind. ReClock v1.01, by Doug Wyatt of Opcode Systems, offers a way out of this dilemma.

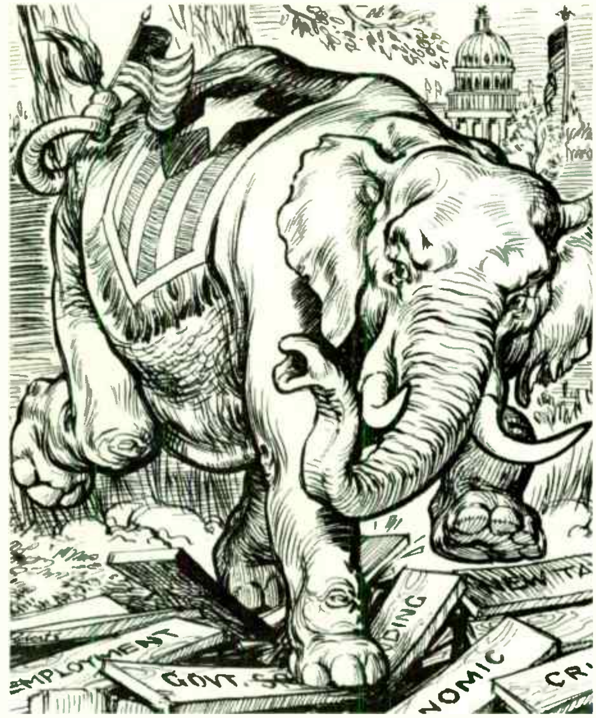
ReClock interprets one track of your sequence as a metronome track, of which each note-on corresponds to a quarter note. Using this information, it rewrites the sequence with a new tempo map that conforms to the metronome track, and *voila*, your sequence's beats and measures fall into line with the *rubato* musical passage. All you have to do is prepare a metronome track on an unused MIDI channel and export the sequence as a standard MIDI file. Then load it into ReClock, selecting the metronome track's MIDI channel when the program prompts you to do so. Lo and behold, out pops a new MIDI file with a revised tempo map.

Preparing the tempo track may be an arduous process, depending on the material you begin with. If there's a kick-and-snare part that contains little else besides quarter-note values, you need only eliminate all non-quarter-note beats and manually add any that are missing. Another option is to copy the track that's closest — say, a rhythmic keyboard part — and manually strip out everything but a quarter-note framework, once again taking care to add any beats that are missing. Of course, you might find it possible simply to overdub the metronome track in real time while listening to the previously recorded material, but this is more easily said than done. In any case, if your metronome track contains each and every quarter note in an accurate relationship to the other tracks, ReClock works like a charm.



Keep time with Opcode's ReClock v1.01.

ReClock is shareware, requiring a $15 payment if you decide to keep it. Inquiries and payments can be addressed to Douglas S. Wyatt, c/o Opcode Systems, 3950 Fabian Way, Suite 100, Palo Alto, CA 94303. EQ



IF ONLY GETTING IN SYNC WAS THIS EASY FOR EVERYBODY.

You see it every day. The quarterback lofts the ball over the wide receiver's head.

The cab pulls up to the terminal as the airplane pushes back from the gate. The guy on tv moves his mouth before the words come out. Classic cases of a world slightly out of sync.

In the audio environment, it's the same problem, times ten. The machines you use, most often don't speak, read, or respond to the same language. You can't just line them up, push the "go" button and have them all operate in unison. How, then, do you get the various formats of audio and video tapes to work in sync?

Well, you probably know about those magic "black boxes" called synchronizers that the guys with the mega-budgets use. What you might not know is that this same magic is now available, and easily affordable, for any home studio.

The breakthrough comes thanks to TASCAM's ATS-500, priced below $800.* It's a two-machine chase-lock system with a SMPTE time code generation capability that can resolve to video sync using your existing VCR and 8 track.

It's got all the professional features—an offset function, wide-band reader, jam sync—yet it's easy to understand and operate. It's also auto-calibrating and requires no mastery of hidden screens or functions. And, it not only works with TASCAM's serial control transports, but by adding TASCAM's IF-500 interface, it'll work with other parallel transports as well.

To see the ATS-500 first hand, just sync up with your nearest TASCAM dealer. He'll show you how easy synchronization can be.

TASCAM

*Suggested Retail Price ©1992 TEAC America, Inc., 7733 Telegraph Road, Montebello, CA 90640 213/726-0303

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Goin' Outta My Head

A do-it-yourself experiment with binaural recording

BY BRENT A. VERRILL

You may be familiar with some of the inadequacies of conventional stereo reproduction — particularly images that can be described as flat. Although the new "dimensional processors" attempt to solve that problem, binaural recording (where stereo sound is recorded into a model of the human head, with omnidirectional microphones placed where sound would enter the auditory canals) is a simpler and less expensive alternative. Although binaural recordings must be listened to on headphones for the proper effect, that's an inconvenience you'll probably accept once you experience the strikingly realistic image of binaural material.

The mechanics of how humans localize sound sources has been the subject of much research, but the main factors are:

- Time delays
- Intensity differences
- The structure of the head and ears
- The interaction between hearing and the other senses

Binaural recording is effective because it takes all these factors (except the last) into account, and applies this knowledge to recording and playback practices.

TALK ABOUT A HEAD TRIP...

Construction of a binaural recording device is easy and inexpensive (mine cost less than $25). The condenser microphone capsules specified in the parts list have surprisingly good fidelity considering their cost, and they're about the same diameter as the average auditory canal.

Referring to the schematic (Fig. 1), the various electronic components bias the mic capsules from either a 9-volt battery or from phantom power (as available on many mixing consoles). If phantom power will not be used, ignore the components inside the "phantom power" box; if battery power will not be used, ignore the components inside the "battery" box. Use shielded cable to connect the circuit board to the mic capsules and XLR plugs.

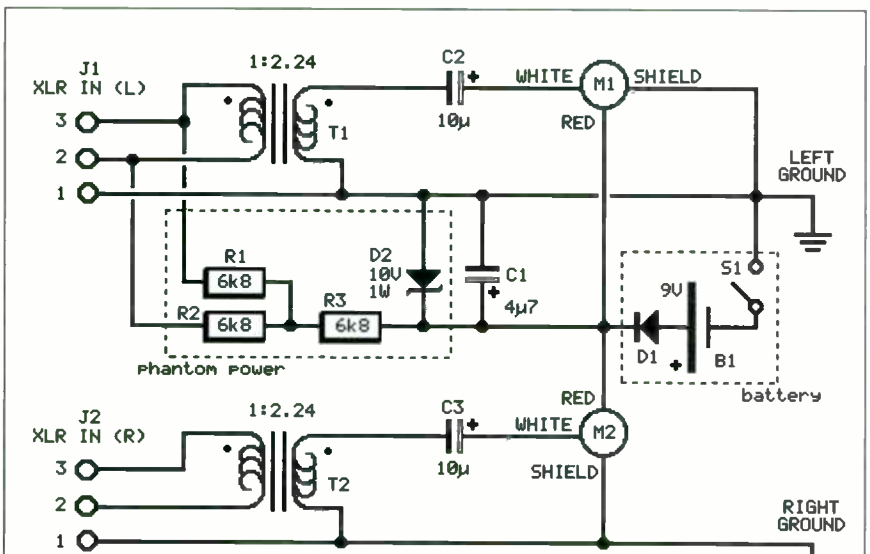

Figure 1

Fig. 2A

Fig. 2D

EARS TO YOU

The next step is to make some ears to stick on a styrofoam wig head. It's easiest to place the mics at the opening of the auditory canal; trying to simulate the entire auditory canal is beyond the scope of the average experimenter.

Making the pinnas (the projecting outer part of the ear) is the fun part. This part is crucial to proper operation, so take the time to do a good job. Casting a suitable pair of pinnas in latex would be near ideal, but I achieved good results by starting with a cardboard oval and fashioning an ear-like shape around it from paper, by trial and error, until I came up with a desirable shape (Figs. 2A and 2B). This became the template for the rest of the construction.

The fewer the structures present in the pinnae, the more difficult it is for subjects to localize sound sources. So I picked three major structures to model: the helix (Fig. 2C), the tragus (Fig. 2D), and the antitragus (Fig. 2E). The tragus and the antitragus form a gap called the intertragal incisure (use

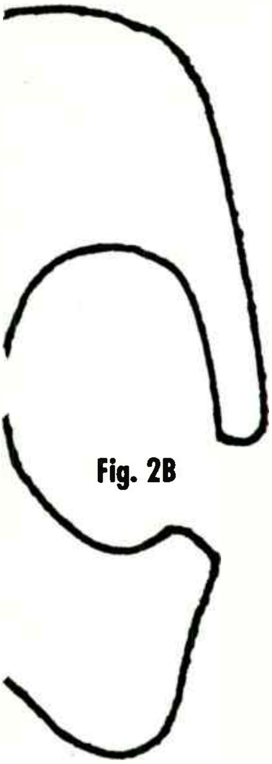

Fig. 2B

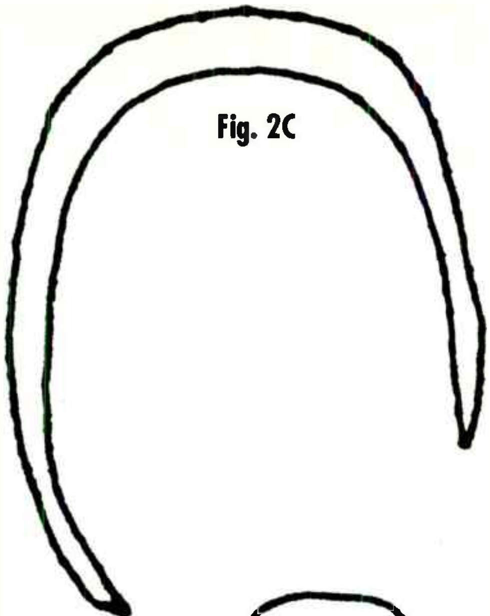

Fig. 2C

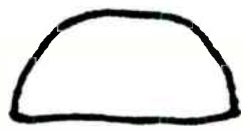

Fig. 2E

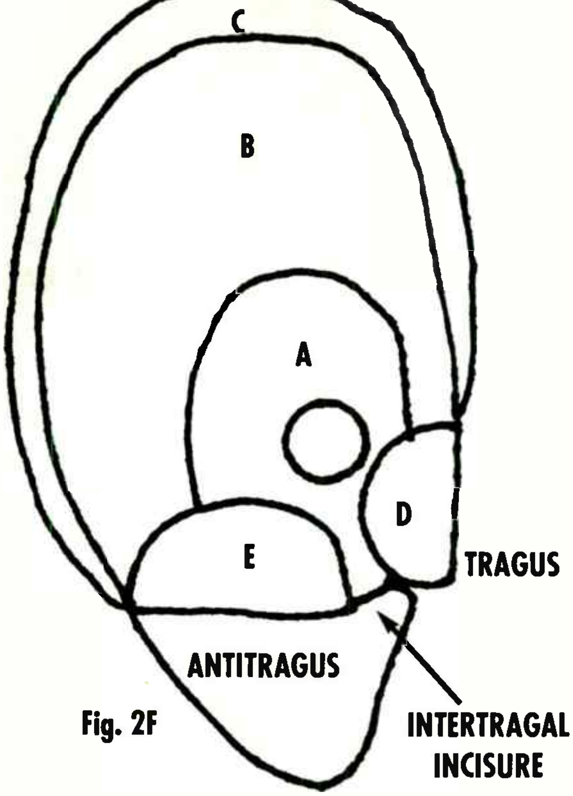

Fig. 2F

that to impress your friends). More information and helpful diagrams can be obtained from an afternoon in your local library — check out research under the topics Auditory Perception, Physiology and Anatomy.

Sheet neoprene (ask your local art supply store) or leather are good, easy-to-work-with, pseudo-pinna materials. They are fairly thick, so you can shave sections to provide a surface against which other parts can be glued. For example, I cut the base oval (Fig. 2A) at an angle to provide a wide, angular edge on which to glue the main hornlike structure (Fig. 2B), and shaved portions of the underside of the tragus and antitragus to eliminate sharp edges. Figure 2F shows the finished pinnas. The basic idea is simple: end up with something that looks like a human ear. Don't worry if both ears aren't exactly the same (neither are yours).

To increase the difference between the front and rear images, I put some foam behind both of the ears to further attenuate high frequencies arriving from the rear. This is somewhat artificial-sounding, but works fairly well for giving location cues.

The styrofoam head, found at any wig shop or hairdresser, is simple to modify: Cut the head in half vertically in a line through the ears, and hollow out receptacles for the battery, wires, XLR connectors, and circuit board. Glue the pinnas on the sides where ears should be, place the capsules through the holes in the base oval of the pinnas, and you're just about ready to record. (Don't use super/crazy-glue adhesives on the styrofoam; it will melt.) Shock-mounting the head for placement on a mic stand is a good idea, but not essential so long as you don't hit the stand.

Parts List

Radio Shack part numbers are indicated with #.

CAPACITORS (35 or more working volts)	
C1	4μ7 or 4.7μF electrolytic (#272-1012)
C2, C3	10μF, tantalum or electrolytic (#272-1013)
RESISTORS (2% or 1% tolerance, metal film recommended)	
R1-3	6k8 or 6.8k
DIODES	
D1	1N4001 or equivalent (#276-1101)
D2	10V, 1W Zener diode
JACKS	
J1, J2	XLR male panel mount jack
OTHER PARTS	
M1, M2	condenser microphone capsules (#270-092)
T1, T2	10,000Ω to 2,000Ω audio impedance transformer
S1	SPST on-off switch (#275-612)
B1	9V battery
Misc.	Battery connector, circuit board, styrofoam wig head, 1' sq. of porous sheet neoprene (or thick leather, foam, etc.), solder, wire, etc.

LOOKING A HEAD TO RECORDING

The recording process is simple: Record each mic's output onto a separate channel. To retain the binaural qualities do not add effects (e.g., reverb or delay), as they destroy the clarity of the sound images. Experiment with various recording techniques; set up the head at a concert, outdoors, at a restaurant, or wherever, and see what produces the most life-like sound. The only rule is that there are no rules! Enjoy your binaural head (and don't forget to name it, lest it feel alienated).

Brent Verrill is in his last semester at the University of Miami's Music Engineering program. When he graduates, he will be on a full-time job hunt (gratuitous plug). He is currently wishing he had a really big computer to see what this virtual reality thing is all about.

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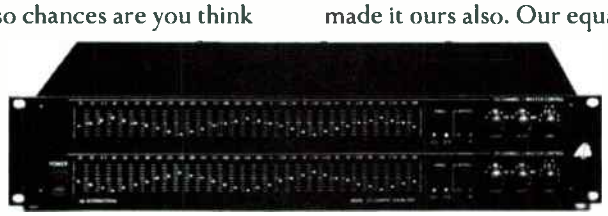

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Acoustically Yours

Don't overlook the most basic of elements in the recording process — *sound*

BY J.D.SHARP

Electro-Voice Sentry 100A Monitor

In the not-so-distant good old days, most recording was accomplished within the confines of commercial recording studios. Even though the science of acoustics and the instrumentation to make acoustical measurements were not as advanced as today's offerings, a considerable amount of time and energy went into the design of acoustical treatments and the construction of good-sounding studios.

In our current era, a great deal of recording activity takes place in "project" environments. These can be anything from a garage or a section of a living room to a fully-equipped studio built on a personal scale. One problem that plagues nearly all these studios is a lack of attention to acoustic realities.Where should that project studio owner begin?

The general goal in a monitoring situation is accuracy. In practical terms, this means that recordings and mixes translate well to other monitoring venues, whether they be professional studios, living rooms, or car cassette players. Several things stand in the way. The most obvious is that the response of a given room is not equal at different frequencies.

MEASURING UP

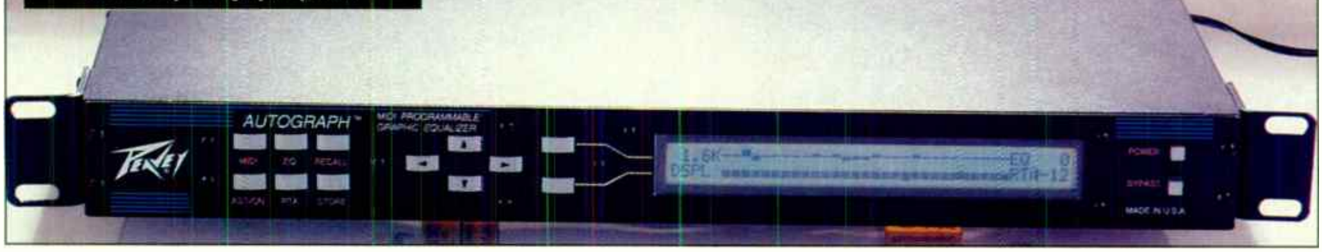
The traditional low-tech solution is to *equalize* the energy in various bands. A *real-time analyzer* (RTA), also known as a spectrum analyzer, is the test tool most often used to make the measurement of energy in each band. These are available in both octave and 1/3rd-octave bandwidths; for our purposes, anything less than a 1/3rd-octave analyzer is effectively useless owing to imprecision. Thirty bands of audio energy are displayed, and the frequency bands correlate to those on properly-designed 1/3rd-octave equalizers (27-, 30- or 31-band). These frequencies are called ISO centers.

Three moderately-priced RTAs currently available are the SA-3050A from Audio Control, the 30M8 from Gold Line and Rane's RA 27 Realtime Analyzer system. Using them is simple enough; take the pink noise output, run it through the playback system, and put the instrumentation mic (included with most analyzers; otherwise use a high-quality extended range omni) at the listening position. Multiple memories in the Gold Line and Audio Control units allow measurements at several points in a room, helping to get a better picture of problem frequencies. Some equalizer manufacturers have built an RTA function into their EQs, as in Peavey's Autograph and Rane's RE 27. A pink noise source is included, and either LEDs or an LCD is used to show the relative energy in each band. The programmable Autograph even provides a function to average together several readings and automatically generate a correct reciprocal curve that flattens response.

All this sounds simple enough, and it is true that results from an equalized room will be an improvement over a raw acoustic environment. Sadly, this is far more effective for taking care of feedback in a live venue than for turning a studio into an ideal recording room. In the grand scheme of things acoustical, this is a band-aid treatment.

Even if energy has been evened out, three crucial problems remain: different reverberation times, resonant frequencies, and reflections. These factors are interrelated. For instance, a room *mode* (resonance) exists when a particular frequency's wavelength is twice as long as the room (this is just the primary mode; there are many others). The sound bounces off the far end and returns just in time to reinforce the new signal emitting from the speaker, adding to the energy and causing this frequency to decay slower than nonresonant frequencies. Reflections can cause a different mess. If a reflection at a given frequency comes back exactly out of phase with the original signal, a cancellation will take place. If this frequency happens to be in the middle of the bass guitar's range, or right where the kick drum's fundamental falls, it is effectively impossible to accurately mix the low end. If the mixing position happens to be situated exactly where a low-end cancellation falls, bass can endlessly be boosted with lit-

The Peavey Autograph Equalizer



Crown's TEF-20 Acoustical Analysis System

tle audible difference — but when your tape is played in another studio, the low end may blow the speakers out! High end reflections, especially those that take place right at the monitor speakers, cause a near-random pattern of phase cancellation and reinforcement in the high end because of the short wavelengths involved.

ACOUSTICAL SOLUTIONS

This is by no means an exhaustive list of acoustical pitfalls, but those mentioned serve to point out that the subject of acoustics is a deep one, so much so that this article can only point in a few promising directions. The answer, needless to say, does not consist simply of tacking up egg cartons or carpets (they're only effective in the upper midrange).

One way to deal with these problems is in the initial construction of the studio. If you're lucky enough to be building from scratch, consult Jeff Cooper's *Building a Recording Studio*, a thorough treatise on the subject that includes both specific plans and excellent detail diagrams showing exactly how things should be done. Unfortunately this publication goes in and out of print, and it's "out" right now.

Most will be stuck with treating a room as it sits. Each frequency band demands a different remedy, but the most difficult task is determining where the problems lie. An acoustical analysis system like Crown's Macintosh- or PC-based TEF 20 can deliver a clear picture of both reflection patterns and the reverberation curve (called the RT60 curve) so that frequency-specific solutions can be devised. RT60, by the way, is the time it takes for a given frequency to decay 60 dB from its original level. Usually, you'll find these devices in the hands of accomplished acousticians and acoustical architects, so if you're not planning to do this sort of analysis on a regular basis you may want to acquire the services of such a technician rather than investing in the hardware and the learning curve to employ it effectively. Gold Line's latest $1,500 DSP-based analyzer provides an RT60 option, as well as displaying over an 85 dB range. They also make the GL60, a Reverberation Time Meter that measures RT60 decay times at six different frequencies — not as sophisticated as the Crown TEF by a long shot, but far better than shooting in the dark.

Or you can go with a common-sense approach, which is less scientific but may still be very helpful. For instance, the primary modes of any room can be calculated by simply measuring it. You'll find information on this in F. Alton Everest's *Master Handbook of Acoustics*. This will identify the likely frequencies requiring treatment in the low end. A bit of intuitive work regarding mids and highs may also help. What you're after is to minimize early reflections from the immediate vicinity of the monitor system, yet leave the room lively

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Craig Anderton, Sound on Sound

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enough to retain a slight sense of ambience. It is also important to improve *diffusion*, which keeps the mid and high energy field relatively even as you move about.

NAILED TO THE WALL

Fortunately, a number of off-the-shelf materials can help. Acoustical treatments from Sonex, Cutting Wedge, Acoustic Sciences Corporation and others can effectively dampen and even out mid- and high-frequency response (for instance, Sonex products are most effective between 250 Hz and 4 kHz). Each manufacturer has specification sheets showing exactly which frequencies their product deals with, and how effectively.

Coping with low end is a stickier problem. Past solutions included building enormous *bass traps* into the ends or sides of studio walls, but these solutions are not applicable to personal recording contexts. Tube Traps from Acoustic Sciences Corporation are a variety of affordable free-standing and wall- or ceiling-mounted bass trap products that combine low frequency absorption with diffusion of mid and high frequencies. Their Super 16 Tube Trap, for example, is a 16-inch-diameter, four-foot-high cylindrical treatment that's effective down to 25 Hz. ASC provides a host of backup materials explaining the nature of the problems, the specific characteristics of each of their products, and how to best use those products.

A final less-than-perfect, but very cost effective, solution is to move your monitors. Near-field monitoring seeks to minimize the role the room plays in the sound that reaches one's ears, by putting the listener close to the sound source. This is far more effective if monitoring is kept low, since higher levels "excite" the room modes and lower intelligibility. The trade-off is that most compact monitors don't offer great bass response. You might look at full-range models such as Tannoy's PBM 8s and System 8 NFMs; Peavey's PRM308S' from their AMR division; JBL's 4208s; or Sentry 100s from Electro-Voice. All these monitors share the characteristic of being small enough for near-field positioning without sacrificing too much low end. If your studio is a control room and recording room in one, consider placing your monitors away from either end and toward the middle — this increases the time it takes for reflections to reach the listening position and can contribute to intelligibility.

Most aspects of studio performance can be improved simply by adding or replacing hardware; if you don't like your reverb, for instance, you can get a better one. But acoustics need to be handled differently. Here you're dealing with the response of the room, and this requires architectural and/or acoustical solutions. If you seek to compete with the big boys and achieve the same degree of clarity and punch from your recordings, you need to arm yourself first and foremost with knowledge about the science of acoustics. After that, take some measurements, and only then work out the answers. Acoustics is an area where "faking it" will get you nowhere. EQ

Of Signals and Noise

Photo by Peter Monroe

If you are purchasing by signal-to-noise specifications, then buyer beware

BY LEN FELDMAN

Let's begin with an example. If I told you that a given amplifier was said to have a signal-to-noise ratio of 100 dB and that I measured the amplifier and found that it actually had a signal-to-noise ratio of only 80 dB, you'd probably conclude that my test equipment was faulty or that the manufacturer was engaging in what I like to call "specsmanship" (also known as lying). Surprise! Both the manufacturer and I would be telling the truth!

So begins our continuing discussion of basic audio performance specifications, and one of the most confusing and misunderstood specifications published by manufacturers of audio products: signal-to-noise ratios.

SIGNAL REFERENCE LEVELS

The discrepancy between my findings and the manufacturer data arises from the fact that not all of us use the same reference signal levels in arriving at the signal-to-noise ratio. Some years ago, the Electronic Industries Association (EIA) issued a Standard of Measurement for audio amplifiers. That standard dictated that signal-to-noise ratios should be measured relative to a constant 1-watt output.

Let's take another look at the amplifier that boasted a signal-to-noise ratio of 100 dB but that I measured as having a signal-to-noise ratio of "only" 80 dB. That particular amplifier had a power output rating of 100 watts per channel, when driving 8-ohm loads. The manufacturer (who, incidentally, is not obliged to use the voluntary EIA Amplifier Measurement Standard) simply used his rated 100 watts per channel as a reference level against which to measure signal-to-noise. Using that reference level, the manufacturer legitimately came up with the figure of 100 dB for signal-to-noise. I, on the other hand, used a reference level of 1 watt, as prescribed by the EIA Standard. Since 1 watt is exactly 20 dB lower than 100 watts, my signal-to-noise reading was — you guessed it — 80 dB. It's easy to understand why many manufacturers prefer to use their rated power output as a reference level against which to measure signal-to-noise ratios. Doing it that way yields a higher (better?) number. One hundred dB sounds a whole lot better as a signal-to-noise specification than does 80 dB, right?

WHY THE ONE WATT REFERENCE LEVEL

Higher numbers notwithstanding, the EIA had a very good reason for standardizing a 1-watt output as the reference level against which to measure signal-to-noise ratios. For one thing, unless we're talking about huge sound reinforcement systems used in rock concerts and the like, the average power delivered under typical studio monitoring conditions is of the order of 1 watt, even though peaks may well be 10 or even 100 times as powerful. (That gets us into the subject of dynamic range, which I'll get to presently.) But there's a more important reason why the EIA decided that all amplifiers, regardless of their full power output rating, should use the same reference level when measuring signal-to-noise ratios. The following example will illustrate why this is so:

Suppose you had a couple of amplifiers — one with a power output

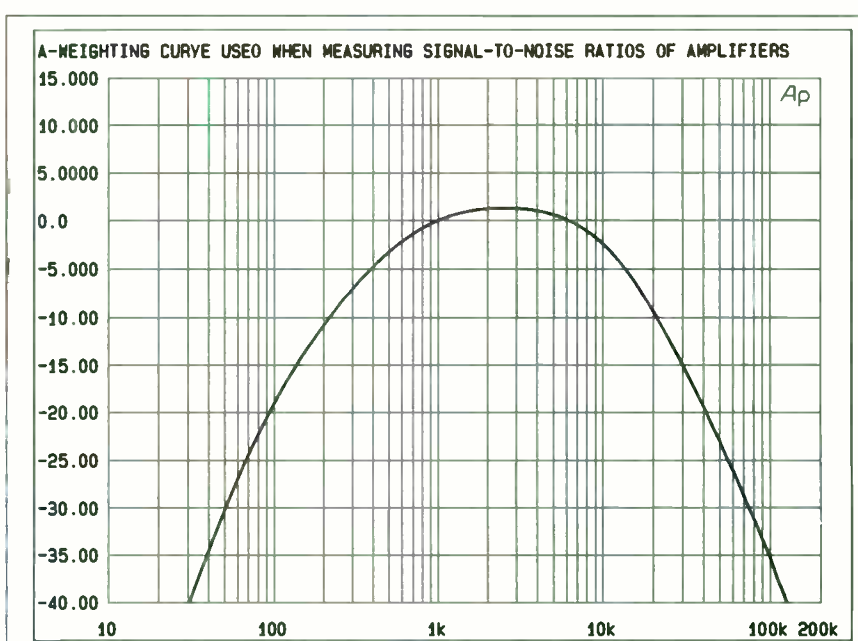

Figure 1

rating of 10 watts per channel, the other with a power output rating of 100 watts per channel. If both amplifiers produce 1 milliwatt of wideband noise when no signal is applied, and if each manufacturer measured signal-to-noise with reference to the rated output of his product, the maker of the 10-watt amplifier would come up with a signal-to-noise ratio of 40 dB while the maker of the 100-watt/channel amplifier would come up with a signal-to-noise ratio of 50 dB. Yet, both amplifiers produce exactly the same amount of noise! Under these circumstances, the maker of a more powerful amplifier would always come up with a better signal-to-noise figure for his product.

Worse still, if that 100-watt amplifier produced 2 milliwatts of noise while the 10-watt amplifier produced only 1 milliwatt of noise, the 100-watt amplifier would still come up with a better signal-to-noise number (47 dB), while the 10 watt amplifier would only show a 40 dB signal-to-noise even though the higher-powered amplifier actually produced double the noise power of the lower-powered amplifier! By referencing all amplifier signal-to-noise ratio measurements to the same nominal 1-watt output level, interpreters of such published specifications have a fair chance of comparing "apples to apples" instead of "apples to bananas."

CURVES ADD FURTHER COMPLICATIONS

If you are a careful observer of published specifications, you may also have noted that some statements of signal-to-noise ratios are accompanied by the statement, "A-weighted Signal-to-Noise Ratio." Ever since scientific studies of human hearing began, we've known that our ears do not respond equally to all frequencies. This is especially true at low levels of sound — the levels at which amplifier noise might typically be reproduced. Our ears are much more sensitive to mid-frequencies (with sensitivity peaks at around 2,000 Hz) than they are to low and high frequencies. Thus, if an amplifier produces a fair amount of hum, at power line frequencies of 60 Hz (or harmonics at 120 Hz or 180 Hz), such hum components will be less annoying and even less audible than noise of the same amplitude reproduced at mid-frequencies. Tests of human hearing sensitivity led to the development of several so-called weighting curves, the most popular of which is the A-weighting curve (shown in Fig. 1). If you measure signal-to-noise ratio using an A-weighting curve (as the EIA Standard requires) you will almost always come up with a better (higher) signal-to-noise figure than you would if the measurement were made giving equal weight or importance to noise at all audio frequencies.

In a typical example, I measured the signal-to-noise ratio of a very high quality professional amplifier and plotted the residual noise content as a function of frequency. As is almost always the case, the chief noise component occurred at the power line frequency (60 Hz). Without the use of an A-weighting curve, that component was about 85 dB below the reference level of 1 watt. Inserting the A-weighting network resulted in a 60-Hz component that was now down around 112 dB below the reference level — an "improvement" of around 27 dB for that particular "noise" component. The point of all this is that when comparing signal-to-noise ratios of competing products, it's important to make certain that the reference level against which the signal-to-noise is being measured is the same for both products (or, if it is not, that you translate the readings to the same reference level) and to determine whether a weighting network is or is not being used to arrive at the signal-to-noise figure for both products.

> It's easy to understand why many manufacturers prefer to use their rated power output as a reference level. Doing it that way yields a higher (better?) number.

DYNAMIC RANGE

Many audio professionals are often confused by the term dynamic range. They tend to think that it means the same thing as signal-to-noise ratio, and in fact, often, the figures quoted for dynamic range of an audio product are similar — if not identical — to the figures quoted for signal-to-noise ratios.

Dynamic range is simply defined as the difference between the loudest (or most powerful) sound that an audio product can produce and the softest or quietest sound that the same audio product can produce without the sound being "buried" in the residual noise. In determining dynamic range, the manufacturer once again must establish a maximum reference level. In the case of an analog tape deck, for example, that upper reference level might be the level at which playback of the tape results in a total harmonic distortion of 3 percent. In the case of an audio amplifier, the manufacturer might use the power level at which the rated distortion level of the amplifier is reached, or some arbitrary power level at which an even higher THD level might occur (e.g., 5 percent, or even 10 percent). As for the lower boundary of dynamic range, that's generally defined as the level at which the reproduced signal can still be distinguished from the residual noise level of the product being measured.

In these days of digital audio, compact disc players often exhibit dynamic ranges in excess of 90 dB — sometimes as high as 100 dB or more. Clearly, if you hook up such a program source to an amplifier whose dynamic range is limited to 80 dB, the full dynamic range of the compact disc player will not be realized. At the moment, there is no industry agreed-to standard for measuring the dynamic range of audio amplifiers, which may account for the fact that you'll rarely see dynamic range listed among the published specifications relating to amplifiers. Still, with digital audio resulting in ever-higher dynamic range capabilities, it would be a good idea if the industry did arrive at some standard and meaningful way of specifying this important parameter for amplifiers, mixers, preamplifiers and other audio electronic components that are expected to handle and process today's wide dynamic range digital program sources. EQ

Audio School Daze

Today there is a vast choice of audio schools — some good, some not so good

BY MARTIN POLON

Audio education — a curiosity. So curious, in fact, that some would consider the phrase to be an impossible contradiction in terms. In the "good old days," audio education meant spending a great deal of time with one's head glued to several worthwhile books such as Oliver Read's *The Recording and Reproduction of Sound*, Howard Tremaine's *The Audio Cyclopedia*, and Harry F. Olsen's *Musical Engineering*. One learned the hard way to take a long hallway with a toilet at one end and from it create an "echo" chamber by means of a dynamic microphone suspended over the bowl and a portable enclosed coaxial speaker system at the other end of the hall. One became an indentured apprentice at a recording studio — or at least it felt that way. There were no audio schools of any kind, shape or form. One was either self-taught, or a distinguished graduate of the University of Street Smarts — on-the-job experience. There were a few sparse curriculums in engineering or physics with a specialization in audio at some four-year state schools and one or two good two-year programs by the 1970s, but the explosion of audio education was a phenomenon of the 1980s.

Today, there are nearly 250 audio-education programs in the United States. They range from four-year programs in music and engineering at public and private universities that closely resemble the European "Tonmeister" training system, to two-year vocational repair and maintenance courses at community colleges, to private-recording-studio curriculums, to short courses taught by industry professionals at extension programs. Overall, their scope and diversity is almost dazzling. I say "almost" because the various programs run the gamut from being of extraordinary quality to being virtual frauds.

There are excellent educational programs offered in audio today. The diversity is a veritable buffet of audio self-improvement. Those desiring to increase their knowledge of both the technology of audio and the business of studio operation are fortunate in having so much more available to them than in the past. But with all the interest in audio education have come the fringe elements who are not interested in helping people to learn and prosper, and whose primary concern, instead, is their monthly cash flow.

FINDING YOUR SIZE

Many assume that problems are found exclusively in private two-year and short-course programs. Others assume that four year studies in colleges are suspect. In fact there are no categorical rules by which to identify the unscrupulous educator. There are short-course and two-year programs that grew out of failed recording studios as an alternative to bankruptcy. There are four-year, state-run curriculums where a music department has a recording program to provide "cannon fodder" for capturing on tape the classical concerts that are the department's "bread and butter." There are four-year and two-year schools where the equipment is 10 years old and the faculty has not seen the inside of a real recording studio for at least that long — if ever. There are short courses and long courses and medium-size courses taught by individuals whose ego is greater than, or at least equal to, their knowledge of audio and recording.

PAYING WITH TIME

For the many students who have attended a course of instruction and found themselves ill-equipped for the "real world" of audio, the most common cry is "I was ripped off!" The real rip-off is not measured in money, but in time stolen from a life. Two years, four years, or even six months, cannot be replaced from our busy lives in the 1990s and students who leave a program knowing little more than when they entered frequently pay for their career mistake for years.

Programs can fail to deliver a useful educational experience if:

1. The instruction is equipment specific — i.e., training is focused on how to use a specific type or grouping of equipment without the technical principles of generic operation.

2. The instruction is focused only on recording and recording studios. There are 10,000 professional jobs in audio in the United States but only about five percent of them are in recording studios. Broadly-based programs that prepare a student for employment anywhere within the audio industry are a much better option.

3. The instruction is narrowly focused on one kind of operational skill. Even if the decision is made to specialize in studio recording, a program that teaches only classical recording or location recording or postproduction does not deliver enough depth. This is not a consideration with short courses where a specific skill is indeed being acquired or touched up.

One of the best ways to use an audio educational program is to learn how to better manage operations and technology in the personal and project studio. Many personal and project owners are doing exactly that. In the next column, we will look at specifics in actually choosing the right audio educational experience for you. EQ

IN REVIEW

Tech 21 SansAmp Rackmount

EQ LAB REPORT

MANUFACTURER: Tech 21, 1600 Broadway, New York, NY 10019. Tel: (212) 315-1116.

APPLICATION: Amp simulator/direct box for guitar, intended to give guitar amp sounds with full-response systems (mixer, PA, etc.).

SUMMARY: It's pretty basic, but the SansAmp Rackmount delivers gritty, vintage, realistic guitar amp sounds.

STRENGTHS: Rotary controls for easy real-time adjustment; sound quality; good construction; multiple outputs.

WEAKNESSES: No patch memories; funky cosmetics.

PRICE: $595

EQ FREE LIT. #: 114

EVERY NOW AND THEN, a little company with a cool idea pops up, stares down the big guys, and carves out a niche with a particular product. The original SansAmp Pedal did just that by producing an excellent direct guitar sound. If you didn't want to go through miking hassles, or enrage the neighbors by blowing your stack up to 11, the SansAmp delivered a credible guitar amp sound into a mixer or PA system.

The main limitations of the SansAmp — DIP switches to change tone, nonstandard (albeit compact) size, and interfacing to studio gear — have all been addressed in the single-rack-space SansAmp Rackmount. The soul of the original SansAmp is there, but being able to twiddle actual rotary controls instead of diddle DIP switches is a major improvement, since you can more fully customize the sound to your liking.

MOVING IN (AND OUT)

The front panel 1/4-inch phone-jack guitar input is paralleled by a rear panel 1/4-inch phone input with switchable line/guitar sensitivity. Plugging into the front disconnects the rear input. Outputs include XLR balanced (with associated XLR level trim control) and two unbalanced 1/4-inch phone jacks, one for -10 dB and one for +4 dB signals. There's a footswitch jack and bypass loop. Bypassing the SansAmp brings whatever is plugged into the loop into the circuit path. One application would be to use two SansAmps, one set for overdrive and one set for a clean sound, for the equivalent of two-channel operation.

Each of the eight controls contributes its own character to the overall sound. *Preamp*, the input level control, goes from clean to high overload. *Buzz* sets the low end overdrive, and determines the "bottom." *Punch* does the equivalent job for the midrange, and *Crunch* for the upper midrange/treble. These controls shape EQ and distortion at the same time. *Drive* sets the overall amount of overdrive effect.

There are two bands of EQ, low and high, that boost and cut ±12 dB. An Output Level control and Active/Bypass switch (with associated yellow LED) round out the front panel. If you want to use the SansAmp Rackmount with a guitar amp, a rear panel Live switch boosts the low and high end to offset frequency response losses in standard guitar amp cabinets.

CONSTRUCTION

The SansAmp Rackmount uses a "wall wart" AC transformer, but at least there's an XLR-style plug and jack instead of those little mini jobs. The internal construction gets high marks: mylar capacitors, TL072 op amps (which have the least noise of the TL0XX family of parts), film resistors, and an epoxy glass circuit board. The op amps are not socketed, which is a drag for servicing, but can be more reliable on the road since there's no chance of the IC shaking out of its socket or the contacts becoming corroded.

SOUNDING OFF

So the big question is, can you forget about your amps and mics and use a SansAmp? In many cases, the answer is yes. One of the nicest characteristics of the SansAmp is that it colors your signal without bludgeoning it — the essential character of the guitar

comes through, even with fairly extreme settings. Each control makes a significant difference in the sound, and the SansAmp passes the test of being able to produce both subtle and heavy distortion. Does it sound exactly like a tube amp? Well, let's just say that it sounds really good, and you don't have to replace the tubes when they start to go "soft."

Giving access to these controls, however, means that not all possible settings are gold record material. It takes a little tweaking to get the most out of the SansAmp, so here's a tip: keep plenty of blank patch sheets around. If you come up with some great-sounding control settings, you may not get them back unless you write down the details (I learned this the hard way).

In fact, my biggest complaint about the SansAmp is that you can't store programs in memory. Considering that this unit has an obvious place in the studio, you'd think engineers would want to be able to store their 20 or so "killer" sounds for fast recall and not have to worry about setting knobs. Well, there's always "hard copy memory." I wouldn't be too surprised if someone came up with a retrofit to add presets to the SansAmp.

OPINIONS

The SansAmp Rackmount is sort of a throwback in this era of microprocessor-controlled gadgets, but you can't argue with an honest sound — and the SansAmp delivers it. The closest point of reference is the ART Power Plant, but the units have significant differences. The Power Plant is a little over half the price, offers two-channel operation, and includes an effects loop. However, the Power Plant mostly gets variations on one particular (albeit excellent) timbre; the SansAmp is more sonically versatile, and seems to retain more of a guitar's particular personality.

Overall, the SansAmp will surely be a hit with studios who don't want to go through the hassle of miking amps and with studio musicians who want to be able to just plug in and play without carrying around a ton of gear. SansAmp is a cool box that does what it promises to do.

— Craig Anderton

BRYSTON

Introducing Bryston's 4B NPB Professional Stereo Amplifier

20 Year Warranty – A Generation of Music

Bryston's Model 4B, at 250 watts per channel has long been considered something of a legend in professional applications such as studio monitoring, theater and soundstage amplification, even arena and nightclub installations. Although there had been many improvements in the design over time, the basis of the 4B had remained relatively stable. Recently, however, our research showed we could make a substantial step forward in amplifier design. The result is the 4B NPB.

The power supplies employ multiple smaller filter capacitors per channel, rather than the single pairs of large filter cans as has been traditional practice. This allows better high frequency response and reduces overall losses in the supply. The wiring harness has been eliminated, and the channels plug directly onto the power supply PCB, further reducing resistance for improved current flow and better filtering.

The inputs feature a special, proprietary buffer circuit which establishes the inverting input for bridging the amplifier, and also is used for the balanced inputs, which are standard. This buffer can actually help the performance of source preamps because its input impedance is perfectly linear at high frequencies, a situation which is not always the case with power amplifiers.

Even such simple areas as the clipping indicators have been improved for more meaningful operation. The LEDs now indicate whenever there is any deviation from linearity, including shorted outputs, or strong out-of-band information, like RF or DC. They turn red during the actual presence of any distorted or inappropriate signal, however brief in duration, down to the millisecond level.

As may be seen, these differences amount to virtually a total re-design, with the purpose of further strengthening the traditional Bryston values of reliability and sonic accuracy. The performance improvement is what is really meaningful to the user, of course, and here we feel the results are most obvious, and most worthwhile. Bryston has always felt that intermodulation distortion is the main source of grain and pain in reproduced music, so we are pleased that this numerical specification has improved by about a factor of four in our new designs. The fact that this is accomplished without large amounts of feedback is testimony to the new circuitry's remarkable linearity.

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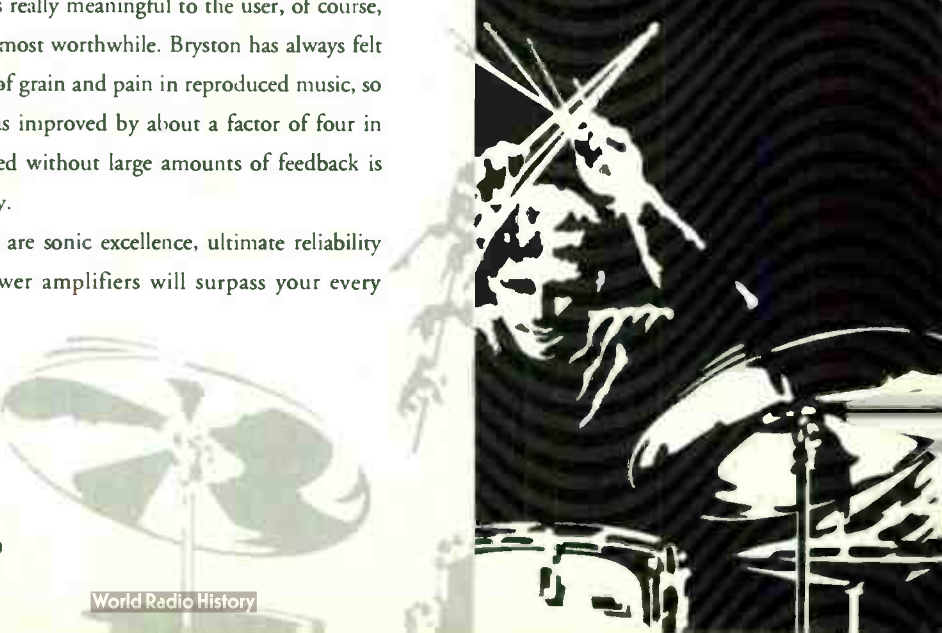

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CIRCLE 15 ON FREE INFO CARD

IN REVIEW

Kurzweil K2000

EQ LAB REPORT

MANUFACTURER: Kurzweil/Young Chang 13336 Alondra Blvd., Cerritos, CA. Tel: (800) 421-0748.

APPLICATION: Keyboard synthesizer with expandable memory and sampling capabilities.

SUMMARY: The K2000 is an excellent synthesizer, capable of many feats that you don't normally find synthesizers able to perform.

STRENGTHS: Powerful memory, some great supplied sounds, strong signal processing capabilities.

WEAKNESSES: Some supplied sounds don't quite make it, unusual keyboard feel.

PRICE: $2995 **EQ FREE LIT. #:** 115

IT SEEMS AS THOUGH there's a new keyboard synth released every month, and it's just too easy to become jaded and think of each new one as simply being a variation on an old theme. Then again, every once in a while, a manufacturer tosses out a curve ball which breaks so sharply that you have to sit up and take notice.

One synth that's definitely in that league is the Kurzweil K2000 (list price: $ 2995). It provides a wealth of innovative features and most definitely has a character all its own. An extraordinarily flexible synthesizer that works primarily with sampled sounds (like so many contemporary instruments), the K2000 can perform an unusually large number of digital signal processes to those waveforms. These processes include many different filtering and equalization operations, as well as a number of "nonlinear" functions such as shaping, warping, and distortion, all of which alter the sound in various interesting ways.

Out of the box, the K2000 comes with 200 ROM programs, along with 100 setups (performance collections of up to three programs in a layered, split, or zoned configuration) and a whopping 8 megabytes of ROM waveforms (as an option, you can also add another 8 or 16 MB of ROM samples). The sound quality of the factory programs varies greatly, but it is clear that the K2000 excels in reproducing all the great analog sounds from the era of the Minimoog, Arp Quadra, and Prophet. This is hardly surprising, considering its powerful filters and the myriad of programming options. There are a number of other impressive factory sounds, including a gorgeous 12-string acoustic guitar, very realistic trumpets, and quite a few excellent drum samples (most of which, by the way, can be improved still further with some judicious equalization).

On the other hand, the acoustic pianos, as well as some of the organ, bass, and sax sounds, were quite disappointing. I also found the onboard effects a bit grainy, often having the unwanted effect of slightly muddying the sound. Reading about sounds, however, can be as pointless as, well, listening to a rainbow, so I strongly recommend that you check out the instrument and decide for yourself!

K2000 sounds aren't limited to using just the factory ROM samples, since the instrument's onboard disk drive also accepts Akai S1000 disks. In addition, the K2000 gives you the option of adding sample RAM memory (in the form of standard SIMMs chips), so you can import up to 64 megabytes of your own custom samples via MIDI Sample Dump Standard. Bear in mind, however, that the stock K2000 has no sample RAM; this is optional. While the K2000 is certainly functional on its own, it seems to have been designed as the central component in an expandable workstation. For example, its built-in SCSI port makes it easy (and tempting) to add hard drives for saving and loading K2000 data "objects," including samples stored in K2000 format. Other options to be made available soon will include a sample input module (projected list price: $795) providing AES/EBU digital — and optical — inputs and outputs for direct importation of generic sample data.

In terms of its standard hardware, the instrument has a 61-note nonweighted keyboard with 24-note polyphony, an onboard signal processor that contains several multieffect programs but has only a single input, and six hardware outputs (four "dry" individual outputs plus a stereo pair that can deliver dry or processed signal). There are also input jacks for one continuous and two switched foot controllers. Personally, I found the keyboard to have a rather strange feel; there's a lot more "give" in the keys, in terms of both downward pressure and horizontal movement, than I've encountered in other synths, but this is very much a matter of individual taste. Custom velocity and channel pressure curves can be designed, or you can choose from among a series of preset response curves.

There's also a rudimentary onboard sequencer, although this has been designed primarily for sequence playback, as opposed to serious compositional use. Happily, the K2000

disk drive, which accepts both double-density and high-density disks, uses standard MS-DOS formatting and plays back Type 0 MIDI files, so if you're using an IBM or Atari sequencer, you can pop your data disk straight in. (If you're using a Mac sequencer, you'll need to first copy and translate the file to an MS-DOS formatted disk.) Each individual sequence file, however, is limited to a maximum of 64k, or about 15,000 notes. The sequencer can also record sysex data (up to 64k of it per file), so it can also be used as a MIDI librarian. All sequence files are held in battery-backed RAM so that they don't need to be reloaded before every gig.

In fact, except for RAM samples, all K2000 information is stored in battery-backed memory. Out of the box, there's about 116k of this memory (optionally expandable to 760k), which is flexibly allocated. In other words, if you store lots of sequences, you won't have much room for additional programs, and vice versa. As a guideline, I was able to squeeze about 200 programs, 15 custom effects, a dozen custom keymaps, two short sequences, and one master table into a stock K2000, with just a little bit of room to spare — all this, of course, in addition to the ROM memory. An unusual feature is that the battery-backed memory is protected by three standard AA batteries that you can replace yourself.

For live performance, the K2000 has a number of thoughtful features. For example, there's a MIDI panic button always handy, and it takes only one button push to change the transmit channel and/or perform octave transpositions. There's also a good "quick access" system for organizing programs and setups into banks of 10, making it a snap to instantly call up all of the sounds you need in each song.

The K2000 will also function admirably as a multitimbral instrument in a studio situation. Up to 16 programs (not setups) can play back

IN REVIEW UPDATE

At presstime, Kurzweil announced a number of important K2000 upgrades and options. These include the imminent release of Version 1.1 firmware, which comes with a new factory ROM program set (the original factory programs will still be available on disk). We previewed some of the new factory sounds at the recent NAMM show, and came away very impressed — many of them are much brighter and punchier than the original ROM sounds. Version 1.1 also includes the implementation of the SMDI (SCSI Musical Data Interchange) format. SMDI allows generic sample dump between the K2000 and other samplers via the onboard SCSI port (it's somewhat similar in concept to MIDI Sample Dump Standard, but much faster).

Another important announcement is the release of the K2000R rackmount unit (list price: $2895). This is essentially identical to the K2000 keyboard, but with an additional four audio outputs (making for a total of 10), and two SCSI ports, making it possible to integrate multiple K2000Rs with a single hard drive.

Purchasers of the soon-to-be-available Sampling Option for either the K2000 or K2000R will receive Version 2.0 firmware, which not only adds a graphic sample editing environment but also allows the instrument to directly read samples from Ensoniq EPS disks (as well as Akai S1000 disks).

IN REVIEW

simultaneously, each listening on its own MIDI channel. There is, however, no memory area where these multitimbral setups are stored — to save a particular setup, you need to do a sysex dump or save all data to disk. Program change, volume control (cc #7), and pan (cc #10) reception can be switched on or off for each channel, and you can even direct output assignments (among the hardware outputs) by MIDI channel, enabling you to audition different programs without altering these routings. Another nice touch is that by simply plugging stereo insert cables into any of the K2000's four individual outputs, outboard signal processors can be used in a loop configuration. As an extension of this concept, you can route any external signal into the K2000 and apply its onboard effects to that signal — similar to the analog input provided by the Wavestation A/D.

Space doesn't permit us to get into the enormous depth of the K2000's programming capabilities — suffice it to say that you'd have to work with a huge modular system to find the equivalent in power and flexibility. This doesn't mean that programming the instrument is particularly difficult, either — thanks to an excellent owner's manual, video instruction tape, and user interface, the process is fairly straightforward. For example, many edit displays are graphic, and they all show actual units of measurement (i.e., seconds for envelope rates, dB for gain and attenuation, and Hz or cents for pitch parameters). A number of rarely found but important functions are supported, such as the ability to alter LFO rates in real time. You can even use the MIDI clock signal as a control source for various programming functions, allowing a sound to change in real time in sync with a sequencer or drum machine.

The K2000 very much stands out in the crowd. Whether you're a novice or a pro, working live or in a studio, this instrument has much to offer — and it probably can do things your other instruments can't do.

—Howard Massey

Howard Massey heads up On The Right Wavelength, a MIDI consulting company, and is the author of several books on music technology, including NAMM's recently published "Taking The Mystery Out Of MIDI."

Ensoniq DP/4

MANUFACTURER: Ensoniq Corp. 155 Great Valley Parkway, Malvern, PA 19355. Tel: (215) 647-3930

APPLICATIONS: A comprehensive four channel digital signal processor for use in both recording and live sound work.

SUMMARY: This very versatile DSP can perform complex signal modifications but can also easily create straightforward effects.

STRENGTHS: The powerful range of effects produce excellent results with a well thought-out choice of adjustable parameters.

WEAKNESSES: Unbalanced audio inputs and outputs that could accept higher signal levels.

PRICE: $1495 **EQ FREE LIT #:** 116

THE DIGITAL SIGNAL PROCESSOR (DSP) has become a fixture in the effects racks of recording studios and sound reinforcement systems alike. On the plus side, it replaces the functions of a dozen dedicated devices. On the negative side, it works fine when laying one track at a time to tape, but when mixing, usually requires you to process a dozen different sounds one at a time. This latter fact has kept the modern effects rack from diminishing down to just two or three DSP boxes; that is, until the recent introduction of the Ensoniq DP/4.

The DP/4 allows the user to configure its four DSP channels in almost every conceivable combination. The four processors are referred to as "Unit A" through to "Unit D," and can be used as either four completely separate signal processors or patched together. Patching is called a Configuration and is operated as though it were the fifth unit. The 32 character LCD display uses abbreviations to show the current program and patching of all four units. When one unit is selected then the display becomes dedicated to that unit. All of the front panel controls, except the input and output level controls, perform multiple functions. This design saves time by speeding up menu selections, comparing edited presets, etc., but can also make operation a little confusing at first.

The DP/4, while only two rack units high, is nearly 16 inches deep. A rack 20 inches deep will fit the DP/4 and its cables nicely. The DP/4's rack mounting holes are a bit too close to its front panel handles, making it difficult to bolt the unit into slightly narrow racks.

The DP/4 offers the user a large number of default settings for control and system configurations. These have been well thought out to provide the ability to suit a number of situations without having to set the mode each time. Everything from the function of external footswitches to sending MIDI control data can be set to suit the user.

The rear panel includes four 1/4-inch (tip and sleeve) input and four output phone jacks, which provide very convenient signal switching depending on which inputs and outputs have connectors plugged in. There are also two 1/4-inch phone jacks for foot controls, a dual foot switch and a CV (control voltage) pedal. These can be mapped to control a variety of parameters within the processor. Also, there are the standard MIDI jacks, In, Out and Thru and a detachable power cord with the standard IEC type connector.

The electrical specifications are more typical of musical instruments than audio equipment. The input impedance is one mega-ohm, one hundred times higher than typical professional audio products. This will allow anything from a guitar to a mixing console to plug directly in, but does not yield the optimum headroom that conventional inputs of 10 kilo-ohm would provide. Input clipping occurs at +18 dBv — 3 to 6 dB less headroom than most professional equipment. This lack of headroom does limit applications such as hard limiting and even heavy compression, which require a lot of headroom to operate effectively. The output clips 3 dB lower (+15 dBv), but this is less of a problem than input headroom in most types of processing except for applications such as dynamic expansion and equalization where additional level is created by the effect.

The DP/4 can provide completely independent signal processing of up to four inputs in just about any combination. It is also able to combine the

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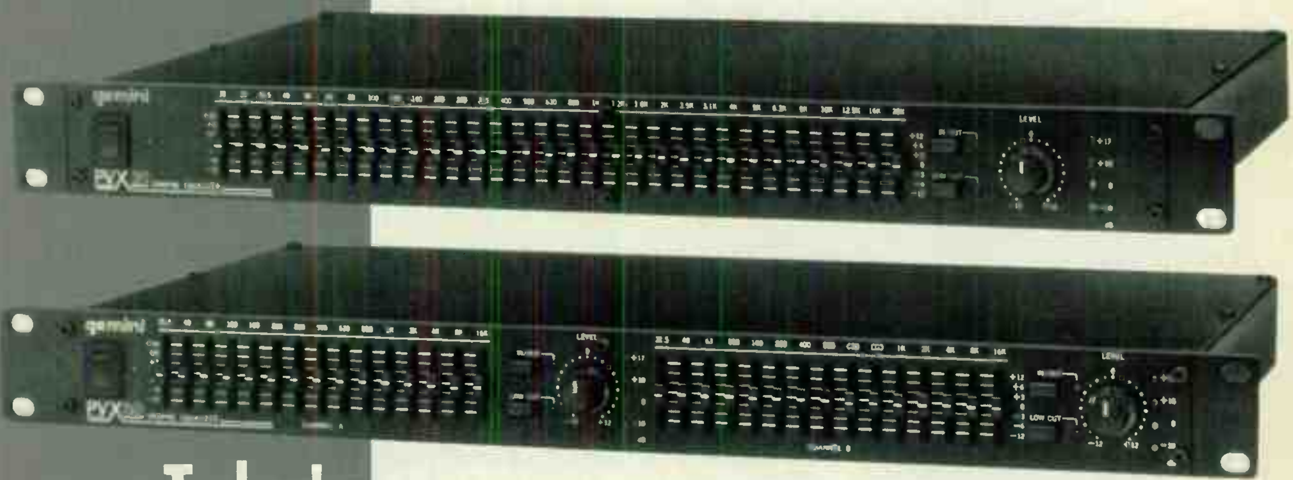

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results in ways that would use up as many as eight FX returns on the mixer if you were to use separate processors.

The DP/4 has been designed with the parallel aspects of the processor usually being to your advantage, instead of simply more confusing. I have used effects processors that could independently process separate inputs but in each case it was difficult to adjust and monitor each input. Ensoniq, however, has drawn on their considerable experience in designing musical instruments to create a user interface that, although small, is relatively quick and helpful when setting the large number of parameters that each effect offers.

The display of gain change when editing a dynamics algorithm is both useful and familiar. The unit can be left in this mode when exiting to edit the algorithm of another unit and, when you return, it will be in the same display mode. The display doesn't show the current range being assigned to an external MIDI or foot controller. The range of 0 percent to 100 percent is shown instead of the actual parameter value, such as frequency of the parametric equalizer band. Hmm? You must assign this to the controller and discover what values the range of frequencies you want to control will fall into. It can be a little tedious to go back and forth between pages of the edit display. Perhaps the fact that any editable parameter, except the algorithm itself, can be assigned to one of two external controllers makes up for this flaw.

Ensoniq has tried to include nearly every digital effect algorithm possible into the DP/4, from reverb and delay, dynamics and EQ, to guitar amp and speaker distortion emulation. The DP/4 can also generate sinewave tones and broadband noise for inclusion in other effects or as a test source for your audio equipment. An algorithm called "No Effect" can be used as a bypass setting or as a remote-controlled audio volume control with preset minimum and maximum level settings. Many of the algorithms include multiple effects such as EQ, compression and noise gating or EQ, chorusing and delay, before we even start to stack the four effects processors!

When chaining effects together care must be taken with the processor's internal signal levels. It is very easy to add level, with EQ for instance, and overdrive the next stage of the processor. The DP/4 is not very forgiving when you have run out of internal headroom and it can be a merry chase to find which parameter is the culprit after four units have been chained together.

There are 50 memory locations for storing your own presets in each of the four banks of patches — 1 unit, 2 unit, 4 unit and Configuration. There are also 50 factory presets permanently stored for each of those four. The multiple unit presets also store the signal routing information. The configuration preset stores the overall routing and system configurations. This allows the DP/4 to become a mix processor one moment and a programmable guitar stack the next with a simple MIDI program change command! The manual includes nearly 100 pages just to describe the algorithms and their adjustable parameters.

The DP/4 is one of the best DSP based processors I have used in live sound reinforcement because it's more flexible than those with dedicated controls and generally more helpful than many of the small display processors. I used the DP/4 with a wide range of performers in different venues and found an extremely useful algorithm each time I had a problem or needed an effect.

I found the compression algorithm to be as good as any of its analog counterparts, most of which lack the DP/4's additional controls. In one instance, I had a bass player that went from a silky fretless style to a funky slap style; inserting the DP/4 kept it all in line without pounding the life out of it. The noise gate was a welcome addition to the compressor algorithm, especially on another channel where a video clip had to be played back at high volume. Compression kept it from getting too loud or being lost in the mix, while the noise gate could be set to gently turn down the noise between sentences of dialogue. Before the sound check I assigned the Compression and Gating Thresholds to MIDI Controllers and, using a Lexicon MRC, set them by ear on the fly. Once I found a setting that sounded good, I stored it in memory for the run of the show.

While using Units A and B as compressor/noise gates on the bass and videotape inputs, I used the last two channels of the DP/4 to provide reverb for the backing tracks, which were recorded dry. It is always difficult to find a reverb that can sweeten a fully-mixed track without muddying the vocals or washing out the rhythm section. With the DP/4 I called up a concert hall and a plate reverb, combining their stereo outputs. Each reverb parameter could be set independently, so a clear, lush reverb was easily achieved. I then assigned both the reverb output level and reverb time to a CV pedal for the run of the show. This allowed me to produce long reverb decays where required and to fade the reverb out for spoken parts. I could change the reverb added to the track to suit the different feel of the band each night and still return to the rehearsed settings programmed into the pedal range of the algorithm.

The use of one box to take on the work of many is always a compromise. Many of the effect units that claim to provide multiple effects are limited by their lack of multiple inputs and outputs or by an awkward system for controlling the interaction of the processors. Ensoniq has provided a solution to these problems as well as excellent effects.

The DP/4 certainly gives you your money's worth and saves space. Ensoniq's understanding of controlling parameters with MIDI or foot pedals, combined with a comprehensive range of parameters, make the DP/4 a powerful tool in anybody's effects rack. Multi-effect processors may put all of your eggs in one basket, but at least this basket has four handles!

—Wade McGregor

LETTERS

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HUM-ZINGER

I am shocked (pun intended) to read Eddie Ciletti's statement ("Buzz Words," April *EQ*) that proper grounding "...requires that all three-pin power cables be lifted." Not only is lifting of POWER grounding connections risky, it is also unsafe, ill-advised, and illegal in any state that enforces the National Electric Code.

Let's examine the grounding issue from my perspective — that of an electrical power design engineer and electrical inspector who also happens to be a recording studio owner. In my studio, and especially on location, the integrity of the electrical power grounding conductor is paramount! This is because its only purpose in life is to ensure the safety of me, the public, the performers and the equipment. When I supervise the electrical wiring at a venue, I know that the power conductors are sized and routed correctly and have correct overcurrent protection, the connectors are rated for the voltage and current, color codes are correct, connections are secure, and above all, the equipment grounding conductor — the green wire — is connected to each item of utilization equipment by a conductor run alongside the supply conductors and solidly connected to the power panel ground bus which is, in turn, solidly connected to an effective grounding electrode. In one long sentence, you have the essence of the grounding requirements of the National Electrical Code. Simply put, the safety of people and equipment ranks WAY above hum and buzz problems!

I absolutely agree with Mr. Ciletti's premise that inadequate and/or incorrect wiring at venues or in studios may create or exacerbate hum and noise problems. However, I insist that the power wiring must be brought into Code compliance first; then the use of balanced circuits, lifted shields, and/or audio isolation transformers will achieve a hum-free audio environment. Those who advocate tinkering with NEC-mandated safety grounds to fix hum problems are woefully misguided, and the practitioners will die young!

Eric G. Lemmon
Owner, chief engineer
Videotel Sound
Lompoc, CA

Eddie Ciletti Responds: I believe you may have taken my comments in the article"Buzz Words" out of context. The portion of the sequence you "lifted" (pun also intended) was preceded by:The proper way to ground your system is to use the 'Star Ground Technique.'" The optimum ground source is stated and then followed by "...individual ground wires are connected to each piece of equipment." Two paragraphs later, we seem to be in agreement as I warn that "'Floating' the ground on on all devices...greatly increases the risk of a shock hazard." My suggested use of "ground lift adapters" is as a tool to determine the source of ground related problems, not the solution.

Unfortunately, "floating" the ground (by lifting the third pin of the AC plug) is common practice. This is especially true in the "project studio" environment, where systems are put together by users rather than technicians. Maybe you and I could start a crusade to insist on a common standard amongst manufacturers. All equipment should at least have "Balanced" inputs. This would allow users unfamiliar with the National Electrical Code a safer option when trying to eliminate hums and buzzes.

DOODLINGS

continued from page 114

formed yet another converter shoot-out. The three converter packages I tried were the Apogee, the Wadia and the Drake. As the source, I used the mix I was getting ready to print. The console was an SSL and the multitrack was an Otari 32-track digital. All of the analog levels were calibrated to within an inch of their lives (I guess that means they were real close). As the "control," I patched the console output right back to the monitor section so that one of the converter choices would be a straight wire.

Let the games begin. Without listening to the output of the console (just the converter outputs) my initial response was that the Wadia sounded the best. Upon comparing the output of the Wadia chain to the straight wire, the Wadia sounded "better" than the straight wire. A little extra frilliness was being added to the top end that made the stereo image a little wider than the mix I was sending into the Wadia. The strange thing was that the wider stereo field was content dependent. Sometimes it was there and sometimes it wasn't. Overall, there was a difference. The Drake to me sounded smoother than the straight wire. It seemed to sound as though the mix was on a piece of analog tape for just a second. I can see why people would like both of these systems.

Now lets talk about the Apogee. The Apogee A/D convertor used to have a problem with its low-end response. Bigger capacitors in the power supply have solved the problem. No one who listened to this test could tell the difference between the Apogee converters and the straight wire. That is all I ask of a converter: to do absolutely nothing. The perfect converter should not have any coloration of its own. It should sound exactly like a straight wire. The Apogee filled the bill. The Apogee has a few additional features that I like. First, you can have more than one digital input to the D/A converter, which is nice if you are printing to multiple formats and want them all to sound the same when you play them back. Second, the whole package, A/D, D/A and power supply fit into one rack space. Third, there is a soft limiting feature that will limit the analog signal before it reaches clipping level at the converter — a definite plus for field recording. Fourth, the Apogee will run on 12 volts DC. As a result, a car battery, an Apogee and a portable DAT machine will turn you into a sampling fool (on wheels). I have also heard that Apogee has sold quite a few D/A units to people with CD players in their Porsches. (How disgusting that I don't have a Porsche to put one in.) Fifth and finally, the Apogee is about half the price of the other converter packages.

Let me say, though, that If someone held a gun to my head and forced me to mix through the Wadia or Drake convertors, I wouldn't be ashamed of the results. They are fine pieces of hardware. I just decided to buy the Apogee. (And I'd rather put mine in a Lamborghini.)

20 BIT STUFF

One more thing before I go. I just heard that Sony has showed their new 20 bit "Super CD" to a few key people (well, except for me). It supposedly plays back as a normal 16 bit CD on 16 bit CD players, but produces the entire 20 bits on super players. In addition, it provides a higher-resolution archival storage medium for record company masters.

Twenty bits seems to be the big buzz word of late. The trouble is that there is no such thing as truly accurate 20 bit conversion in the real world. With 20 bit converters, you can produce a pretty accurate 16 bit image of the analog input, but as the resolution gets smaller, the linearity gets less and less accurate. When 16 bit converters first came out, nobody was going to advertise their new CD player as having "true 12 bit linearity" while the one next to it was claiming 16 bit conversion. The consumers would all line up to buy the 16 bit model.

Don't get me wrong here. A converter that claims 20 bits but produces good 16 bit results is better than a 16 bit converter that only produces good 14 bit results. On the other hand, a perfect 16 bit converter can give better results than a not-so-good 20 bit converter. The Wadia and Drake converters are 20 bit packages. The Wadia has a mode for optimizing dither for 16 bit operation. The Apogee uses 18 bit A/D convertors tweaked-for-optimum 16 bit performance. So don't rely completely on the numbers. Listen and choose the one that is right for your application.

Finally on 20 bit, I think that 20 bit data storage is going to be necessary for the final product, not because of more accurate A/D converters, but because of mastering and mixing through totally digital consoles. If you have an accurate 16 bit digital source and then perform digital domain EQ or reverb or limiting, the mathematical results will require 20 bit, 24 bit, or even greater data paths to maintain the resolution of the original signal. It then makes sense to provide 20 bit or better storage for the results.

WINDING DOWN

Well, that about over-does it (digitally) for this issue. I just went through my CD collection and came up with 74 minutes of music comprised of my favorite cuts from various artists. I transferred the tunes over to my DD-1000, sequenced them in the order I wanted and then cut a recordable CD. I'm going to listen to it in my car on the way down to the newsstand to pick up the latest issue of *Teenage Mutant Bimbo Nurses.* Maybe there is something of worldly importance in this month's issue? I notice that there hasn't been anything in the last few issues, but everybody is entitled to the benefit of doubt, right? EQ

TEDDY RILEY

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neer, Jean Marie Horvat, whom I like to call "Little Swedien." Currently, I'm finishing up the funk flavor for Bobby Brown's new album. I think that as Bobby creeps up to the level Michael Jackson is at, Michael will move up to the Sammy Davis Jr. level of stardom, where legends are made.

The only tips I can give to young producers who may be trying to capture the New Jack sound is to keep doing what they're doing and we can make this music continue to grow. It's not unfeasible to think that we can make New Jack Swing a big category in music. They have R&B charts, dance charts, rap, pop — everything except New Jack Swing, which is a mixture of all types of music. The latest boom in the building of rap and R&B studios is sure to take this new music in a new direction and I'm interested to see how it will evolve. It's funny, because a lot of the guys I used to go to for help are now coming to me for advice on building their own studios. I guess, when I really think about it, I've learned a lot over the last few years. EQ

DRUM MACHINES

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milliseconds will slip right by. I have used the Alesis D4 (with it's 500 built-in drum sounds) in real time to add fatness to a snare that was already on tape and the six millisecond delay didn't show at all. The resultant snare composite sounded great.

The Akai ME35T has no sounds of its own, so it must trigger a sound module via MIDI. The Simmons MTM must either send out MIDI to an external sound module or send trigger pulses designed for Simmons Drums. With these devices you have to add the MIDI delay of the sound module to the trigger-to-MIDI delays. In the case of an Alesis D4 triggering a sound in an Akai S1000, this could add 10 to 14 milliseconds. Now anybody can hear that, even with one ear tied behind his (or her) back.

If you are recording on analog tape, you can compensate for the delay by turning the tape over and delaying the trigger source over to another track. When the tape is turned back over to travel in the right direction, the trigger source is early. You can then run the trigger source through a delay on it's way to the drum trigger. Now you can slip that new snare right to where it is supposed to be.

If you are recording digitally, you can't turn the tape over 'cause it won't play. You can only lock up a second machine with a copy of the trigger and offset it earlier. I have used the Fostex D-20 SMPTE DAT machine for this purpose, but you could use a two-track analog machine or just about anything that allows SMPTE synchronization. The other choice would be to use a device that triggers with no delay.

The Wendeljr drum replacement module has the fastest real-time trigger, measured at three microseconds. Yup, that's 3/1000 of a millisecond. The Wendeljr, however, does not follow dynamics, so you have to either automate the dynamics during mixing or use multiple Wendeljrs when replacing. There was a prototype Wendel dynamics unit that would trigger in 60 microseconds, but it was never produced. Wendeljr could fake dynamics by switching between two different samples during fills, eliminating the machine-gun effect common to drum machines. Wendeljr is not made any longer. Fostex was distributing the remains, but I think they are all gone. Some rental places have them, but they always seem to be rented out.

Bob Clearmountain sells a little box that alternates between samples during fills by sending out separate MIDI events. I haven't used one yet, so I don't know the delay situation. It contains no sounds of its own, but the idea is to trigger two separate sounds in a MIDI sound module.

The Forat F16 (drum sampler) is pretty fast when dynamics are turned off and can be used in real time. It slows down considerably when following dynamics. I have seen a few drummers with them in their racks.

I know that there are other samplers that trigger from audio, other tempo mapping devices and other audio-to-MIDI converters, but the principles remain the same. Just substitute the machines that you want to use and go to work.

Well there you have it, everything that you ever wanted to know about replacing drums, and some things that you probably didn't want to know. This should give you enough background to tackle just about any problem you come up against. Good luck! EQ

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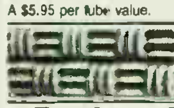

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HARPS' ACCORD

continued from page 31

WHEN IT RAINS, IT POURS

The ironic part is that Sanford had already given up the fight when the April 3 raid occurred. He'd decided to sell up and move to Hawaii. He was there house-hunting when the inspectors moved in on Secret Sound. But now for the real kicker: The heavy rainfall and flooding that hit L.A. earlier in the year caused a landslide in Sanford's backyard. Part of the patio collapsed right near the swimming pool. Now the Department of Building and Safety says the entire house is a hazard and Sanford can't move until he completes all the repairs required by the department. Sanford says he's the victim of deliberate harassment.

Maybe it's a climatic thing. In warm climates, people tend not to trouble themselves unduly about laws. They're just words in dusty books somewhere. Why bother to update them? In some Mediterranean countries they still have medieval statutes on the books banning things like extramarital sex or drinking on Sunday. Almost everyone runs around in flagrant violation, of course. But nobody takes much notice — until there's a vendetta....

One thing is certain. Whenever technology changes, someone gets hurt. The advent of the talkies imperiled lots of big stars and major film studios. Gutenberg's printing press put a serious dent in the Church's worldly power. That's the funny thing about these sweeping technological innovations: when they first happen, they threaten the established order, the big guys. The big guys generally survive, although they usually lose their monopolies. Maybe it's only natural that they try to block change by dredging up those dusty old laws.

As I leave Chas Sanford's place, I suddenly realize what's so weird about his neighborhood. The trees. "Oh yeah..." he grins and explains how all the land around the area used to belong to an old movie actor who imported and planted tons of trees from Europe and the East Coast. Driving away from Secret Sound, it's fun feeling as though I'm back out in the Hamptons, or some other place where guys still wear neckties at dinner. But the thought of all those cops swarming on Sanford's front lawn makes it impossible to forget that I'm in dear old Los Angeles. **EQ**

Digital Doodlings

Photo by Ed Colver

DCC, MD, Ruby Crystals and other bits from the digital domain

BY ROGER NICHOLS

EVERYONE I TALK TO who has read an ad about the new DCC machines thinks that the DCC tape will play back on their old cassette deck with near CD quality. The cross section ranges from 60 year old doctors to 12 year old school girls (my niece). When I explain how it really works, their mouths drop open with amazement. The ads say the DCC is completely compatible with standard cassette formats. I guess Philips thinks that if a bunch of people buy DCC tapes that won't play on their current decks, they will all run out and buy DCC machines instead of throwing away the DCC tape they had mistakenly purchased.

I really don't want to make waves about DCC. I view this whole digital audio tape issue as sort of like the war against sex magazines. The record companies didn't like DAT tape because they thought everyone was going to do something nasty with it, like copy CDs. Other people thought it was a useful format. DAT has some "socially redeeming value." DCC, on the other hand, has no reason to exist. Everything it tries to do can be done better with some other format. If I likened DAT to the more "socially-redeemable" aspects of *Playboy*, I would have to compare DCC to something more like *Teenage Mutant Bimbo Nurses*.

HARD DISK STORAGE

Storage capacities of computer hard disks seems to double every couple of years, while the size of the unit, its access time and price regularly drops. In 1981 a 5 1/4-inch hard disk stored 5 megabytes of data and cost about $3500. In 1992, $3500 buys you a hard disk the same physical size, but storing about 2.4 gigabytes (2,400 megabytes) of data. If you prefer a smaller size, you can get hard disks that are only .5-inches tall with platters under 2-inches in diameter, offering 120-plus megabytes that will fit inside your shirt pocket. These high capacities have been the reason for low-cost digital audio workstations and multitrack hard disk recorders. Remember, it takes 10 megabytes to store about one minute of stereo digital audio at a sample rate of 44.1 kHz, so you're going to need 600 to 1,000 megabytes of space to fit a whole CD's worth of music on your hard disk.

Just when you thought it was safe to base your whole existence on the future of conventional hard disks, along comes the erasable optical drive. Their larger capacities and "removability" were attractive features, but the slower access times made them hard to use for any more than two tracks of digital audio. The Sony drive, which is used by most of the digital audio people, stores 325 megabytes per side on a removable disk. This calculates to about 30 minutes of stereo per side at 44.1 kHz.

The first commercially-available digital audio product to incorporate erasable optical drives was the Akai DD-1000. The Akai DD-1000 is capable of four track performance because it does not have the same computer operating system overhead as PC- and Macintosh-based systems. Access times have been improving, and now erasable optical drives are available to interface to all of the two track computer-based hard disk recording systems. Sony is making further storage and access time improvements in their drives and within the next six months to a year, optical disk storage will at least double while improved access times will allow four- and eight-track digital audio workstations to employ the optical format.

This optical disk technology will have direct impact on the future of the forthcoming Sony Mini Disc (MD) recorder. Upon its release, data compression will be necessary to enable the MD to store enough playing time to be compatible with CDs. Within the next few years, the amount of storage will increase to the point where a new version of the Mini Disc recorder will be able to store as much as a CD with full 16 bit resolution.

Scientists have discovered that ruby crystal can be used as an erasable storage medium when written and read by a laser. It turns out that the ruby is very sensitive to the wavelength of the laser light. If you had a thin layer of ruby as the substrate of a CD, you could then record it exactly as you would an erasable optical disk. If you change the wavelength of the laser just slightly, the previous data becomes invisible to the laser and the process can be repeated all over again. The wavelength can be changed over 1000 times without disturbing any of the data written by any of the other wavelengths. To access the original data, just change the frequency of the laser back to its original value. This means that on a disc the size of one CD, you could store the data contained on 1000 CDs. It may be a while before these drives are available for audio storage (the U.S. Government needs them first for data storage in order to keep more information on your personal lives).

A/D AND D/A CONVERTORS

None of the A/D converters supplied with digital recording devices yet match the quality of dedicated standalone converter packages. When I mix, I use external converters whenever possible. A couple of weeks ago I per-

continued on page 106

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