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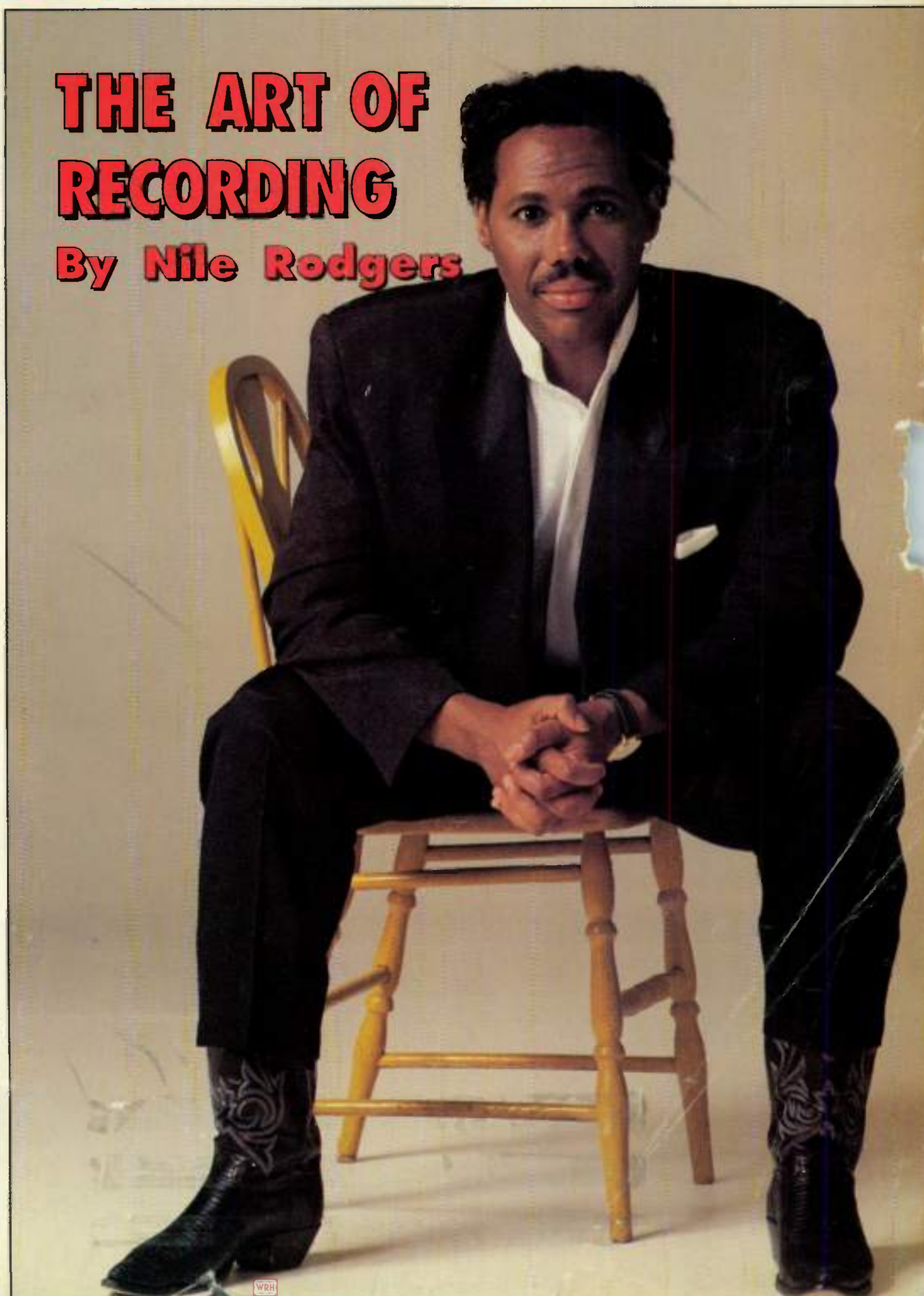

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PROJECT RECORDING
& SOUND TECHNIQUES
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NOV/DEC 1990

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Nile Rodgers by Renaldo,
Bob Ludwig by Peter Monroe.

Roland breaks th

If we were to tell you that our new S-770 is the best digital sampler in the world, you'd probably mutter something about truth in advertising and go on about your business. When, as you'll discover momentarily, it's absolutely true. And, as you'll also discover momentarily, the reason for it has less to do with any one feature in particular than it does with several features working in conjunction.

Such as the fact that the Roland S-770 is equipped with AES/EBU Digital I/O, so it's actually possible to set up a fully integrated digital production facility.

We've also equipped our S-770 with both 20 bit D/A conversion and Differential Interpolation, thereby giving it higher resolution than any other stereo sampler.

And while we're making comparisons, allow us to offer another one. With 24-voice polyphony, the Roland S-770 has more voices than any other comparably-priced sampler. So you're not only assured of getting

EQ plans to become your pipeline to the industry.

Patching Into the Reader

Things change. That's one of life's many lessons. Another, more positive lesson is that life goes on. Such is the case with EQ magazine where, with this issue, readers will notice that new ownership, staff and ideas have come to these pages. The new staff, loaded with experienced, industry publishing professionals, and bolstered by an advisory board of musicians, recordists and sound reinforcement technicians, knows that one of the things that never seems to change in this ever-changing business is the need for knowledge.

Professional players, engineers, producers — anyone involved in sound and recording — need to know about the latest equipment. Furthermore, they need to know how (and how well) that equipment operates. It's necessary for their — your — very existence. Knowing that a new limiter, MIDI merger, synchronization device, or software-based sequencer is on the market, and that it works effectively, is essential to you — whether for keeping and getting clients, or for advancing your own creativity.

Disseminating useful information is what EQ is all about. Hands-on tips and techniques designed to help you do things faster, more efficiently and with greater confidence are what you'll find in these pages. To that effect, you'll find new columns dedicated to answering your questions, be they about particular products, or about generic audio/music problems. Manufacturers and industry experts will answer your queries. We welcome your letters. EQ plans to become your pipeline to the industry.

The past five years have produced many professionals who have put together personal production facilities — places where they can work at their own pace. These individuals have, by necessity, simultaneously taken on a number of jobs — engineer, producer, arranger, player, technician. Along with those musicians putting together touring racks and soundmixers working in clubs and mid-size venues, they need a comfortable place to ask questions and learn. We plan to make EQ that place.

So, throw your hat on the console and put your feet up on that blown speaker cabinet. You're home.

Hector La Torre

Hector G. La Torre
Executive Director

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PRESENTING THE LEXICON 300

A unique digital effects processor with analog and digital inputs and outputs, time code automation, and magnificent sound.

There may be digital effects processors that rival some of the 300's features, but you'll never find one with them all. The 300 delivers precise delay and stereo pitch shifting, as well as stunning

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The 300 is a powerful tool in audio for video applications where time code synchronization is essential. And in digital video editing, the 300 ensures that scene changes are handled smoothly — in the digital domain. Because the 300 has digital inputs and outputs, it's the perfect choice for RDAT and CD mastering. And for music production there is an incomparable set of sounds, as you would expect from Lexicon.

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The 300 redefines connectivity standards for digital signal processors. Unique analog and digital circuits accept analog signals or digital signals in the consumer SPDIF or professional AES/EBU formats. Whether the source is a CD player, RDAT recorder, or open reel digital deck — the 300 identifies and locks onto the incoming format.

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With the 300's real time MIDI automation you can record parameter changes on most any sequencer. The 300 also includes Lexicon's Dynamic MIDI® allowing you to control the 300's effects parameters from any MIDI controller.

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The 300 joins the Lexicon family of digital effects processors. From the economical LXP-1 to the world renowned 480L, they all share the Lexicon Sound.

For more information about the 300 or any of Lexicon's products, call (617) 736-0300, FAX (617) 891-0340, or write Lexicon, Inc., 100 Beaver St., Waltham, MA 02154.

lexicon

The Art and Science of Sound

CIRCLE 26 ON FREE INFO CARD

EQ&A

This is where your technical questions get answered. Send your query with your name and address to: EQ Editorial Offices, 939 Port Washington Blvd., Port Washington, NY 11050

Q I run a jingle studio in Manhattan and I'm confused by all the "professional" DAT decks that are being sold. What's the basic difference between "pro" and "consumer" units? What features should I look out for?

David Mitchell, Syosset, New York

A Pro units are different from consumer DAT machines. Pro units should give you better performance and should have features you'll need to integrate them into a professional studio environment. Among the most important features to look for are balanced, +4 dB inputs/outputs, AES digital inputs/outputs and IEC958 digital inputs/outputs (SPDIF). Also keep an eye out for a jog/shuttle wheel, and sophisticated editing features for Start IDs (program numbers) and Skip IDs. The jog/shuttle wheel turns out to be extremely useful for precise positioning of these IDs. Don't forget that your pro DAT unit needs rack ears. And make sure you are getting a professional implementation of SCMS (Serial Copy Management System), the copy inhibit standard for consumer DAT machines. SCMS never affects the audio quality but a consumer DAT with consumer SCMS will inhibit you from making certain kinds of copies, which could be a problem in a studio. A pro DAT, on the other hand, should never prevent you from making a copy, and should give you a way of selecting the SCMS level on your tape at least when you're using the analog or AES ports.

Chris Foreman, Marketing Manager
Panasonic Professional Audio

Q I recently purchased a Nady 650 VHF wireless guitar system for my guitar. Whenever I turn off the guitar volume at the end of a song there is a very loud series of pops. It only occurs with one of my guitars, a Charvel "Fusion Deluxe" (passive electronics), which has one stock, single coil, ceramic-type, high-output pickup (neck) and one Seymour Duncan "J.B." high output humbucking pickup (bridge). I had the guitar completely checked out and replaced all pots and wiring, but to no avail. The sound does not occur with my other brands of guitars that have the same electronics, nor does it occur when I am plugged directly into my amplifiers. Is the problem the high output pickups, or some freak grounding problem common to Charvel? Have other guitarists using this wireless system encountered the same situation? How can I rectify it?

Rello T. Slick, Bronx, New York

A Metallic scratching noises result if bare metals are rubbed in the vicinity of a wireless transmitter or receiver. This happens when you turn a volume pot all the way off (it goes from metal on carbon to metal on metal). The phenomenon is common to all FM transmissions. Make sure all guitar volume and tone pots are clean and contacts solid. A 220 pf capacitor soldered across the hot to ground terminals of the pots will eradicate pot noises by effectively shorting any RF hot spots caused by the proximity of the transmitter. The capacitor will not affect the transmitter or a regular guitar cord.

Tono Rondone, Advertising Manager
Nady Systems

Q I am using 60-degree and 90-degree Flat-Front Bi-Radials (c) and find I can't get the response matched up (using the same drivers on each). I almost need different equalization for the two horn groups. Is there a problem here I'm not aware of? Is there a remedy? The 60-degree horns have a peak at about 7-8 kHz, and a notch just above it. The 90-degree horn doesn't show this glitch.

Andy Pensa, Eau Claire, WI

A You don't specify the exact nature and number of elements in the 2380/2385 array, so it is not possible to pinpoint the problem. Looking at the JBL 2380 and 2385 data sheets, both horns have very smooth on-axis response, and relatively less high-frequency boost is needed to achieve flat on-axis response. Identical high-frequency power-response correction equalization can be used on both horns. This can be applied via an active circuit in the electronic crossover or via matching passive JBL networks. Field modifications to implement this equalization are described in one of JBL's *Technical Notes*, Vol. 1, No. 5.

The response differences described do not appear on the measured curves in the horn's specifications, so identifying their source is difficult. If you are certain the differences are due to different horns and you feel they need to be corrected, then separate equalization will need to be applied to each horn type. This cost would probably not be worth the benefit as interaction might cause response perturbations equal or greater than the differences you are trying to correct. The differences are more likely due to mounting or local acoustic environment differences, or perhaps there is a discrepancy between the two compression drivers.

In normal array usage, a single 2385 (60-degree x 40-degree) might be used for far coverage, and a splayed pair of 2380's (90-degree x 40-degree) for front coverage. We stress the importance of physically aligning the drivers in such arrays as these in order to minimize peaks and dips in response in the area of overlap of the horns.

Ted Telesky, John Eargle
JBL Professional

(Editor's Note: JBL's Technical Notes are available from the company at no cost by writing to Ms. Lynn Cope, Literature Dept., JBL Professional, P.O. Box 2200, 8500 Balboa Blvd., Northridge, CA 91329.)

Q Could you please give me an explanation of the terms "side-chain" and "frequency dependent" as they apply to compressor/limiters? My equipment offers these possibilities, but because I'm not sure what these functions will do for me, I'm presently not using them. Specifically, how should I set up to use these functions, and what do I listen for to know the process is working?

Ed Gusser, Ellenville, New York

A Side-chain refers to the way through the signal path, which starts at the microphone or instrument, and ends at the recording heads or speakers. The side-chain is a detour from the direct signal path much like an effects loop. However, unlike the effects loop the side-chain is used for controlling the signal path. This control can be accomplished by means of equalization, compression,

continued on page 74

AND WE QUOTE:

"We also tested this monitor for each of its EQ settings. The EQUALIZED mode for the PRM 308S was very strong in the rock and dance music categories, proving the **highest scores in the gestalt, clarity, and depth areas of the dance music categories**. On top of these honors, the 308S monitors were felt to have some of the **best bass extension of any monitors tested**. In the EQUALIZED mode, they were smooth and provide a very deep stereo image that must be heard to be appreciated.

In REFERENCE mode, the PRM 308S took **first place as the best-liked speaker of any tested**. In this mode they were thought to possess the **clearest and most musical sound** for all three types of music. The main difference between the two EQ settings lies in the frequency balance characteristic, where the REFERENCE mode is much more even sounding than the EQUALIZED mode. Many of the near-field monitors tested had a light midrange feel to them, so it's nice to be able to flip a switch to get an idea of how your music sounds in an alternate environment. **The PRM 308S system is truly wonderful and must be heard to be believed!"**

Rolf Hartley • Audio Consultant/Journalist • EQ Magazine*

"I read with interest The Electric Near Field Monitor Test in your premier issue. I spend most hours of my rapidly dwindling life in front of the little beasties and don't have the time or the money to buy every speaker you reviewed. On the basis of your admittedly subjective evaluation I acquired, against my better judgement, a pair of Peavey PRM 308S's. I mean, just the logo, you know?

Boy, was I wrong. As you say, **the speakers have to be heard to be believed.** So far I've mixed two albums through them (Steve Earle and Colin James, both to be released in June) and **neither I nor the artist could be happier with the results.** Thanks for the tip and, if you're in the market for several pairs of NS-10s, please give me a call."

Joe Hardy • Ardent Recordings • Memphis, TN

Peavey Audio Media Research™ 308S ™ Phase Reference Monitors

From the 14 models of near field studio monitors tested by GPI Publications in *EQ Magazine*, the Peavey Audio Media Research™ PRM™ 308S ranked number 1 in Reference Mode*. In categories such as stereo imaging, spectral balance, transient handling, clarity, and gestalt, the 308S was picked as the best-liked of those tested. . . If that isn't enough, it also placed third in the Equalized Mode.

The PRM 308S features a unique foam blanket surrounding the high frequency and midrange components to provide more accurate "imaging" and "transparency." A highly desirable reference/equalization response mode switch is provided, simulating different listening environments . . . tracking or mixdown.

Experience what the test panelists of pro engineers and producers discovered. Call us at the factory for the name of your nearest Peavey AMR dealer to arrange a "test listen" of a pair of 308S reference monitors for yourself, and perhaps your ***frame of reference* will change, too.**

**GPI Publications, EQ Magazine MAR/APR 1990 "The Electric Near-Field Acid Test"*

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EQ PRODUCT VIEWS

CAUSE AND EFFECTS

Lexicon's recently-introduced **LXP-15** digital multi-effects system offers Dynamic MIDI effects automation and remote control. Dynamic MIDI provides control of 27 variable parameters, and all major functions can also be controlled from the front panel with a switch, an adjustment knob, and a soft-key controlled display indicating an effect's parameters and operating status. Five inputs for external foot switches are included, and can be patched to any of the effects parameters. It can be used for recording as well as sound reinforcement applications. There are 128 preset effects and there is room for 128 custom effects. Included among these are pitch shifting, stereo delays, gate, plate, and reverb. It retails for $1050. For more information contact **Lexicon**, 100 Beaver Street, Waltham, MA 02154, 617-891-6790. Circle reader response number 145.

SOUND OF MUSIC

A line of sound reinforcement loudspeakers ranging from the compact CSX25 to the full-size CSX70, the CSX Series from **Community** offers high current quick disconnects to the speaker leads, large steel input panels, and a new crossover network utilizing PowerSense proprietary circuitry. The PowerSense circuitry monitors operating power levels and provides positive indication of overload conditions. They are designed for concert sound reinforcement, nightclub, church, or public speaking applications. Prices range from $321 to $1066. For more information contact **Community Light & Sound, Inc.**, 333 East Fifth Street, Chester, PA 19013, 215-876-3400. Circle reader response number 119.

AMPLE FIRE

QSC Audio's EX 4000 dual monaural power amplifier lets you turn up the heat — with the extra protection of a built-in limiter that prevents distortion during clipping, and a thermal management system that varies fan speed with heat requirements. Open input architecture allows for second generation signal processing inputs or computer control, digital audio, or fiber optic systems inputs. The dual mono configuration allows its protection systems to be placed independently on each channel. It also offers detented gain controls with 2 dB steps, LED status arrays, and Neutrik SpeakOn and 5-way binding post output connectors. It lists for $1998. For more information contact **QSC Audio**, 1926 Placentia Avenue, Costa Mesa, CA 92627, 714-645-2540. Circle reader response number 166.

MIX MASTER

Tascam's M-3500 console series is designed for space-conscious studios who require ample facilities for 24-track and MIDI equipment. The M-3500/24 ST in-line mixing console offers 24 mono and eight stereo inputs, and 28dB headroom and -130dB (EIN) mic preamps. It features eight group busses, an in-line distributed monitor section with a dedicated fader and mute, four assignable effects returns and eight stereo channels. The board also includes four-band equalization with two mid-range frequency sweeps and a high-pass filter. Prices range from $7499 to $9499. For more information contact **TEAC**, 7733 Telegraph Road, Montebello, CA 90640, 213-726-0303. Circle reader response number 187.

ALL DAT JAZZ

Frequent fliers take note: **Bruel & Kjaer** and **Panasonic Professional Audio** have jointly introduced the first completely portable R-DAT recording/archiving system housed in a padded flightcase. The system couples two B&K Type 4006 omnidirectional microphones and Panasonic's SV-255 portable R-DAT recorder. Accessories included are a stereo mount, P-48 power supply, UA-0777 nosecones, battery charger and battery. B&K Type 4011 cardioid microphones are optional. The package, contained in a lightweight Zero/Halliburton flightcase, meets F.A.A. airline regulations for carry-on luggage. The system is available from Bruel & Kjaer and costs $6,200. For more information contact **Bruel & Kjaer**, 185 Forest Street, Marlborough, MA 01752-3093, 508-481-7000. Circle reader response number 115.

SPEAKER OF THE HOUSE

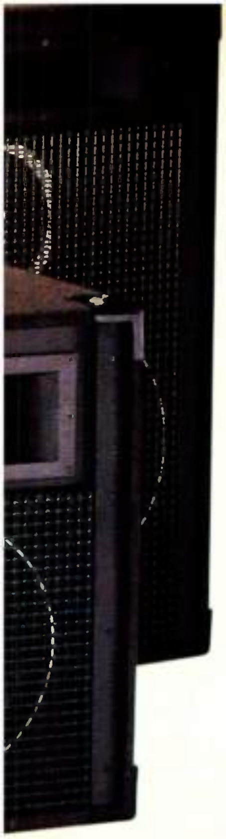
JBL's newly introduced **SR4700** Series of sound reinforcement loudspeakers consists of six models featuring a durable, lightweight cabinet design. These high-tech loudspeakers feature titanium diaphragm compression drivers with patented diamond surround, Bi-Radial horns, and new Vented Gap Cooling low frequency transducers. Prices range from $795 to $1695. For more information contact: **JBL Professional**, 8500 Balboa Boulevard, Northridge, CA 91329, 818-893-8411. Circle reader response number 135.

HEAD BAND

Incorporating a new "hinged" earphone design that lets them be angled away from the ears, the **AKG K1000** reference listening system headphones connect right to an amplifier's loudspeaker terminals rather than the headphone jack. Its cord leads are recessed into the underside of each earphone so that the headphones can be placed upright. The transducer design delivers an acoustically transparent perspective, and offers extended low frequency performance. Its suggested retail price is $895. For more information contact **AKG Acoustics**, Inc., 77 Selleck Street, Stamford, CT 06902, 203-340-2121. Circle reader response number 102.

DYNAMICS DUO

The **MDC 2001** from **A.R.T.** is a two-channel stereo device designed as a master controller for live sound reinforcement, vocal/instrument tuning and enhancement, or as a final level controller/signal enhancer for tape mastering. You won't miss a beat with more than 45 LEDs for monitoring all functions and level variations. Its compressor features independent control of all needed functions including input gain, slope, attack and release time. For external control, a switchable detector loop offers gating, keying or ducking functions. The controller comes in a standard 1U rack design, and retails for $499. For more information contact **Applied Research & Technology**, 215 Tremont Street, Rochester, NY 14608, 716-436-2720. Circle reader response number 110.



BITS & PIECES

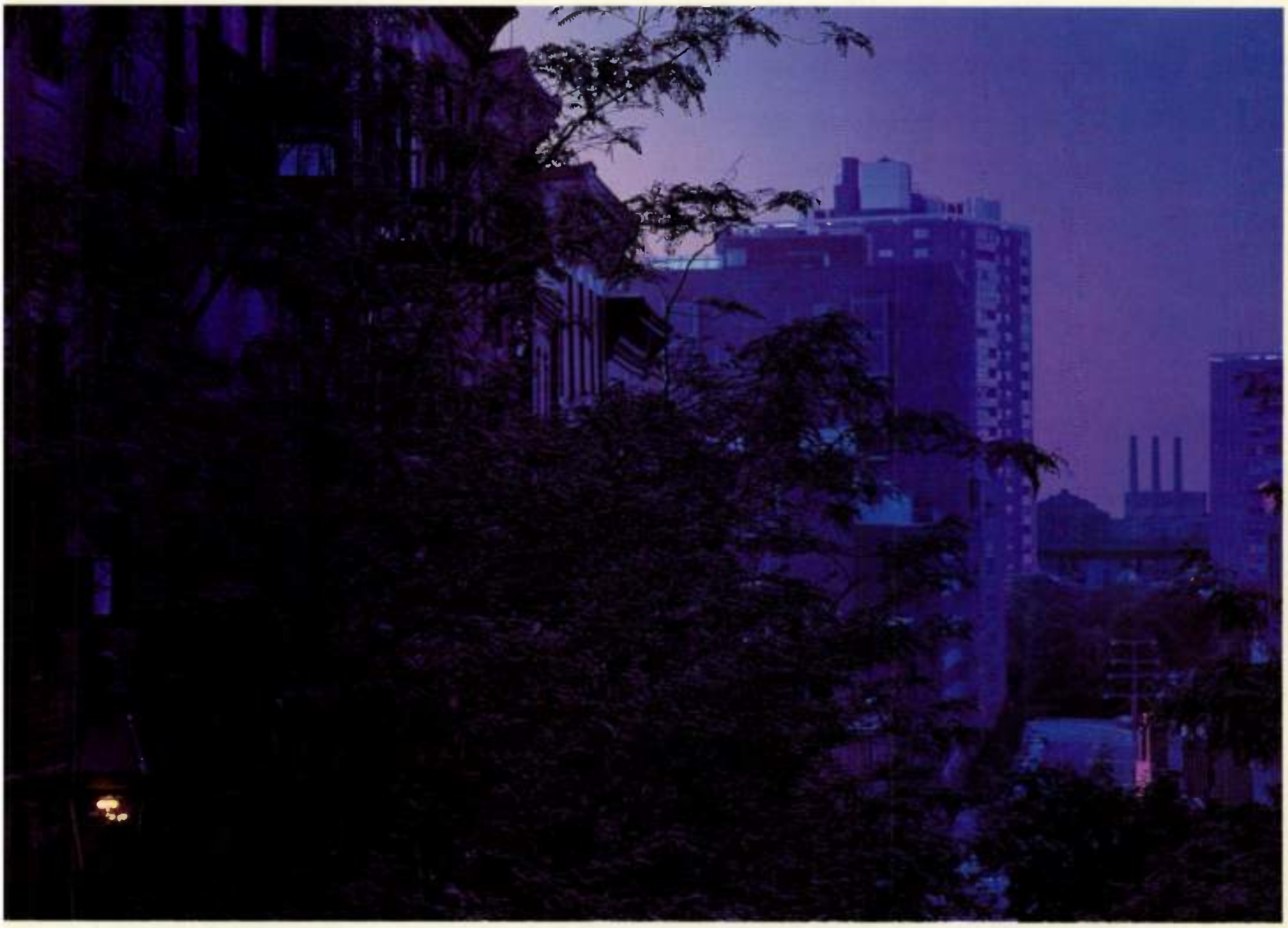
Get seamless recordings with **Panasonic's** newest professional DAT deck, the **SV-3700** — a high resolution 4-DAC system featuring high performance A-D one bit convertors. The DACs ensure replay quality by removing zero cross distortion and enhancing linearity. The recorder is equipped with AES/EBU and IEC 958 consumer format digital inputs and outputs. A comprehensive display allows for monitoring of error correction rates during playback. An infrared remote controls all front panel functions and a shuttle wheel with two variable speed ranges lets you review program material slowly or at high speeds in forward and reverse. It lists for $1599. For more information contact **Panasonic**, One Panasonic Way, Secaucus, NJ 07094, 201-348-7000. Circle reader response number 163.

COMPOSE YOURSELF

Looking for an all-in-one recording system? **Roland's MV-30 Studio-M** allows you to create music, edit internal sound, and automate or control mixes in real-time. It can work as a standalone or control a large MIDI configuration. Its built-in sequencer has an internal memory capacity of 50,000 steps and 16 tracks (eight internal, eight external). A real-time pattern trigger mode provides fast access to ten assignable patterns, which can be triggered individually or simultaneously and recorded as a sequence. For more information contact **Roland Corp.**, 7200 Dominion Circle, Los Angeles, CA 90040-3647, 213-685-5141. Circle reader response number 170.

Don't eat. Don't drink. Don't sleep. Don't stop.

Yamaha Corporation of America, Professional Audio Division, P.O. Box 6600, Buena Park, California 90622-6600.



UP DATE

News & Notes From The World of Modern Recording & Sound

Have Tape Will Travel

RANDY EZRATTY'S mobile recording studio, Effanel Music, has gained industry attention because of its automated **SSL** console and 48-track **Sony** DASH recorders. However, this is only the latest evolution of portable systems that Ezratty has designed and used since the late 1970s, when he assembled a portable system into road cases to record Mick Fleetwood in Ghana for RCA. The resulting LP, "The Visitor," was a breakthrough album in international music, and put Ezratty's style of portable recording on the world map too.

Since that time, Effanel Music has mixed and tracked everywhere from the Chilean Andes to Suzanne Vega's living room, and the portable recording concepts that Ezratty pioneered are of value to anybody eager to take their gear on the road.

"One of the secrets is to have all your cable interconnection and grounding pre-configured all the time. Once you get to your loca tion only a few cables are hanging there and the grounding is already taken care of. All the cables should always lead to the same place with its own multi-pin connector." He adds that, "the entire system should be built in a room and made quiet and reliable beforehand so you get where you're going and simply set up."

This doesn't mean buying the most expensive gear you can afford, either, Ezratty advises. His first system featured a vintage 1970s Stephens 814 tape machine ("it had no capstan, no brakes and it handled tape like a **Studer A-800** ten years later") and a three-piece **Sound Workshop** Series 30 console ("so it could be carried by one person").

Though Ezratty is particularly fond of older Stephens, Ampex MM-1200 and ATR-124 tape machines, and though he credits Daniel Lanois, whom he worked with on U2's "The Unforgettable Fire," with configuring one of the more innovative, vintage systems currently in use (featuring **API** and **Neve** equalizers), he advises that reliability is an essential part of any system configuration. "I wouldn't want to have a system that is so esoteric because it may not be as dependable. No artist trying to capture a moment wants to have a 24-year old capacitor going out on him."

As such, he is intrigued by the number of smaller remote recording assignments that are currently being recorded with the **Akai** A-DAM. He also advises that although some of the older tube mics may be preferable from a sonic standpoint, they don't necessarily stand up to the rigors of the road. "We find that for live recording, we tend to lean towards smaller capsule mics because in the real world there seems to be overabundant bass — especially in a concert situation."

Photo by Peter Monroe

Ezratty asserts that miking is crucial to the system's portability. "The better suited the mic is for a particular environment the less you have to compensate externally with extra outboard equipment," he says.

Of course, working with the right artists never hurts. Doing vocals with Suzanne Vega on her "Days of Open Hand" album in her living room afforded the artist the comforts of her home.

Bryan Ferry's "Boys and Girls" was set up and tracked in a TriBeCa loft. "The right artists don't fight the environment, and they want to record in unconventional places because they're looking for something that goes beyond acoustics," Ezratty says.

"It's basically a mindset. You have to realize that the recording doesn't have to sound like every recording studio in town. If it sounds pleasing to you then it's fine."

One MO' Time

The first recordable/erasable magneto-optical (MO) CD systems were introduced at AES, and will be made available to professional audio users within the next few months.

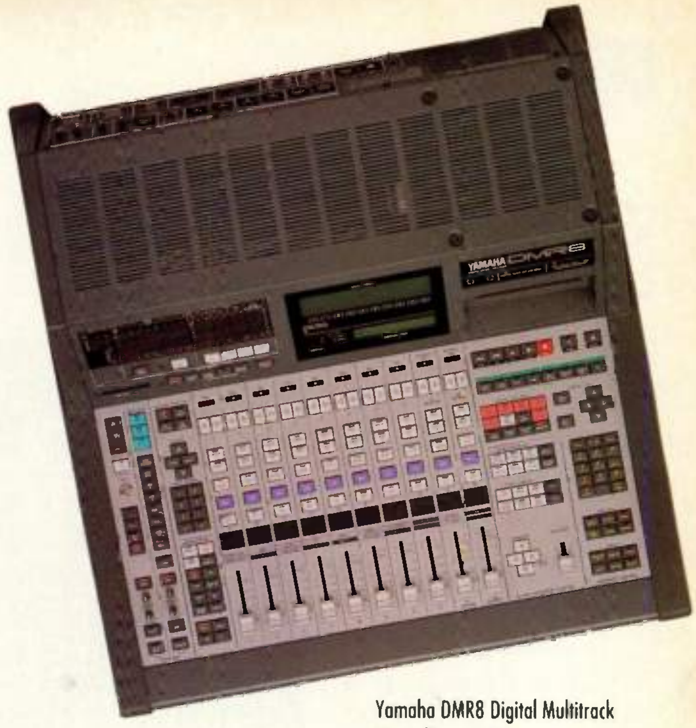
MO is the only recordable/erasable CD format that is currently being manufactured. Developed by both professional and consumer audio manufacturers, it uses 5 1/4-inch disks and can only be played on these soon to be available player/recorders.

Akai, New England Digital, and **WaveFrame** debuted various MO products for their recording systems. The Akai and WaveFrame systems are recordable and erasable, while New England Digital's unit is Write Once/Read Many (WORM). The **Solid State Logic** ScreenSound utilizes the WORM system from **TEAC** for sound archiving.

The Akai system is a single drive unit that will be incorporated into its DD1000 recorder/editor, and is designed primarily for remixing. The WaveFrame version is designed for real time two-track recording to be used with the company's AudioFrame and CyberFrame digital audio workstations. Meanwhile, the NED WORM system is primarily a backup sound storage device, which the company eventually plans to supplement with a sound library.

The three MO systems are incompatible with each other. But because of better storage and faster access capabilities, many observers feel MO systems could replace DAT. Stay tuned.

First Look: DAWs Get Personal



Yamaha DMR8 Digital Multitrack Mixer/Recorder

NOW THAT you've read all the news you can stand about the wonders of the new digital audio workstations — make room for some news worth hearing:

At this year's AES a range of new, affordable digital audio workstations, hard disk and tape-based digital storage systems with signal processing and equalization, built-in consoles and other cool complements, were introduced.

The operative word is *affordable*, a nebulous term ranging between $20,000 to over $40,000 but still a far cry from the quarter-million-dollar systems that introduced the concept. Even more significant is the entry of manufacturers ordinarily associated with MI into what had previously been exclusively a high-end pro audio domain. Korg, Roland and Yamaha have added the experience gained from producing innovative and sophisticated keyboard, sequencing and signal processing systems to mature tape and hard drive technologies to produce workstations in a price range that, while expensive from a purely MI point of view, bring powerful digital audio workstations to a new, cost-effective price-point.

But how good are they and when will they truly appear? Unfortunately, only time will tell the full story. However, **EQ** had an opportunity to preview some of these units and tweak a few knobs ourselves at AES.

Yamaha's DMR8 Digital Multitrack Mixer/Recorder is a tape-based eight-track 20-bit digital recorder with a 24-bit digital mixer, timecode reader generator, 32-point auto-locator and automated mixing system in a single integrated unit. The DMR8 uses a proprietary Yamaha tape cassette loaded with 8mm tape. In addition to the eight audio tracks, two FM auxiliary tracks and a timecode track are included, all running over a stationary head. Running time is 22 minutes per tape at 44.1 kHz; 20 minutes at 48 kHz.

It also records at 32 kHz. The real-time automation system stores all moves and snapshots (storable to card), with three-band EQ and three digital signal processors (derived from the SPX900/1000 chip) and the DMR8 uses a moving fader system. Eight faders are multi-assignable in three banks to a range of functions, starting with input level control through automated mixdown. These eight faders can handle 24 channels of digital audio in groups of eight during a mix when connected to other digital audio sources (such as a DRU8, a rack-mounted version of the DMR8 without a mixer). The Yamaha system, which originally debuted at last year's NAB, is one of the most evolved products in the current crop of DAW debutantes. The tape-based, stationary head system provides relatively fast access times and an easily removable storage medium. The proprietary tape format could be a problem in terms of supply but Yamaha product specialist Michael Nicoletti maintains that Yamaha will provide strong support. The multiple assignment of the faders to many functions keeps the system compact but makes for a cluttered front panel.

This may make the learning curve on the DMR8 steep, at least initially. The mechanical faders are the only tactile links to more conventional console technology. However, the unit has just entered production in Japan and U.S. release is expected for early 1991. The total system cost will be about $40,000.

The new **Korg** Digital Audio Production System has a similar, though considerably less cluttered desk layout. This integrated eight-track, hard disk-based system comes with about 100 minutes of audio

continued on page 22

Sampling Clearmountain

WISH YOU could have Bob Clearmountain mix your new recording?

Sorry, stand in line. But, courtesy of East West, you can use some of the 259 drum sounds he recorded direct to DAT at **Bearsville** and **A&M Studios.**

"I'm tired of having to listen to demos with crap drum sounds," Clearmountain says.

"I've had access to two of the best recording studios for recording drums on the planet, and over the years I have gained a bit of experience at recording drums," he says, "so I thought some of you sample freaks just might enjoy this."

Though some were sampled into an **Eventide** H3000SE with 44.1 kHz sample rate and 16-bit resolution, all were played live. Those recorded at Bearsville and A&M Studio A utilized Neve consoles, while samples taken at A&M Studios B and D were recorded through Solid State Logic boards.

Clearmountain's samples includes "dry" (minus effects) bass and snare drums in mono with stereo ambient versions.

For some tom tom series, the "dry" signal was recorded on the left and the ambience on the right. Another series is centered with stereo ambience. The result is an extensive menu of drum sounds.

Use them and you'll have one up on Bob himself. None of the samples have yet to be used in any of his own projects.

Now how should your lawyer handle that album credit?

If you need any further information, please feel free to contact East-West at: 8787 Shoreham Dr., Suite 807, Los Angeles, CA 90069; (213) 659-2928.

Snuff Walden has recently purchased a Sony APR-24 24-track analog recorder to be used on future TV music projects including the upcoming shows "The Outsiders" and "Working Girl."

EQ's Digital IQ

1. In DAT, what is the method that insures rotating record or playback heads remain centered along the helical track on tape?
A. FTA **B.** ATF **C.** AFT

2. Coercivity is measured in
A. Hertz **B.** Megabytes **C.** Oersteds

3. What is the residual magnetism of a magnetic medium after it has been recorded to maximum capacity?
A. Retentivity **B.** Coercivity **C.** Quantivity

4. Parity is a simple form of error detection that
A. subtracts an extra bit from each word
B. adds an extra bit to each word
C. splits extra bits

5. Dither is added to an incoming signal at the time of encoding to reduce or mask the effects of
A. distortion
B. magnetization
C. quantization

6. The process of combining two or more channels of information into one is
A. duplexing
B. multiplexing
C. oversampling

7. In a digital word, the first bit in the word sequence defining the largest increment of resolution is the
A. Least Significant Bit (LSB)
B. Most Significant Bit (MSB)
C. Significant Original Bit (SOB)

8. Irregular low level signal fluctuations are
A. artifacts
B. digitally generated
C. random noise

9. A type of digital filter that uses oversampling to enable implementation of anti-aliasing without undesirable analog side effects is known as a
A. Finite Impulse Response (FIR) Filter
B. Infinite Impulse Response (IIR) Filter
C. both A and B

10. What is caused by the creation of false signal components during the analog-to-digital conversion process?
A. Anti-Aliasing
B. Aliasing
C. Angst

EQ would like to credit DIC Digital's Pocket Guide To Digital for the questions and answers in this quiz. For a complimentary copy of this valuable reference booklet contact **DIC Digital Supply Corp.,** 222 Bridge Plaza South, Fort Lee, NJ 07024, 800-DAT-1-DIC.

Answers: 1-b, 2-c, 3-a, 4-b, 5-c, 6-b, 7-b, 8-c, 9-a, 10-b.

Random Access

Dave Edmunds recently purchased a 10-channel, 20-input **Tascam** 688 Midi studio to work on demos in his home recording studios. ☐ **Alwynn Willis Ministries** have purchased **Tascam** 112 and 122 cassette recorders/reproducers. The 4-track, 2-channel cassette machines will be used to record sound tracks and sermons. ☐ **Greg Allman** recently purchased a **Tascam** 4-track Porta II ministudio to record some ideas on the road. ☐ **Bruce Springsteen** has taken delivery of a **Sony** PCM-3348 48-track digital audio recorder for his new project studio. The **DASH** multitrack recorder is compatible with the two PCM-3324 24-track recorders he already owns, and is being used to record material at A&M Studios. ☐ **Adrian Lee** of Mike and the Mechanics has purchased a **Soundtracs** IL 3632 console with Tracmix automation to be installed in his private studio. ☐ **Herbie Hancock** has recently purchased two **Akai** 12-track A-DAM digital recorders which he expects to combine at one 24-track digital installation. Currently touring with guitarist **Pat Metheny**, bassist **Dave Holland** and drummer **Jack DeJohnette**, Hancock is using one of the machines for on-location recording. Meanwhile, the second A-DAM has already been installed in his project studio, Hancock Music, in Los Angeles. At the same time, **Stevie Wonder** has purchased his third **Akai** A-DAM recorder. ☐ Artists currently touring with **Electro-Voice** N/DYM microphones include **Aerosmith, The Jeff Healy Band, MC Hammer, Prince** and **Santana**; the mic is acclaimed for providing very high gain before feedback. Aerosmith and Prince are using Electro-Voice's DeltaMax concert systems as well. ☐ Drum roll, please. And the winners of the recent **Ampex** Golden Reel Awards are rockers **Joan Jett & the Blackhearts**, red hot rapper **Young MC**, dance music don **Stevie B**, and soul sensation **Luther Vandross**. Joan Jett's LP "Up Your Alley" and Young MC's album "Stone Cold Rhymin'" were recorded exclusively on Ampex Grand Master 456 audio mastering tape. Country and western singer **Keith Whitley** was posthumously awarded a Golden Reel Award for his album, "Don't Close Your Eyes."

Just The Snuff

A PROJECT studio is responsible for several of the most popular soundtracks you're hearing on this year's TV schedule, thanks to the work of Snuff Walden Productions, which wrote the themes to "Wonder Years" and "thirtysomething."

Company owner Snuff Walden also wrote and recorded the tunes to the Emmy award-winning drama "Roe vs. Wade" and, as part of his production studio upgrade, he has purchased a **Sony** APR-24 24-track analog recorder to be used on other TV music projects including "The Outsiders" and "Working Girl" and an upcoming NBC movie entitled "U.F.O. Cafe."

The recorder has a bar graph metering system, multi-features synchronization facilities, and amorphous record and play heads. "Its simplicity is what makes it so great," says Walden. "Operationally, it works great because it has a jog shuttle wheel that lets me move along one frame at a time as I'm going through the video, allowing me to keep track of the SMPTE timecode." The APR-24 also offers 14-inch reel capability, 15/30 ips selection, processor-assisted keystroke alignment, and remote audio, transport, locator and synchronizer control.

Space Is The Place

THERE'S yet another spatial imaging process to consider when you're considering mixing your next project with **Q Sound** and **B.A.S.E.**

Roland has introduced the Sound Space Processing System, a technique that reproduces a three-dimensional aural environment on a conventional two-speaker system. It too can be used for mixing, mastering, video soundtrack production, and TV and FM radio broadcasting. What's more, the processing system requires no additional equipment for playback.

The basic components include a Sound Space Processor unit, two A/D/A converters and the Sound Space Controller unit itself. The processor unit processes four signals simultaneously, and transaurally processes binaural sound for reproduction through two-speaker systems.

Azimuth and elevation of sound images are adjusted by the rotary dials of the controller unit, while MIDI I/O connectors provide manipulation of channel signals via MIDI messages from an external device such as a MIDI sequencer. The processing system can be used to design a particular soundfield on every track during mixdown on a multitrack recorder.

Personal DAWs

continued from page 19

storage at 44.1 kHz and is expandable to five times that size. In an attempt to overcome the inherent up/downloading time requirements of Winchester disk systems, an integrated streamer backup tape system is standard, although the opportunity to see it in action didn't arise at the show. A 16-track sequencer is onboard, along with good video interfacing (LTC, VITC and RS422 ports), both analog and digital I/Os, external digital word clock sync and a pair of MIDI connections.

The front panel is rather pleasing to the eyes; while faders are assignable to various functions, bussing is done more conventionally, a jog wheel is provided for scrolling, and audio program controls are activated by the familiar "PLAY, FF, REWIND," etc. buttons. These make the Korg system quite instinctive. Still under development, the company maintains that it will be available by late summer/early fall of 1991 for about $30,000.

The **Roland** DM-80 Hard Disk Recorder is also an exciting new technology statement. Its 100 megabyte hard drive holds four tracks, which can be doubled to eight internally as an option with a 200 megabyte drive. The DM-80 is interfaceable with a Macintosh computer as a controller, and recording time can be additionally extended by hooking up to a magneto-optical drive via a SCSI port. Recording time at 32 kHz is 25 minutes; at 44.1 kHz it's 18 minutes; 16 minutes at 48 kHz. The unit has digital EQ and the system requires an external mixer.

Unfortunately, the demo system at AES wasn't ready,but a discussion of layout should help. The DM-80's main interface is the remote, and its outward appearance will be more familiar to sequencer hardware users than to recording engineers, although a Macintosh screen should give some comfort when using multiple units for more tracks. A 240 x 64-dot LCD screen provides clear, easy-to-read access to the unit's functions, and a jack is provided for connecting the remote to a QWERTY keyboard. A jog wheel gets you around the functions and allows for scrolling. There are eight location markers hard-assigned to the front panel. And PC-type function buttons coexist well with tape recorder-type controls here. Roland says the system should be ready next year at an as yet undetermined price.

Incidentally, all three systems address video applications with SMPTE and other coding systems and sample rate conversion capability (real-time or otherwise). The fact that they are not fully ready to market means that the companies are watching wind direction very closely in terms of features and software updates before committing themselves fully. The bottom line, though, seems to be that a variety of systems using a variety of approaches and a limited number of formats is on the very immediate horizon. And now because they're being offered by MI manufacturers, you can also see these digital audio workstations at NAMM in January.

—Dan Daley

New Heads Are Better Than Old

TO THE delight of many vintage analog tape machine owners, **JRF Magnetics** has recently introduced the PLX line of replacement heads, which includes a set for the popular TEAC 80-8 1/2-inch eight-track tape machine.

"We're trying to make maintenance cost-efficient for people who want to use discontinued tape machine models," said John French, JRF Magnetics president.

The TEAC 80-8 is a very widely-used analog workhorse that had been difficult for some owners to maintain due to high replacement parts costs.

"The new heads have also been designed to improve low end response at higher speeds, and retain that quality two or four times longer than the standard heads," French added. A testament to the TEAC 80-8's widespread popularity and longevity is the fact that French recently bought one himself and added the new heads.

Replacement heads for two other vintage models, the **Tascam** 8516 one-inch 16-track and the **Ampex** MM1200 two-inch 24-track machines, have also been made available.

French describes the niche his company has carved by saying, "We're trying to service the users of older machines who don't want to stop using a workhorse machine." For those who want to deal direct with TEAC, replacement record/playback heads are available at a retail price of $778.

For more information contact **JRF Magnetics**, 249 Kennedy Road, PO Box 121, Greendell, NJ 07893, 201-579-5773, or **TEAC Corporation**, 7733 Telegraph Road, Montebello, CA 90640, 213-726-0303.

TEAC 80-8

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ROOM WITH A VIEW

Maui Wowie

Steely Dan's Walter Becker builds himself a place in paradise – with a little help from his friend, John Neff.

STUDIO NAME: Walter Becker's Home Studio, Ulupalakua, Maui
OWNERS/OPERATORS: Walter Becker, John Neff
CREDITS: Windham Hill Records, Warner Brothers Records
STUDIO DESIGNER: Peter Maurer, Bau-Ton, Los Angeles, CA
DESIGN NOTES: The 18 x 22 foot control room was built in a redwood house where Becker formerly used to practice, featuring 15 feet of glass looking out on the islands of Kaho'olawe and Molokini with views of Lanai and West Maui...The studio is located in a separate building and is viewed through a closed circuit monitor from the control room...The iso room is built on a separate slab and physically doesn't touch the studio itself...Wiring is in accessible conduits under the ground.
CONSOLE: Soundtracs IL 4832, 48 x 32, two effects return modules for an additional eight effects returns, extended patchbay for additional 96 tie-lines, Penny & Giles faders, automation-ready.
TAPE MACHINES: 3M 32 track and four track digital, Mitsubishi X-80 two track digital, Fostex E-16 16 track, Fostex Model 20 center timecode two track, Sony DTC-500 DAT, Panasonic SV-3700 DAT.
MONITORS: Meyer HD-1, Toshiba closed-circuit TV link to the studio.
OUTBOARD GEAR: UREI LA-4, dbx 161, UREI 1176LN, dbx 160x, Rane DC24, Sony MUR-201, Alesis Midiverb, Alesis XT:C, Roland SRV-2000, Yamaha SPX90, Korg SDD2000, Roland SDE-1000, Omnicraft GT-4, Drawmer DW201, Massenburg GML 8300 preamps, Massenburg GML 8200 parametric EQ, Meyer CP-10 dual parametric EQ, Tube Tech PE-1b, Summit four band parametrics, Yamaha Q2031, Wendel Jr, Russian Dragon time comparator.
KEYBOARDS/SAMPLERS: Korg SGX-1D digital grand piano, Yamaha DX7 II FD, Yamaha DX7, Akai S1000, Akai S950, Akai MPC-60, Oberheim Matrix 1000, Roland D-550s, Kurzweil 1000PX plus, Korg M-1REX, Yamaha TX802, Roland Super Jupiter MKS-80, Yamaha TX816, Rhodes 73, Hammond B3.
MICROPHONES: Neumann TLM 170, Neumann U 87A, AKG C414EB, AKG D112E, AKG 451EB, Schoeps CMC5-U, B&K Series 4000, Shure SM57, SM58, RCA 77DX, RCA 44DX, Countryman Omnimax.
MIDI: Opcode Studio Vision, Digidesign SoundTools, Elka removable hard drive, Apple Macintosh IIcx.
EQUIPMENT NOTES: Says John Neff: "I like the inline design of the I/O of the Soundtracs IL because you can maximize the number of inputs on the console in a relatively small space; it is extremely well laid out, the EQ is flexible and very British sounding and it is very quiet to the digital multitrack–better than some of the older Neve consoles that Walter (Becker) likes; we needed a 32 buss console to accommodate our digital 32 track – this was the least expensive one that still afforded all the features associated with higher end consoles." He adds, "With the Meyer HD-1s whatever you hear translates to any other medium, car stereo, home stereo, boombox. If you have it together in the Meyers, it stands up in everything."
PROJECT TIPS: Neff says, "Everyone builds a studio so they don't have to pay an hourly rate when they work on their own projects. However, no one has an unlimited budget—you have to make some compromises in order not to go commercial with the room. The main advantage of a project studio is that you can take the time to experiment with sounds or compositional ideas without always keeping an eye on the clock."

Becker's fully-loaded, fully-floated room is in an all-redwood house and features 15 feet of glass looking out on the islands of Kaho'olawe and Molokini with views of Lanai and West Maui...


SOUNDTRACS
IL4832

Private Ears

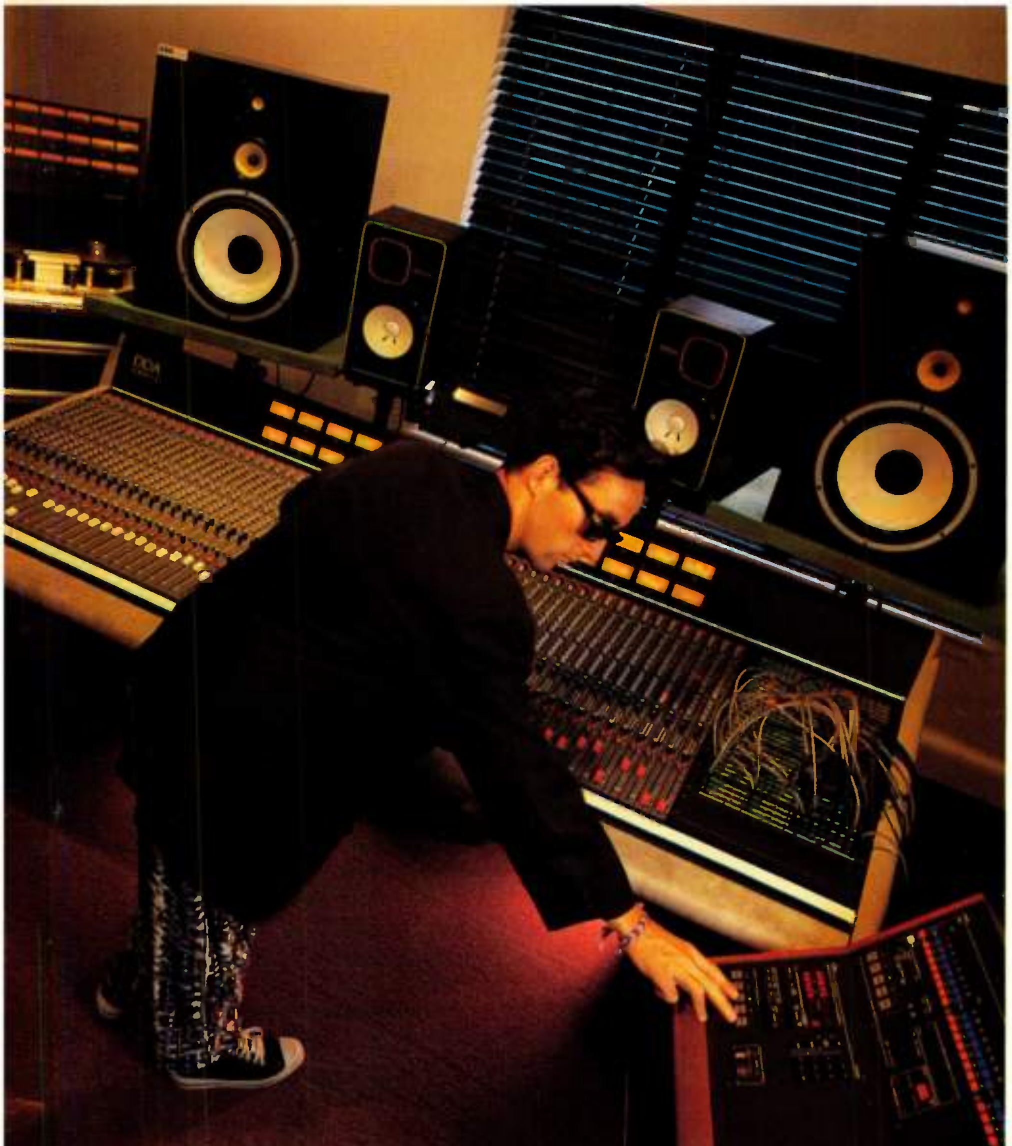

Photo: David Safian

Songwriter/producer Steve Harvey has built an expansive MIDI-recording facility in an ordinary Hollywood office complex.

STUDIO NAME: Scotland Yard, Hollywood, CA
OWNER/OPERATOR: Steve Harvey, songwriter/producer
CREDITS: Nia Peeples, Karen White, The Commodores, Gerry Woo, new solo LP (A&M Records)
STUDIO DESIGNER: Audio Intervisual Design, Los Angeles, CA
CONSOLE: DMR 12 (DDA/Klark Teknik), inline or split with separate monitor section, 32 x 12 x 24 (or 56 x 12 x 2), dual line inputs, full patch bay, DDA MIDI Muting Automation System capability.
TAPE MACHINES: Studer A827-24 track, Panasonic/RAMSA SV-3700 R-DAT.
MONITORS: KRK-1303 three-way near-field, KRK-703 at synthesizer rack, Yamaha NS-10.
OUTBOARD GEAR: Sony MUR-201 stereo reverb, BSS DPR-402, BSS DPR-504 limiter/compressor, Audire power amps (Forte, Crescendo), Roland R-880 reverb, GML 8200 parametric EQ, GML four channel mic preamp.
WIRING: Mogami and Yaleflex
KEYBOARDS/SAMPLERS: Roland S-770 (16 megabytes), Akai S1000HD, Roland D-550s, Korg M1R, Yamaha TX802, Roland MKS-70, MKS-50, MKS-20, MKS-80, Roland Vocoder SVC-350, MIDI Moog, Harvey 808, Akai S900, Proteus, Roland U-220 sample playback unit, Yamaha TX816 (eight modules), Roland A-50, Octapad Two, Roland PD-11 bass drum pad.
EQUIPMENT NOTES: "The DMR 12 is extremely flexible; if I decide to mix here I have a lot of inputs to work with; my engineers say it's a very musical board and for the money it is excellent... Some think the Roland R-880 is a poor man's Lexicon but it actually has its own sound; for the money it is also extremely powerful and features a remote that gives you a clear visual readout so you can see what you are doing...I spent a lot of money on the tape machine but getting the Studer was essential; it sounds great and it is extremely reliable; you get more than just the name when you get a Studer."
PROJECT TIPS: "Building my own studio enables me to spend as much time as I need as opposed to having to spend as much time as I can afford...It also cuts down on the time it takes me to set up; before, I would go to a studio and take a half day just to set up my keyboards...I take my projects as far as I can here and take my tapes to a studio like Larrabee for final mixdown by one of the leading mixers in town...I've been recording my keyboards to 48 track by first filling up the 24 track, then striping another piece of tape and dumping a stereo mix with the same SMPTE offset; I end up slaving them both up when I get to the mixing suite...The way I've set up my control room, all my keyboards are wired into a separate patchbay and I use Roland M16E mixers (16 x 2) to audition the sounds. When recording, I put them through the GML equipment and the DDA console to tape...I'm being very careful about EQ'ing too much these days since I find it takes the life out of the sound; my demos were actually sounding better than my masters because I was going in and EQ'ing everything when it was going down and when it was coming back and then going to the mix process."

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Inside DAT Mastering

Project studios need to prepare their DAT correctly to make it work in the mastering suite.

BOB LUDWIG

IN JULY 1987 I mastered my first Compact Disc recorded entirely on a DAT machine. Bob Clearmountain had mixed Robbie Robertson's solo album both to analog 1/2-inch 30 ips and on his new consumer DAT machine. We listened to both formats and decided that, for this project, the DAT seemed to sound better than the analog tape and we mastered the recording from that digital source.

Bob Ludwig (left) with Lou Reed (photo: Peter Monroe).

Since Robbie's album, a lot of DAT recordings have been used as the master tape for many projects even where the budget was no object. DAT has found a home with the audio experts, musicians and with project recordists.

What do you need to consider when preparing the tape for mastering? Consider these: tape back-ups, maintenance, tones, formats and editing.

BACK TO BACK

The cardinal rule for any digital project, be it multitrack or two track, is to make multiple masters of the original or immediately clone your only master to a backup tape. We have a slogan at Masterdisk saying, "Never turn your back on digital." If you don't have a system like the PCM-1630, which can computer verify your transfers as being accurate, one can never leave the monitors while copying digital. Simply because you have played your tape back a dozen times successfully does not mean it will play back correctly on the next pass.

Lots can go wrong with digital if you don't pay close attention to it. For example, it is easy to pick up ticks on your copies if the dubbing cables or word clock are not correctly synchronizing the transfer. Sometimes these ticks are very soft, so be sure to listen carefully.

At least make two copies on the same piece of tape stock. It is better to make a copy on two different pieces of tape stock and better still to record on two different transports. If you are traveling any great distance to your mastering session you'd better bring both copies (with a clone left at home) as things going wrong at a mastering session are often measured by the square of the traveling distance to get there.

While the above suggestions apply for all digital formats, they have special meaning for DAT. The metal tape formulation used to record on DAT is very robust; indeed you can put your DAT master on the average degausser and nothing

You can't afford to spend less on a DAT. And you certainly can't afford to spend more!

Introducing the new, improved Pro-DAT Recorder from Panasonic: the SV-3700. "New?" "Improved?" Aren't those just overused buzz-words for breakfast food or laundry detergents? Sure. But the SV-3700 is new, and it is improved, and we want to tell the professional user about its 4th generation technology as directly as we can.

You know the sonic and performance excellence of the Panasonic SV-3500 DAT. Who in the industry doesn't? The new SV-3700 incorporates significant advances in analog-to-digital converters, digital-to-analog converters, transport accuracy and reliability plus control and interface technology. In the SV-3700, one bit A-D converters bring dramatically improved performance and linearity compared to conventional successive approximation A-D converters. These high-performance one-bit ADCs reduce both signal and zero cross distortion producing cleaner, clearer audio at low as well as high levels.

Subjectively, these technical advances translate to accuracy in the spectral balance, ambience, and "space" around instruments that form the most important part of natural musical and "correct" sound.

Oh yes, there's another very important improvement in the SV-3700. Price. When you visit your dealer, you'll find that in professional applications you can't afford anything less than the SV-3700. And you certainly don't have to spend more. For more information, contact Panasonic Pro Audio, 6550 Katella Avenue, Cypress, CA 90630. 714-373-7278.

Panasonic
Professional Audio Systems

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will happen to it. Yet, because the tape is so thin, it is always possible to jam and be destroyed within the tape transport. The advantage of recording on more than one transport is that on rare occasions a deck will not have its tape guides aligned properly or it will be electrically misaligned. While it will record and play back fine within itself it may not meet DAT specifications and thus will not replay properly on another machine. Recording on two machines lessens this (thankfully) rare predicament.

What about tones? If you record on a professional DAT machine a 1 kHz reference and balance tone may be all that is needed. Feed a tone into both channels of your DAT recorder and set the meters so that the indicator lights just barely come on at a level of -14 dB from the maximum "0." (If you're a pro, your mastering facility can make you a digitally generated 1 kHz tone that is precisely balanced for machine set-up.) If you start a recording and find you are going into the "OVER" area beyond the "0" maximum, do the alignment over again at a lower level such as -16 dB. Do this until your normal mixing level never goes into the "OVER" and yet hits the "-3" to "0" area often. If you go into the "OVER" area but don't hear anything wrong you are probably okay. On some machines the "OVER" light comes on if a single sample is saturated. If it is a severe "OVER," bad distortion can result, especially with piano or other transient instruments. Many CD plants will not tolerate a single "OVER" on the final CD master tape, so my advice is not to have "OVER"s on your tape.

If you have a professional DAT machine normally set the emphasis OFF, the Copy Prohibit OFF and the sampling rate to 44.1 kHz to avoid having to use a sample-rate converter in the digital domain and to make editing simpler. Most digital editors only operate at 44.1 kHz. In the analog domain, 48 kHz may be appropriate.

PRO DAT?

Recording on a consumer machine can be tricky: many of them are made with only a single analog-to-digital converter that is used for both left and right channels causing an 11 microsecond delay when played on a professional CD mastering system. If you are not sure about the machine you are using, record a 10 kHz or 15 kHz tone as well as the 1 kHz onto both channels (mono) and the mastering engineer will see the timing error on his oscilloscope and correct for it with an Harmonia Mundi digital domain card. It can also digitally handle the pre-emphasis that your consumer machine has probably used.

Razor blade editing on a reel-to-reel digital tape machine makes several millisecond editing accuracy possible. Such is not presently the case with your DAT machine. If you need accuracy to within a few milliseconds or require waveform editing, you need to locate an editor. If you are not attending the mastering session, be sure to enclose a note with any suggestions you want to make to the engineer. Never assume they'll fix something automatically.

DAT has a secure place in professional recording — regardless of your budget. If you use a little care and common sense, mastering from DATs can be effective, inexpensive and relatively risk-free.

Ludwig *is chief mastering engineer of Masterdisk in New York, with too many credits to list here (or anywhere).*

Home With A Range

Nashville virtuoso Mark O'Connor builds his studio from the ground up.

THOMAS GOLDSMITH

NASHVILLE SESSION musician Mark O'Connor has built a reputation as a guitarist and fiddler for artists ranging from Emmylou Harris and James Taylor to Andreas Vollenweider. But he has built the newest instrument in his repertoire all by himself — the new 24-track Hometone Studios that he constructed inside his hillside suburban home.

"I was here almost every day while the studio was being built, and had time to correct problems and minor flaws as they came up," O'Connor says. "I did most of the construction myself; my dad was a carpenter, so I know how to build. For technical assistance, I just started reading a lot of books and asking engineers a lot of questions. The most common but most valuable advice was to use your ears first — and if you like the sound, you're probably doing something right."

Laying down tracks with his CAD board and Otari MTR-90 II (photo: Beth Gwinn).

O'Connor has been turning heads with his musicianship since he started winning fiddle contests at the age of 12. Stints playing guitar, fiddle, and mandolin with David Grisman and the Dregs followed, along with a mid-80s move to Nashville, where he established a jazz recording career and became one of the brightest lights on the session scene.

O'Connor is not putting aside his studio career for the recording business. For the most part, he simply enjoys the convenience of doing master-quality work just a few steps from his living room. In fact, all the solo passages for his recent Warner Bros. LP "On the Mark" were recorded at Hometone. He's also using the room to write a string quartet commissioned by the Santa Fe Chamber Music Festival.

SOUND FOUNDATION

Planning for the basement studio, which consists of a 14 x 24 foot control room and a 12 x 20 foot recording studio, began concurrently with construction of the entire house. Before the sheetrock was laid, Nashville engineer Larry Cummings helped install power and audio wiring for the recording equipment, as well as for microphone and headphone lines. Then there came the process of "floating" the studio in order to minimize noise. The studio's walls aren't connected to the outside walls, and rest on dense rubber. Meanwhile, the ceiling was built into the inside wall and was isolated from the ceiling above it. "The wooden floor rests on a dense foam pad, has a 3/4-inch layer of plywood underneath, a soundboard underneath that, and the regular 3/4-inch ply subfloor underneath it all," O'Connor says. "All the plumbing, too, has huge pipe insulation around it, so you don't hear the water running. In a home, that's not an easy thing to do, so I had to watch carefully where all the pipe was going. Knowing that gives

THE SOUND OF PERFECTION

WAREHOUSE SNARE
REV:POSITN= 83%

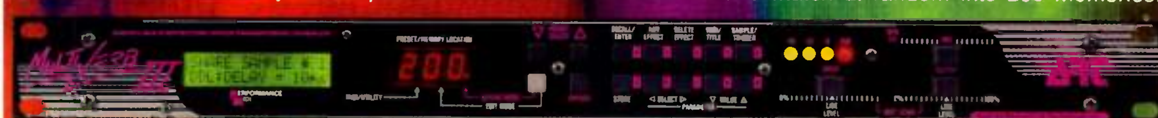
INFINITE PLATE
REV:DIFFUS= 100%

DREAM FLANGE
7 CMP+EQu+EXP↗

NIGHT DEMON GTR
7 EQu+CMP+DST↗

MIDICTL=176 CH01
MODWHEEL=001:054

A short time ago, A.R.T. stunned the recording world with the release of the SGE Mach II. Offering 12 simultaneous effects and a 400% more powerful processing section, the Mach II offered spatial realism that defied description. The all new DRX uses that same processor and is expressly designed for studios and live sound applications. The DRX will do 10 simultaneous audio functions and features an exciter, compressor, limiter, noise gate, expander, envelope filter, 24 different reverb algorithms, 21 different delays, sampling, pitch transposing, panning, equalization, leslies, stereo flange and chorus and more—over 60 effects to choose from with bandwidth to 20 KHz! The creative power is astonishing. The noise gate can gate off microphones so the wash from live drums doesn't trigger your effects buss. The compressor can smooth out wild dynamic swings on vocals. The exciter will increase the edge and clarity of any type of material. The noise gate can "turn-off" noisy guitar amps in between songs or allow you to run higher gain levels without feedback on vocal and drum mics. The limiter can hard limit any source so that clipping can be totally prevented. And you can pick and choose effects and mix and match at random into 200 memories!

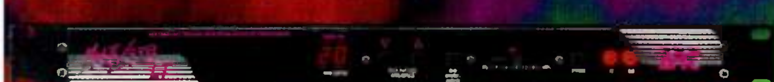


SHIMMERING CHOR
7 EQu+CMP+EXP↗

BRITISH THUNDER
PTr:PITCH = -12

PTCHBND=224 CH01
LSB=126 MSB=127

The all new Multiverb III uses the same revolutionary processor as the DRX and offers more than 50 effects to choose from! The Multiverb III features everything the DRX does except the dynamic effects section (comp/limit/gate). It will do four simultaneous effects and unlike other units allows you to pick and choose effects at will and change their locations-you're not limited to confusing configurations. Like the DRX, it features a Midi Data Monitor that allows you to see the digital midi data stream—simply connect a keyboard, foot controller or any other midi device and the LCD will give you a real time readout of channel pressure, patch change or any other midi info! And the sound and spatial realism of the Multiverb III is absolutely stunning.



The Multiverb LT offers the power of the Multiverb III in an ultra simple format. It will do 3 simultaneous effects and contains 192 of the finest studio effects combinations ever created. Lush reverbs, delays, flange, chorus, and special effects combinations are available at the touch of a button! The sound for the price is unbelievable—and midi addressable. The NEW X-II Midi Foot Controller works with all midi effects units—at a great low price.

CIRCLE 03 ON FREE INFO CARD

ART

Applied Research & Technology • 215 Tremont Street • Rochester, NY 14608 • (716) 436-2720 • Fax (716) 436-3942 • Telex 4949793 ARTROC

you more confidence. You don't have to worry that somebody's going to flush the toilet."

O'Connor installed double-glass in the windows between the control and recording rooms, as well as in the windows facing the woods surrounding his house.

SOUND SYSTEM

A semi-cylindrical bass trap/diffusion device covers the back wall of the studio, and insulated, fabric-covered wall panels deaden the room when necessary. O'Connor's 24-track Otari MTR-90 II sits in a separate sound-insulated closet, which has a glass door for monitoring the machine's VU meters during recording. Near-field monitoring is supplied by a pair of Tannoy Little Gold monitors. His console is a 24-input, 12-output board manufactured by Conneaut Audio Devices, a company in Conneaut, Ohio.

Of the console arrangement O'Connor says, "I don't use the board to record with, only to monitor with." Recording signals from microphones or direct boxes run into the patchbay, into Massenburg mic preamps and directly into the recording machine.

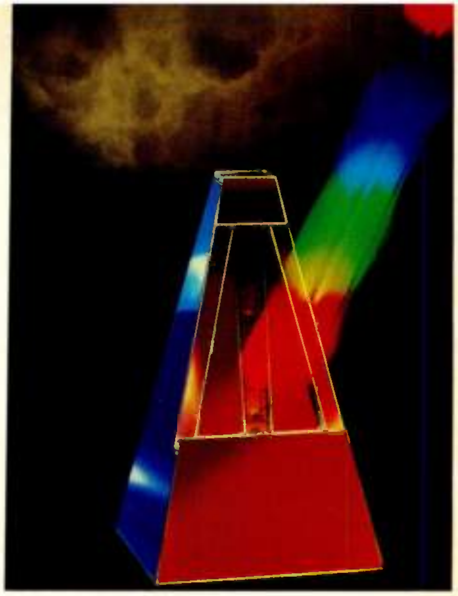
"If I need EQ, I go to my outboard gear; I have two API 550s," O'Connor says. "I go out of the API into the machine." The studio is also equipped with two Teletronix limiting amplifiers, a UREI 1178 stereo compressor, and a Drawmer stereo noise gate.

The main recording microphone in Hometone Studios is a vintage 1957 AKG C24 stereo tube mic. "I use it for everything, but I especially like it for my violin," O'Connor says. "Though most of the time I just record mono, the stereo is there if you want a big, across-the-room fiddle or violin sound."

O'Connor has also had to devise his own method of punching himself in on the overdubs. "Producing, engineering, and recording my own sessions means using a hands-off method of recording myself. Basically, I take the remote control off the stand and put it on the floor. Then I put the right amount of pillows next to it so I can put my foot on it, and be able to hit the RECORD button with my toe."

Now that the construction of Hometone studios is over, the only worry that O'Connor has is that its location might sometimes be too convenient. "You have the feeling sometimes that you'd rather watch the ball game instead of using the studio. The only real pressure comes from my own discipline."

The MING Dynasty



Using a metronome to ring out a live sound system

WADE MCGREGOR

THE PREDICAMENT is not hard to imagine: You are facing a concert sound system that includes the usual array of microphones, mixer, amplifiers and loudspeakers. The last time you dealt with it, a soft-spoken singer needed so much amplification that the whole system began to scream, howl and squeal like the Siren Research Unit at Bell Labs. How can you amplify those vocals without the feedback?

You could get the assistance of a bystander to jabber into the mic while you try to equalize out the worst feedback rings, but he will probably wander off after the first screech. You could clap your hands, turn up the pink noise, or even just let the room noise excite the system into feedback — but wasn't that what you did the last time with such painful results?

There are sophisticated tools to assist you in this endeavor — the Techron TEF, the Meyer SIM-CAD, S/T RTA-2 or one of the many automated equalization systems. Unfortunately, the person paying for this job can't afford such an expensive solution. They would rather you just amplified things, as necessary, without all that feedback — thank you very much.

MING RING

Fortunately, there are methods that require much less expensive hardware. Don Davis showed me a technique at a Syn-Aud-Con seminar that is very effective for "ringing out" a sound system (ringing meaning measuring feedback oscillations). You apply a tone generator, sweeping through the audio band, to the system through the same loudspeakers that the microphone is routed to. Then you increase the gain of the system after each sweep, until the system begins to "ring." The frequency that sustains after the tone sweeps by is the one that's close to feeding back.

At that point, you notch out that particular frequency with an equalizer, and increase the system gain until the next ring becomes audible. All you have to do is keep repeating the process until you've got sufficient gain or you pass the point of diminishing returns in equalization, (i.e. multiple frequencies are ringing together, or more sliders on the EQ are down than are at zero).

There are only a few problems with this method. First, to be truly effective, the tone has to be loud enough to swamp any other noise in the room and the sound system. No one wants to listen to it sweeping through the upper mid-range at high volume. Second, you have to manually turn a knob to sweep the tone, leaving you with only one hand to either adjust the EQ or grab the fader if the feedback suddenly takes off — and feedback can not only be painful, it can permanently damage loudspeaker drivers, especially at higher frequencies.

I found a better way to get the job done: using a spring-activated metronome available at any local music store. The model I use, a Wittner Taktell Piccolo, produces approximately 70 dB SPL C weighted, RMS at one cm. This meant that its acoustic output is similar to conversational speech measured at the same distance.

I decided against the smaller and "higher tech" electronic models because their output is more of a beep than an impulse; as such, they lack the harmonic content of their mechanical counterpart. The mechanical metronome has an acoustic output that is consistent over time, does not require calibration, maintenance or batteries, costs less than $60 and will fit into a good-sized pocket.

The metronome was thereafter called the Mechanical Impulse Noise Generator, which looks great on a purchase order, and was dubbed the MING by my assistant. The harmonics that are generated by the MING impulses are sufficient to allow the trained ear to make reasonable guesses about the qualities being effected by the EQ, as it is applied.

HOW TO MING

To get rolling, just wind up the MING, put it in front of a microphone at about the same distance as the singer, talker or musical instrument and set it in motion. The sound technician then returns to the mixer and gradually increases the gain of the microphone in the sound system until he hears a feedback oscillation.

As the feedback becomes audible, you must decide whether to make mic and loudspeaker placement adjustments or make adjustments in the frequency response of the sound system (EQ). Keep upping the gain of the microphone until you can't make any more reasonable adjustments to eliminate feedback oscillations or until you arrive at a gain setting far in excess (>10 dB) of what the situation demands. If modification of the sound system physically or electronically is not appropriate, then at least you are aware of the gain settings that are below the threshold of feedback — as well as those that are above it.

Keep repeating the process for each mic that could be in danger of feeding back during the event. Turn on all foreseeable combinations of microphones to see if any of the combinations will cause a shifting of the feedback threshold. Moving the MING from one mic to another will help make all potential problems more audible.

MEASURING MING

With the metronome in place and ticking at a reasonable rate, it is easy to compare EQ settings, gain before feedback margins, stage monitor levels and their effects on "front of house" sound qualities. The impulse will also show up flutter echoes, slap echoes, and other acoustic anomalies that may be masked when simply playing back music to check a system. The MING's impulse will also make open microphones that are near feeding back more obvious, as they cause an increase in the apparent reverberation time of the room, which is often audible as a hollow quality that disappears when the system/mic gain is reduced or when you apply appropriate equalization.

By muting the sound system, you can make a direct comparison of the amplified and acoustic qualities of the impulse. This lets you make a judgement on whether tone or perspective is being altered in a positive manner. If you have someone to assist you, walk around the venue while the system is occasionally muted/unmuted, so you can discover what changes in acoustic imagery are being achieved in amplifying the original sound source.

The impulse creates many time domain distortions within a sound system, especially those that are the product of speaker placement, microphone placement, array interaction etc. The use of the MING as a source for checking signal delays within a distributed system will reflect whether the arrival is within the integration time of the human auditory system, and can be directly compared to the acoustic arrival time.

After a season of using the MING, I found I had been able to equalize a sound system in a shorter period of time, using less drastic EQ, achieve more consistent results, and have a high degree of confidence that the sound system was optimized to the venue. I found it worked much better than handclaps, tapping the microphone or depending on incidental room noise to excite feedback oscillation. In those worst case situations where an amateur performer would arrive too late for a "mic check," or a professional performer would ask for a lot more monitor level during the performance, I was always confident that I had maximized the gain of the system without limiting the quality or intelligibility of their voices.

The metronome has an acoustic output that is consistent, does not require calibration, costs less than $60 and fits into a pocket.

The Mechanical Impulse Noise Generator is certainly not the only tool that can achieve this result, but it is a small, inexpensive, and wireless acoustic radiator. With a reasonable theoretical understanding of acoustic sound transmission and audio signal processing, it can yield results quickly and effectively. To be an effective tool, the MING requires a sound system that is in need of small adjustments and not wholesale revision, as well as well-trained technicians with "ears of experience" to perform the appropriate tweaks and adjustments.

***McGregor**, former Head of Sound for the Calgary Center for Performing Arts, is currently a systems designer for DSE Production Equipment and Services.*

Mostly Mozart

What you need to know about Amek's new automated console. Or should we call it a workstation?

IRA CORD RUBNITZ

RECORDING CONSOLES are slightly intimidating until you realize that, just like cars, they all basically do the same thing. This is important for all recordists and engineers to understand once they leave the security of the project studio for the thrill and challenge of finalizing the mixdown at that platinum studio down the road.

Amek has come up with a very hot and exciting new console in the form of its new Mozart. Most musicians and engineers will understand how to get it moving, but it takes considerable experience to throttle its creative power to the max. Amek has built a technological support system for the board that opens up worlds of possibilities to the creative engineer. What's more, the company has recently hired Rupert Neve as a consultant, and he has come up with some great new designs for the company.

As a result, I was only too keen to get my hands on a custom Mozart recently installed in the Bakery Studios in North Hollywood, California. Basically, I discovered, consoles are becoming workstations — and the Mozart is a case in point.

BOARD ROOM

First impressions do make a difference. The console was very well laid out and after browsing through the operations manual, I felt familiar with the extension effect of having pots and switches where you get used to them and not have to play hunt and peck.

The faceplates on the console I drove were a light grey with white silkscreen letters embossed on them; as a result, reading this particular desk was virtually impossible. Today's Mozarts have a dark grey background with white lettering that makes quite a difference. My only other complaint was the buss switches were placed much too close together.

The first indication that the Mozart is a performance vehicle is its comprehensive Group Master Computer section. Located in the center, it is divided into three sections. The Master Status Control switches control the status of the grouping system by selecting the channel number and page number of the switch function in addition to changing modes. Underneath are the Channel Function Group Master switches, which are assignable Group Master switches for channel functions of the Input Module soft keys such as the EQ, Sends, Inserts, etc. Finally, the Mute Group Master section consists of eight auto mute group switches assignable to any groups of channels — extremely useful for creative muting using different switches for each part of a song. You can store up to 99 pages of console setups in memory.

> "It controls just about everything but the coffee maker."

The Mozart was designed in conjunction with a graphic moving fader and recall computer package utilizing an Atari Mega ST computer and the Amek/Steinberg Supertrue software mixing automation. This easy-to-control package actually makes the Mozart/Supertrue system a virtual workstation, letting you control just about everything except the coffee maker. Reading SMPTE, Supertrue is not only used for fader moves and mutes, but it can also handle many different events with quarter frame accuracy — accurate to 1/120th of a second.

The program is interfaced with and can be accessed by a keyboard or trackball with two buttons located in the center of the desk. Point, click and pull down those menus. The main page/screen resembles a mixer with faders, groups and mutes plus a camera icon for snapshots and SMPTE readouts in any size and transport controls for your sequencer—yes, that's right, you can slave your sequencer to the system for yet more of a workstation approach.

Basically, Supertrue is a well-developed graphic sequencer used to create automated mixes. Besides bypassing the VCAs (which, on this console, are dbx 2151's) and automating the moves of the Penny & Giles faders (equipped with self-cleaning wipers), Supertrue can also automate eight mutes for each channel mute and 15 switches per channel, plus it can handle MIDI event triggers, cue lists, merging/copy/paste, and automatic drop-in functions. Software-defined noise gates add even more workstation-style power.

Another strange but useful feature are the Fader Jobs — graphic representation by "drawing" prerecorded fader movements for advanced and highly elaborate offline control. You can hand-draw fader moves or call up libraries of 16 varying movements. The automation graphically follows fader moves and mutes. Mozart and Supertrue can be used quickly and simply, like an "analog workstation."

After doing tracking, some overdubs and a Supertrue mix, I found that the Mozart/Supertrue was the most innovative package I've experienced in some time.

Mozart has all the basic tools you need. Its EQ section, which includes High (from 80 Hz) Pass and Low (from 10 kHz) Pass filters, for example, is a four band (Lows: 30 Hz - 1.2 K; Mids 1: 40 Hz - 2.5 K; Mids 2: 350 Hz - 15 K; Highs: 500 Hz - 20 K) semi-parametric with shelving or bell curves, and dB cut and boost is +/- 18 dB. It comes through the way you want.

Basically, the board combines a hot sonic package with the tools that put a wide variety of producer's techniques at your fingertips. When you're driving the Mozart, creating music is elementary. The only question is whether to call it a console or workstation.

Or maybe you should just forget it and enjoy the ride.

CIRCLE 11 ON FREE INFO CARD

Captain Timecode

NOT SO VERY long ago timecode was a mysterious term that couldn't find its way past the receptionist in even the highest tech recording studios. Then again, such video verbiage was the exclusive domain of those other tweaks who played with pictures and knew little or nothing about sound.

But now that even heavy metal rockers recognize the importance of video to making their daily bread, it is important to remember the contributions that, for better or for worse, a New York studio owner/engineer named Bob Liftin made to making the technology match between sights and sounds.

In the latter days of his pioneering facility, Regent Sound, Liftin was responsible for the audio for "Saturday Night Live" as well as for the new cable upstart channel MTV. Just as video would ultimately impact audio recording from the broadcast networks to the MIDI studio, these projects were the inevitable offshoot of a music career that began recording singles in the 1950s and 1960s for artists ranging from Little Anthony and the Imperials to Roberta Flack, Aretha Franklin and Bette Midler. By the time of his death, there were few in the industry who remembered these credits, however; most knew Liftin as an apostle with muttonchops for audio-for-video and SMPTE timecode.

For Liftin, the innovation that would change the way audio-for-video was handled began during an audio date. In an interview he conducted ten years ago this month, Liftin retold the story in his own words:

"In 1973 I was recording Les McCann who was doing an album in which he was playing all the keyboard instruments himself, which was something special at the time, and he quickly ran out of tracks. We had 16 tracks at the time and two studios and he wanted to know why we couldn't take our machine from Studio B and put it in Studio A and then have 32 tracks and keep on going with overdubs. So I sat down with one of my engineers, trying to figure out how we could synchronize these machines.

"Then two or three days later, as coincidence would have it, I received a letter in the mail from the Electronic Engineering Company of California (EECO) and in it was the description of SMPTE timecode which had just been approved. SMPTE

When they write the history of audio-for-video the text should be dedicated to the memory of one man, Bob Liftin.

MARTIN PORTER

was an 80 bit code identifying a certain spot on the tape by putting electronic sprocket holes on the tape and thereby allowing synchronization. At the time, a lot of people said, 'who would ever want to lock up two 16 track machines and synchronize them, that is ridiculous.' But I foresaw this as the way the industry was going to have to go because it was the only logical way of increasing your track capacity without substantially increasing your investment. Also 24 track was happening at the time and I felt very strongly that if you put 24 tracks on the same size tape you were going to lose your signal-to-noise ratio and your dynamic range. I was not a strong advocate of noise reduction simply because I felt it was a bandaid on a situation that was simply created by making multitracks even narrower.

"I guess it was around 1975 that many of our advertising clients were coming by and wanting to look at a picture while we were doing their audio work. So it occurred to me that, instead of going out and buying film equipment, the easiest thing to do would be to buy a videotape recorder and lock this videotape recorder to my multichannel machine, basically via the same SMPTE timecode system. We had the videotape and stripped off the music and put it on our multitrack machine, redid the vocals, did some sweetening and then dumped it back on the videotape. This was a horse and buggy system because you had to cue up both machines by hand and hope they would synchronize.

"We explored automatically cueing at that time. In fact, my cousin was going to Rensselaer Polytechnic Institute and he used to work for me during the summers. He went back to school and told some of his friends that I had this fabulous thing in the studio where they could synchronize two machines. So they came back down to New York and took a look at it. And one of the kids, I don't remember who it was, said, 'How would you like to automatically cue this stuff up?' I said, 'I'd love to.' So he built me a 'search' to cue up the unit using SMPTE timecode. They went on to form a company called BTX.

"About that time I was getting a little scared about the business. The reason was that I just did two albums which ran up $160,000 bills in studio time. And I said to myself at that time, 'This is outrageous.' We had low rates and we weren't socking it to the record companies, it was just that they would block book for weeks and months just to do an album. And I said to myself at that time, 'I don't care if they sell a million albums, this cannot go on. Economically this is not going to work.'

"The result of this was I decided that television, which has basically an economic structure to support these kind of costs and which was much more efficient, was a better way to go. Also it was a challenge. The sound was so bad, nobody really cared about it. There were probably a handful of studios in the United States, two handfuls for that matter in the world, dealing with television sound on a practical level at that time.

"The key to success in audio-for-video is somewhat the same as any service business: produce a superior product at a price your producer can afford. In other words, help keep your client in business — he won't remain your client if he goes out of business or finds a better price elsewhere."

If every wireless company claims theirs is the best, why do they try to copy ours?

First, let's talk technical.

A lot of newcomers to the wireless field are making pretty bold claims these days, but they all have one thing in common. They all say theirs are the best. Well, we've got news for them: all wireless systems are not created equal.

At Nady Systems, when we say we are the leader, we can back up our claims. Everyone knows modern wireless systems use companding for noise reduction, yet few realize that we invented and patented audio companding circuitry for wireless over 12 years ago.

Sure, other companies have attempted similiar designs. Unfortunately, most of them simply fail to utilize the existing technology properly, so at 120 dB dynamic range, we still enjoy up to a 20 dB edge over many of our competitors.

In fact, what we find interesting is that even our early prototypes are still significantly quieter than these so called "new and improved" wireless systems. You may not always need the kind of performance our wireless systems can deliver, but isn't it good to know that the technology is there if it's required?

Let's talk product.

Our Nady 650 VHF Wireless System packs a surprizing number of professional features into a mid price system. It's rack mountable and can be used with up to 10 channels operating on 10 different frequencies at the same time. And, of course, the 650 VHF has a highly sensitive True Diversity front end for drop out free performance in any environment.

The top of the line Nady 1200 VHF Wireless System defines state of the art wireless technology. Features include 20 channel simultaneous operating capacity on custom frequencies. The 650 and 1200 are ideal for any wireless application, such as the theatre, churches, meeting rooms or live entertainment. Musical instrument and lavalier mic body pack transmitters are available. Our 1200 HT handheld is a sleek, metal wireless mic with no protruding external antenna—another much copied Nady innovation.

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So whether you need a wireless system for a modest facility with a small budget or a multi channel wireless for a large location, you can count on Nady to work flawlessly, give you complete coverage and sound superb doing it.

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Nady has been in the wireless business for over 12 years. And we now sell more wireless than the rest of the industry combined. Nady almost singlehandedly pioneered the use of wireless in the live entertainment industry, where road ruggedness and dependability in a wireless system is essential. This means that when you install or use Nady, it will not only work the first time—but keep on out performing other wireless units for a long time to come.

Being on the leading edge of wireless technology practically since its inception, we have given the competition a lot of things to try to copy.

So why settle for imitations? In Wireless there's really only one leader—Nady.

NADY™ SYSTEMS INC.

CIRCLE 27 ON FREE INFO CARD

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Nile Rodgers talks about the art and passion that turn a good performer into a successful producer.

ORGANIC RECORDING

I didn't start off in the music business with my eyes set on being a producer—and to tell you the truth, I still wouldn't describe myself that way, no matter how many production credits I collect. First and foremost I'm a musician.

When I'm in the studio control room, I don't stop being a musician. That's what I bring to every project I work on. And that's what I want to talk about — how you, or anybody, can combine the art and passion that make a good performer with the know-how you need to become a good producer. To me, the "two arts" are really one and the same thing.

Sometimes I think I'm extremely lucky because I get to do the only thing I know how to do, making music, I get to do it full time and I get to do it in both capacities. Frankly, it happened for me the way it's happened for a lot of people in the business—by first producing my own records.

It actually began when Bernard Edwards and I started the band that we'd later call Chic. At first we were a really good rock-and-roll band, but people thought it was weird that we were a black band, because to them we sounded white. Since I'm an orchestrator, we also did a big band arrangement, then we played around with jazz for a while. But then we took a clue from the clubs we were hanging out in and turned to disco. I thought to myself, "Man, this is cool, because you can be a good musician and still go out and play commercial music." Since we weren't the types to sit around and wait

BY NILE RODGERS

for a record company to discover us, we went into our friend Robert Drake's studio in New York and started recording and producing ourselves.

This brief history illustrates how much your life and career can change. For me, I went from being a rock-and-roll musician to launching my career in disco as a musician—and I became a producer as part of the bargain.

ART & SCIENCE

Being a producer today — technically, at least — is very different from what it used to be. When we were doing Chic, we were all just kids starting out. We didn't have sampling; we did everything by hand. But we were all good musicians — that's an extremely important point. Of course, Bob Clearmountain was doing our engineering, which didn't hurt. But in general, if it didn't sound right, we'd play it again. Only rarely did we do tape loops and edits from different parts of the song and move them around. This is an attitude I still like to see in the studio: Always try to keep it natural or, as I call it, "organic."

This is not to say that technology isn't important. Frankly, I love music technology and would be the first to hold up a banner for many of the new recording and sound techniques. But I think what really makes music is the artists' ability to mix the technology with live recording and human emotions. That's essentially how I see my job as a producer.

I have lots of friends who are good producers but who are only techno-producers. They make records — great records — but they can't play an instrument. It works for them, but it just isn't me. Production is an extension of my musicianship. The studio is yet another instrument that I can play.

So the question is, if you're a musician first and foremost, how do you combine the art with the science in the control room?

The answer is different for everybody, but personally, I think my fundamental job as a producer is keeping an eye on the arrangement. In this sense, being an artist and an arranger is the foundation of my playing and of my musicality. In fact, the first thing I send for when I walk into a rehearsal is manuscript paper. I keep track of the arrangement so that when I talk with the band about changing parts, I can speak musician to musician. The role of the producer at this point is to let the performers do their thing; you redirect them and make suggestions only if their music starts veering off in some strange direction.

Of course, you've got to vary your approach. For example, I was just working with the Stray Cats; everybody in their band reads music, so I could talk to them in that language. But recently when I was working with the Vaughn brothers, I

EAR CANDY

Nile Rodgers' personal production system is about to expand from the Tascam 688 Midistudio he currently carries around on the road. In fact, he's scheduled to open a New York studio to accommodate productions for his new record label — Ear Candy —"what arrangers call strings and horns and background singers," he explains.

"Ear Candy will be a complete high-tech environment, but it will have a completely organic feeling," Nile says.

While the Ear Candy team is still scouting for a Manhattan location, Nile reports that the studio will feature both Solid State Logic and Neve consoles with the full gamut of outboard gear and digital (reel-to-reel and DAT) recorders.

"Every room will have complete MIDI facilities," he says. Currently, everyone in his New York offices has at least one Macintosh workstation; they predominantly use Alchemy and Performer software. He reports that a full battery of hard- and software will be on line at Ear Candy.

Right now, he is concentrating on finding the perfect location. "Personally, I don't like to travel 80 miles outside the city and stay there to get my work done. Certain musicians love that — the communal feeling. I do too, but I only need it when I close the door behind me — I don't need it when I wake up or when I'm not doing a record. The magic is the fact that we're making music, not that we're making it in a commune in the middle of Nova Scotia. I say let's be communal, but let's do it near some really good clubs and restaurants."

—J.T. Way

had to improvise. They knew chords, but put some sheet music in front of them and they laugh at you. It was a challenge, since I was doing the arranging. I ended up making flashcards and having my assistant hold them up so they kept to the score. You've got to be versatile like that.

Of course there's the equipment to consider. You don't want to blow it when the band gets the spirit and plays the tune perfectly. For the last few years, I have been a fan of getting every sound right in the room coming from the instrument and from the players' hands, so almost the entire record is completely flat with very little compression and EQ. The only time we adjust that compression is if the ambient level of the room in the mic is coming back and effecting the sound.

THEORY & PRACTICE

The best way for a producer to get a real, emotional sound is often to take a hands-off approach. I want every instrument to be captured the way that it is. Sure, I could change every track, sample every note, and change everything — but when you're making an organic record, that defeats the whole concept.

These days this may not be a popular opinion. But I'm an arranger, musician, songwriter and a producer. I'm a programmer too, but that's the least of my credentials and I only program to aid our music. Most of the time, you really don't have to program anything. You just play it. In that regard, it's a totally organic record. You may not think it is difficult to produce this way but, to the contrary, it is an art form to get an organic record to sound as good and as consistent as a completely programmed techno record.

You have to remember that nowadays, when a kid hears a bass drum, they're all the same audible level. Somebody's programmed the dynamic levels so it's all perfect and it always does what it's supposed to do. I know — as every producer has to know — that I have a responsibility to take that pressure off the performers and just let them play and make sure we are recording it in such a way that it fits the taste of contemporary listeners. The trick is to keep it true to what they do and true to what I like.

The fact is that music technology is there to serve musicians, composers and producers. It isn't the other way around — like in some science fiction movie where the computers take over the planet. Of course, human beings aren't perfect. But we don't want perfection. We want style. A lot of people that we think are truly beautiful aren't perfect — it's the ones that have something weird or quirky about them that you really notice and find so appealing.

It's the same with music and technology. I hate it when I hear the same drum sound on every record. I say to myself, "Damn it, I know that drum machine. Why don't they do something to change it around a little bit?" At the same time, I love it when someone takes that same old drum sound and builds a new, totally clever record around it. You see, you can approach novelty and innovation from two sides — that's something every producer should have in the back of his mind.

SENSE & SENSIBILITY

I'll let Bob Clearmountain talk about production from the engineering side — because there's no one in the world who's better. I take it from the musician's angle.

Your first job as a producer is to make the song make sense. I'm a songwriter, so if I have an idea in the studio, I voice it and see if the band can hear what I'm saying. The only way to see if it's right is to try it. Then we try to get to a place artistically where I'm happy, they're happy and it all makes sense. It's not that hard of a job when you're working with good people. It really only becomes difficult if people don't know who they are, what they are and what they want. Or if they don't have enough faith in you to understand that you know them and care how they sound.

I always look at a new project as if I am joining the band. We should be able to have the same free exchange of ideas and arguments you'd have as a bandmember. That helps me, because I know I am on the artist's side first — and I know that artists don't have that many allies in this world. Once we deliver the record, chances are someone at the record company is going to have complaints. But you know and I know that this happens every time you deliver a record. If you work as one with the band, you can always call up and say, "We made a great record, didn't we? It was a great and honest experience, right?"

To make this type of musical connection you can never walk into a session in a vacuum. You have to know the artist you are going to work with — what the band's sound is, what kind of message they're trying to get across. You should be as prepared as they are, even though you're not going to walk out on stage and play.

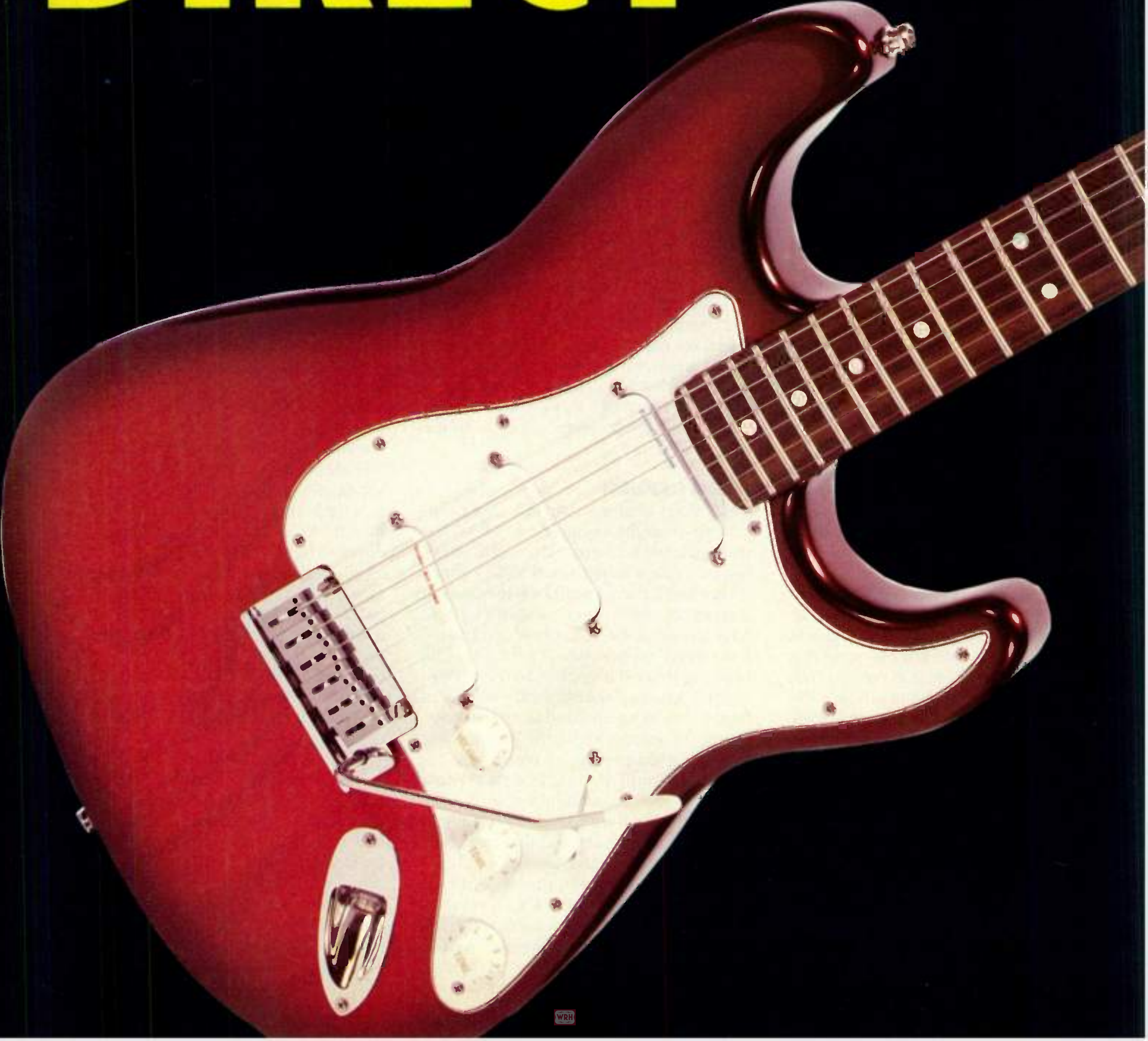
Then there's the business side of producing to consider. As a producer, there are a lot of things you have to take care of. You are responsible for the product, which in turn means that you are responsible for the budget, for the players, for where the project goes and all sorts of things you rarely have to consider as a musician.

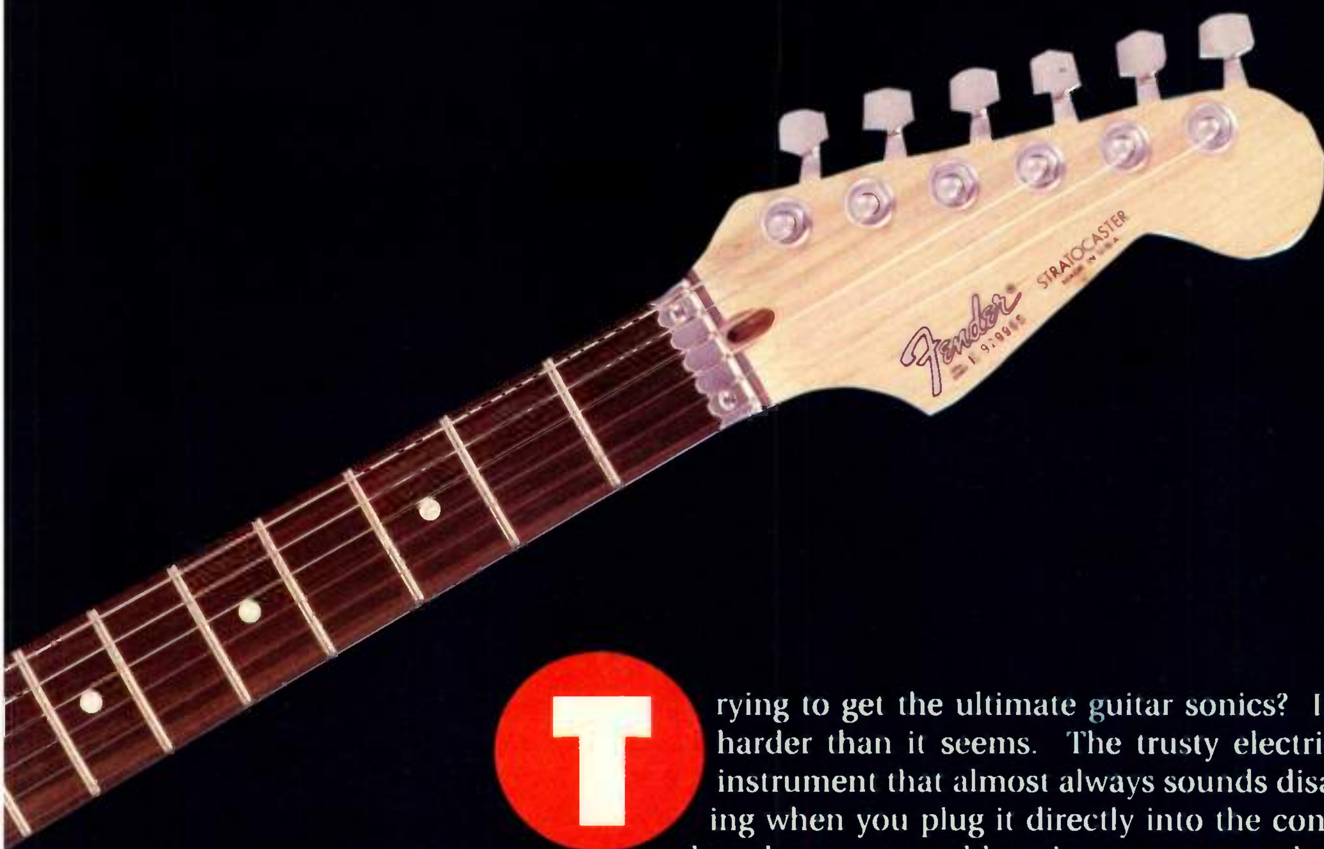
I hate that side of the business. You have to call somebody's manager if you want to rent the simplest piece of gear. You might hate it too, but believe me, it's got to be done — right. I'm from the old fashion hippie school, where the record company buys you a farm and you move in with your band. Of course, that's not how it is. And I wasn't brought up like that either; Bernard and I had to hock our equipment to make our first record. I used to work at the Apollo Theatre and one of the highlights was that there was food backstage. You had to buy it, but it was food.

If I have any last words, let them be this — you've got to love what you do. Never stop being a musician and never let that music out of your heart. Use technology effectively, but don't let it overshadow the performers' inspiration and spontaneity. And don't blame yourself if the band just isn't together; you'll always find you produce the best records with the best artists. Enough advice? One more piece: always have fun — and think *organic.*

Photos of Nile by Renaldo

GOING DIRECT





Three new direct boxes give your guitar that authentic sound when plugged into the mixer or P.A.

RANDY CHANCE

Trying to get the ultimate guitar sonics? It can be harder than it seems. The trusty electric is one instrument that almost always sounds disappointing when you plug it directly into the console. Although programmable guitar preamps and outboard effects offer more options than Fender Tweed amps and Ampeg B-15s, they often still don't sound good enough.

It seems no high-tech solution can consistently offer the complex tonal properties and envelope characteristics that make guitarists and engineers reach for those old tube amps.

True, the recent crop of programmable, MIDI-controllable, rackmounted guitar preamps (many with tubes) provide us with some wonderful new sounds, and attempt to recreate some old ones. And, of course, many engineers enjoy the sound they can get from a tube amp's preamp output. However, it is the amp tubes, not the preamp tubes, that provide the crunch lacking in solid-state designs. It seems unless you're actually over-driving your power-amp tubes, your rock 'n' roll sound is never quite authentic.

I recently examined three devices that try to solve this problem. These "speaker simulator" boxes are designed to receive a speaker-level signal from an amp and reduce it to a line-level signal that's suitable for a mixing console. They also transform the amp's source impedance (4,8, or 16 ohms) to an appropriate line impedance (600 ohms or higher). This lets you run an amp head directly into a mixing or P.A. board, ignoring the preamp output. The guitarist can use a cherished pre-CBS Fender Twin or early-model Marshall head, run its speaker out into one of these nifty boxes, and feed the signal into the latest stereo digital effects processor to achieve that "ultimate" sound.

DIRECT BASICS

If you find yourself dependent on a certain amp-microphone combination and have to jump through hoops at every studio to get it set up — or if you're having trouble with "wooly" sounding miked guitar tracks in a mainly synth- and sampler-based mix, Direct Injection (DI) may be the way to go. How do you do it?

☐ Feed the guitar amp output (usually post-power amp) into a device that electronically emulates how a speaker and cabinet alter the sound. The processed sound then feeds the mixer console.

☐ Insert a signal processor that can create an "amp" sound between the raw guitar output and mixer input. Some of these devices are actually miniature guitar amps (without output transformers and speaker box emulators) using traditional analog design techniques (e.g., tubes or solid-state); others are digitally-based multi-effects units whose repertoire of effects includes speaker emulation.

Most DI devices designed specifically for guitar process the sound to one degree or another. If you want an unprocessed sound, note that plugging your guitar into a mixer won't usually give satisfactory results because the console input impedance will be too low. In addition, balanced console inputs need a suitable adapter to handle the guitar's unbalanced output. The same problems can also apply to feeding an older effects output, not just a guitar, directly into a console. Check the input impedance since transformer-based models generally load down a guitar more than active direct boxes, whose impedance can be as much as several megaohms.

—Craig Anderton

Does this setup sound that much better than a good programmable distortion box or tube preamp? Is it worth dragging around a tube power amp just to use as a preamp? Can you really give your instrument an authentic feel by running direct into the mixing board, with no Celestions or Marshall bottoms to provide that vital "thwonk" of many contemporary guitar sounds?

To find out, I investigated three units — the Alan Holdsworth Juice Extractor, the Groove Tube Speaker Emulator, and the Red Box from Hughes & Kettner. I set up a Marshall 800 JCM Lead series amp head on the BOOST channel, a Metaltronics all tube 100-Watt head, and a Music Man 75-Watt head. Then we created an appealing sound through the speakers, and ran the signal through each speaker simulator to see how it stacked up. The "benchmark" axes were a Jackson Charvel 475 guitar with Jackson pickups, a new Strat with LACE Sensor pickups, a Yamaha RGX-612S with Tom Anderson pickups, and a custom Strat with EMG pickups.

GETTING JUICED

At $529 (the most expensive of the three), the Alan Holdsworth Juice Extractor offers a wide range of sounds. This 19-inch rackmount unit has a flat curve and an infinite number of EQ-based curves. You can run it flat, which lets the amp sound straight through with no added EQ, or you can create your own sound with the help of a three-band sweep EQ, which has controls for 26 to 640 Hz, 2.5 to 5 kHz, and 8 to 20 kHz. All three bands have separate -15 to +15dB boost/cut controls. The one input and six output connectors are 1/4-inch phone jacks, which makes it easy to run stereo feeds to separate mix and effects processors through a recording board and P.A. board simultaneously.

The owner's manual says that if you create your own curve from the flat curve, you should start with the high frequency of the parametric cutback, and then add as desired. The best results with a hard rock sound (on all the heads) were achieved by greatly boosting the extreme lows, with the frequency set to the two o'clock position (the controls aren't calibrated), midrange set at 12 o'clock with a boost of about 6 dB, and highs rolled off slightly with the bandwidth set at about 12 o'clock. I didn't need as much extreme low boost with the older head, and had to roll down the highs more when using the active EMG pickups. Turning down the Marshall gain to about nine o'clock produced a good, clean sound, especially from the new Strat. I was able to get old amp reverb sounds at line level, which is hard to do with modern gear.

The Juice Extractor also has a downward expander circuit that uses a voltage-controlled amplifier to provide up to 50 dB of gain reduction. At input levels above an adjustable threshold, output levels are unaffected; below the set threshold, the device decreases output level exponentially in relation to input level. As the signal drops 20 dB below the threshold, for example, the output stage decreases 30 dB. If the incoming signal drops to zero (no notes played), the Expander causes the amp noise to drop below audibility. In practice, this worked very well, although (like everything else on the unit), it took some experimentation to get a satisfactory sound. But we buy things with lots of knobs so we have control over the parameters, right?

It was funny to hear the same pickup squeal at a high threshold that often occurs when you run a guitar through speakers at high volumes, but — with the right amount of tweaking — it exhibited excellent signal-to-noise. I had to work with it a while before it sounded better than a normal direct box or bad mic placement. However, if it were a permanent part of my studio rack, I soon would become a very fast and proficient operator.

These units do eliminate the problems of inconsistent mic quality, fees for speaker-cabinet cartage to sessions, and all the headaches of blown speakers and rattling cabinets. The serious engineer will want to try all three.

GROOVE TUBE

The Groove Tube Speaker Emulator ($450) also worked well and it is considerably simpler to use than the Juice Extractor. This 19 inch rackmount unit has a 4/8/16 ohm impedance-selector switch, one input, one output, I/O level controls, and three bands of EQ fixed at 220 Hz, 1.7 kHz, and 3.5 kHz. It works as a speaker load even when turned off. When put to use, the input and output controls somewhat affected overall tonal response. The manual recommends a low input level (below 12 o'clock) to keep the resonance filter from overloading, and I found this useful. The manual also recommends EQ settings to obtain the sound of a U73 Marshall 100-Watt head and a U62 white Bassman top: After fiddling awhile, I got the idea — but if you want those sounds, the best idea is to use those heads in the first place. I produced excellent sounds with the Emulator, with as wide a range as I was able to produce with the Juice Extractor.

Left: Red Box from Hughes & Kettner; upper right: Groove Tube Speaker Emulator.

(NOT SO) BIG RED

The Red Box ($99) is the simplest device of the three. It doesn't even have a manual; instructions are printed right on the unit, in German (which doesn't help if you can't read the language). The Red Box is about the size of a guitar footpedal, and is powered by a 9 volt battery. (It also includes a connector for a power supply, which must be purchased separately.) This thing has no controls. You simply plug your amp head into one phone jack and your speaker bottom into the other, then run the line out to your console. The XLR line-out lets you bring up the signal as a low-impedance mic level. I found it satisfactory for general guitar sound, but all the equalization has to be accomplished through the board or outboard gear. The Red Box also has a line input, but its only use, as far as I can tell, is to run from a preamp output into a low-impedance snake on stage. Please note: the Red Box has no dummy load, so you can use it only with the speaker cabinet plugged in.

These three boxes all apply the theory of negative feedback between amp and speakers, with good results. Ultimately, comparing them is a subjective experience. I'm not sure I ever achieved a real cabinet thump, but it's also difficult to get one on a tape from microphone techniques, after the usual array of effects have been added. These units do eliminate inconsistent mic quality, fees for speaker cartage to sessions, and blown speakers and rattling cabinets.

With enough tweaking, these boxes, particularly the Juice Extractor and Speaker Emulator, provided more satisfactory sounds than I generally get from line-level preamps or distortion units. The serious engineer of guitar sounds will want to try all three — over and over again.

Manufacturers:

Alan Holdsworth Juice Extractor, Rocktron, 2870 Technology Drive, Rochester Hills, MI 48309, (313) 853-3055; Groove Tube Speaker Emulator, Groove Tube, 2866 Foothill Blvd., Sylmar, CA 91342, (818) 361-4500; The Red Box, Hughes & Kettner, Inc., 35 Summit Ave., Chadds Ford, PA 19317, (215) 558-0345.

Chance *is a programmer, composer, and session guitarist, whose Los Angeles-based production company specializes in making alternative music.*

NEW AT AES

The next six pages contain many of the important new products you saw (or may have missed) in Los Angeles.

There's nothing quite as exciting – and exhausting – as an AES convention. Besides the fact that it's a powwow where ideas are exchanged and friendships are made, it's also a forum for the latest and greatest in creative sound technology. So if you were too busy wheeling and dealing to see the whole show floor, or if you were at the home front mixing, relax. Our editors have compiled a sampling of the gear that stole the limelight in L.A. Be sure to use our exclusive reader service and manufacturer fax numbers for more information.

INFO#	FAX#	BRAND	MODEL
100	(201) 440-2865	Agfa	SR-XS —
101	(817) 870-1271	Akai	S1100 DD1000
102	(415) 351-0500	AKG	K270HC C406 C407
103	(213) 836-9192	Alesis	RA-100 3630
104	(804) 358-9496	Alpha Audio	DR-2
105	(415) 367-4132	Ampex	617 C-60 618 C-90
106	(002) 823-9542	AMS	AudioFile + Logic 1 Soundfield
107	(617) 329-1241	Analog Devices	AD1862
108	(213) 399-7665	Apogee	D160
109	(201) 249-2123	Ariel	DAT-56
110	(716) 436-3942	A.R.T.	Power Plant MDC 2001
111	(615) 689-7615	Audio Animation	Paragon
112	(516) 935-8018	beyerdynamic	MC 742 TG-X 480
113	(213) 676-6713	BGW	BGW 200
114	(213) 329-9895	Brainstorm	TDC-24
115	(508) 485-0619	Bruel & Kjaer	—
116	(415) 351-0500	BSS Audio	DPR-901 TCS-804 TCS-803
117	(206) 778-9453	Carver	PT-2400
118	(301) 694-5152	Cipher Digital	CDI-328
119	(215) 874-0190	Community	RS220
120	(201) 423-2977	Crest Audio	NexSys
121	(219) 294-8329	Crown	Macro Reference
122	(037) 274-3532	DAR	DASS 100 SoundStation/DSP
123	(415) 351-0500	dbx	160XTDO
124	(516) 420-1863	DDA	DCM224V
125	(415) 327-0777	Digidesign	SampleCell
126	(612) 544-5573	Digital Audio	The CardD
127	(313) 471-2611	DLC Design	SPEAK
128	(415) 863-1373	Dolby Labs	Model 422
129	(508) 234-8251	EAW	KF1000 SM300
130	(213) 802-1964	Fostex	820
131	(212) 265-8459	Gotham Audio	Penguin
132	(916) 273-6948	Hedco	HD-16X16AS
133	(213) 841-0346	Hybrid Arts	ADAP II

** Note: All information compiled from manufacturer provided literature, and is subject to change without notice.*

PRODUCT TYPE	APPLICATION	NOTES
audio cassette	recording	C-60, C-90 configurations
mastering splicing tape	recording	white, polyester tape for all audio production
digital sampler	recording	RAM expands from 2 to 32 Mb
optical disc recorder	recording	utilizes re-writeable optical disk
headset	recording/monitoring	auto on/off switching headphone, electret condenser mic
instrument microphone	recording	mini lectret gooseneck
lavalier microphone	recording	omnidirectional
power amplifier	monitoring	100w into 4 ohms, independent channel volume controls
compressor	recording/monitoring	switchable RMS/Peak, Hard/Soft Knee compression styles
edit controller	recording/editing	control interface for hard disk recorder
audio cassette tape	duplication	interlocking, symmetrical hubs
audio cassette tape	duplication	interlocking, symmetrical hubs
hard disc recorder/editor	recording	8 simultaneous inputs, read/write optical drives
digital mixing console	recording	for use in digital post production environment
stereo microphone	recording	produces B format signals for UHJ surround systems
D/A convertor	recording	20-bit, 119-dB SNR
dithering module	recording	proprietary module
DSP development system	recording	for IBM-PC, AES/EBU SPDIF digital audio I/O
guitar preamp	sound reinforcement	remote channel switching, effects loop
master dynamics controller	recording	21 LED gain control metering
broadcast signal processor	processing	4-band limiter, 4-band A/B comparison
stereo condenser mic	recording	2 vertical double diaphragm capsules
wireless microphone	recording/sound reinforcement	hybrid super/hyper cardioid polar pattern
power amplifier	sound reinforcement	100 w/ch or 200 w/ch mono
tach/dir converter	recording	for use with Sony APR-24 onboard chase synchroniser
R-DAT archiving system	recording	developed in cooperation with Panasonic Pro Audio
dynamic equalizer	recording/sound reinforcement	4 parametric bands of frequency selective compression
stereo time corrector	sound reinforcement	designed for speaker system time correction
time corrector	sound reinforcement	mono, 1-in 3-out, multitap digital delay line
amplifier	sound reinforcement	1156 w/ch-8 ohms
random access recorder	recording/editing	available in 2 or 4 track versions
loudspeaker	sound reinforcement	200w pink noise/500w program, 127 db output at 1m
computer control system	recording/monitoring	remote monitoring and system control
amplifier	sound reinforcement	750 w/ch-8 ohms
synchronizing system	recording/monitoring	multiple device interface and signal processor
digital audio workstation	recording/editing	16 channels, Segment Based Processing
compressor/limiter	recording	separate active-balanced, single-ended outputs
console	post production	4 stereo sub-groups, 24 routing buses
playback card	recording	stereo, 16-bit, 16 voice, 8 output for Mac II series
buss card	recording	for IBM AT, 16-bit digital recording to disk
software	monitoring	loudspeaker simulation program
encoder/decoder	processing	4-channels of Dolby B, C, S noise reduction
loudspeaker	sound reinforcement	two 12-inch speakers, 2-inch high frequency driver
stage monitor	sound reinforcement	3-way, horn loaded design
mixer	recording	20-input, 8-buss output
graphics interface	recording	digital audio mixing, editing
router	recording	available in stereo, dual, mono configurations
portable recorder/editor	recording	for use with 4Mb laptop computer, turnkey system

NEW AT AES	INFO#	FAX#	BRAND	MODEL
	134	(608) 273-5483	**Intellix**	Studio Psychologist
	135	(818) 893-3639	**JBL**	SR4700 Series
	136	(818) 763-4574	**Jensen**	69JT
	137	(213) 822-2252	**JL Cooper**	CS-1
	138	(708) 864-8076	**John Hardy Co.**	MPC-1R TS-1R
	139	(201) 523-2077	**JVC**	DS-DT900U
	140	(201) 579-6021	**JRF Magnetic**	P.A.W.
	141	(213) 604-4487	**Kenwood**	DA-7000/DD-7200
	142	(516) 420-1863	**Klark-Teknik**	Midas XL3 Midas XL88
	143	(501) 777-6753	**Klipsch**	KP-320 KLiP Circuit
	144	(516) 333-9108	**Korg**	DAW
	145	(617) 891-0340	**Lexicon**	300 LXP-15
	146	(818) 718-2886	**Martin**	CT Series
	147	(818) 781-3828	**Massenburg Labs**	8900
	148	(415) 486-8356	**Meyer Sound**	VX-1 DS-2
	149	(503) 293-0562	**MicroAudio**	DCM 6200
	150	(516) 420-1863	**Milab**	LSR2000
	151	(062) 866-7002	**Minim**	E250
	152	(415) 871-6555	**Monster Cable**	LightSpeed 12
	153	(619) 431-8077	**Morenz**	SA-1000
	154	(512) 891-2947	**Motorola**	96002
	155	(212) 302-1627	**Nagra**	Nagra D
	156	(603) 498-3684	**NED**	SoundDroid MultiArc
	157	(312) 975-1700	**Neotek**	Encore
	158	(203) 792-7863	**Neve**	DTC-2 VRP
	159	(916) 265-1010	**NVision**	NV2000
	160	(213) 946-6030	**ODC**	617 CX
	161	(415) 341-7200	**Otari**	MK-IV series DDR-10 DTR-900-II Series 54/Film
	162	(205) 985-9966	**Oxmoor**	RMX-44/62
	163	(714) 373-7242	**Panasonic/RAMSA**	SV-255 SV-3700 SV-3900
	164	(601) 484-4278	**Peavey**	Chase-Lock 3680 Ultraverb II PMA 250

Community VBS210, RS220 & 220 Controller

JBL SR4700 Series

Milab LSR2000

Note: All information compiled from manufacturer provided literature, and is subject to change without notice.

PRODUCT TYPE	APPLICATION	NOTES
monitor mixer	monitoring	remote controlled
loudspeakers	monitoring	titanium diaphragm compression drivers
electronics package	recording	replacement for Magna-Tech 69C reproduce electronics
control station	recording	hardware control interface for disk systems
mic preamp card	recording/sound reinforcement	vertical format card-cage version
transformer splitter card	recording/sound reinforcement	3 transformer isolated outputs
DAT recorder	recording	synchronizes with TV signals, uses SMPTE time code
alignment tapes	testing	1/4-inch time code and pilot tone test tapes
CD write-once system	recording	consists of encoder, writer, software
console	sound reinforcement	8 mute groups, 8 VCA masters, 18 discrete sends
matrix mixer	recording	standalone, 8-channels, contained in 4U rack mount
loudspeaker	sound reinforcement	15-inch, 2-way, 12-inch passive radiator
limiter protection circuit	processing	available in KP, KSM series speakers
digital audio production system	recording	8-track recorder, 16-track MIDI recorder/sequencer
digital effects system	MIDI	50 event program change list
multi-effects processor	MIDI	remote control of 27 variable parameters
loudspeakers	sound reinforcement	disco, theater, A/V applications
limiter/compressor	recording	2-channel, variable crest, selectable soft/hard knee
stereo program equalizer	sound reinforcement	5 controls per channel, overall stereo gain control
mid-bass loudspeaker	monitoring	frequency range: 50-200 Hz
mixer	recording	programmable from PC, Mac, front panel
condenser microphone	sound reinforcement	hand held type for live applications
ambisonic encoder	recording	converts B-format signals into UHJ 2 information
mic distribution system	recording/sound reinforcement	digital multiplex fiber optic cabling
amplifier	recording	switching Class D power amp
32-bit processor	processing	for multi-chip implementation
digital audio recorder	recording	4-track, 1 analog track, time code
audio editing system	recording/editing	handles dialogue, effects, Foley applications
processing system-DSP option	recording/editing	multitasking, multiuser
console	post production	1-4 positions, LCRF or 6-track format/monitoring
digital transfer console	recording	A/B store system, optional dither
console	post production	recall, flying faders, Dolby DS4 matrix monitoring
high definition audio system	recording	supports interconnections for ATV, D1, D2, type-C VTRs
encoder/decoder	processing	encoding/decoding 2 channels, stand-alone rack-mount
tape machine	recording	2, 4, 8-track configurations
disk-based editor	recording/editing	2-track
digital audio tape recorder	recording	8x oversampled D/A convertors, 32-channels
console	recording	4-channel panning, monitoring
amplifiers/mixing matrix	recording	1U, rack mount chassis
portable DAT recorder	recording	true gain control
DAT recorder	recording	jog/shuttle dial, AES/EBU I/O
DAT recorder	recording	RS-422 control port
software	MIDI	updated for SyncController synchroniser
console	sound reinforcement	100mm faders, 36 channels, 8 aux sends, 8 sub masters
multi-effects processor	recording/MIDI	256 internal programs, 128 effects slots
amplifier	sound reinforcement	125w RMS per channel at 4 ohms

NEW AT AES

INFO#	FAX#	BRAND	MODEL
165	(213) 530-2384	**Professional Audio**	TOC Stage Wedge 2
166	(714) 645-0401	**QSC Audio**	EX 4000
167	(206) 347-7757	**Rane**	FPL 44
			FLM 82
168	(714) 250-1035	**Renkus-Heinz**	CM Series
169	(604) 734-3901	**Richmond Sound**	816-S
170	(213) 722-0911	**Roland**	DM-80
			MV-30
			SBX-1000
171	(516) 932-3815	**Samson**	True Diversity UR-4
172	(212) 924-1243	**Schoeps**	VMS-021B
173	(203) 434-1759	**Sennheiser**	WM1
174	(212) 315-0251	**Solid State Logic**	SoundNet
175	(415) 394-8099	**Sonic Solutions**	NoNoise II
176	(201) 833-2880	**Sony**	BVG-200
			PCM-7010
			PCM-7030
			PCM-7050
			RM-D7300
177	(416) 886-6800	**Sound Ideas**	LucasFilm Library
178	(818) 893-3639	**Soundcraft**	Venue
			Delta 8
179	(613) 696-4864	**Soundtracker**	—
180	(081) 399-6821	**Soundtracs**	PC 32
181	(415) 457-6250	**Spatial Sound**	SSP-200
182	(206) 487-3431	**Spectral Synthesis**	AudioVision
183	(916) 635-1787	**Stewart**	PA-500
184	(615) 256-7619	**Studer Revox**	Dyaxis 2+2
			D820-48
185	(206) 283-5504	**Symetrix**	DPR44
186	(519) 745-2364	**Tannoy**	PS-88
187	(213) 727-7635	**Tascam**	M-3700
			M-2500
			488 PortaStudio
			424 PortaStudio
			Porta 03
188	(612) 736-1246	**3M**	996
			AUD 80+
189	(415) 351-0500	**TurboSound**	TMS-5
190	(415) 657-4225	**UltraAnalog**	ADC 20048
191	(818) 444-1342	**Vega**	Pro Plus
192	(303) 447-2351	**WaveFrame**	CyberFrame
193	(714) 739-2680	**Yamaha**	DMC-1000
			DMR8
			AD8X

Panasonic SV-3700 DAT

3M 996

Soundcraft Venue

Tascam M3700

Yamaha DMC-1000

** Note: All information compiled from manufacturer provided literature, and is subject to change without notice.*

PRODUCT TYPE	APPLICATION	NOTES
loudspeaker	recording/monitoring	15-inch coaxial speaker
power amplifier	sound reinforcement	input connectors mounted on removable module
limiter	recording	quad program, Auto Slave channel switching
line mixer	recording	8 mono, 4 stereo line inputs
loudspeakers	monitoring	multi-angled cabinet design
special effects mixer	sound reinforcement	show controller for live performance applications
hard disk recorder	recording	4 recording tracks expands to 8, 100Mb memory
music production system	MIDI	built-in digital effects, tape sync
controller	MIDI	cueing system
rack mount receiver	recording	signal level of +10dbm without clipping
stereo microphone	recording	user-selectable recordings, X-Y or M-S recording
wireless mic mixer	recording	5 inputs, incorporates RF receivers as inputs
digital audio network	recording	works with 7 ScreenSound audio post systems
digital signal processor	processing	enhanced broadband removal program
timecode reader/gen.	recording	portable DAT component
DAT recorder	recording	timecode-equipped
DAT recorder	recording	timecode-equipped
DAT recorder	recording	timecode-equipped
DAT edit controller	recording/editing	timecode-equipped
sound effects library	recording	6-CD collection
console	sound reinforcement	8 buss, 40 inputs, 4 stereo effect returns
console	sound reinforcement	8 buss, 36 inputs, 16 monitor returns, 4-band EQ
digital sound editing system	film sound editing	multi-user system with independent workstations
mixing console	recording/MIDI	32 I/O channels, 72 line inputs with MIDI control
processor	MIDI	19-inch joystick remote, 2 rack spaces
graphic soundfile editor	recording	application for SynthEngine Sampler
amplifier	sound reinforcement	two rack space dual mono, 250w/ch into 8 ohms
digital recorder/editor	hard disk recording	2-, 4-channel simultaneous play, multitake record
digital recorder	48-track recording	4x oversampling D/A convertors, passive filters
recording/editing station	recording/editing	4-track random access
subwoofer	recording/sound reinforcement	100w self-powered for near field monitors
console series	recording	24- or 32-channel configurations
console series	recording	16- or 24-channel configurations
portable recorder	recording	8 mono/2 stereo inputs, 8-track
portable recorder	recording	4-track, 3 tape speeds
portable recorder	recording	4-track, 2 channel inputs, 1 7/8 ips, mini studio
analog mastering tape	recording	79.5 SNR
digital U-matic cassettes	recording	83.5 mins. recording time
loudspeaker	sound reinforcement	full range, 3-way enclosure
A/D converter	recording	20-bit converter; 128-times oversampling
UHF wireless mic system	recording	includes R-662 receiver, T-667 transmitter
recorder/editor	recording/editing	multitrack 8 channel disk, sound looping
console	recording/mixing	digital, automated
digital audio workstation	recording/mixer	multi-track mixer/recorder
A/D convertor	recording	8-channel, 48, 44.1kHz sample rates

Miking Guitar Amps

Getting the right string sound without going direct injection requires some basic miking know-how.

TOM LUBIN

STACKS OF MARSHALL amps may be macho, but a lot of times, you can get those great-sounding, loud, distorted guitar tracks better with small amps and speakers and a few well-placed microphones. Too many times, I've seen engineers start compensating for a bad guitar sound with EQ, when the problem was simply that the mics weren't in the right spot — or weren't the right mics to begin with.

Where you place a mic can make or break your guitar tracks. First, the standard rock'n'roll procedure is to position a "close mic" no further than about 18 inches away from the speaker. Depending on how much of the room's acoustic reflections you want to capture, a second "room mic" can sit six feet or further from the sound source, on a stand several feet off the floor.

Picking the positions for the close and room mic isn't a matter of formula nor an act of magic. You simply go out into the studio, stand, kneel and otherwise contort your body until you find the spot that sounds best. Then you mike it.

So much for the basics — of course it gets more complex than this. The search for the ultimate guitar sound is a fine art that most of us keep honing and perfecting as we pack away experience and develop our "trademark" sounds.

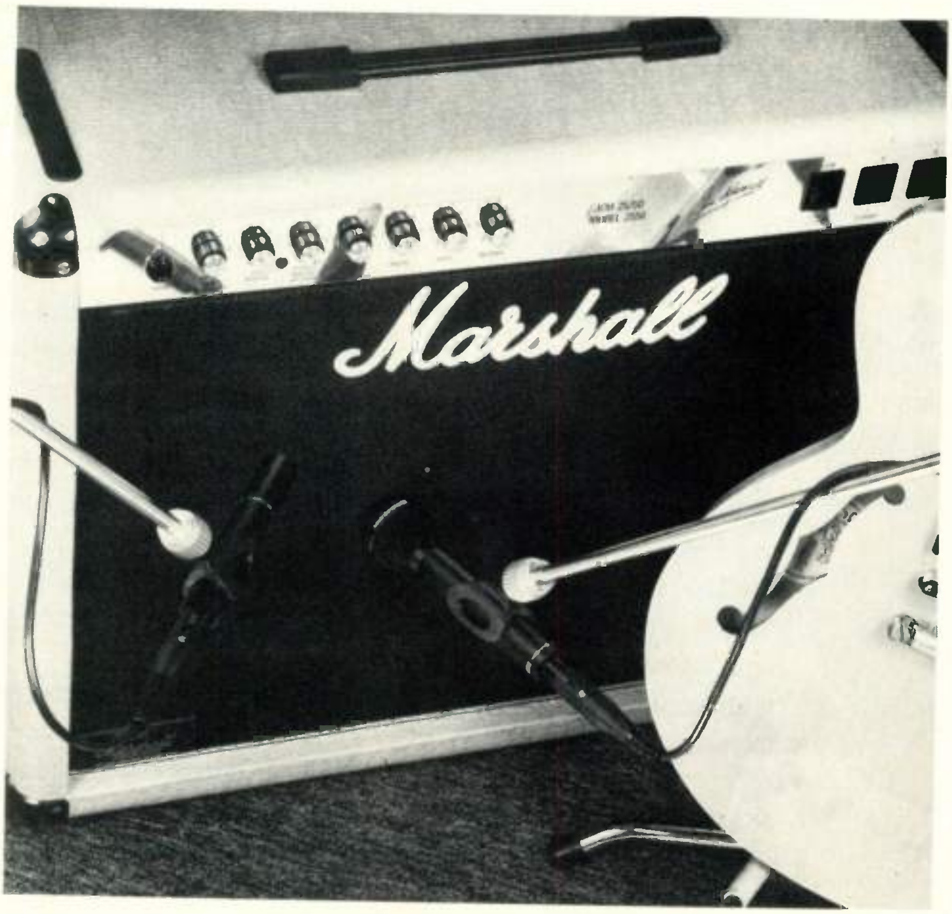
'PHONE CALLS

Just ask any engineer what his favorite microphones are. You'll hear a thousand different answers and a score of different formulas — although in dynamics, the names Shure SM-57 and SM-58 keep popping up, as do the Neumann U87 and AKG's 414 in the condenser ranks.

I tend to use condenser microphones for their brilliance, but many people like the chunkier quality of a dynamic. They also tend to be more reliable on the road (not to say that you can continually abuse them with floorshaking volume without eventually paying a price).

Don't be afraid to experiment; try placing the mic at different distances to create a more in-your-face or ambient sound. If the mic is right up against the speaker, low-end boost will result (this is known as the proximity effect).

For lead guitar, I usually place a stereo pair of mics a few inches off the floor, one to three feet in front of the guitar amp, about three feet apart. I often put a reflective surface (Masonite or thin plywood) on the floor in front of the amp, under the mics. I elevate the amp about a foot off the floor on a large rubber block, so the mics are in direct line with the speakers, and the amp is physically decoupled from the recording studio.



Beyer M-201 (left) and M-88TG miking a Marshall 25/50 Jubilee Combo amp.

It's never a bad idea to indulge some of your crazy ideas, either, and great success has been had by miking an amp in a stairwell or a hallway to catch some dramatic acoustic reflections for a full, big sound.

AMPS & SPEAKERS

Choosing and placing the mics is ultimately the arbiter of your final sound quality — but the amp and speaker can make an enormous difference too.

In my opinion, smaller amps simply offer more control over the sound. Moreover, the speaker distortion of smaller speakers has warmth, brilliance and presence. Large cabinets, on the other hand, tend to have a harsher sound.

Many guitarists use the amp without the speakers, by feeding some sort of speaker-substitution box to the speaker output, simulating the speaker's impedance load. Many amps now include preamp-outs, so the preamp's distortion can be used without the other amp circuitry.

Another viable alternative is to plug directly into the console via a direct box and an effects processor like the Rockman, ADA MB-1, or Boss GL-100 (see the Guitar DI

A few important words about the new A-T 40 Series:

Tony Bongiovi
Power Station

"The 4051 is a great mike, especially for rock. It sounds fat and you can bang away at it with a lot of level without a pad…for a rock studio like the Power Station that's important. When you put it on horns it has a nice clean sound and it holds the dynamics well…it's just an excellent sounding mike."

David Cook
Dreamland Studios

"…real nice top end and a warm bottom end…very versatile. I didn't have to pile on a lot of EQ to capture the air in the studio…very present, very natural sounding mikes."

Milan Bogdon
Masterphonics

"The S/N ratio is superior to some of the other mikes we used. They're bright and clean so we don't have to push the EQ. Superb mike…great for vocals, overheads, snare, toms, electric and acoustic guitar...it seems to work well wherever we put it."

Jeff Baxter
Producer/Artist

"If I'm not getting what I want from another microphone…I've been putting up the 4051 and it nearly always does the job."

Mack Emerman
Criteria Studio

"The response is very flat…it holds the natural tonal qualities even at high sound pressure levels."

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feature in this issue). Other guitarists run these devices into a powered, 4-inch speaker that is miked.

Want feedback? Put the small speaker on a stool close by, so the speaker blasts right into your pickups. This results in big amp feedback and sustain without shaking the building apart. You can mike the little speaker or run direct out of the distortion box, just using the speaker for feedback. It's all a matter of taste, and what you're doing. Often I cut rhythm tracks direct through some distortion box, then use an amp for the leads.

Getting fancy — or even setting up a simple miking configuration — can cause phase cancellation, which you'll notice as combined signals lose their punch. Make sure all the sources sound good when combined in mono. If they don't, for a two mic setup, rearrange the mics. If you combine microphones with a direct signal, or take a direct signal from a few different places along the signal path (a preamp-out combined with a speaker-out, for instance), those different sources can be out of phase. In this case, you can either play around with different combinations of the board's phase-reverse switch(es) or reposition the mics to solve the problem.

MAKING TRACKS

Once it all sounds good together, how many tracks should you use? The number depends on the guitarist. Some want each output of each take recorded on a separate track — one for the guitar's direct output, two for the delay's output, one for the preamp's output, one for the speaker out, and one for each mic.

On distorted rhythm guitars, I usually mix the various signals together for a single, strong, monophonic track. Then I double the part on another track and pan the two tracks left and right. Other incidental rhythm parts would go on separate tracks and get positioned in the mix to fit with everything else.

Many players love amplified string noises because they make guitar parts sound more exciting. I've recorded separate tracks of finger squeaks, string noise, and whammy-bar groans, sometimes in stereo, sometimes even doubling them. These get mixed in with the guitar performance tracks.

A few final notes on the electricity: if the board uses balanced inputs, and you use direct boxes for the line-outs from amps and effects, you have to try various combinations of the ground-lift switches on the direct boxes. If you don't use direct boxes, you probably need to lift the grounds of the devices in use, leaving the console's earth ground the only one connected. All other devices should be connected to a 3-pin ground-lift plug. Don't break the ground pin off; it's the only thing that stands between you and electrical shock. And that can ruin any session.

Lubin *is an American teaching recording at the Australian Film, Television, and Radio School.*

Sound Studio Business

Save big tax bucks with a Subchapter S corporation.

STEVE HARLAN

I ONCE HEARD that a good example of wasted space is a busload of tax accountants going over a cliff — with three seats empty. Being a CPA, I tried to subdue my laughter. But really, folks, we card-carrying members of the briefcase brigade come up with some really cool tax-saving ideas every once in a while. Our idea for today? Say "yes" to Subchapter S.

April 15th is already approaching, promising to take with it more hard-earned studio and music royalty profits than you care to think about. Now, I know taxes might be a sore subject, but I promise not to throw salt in the wound. Instead, I've got a tax idea that just might save you some real dough.

But first, as always, I have to point out that you shouldn't take any action on what you read here without first consulting a qualified tax professional.

Now go and dig out your most recent income tax return. Flip through the forms until you come to Schedule SE Social Security Self-Employment Tax (SE Tax) and take a look at line 9. That's the tax I want to talk about and, specifically, how it can be reduced.

For those of you who don't have a Schedule SE, it's the way self-employed individuals pay Social Security. If you're a wage-earner, social security is taken from your pay as FICA, but you should still read this article if you're thinking about starting a business or might have self-employment income in the future.

The self-employment tax rate was 13.02 percent for 1989 as it was for 1988. For 1990, the rate went to 15.3 percent. That means for every $1,000 your studio or music business netted you in 1990, you'll end up turning $15 of it over to Social Security — to say nothing of income taxes.

A Subchapter S Corporation (S-Corp) can help. The best way to illustrate this is by example. Assume your music business had total sales of $100,000 and expenses such as rent, utilities, phone, supplies, depreciation, etc. of $60,000, netting you a cool $40,000 before taxes. As a self-employed artist you'd pay $6,120 SE tax on $40,000 of self-employment income.

But let's say you had incorporated your business and elected Subchapter S status. Using the same numbers, before you pay yourself a dime out of your corporation, the business has $40,000 ($100,000 sales less $60,000 expenses). Now, there's at least two ways to take money out of an S-Corp. First, you have to pay yourself a reasonable salary — say $24,000. After your S-Corp pays that salary out of the $40,000 net income, it still has about $13,600 left over ($40,000, $24,000 salary, $2,400 in FICA and payroll taxes). That $13,600 can be paid out as a "distribution" and isn't subject to the 15.3 percent Social Security tax.

You might be asking yourself "why not just pay out the entire $40,000 as a non-FICA taxable distribution?" The reason is that you must pay yourself a reasonable salary. Unfortunately, "reasonable" is not defined further than "what a reasonable salary would be given all the facts and circumstances surrounding your situation." In situations where there are similar businesses, you could find out what a similar position would pay in wages. Obviously, determining the minimum salary is something best done with the help of a tax professional.

One more thing to consider. The above strategy could possibly reduce your Social Security benefits down the road because benefits are more or less tied to what you've paid over the years. Accordingly, it would be prudent to contribute the FICA tax savings to an IRA or other qualified corporate retirement plan. The general consensus is that dollars put into tax deferred retirement plans will yield a greater retirement benefit than the same dollars put into Social Security. However, this may be more the case for younger than older individuals.

> Spend those saved taxes on whatever new gear you need most for your project studio system.

Of course, another way to look at it is that the Subchapter-S FICA savings strategy can make a part of your Social Security taxes optional. You can have the option of spending those saved taxes in whatever manner makes the most sense to you.

Setting up a corporation is a job for a CPA and an attorney. When your attorney asks what assets you want transferred into the corporation, consider retaining all your equipment and leasing or renting it to the corporation. Rental income may also be non-taxable for Social Security purposes.

In addition to the limited liability benefits of forming an S-Corporation, an important side benefit for individuals with self-employment income below the SE tax maximum ($48,000 for 1989) is the ability to reduce Social Security taxes. This is accomplished by structuring a compensation package composed of (1) a reasonable salary which is subject to Social Security taxes and (2) distributions, rent and other items which are not taxable for social security purposes.

The bottom line? Even if you're in the studio for love not money, that money is better served feeding your outboard rack than paying unnecessary taxes.

Doesn't that put it all in perspective?

10 Power Mixing Tips

This is the first in a regular series of mixing tips by leading recording engineers and producers.

BOB HODAS

1 WHEN MIXING WITH automation, make sure to save the mix often. Even the newest computer systems are quirky at times; if they crash you lose hours of work in a moment.

2 Use the fader reverse while mute writing. Monitoring the track on the console's small fader allows you to hear everything and it also gives you cues for elements you want in or out. Besides, it lets you concentrate on the music instead of on the meters.

3 Before doing a console reset, make sure to mark your fader positions. Faders don't always come back to the same position if they have not been properly calibrated or if the computer is stricken by a bug. It saves lots of time to visually verify your levels rather than wondering after the fact, "what's different about this mix now?"

4 Want to avoid MIDI time lag on your triggered instruments? Consider these: (a) Assign the instrument you are replacing to the sync head. The signal will naturally occur before the repro head. Delay that output to trigger the sample until it fits into the track where you want it. (b) When working with a Studer tape machine with dual outputs, assign the second output to the sync head. The trigger signal (the original snare, for example) is then delayed until the sample snare you're mixing with the original lines up perfectly.

5 Reverse reverbs, pre-verb ghosts or predelay can be created simply by taking the signal off the sync head and processing it before it occurs on the repro head.

6 Use outside gates for transient material. Even the latest and greatest consoles have very slow attack times on their gates (i.e., 100 microseconds). As a result, they can lose a lot of transients and rob the life out of your mix. I prefer to use independent gates, the fastest of which is the Aphex 612, with a four microsecond attack time.

7 When using EQ, try to cut instead of boost. If something sounds too dull, cut the offending frequency instead of brightening the channel. This way you won't get into brightening wars with other channels, which often leads to a condition I call "fade-itis," in which faders creep up until you lose the console headroom.

8 Just because reverbs have stereo outs doesn't mean all reverb information should be far left and right. Placing a verb panned with the instrument itself often creates a nice sonic space around that instrument.

9 Use gates for more than just noise control. For example, if a bass player is leading the kick on a tune, gate the bass keyed by the kick so the entry sounds tight. Create an envelope around background vocals by using hold and release times and key them with the lead singer.

10 Use EQ on effects sends. It is a lot easier to tune what is going into a digital reverb than what comes out in many cases. Different devices (and programs) have different personalities and require individual treatment. The right EQ for the instruments in your mix may not give the desired result in the reverb.

11 *Here's one more tip for good measure:* You don't always need automation to fit a lot of moves into a tune. So, don't forget the lost art of tape splicing. Mix the song in sections and then cut the pieces together with your trusty razor blade and splicing block.

***Hodas** has worked with the Doobie Brothers, The Village People and Windham Hill Records.*

User Unfriendly?

Manufacturers need to address their products to a new generation of artists who record their own music.

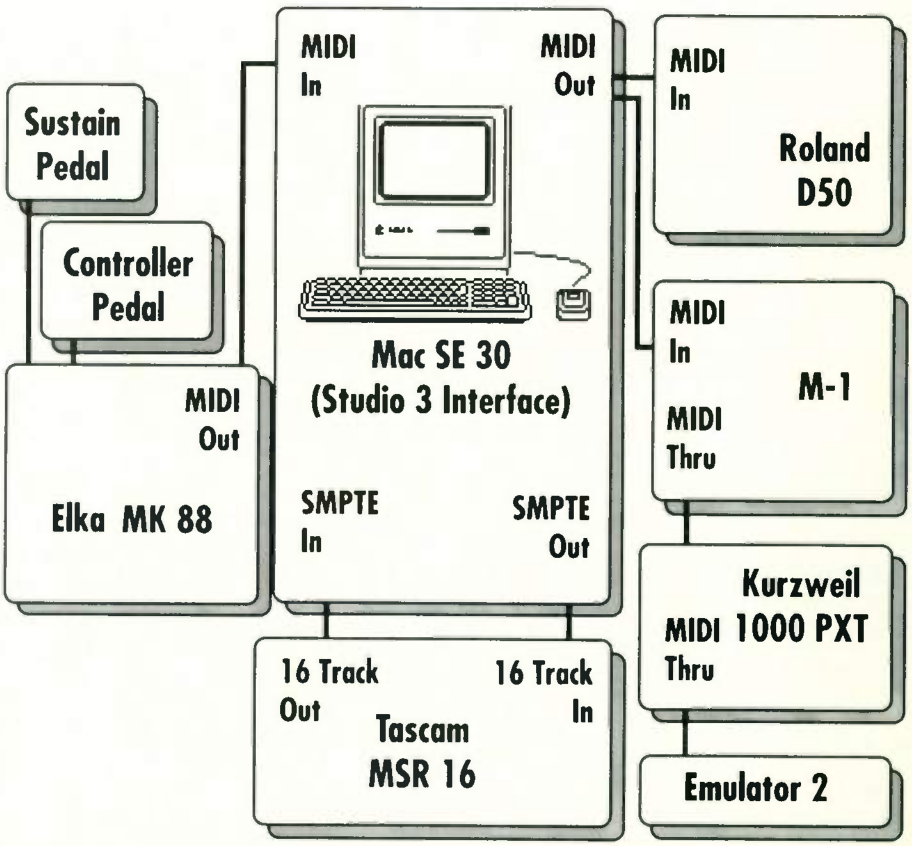
J.D. SHARP

A VAST CHASM has opened between musical artists and their equipment into which many good musical concepts have plummeted to their deaths. The role of a new generation of project engineers may be to span that gap, so artists can concentrate upon what they do best — creating art.

I recently helped a client implement her new personal production studio. Her system expanded from acoustic piano and Emulator to Kurzweil module, Elka MIDI controller, Roland D50 synthesizer, and Korg M1, plus a Tascam MSR-16 multitrack recorder, Macintosh SE30 running Studio Vision software, and Opcode Studio 3 SMPTE interface.

Her normal mode of creation involved spontaneity, layering synths and samplers somewhat randomly until something came along that resonated in her imagination, at which time she'd switch on the tape recorder and lay down a track. The results were fantastic, reflected in healthy sales of her independently distributed tapes and CD's. But she bought a computer so she could record "virtual tracks" directly to DAT for outstanding and crystal-clear results–making those CDs and tapes even better.

The MIDI system that Sharp designed made it so his client could just switch on a tape recorder and lay down a track. ▼



REAL-TO-REEL

Yet something got lost in the translation. MIDI was a blessing and an obstacle to her way of doing things. MIDI sequencing software is oriented more or less toward working one track at a time, using one instrument; as soon as this artist wanted to assemble impromptu combinations of instruments, things got awkward. She used to keep all instruments on a single MIDI channel, turning them up or down to create layers. There was no way to do this and keep data discrete on different tracks and channels.

We dove onto the Studio Vision manual and screen menus and discovered that a provision does exist for sound layering. She could override the normal "rechannelization" of data to a single MIDI channel by using an input map. We created the input map and moved on, since this only partially solved her problem. Seems that with an input map in place, Vision simply echoes exactly what's sent in, and my client didn't know how to make the Elka output on multiple MIDI channels simultaneously. So we spent almost an entire day translating MIDI engineering gibberish into humanly understandable concepts.

In the end all was well — but it was a bloody struggle the whole way. In the end her entire way of layering and creating sounds had been altered, for better or worse.

REAL DEAL

This experience clarified several points:

(1) Many products in our field are designed by engineers for people of an engineering bent, and have little to do with how musicians think. Yet often these products are used by artists, not engineers. If these products weren't put in the hands of musicians before making it to market, manufacturers who think they've made progress in the "user-friendly" arena are severely deluded.

(2) The owner's manual is probably the most neglected piece of equipment supplied with every device. The reason for such neglect? The vast majority of them are useless to anyone without engineering experience and many might as well have been written in Sanskrit.

(3) Equipment dealers are far too glib in extolling the virtues of the equipment they market, and don't make customers aware of the potential down sides.

(4) Many artists would be better served by leaving the engineering to someone who understands their creative needs and can bridge the gap between concept and execution.

The industry-wide trend is going in the opposite direction; more artists opt for self-production and self-engineering. This opens creative channels for some people, as they become masters of recording and computing media. For others, it merely interposes an all new discipline that may have nothing to do with where the artist's talent is.

Many personal project studios don't have a wall between control room and recording space; similarly, the new breed of engineer must emerge from behind the console and expand his or her knowledge to encompass the worlds of MIDI, SMPTE, and production software.

Armed with such skills, a studio staff engineer can enjoy security, instead of bemoaning the fact that personal studios are running small- and mid-sized operations off the map. Meanwhile, musicians can concentrate on sharpening their musical skills and expanding the boundaries of their imagination.

***Sharp** is the owner and proprietor of Bananas At Large, a Northern California pro audio and music dealership.*

Puttin' On The Tweak

VU Calibration, Reproduce Alignment, & Fun with Nanowebers.

DR. RICHIE MOORE

NOW THAT WE'VE tweaked, cleaned, and demagged the tape transport (as described in EQ Sept/Oct '90), let's calibrate the recorder's VU meters and heads.

GETTING STARTED

I can't emphasize enough how important it is to demagnetize and clean the entire tape path properly before you start the alignment. Taking these all-important preliminary steps prevents damage to the costly and fragile test tape, and prevents any debris from causing improper readings during the alignment procedure.

To align properly, you need a test tape, a good AC RMS VOM (volt-ohm-meter), and a steady sine-wave signal source such as a test oscillator. Required tools usually are a 2, 2.5, and 3mm hex driver, small (#1) flat-blade screwdriver or plastic tweaker, and a non-inductive screwdriver (depending on the machine). [Ed. Note: An oscilloscope is ideal, but we'll also describe a method that gets you close without one.]

Electronically aligning a tape machine involves several stages: machine input/output (I/O) calibration, reproduce alignment, and record alignment. We'll take them one by one, from the top.

The first step is to read the label on the side of the test tape box to find out the reference level, or listen to the instructions at the beginning of the tape to get that info. The announcer will say something like, "The following tone is 1,000 Hz recorded at the reference fluxivity of 250 nanowebers per meter." As explained in my last column, 250 nWb/m = +3 dB over 185 nWb/m, so we set the reproduce gain to 0 VU.

VU METER CALIBRATION

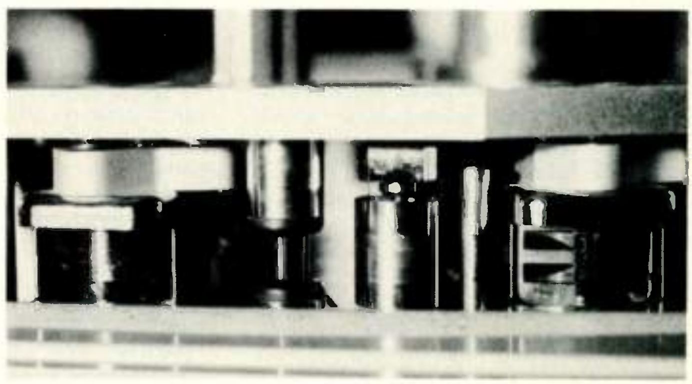
Using a VOM every time you align a tape machine is tedious. That's why all tape machines have built-in voltmeters: the VU meters. You need only use the VOM for occasional calibration checks and proof-of-performance verification.

(1) Plug the oscillator into your console (unless the console already contains one) and set it to 1 kHz at line level; that's +4 dBm (1.23 VAC) or -10 dB (.317 VAC), depending on your machine.

(2) After measuring to ensure that all levels coming from each channel of the console are the same, route the oscillator to all busses and set the tape machine to input.

(3) Attach the VOM to the output of track #1. Adjust the input or monitor gain trim pots on that track, so the voltage level coming out is the same as that going in from the console. Check the location of the meter indicators for each channel. If they read 0 VU, don't touch them. If they are not reading 0 VU, adjust the VU meter cal trimmer to achieve 0 VU. (Not all tape machines provide such a pot. Some manufacturers install meters permanently calibrated to the circuit, in which case there is not much you can do. For the most part, they're close enough for rock 'n' roll.)

(4) Perform step #3 for all remaining channels. You now have a very handy calibration tool with which to do alignments — your tape machine.



Don't tweak your machine until you've cleaned the tape path.

REPRODUCE ALIGNMENT

With the test tape loaded on the machine, grab your faithful tweaker and follow me:

(1) Using the 1 kHz section of the reproduce tape, set the repro gain on each channel to 0 VU. This is a preliminary adjustment.

(2) Connect an oscilloscope to the output of two outside tracks of the tape machine, with the 'scope in the X-Y mode. If aligning a multitrack machine, use the tracks just inside the edge tracks (tracks 2 and 15 or tracks 2 and 23, for example).

(3) Before adjusting the azimuth of the repro and record heads, make sure you positively know which is the azimuth screw (if in doubt, refer to the documentation). Locate and play the 10 kHz portion of the test tape. In repro mode, adjust the head for maximum output on the 'scope, achieving as close to a 45 degree angle as possible. Setting the correct azimuth involves turning the screw slightly in one direction or the other.

[Ed. Note: If you don't own an oscilloscope, connect the outputs of two outside tracks to two line inputs on the console. Assign these channels to a single VU meter on the console. While playing the 1 kHz tone in repro mode, bring up each fader individually until the common meter reads -6 VU. Turning both on should yield approximately 0 VU. Play the 10 kHz portion of the test tape in repro mode, and adjust the azimuth for maximum output on the VU meter.]

(4) Put the machine into sync mode and adjust the record head for the same conditions as step #3 above.

(5) Again play the 1 kHz tone and adjust the repro gain in repro mode for 0 VU reading.

(6) Using the 1 kHz tone, adjust the sync gain in sync mode for a 0 VU reading.

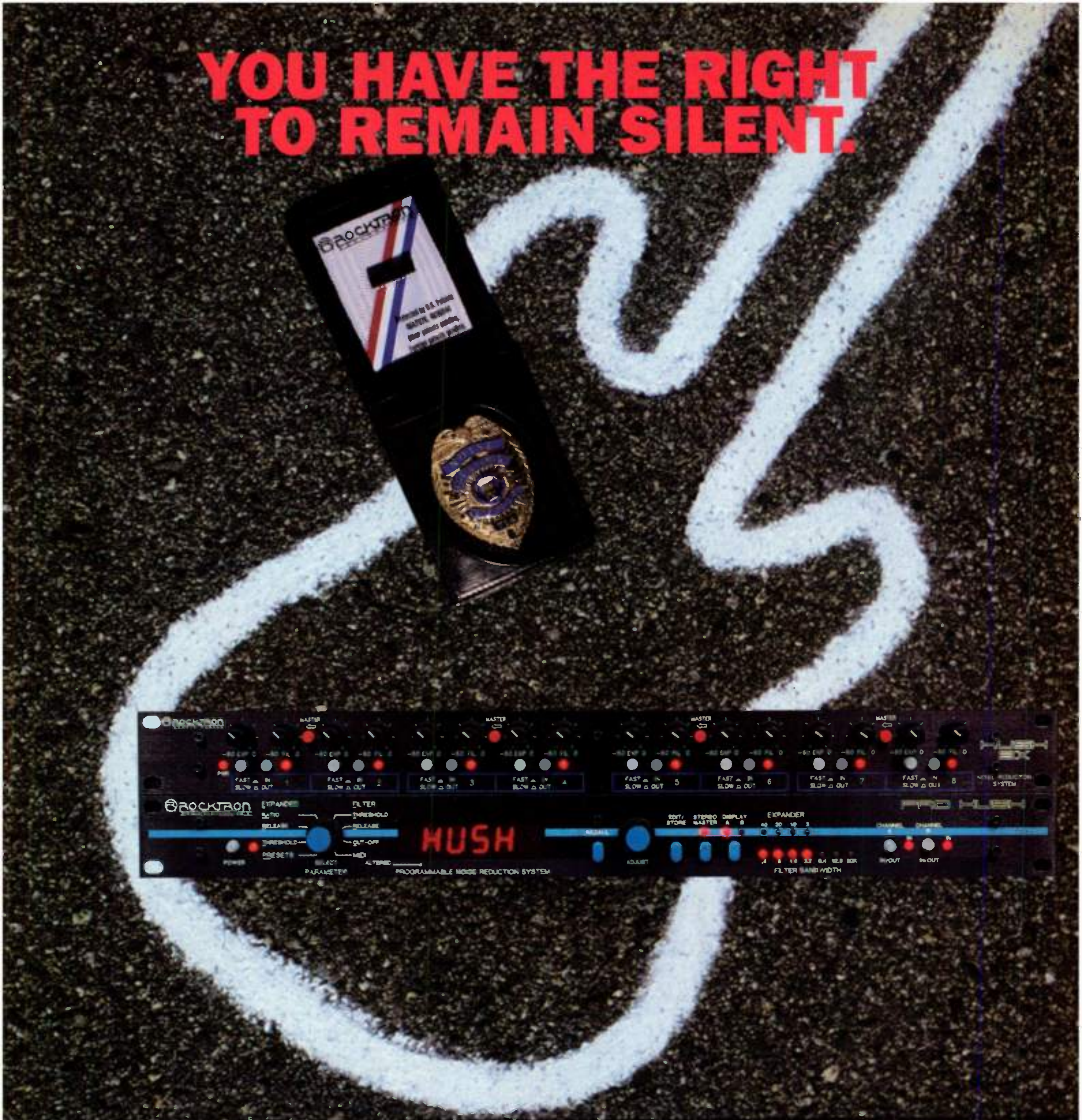
(7) In repro mode, play the 10 kHz tone and adjust the high-frequency equalization control for a reading of 0 VU. This control probably is labeled EQ or HF gain, or something close.

(8) If your machine provides a sync HF trim pot, adjust it in sync mode for the good old 0 VU reading.

The only accurate way to calibrate low-frequency gain is in record mode. We'll cover that next time when we look at recording alignment and the mysteries of tape recorder bias.

Until then, have a good tweak.

Moore *is chief technical guru at The Plant Recording Studios in Sausalito, California.*



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Enter HUSH circuitry. Our single ended noise reduction that effectively nabs noise at all signal levels and all frequencies while preserving the sonic integrity of the original. In fact, the arresting sonic improvement that HUSH provides has prompted one reviewer to testify: "This is high quality pro audio equipment with the flexibility to remedy a long list of noise problems."

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So take a bite out of noise. Get HUSH into your system. We'll help you put noise away for a long, long time.

ROCKTRON
STUDIO SERIES

CIRCLE 30 ON FREE INFO CARD

Audio & MIDI: Closing a Gap

Face it — music and MIDI are two different languages. Today's new software is helping the studio speak both, but the real thrill is yet to come: a universe where audio and MIDI can be treated as one and the same.

TED GREENWALD

MIDI MAY BE a creative savior — but it can also be a technical nightmare. When combining audio and MIDI, you always have to remember that you're speaking two different languages, and bridging the gap can be a real tongue-twister.

That's because these two types of information are incompatible on several levels. First, unlike music tracks already on tape, sequencer tracks aren't necessarily linear (the material can be looped, reversed, cloned, etc.) — plus, they don't necessarily have a fixed temporal relationship (they can be slid forward or backward, sped up or slowed down, shortened, and so forth, independently). And unlike audio, MIDI data is organized clearly according to pitch, duration, inflection, and other musical content, elements you can manipulate in isolation. On the other hand, audio data has a full definition of timbre and amplitude envelopes — information entirely lacking in MIDI.

STUDIO VISION

How do you put the two together? In the world of audio, the now voguish "music workstation" is a crucial first step. In fact, music software pioneers Opcode and Digidesign have debuted products that refine the integrated recording/sequencing environment, portending new breakthroughs in price and functionality.

Studio Vision ($995), from Opcode, is a software application that integrates Vision (Opcode's state-of-the-art MIDI sequencer for the Macintosh) with Digidesign's SoundTools ($3,285), a Mac-based two-track hard-disk recording package. Concurrently, Digidesign has introduced Deck ($349), a software front-end for SoundTools that provides a familiar-looking four-track recording/mixdown screen and plays back standard MIDI files along with digital audio.

The significance of both products is that they locate MIDI sequencing and digital recording (along with mixing and signal-processing effects) in a single user interface. Studio Vision and Deck offer a fresh glimpse of the power that awaits the music production community — levels of spontaneity, productivity, and flexibility unavailable today, but likely to be taken for granted in the next several years.

There are two ways in which Studio Vision significantly narrows the gap. The first is its price, which brings an unparalleled degree of audio/MIDI integration within reach of the smallest studio owners. The second is its user interface, which doesn't differentiate between audio tracks and MIDI tracks, at least for the purposes of large-scale viewing and editing. Regions selected for cutting and pasting can include both sequenced and recorded tracks; multitrack editing is performed as though they were one and the same.

FUTURE VISION

Studio Vision can give us all a taste of the future — a future in which the differences between the sequencing and recording will become more or less irrelevant in production. Cutting and pasting en masse is only the beginning; someday, we'll be able to take sequenced and recorded tracks and quantify, time-compress (or expand), tempo-map, bounce, crescendo and transpose them together.

Sound good? The future will also see the end of the difference between disk-based audio (associated with conventional recording) and RAM-based playback (more often employed in conjunction with sequencing). That means fragments of audio might be swapped from hard disk to RAM as necessary, automatically, upon invocation of enveloping, looping, filtering, or triggering commands. The result will be enormous creative power at the touch of a couple of buttons.

Studio Vision offers a fresh glimpse of the power that awaits the music community as MIDI and audio recording technologies are combined.

The New England Digital (NED) Synclavier triggered this trend with the introduction of the Sample-To-Disk option in 1982, and has offered integrated features in a multitrack environment since 1985. "Our sequencer doesn't discriminate between the sounds created in the synthesizer, in RAM, on hard disk, or via MIDI," says Ted Pine, NED's marketing manager. There's no practical difference between punching into a sequence track and an audio track; in fact, one command converts a sequence track into audio data. But the sequencer doesn't give a coordinated, comprehensive view of various levels of data organization or apply MIDI-style functions to audio tracks.

In its debut edition (v1.1), Opcode's Studio Vision, linked up with Digidesign's SoundTools, would perform some MIDI operations on audio tracks. For instance, the start times of user-defined audio regions could be quantified to a specified rhythmic value. That was the first step toward providing audio parallels for the microscopic editing capabilities already taken for granted in MIDI production. The next will be to automate the process, so that the system can identify the sounding portions of live track, isolate them, eliminate the intervening silences (to save memory), and quantify them, all in a single command.

With the addition of a pitch-recognition algorithm, the system might parse an audio track into its component pitches and durations, making them available for MIDI-style manipulation. Ultimately, it

would be able to eliminate redundant notes, compiling a recorded performance as a conventional sequenced multisample, complete with controller data for pitch bends and dynamics, and available for re-sequencing. It could alter individual pitches to fix a clam — and indeed, might identify clams automatically — and edit individual timings to tighten up an uneven performance. Spot wiping would be reduced to deleting an on-screen audio event. It could even apply esoteric MIDI functions like key-coherent transposition (in which some pitches are transposed more than others) to recorded audio.

David Frederick, in charge of product development at WaveFrame, makers of the AudioFrame workstation, offers the concept of "multi-purpose tracks" allocated by the machine itself when presented with a given type of data. "We need machines capable of determining the most efficient way of storing a performance," he asserts. If you were to connect a mic to an analog audio input, the machine would allocate a hard-disk track. If you fed it MIDI data on a currently unused channel, it would create a sequence track automatically.

> An integrated digital recording/sequencing system might parse an audio track into its component pitches and durations, making them available for MIDI-style manipulation.

USER VISION

How would such functions appear to the user? Opcode founder Dave Oppenheim, "designing on the fly" as he puts it, imagines a visual display depicting audio and sequence tracks, allowing the user to "zoom in" to a music-notation view of the pitch and duration materials of sequence tracks and audio tracks alike. The user wouldn't need to know if it was MIDI or audio until he wanted to make an alteration peculiar to one kind of data or the other. At that point, he could select to view the data level.

If the data happened to be MIDI, the display would be alphanumeric; if it were a digital recording, the screen would show an editable waveform graph, or perhaps a Fourier spectrum. The MIDI display, ideally, could be toggled to a synthesizer edit window, making the parameters of a synthesized sound accessible. Likewise, a software toggle could replace the audio waveform display with sample editing tools for looping, enveloping, filtering and other DSP functions.

What stands in the way of realizing such notions? One of the biggest roadblocks is simply research and development, particularly in software. "Most of the technology, in terms of components, is already available," according to Ted Pine. "The real advances in the 1990s are going to be in the user model." MIDI sequencers have been designed around a tape deck model for years; the opposite has yet to be attempted. With the range of other functions that might be built into such a system, designing an interface for a practical, integrated workstation is no mean feat.

There are technological difficulties as well. After decades of research, a solution to the problem of polyphonic pitch tracking — that is, the ability to identify the pitch components of recorded tracks so that they can be manipulated — remains elusive. Real-time pitch-to-MIDI converters such as IVL's Pitchrider suggest, however, that monophonic tracks may no longer pose a problem. Refinements of existing technologies — faster hard disks, faster microprocessors with wider word lengths, larger RAM chips, cheaper analog-to-digital converters — appear to be well within the foreseeable future. The day of the fully integrated music production system may not be so far off.

Greenwald *is a music producer, composer and journalist. Recent projects include "The Beatles Companion" and music that has appeared on Windham Hill and Sonic Atmospheres Records.*

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CIRCLE 15 ON FREE INFO CARD

Audio at Your Desktop

Three ways to match your desktop sights with digital sounds.

HAROLD OSBORNE

WHAT DOES THIS new buzzword "multimedia" mean to the world of audio recording?

Multimedia includes such diverse concepts as interactive videodiscs, games, and educational software. What the "revolution" is about, primarily, is that the tools for creating these things are no longer the exclusive property of expensive studios, but are now available to anyone with access to a personal computer.

The function of sound in multimedia is to link with images, to act as bridges between program segments, or to convey information when the viewer or user takes a certain action, like pushing a button or selecting an item from a menu. "Authoring systems" are the software used to create the presentations, and they use sound as "cast members," or items to be called up at specific times.

DIGITIZED

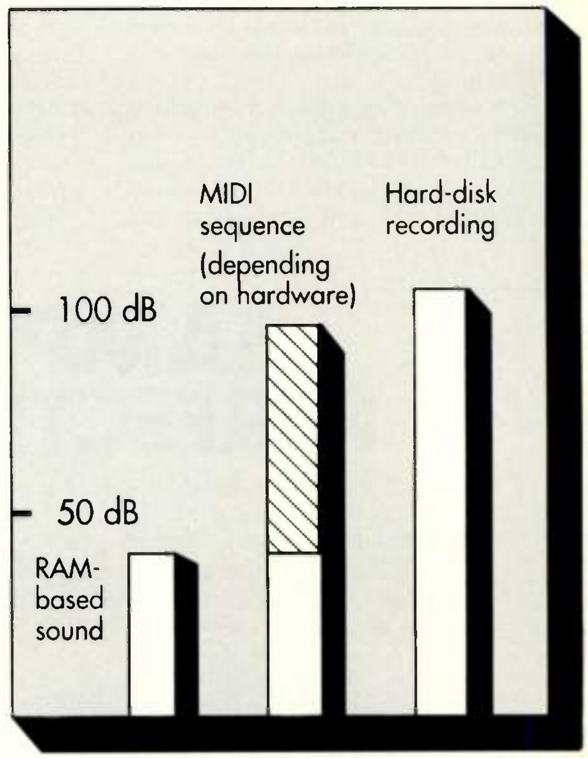
There are basically three ways to incorporate sound into a multimedia presentation. The simplest is with a digitizer that records audio from a microphone into RAM. Each sound in RAM is given a label, and can then be played back on cue. Its chief advantage: it's cheap. An audio digitizer costs only about $100 or so, and if you're using short sounds, you don't need a lot of RAM to play them. RAM is itself relatively inexpensive—current prices are hovering around $50 per megabyte, which is enough for up to about four minutes of sound at minimum fidelity. External storage, once you're done with a sound, is relatively easy, especially with high-density floppies and hard disks. Digitizers are available for just about every type of computer, so this method of recording sound for multimedia is very popular.

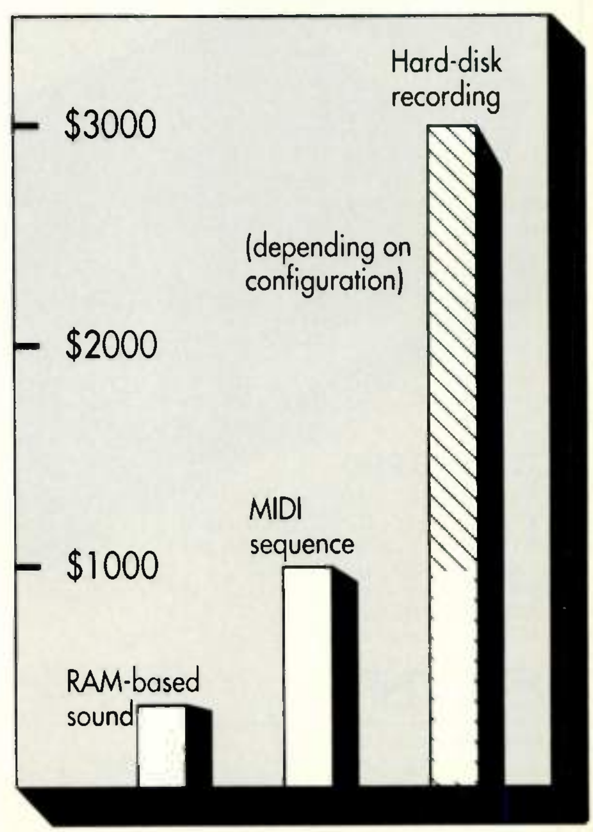
The big tradeoff, of course, is quality. Digitizers work in 8-bit samples, which gives a best-case dynamic range of only about 48 dB — and due to design compromises that keep prices down, it usually works out to be much less than that. The most sophisticated systems have a maximum sampling rate of 22 kHz, which translates to a possible top end of about 10 kHz, but again, performance is often significantly poorer in practice. But for narration, dialog, and even some effects, RAM recording is perfectly adequate.

DISK DRIVING

A high-fidelity alternative is hard-disk recording, now available on Macintosh, IBM, and Atari platforms. In this method, incoming audio is recorded directly to a high-speed SCSI hard disk, bypassing RAM almost completely. Audio performance is much improved, with 16-bit recording and sampling rates as high as 48 kHz. Much attention is paid to filter and convertor design, and the systems sound so good, some can even be used to master CDs. The hardware to accomplish this, of course, is significantly more expensive than simple digitizers: typical cost is $2000 to $3000 (not including the computer), although a recently-introduced system for the Macintosh, Digidesign's Audiomedia, retails for about $1000.

Storage, however, is a major issue. A one-minute stereo sound file at maximum fidelity will occupy 10 megabytes on a disk. While having 10 megabytes available on a typical computer hard disk is not uncommon, the question remains of what to do with the sound when you're done with it. Removable hard disk drives, read/write magneto-optical sys-





Above: Comparative dynamic range; Below: Minimum hardware costs (not including computer).

tems, and tape streamers all can provide adequate storage for archiving such storage for archiving such huge files, but they aren't cheap.

Accessing a hard-disk recording from within a multimedia application is not as straightforward as dealing with RAM-digitized sound, and special routines have to be included in the software to locate and play the sounds from disk. In the Macintosh world, these are known as "X-objects," and take the form of small programs that are inserted into Hypercard, Macromind Director, or other platforms, and are supplied by the makers of the audio hardware. Once these programs are installed, calling up hard-disk files on cue in a presentation is simple.

MIDI MINDED

The third method of doing sound with Multimedia is to use MIDI. MIDI music and sound effects can be created with any sequencing program, and the sequences stored as standard MIDI files. On the Macintosh, they can then be played back through multimedia software using X-objects similar to those used to play hard disk recordings.

MIDI's great advantage is that it is almost infinitely flexible. When a MIDI sequence is playing, it can be altered in real time; tempo and volume can be changed, tracks can be muted or brought in, musical lines can be transposed, loops can be entered and left, and pieces can even be re-orchestrated on the fly using program change commands.

> Some manufacturers, like Roland, are directly addressing the MIDI-for-multimedia market with inexpensive standalone multi-timbral modules.

MIDI also gives you the widest range of choice in terms of playback hardware, whether you use a single keyboard-less synthesizer module, or an automated 32-channel studio. Some manufacturers, like Roland, are looking at directly addressing the MIDI-for-multimedia market, and are making inexpensive standalone multi-timbral modules featuring high-quality synthesized and/or sampled sound with no controls at all, designed to be operated entirely from a computer.

MIDI's chief disadvantage is that you can't record "real" sounds, like speech, without samplers, which increases cost and complexity dramatically. But there's no rule that says you can't combine methods, and use MIDI for music, hard disk recording for those bone-jarring sound effects, and RAM recording for voices.

As the field develops further, we should begin to see more such "hybrid" uses of different audio technologies, which will allow sound to take its rightful place as an equal partner to sights in the world of desktop multimedia production.

EFFECTIVE VIDEO F/X

What's new in desktop video? The Macintosh community is buzzing about the newest desktop video production system, Video F/X, an integrated video, audio and graphics editing system that lets you create pro-quality videos from a Macintosh II keyboard.

Video F/X uses proprietary analog and digital technology to translate video signals into digital computer format and back to video. Its hardware includes an external video/audio control box and a video frame buffer add-in card for the Macintosh II. The control box contains video and audio switchers which mix video sources and stereo sound tracks while placing all video tape recorders and other peripherals under computer control.

Other features include an alpha keyer which superimposes titles and graphics cleanly over live video with multiple degrees of transparency; a transition generator which fades, wipes or dissolves scenes; and circuitry to synchronize the Macintosh with VTRs. The digital frame buffer is a 32-bit color QuickDraw compatible card which can capture frames of video and output Macintosh graphics to a VTR.

For video clip selection, definition and review, producers use the Macintosh to control playback from the VTRs, with accurate control to a single frame. Single frame accuracy is needed to create professional quality video presentations. Users can select start and stop points and capture frames of video in real time.

Video F/X will support interformat editing, utilizing VTRs in a variety of formats, from Hi-8 and S-VHS to U-matic, Betacam and videodisc. The product can receive input from two video and three stereo audio channels, as well as Macintosh audio or microphone input to the audio mixer.

Video F/X's key software element is its Organizer Window, which is used to script, link scripts to video footage and log clips. As the user selects a clip, Video F/X records the reel name, timecode position, duration, clip name, and the head frame (a picture of the first frame of video selected). This information is displayed on the computer screen as well as recorded in an Edit Decision List (EDL).

Users can cut and paste video images, insert graphics, trim and rearrange segments, add audio, and select scene transitions such as fade-ins, directly from their Macintosh screen. Video F/X can then assemble their program automatically and with frame accuracy. The user can preview edit decisions on a video monitor, continue to edit if necessary, and then produce a finished master video tape.

A minimum configuration requires a Macintosh II, eight megabytes of memory, 40 megabytes of storage, a Macintosh RGB color monitor, two VTRs, an NTSC video monitor and a System 6.0.5 (or higher) operating system. The product was scheduled to begin shipping in October 1990 at a price under $10,000.

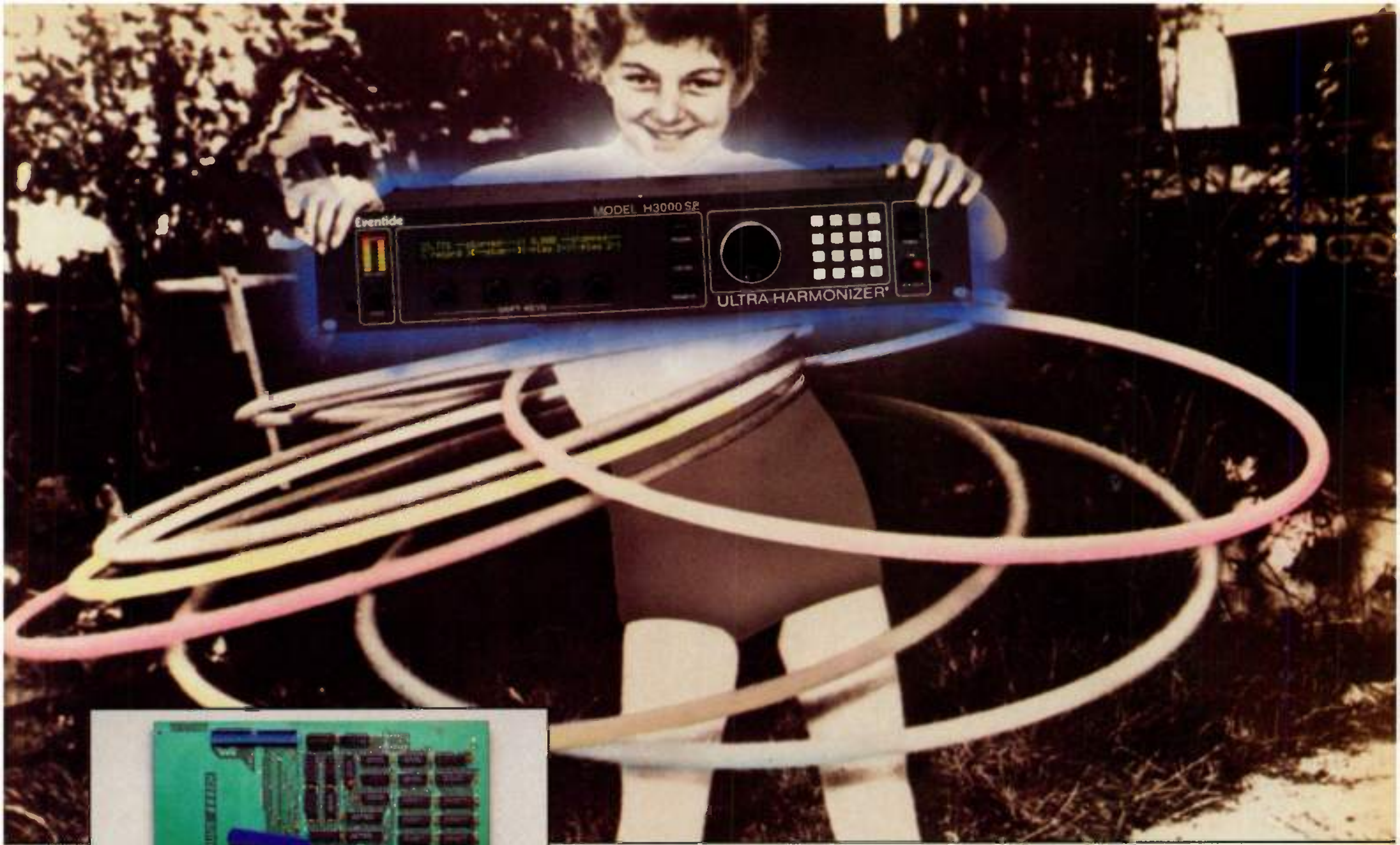
Video F/X is an integrated desktop system requiring a fraction of the knowledge, cost and space investment of traditional editing systems.

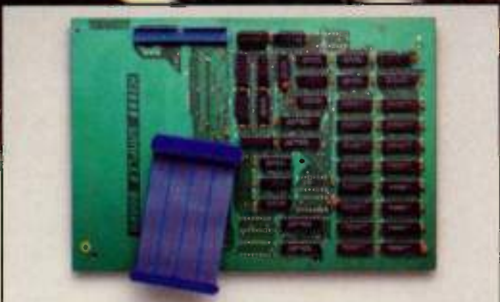
DataDisk Goes Live. Musicians can now use the Alesis DataDisk for playback in live applications without having to use sequencers and computers, thanks to recent software upgrades that include a MIDI sequence recording/playback feature. Along with this upgrade, Alesis is also offering new software for the Quadraverb simultaneous multi-effects processor, featuring 1.5 seconds of sampling at full bandwidth, which can be triggered manually or via MIDI. Flexible multitap delays, a ring modulator, resonators, and MIDI-controlled dynamic panning round out the new Quadraverb features. The software updates will retail for $30 each. *Contact: Alesis, 3630 Holdrege Ave, Los Angeles, CA 90016; (213) 467-8000; Fax (213) 836-9192.*

Hybrid Intros New ADAP II Software. Users of the ADAP II digital audio recorder/editing system from Hybrid Arts can now change the length or tempo of an audio file uniformly — without changing its pitch or introducing artifacts — using the new TimePage software. This new algorithm can scale speech by plus or minus 30 percent without artifacts, says Hybrid's Director of Research and Development, Chuck Peplinski. *Contact: Hybrid Arts, Inc. 8522 National Blvd, Culver City, CA 90232; (213) 841-0340; Fax (213) 841-0346.*

New Application Notes From Lexicon Lexicon is offering a range of free Application Notes as part of a continued effort to provide tips, tricks, and techniques to owners of Lexicon PCM-70 and LXP-5 digital effects systems, as well as the MRC MIDI Remote Controller. These include: "PCM 70/MRC: Chorus 300DLY. Stereo Chorus and Delay with Remote Control Using Dynamic MIDI(r)"; "LXP-5/MRC: Interactive Improvisation and Alternative Performance Techniques"; "MRC: MIDI, Mixing, and the MRC"; "MRC: Mono Mode on Yamaha Synthesizers"; LXP-5: Too Many Nouns? Try These 'Verbs' (Getting More Reverb From the LXP-5)"; "MRC: Two MRC Set-Ups For The Kurzweil 1000 Series"; "MRC: Using the Lexicon MRC MIDI Remote Controller with the Alesis Quadraverb"; "LXP-1: Controlling LXP-1 Decay and Delay With Your Drum Machine." *Contact: Lexicon, Inc., 100 Beaver Street, Waltham, MA 02154; (617) 736-0300; Fax (617) 891-0340.*

Peavey Adds Sampling Option to the DPM 3. Peavey Electronics has introduced upgraded software for the DPM 3 keyboard, along with a sampling option called the DPM VE. The new software provides local edit capability for samples loaded via MIDI Sample Dump Standard, and it also supports dumping of samples from internal memory via MIDI Sample Dump Standard, with expansion Sample RAM capability of up to one megabyte. Changes in the software can totally reconfigure the method used to generate sounds. A new sampling option is also available for the DPM 3 keyboard, the new DPM VE. The VE is rack-mountable in a standard 19-inch rack, occupying only a single rack space, but holding plenty of power. The samples are 16 bit; the unit supports the MIDI Sample Dump Standard. By coupling the DPM VE module with the 2.0 software version of the DPM 3, a player is able to take his own 16 bit samples and load them into the DPM 3, as well as performing editing on the samples. MIDI IN, OUT, and THRU ports are provided. The DPM VE supports any sample rate from 16 kHz to 48 kHz. *Contact: Peavey Electronics Corporation, 711 A Street, PO Box 2898, Meridian, MS 39302; (601) 483-5365; Fax: (601) 484-4278.*

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CIRCLE 18 ON FREE INFO CARD

continued from page 10

limiting, expansion, or by a noise gate. The means of control that is used depends on what is needed to be done to the signal.

Each of the controllers mentioned above can be installed into the direct signal path, however, the use of the side-chain can add more control over the signal. For instance, a compressor installed in the direct signal path may cause some undesirable effects on the signal. The effects are known as "pumping" or "breathing." Both refer to the same side effects, and to a suppressed signal. The "pumping" generally occurs at low frequencies. The suppressed signal will occur at high levels, due to drastic gain reduction.

In an effects loop the direct signal path is interrupted, sent to an exterior device, processed, and then returned to the direct signal. A side-chain effect will not interrupt the direct signal path, it is a branch from the signal path to an exterior device, processed by that device, and when returned, then summed with the direct signal path.

To eliminate the pumping from the system put a low frequency dipping curve on a graphic equalizer to desensitize the compressor when those frequencies are present.

An example of this configuration of a side-chain would be used on a compressor. By adding an equalizer in the side-chain, the compressor becomes "frequency dependent." This technique is often used on vocal tracks and is known as a "de-esser." De-essing refers to the removal of the high level of sibilant frequency in the vocal track, which will cause distortion. A good example of excess sibilance would be on a live local newscast. When watching the news, listen to the field reporters. At the end of the words listen for a prolonged "s" or during the word for a prolonged "sh" sound. This is when a de-esser needs to be employed.

This method of installing an equalizer in the side-chain of a compressor can also be used on low frequencies. This would be done when too much low frequency energy is present in the program. (Normally a graphic equalizer is used for this technique.) When this occurs the compressor will ride the gain of the system up and down in time with the bass beat. As a result "pumping" will occur. To eliminate the pumping from the system put a low frequency dipping curve on a graphic equalizer to desensitize the compressor when those frequencies are present.

When adding controllers to the side-chain the overall system noise will not be increased. However, when adding anything into the direct signal path or the effects loops, the system noise will increase. Therefore, the use of the side-chain is the preferred method.

Bruce R. Griffen, Technical Support Mgr.
Rane Corporation

Q **I'm thinking of replacing our multi-box speaker system with a small package processor system. However, I'm wondering how the array of boxes will compare sonically when highs/mids/lows are stacked and pointed where I need them. The sensitivity and coverage charts don't tell me all I need to know. What's the reach of a small box system? The idea of a smaller, flyable system appeals to me, but what am I giving up?**

Feliciano Sevilla
Catano, Puerto Rico

A The sonic quality of a speaker system is dependent on the caliber of the speaker components and the way in which they are combined and aimed. Start with top quality components to get an excellent sound.

The purpose of an array is to reduce the interference between speaker systems by minimizing overlap of coverage patterns. However, speaker systems do not magically combine merely by placing them in certain configurations. All systems, even those that are "trapezoidal," must be aimed to adequately cover the audience area. The trapezoid shape is designed to get the rear of the cabinets closer together for proper coverage.

The "reach" of a system, also referred to as the "throw," is determined by the directivity of a speaker system across its bandwidth. Therefore, as a rule of thumb, three- or four-way systems with horn-loaded mids and highs will have a longer throw than a two-way system with direct radiating low/mids and horn-loaded highs. Since a horn mouth must be larger to control lower frequencies, most effective "long-throw" systems are larger than the typical small "processed," or electronically-controlled speaker systems. However, recent developments in fairly compact one- and two-box "all-in-one" speaker systems with horn-loaded mids and highs have made these ideal for medium and large applications where longer throw is desired. Often, the smaller systems are used to augment larger systems for front- and side-fill use, or by themselves for short to medium-throw applications. Try to find these types of systems being used for events, listen to them, judge the environment and then determine what will work best for your needs.

Ivan Schwartz, Concert Sound
Marketing Manager
Electro-Voice

EQ&A

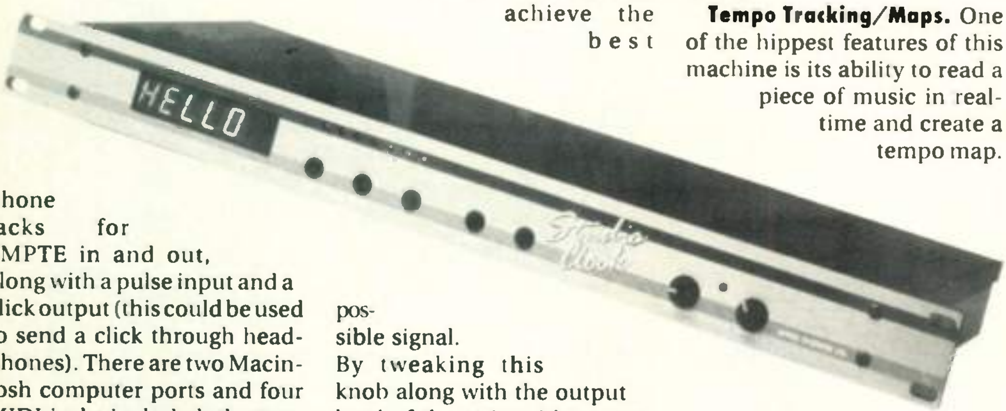
Send your technical queries to: EQ Editorial, 939 Port Washington Blvd., Port Washington, NY 11050.

Aphex Studio Clock

Aphex's Studio Clock is a sophisticated master clock with lots of power packed into a single rack space. In a nutshell, the Studio Clock reads and writes all four SMPTE rates (30 fps, 24 fps, 29.97 fps, 25 fps) and can convert that code into MIDI clocks with song position pointer, MIDI Time Code, or Direct Time Lock. It also incorporates a tempo tracker that can sync to existing music even when time code isn't present. The ability to create a tempo map along with a Macintosh MIDI interface completes the package.

Back & Front. The back panel of the Studio Clock includes phone jacks for SMPTE in and out, along with a pulse input and a click output (this could be used to send a click through headphones). There are two Macintosh computer ports and four MIDI jacks included; the normal In, Out, and Thru, along with another leading directly into the second Mac input. Also on the back are the power switch and the input jack for a 9 volt power transformer.

The working end of the Studio Clock includes a large six-digit LED display. This display is used to show both the timing information and the parameter settings. Programming the Studio Clock is accomplished by accessing different pages of functions. A series of five LEDs are used to inform you of the current page, and three buttons (up, down, and yes) are used to access the different parameters and to set their values. Additional LEDs let you know when the unit is sending MTC, storing a tempo map in memory, running, or listening to information at the audio input. Two level knobs, one for trigger and the other for SMPTE, round out the front panel.



Readin', Writin', and Rhythm. When striping time code with the Studio Clock, options are available for setting the SMPTE starting time. For example, you might want to stripe a tape with the time code beginning at 1:01:05:00 (hours:minutes:seconds:frames).

The Studio Clock can read a SMPTE signal from -30 dB at a -10 level, up to -2 dB at a +4 level. The unit's sensitivity can be adjusted with the SMPTE level knob in order to achieve the best possible signal. By tweaking this knob along with the output level of the stripe, it's easy to get a strong and consistent lock. If, for some reason, the Studio Clock has trouble reading the code, the front panel's LED lets you know by displaying BADBAD. Error correction in the software makes it possible to read pretty bad code, and even read code on tapes that have been spliced.

Operating as a standard SMPTE to MIDI converter, you can specify a starting time for the first measure of music (perhaps the song should start at 01:01:15:00), tempo in beats-per-minute, and a SMPTE offset value. From there, you can access another programming page to set different values for certain measure numbers. For example, the tempo, meter, or the offset could be changed at bars eight, 12 and 24. There are also options for editing or deleting any changes that you may have programmed into the Studio Clock.

Synchronization is available in three different flavors. MIDI clocks and song position pointers would be used when slaving most drum machines or software sequencers to tape. Direct Time Lock is available for those sequencers that can respond to it, and MTC is supported. Of course, when outputting MTC, tempo changes, offsets, and meter changes are irrelevant as MTC carries no timing information within its code.

Tempo Tracking/Maps. One of the hippest features of this machine is its ability to read a piece of music in real-time and create a tempo map. There are three ways the Studio Clock can accomplish this bit of magic. First, it has the ability to listen to an audio source routed to the pulse-in jack. Second, it can listen to MIDI clocks. This is great if you want to dump a tempo map from an external sequencer into the Studio Clock. And third, the unit can create a tempo map from reading Note-On messages from any MIDI controller.

The tempo tracker (used when tracking from audio or MIDI) makes use of four different algorithms. Each algorithm is designed to provide a different window for capturing the timing of the performance. For example, Algorithm 'A' allows the tempo to be changed drastically over a short period of time. It is best used when the tempo tracker is looking for quarters and eighths (perhaps the snare and kick of an acoustic drummer). Algorithm 'C', on the other hand, will understand very complex rhythms, but the tempo can only be pushed or pulled gradually.

Using the tempo tracker feature, the Studio Clock can be used to sync a drum machine or software sequencer to a real musician. Since it can track audio and MIDI note-on messages, the musician could be performing live on acoustic or MIDI instruments.

The Studio Clock will even let you change the tempo tracking parameters at specified measures. Along with programming the tempo, offset and meter of any bar, you can also tell the Studio Clock to listen to a different source (MIDI notes, MIDI clocks, or audio) and apply a different algorithm at any particular measure.

The tempo tracker can even track a full audio mix (strings, winds, vocals, etc.) because it listens for transients from the drums and other rhythmic instruments. It has trouble tracking a smooth flowing passage

MANUFACTURER: Aphex Systems, 11068 Randall Street, Sun Valley, CA 91352; Tel: 818-767-2929.

APPLICATION: A powerful SMPTE to MIDI converter for situations where you must sync to recorded material, or have a live drummer control a software sequencer in real-time. production/performance, recording, post production.

SUMMARY: Its ability to lock a software sequencer to a prerecorded material was nothing short of amazing.

PRICE: $695

that has little rhythmic movement (this is perfectly understandable), but if the mix has a rhythm section, this baby can lock to it.

When creating a map, you instruct the Studio Clock that you want to make a recording of the changes taking place in the tempo tracker. According to the folks at Aphex, when a map is being recorded, each eighth note is tagged to a particular SMPTE location (accurate to the bit). In addition to this, each downbeat of every bar has a special tag which locates its position in the SMPTE code.

When playing back a map, the Studio Clock is constantly checking the location of each and every eighth note and comparing it to the tagged SMPTE location. You may not be aware (visually) of tempo changes at the level of the eighth, because the display only updates the beats-per-minute at the beginning of each measure.

Once the Studio Clock has a tempo map in residence, the unit can be instructed to play back the map (in which case the standard SMPTE to MIDI converter is negated), or to disable the map. If you wish to edit the map, you can do so on a global scale. Global editing allows you to adjust the entire tempo map more on top or behind the beat with a resolution of one SMPTE bit (416 microseconds). Another function of the global edit is to move the entire map to a new SMPTE location. The Studio Clock also offers a provision for editing single events in the tempo map. Any eighth note can be shifted forward or backward in time with the resolution of one SMPTE bit.

Strengths & Weaknesses. There are several features that make the Studio Clock a hip piece of gear. The display can be seen from several feet away as the LEDs on the front panel are quite bright. In addition, the unit can perform a Sysex dump from the front panel. This means you could create a new tempo map or a feel for a tune, and then save it out to your sequencer or universal librarian for later use. You could even audition changes in the tempo or the offset in real time while a sequencer is slaved to the Studio Clock.

Now for some minor gripes. The power switch on the back of a rack-mount unit doesn't make sense. You either have to crawl behind your rack to turn it off or leave the box on all the time. In addition, 9 volt power converters seem to be self replicating. Adding another one isn't an idea that drives me wild.

The Studio Clock will let you set time signatures for different measures, but within a limited range. Meters can be programmed to be from three to 12 eighth notes long. While these settings may be sufficient for most pop music, working in meters such as 7/16 or 11/4 creates a nightmare. While the Studio Clock will accept tempo and offset changes, they can only be accomplished at the level of the measure. There are no provisions (other than using the tempo tracker or recording a tempo map) for programming a tempo change in the middle of a bar or performing a smooth accelerando or ritardando.

Conclusions. The Studio Clock's ability to lock a software sequencer to a prerecorded material was nothing short of amazing.

If you're looking for a SMPTE to MIDI converter and run into situations where you must sync to recorded material, or have a live drummer control a software sequencer in real-time, the Studio Clock could be your salvation. If you're interested in having the studio clock follow a live MIDI controller, have a salesperson give you a demo first. *—Norm Weinberg*

EDITOR'S NOTE: New software upgrades for the Studio Clock were just introduced — including a Macintosh editor with extensive tempo map editing features, plus a Performance ROM that turns the Studio Clock into a central stage brain, controlling various MIDI products while syncing the sequencer to a live drummer.

Encore Notation Software

With six different music notation programs already on the Mac market, why would Passport enter the fray with a new program? Perhaps the answer lies in an elusive combination of high-quality printing, MIDI capability, control over the printed page, and ease of use. To date, no single program seems to be all these things to all people.

Encore offers high-quality printing on a Postscript printer using Adobe Systems' Sonata font. The program both sends and receives MIDI: It creates notation from standard MIDI files and creates MIDI data from notation. Encore gives the user a great deal of control over the printed output by being able to adjust the placement of almost any symbol on the page. The program has a friendly user interface and it's as fast as anything else on the market (even on a Mac Plus). So, could Encore be the one we've all been waiting for? Read on.

Data Entry. Encore lets you enter notes in four different ways: step entry with the mouse, step entry with a MIDI controller, real-time entry with a MIDI controller, and importing MIDI files. All four methods can be mixed and matched to best suit your own work habits. Imported files must be standard type one MIDI files or originate from Passport's own Master Tracks Pro, Master Tracks Jr., or Pro 4 programs. Importing is extremely fast. When data is first imported to Encore, it appears as a series of note heads representing pitch and relative position within the barlines. Each track of a MIDI file will appear on a separate staff (maximum of 64) and bring with it track names and tempo maps, along with time and key signatures.

The next step is to ask the program to guess durations. In essence, this quantifies the notation from the MIDI data and assigns traditional note values to the dots. Encore had no trouble transcribing eighth and quarter triplets accurately, but completely missed any other duplet value. Once a hunk of data has duration, the program can "beam on beat." This automatic feature beams groups of notes to coincide with traditional beaming rules depending on the meter.

There are two other ways to use MIDI information when entering note data. In step-time, Encore will listen to a MIDI controller to determine pitches, while you select the rhythm from the Mac's keyboard or the notes palette on screen. Encore is pretty quick at this too, tossing about two notes per second on the screen. In you prefer, Encore will also record your MIDI performance in real-time.

Notes can also be entered with the standard point-and-click technique. Simply select a note value from the notes palette and click a position on the staff.

MIDI Playback and Editing. These two topics are closely related. Encore is not a sequencer, yet the program can be used as a MIDI data generator that will let you hear all of your edits through MIDI compatible gear.

Depending on the particular action, edits to your score may affect the screen, the MIDI data, or both. MIDI data can be massaged in a complementary way for most edits that affect only the screen. For example, you could add a crescendo to your score, and then tell Encore to gradually change the velocity of the MIDI data associated with the crescendo. While not automatic (like Finale's executable shapes), this method is still effective.

Selecting data for editing is quite easy and very thorough. Each staff can contain up to four voices and each voice can be adjusted individually. Data can be selected by note, section, bar, staff, page, or the entire composition. Shift-click selection of discontinuous material is supported.

Perhaps the greatest editing plus of Encore is the ability to move and adjust any notational symbol on the page. All graphics in Encore are extremely malleable. Crescendo, diminuendo, slurs, ties, pedal marking, boxes, circles, oblongs, and lines can be dragged

MANUFACTURER: Passport Designs, 625 Miramontes St., Half Moon Bay, CA 94019; Tel: (415) 726-0280.

APPLICATIONS: A notation program that will adapt itself to the way you actually compose — on a sequencer, with a pencil and paper, or in real time.

SUMMARY: High-quality notational software, tested on the Macintosh, that is every bit as powerful as it is easy to use.

PRICE: $595

IN REVIEW

and stretched after placement. Even beams can be grabbed and adjusted in terms of height and angle.

Interested in adding text or lyrics? No problem. Text boxes can be entered anywhere on the page and up to four verses of lyrics are automatically centered under notes.

Score Control and Layout. Page setup is not difficult. Initially, you let Encore know how many measures to place on a line (from one to eight), and how many systems to place on a page. Once defined, all measures take up the same amount of physical space. You can give a measure more or less room by dragging the barlines. You can also tell Encore to add or subtract measures from the system, as well as add or subtract systems from the page.

The musical symbols for each staff can be scaled to one of four sizes. It's even possible to have different size staves in the same system. The upside of this technique is that you can pick a font size that best fits the look and density of your score. The downside is that you can't mix sizes within a single staff. When it's time to create a set of parts from the score, Encore can perform an automatic part extraction. But, the program doesn't create multiple rests in an extracted part.

Final Verdict. Passport seems to have created a notation program that will adapt itself to work the way you feel most comfortable. Some musicians compose on a sequencer and are looking for a notation program to transcribe the aural to the visual. Encore will do this. Other composers like to work with a pencil and paper, trying out new ideas and listening to how they sound. Encore would make this composer feel right at home. Others might want a program to take down their improvisations in realtime to create both notation and a MIDI file that can be tweaked until perfection. Encore will do this too.

Tim Tully has written a manual that is clear and concise. The tutorials explain most of the features that you'll need for just about any notational project. The user interface is slick, smooth, and logical. There are keyboard equivalents for just about any action. In short, Encore's got a lot going for it.

Yet, there are a few major obstacles that Passport needs to solve before this program can compete in the real world of professional notation. First and foremost is the problem with vertical alignment. This should be automatic. The time consuming task of realigning just about every note by hand is not attractive. A Passport representative has said that this problem would be addressed in the next update.

Next in importance is the ability to perform cross-staff beaming and allow beams to extend over barlines. Encore should also automatically create multiple rests in extracted parts. Last but not least, Encore needs to support double dotted notes and rests.

If Passport could fix these problems, this would be one heck of a program.

—Norm Weinberg

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And Digital Standards To All...

After Season's Greetings Come the Workstation Blues.

ROGER NICHOLS

CHRISTMAS IS just around the corner, and it is waiting to mug you, beat you senseless and take all of your hard earned cash. If you can't think of anything else to waste your money on, how about the quietest consoles known to man or multiple synchronized digital multitrack machines that will allow you to record so many tracks that it will be impossible to find a place to mix them?

And don't forget the digital audio workstation. You will be able to play with the newest, latest, whiz-bang digital audio workstations that are guaranteed to save you time and effort when editing your next project. Or...maybe not.

TIME SAVER?

Maybe it takes longer to do your editing on a hard disk-based editor. Consider this little scenario: You have decided to mix directly to your new hard disk-based digital audio workstation with a 650 megabyte hard disk so that you can store over an hour's worth of material.

Great, so now you take all of your new equipment into the studio to start mixing. Everything is going just fine. You finish mixing the first half of the album and then realize that you don't have enough room on your hard disk for the whole album. You forgot to allot space for the two different versions of one of the tunes or the TV mixes of all of the tunes. Boy, this stuff eats up hard disk space pretty quickly.

Well, you guess that you are going to have to off-load some of the tunes to make room for the ones that you haven't mixed yet. Your choices are to copy the mixes to a DAT machine or some other digital audio two track machine that has the appropriate digital interface, copy all of the data to a bunch of removable hard disks, copy everything to a removable optical disk drive, or back up onto 600 or so floppy disks. I think one of the questions they ask when you call the suicide hotline these days is "are you attempting to backup digital audio data onto floppy disks?"

This is just the beginning of your problems. After you have spent half your life getting your digital information in and out of your workstation and have managed to get your album edited and sequenced, how are you going to get it to mastering? A lot of mastering facilities have some sort of hard disk-based editing system, but I'll bet two tickets to the next Steely Dan concert that the hard disk-based system at Bernie Grundman's or Bob Ludwig's is not the same brand as the one that you bought. So once again you have to download your data to a format that can be used by the mastering facility. This time saving device has just cost you another hour or two. Was it all worth it? Only time will tell.

LIFE SAVER?

I think that digital audio workstations can save time. There are editing and sequencing tasks that can be done on a hard disk system that are simply impossible in a longitudinal system. I think that the bottleneck is in the storage media adopted by the system. Erasable optical disks seem to be the way to go. More and more manufacturers are turning to this type of media for storage. The problem is that the data storage is not compatible from system to system. You cannot take a disk written on one system and read it on another system. There are enough different sources of hard disk-based workstations that virtually every studio could own a system built by a different manufacturer.

> Standards have to be agreed upon to enable the transfer of data between systems, and from system to system.

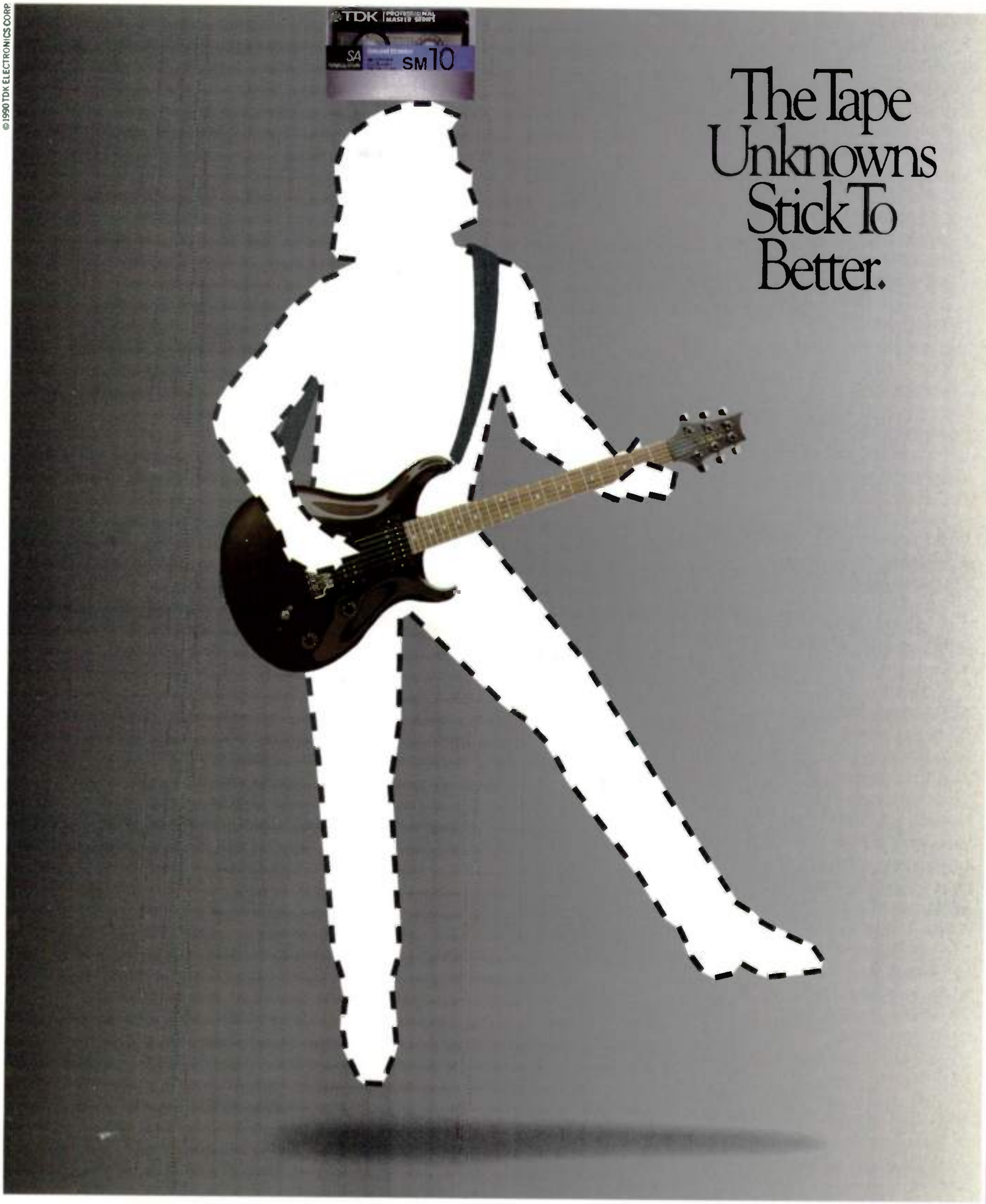
Let me assure you that I am not against digital audio workstations — quite the contrary. I have been using a hard disk-based editing system since 1981. I needed it so badly that I had to build it myself. It was based around an S-100 computer system with the Micropolis 32 megabyte 8-inch hard disk system and a digital interface to the 3M digital multitrack. It's name was Wendel. I used it on the Donald Fagen "Nightfly" album to do things that couldn't be accomplished any other way.

Digital audio workstations today are becoming even more valuable as production tools, but standards have to be agreed upon between manufacturers to not only enable the transfer of digital audio data between systems, but to transfer editing information from system to system.

In the land of video editing a CMX-compatible edit decision list is transportable between systems with excellent results. The same thing needs to happen in the land of digital audio workstations before everyone will have one.

So when you sit on Santa's lap and he asks you which digital audio workstation you want, tell him that Roger said that you should wait until you can perform some edits on your Digidesign, take the optical disc to your friend's Sonic Solutions to clean up some background noise and then go on to the mastering facility where they have a DAR SoundStation II for mastering. Now you've really got something.

Nichols *is chairman of the boards at Soundworks West in West Hollywood.*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