REVIEWS: ALESIS ADAT BRC • DYNACORD DRP-10 • D&R ORION CONSOLE

EQ

THE PROJECT RECORDING & SOUND MAGAZINE

JAN 1994

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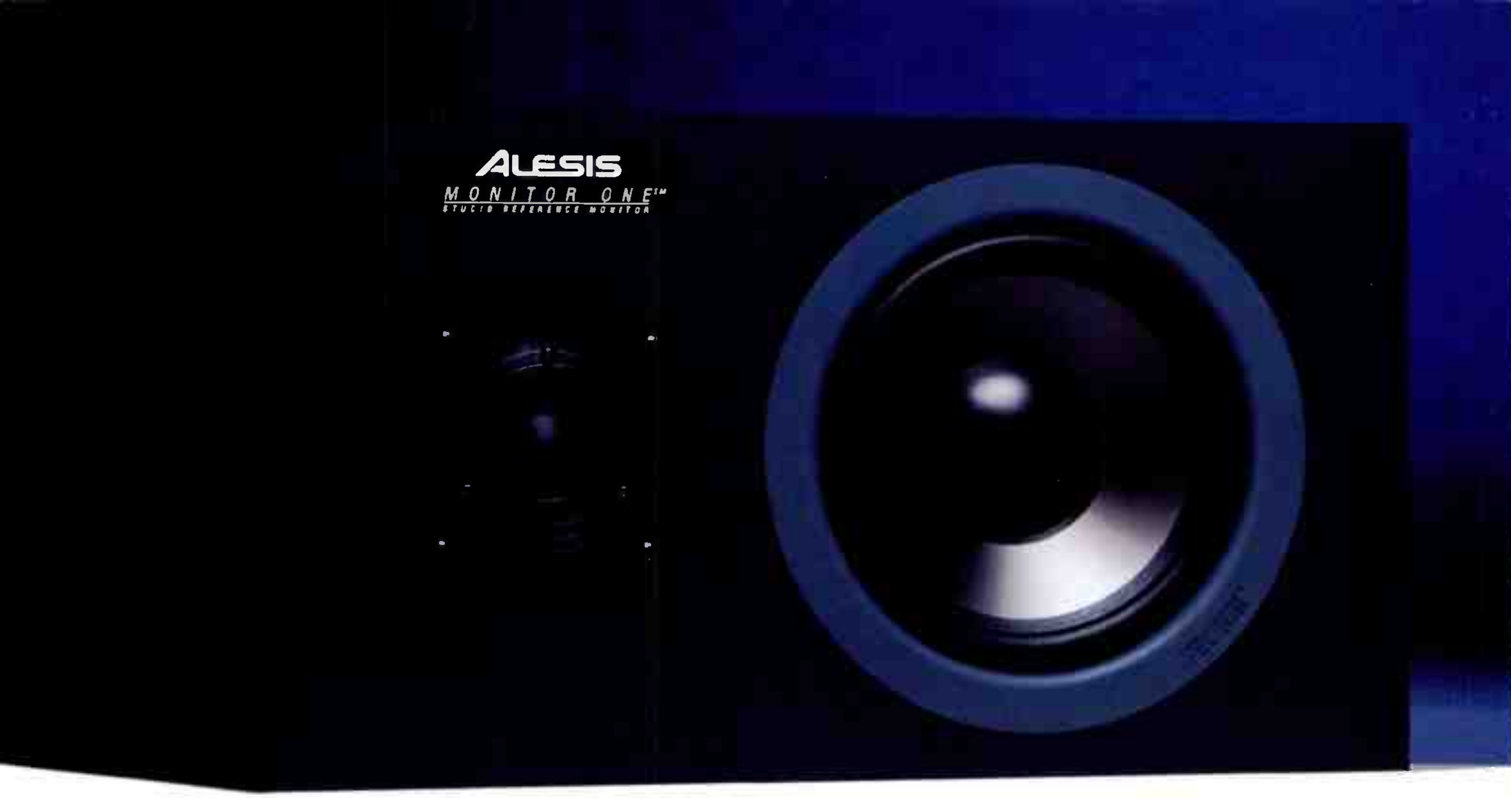
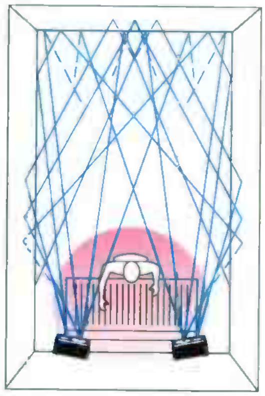


The Truth From

The truth...you can't expect to find it everywhere you look, or *listen*. But when mixing music, hearing the truth from your monitors will make the difference between success and failure. You'll get the truth from the **Alesis Monitor One™ Studio Reference Monitor**.

Room For Improvement

Fact: most real-world mixing rooms have severe acoustical defects. Typical home and project studios have parallel walls, floors and ceilings that reflect sound in every direction. These reflections can mislead you, making it impossible to create a mix that translates to other playback systems. Trying to solve the problem with acoustical treatments can cost megabucks and still might not work. But in the near field, where direct sound energy overpowers reflections, reverberant sound waves have little impact, as shown in the illustration. The Monitor One takes full advantage of this fact and is built from the ground up specifically for near field reference monitoring.



Does your living room double as your mixing suite? The pink area in the illustration shows where direct sound energy overpowers reflected waves in a typical mixing room. The Monitor One helps eliminate such complex acoustic problems by focusing direct sound energy toward the mixing position, instead of the love seat.

Working close to the sound solves the room problem but creates other problems, such as high frequency stridency and listener fatigue (typical of metal-dome and composite tweeter designs). Our proprietary soft-dome pure silk tweeter design not only solves these problems, but delivers pure, natural, incredibly accurate frequency response, even in the critical area near the crossover point (carefully chosen at 2500 Hz).

The Truth From Top To Bottom

The Monitor One gives you all the truth you want in the mids and highs, but what about the low end? You probably know that the inability to reproduce low frequencies is the most common problem with small monitors. Most of these speakers have a small vent whose effect at low frequencies is nullified by random turbulence, or they're sealed, which limits the amount of air the driver can move. Such speakers give disappointing results in their lowest octave.

The Monitor One overcomes wimpy, inaccurate bass response with our exclusive SuperPort™ speaker venting technology. The ingenious design formula of the SuperPort eliminates the choking effect of small diameter ports, typical in other speakers, enabling the Monitor One to deliver incomparable low frequency transient response in spite of its size.

Alesis SuperPort™ technology gives you the one thing that other small monitors can't: incredibly accurate bass transient response. No, the SuperPort doesn't have a blue light, but it makes the picture look cool.

The result? A fully integrated speaker *system* that has no competition in its class. You'll get mixes that sound punchier and translate better no matter what speakers are used for playback. Whether you mix for fun or for profit, you want people to hear what *you* hear in your mixes. The Monitor One's top-to-bottom design philosophy is a true breakthrough for the serious recording engineer.



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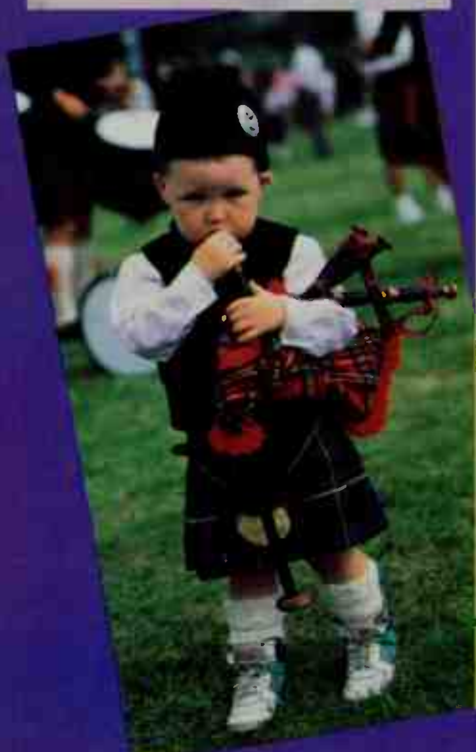

FEATURES

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BUGBUSTERS

Roger Nichols' comments (Dec. '93) regarding software-based problems are very important to all of us. Although I wish that he had used someone else's product as the target example, his points are well taken. The development and support of software-intensive products require a stronger cooperative effort between the supplier and user.

Flying Faders software goes through two stages of testing after it leaves R&D. It is subjected to testing by a customer-biased member of the AMS Neve Marketing team, then to a "beta" test where a controlled number of customers test the software. Beta testing is a probability exercise — when can testing be cut off with adequate confidence that the benefits far outstrip the undetected bugs? The customer's only hope is that the supplier has the resources and conviction to stick with beta testing long enough to reach a level of performance that will satisfy the customer. We are sure that Flying Faders has achieved its outstanding record of success due to its thorough beta testing.

As part of the continuing software improvement process, the customer must strive to provide clear and accurate descriptions of bugs. Since we find that many situations are the result of lack of operator familiarity or training, our first effort must be to separate bugs from "finger trouble." Then the customer must accurately describe the environment, actions and enhancements. If the system doesn't do what they expect, the lack of a feature is reported as a bug!

Roger's suggestion of a clearinghouse for bug reports, workarounds, and fixes is why we set up a Bulletin Board earlier this year (+44-282-415551). Known bug lists have always been distributed with Flying Fader releases, and will be distributed with our other product releases from now on.

However, we must also recognize that not all computer crashes are the result of operator or software failures. All computers are vulnerable to factors such as power line glitches, alpha ray hits on memory, and data corruption on storage devices. (At a processing rate of 20 million bits per second, even an insignificant error rate of one error per trillion bits will occur twice a day!) The resulting errors may appear to be software related because the defect is only made noticeable when the code accesses the damaged information.

Because we cannot completely eliminate these problems, we are evolving our strategy to try and minimize the damage done by these "Acts of God," which lurk beyond the best beta test procedures and reasonably reliable hardware. For instance, the Capricorn automation system automatically backs up all the mix data to hard disk without the engineer needing to initiate the process. A similar system is being actively tried for Flying Faders.

The desire and ability of a supplier to continue to evolve a product should be a major criteria when selecting between potential suppliers. Gone are the days when the customer could assume that purchasing a product implied free software upgrades forever. The annual expense to plan, implement, test, document and support Flying Faders software enhancements is approximately equal to the total cost of developing the Flying Faders hardware for the Neve V-Series console!

Isn't an investment of perhaps $1,000 a year to obtain a better than new performance on a $50,000 to $100,000 system a bargain? I would certainly jump at a deal like that from my car dealer — 1 to 2 percent to completely avoid depreciation of utility and even add some new features!

But what about the customer who chooses not to stay current by upgrading? I can't go back to my Ford dealer and complain that my 1991 model car doesn't have air bags. And Microsoft won't upgrade my DOS or Windows free of charge. Do we try to apply a different standard to our professional audio products? Is this standard fair and reasonable? Our belief is that we need to enter into these "software relationships" for the long haul, not just a one night stand.

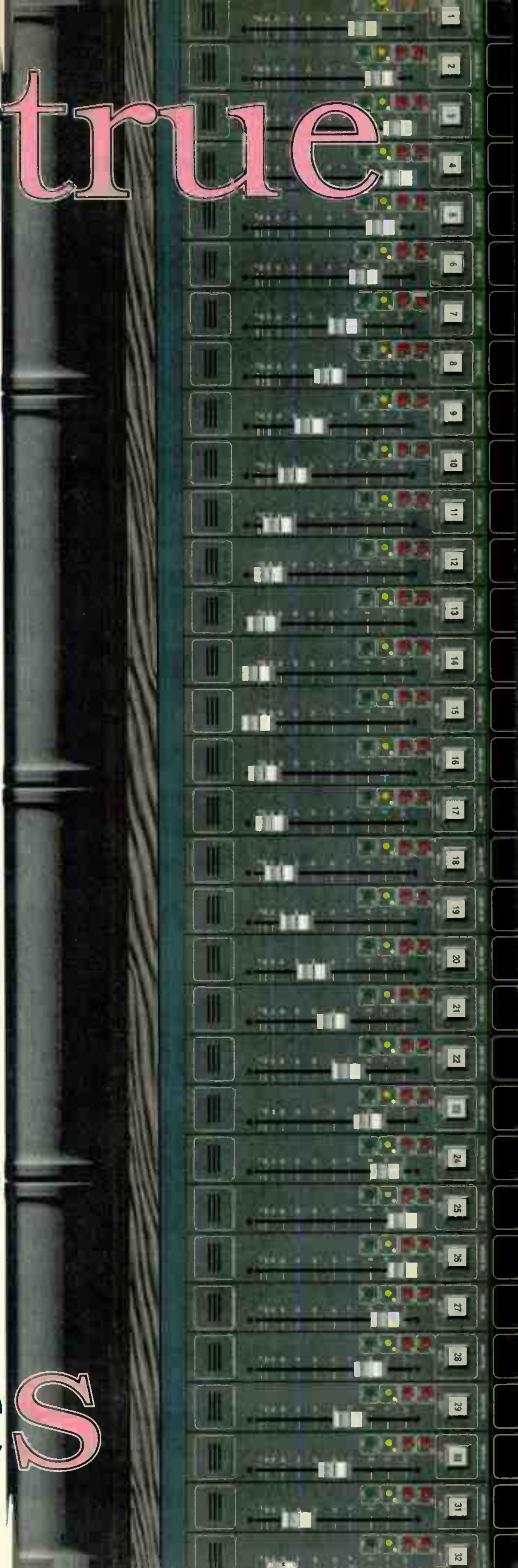
Thank you, Roger, for launching this topic. The software support debate is probably as important as debating whether transistors can ever sound as good as tubes! I just hope the answers are more tangible and universally acceptable.

Antony P. Norris
Software QA Manager
AMS Neve Plc

WRITE TO US

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

Supertrue



Zappa: The Innovator

In the October issue of *EQ*, the Father of Invention himself, Frank Zappa, took a few moments to sit down for a photo session in his home studio, the Utility Muffin Research Kitchen. Unfortunately, these photos were some of the last ever taken for the public and Zappa's devoted legion of fans. His gracious invitation to bring *EQ*'s readers into his legendary home studio demonstrates Zappa's unyielding commitment to his peers in sound recording.

Frank Zappa's passing signals the end of a musical era rooted in satire, versatility, and technological brilliance. Truly uninhibited, he never shied away from mockery when it came to the music business (*We're Only In It For The Money*, 1967), politics (*FZ Meets the Mothers of Prevention*, 1985), or anything else for that matter. When it came to his heroes, however, Zappa was never more laudatory in his pursuits. Up until the end, Zappa was working on *The Rage and the Fury — The Music of Edgard Varèse*, a tribute to his original inspiration, Edgard Varèse.

As a master of modern technology, Zappa pushed his home studio to the limits of record production, racking up countless accomplishments in the realms of digital audio and recording technology. He was a frequent AES attendee and he was one of the earliest adherents of digital multitrack recording. Recording technique was an integral part of all his musical creations — so much so that Zappa built the largest Synclavier system in the world and devoted much of his later work to this technology

Look for an in-depth historical feature on the legendary Frank Zappa in an upcoming issue of *EQ* Magazine.

EQ&A

SNAKE, RATTLE, & ROLL

Q *Is there a possibility of crosstalk among the low-, mid-, and high-frequency signals in a triamped system using a 100-foot speaker snake consisting of six conductors (3 pairs; each conductor is 12-gauge stranded copper) wrapped tightly together? The snake carries 800, 400, and 200 watts for the low-, mid-, and high-frequency signals, respectively.*

Chuck Carlley
Jackson, MS

A In the audio bandwidth (20 Hz – 20 kHz), crosstalk will not be a problem using the speaker snake in your system. Crosstalk is induced either electromagnetically (inductive coupling) or electrostatically (capacitive coupling). The wire and load impedance are low enough that neither mechanism will create a problem. When multiple cables carrying different signals are run together, each conductor pair should be twisted.

Bill Gelow
Director Systems Engineering
JBL Professional

A QUESTION OF COMPATIBILITY

Q *Now that Fostex has released its RD-8 ADAT-compatible recorder, I'm curious as to how interchangeable tapes will be, as I've noticed that in addition to 48 kHz, the RD-8 also has a 44.1 kHz sampling frequency.*

Raphael Septien
Austin, TX

A The Fostex RD-8 offers both 48 kHz and 44.1 kHz sampling frequencies with ±6 percent varispeed, and as such, has two separate clock rates. While the Alesis ADAT primarily records at 48 kHz, later software versions provide a 44.1 kHz sampling frequency accessible by means of the recorder's pitch control capability. The Alesis ADAT accomplishes this 44.1 sampling rate by means of a variable clock rate.

Tapes are fully interchangeable between recorders at either sampling frequency, thus enabling users to work on either machine or with any combination of units. Of particular interest, if the RD-8 is master to a series of Alesis ADAT recorders (Version 3.05) and is playing back a tape at 44.1 kHz, all slave units will "pull down" to the 44.1 pitch control setting automatically. Multiple recorders are connected by simply daisy-chaining the units via the sync ports located on the rear panels of either model.

Roger Maycock
Senior Product Specialist
Fostex

CONVERSION QUERY

Q *Please explain the differences between the Burr-Brown 1700 (PCM1700U/P) and the Analog Devices DA1865N (AD1865) digital-to-analog converters. Also, does the higher model number on the Crystal 5539 mean it's a more advanced converter than the 5536, and if so, why? What's the cost of these units?*

David Nichols
Phoenix, AZ

A The PCM1700U/P and the AD1865 are essentially the same. The Analog Devices part runs with a lower maximum power supply voltage (±6 V vs. 7.5 V, and thus only dissipates 260 mw vs. 300 mw). The PCM 1700 is specified at 108 dB signal-to-noise and the AD1865 at 110 dB. Very close indeed.

The answer to the second part of your question is no. Despite the different model numbers, the Crystal 5339 and 5336 converters are also essentially the same. The 5339 has an "SCLK" that clocks out data on the falling edge, while the 5336 has an "SCLK" that clocks out data on the rising edge. Another difference is that the extended temperature range devices of the 5336 family fall out of spec compared to the consumer-grade 5339. Both the S/N and the Passband range are reduced on the extended temperature versions of the 5336. The 5339 only comes in a 0- to 70-degree range. The cost of any converters in question is based on a function of quantity. One or two may actually be had for free as manufacturers samples, but the final price for any real quantity must be negotiated.

Greg Hanks
New York Technical Support, Ltd.
Chappaqua, NY

SOUND BARRIER

Q *My band plays in clubs (100–400 people) and we're caught in a dilemma. As soon as we get the excitement and energy going, the club owner tells us to turn down. When we turn down, our sound falls apart. We'd like to make the club owners happy, but artistic fulfillment is supposed to be part of the deal, too. Does there have to be a compromise between the two extremes?*

Robert Atkinson
New York, NY

A Not only are there financial aspects to your question (annoy the club owner and you don't get hired back), there are also legal considerations. In this litigious society, lawsuits have been instigated by music fans claiming permanent hearing damage. So don't think the club owner is on the wrong side of the "if it's too loud" equation.

The solution has three elements: speaker positioning and aiming; stage-equipment volume; and timbral balance. Determine if the excessive level is localized (just a few tables near the stage). If your main "front of house" (FOH) speakers are positioned with the high-frequency drivers at ear level for those seated at the front tables, we have diagnosed the problem. The prescription is elevation of your FOH speakers. Ideally, these speakers should be raised and angled down so the high-frequency horn is aimed at a point about two-thirds of the way to the rear of the audience. By so doing, the near tables or dance floor are now farther from the FOH speakers, and therefore the audience in the back of the room will actually hear better. In other words, uniform coverage has been achieved. In practice, this requires some expertise in rigging.

Moving on to stage-equipment volume — when your rock-god hero stands in front of stacks of guitar speakers in a stadium, the amount of sound energy from those stacks reaching each of the 10,000 members of the audience is far less than the energy reaching each member of your club audience from your smaller rig. In large-concert sound, very little of the direct sound from the onstage equipment is heard by the audience — most of it comes from the PA system. In a club, the situation is very different. Direct stage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sound is a major part of the total sound in the room. If the guitar amp or the bass amp or the drum sound coming directly from the stage is too loud, the sound engineer has two choices: bring everything else up to balance, or live with a badly mixed show. The solution is to bring down the onstage volume. Keyboard players and bassists can usually simply turn down their stage rigs.

Guitarists and drummers may need to adjust their technique and equipment to play at lower levels. The guitar player may be able to get an acceptable sound using signal processors and a smaller amplifier, or by acoustically isolating the guitar speaker and miking it. (I've known guitar players who left their speaker cabinet in the road case with a microphone.) If your drummer finds it impossible to alter his technique you can consider putting up "gobos." These are portable low walls that are placed around the drum set. Depending on how attractive your drummer is, these could be made of Plexiglass or some acoustically (and visually) opaque material. The effectiveness of gobos is minimal, but every dB of reduction helps. A better alternative is electronic drums and trigger pads. Since an all-electronic set has virtually no acoustical output, it can be easily balanced with the band no matter how aggressively the drummer plays.

An added benefit of reducing stage volume is that achieving adequate monitor levels will be easier (a note of caution: stage monitors that are too loud can spill into the audience and make the original problem worse), plus you'll still be able to hear when the cutting-edge music you're playing now can only be heard on the oldies-but-goodies stations.

Thus far, I've addressed the how-to-turn-it-down part of your question. Now let's turn to keeping the excitement and energy. Back in the 1930s, two Bell Lab scientists named Fletcher and Munson determined that the human ear perceives changes in sound levels differently depending on frequency (for extra credit, look up "equal loudness contours" in the *Yamaha Sound Reinforcement Handbook*, p. 29). As you reduce overall level, you will have to boost low frequencies (below 100 Hz) and — to a lesser extent, high frequencies (above 6 kHz) — just to keep the *perceived* tonal balance the same (this is exactly what the loudness switch on your stereo does). These very low frequencies are pleasing to the human ear and convey a sense of power (not to mention that all-important, primal kick-drum beat). Consider adding a subwoofer system to your PA. Subwoofers can maintain the energy and excitement you're looking for, but won't be perceived as being obnoxiously loud by your audience or by the guy who signs the check at the end of the night.

Gerald Tschetter
Division Marketing Manager
Yamaha

This is where your questions get answered. Send your query with your name and address to:
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Fax: 516-767-1745

EQ TIPS

EDITED BY DAVE BRODY

AGE OF THE HYBRID STUDIO

Don't chuck your analog multitrack recorder just yet! The big news from the big studios is "hybrid-multi." If they can do it, you can do it. And there're some good reasons why you should.

Back in the music school days (between endless choruses of "Giant Steps"), my old composition teacher bestowed upon us the wisdom of the ages. "Folks," he said, "the definition of orchestration is nothing more than each instrument doing what it does best." This is equally true of studio production — the modern extension of classical orchestration. You choose mics, speakers, and outboard gear for their specific sonics; now do the same for your storage format.

Let's face it, beyond the arguments about which is better, analog and digital simply sound different. Many mothership studios are locking digital and analog machines and making "sound" decisions about what gets recorded where. It's becoming fairly common to record the bass and the drum kit — sometimes sans cymbals — on the analog deck, while vocals, pad-synths, winds, high percussion, and so forth go to the digital. D/A conversion imparts that thin, crystalline, clear, sometimes sizzly top (especially at sampling rates below 50 kHz and architectures of only 16 bits). The good old analog deck rounds and thickens things, particularly affecting low frequencies (they seem to be fatter and "ring" longer). This is primarily due to "head-bump" (which will remind you of the proximity effect you hear with your cardioid mics).

Rock engineers have gotten quite good at knowing when to slam the tape with level to incur "tape compression" — a kind of limiting whose sound can probably be best described as "the world's softest brick wall." You've heard this technique on records for years, and the new generation of analog tape makes it possible to do it without hearing unacceptable hiss in the quiet breaks. Noise reduction systems of the sort used on analog multitracks have their own characteristic sounds; you'll probably find their effects most musically useful on low-frequency material.

Driving this trend toward "hybrid-multi" is the business need to keep older analog equipment productive — a need you and I share with that big studio downtown. The problem is that many of our project studios do not contain analog equipment that is capable of being externally synchronized (or for which doing so would be prohibitively expensive).

One solution is to record the analog-appropriate material first and fly it over to your new digital multitrack. This allows you the creative option of doing a submix on transfer, the way record dates were done in the '60s. In more recent hit-record history, Tonemeister Bruce Swedien has been known to record drums on a 16-track analog machine and bounce the completed tracks to 32-track digital, thus forever "freezing" the good contribution of analog in a medium that won't degrade with each pass over the heads.

If you're working with sequencers, you would stripe one track of the analog multi with timecode, record, say, your drum and bass parts, and fly the result to the digital while regenerating timecode. Another option is to record anything you wish to have "analogized" on tape, sample it, and fly it back to the digital — possibly in the digital domain, if you can work around the competing I/O standards. This brings up the possibility of playing games for feel. A variation on this theme would be to employ a multitrack digital workstation — even if you don't own one, you probably know someone who does — so you could slip-align tracks against each other. However you do it, it's a good tool to keep in your pallet. EQ

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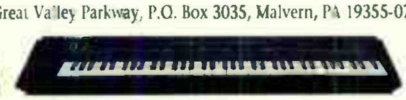

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KS-32 Weighted Action MIDI Studio

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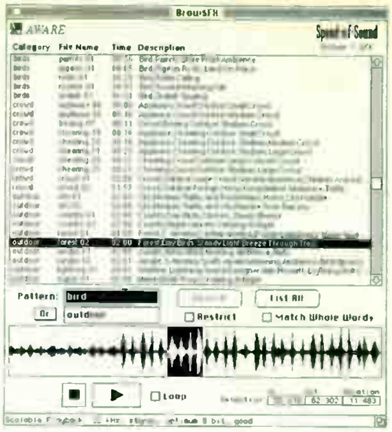
SQ-R PLUS 32 Voice Synthesizer Module

EQ NAMM VIEWS

STAY AWARE

The MegaDisc CD-ROM from Aware, Inc., provides multimedia users with over seven hours of audio. It doesn't require any special audio hardware or CD-ROM drive configuration. The Aware *Speed-of-Sound Library, Volume I: SFX* is now available in an interoperable version that is compatible with both Macintosh computers and Silicon Graphics workstations. Aware *Speed-of-Sound* is a complete library-on-a-disc suitable for professional audio applications. Everything you need is built into the disc itself. For more info, contact Aware, Inc., One Memorial Drive, Cambridge, MA 02142-1301. Tel: 800-292-7346. Circle EQ free lit. #101.



The *Aware Speed-of-Sound* ™ Library, Volume I: SFX. One CD-ROM contains over SEVEN HOURS of top quality digital sound effects—plus built-in *BrowSFX*™ interactive librarian that makes finding and auditioning any sound a snap!

3-2-1 BLAST-OFF

N.E.I.'s new Model 321 electronic crossover is an upgraded version of the popular 321. The device now incorporates ultraquiet ICs and sturdy, soft-touch switches. The open PCB design is serviced with off-the-shelf components. The 321 can be configured in two-way stereo or three-way mono. All ins and outs have balanced XLR and unbalanced 1/4-inch phone jacks. There are high-frequency polarity switches and a rotary frequency-selector knob with a times-10 multiplier switch. The 321 also features smooth Butterworth filters for minimal pass and ripple. Frequency response is 20 Hz–20 kHz and signal-to-noise is greater than 90 dB. Retail price is $319. For further details, contact N.E.I., BXG International, 5337 N. Marine Drive, Portland, OR 97203. Tel: 503-289-3182. Circle EQ free lit. #102.

SONUS MAXIMUS

Following on the success of BBE Sound's previous BBE Process products, the company has introduced two new units: the BBE 461 and BBE 362 Sonic Maximizers. The new BBE II Process reduces noise level. Signal-to-noise ratio for the 461 and 362 is –90 dBu. In addition to lower noise specs, the process sensitivity has been greatly increased, making these models more responsive to low signal levels. The BBE 461 features +18 dBu headroom and a bypass switch for comparisons between processed and unprocessed signals. The BBE 362 features ganged definition and bass controls, +15 dBu headroom, and a bypass switch. For further data, contact BBE Sound, 5500 Bolsa Avenue, Suite 245, Huntington Beach, CA 92649. Tel: 714-897-6766. Circle EQ free lit. #104.

NAMM INSIDER

Yamaha will be introducing the H Series of power amps, which include a Dual Mono mode. You still get stereo and bridged mono modes, but now two sets of speakers may be driven by the same signal. The three models are the H3000, H5000, and H7000 and provide 350, 550, and 750 watts per channel into 8 ohms, respectively. Also new from Yamaha is the A100a power amp. This unit provides 50 watts of continuous power into 8 ohms with less than .2% THD...**Passport Designs** has just unleashed Passport Producer Pro for Windows. Already popular on the Macintosh platform, Producer Pro offers multimedia producers interactivity, path-based animation, external device control, video support, extensive graphics and text capabilities, a variety of video clips and background textures, Photo CD images, and digital audio and MIDI files. Like the Macintosh version, Producer Pro for Windows is a time-based production tool that combines and precisely synchronizes animation, video, sound, music, and presentation graphics in a real-time desktop environment...**SoundTrek** has introduced the Jammer for Windows. The Jammer does not provide "canned" or prerecorded parts, but rather combines music theory, artificial intelligence, and randomness to create new musical parts every time you compose. Jammer for Windows supports all of the features and functions found in the DOS version plus adds many new features. These include graphic animation, an array of colorful and artistic toolbars and icons, support for polyrhythms, new and improved graphic views of measures, and easy chord entry via mouse or keyboard...**Generalmusic** is offering the WX2 and WX400 multimedia keyboard workstations. The 61-key WX2 and 88-key weighted piano action WX400 are full-featured professional music workstations. The WX keyboards can function as compositional tools, using 96 built-in existing patterns with accompanying instruments called Styles. The Styles can be edited, combined or looped and stored as a new pattern or style. The WX workstations can display the melody notes, chords and lyrics of a song...**Korg** will be introducing new multieffects systems at the NAMM Show. The G1 guitar distortion processor houses eight internal effects and features newly-patented hybrid gain structure technology, improved overdrive and distortion, fuzz, pedal control input for wah-wah effects, delay, EQ, speaker simulation, noise reduction, guitar amp or direct line recording outputs, nine user programs, compare function, and headphone out. Also new is the G4 Rotary speaker simulator.

OPEN FORUM

DDA's Forum consoles come in three versions for various applications. The PA is designed for live applications, featuring eight group outputs, while the Matrix offers an 8 x 8 matrix with left and right return on each output. The Composer is intended primarily for 24-track recording applications. All have a switchable 80 Hz high-pass filter with 12 dB per octave low-frequency roll-off that reduces proximity effects and eliminates rumble and other low-frequency noise. The 4-band equalizers feature sweepable mid frequencies that range from 470 Hz to 15 kHz for upper mids and from 70 Hz to 2.2 kHz for lower mids. Aux sends have Aux 1 and 2 switchable to Pre or Post modes. All Aux sends are also internally selectable as pre-EQ/prefader, post-EQ/prefader, or post-EQ/postfader. Aux 1 can be converted to a direct output with level control for separate effects sends or multitrack feeds. All aux outputs are balanced. Balanced group outputs are also featured. Each channel has five-segment LED metering from –13 dB to 17 dB, allowing levels to be determined at a glance. Alps controls and faders are incorporated for precision control. Three modules are available with the Forum: a stereo input module, a digital stereo input module, and a six-into-one mic input module. For more information, contact: Mark IV Pro Audio Group, 448 Post Road, Buchanan, MI 49107. Tel: 800-695-1010. Circle EQ free lit. #105.

HARNESS THE POWER

The new Hot House M500 and stereo S400 amplifiers employ the same technological advancements found in Hot House's new M500HV 600-watt mono-block amplifier. Refinements in front-end design, coupled with a proprietary application of minimal local feedback, have improved the line's sound, particularly in the areas of detail resolution and transient capability. Slew rate is up to 85 volts/microsecond, while rise time is 900 nanoseconds. Istantaneous current delivery is assisted by custom-built low ESR filter caps. Due to the absence of global feedback, the mono blocks can be linked two per side in parallel pairs as well as series bridged pairs, offering a total of eight stereo combinations from 400 to 3600 watts at impedances down to 1/2 ohm. For more information, contact Hot House, 275 Martin Ave., Highland, NY 12528. Tel: 914-691-6077. Circle EQ free lit. #106.

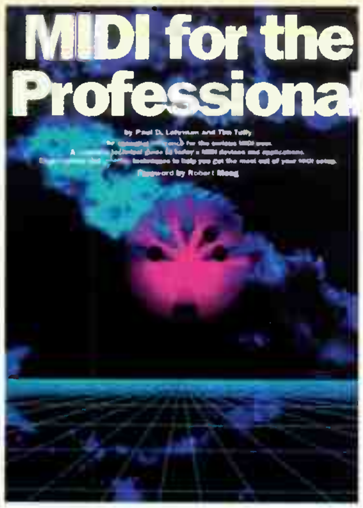
FOR THE LOVE OF MIC

Stedman has introduced the N90, a side-address, dynamic mic designed for critical recording and live-sound applications. It employs a precision low-mass mylar diaphragm, coupled with a large magnet assembly. This dynamic design yields optimum transient and frequency response without the need for phantom power or batteries. The N90 body is machined from massive 1/4-inch aluminum and the windscreen is fashioned of stainless steel. Frequency response is 35 Hz–19 kHz, sensitivity is –57 dB, and impedance is 250 ohms. The N90 features a cardioid pickup pattern. It comes with a standard 3-pin XLR connector. For more information, contact Stedman Corporation, 4167 Stedman Drive, Richland, MI 49083. Tel: 616-629-5930. Circle EQ free lit. #107.

WINNING HAND

The Solitaire console from Soundtracs is a 24-bus in-line production console with the latest Soundtracs ADP dynamics package, providing gates, compression, limiting, expansion, modulation, and autopanning on each of 24, 32, or 40 channels. An automation option of either VCA or moving fader is being made available. With dual-line inputs on each channel, the Solitaire provides 88 comprehensive inputs on mixdown, combined with automation and dynamics. Prices start at around $17,000. For more information on the Solitaire console, contact Soundtracs, Samson Technologies, P.O. Box 9068, Hicksville, NY 11801. Tel: 516-932-3810. Circle EQ free lit. #108.

EQ NAMM VIEWS



MIDI BOUND

MIDI for the Professional has just been released by Music Sales Corporation. Written by *EQ*-contributors Paul D. Lehrman and Tim Tully, it is a reference and sourcebook for the serious MIDI user. It's filled with in-depth technical information, expert advice and techniques, and practical examples and illustrations to help you get the most out of your MIDI equipment. The book's foreword was contributed by Robert Moog. For more information or to place an order for *MIDI for the Professional*, call 800-431-7187. Circle EQ free lit. #109.

MILLENNIA POWER

The HV-3 high-voltage microphone preamplifier from Millennia Media utilizes an octal-matched transistor hybrid front end and 80 V FET-based DC monolithic output stage for optimum sonic integrity. The HV-3 has a high-headroom, high-speed signal path design. It provides front-panel mic selection (standard or high voltage), phantom power, gain control, and overload indication. LED status indicators are provided for all DC power rails. Balanced output stages employ DC-coupled FET-based drivers. THD + noise is guaranteed to be less than .0007 percent over all audio frequencies and typically less than .0005 percent. An installed A/D converter will be available in late 1994. For more information, contact Millennia Media, P.O. Box 277611, Sacramento, CA 95827. Tel: 916-363-1096. Circle EQ free lit. #110.

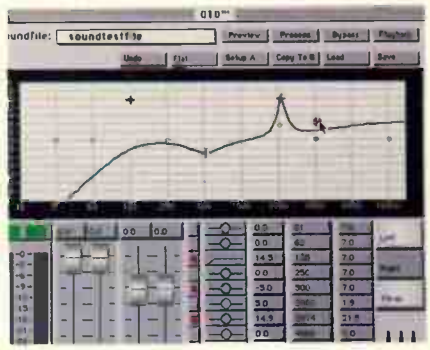
ISOLATIONIST

Continuing its application of acoustic technology to headphone design, Sennheiser has introduced the HD 25SP studio monitoring headphone. Utilizing dynamic drivers in a closed supra-aural design, the HD 25SP offers a lightweight and comfortable headphone for users who spend a great deal of time monitoring through phones and need maximum isolation. The low-impedance, high-sensitivity drivers are efficient, providing sufficient volume even in applications that may formerly have required an additional headphone amplifier. For more info on this and the rest of the company's pro headphones, contact Sennheiser, 6 Vista Drive, Old Lyme, CT 06371. Tel: 203-434-9190. Circle EQ free lit. #111.

HOT MOD

Kurzweil's new MicroPiano is a half-rack sound module with high-quality sampled keyboard sounds and built-in effects. The MicroPiano contains Kurzweil's new Grand Piano sample, plus Ensemble Strings, Hammond Organ, and various electric and electronic pianos. Thirty-two presets are offered. Retail price is $499. For additional info, contact Kurzweil, Young Chang America, 13336 Alondra Blvd., Cerritos, CA 90701. Tel: 310-926-3200. Circle EQ free lit. #112.

CATCH A WAVE



Waves Ltd., has released Q10, a software plug-in for Digidesign recording and editing systems. Q10 gives users of Pro Tools, Sound Tools II, ProMaster 20, and Audiomedia II systems ten bands of high-quality, fully parametric IIR filtering, including high and low shelving and high/low-pass filtering. All parameters can be directly controlled by dragging the equalization curve or by adjusting the displayed digital values. A version for Sound Tools I will soon be made available. List price is $399.95. For more information, contact Rockwell Digital, Santa Monica, CA 90404. Tel: 800-941-7000. Circle EQ free lit. #113.

EQ NAMM VIEWS

ONE FOUR ALL

Fostex recently premiered its top-of-the-line 380S 4-track cassette recorder/multitrack, multifunction mixing console combination. It features 12 inputs and incorporates double-speed tape capabilities that, combined with Dolby S, ensure quality analog sonic performance within the format. The 380S also features sweepable midrange EQ. List price is $995. For more info, contact Fostex, 15431 Blackburn Ave., Norwalk, CA 90650. Tel: 310-921-1112. Circle EQ free lit.#114.

PATCH THINGS UP

The Akai DP88 digital signal patchbay allows you to patch any combination of 10 digital inputs and outputs, and you can store these in up to 128 memory locations. The DP88 is a 2U-space rack-mountable unit that provides eight XLR and two optical ins and outs. You can therefore patch between electrical and optical equipment. Also, one XLR in/out pair and one optical in/out pair are located on the front panel, making it easy to patch additional devices even when the DP88 is rack-mounted. The DP88 is fully programmable. Up to 128 patch configurations can be stored in its memory, and programs can be selected from the front panel, or via foot pedal, or by remote MIDI program changes. Suggested retail is $1095. For further information, contact Akai, 1316 East Lancaster, Fort Worth, TX 76102. Tel: 817-336-5114. Circle EQ free lit. #115.

WE CAN SHOW YOU ALL THE FEATURES THAT SET OUR NEW MPL 2242 APART EXCEPT ONE.

- 22 inputs. 10 XLR balanced inputs featuring a low noise, padless preamp design. Six true-stereo inputs.
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- The only 4 buss design in its class (using 4 dedicated sub group faders) with discrete Group and Main summing outputs.

SOUND.

SAMSON AUDIO

For more information about the MPL 2242, please contact Samson Audio, a division of Samson Technologies Corp.,
P.O. Box 9068, Hicksville, NY 11802-9068 (516) 932-3810 FAX (516) 932-3815

CIRCLE 74 ON FREE INFO CARD

Let's Make A Deal

An independent pro audio dealer makes a case for protecting this endangered industry resource

BY J.D. SHARP

Purveyors of audio and musical equipment often feel that they are caught between the proverbial rock and a hard place. Both the end user and the manufacturer have been known to perceive the dealer as an obstacle that stands between them, rather than as a facilitator who connects the customer's need with the appropriate manufacturer's products.

For the manufacturer, this translates, in some cases, into a strategy that calls for opening the maximum number of dealerships that the planet can sustain. The concept is that dealers will engage in insane cutthroat competition and drive the retail price lower, thus making the product more affordable to the consumer, and increasing sales. All this (theoretically) improves the bottom line, because if more sales volume can be achieved, there are economies of scale to be realized.

The consumer sits on the other side of this equation: the primary concern, in the majority of instances, is that he or she acquire the desired product at the lowest possible price. What if you lose a few dealerships in the process? Hey, those are the breaks.

But I'm getting ahead of myself. It's time to take a few steps backward and take a look at the psychic makeup of the typical independent audio and/or musical equipment dealer (we'll take a look at the large-scale operations next). The first fact to keep in mind is that very few store owners got into the business [strictly] as a money-making proposition. There are a few reasons why this is true.

First, most ambitious people would not select the music and audio retailing industry as their first choice for amassing a fortune. Even though

MONTY HALL knew the art of the deal. Wouldn't it be great if you could pick a console from behind curtain one, two, or three?

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When factory presets alone are just not enough. When you know it's time to exercise more control over the sound of your music. Move up to **MIDIVERB®III.** The powerful simultaneous multi-effects processor that's the latest incarnation of the award winning* MIDIVERB from Alesis.

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ALESIS
STUDIO ELECTRONICS

CIRCLE 05 ON FREE INFO CARD

music is universal and pervades our everyday life in every situation; the fact is that playing an instrument, performing, and recording are pretty far down in most folks' priority schemes. So the audio and musical specialist is inherently addressing a limited market, which is one strike against accumulating the aforementioned fortune.

The second fact is that most owners have become music or audio store proprietors out of their love of the industry and involvement with its products. Virtually every one of them has spent time either as a member of a band, the operator of a sound-reinforcement company, or the owner of a studio. The retailing aspect usually develops as a way to make ends meet. So in the big picture you're not likely to find a less mercenary bunch than the heads of music and audio operations. This also explains why some operations seem poorly managed.

The problem for your dealer is that as time goes on, fewer people understand that there is a cost involved in dealing with them, and that profit is far-from being a dirty word.

THE BIG GUYS

The above remarks apply to that endangered breed, the independent dealer. As is the case with so many other industries, the trend in music is inexorably toward large, multistore, multicity operations — and it is pressure from both consumers and manufacturers that make it so. The manufacturers, with some notable exceptions, heed the siren call of cash on the barrelhead, even if it means cutting a special deal with a big chain behind the backs of their loyal independent dealers. It's not hard to empathize with their point of view: they get to move a whole bunch of stuff in one bulk transaction with a minimum of paperwork and fuss, and rather than financing the purchase (which burdens them with borrowing costs), they get a check up front. And it's certainly not hard to see what motivates the chains: they get a lock on the marketplace, at least for that particular product. The only potential victim in all this is the consumer, who might end up buying a product not because it is the best for a particular need, but rather because a corporate giant has put the weight of investment into it and instructed the sales staff to move the product or suffer the consequences.

Naturally we all want to purchase the right thing, so the intelligent consumer goes out and shops. Shopping can take on several forms, including reading magazines like this one, talking to friends and experts, and visiting retail stores. Many consumers have come to accept it as a given that they can go down to their local emporium, ask a series of questions, perhaps listen to the product, maybe purchase it and take it home for a night — and then still be free to return it. An increasing number of these consumers also believe that it's a basic American right to take their new-found knowledge and then shop around for the very best price. There's no particular connection made between the time the dealer has spent with the customer and the fact that the only benefit in all this for the dealer is to "close the sale." After all, it is profit, plain and simple, that makes this all possible.

THE PRICE TO PAY

The problem for your local beleaguered dealer is that as time goes on, fewer people understand that there is a cost involved in dealing with them, and that profit, far from being a dirty word, is the fuel that drives the whole process. Dealers, typically, believing in the righteousness of their clientele, think that right-thinking customers will reward them by purchasing the product from them, even at a higher price, when they are offered the right information and the sale is backed up with meaningful support (like swapping defective equipment over the counter should it fail in the first 30 days). In other words, the up-front info and after-the-sale support are part of the deal, and the deal is more important than the price. Consumers, typically, increasingly accept the information offered to them and think noth-

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CIRCLE 77 ON FREE INFO CARD

ing of shopping the price across town or across the nation. Their orientation is toward the price and not the overall "deal," and any fleeting sense of moral obligation went down the drain with Michael Miliken and the excesses of the '80s.

Let's follow this scenario to its logical conclusion. If price is all that matters, then organizations that can provide the lowest price will predominate. To offer the lowest price, it's necessary to reduce overhead to a minimum and maximize volume. To reduce overhead you have to pay lower wages and offer fewer services, plain and simple. To maximize volume you need to concentrate on fewer products and purchase more of them at one time. Where does this leave the consumer of the future?

First, the sales personnel that will be encountered will be far less informed than at present, because the wage scale will be lower. Anybody with intelligence and communication skills will seek commensurate compensation in another industry, leaving only clerks behind the counter. In fact, some catalog operations show you a picture of their lovely headset-wearing receptionist, ready to take your order. It doesn't take a sexist fanatic to realize that she wasn't hired to provide you with the keys to the inner mysteries of MIDI. You're supposed to have formulated your decision from a single paragraph of product description and a 1-inch-square color picture (or better yet, by visiting your local music store).

Second, the luxury of choice will ebb away; instead you'll be offered only the choices that fit into this mass-marketing scheme. This also means that products that require more explanation and sophistication to understand will either disappear or become very expensive, as only a handful of specialists will be able to support them. If you don't believe it can happen, go to your local warehouse club operation. Ask one of the folks driving the forklift around for some pointers on how to operate the VCR or computer on the shelf in front of you.

All of this is on the mind of your local merchant when you walk through the door or call on the phone. Aside from the pressures generated by epidemic price-shopping, dealers are also in a squeeze from the manufacturers. For many years there have been, at the wholesale end, traditional levels of discount from suggested retail levels. These discount levels vary according to product group (more for guitars, basses and amps; somewhat less for keyboards and popular signal processing; and even less for high-end consoles, recorders, and signal processing). In the last couple of years the manufacturers have hit on a new trick: keep the retail price low so it's palatable to the end user, but increase prices by squeezing the dealer's profit margin. Japanese manufacturers have been under particular pressure because the yen has shot up against the dollar. If they passed through every change in currency value as a retail price increase, everyone would just laugh and ignore their products. So if you notice an increasing reluctance on the part of your equipment purveyor to offer you the discounts to which you've become accustomed, this is the reason.

As the market evolves it is likely that music and audio dealerships will undergo several changes. The ones that are service oriented will expect you to pay for the privilege. This can take the form of higher prices, or may come in the form of what the computer industry calls "unbundling," where nothing is thrown in on the deal. The hardware has one price. If you're self-sufficient and can take it home and run it no-questions-asked, you're out the door with the lowest possible price. If you need guidance you may be given a 900 number to call, and the meter will start running as soon as the call is picked up. Warranty service will be similarly unbundled: basic warranty service will require that your device be sent back to the manufacturer or a regional repair center, at your expense, while in-store or on-site service (perhaps with loaner equipment) will require the purchase of a separate service contract. These adjustments will bring the selling and buying equation back into balance: you'll get exactly what you pay for. After all, you wouldn't want to work for free, would you? Equipment dealers, are people too, and tend to think in exactly the same way.

J.D. Sharp is the owner of Bananas At Large, an independent pro audio dealership. We encourage manufacturers and customers to voice their opinions on this important issue.

KNOWING THE SCORE: Eric Guthrie (left) and Chris Knudsen

MARIACHI MEN

Eric Guthrie

"With MIDI, you can make it sound as good as you need to at home. And if you can't get your point across on a 488, you probably can't do it at all."

As the essential tools of multitrack recording and music production become very affordable, the gap between "professional" and "amateur" is increasingly defined by talent and effort rather than economics. Case in point: Eric Guthrie, an undergraduate biology student at the University of Texas at Austin who scored last year's surprise hit film *El Mariachi* in his apartment on a shoestring budget. Along with his partner, Chris Knudsen, Guthrie produced the critically acclaimed film score using an 8-track Portastudio, a small MIDI system, and a borrowed DAT machine — and without the benefit of timecode synchronization.

Before the overnight success of *El Mariachi*, Guthrie's musical "career" consisted mostly of childhood music lessons and his high school jazz band. In college he maintained an interest in music composition as a hobby and got some equipment for his own enjoyment, but never planned on doing it professionally. He became acquainted with fellow U.T. student Robert Rodriguez through a friend and subsequently created the music for the young director's first student film. On the basis of that, Rodriguez asked Guthrie to score his next film, which turned out to be *El Mariachi*.

Knudsen, yet another U.T. guy with some film and music production experience, came on midway through the project, when it proved to be too much for Guthrie to handle solo along with his studies. As a result of their successful collaboration, Guthrie and Knudsen are committed to an ongoing musical partnership, called GK Productions. And now the two former unknowns are getting a shot at Hollywood scoring projects.

Without much film scoring experience to work from, Guthrie referred to what some of his favorite musical influences had already done on soundtracks for other films, in particular, the Lyle Mays and Pat Metheny score for *Falcon and the Snowman*, and Peter Gabriel's score for *Last Temptation of Christ*. Guthrie explained that the music was intended to create a very definite sense of the film's location. "Robert wanted a real 'bordery' feel, and I expanded on that with my little bag of tricks: some looping, cool effects, some neat pads, chord voicings. We tried to make it sound a little bit more brooding and cerebral. We would give Robert a bunch of different versions, different mixes, different lengths...variations on everything, so he could edit any way he wanted."

Scoring *El Mariachi* was a learning experience — sometimes the hard way, as Guthrie tells it. "This was Film Scoring 101 for me. There was no SMPTE, so I had to visually cue everything, which takes a lot more time. In some scenes, the music was very highly synchronized and spotted to the action on the screen. And there are places where you hear a lot of layering, because we wanted a lot of separation and different effects on each element. We had to keep a notebook of how we stacked multiple elements onto each track, and then there were three people here during the final mixdown, twirling knobs and changing effects. It was crazy."

Guthrie's multitrack system is a Tascam 488 Portastudio, which integrates an 8-track cassette recorder and a 12x4 mixing console. His MIDI system consists of Ensoniq SQR and EPS-16+ keyboards, a Korg WaveStation, and a Yamaha TG77, run by Opcode StudioVision sequencing software on an Apple Macintosh Classic 2. Not much by Hollywood standards, but it's enough to get the job done, as Guthrie attests, "With MIDI, you can make it sound as good as you need to at home. And if you can't get your point across on a 488, you probably can't do it at all." —*Tony MACaroni*

MONKEY BUSINESS

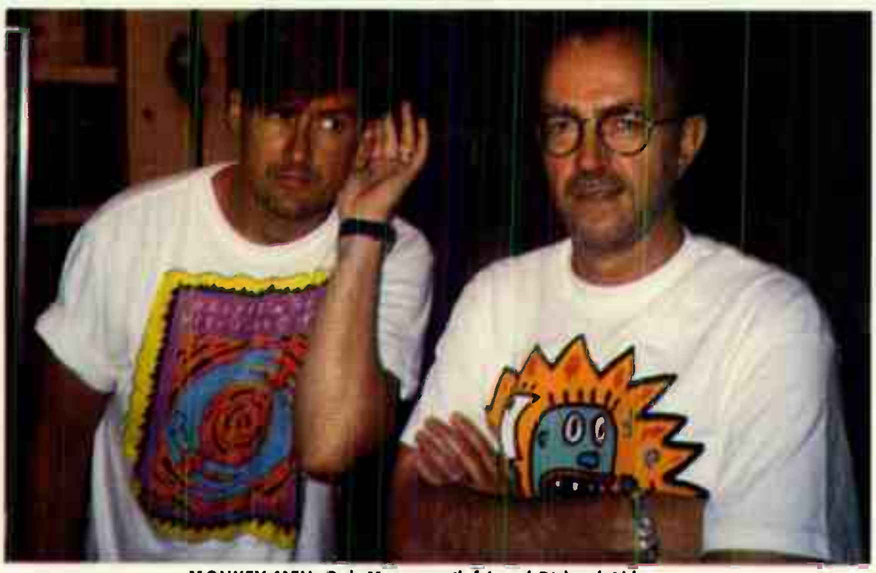
Richard Alderson & Rob Mounsey

"I don't like that tradition of hiring brilliant musicians and overdubbing them into a claustrophobic little eight-bar box."

Every creative endeavor is unique in some way, and while Rob Mounsey and Richard Alderson are obviously proud of their previous work together and separately, they feel strongly that their latest joint venture, *Back in the Pool*, is genuinely unique.

Several things support that belief. One is the label, Monkeyville Records, which Mounsey calls "the world's smallest record company." (He is it.) More important, however, is the kind of music on it (what Mounsey calls "New York-Brazilian-Afro-Pop-Jazz") and the way that music was performed and recorded.

"*Back in the Pool* could not have been created without computers," says Alderson, house engineer at Mounsey's Flying Monkey Studio and the man who recorded and mixed *Back*



MONKEY MEN: Rob Mounsey (left) and Richard Alderson.

in the Pool. While building music on computers is far from a new story these days, the way Mounsey and Alderson began this project definitely is.

Once Mounsey, who has served as arranger, keyboardist, and producer for many high-profile recordings, notably Steely Dan's *Gaucho*, Paul Simon's *Graceland*, Donald Fagen's *Nightfly*, and albums by Chaka Khan, Eric Clapton, Sinead O'Connor, had a rough idea for the 11 tracks that comprise the recording, he brought in various New York musicians to contribute performances. "I don't like that tradition of hiring brilliant musicians and overdubbing them into a claustrophobic little eight-bar box," Mounsey explains. "It's so much more musical and more fun to give them lots of room to play the way they play and to edit their performances later, rebuilding the tracks around them."

Alderson continues, "We made minimal rhythm tracks for each future composition, then recorded the featured musicians in an open musical framework. Rob performed live during this initial step, providing melodic and chordal guidelines. Then he built sequences for each piece based on his concepts and on the inspirations of the performers."

The musicians involved included some of the top names in New York: Michael Brecker, Kacey Cisyk, Lani Groves, Steve Khan, Will Lee, Jeff Mironov, Christopher Parker, Lew Soloff, Toots Thielemans, Nana Vasconcelos, and George Young.

"I took great care to construct a true three-dimensional acoustic during the recording and mixing," Alderson says. "To accomplish this, live performances were recorded in stereo with an AKG C34 single-point stereo mic, a pair of Josephson C640 mics, and a pair of Milab VIP 50 mics. Also, tracks were laid onto the multitrack in stereo configuration as much as possible. Processing was accomplished using an SPL Vitalizer, an Aphex Compeller, and an Eventide H3000SE. Our reverbs were two Yamaha SPX1000's, a Rev 5, and two Roland SRV2000's. Quested Q108 self-powered monitors were used for tracking and mixing the entire process. A Studer A-80 24-track with Ampex 499 tape at elevated level was our mastering format." Recording this way left Mounsey and Alderson with a wealth of raw material.

"During the production process," Alderson explains, "we made extensive use of Digidesign Sound Designer and Opcode's StudioVision to provide Rob with a unique musical palette. It allowed us to build the equivalent of up to 40 'virtual tracks' on the master tape. We mixed to a JVC DS-DT 900 DAT recorder. *Back in the Pool* is an analog recording that is highly finessed with digital processing and mixing."

Mounsey has worked out of his Flying Monkey studio full time since 1985, when Alderson converted what had been the living room of Mounsey's downtown loft into the heart of a recording facility. "I thought I wanted to build an 8-track studio," Mounsey

recalls, "and a mutual musician friend introduced me to Richard. He sold me on the idea then of going 24-track, and we haven't looked back since."

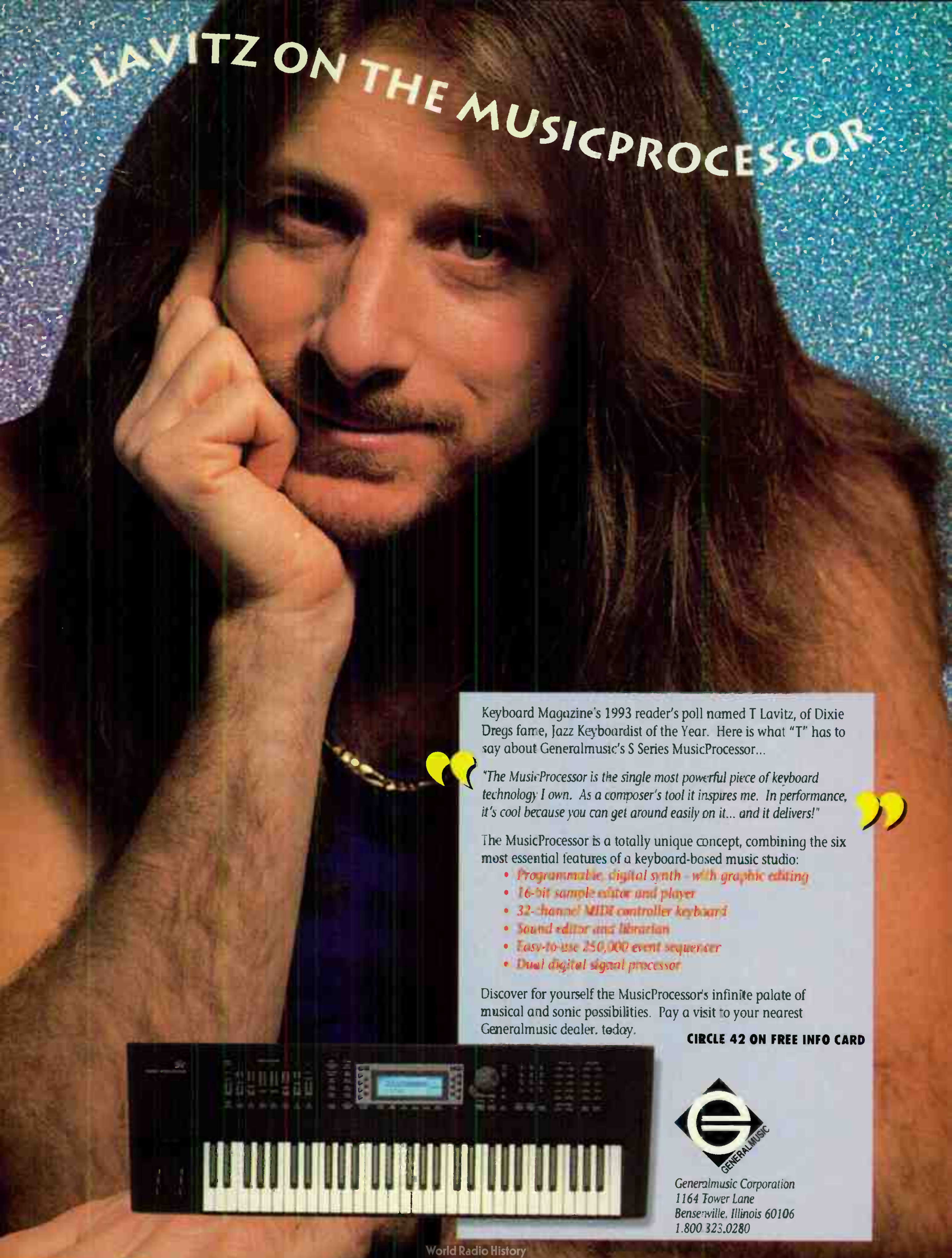
In 1991 Alderson completely revamped the original studio, adding an extensive Macintosh-based sequencing and digital recording workstation using Digidesign Sound Tools, two Sample Cells, Opcode StudioVision, and Digidesign's Pro I/O interface. Later, they replaced the original synth and sampler setup with a Korg Wavestation, a Yamaha TG77, and an Emulator EIIIXP. Recently they installed Optifile automation in the Amek Angela 36-input console and added a TC Electronics 5000 processor and a Lafont high-end audio rack with two mic pres, two compressors, and two EQs. Currently, Flying Monkey's studio can handle any technical or musical complexity. Future plans call for the addition of a 24-track hard-disk-based digital recording device to complete the world-class project studio environment.

Alderson has designed and built several other top audio facilities in New York and around the country, including Bass Hit Recording, Random Bus at Dennis Hayes & Associates, C-5 Inc.'s film sound editing facility and Foley room, 4/4 Production's Wharehouse Studios, and the soon-to-open Crew Cuts Film & Tape's Buzz Inc. SSL Scenaria-based audio suites.

In addition, Mounsey produced the Oscar- and Grammy-winning song "Let the River Run" from the feature film *Working Girl*, which he also scored. And Mounsey won an Emmy for composing the new theme music for the daytime drama *Guiding Light*.

Together he and Alderson have worked with a long and growing list of top recording stars, including Phil Ramone and such performers as Aztec Camera and Gloria Estefan.

But *Back in the Pool* is where their passion lies these days. Mounsey's hope is that serving as his own label will help and not hurt the recording's chances of success. No matter what happens, he and Alderson plan to repeat the recording process and techniques they used on this project on future efforts. Says Alderson, "It's the only way to go." EQ


T LAVITZ ON THE MUSICPROCESSOR
Keyboard Magazine's 1993 reader's poll named T Lavitz, of Dixie Dregs fame, Jazz Keyboardist of the Year. Here is what "T" has to say about Generalmusic's S Series MusicProcessor...
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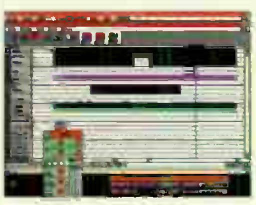

The challenge was to build a DAW so affordable that smaller facilities, project studios, and our customers installing multiple rooms could meet professional demands at the lowest possible price. That DAW is the ***Prisma***™ system.

A NEW LIGHT ON THE SUBJECT. The ***AudioPrisma***™ non-linear audio multitrack is integrated onto a single system board. It locks to black burst, chases timecode, and records on fixed and removable hard and magneto-optical disks. It provides sophisticated in-system digital patching, features real-time mixing and automation, uses three different editing windows for different production environments, and packs over 80 MIPS of 24-bit DSP power! ***Prisma***™ delivers sophisticated editing and powerful sound design features at a fraction of the price of lesser systems.

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*Suggested Professional Net price, dealer prices may vary. Requires 386 or higher PC and one or more SCSI storage devices.

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CIRCLE 63 ON FREE INFO CARD

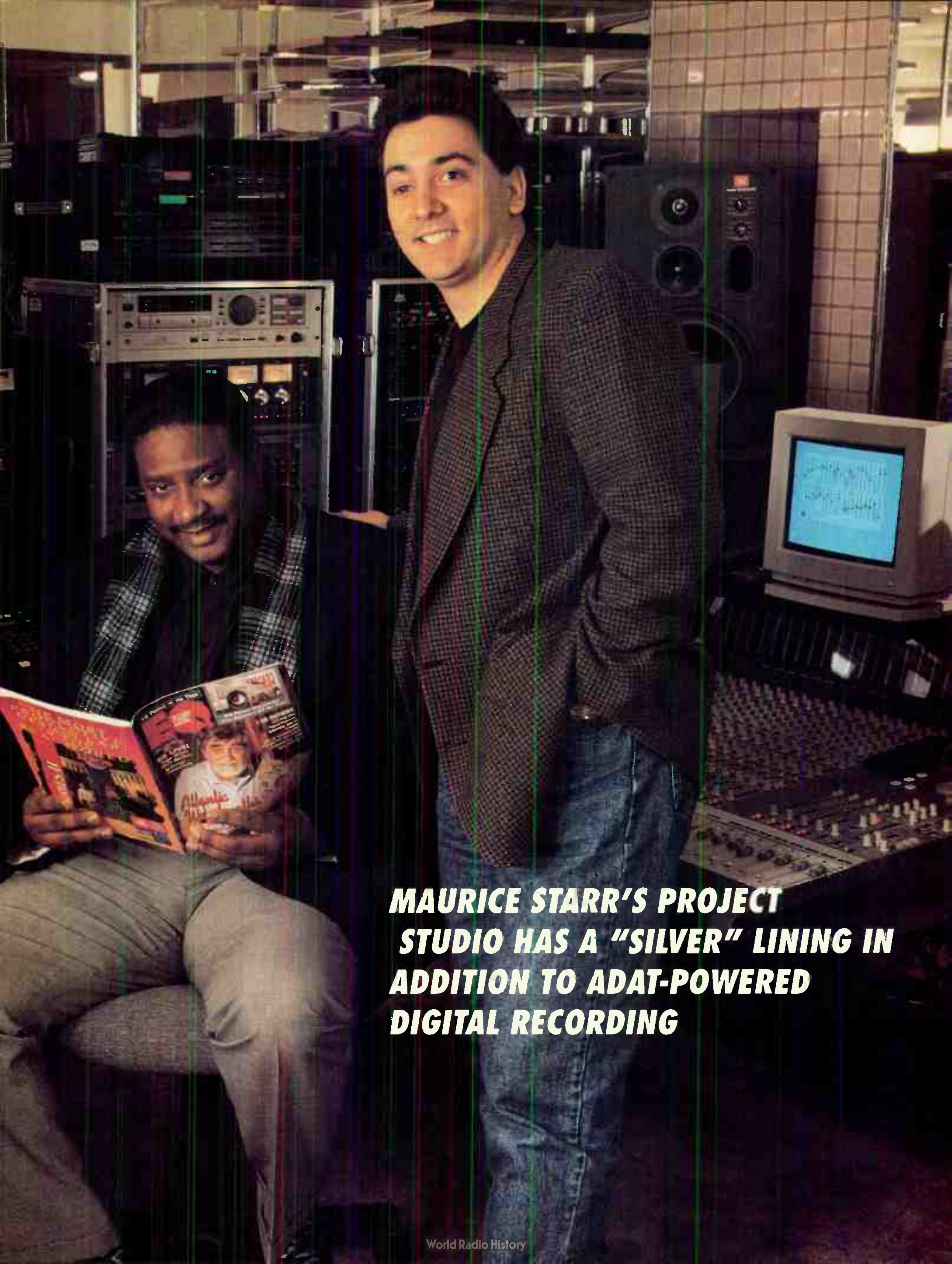
Starrmaker Machinery

STUDIO NAME: Nehemiah Music
LOCATION: Bloomfield, CT
KEY CREW: [left to right] Maurice Starr (co-owner); Silver Sargent (co-owner/producer/engineer); David Frangioni (chief engineer/main man)
CREDITS: New Kids On The Block; various gospel projects
CONSOLE: TASCAM 3700 32-input automated console with meter bridge and balanced ins and outs
KEYBOARDS: Roland JD990, U220 [2], and D110; E-mu Proteus 1 [2], Proteus 2 [2], Proteus 3 [2], and Procussion [2]; Yamaha TG77; Alesis D4; Korg M1 and O3W/R
SAMPLERS: Kurzweil K2000 [2]; Akai S1000 and S1100 with IB104 and hard-disk recording; Ensoniq EPS16+R
MONITORS: Yamaha NS-10; various JBLs and Auratones
AMPLIFIERS: Hafler 1200; Carver PM600
COMPUTERS & SOFTWARE: Apple Macintosh Quadra 950 20/120 with 2 color monitors; Opcode Studio Vision, Galaxy Plus, and Studio 5 [2]; MOTU Performer; Passport Alchemy and Pro 5; TASCAM Faderview; Digidesign Sound Tools 2
RECORDERS: Alesis ADAT with BRC [3]; Soundcraft 24-track 2-inch with autolocator
DAT MACHINES: Panasonic SV3700
OUTBOARD GEAR: Yamaha SPX900 [2] and SPX1000; dbx 166 [2] and 160XT; Alesis MIDIverb III [2]
MICS: AKG 414 and others; Shure SM57 and SM55; various Sennheisers
EQUIPMENT NOTES: Frangioni states: We love the 3700 with Faderview. It's very flexible, and the automation is very powerful. It's great being able to work with either scenes or real-time moves. FaderView allows us to do a Total Recall of everything on the console.

The ADATs are also awesome. They sound fantastic and the BRC is amazing. It does things that very high-end remotes can't do. Samplewise, the K2000 is Silver's favorite new toy. We just added the orchestral ROM, PRAM, and a 240 MB internal hard drive.

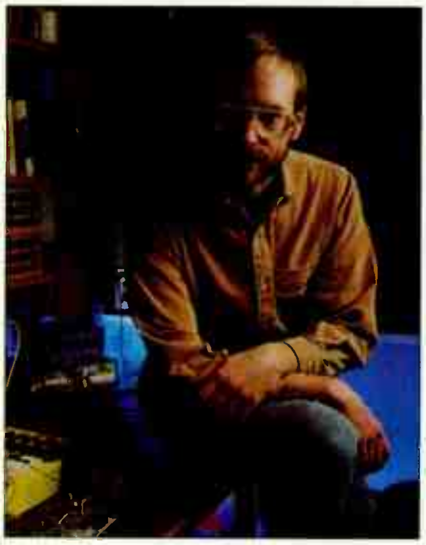
PRODUCTION NOTES: We often fly digitally into Studio Vision and arrange the parts before flying back to the ADAT. The Fostex synchronizer locks the ADATs and 24-track together to allow either to be master or slave. Special thanks to Chuck and Brenda Surack of Sweetwater Sound and DFE. EQ

Photo by Michael Partenio



MAURICE STARR'S PROJECT STUDIO HAS A "SILVER" LINING IN ADDITION TO ADAT-POWERED DIGITAL RECORDING

Who's Minding the Storage?



When your hard disk can't hold any more, who you gonna call?

BY CRAIG ANDERTON

The Achilles heel of digital audio is storage — at 10.5 MB per minute of stereo recording, you can chew through a hard disk in no time. But recording the signal in the first place is not as much of an issue as backing it up. Even a gigabyte hard drive probably won't provide enough storage to keep all your data for current projects along with backup data for previous projects.

Consider the options. A 44 MB SyQuest cartridge? It can back up a short pop tune, but at $60 a cartridge, that's an expensive way to go (SyQuests are not known for extreme reliability, either). An 88 MB cartridge? Better, but still a problem.

Magneto-optical drives are a big step up. A 128 MB drive costs under $1000 and cartridges run around $40, but we're still talking about backing up only 12 minutes of stereo, which is not enough for many projects. And 650 MB optical drives? Nice, but a bit pricey (around $2000 for the drive and $100 for each cartridge) and the technology isn't really fast enough for multitrack hard-disk recording if you want one piece of gear to do double duty.

DAT backup? I don't know about you, but DAT — which was never intended for pros — seems like a somewhat tentative backup medium. Those little teeny tracks on those little teeny fragile cassettes do not inspire feelings of tremendous confidence. Still, DAT backup does offer the advantages of being cheap (about $10 per DAT) and storing humongous amounts of data.

And that's what I used until I learned of a special offer for ADAT owners: a removable cartridge drive for $895 that stores up to 1.6 GB of data (partitioned into four 400 MB sectors) on an $11 cartridge that's much more robust than a DAT. A similar offer is also available to owners of the DA-88 (who will obtain even more storage — over 4 GB partitioned into four 1.1 GB sectors) and other digital audio recorders capable of recording and playing back through an AES/EBU or SPDIF interface. Sound good to you?

TALE OF THE TAPE BACKUP

The workhorse for inexpensive tape backup is the digital multitrack recorder itself, but the secret ingredient is an AES/EBU-compatible interface. The "$895" referred to above is the cost of an Alesis AI-1, the AES/EBU interface/sample rate converter for ADAT. This is what enables you to dump digitized audio data directly onto the multitrack tape. For DA-88 owners, the Tascam IF88AE digital interface provides an equivalent function.

If a device can send and receive audio as AES/EBU digital data (DAT machine, Digidesign DAT I/O, etc.), that data can be converted into the multitrack's proprietary digital format by the AES/EBU converter and stored on tape.

There are several advantages to backing up audio on digital multitrack tape:

• You're dealing with an inexpensive, universal storage medium. Rather than having some audio data on SyQuests, some on DAT, and some on floppies, everything can be saved onto a common format.

• It's easy to "clone" tapes and make safeties if you have access to a second digital multitrack.

• Data start times can be referenced to the recorder's time reader, so if you have an autolocator you can simply enter the time where the data resides, then retrieve it. (With the ADAT BRC, you can name location points and store them as a "song." This makes it really easy to find and recall data.)

• If you're backing up mono audio, you can store twice as much as mentioned above (3.2 GB for ADAT and 8.8 GB for the DA-88). That's an entire sample library for most normal human beings.

• It's easier to exchange audio data with other studios since it's more likely they'll have a digital audio multitrack recorder than something like a 600 MB optical drive.

You don't get something for nothing, however. The main trade-off is that backups generally have to be done in real time, and restoring is a real-time process, too. Also, tape falls somewhere in the middle of the robustness scale; it's probably better than a SyQuest cartridge, but not as good as a magneto-optical cartridge. The biggest disadvantage is that you can't back up other types of computer files, since only digital audio is eligible.

DETAILS, DETAILS

Because of the "partitioning" effect of having several different tracks, you'll have to keep time limits in mind as you decide which files to record on particular tracks. It's also important to maintain accurate records of what data is located at what times on what tracks (I made up a form for entering this information — it definitely helps). Still, ADAT's 400 MB and the DA-88's

1.1 GB partitions are not exactly chintzy, and you'll probably encounter few projects that need more contiguous storage space than that.

Backing up DAT is simple: just plug the DAT player's digital out into the appropriate digital input of the AI-1 or IF88AE, put the tracks you want into record, and play the DAT.

Backing up samples is a different matter. For my Mac-based system, once the sample is imported into sample-editing software (Alchemy, Infinity, or Sound Designer), it's saved to hard disk. I then use Digidesign's DAT I/O hardware and DATa software to shuttle the data from disk to the AI-1. (DATa can save and restore either individual files or folders of individual files.)

A FEW FINE POINTS

• If you're using DATa, save all your files as Sound Designer II files before backing them up. DATa will also back up AIFF files, but the process takes longer.

• DATa writes a header that includes loop points and other file characteristics as well as the audio itself. Sound Designer II files, however, are recoverable as standard audio should they become corrupted.

• I try to keep everything in my studio at 44.1 kHz, which initially presented a problem because of ADAT's default 48 kHz sampling rate. However, setting ADAT's pitch control to –1.47 provides a 44.1 kHz sampling rate (as confirmed by the pitch control readout), so you can do a straight 44.1-to-44.1 transfer for backup. (The AI-1 does allow for format conversion, so you can bounce 44.1 to 48, or vice-versa, if need be.)

• Store your tapes properly (a cool, dry environment is optimum).

GET BACKUP TO WHERE YOU ONCE BELONG

Although multitrack tape recorders provide very inexpensive backup, they can't do everything and are not always the optimum choice for backup. Tape works best for archival applications; I still use a 44 MB SyQuest drive as a "holding tank" for current work to avoid the time required to do a tape-based backup/restore onto my main hard disk.

Unfortunately, a 44 MB disk only has enough space for backing up samples or a short tune. Eventually I'd like to replace it with a 600 MB optical drive when the prices come down a bit. Even then, though, I expect to continue making backups on tape; at $100 apiece, 600 MB cartridges are still much more expensive than something like an S-VHS tape. For archival files that may or may not get used again, tape seems like a much more cost-effective way to go. In any case, it certainly has worked well for me.

[Editor's note: For more information on the Alesis AI-1, check out Roger Nichols's review (a sidebar to the Alesis BRC review) in this issue.]

Craig Anderton, EQ's West Coast editor, is the founder of a 12-step program specifically for musicians. The 12 steps are C, C#, D, D#, E, F, F#, G, G#, A, A#, and B.

Suggested retail list $279.

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CIRCLE 09 ON FREE INFO CARD

The Seriou

The TASCAM DA-88 The Digital Multitrack Deck For Serious Production

It's true. The first machine designed specifically for low cost digital multitrack production is now available. And it comes to you from the world multitrack leader, TASCAM. It's simply the most advanced, well thought out and heavy duty digital 8-track deck you can buy. The best part is, it's incredibly affordable.

The DA-88 is built for production. The integrity of TASCAM's design is evident in every facet of the deck. From its look and feel — to its exceptional sound, unsurpassed features and expansion capability.

Goes Faster, Lasts Longer And Takes A Beating

While we admit that it's an elegant looking machine, it's tough to see its finest asset. The tape transport. Designed and manufactured by TASCAM specifically for the DA-88, it's fast, accurate and solid. And that's what counts in production — in personal studios, project studios or in those demanding high-end facilities.

You'll notice it uses superior Hi 8mm tape, giving you a full 108 minutes of record time. What's more, the transport is lightning fast and yet so quiet you'll barely hear it blaze through a tape.

We didn't stop there. Because production environments are notorious for constant, if not abusive, shuttling, punching, 24-hour operation — you get the idea — the transport was designed and built to take a beating.

Even more impressive is the transport's responsiveness. Take a look at the front panel. Notice the shuttle wheel? Turn it just a bit and the tape moves at one fourth the normal play speed. Turn it all the way and it flies at 8 times faster. Do it all night if you want. It's quick, smooth and it's precise. Need to get to a location quickly? Accurately? Shuttle a bit and you're there. The location is easily viewed on the DA-88's 8-digit absolute time display — in hours, minutes, seconds and frames. With the optional SY-88 sync card it displays timecode and offset, too.

You Already Know How To Operate It

Unlike other digital multitrack decks, the DA-88 works logically and is simple to operate. Like your analog deck. All functions are familiar and easily operated from the front of the deck.

Adding the optional SY-88 synchronizer card is as easy as changing a Nintendo® cartridge. With it you're SMPTE and MIDI compatible. And no matter how many DA-88s you have locked up, you need only one sync card. Other optional accessories include AES/EBU and SDIF2 digital interfaces allowing the digital audio signal to be converted for direct-digital interfacing with digital consoles, signal processors and recording equipment.

s Machine

Take punching-in and out, for example. You have three easy ways to do it. You can punch-in and out of single tracks on the fly. Just hit the track button at the punch-in point. Hit it again to punch-out. You can use the optional foot switch, if you like.

Or, for multiple tracks, simply select the track numbers you want to punch, push play, and when you're ready, hit record to punch-in, play to punch-out.

Finally, for those frame accurate punch-ins, you've got auto punch-in and out. In this mode you can rehearse your part prior to committing it to tape.

No matter which way you choose, your punch-in and out is seamless and glitch free due to TASCAM's sophisticated variable digital crossfade technology.

That's not all, you also can set your pitch (± 6%), sample rates (44.1 or 48K), as well as crossfade and track delay times. All from the front of the DA-88.

Complete Synchronicity

There's more. Add the optional SY-88 synchronizer card to just one of your DA-88s and you've got full SMPTE/EBU chase synchronization. The best part is, you can record time-code without sacrificing one of your audio tracks. You also get video sync input, an RS-422 port to allow control of the DA-88 from a video editor, and MIDI ports for MIDI machine control.

A Digital Recording System That Grows With You

The DA-88 is truly part of a digital recording system. Start with 8 tracks today — add more tomorrow.

Adding tracks is as simple as adding machines — up to16 for a total of 128 tracks. They interconnect with one simple cable, and no matter how many DA-88s you have, they'll all lock up in less than 2 seconds.

Controlling multiple machines is made simple with the optional RC-848 remote. With it you can auto locate and catch 99 cue points on the fly. It comes complete with shuttle wheel, jog dial, RS-422 and parallel ports, and it controls other digital and analog machines, too.

Listen To The Rest

Of course, the sound quality is stunning. With a flat frequency response from 20Hz to 20kHz and dynamic range greater than 92dB, it delivers the performance you expect in digital recording.

So get to your authorized TASCAM dealer now. Check it out. Touch it. And listen to it. Once you do you'll know why the TASCAM DA-88 is the serious machine for digital production. The TASCAM DA-88 is the choice of studios worldwide. And at only $4,499, it should be your choice.

TASCAM

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CIRCLE 56 ON FREE INFO CARD

The Shure 520D

A classic mic that really brings out the blues

Photo by Ed Colveer

MICROPHONE NAME: Shure 520D "Green Bullet"
MICROPHONE TYPE: Controlled Magnetic
POLAR PATTERN: Omni-Directional
FIRST RELEASED: May, 1949
ORIGINAL COST: $16.50
FREQUENCY RESPONSE: 100 Hz – 5 kHz
OUTPUT IMPEDANCE: Switchable between 150 ohms (low) and 15k ohms (high)
SONIC NOTES: The 520D is still in production mainly because of its popularity with harmonica players. It's a dual impedance version of the original 520 "Controlled Magnetic" high-impedance mic.
USER NOTES: Says Chicago blues master, Sugar Blue, "I've been using a (Green) Bullet mic for the last 20 or 25 years. I like it because it's got a really rough signal. It has a 'bubbling' sound that others don't have, and I've tried lots of different microphones. I look for a 'picked' sound, like a guitar pick against strings, and the 520D delivers it.

"I also appreciate the shape of it. With its big grille, you don't have to move the instrument from side to side. The harmonica is built with the low reeds to the left, high reeds to the right; with most microphones, the face is too small to incorporate the full range of the instrument without moving the microphone to the reed that you're playing and thereby possibly losing your embouchure. With the 520D, the breadth of the face allows every reed access to the element without moving."

The 520D is a dual impedance mic that can be connected to balanced low impedance connectors after a simple internal change, and can also be connected to balanced high impedance connectors via an easy rewiring at the plug. Its frequency response rises steadily from 100 Hz to 4 kHz where it takes a 5 dB drop and then rises 5 dB at around 5000. There is a sharp drop in response at 10 kHz. This curve seems to suit the sonic characteristics of the harmonica to a tee. EQ

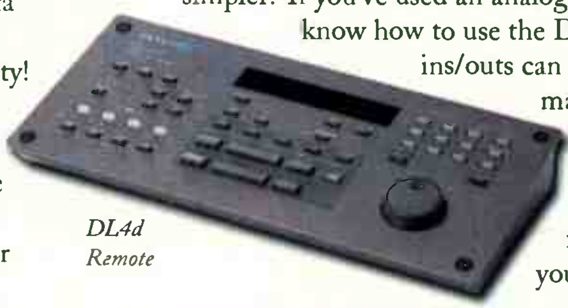
Ten Reasons Why You Should Cho

1. **Tapeless Editing** The DR4*d* can simultaneously record 4 tracks directly to standard SCSI-compatible hard disks, not tape. Tape recorders which use a cassette format (VHS, 8mm, etc.) have a huge problem: without at least two machines, you can't edit. But even a single DR4*d* allows random access editing that tape recorders just can't offer. Move, Copy, Insert, Copy + Insert, Move + Insert, Erase, and Delete with ease. Edit with complete confidence, because if you try an edit but change your mind, the Undo function will instantly restore the previous arrangement. It's a breeze to copy any part of a track and paste it anywhere on any track, even with a specified number of repeats. Or perhaps use the Insert commands to instantly slide track data in time against other tracks. This editing power encourages experimentation, and thus, your creativity! Imagine it. Do it.

2. **No Waiting** Another problem with tape is the time required to physically move from one point on the tape to another. Concentrating on your music is what's important, not waiting for tape to shuttle back and forth. Never again waste such precious time: the DR4*d* allows you to instantly move to 108 different locations. Set up repeat sections, jam along with your tracks, then drop into record to capture it all while it's still immediate, fresh.

3. **Jog/Shuttle** Another cool DR4*d* advantage is the ability to offer scrubbing of audio, like "reel-rocking"on analog decks - only with much better quality. Our Jog/Shuttle wheel lets you scrub through the audio at various speeds, forwards or backwards. So finding precise editing points is only as complicated as using your ears.

4. **Familiar Operation** One concept we *did* want to carry over from tape recorders is the user interface. Friendly, tape machine-style controls make the DR4*d* by far the easiest hard disk recorder to use. With dedicated buttons for Play, Stop, Rewind, Fast Forward, and so on, what could be simpler? If you've used an analog deck, then you know how to use the DR4*d*. Punch-ins/outs can be performed manually or automatically from the front panel, or via footswitch. Like you'd expect.



DL4d Remote

5. **Expandability** Up to four DR4*d*s can be chained together to create a 16-track system, simply by plugging an optional cable between units! And the optional DL4*d* Remote makes it a snap to

ose the DR4*d* Hard Disk Recorder

control all of them. An optional, factory-installed 200 MB internal hard disk offers 32 track minutes of recording right out of the box. The DR4*d* can handle up to seven hard disks and supports seamless overflow recording across multiple disks. With enough disk storage space, you can actually record on all four tracks for an incredible 24 hours!

6. EXCELLENT CONNECTIONS Four balanced TRS 1/4" Input and Output jacks, easily switchable between -10 and +4 dBu levels, simplify interfacing with any type of console. The DR4*d*'s pair of digital I/O ports allow communication with other digital devices in the form of both XLR and RCA connectors (AES/EBU or Type II selectable), as well as provide DAT backup. And then there's the supplied SCSI port for access to external hard disk drives. Just plug and play!

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

8. DEDICATED DESIGN The DR4*d* is a dedicated digital audio product rather than an add-in board for a computer. It's a tool designed for a single purpose: to record and edit audio precisely, effortlessly, and affordably. We think you'll agree that it succeeds on all counts beautifully.

9. SOUND QUALITY The DR4*d* contains Akai's own advanced digital technology, including super-clean 18-bit 64x oversampling A/Ds and advanced single-bit 8x oversampling D/A convertors with 18-bit resolution. Industry standard sample rates include 48, 44.1, and 32kHz. In short, the quality is superb and with a full 96dB dynamic range, you can rest assured of always sounding your best.

10. $1995.00 Simply put, the DR4*d* is the best value in digital recording today. For the first time, the nucleus of a professional quality 4-track hard disk recording system can be yours for only $1995.00! Just add internal or external hard disks, and you're ready to use our latest masterpiece for creating your next masterpiece.

DR4*d*

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recording is actually a 15-bit recording, not 16-bit.

• And now, from the "rules-are-made-to-be-broken" department: if a piece of digital audio has just a few extremely short transients that hit maximum headroom, you can go a little over zero and never hear the differ-

CONDUCT IN THE ALL-DIGITAL WORLD

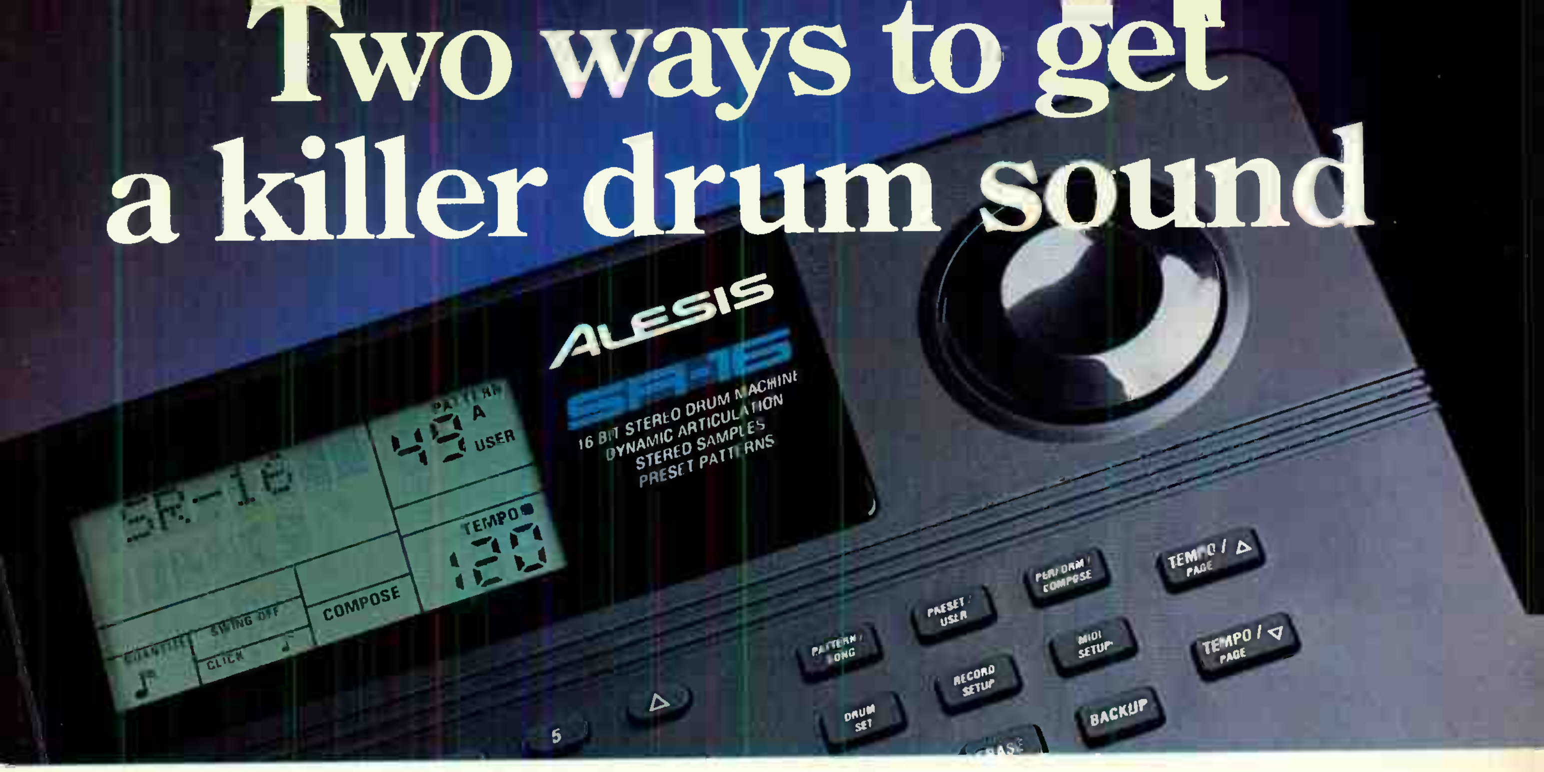
In an all-digital world, all units must be asynchronous and be slaved to a common word clock. To give an idea of what this means, here's a real world example from the "Frangioni Files."

Consider an all-digital studio with an Akai S1100 sampler, Digidesign Pro Tools hard disk recording system, Fostex D-20 DAT, and a Yamaha DMP7D digital mixer. All devices need to lock to a common word clock. The word clock carries the sample-rate information, which digitally controls the timing of all devices.

An asynchronous unit essentially means that it does not require the external word clock to function. For example, the DMP7D uses an interface called the D2-IB to get AES/EBU to and from the outside world. This unit is asynchronous, receiving its word clock from the DMP7D and the AES/EBU device to which it's locking. Therefore, the D2-IB is actually locking to the DMP7D as well as the input device (D-20, Pro Tools, etc.). Pro Tools requires the Video Slave Driver to accomplish external word clock synchronization. Word clock needs to be distributed to all devices through a word clock distribution amp; the S1100 has no word clock input, so it receives timing information through its AES/EBU port. The D2-IB allows this to work because it's asynchronous.

Of course, this is just a crash course (and the system did in fact crash a few times) about typical digital audio interconnections. There is a lot more to discuss — and you never know, such a discussion just might fill some future pages in *EQ*.

Two ways to get a killer drum sound



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ALESIS STUDIO ELECTRONICS

ence (don't worry, we won't tell anyone). And yes, you can light the red LEDs from time to time on Digidesign's digital I/O interfaces without harming the audio quality.

• As far as digital metering goes, most units differ just enough to be useless. Even though a meter may read zero on peaks, there is often 2 to 3 dB of headroom left. Most units will flash zero continuously if you really go over.

• For those of you who mix in your own studios and aren't too concerned about the clock, run through a mix a couple of times at different levels to find out where the true maximum headroom really is. Very often you'll find that you had a bit more headroom to go than initially indicated; giving the signal a little more boost can yield a better signal-to-noise ratio and make better use of the available headroom.

• Always turn down your monitor volume when plugging and unplugging digital connections. Here's why: let's say an ADAT is hooked up to the AI-1 sample rate converter, which is receiving AES/EBU data from Pro Tools. If the speakers are live in the studio and you unplug the connector from the AI-1, the system will lose its digital clock signal and noise will appear. The noise can be very loud and generate nasty digital spikes (goodbye tweeters). Some devices, such as Pro Tools, handle this with little fuss; other devices (such as the original Sound Tools) pass along the digital clock noise and emit very loud and disturbing bursts. Be careful!

• Digital patchbays are just as useful as analog ones. For instance, the David Frangioni setup uses a digital audio patchbay (custom designed by Andy Topeka, known for his work as equipment tech for groups such as the Cars). All of the digital ins and outs go through the bay, which uses Cannon-style BNC connectors for S/PDIF lines and balanced 1/4-inch for AES/EBU.

• Speaking of patchbays, make up

AKAI DR4D TIPS

An insider's look at getting the most from this new digital four-track.

(1) You can use the DR4's UNDO function to audition and choose between two recorded sections, such as a guitar solo. Record a solo on a track. Now record over that solo on the same track. When you press the UNDO button, the first solo will be active. Press the UNDO burtton again, and the second solo becomes active. You can choose which one you wish to keep. Make your choice prior to doing any more recording or editing, as the DR4 has only one level of UNDO.

(2) Need to practice a section over and over to get a solo right? The DR4's REPEAT key will allow you to seamlessly loop a section of your song. To use the REPEAT function, mark an IN-point and an OUT-point. Stop the DR4, hit the REPEAT button, and then hit the PLAY button. The DR4 will now loop the section between the IN-point and the OUT-point, and of course there is no lag when it starts the loop each time.

(3) The DR4 uses 18 bit A-to-D converters. These can be used in place of lower quality 16-bit converters with DAT, tape or hard disk systems. Arm tracks 1 and 2, feed your audio into the DR4, then go out of either the AES/EBU or S/PDIF jacks on the back and go digitally into your other digital machine. You'll end up with a higher quality recording. Make sure to specify either Type 1 (AES/EBU) or Type 2 (S/PDIF) in the SUBMENU DIGITAL OUT function.

(4) The DR4 can serve as an editor for either digital tape machines, like the ADAT or DA-88, or for analog tape decks. With a tape machine, it requires two machines, a transport controller, lots of tape shuttling, and lots of patience to edit. The DR4 can perform edits internally and very quickly. Use the IB112T SMPTE interface on the DR4, which does not require a track for sync. Stripe a track of timecode on the tape machine, and then synchronize the DR4 to the tape machine. You can use the DR4 to fly in vocal or instrumental

parts to the tape machine, or to copy vocal choruses without having to sing them again. In addition to the edit features, you also have four more tracks to work with.

(5)If you get stuck in the display, such as capturing a time location you don't want, and want to return to the main screen, press the ESCAPE key. This will always return you to normal operation.

(6) When working with the IB113M MIDI interface and a sequencer, there is a MIDI limitation of 1024 bars for song position pointer. When starting a new song, reset the clock to zero by pushing the RESET key. This will also reset the song position pointer.

(7) The clock on the DR4 is a 24-hour clock, and allows you to record anywhere within that 24-hour period, as long as you keep within the limits of the hard disk capacity. This allows you to start recording wherever you like, such as at even hour marks. Certain time regions, such as hour -ten or hour-fifteen can be used as "scratch-pad" areas for editingh and assembling in. Copy or move your material to a different location and when you are happy with the edits, move it back to the original time."

(8) When using SMPTE timecode as a source of sync, you often need to offset the time you recorded your piece. For instance, you might want a song starting at an absolute time of ten minutes to play back at one hour. To do this, locate the DR4 to the start time of the song, (00:10:00:00). Press the STORE/ENTER key, and then press the RESET key twice. The display will flash -.-.-.-.-.-.-. Now use the keypad to enter the desired time (1:00:00:00). Your song will now play back at one hour.

some BNC to 1/4-inch patch cables. With most combinations of equipment, this allows for effortless format conversion between S/PDIF and AES/EBU. (Yes, it really does work most of the time.)

• Digital feedback loops can exist just as easily as analog ones. If you're using Digidesign's DAT I/O with a DAT recorder, and both the digital ins and outs are connected, under some conditions you can get a feedback loop (as evidenced by flashing lights, nasty noises, and the appearance of poltergeists in extreme cases). The solution is simple: unplug one end of the cord that's not being used for a given situation. In other words, if you're sending data from DAT I/O to DAT, unplug the digital input to the DAT I/O but leave the output connected.

• Samplers that include digital I/O are just one more reason why living in the 1990s isn't really so bad. If your sampler offers the option of digital I/O, get it immediately! This lets you use outboard converters, backup data to DAT, and transfer sound in the digital domain (from CD, DAT, sample editors, and so on). The K2000 will even let you use it as a standalone analog-to-digital converter — just send audio into the analog inputs while in Sample mode, and the digital outs will output the audio in real time. Pretty cool!

• For those of you not yet accustomed to working with digital transfers, don't delay any longer. Working in the digital domain not only retains the high fidelity of the source material but also offers one critical feature: standardized gain structure. Whenever you transfer between two digital devices, you do not have to realign your gain. This saves so much time in the studio that many engineers transfer digitally solely for this reason (hey, whatever works).

Just remember that the most important point about digital is that it really does work. If you do encounter problems, it's a pretty good bet that the problem doesn't lie with a particular piece of gear, but rather how it interfaces with the rest of the world. Sometimes just one little switch thrown in the wrong direction can upset an entire system, so it pays to know your gear and once you find a working setup, stick with it. **EQ**

PARSONS, PLAC

ES, AND THINGS

Alan Parsons describes his quarter-century of recording and how he'll "Try Anything Once"

BY STEVE LA CERRA

Almost 25 years ago, Alan Parsons landed a gig at a recording studio where the musical output would become legendary. As an assistant engineer at Abbey Road Studios he worked on The Beatles' *Abbey Road*. In 1973 he engineered an album which has been on the Billboard Top 200 Albums chart for the better part of the 20 years since its release. That album, Pink Floyd's *Dark Side of the Moon*, has certainly made its mark on the music industry and remains a self-contained course in recording technique for future generations of "wannabe" engineers.

That was only the beginning for Parsons. He took a step to the other side of the glass and began a partnership with singer/songwriter Eric Woolfson to form The Alan Parsons Project. The Project released a series of albums that have spawned hit singles such as "Games People Play," "Time," and "Eye In the Sky," and have brought Parsons ten Grammy Award nominations.

Parsons's latest release, *Try Anything Once* (Arista), carries his trademark recording techniques: the instruments are clean, the vocals intelligible, and you can hear every part as it contributes to the overall recording. *EQ* recently spoke with Alan Parsons about *Try Anything Once* as well as about production, engineering, and his project recording studio named Parsonics.

Q. Your last album, *Gaudi*, was released six years ago. How have developments in recording technology changed your approach to engineering?

A. I wouldn't say that there have been any dramatic changes in the last six years. I think recording technology has evolved in a natural way ever since I started; doubling up the number of tracks approximately every five years. I think what really is undergoing more radical and more rapid change is musical instrument technology, as well as the attitude of engineers and producers.

The attitude we have in the changing world of the musician is that we are having to look toward the music end of the market to find anything new. That isn't to say that the audio industry is not undergoing change, because it is. It is just that we are seeing more rapid change on the musician's side of things.

Perhaps some of the technology that is often associated directly with pro audio is actually musician's technology — such as sampling, sequencing, and all that kind of stuff. Sampling, sequencing, and drum machines — those are the sorts of things that have really changed the industry. Nothing has really happened in the last seven or eight years in recording and studio technology. Digital tape machines and, digital multitrack is now within everybody's reach; that is a development. But the actual recording technique, to answer your question, has not really changed much.

Q. How have the changes in instrument technology affected you?

A. I often record a part to a sequencer for the first time and then put it on tape. That is done to clean it up, or make changes in the sound, or whatever. It's quite a common practice now, and I accept that as an advance in a way. But I'm always very careful to hold on to the human touch. I don't like to rely too heavily on sequencers to bring a product home. I have always been a great one for trying to get as many people as possible playing together on a record at the same time and I am still sticking to that. I think that generally you get the best results that way.

Q. So do you do your rhythm tracks live now?

A. Well, I am a little short of space at Parsonics. If I had the space I'd love to be able to record everything live. Certainly live drums. But having said that, what I really mean is a live drummer, because I'm using sampled sounds played with pads and all the drum brassware is being recorded for real. I think that there is no substitute for a good drummer — whether he is playing pads or programming a drum machine, whatever, as long as he has some of his character in there.

Q. What were your intentions when building Parsonics?

A. This latest version of Parsonics is actually my third. My first studio was fairly ambitious, fairly expensive. I didn't cut any corners. I made the mistake of thinking that it was forever. And then having done that, I built a second studio thinking that it, too, was forever. The only reason I didn't complete the second studio was I thought I was going to be leaving the country to live in the States. But plans changed and now I'm back here (in the UK). So my new studio essentially is a compromise between the two. On second thought, I wouldn't say that it's a compromise exactly, but I thought very carefully about where to spend the most money.

This time, I was really just intent on making sure that I had a shell that was soundproof and acoustically sound. It's very much a workshop sort of atmosphere, so I spent the most on acoustic treatment and on just getting a place where I felt comfortable. I didn't think of other people as much. I've got no accommodations. There is a control room that is adequately sized, a rather smaller studio than I would have liked, and a kitchen.

Q. What gear are you using?

A. Two 24-track digital Sony machines locked together. They are the original 3324's, but with Apogee filters that are undoubtedly an improvement to the sound. If money were no object, I would

"There is no substitute for a good drummer."

"I think that great records come from great moments — not from great equipment."

buy the new series, the 3324S, but I'm waiting to see how this album does and then I'll take a view on whether I can afford to do so. I have an Amek Angela console, which I am very pleased with. I see no reason to update that just yet.

Q. How about this latest interest in tube processing gear? How do you feel about that?

A. Well, I worked with it in the old days. Basically it gave me nothing but problems. It has its pluses and minuses. I'm a great one for taking what you have got and making the best of it and not laboring with trying to get the best sound by endless experimentation. If I had a great tube mic, a great tube limiter, and a great tube mixer to put them through, I would use them provided I could deal with the unreliability and what have you. But I'm a great one for just getting the job done, and I think that great records come from great moments — not from great equipment.

Q. Is there any particular equipment that you have found that makes your job easier?

A. In a sense, digital recording has made my life a lot easier. I like the idea of being able to edit selectively between tracks, where you don't have to take a razor blade and cut across the whole tape. It's like cut-and-paste audio, which is very easy with digital audio machines. And you have the advantage of no generation loss.

I think that sequencers, to a degree, are making the job easier. The ability to clean up and correct a performance when the need demands makes it very quick and easy. Obviously digital reverb units are infinitely flexible, which saves time also. I don't have to mess around with live chambers and tape delays the way I used to.

Q. Do you have any favorite digital reverb at this point?

A. I often enjoy the quick setup of the

"I often enjoy the quick setup of the cheaper low-end Japanese boxes."

cheaper low-end Japanese boxes. But when it comes to mixing, I tend to go for the more-difficult-to-program-but-better-sounding units like the Lexicon PCM-70, where you do have to fight a bit to get what you want out of it. I also use a Yamaha Rev 7 and the new Yamaha SPX-990.

Q. You mentioned that you are doing some triggering of samples played from pads. Are these samples that you have recorded specifically for the project?

A. No. Stuart Elliott (the drummer) is very much into carrying around a large library of his own samples for the AKAI S1000. He also works at home on his own. We add a combination of my own S1000 and his.

We have since both traded in our S1000's for S3000's. One thing I have recognized with experience is that you cannot be expert with every machine that comes on the market. And the S1000 seems to be Stuart's particular forte. I am quite happy to leave the drum samples to him. He will always offer up a whole bunch of different sounds for me to choose from, which is a nice thing to have. Ultimately it is very timesaving, because to achieve all the different drum sounds by real means — such as through mic technique, changing drums, tuning drums, and special effects — would literally take days.

Q. Do you still record acoustic drums?

A. I am not saying that I never will, but on this particular album all the drums with skins were samples, and all the cymbals and high hats were recorded as usual. But the important point to notice is that everything was being played at once. Stuart had a kick sample, a snare sample, and a series of tom samples all being played live with the cymbals and high hats. It was a real drummer's performance — not a drum machine or sequencer in sight.

Q. Did you and Stuart find that there is a certain amount of delay between when he actually strikes the pad and when the drum sample sounds?

A. I have never regarded that as a problem. I just said there was not a sequencer in sight, but occasionally we recorded for safety on a sequencer. And you might find that you want to take a couple of beats and trigger them a few milliseconds earlier. I think the equipment he is using is giving a delay, but it's pretty insignificant. Certainly not enough to be a concern. I think there is a little bit of paranoia attached to MIDI delays. Basically until you are into delays above, say, 10 ms, there is no problem at all. Even at 20 ms you could say there aren't really problems.

Q. Any general engineering advice?

A. My only solid advice would be not to try to master everything. You can't possibly do it. There's too much technology around these days to be able to know everything about recording, everything about MIDI, and everything about computers. Find an area that you want to specialize in, and go with that. EQ

STUDIO HAT TRICK

Alan Parsons secluded himself in his state-of-the-art Parsonics Studio in Sussex, England, and emerged 100 days later with *Try Anything Once*, his first album in nearly six years. The by-product of 25 years of twiddling behind the board, *Try Anything Once* sees this studio legend further exploring a plethora of musical themes and styles.

Unquestionably, Parsons's musical roots run deep. After a stint playing guitar in a high school blues band, Parsons's career took an eventful turn when he served as an assistant engineer at the famed Abby Road Studios. After working under the tutelage of Beatles producer George Martin, Parsons went on to lend his engineering skills to the Hollies, Roy Wood, and Olivia Newton-John. And after the Beatles split, he worked with producer Phil Spector on George Harrison's *All Things Must Pass*. He also worked with Paul McCartney's Wings on *Red Rose Speedway* and the hit single, "Hi Hi Hi."

In 1973, Alan Parsons achieved perhaps his greatest moment as an engineer working on Pink Floyd's *Dark Side of the Moon*, which earned him his first Grammy nomination and a perennial slot on *Billboard's* Top 200 album chart. Three years later, he launched the Alan Parsons Project with Abbey Road alumnus Eric Woolfson and scored a number of hit singles such as "Games People Play," "Time," "Eye In The Sky," and "Don't Answer Me."

Having completed the proverbial hat trick as he's enjoyed success as an engineer, producer, and artist, Parsons says that his next challenge lies in bringing the songs on *Try Anything Once* to the concert stage later this year.

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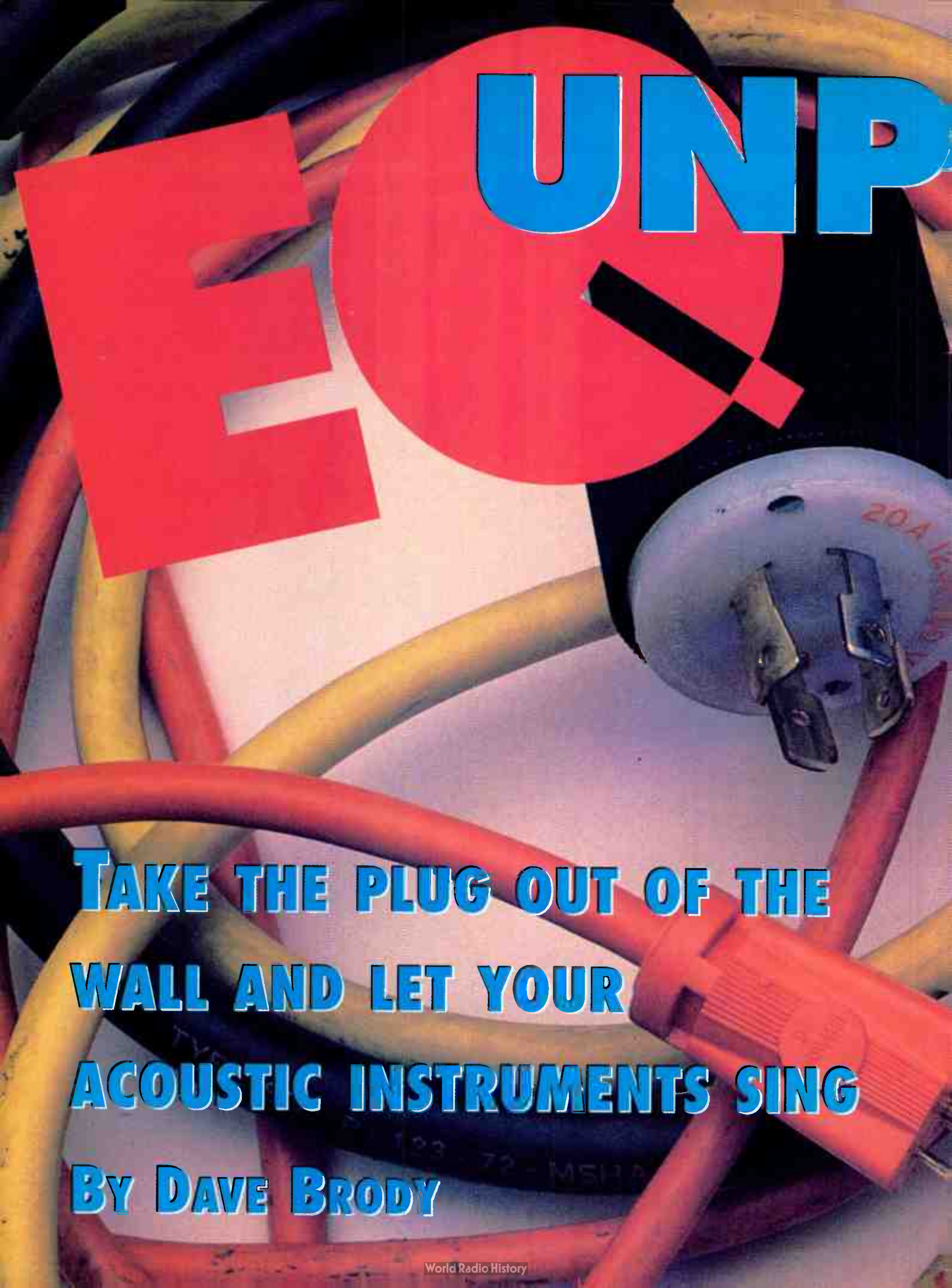


EQ UNP

Take the plug out of the wall and let your acoustic instruments sing

By Dave Brody

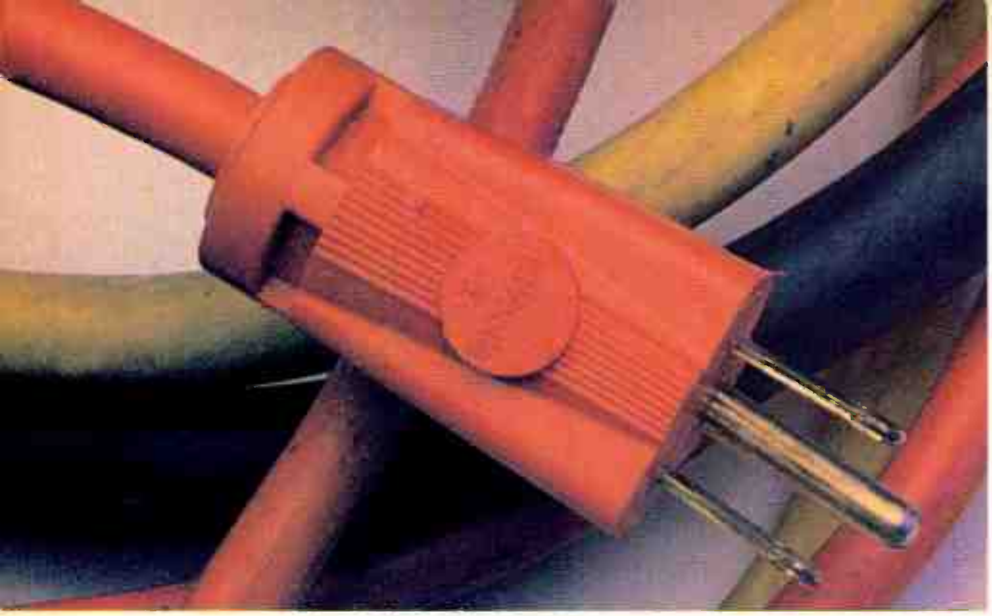
LUGGED

AN ACOUSTIC CLINIC FOR THE DIGITAL AGE

They're baa-aack!

Just when you thought you could impose order on the audio universe with one click of the MIDI panic button, those marvels of 19th-century engineering, the acoustic instruments, have returned with a vengeance.

Aw, c'mon; you remember them. They came in funny-shaped cases, required constant attention to intonation, and were generally operated by hypersensitive folks with spurious and intermittent personality disorders. Yet despite their cranky nature (both player and played) they could produce gorgeous, constantly changing waveforms reflecting subtle, intuitive musical nuances spoken directly — and in real time — from the soul of the performer.



Are you ready to capture these gems with all the clarity that the new generation of project studio gear is capable of?

Well, by the end of this series of articles you will be. Over the next several issues, we'll present an acoustic recording clinic — not to tell you what to do (you're the artist), but to help you expand the colors of your sound pallet. We'll give the most attention to pop music, but we'll not fail to give good insights into orchestral instruments as well as into the musical rainbow of ethnic instruments through which "world music" flows.

HERE WE GO

The rise of the project studio has changed — fundamentally and forever — the job descriptions of people who make recorded music. Often, roles have been merged. Writer, arranger, engineer, musician, and producer are frequently the same poor slob (you and me). Live, real-time performance — and the capture thereof via microphones — introduces enough variables to keep all those folks very busy. It's exciting, exhilarating, and a potential pain in the ass. But, as Hugh Padgham and Sting recently proved yet again, it's perfectly possible to surpass the finest studio while recording in your house. Doing so gives color to the performance and comfort to the performers. And the new generation of digital gear (ADAT, DA-88, et al.) is lightweight and roadworthy enough to do great remote multitrack work, whether via live performance or overdub.

Whereas recording sequencer-driven synths and samplers can approach being a science (at least in terms of controllability), the recording of acoustic music is still very much an art. Which means that there's always room for experimentation. (Just how much room, of course, is governed by either your client's budget or by your lover's patience with your addiction to your muse.) What follows are a few very general suggestions to help guide you as you transform experimentation into technique. The tips can't (and won't) turn you into a Bud Graham, a Bruce Swedien, or a Rudy Van Gelder overnight, but they're a place to start.

MIC PLACEMENT

In general, the closer the music is to pop style, the closer to the instrument the mic will be placed — which is to say, the less the room acoustics will influence the sound. Conversely, "serious music" (often — and perhaps wrongly — called "classical music") demands serious attention to how the instrument interacts with its performance space. This same principle also applies to recording the human voice: pop singers tend to sound whimpy when distance-miked, but operatic divas sound comical up close.

If you analyze it, you are giving the listener contextual cues by your choice of mic and placement. And, like it or not, the pop ear has been conditioned to expect close miking on featured musical elements with any associated "acoustic" space tacked on as an effect — usually by means of an artificial "chamber" device.

One nifty (and too often overlooked) trick is to don a set of cans and feed yourself the soloed output of a mic as you're trying to position it. In fact, the only excuses for not doing this are time pressures or the need to impress a client with your supposed superior experience. Personally, I'd go for the cans.

The project studio is changing the way we mic things, too. Many of us are perfectly comfortable with playing wired axes in the same room as console and tape machine. There is no law that says this can't produce a great "unplugged" track, as well, but there are some obvious pitfalls. You won't realize, for example, how piercing the sound of your hard disk spinning up really is until you've heard it intrude upon your beautiful acoustic guitar harmonics. So the general principle is to use the angle of rejection (the "back" of the cardioid pattern) to exclude those annoying fans and creaking floorboards. Those of us who sing and punch-in our own vocals usually have no choice but to do this (unless we've built one of Craig Anderton's stomp 'n punch pedals with an extra-long cable). But we can console ourselves with the fact that many big name artists — Paul Simon and Cyndi Lauper come to mind — often record their vocals in the control room. Some don't even use headphones; to minimize leakage, they rely on exact placement of the mic relative to the monitors.

MIC SELECTION

In general, the softer the instrument, the fewer harmonics it will produce. This tells you what kind of mic to reach for.

This is one reason why quieter instruments seem to like condenser mics, while louder instruments may prefer dynamic mics. If you're after short, sharp punch in the sound, use a rapid-response mic. Because condensers suspend their diaphragms within charged fields, they are able to respond quickly to changes in air pressure. But a big blast of wind can overwhelm them. So if you're close-miking drums and brass instruments, you might want to go dynamic.

Conversely, if you're looking for more resonance (more lyrical "sing factor") you might want to try a tube mic or ribbon mic. Ribbon mics take longer to get rolling (attack transients can become "ramped and rounded"), but ribbons tend to ring out a little longer. Take care, though; ribbon mics, especially older ones, are comparatively fragile and can be "blowed up real good" by too much air pressure. Don't use 'em close to drums, loud cabinets, or wailing horns.

The current trend toward "classic" mics has its dark side: there's a lot of noisy old crap out there in the marketplace selling for ridiculously high prices. Just because something has a tube doesn't necessarily mean it'll sound good! It does mean that you'll have to wait for it to warm up and stabilize (half an hour is not too long!), treat it gently (hoping the long string of previous users has done the same), and put up with hiss and hum (which — unlike that with solid state gear — will constantly fluctuate, thereby calling attention to itself). If you're not independently wealthy, develop a relationship with a reliable rental company that stocks a decent selection of microphones at reasonable rates.

"SEEING THE SPACE"

In general, the greater the number of mics used, the less accurate the psychoacoustic picture of the performance will be.

This probably doesn't mean squat in a pop context. And of course, combining sound fields by layering them within a multitrack mix can throw things quickly into illusionland. But if re-creating an accurate audio image of the performance space is a requirement, try to use as few mics as possible. The reason is obvious: each additional mic will contribute its own idea of arrival times from the various sound sources. Combining mic feeds will cause constructive and destructive interference of the waveforms, the end product of which will be blurring of the mind's vision. Resurrecting some of the techniques of the old masters of big band and orchestral recordings can help us a lot in the digital project studio age. More on this in later episodes...

THE GENTLE ART OF COMPRESSION

In general, the greater the number of elements in the final mix, the higher the degree of compression you'll need on featured instruments, especially if a rhythm section is playing.

Once again, this is a style-based bargain. The more your acoustic instrument has to compete with other colors for "ear-space," the tighter the range of amplitude variations it can permit. Another way of saying this, of course, is that the better the musical arrangement, the less you'll have to fight the compression battle.

One valuable old technique often overlooked these days is known as "chain compression," where the signal passes through two compression devices set differently from each other. The idea is for the signal to constantly float in one device set at a gentle ratio (say, 3:1) and then be passed to a second device set for true limiting (20:1, or so) whose threshold is adjusted to react only on peaks. This practice is also called "parallel limit-

continued on page 105

AN ANCIENT AND WEIRD WAR STORY

Submitted for your consideration: a twilight zone episode in the annals of recording as told by one who lived it.

Yep, the strangest one I ever did was probably the bagpipe record. No, it wasn't that traditional Celtic/Pictish marching o'er the hillsides stuff. Not even a city police battalion vanity album. He (who shall remain nameless here) asked us to record an LP of American top forty pop tunes of the soft-rock variety — backed by a full rhythm section, strings, and horns. And, of course, the melodies were to be performed upon that which has been described (appropriately) as a man strangling an octopus.

The concept was to telemarket the album through late-night, cheap-rate TV advertising — sort of a "Zamfir of the bagpipe" vibe (I'm talking Broadway tunes and Jim Croce songs here, okay?). The idea was, well, perhaps ahead of its time.

Nowadays, the technology exists to make a realization of this dream almost workable (if though still not in good taste). At the time — 13 years ago — two major obstacles reared their unsonorous heads.

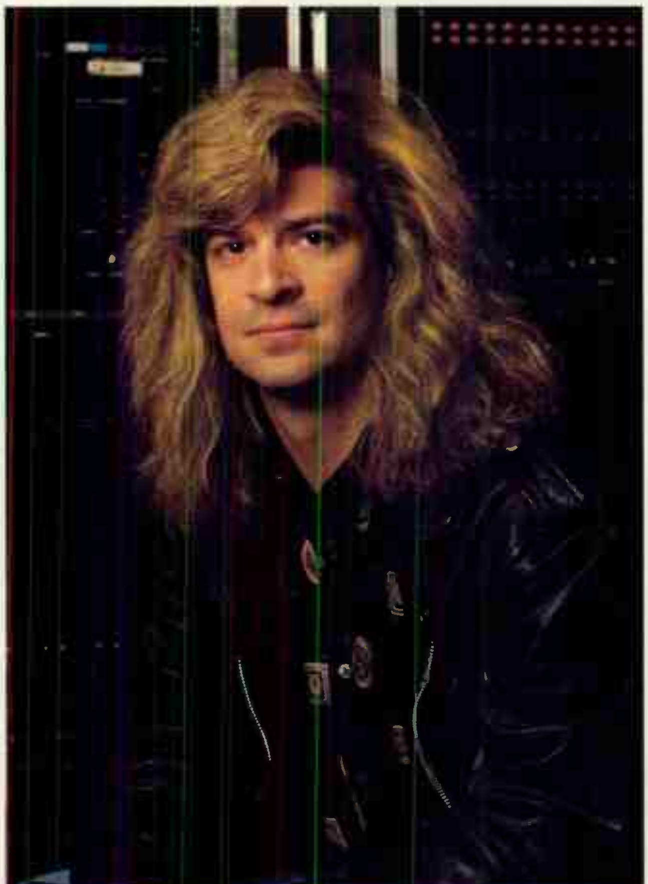
The first was the miking problem: Just how do you get a strong, focused audio image of an instrument that has several pipes sticking out of it — one of which is giving you melody (the chanter) whilst the others are honking long-drawn-out pedal tones with lots and lots of buzzy harmonics (the drones)? What we ended up doing was to stick the guy (complete with kilts and sporran) by himself in a studio with a large diaphragm hypercardioid condenser mic about four feet in front of him, chest high. Then we placed two omnidirectional pencil condensers high up in back of him with about seven feet between them, directed inwards at the pipes up over his shoulder. These were hard-panned left and right; the hypercardioid was brought up the middle.

The second — and by far tougher — problem had to do with intonation. Bagpipes do not produce a standard, well-tempered, Western chromatic scale. Not even close. To make matters worse, the gasbag-based design concept of the instrument requires the player to ramp-up the drones to playing pitch before each entrance — and down again in order to stop. [Imagine looping a Daffy Duck sample and pitch-wheeling it up and down.]

These complex issues were attacked (not to say "fixed") in a number of ways. The project's arranger was transposing charts like a madwoman up to the moment of the first take to get the tunes in keys centered around where the particular melodies seemed most in tune. The ramping-up problem was solved by supplying the lowly engineer (me) with a score so I could simulate an in-tune entrance by quick-fading the mics in after the piper's drones were up to pitch but before the actual melodic entrance. That left those all-too-frequent outrageously out-of-tune melody notes to deal with. We solved the worst of these by VSO-ing the machine to tune it to the bagpipe note, recording the melody on a separate track pair for each separate pitch center and compositing (via cross-fades) a new, somewhat more in tune, lead bagpipe track. All in all, a living, breathing hell.

These days, you would just suck the part out (in the digital domain) to your waiting workstation, perform the necessary fade-ups (complete with your choice of algorithmic taper), resample and time-stretch those regions requiring pitch adjustments, fly the fixed track back, and mix with ambience to unify the sound. Who knows? The result might even be listenable. *—Dave Brody*

BILLY JOEL'S PIANO MAN



DAVID ROSENTHAL PILOTS THE KEYBOARDS DOWN THE "RIVER OF DREAMS" TOUR.

BY DAVID FRANGIONI

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The call for the Billy Joel gig came just after the release of Red Dawn's first album on EMI Japan (right now the album is only available in the States as an import — check it out though, it's great). David only had a few weeks from the time rehearsals started to the beginning of the tour. Prior to rehearsals, he had to learn 54 songs in ten days. Once he had accomplished that feat, it was time to start designing the "rig of mortis."

Underneath the stage are his dual 25-space racks that contain the necessary fire power for Billy's tour (one of the biggest tours ever!). Rack Number One sports a Furman PL8 Plus, Opcode Studio 5 #1, Roland MKS 70 and D550, Dynatek 44 meg Syquest drive and CD ROM drive, Kurzweil K2000R {w/32 meg RAM}, PRO 1, HX, and AX Plus, Yamaha TX816, and a PS1 Surge protector. Rack Number Two contains another PL8-Plus, Opcode Studio 5 #2, Roland M480 48 Input Line mixer, Voce Micro B Organ module (with an extra one as a spare), (2) Yamaha SPX90II, Lexicon 300, PCM70, and (2) LXP-1, Eventide 910 Harmonizer, Strobe tuner, and a Yamaha DX7II on top of the racks. On-stage, there is a Yamaha KX76, Roland JD800, Korg CX3, Kurzweil K2000s (w/32 meg RAM), Moog Taurus pedals, (2) Mitigator MIDI controller pedal boards, 147 Leslie (for the CX3 and Voce), and an uninterruptable power supply. Of critical importance is a Joe Scacciaferro (of Triple-S electronics) designed custom foot switch, which acts as a "Panic" button for the entire system. In the event of a MIDI meltdown (stuck note, system crash, etc.). David can step on this pedal from the stage and send "all notes off" to all of the synths instantaneously. Computer-wise, he's using a Macintosh IIci with 20 meg of RAM, 1.2 gig external HD, 120 internal HD, and a 60 meg external HD. Software includes Studio Vision, Galaxy Plus Editors, Sound Tools all controlled from OMS. Galaxy edits and stores all of the patches in the setup. Should any of the gear go down, David can simply reload the patches back into the offending synth(s) quickly and efficiently. This also applies to those times when David must use rented gear (i.e., *Saturday Night Live*).

This combination of gear was chosen for its ability to recreate the rich keyboard textures found on all of Billy Joel's albums from the last twenty-plus years. Fortunately, David had the bulk of this rig together before he even got the Billy Joel gig. Even though he hadn't toured in the last few years, his band Red Dawn kept him busy in the studio. It was during this time that he began checking out the Opcode Studio 5 MIDI processor/patch bay. The Studio 5 is a 15-port MIDI patchbay with virtually unlimited processing power. Up to six Studio 5's can be daisy-chained together to create a 1440 MIDI channel system! David is using two of them (plus one for a backup) in his 480 MIDI channel rig. Each one is customized with expanded RAM and new ROMs, which allow the units to be linked together without the need of a Macintosh (*Editor's note: Opcode has since released the Studio 5LX which has all of these features*). The Studio 5's make it possible to immediately access any of the 54 songs that Billy might call at any time during the show. It acts as the "brain" of the system, remembering all of the settings for patch changes, volume, MIDI processing, splits, as well as the mapping of all devices in the rig. For instance, on "Piano Man" David plays the Mandolin part, which consists of straight Mandolin samples (on MIDI channel 1) and tremolo Mandolin samples (on MIDI channel 2). To switch between the two samples, he converted the sustain pedal to send MIDI volume 127 to one program and MIDI volume 0 to the other program, always alternating between the two as the pedal is depressed. This allows him to play either part by simply stepping on

Photos by Julian Jaime

The *EQ* Guide To Live EQ

Photo by Steve Jennings

No, we haven't stuttered. Here are some new takes on that ever-present tool engineers love to hate.

By Robert Scovill

■AFTER BEING approached by the editors of *EQ* to contribute an article for the magazine, my initial challenge was to come up with an inaugural subject. After a degree of pondering, it seemed so obvious — equalizers. Ah yes, equalizers, you know, those 1/3-octave, parametric, transversal, inductive, resistive, phase-distorting, phase-shifting, signal-colorizing, beautiful, horrible, cheap, outrageously expensive boxes of salvation that every engineer swears he will try to avoid using, but ends up counting on at one point or another during the music-making process.

In the next few paragraphs, I am going to explore a new twist on an old ideology. For example, the use of equalizers to effect the sonic texture of a given instrument without hearing the actual output of the equalizer. Sound too good to be true? Of course, the ideology that I am speaking of is the ever-mysterious use of the "sidechain" in dynamics processors. Are you enticed? Lets go...

CHAIN GANG

First a few basics. The idea of sidechaining a dynamics processor is not a new one. You probably see it used in studios, PA systems, and broadcast facilities every day in compressor/expanders, noise gate/duckers, and so on, via the key input switches to select internal or external sidechain circuitry. Deessers in and of themselves are usually nothing more than a frequency and Q-specific compressor.

With the advent of inexpensive graphic and parametric equalizers, it is possible to set up a number of frequency-dependent dynamics processors without potentially breaking the bank. I often incorporate these hybrid systems in my live setups, as well as in the studio, and in this article I will show you some ways to take these setups to a higher level of effectiveness. In fig. 1 I have shown a common layout for a contemporary compressor. Let's examine that diagram and then we are going to construct our own frequency-dependent compressor using any old equalizer.

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Compressor/limiters such as Brooke Sirens's DPR402 offer many variables on this theme. Using the barrier strips on the back of the unit, you can access many different points in the circuit to create different types of sidechains. It has proven to be a very handy unit to have around for specialty situations such as the ones discussed in this article. Note that the equalizer can really be of almost any quality because it does not exist in the audible signal chain.

A good example of a use for this type of setup might be with a frequency-unfriendly guitar solo. Find the frequency that is really bothering you, then go to your sidechain, insert the equalizer, and boost that frequency. Adjust your threshold, attack, and release times to taste, and I predict you will be stunned by the results. Notice that you do not have to listen to all the negative effects of using poorly designed equalization on your guitar because, as you see in the diagram, the equalizer is not in the audible chain — you just hear a smoothed-out response of the source, and the sound of your favorite compressor working.

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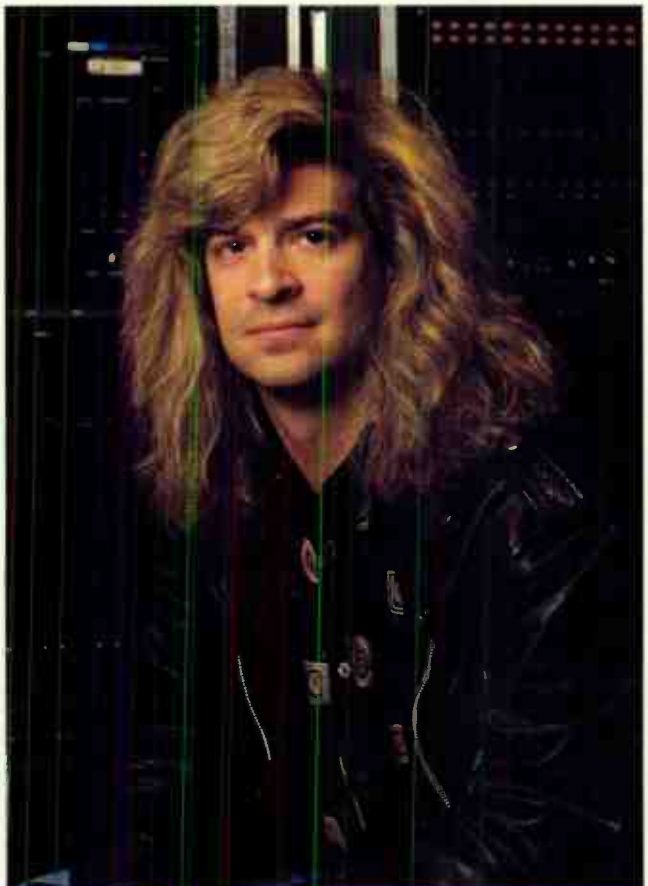

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Photos by Julian Jaime

the sustain pedal (to switch between samples).

In addition to the Studio 5's, David is using two Mitigator foot pedals. The Mitigators have a bank of five assignable pedals for each of their 128 presets. Each preset can then be put in any order within one of three lists. For example, each preset can be named (and viewed from a BIG display) by the song and each song could have up to five different patch configurations. David would call up, say, "Goodnight Saigon." This would send the appropriate program change to the Studio 5's. Now he can still send five more program changes as needed without having to leave the "Goodnight Saigon" preset. This allows each song to require only one preset. In addition, the order of presets can be arranged in any order and played back as one of three lists. In David's setup, he has one Mitigator programmed to cycle through the songs as the set list is written. The other Mitigator serves as a backup and is programmed to access any of the songs in alphabetical order. Therefore, he can instantly go to any song should Billy stray from the set list.

As far as the keyboard programming went, Rosenthal once again met the challenge head-on. He created most of the sounds from scratch, rarely using factory sounds except as a starting point. Instead of taking his Mini Moog on tour, he multi sampled his own custom Mini Moog sounds into the K2000. The Accordion on "Scenes from an Italian Restaurant" was made in the dressing room thanks to T-Bone Wolk (Billy's bass player and part-time accordion player) and a portable DAT machine. Again, multiple samples were taken and played back through the K2000.

Rosenthal can't say enough good things about the K2000. "I absolutely love it. This is the keyboard that I've been waiting years for. I went into the studio and sampled direct-from-digital (via the AES/EBU bus on the Kurzweil) all of the background vocals on 'River of Dreams.' We only sequence one song in the show and that's because of the real characteristic choir sound on 'River.' The vocals were originally recorded onto a Sony PCM3348 digital 48 track and mixed to two tracks. I went in and took the stereo mix digitally into the K2000. During the show, I trigger Studio Vision with a MIDI Key to start the song. The click (which is a shaker sample) goes to drummer Liberty Devito's headphones via one of the Kurzweil's separate outputs. The background vocals go out of another pair of outputs to

IN A BAND VAN

the FOH and out of a third pair as a monoblend to the monitors. Two-thirds of the way through the song, the whole band stops and Billy plays around with the audience. MIDI keys allow me to start and stop the sequence at will, because I never know how long we'll stop for. When Billy sings the pickup phrase, I manually kick the sequence back in. The K2000 and Studio Vision allow me to easily pull this off every night."

David also designed Billy's MIDI rig on stage. It consists of two MIDI'd Steinway Grand Pianos with a Korg O1W/Pro on each side of the stage. All of these controllers (the two Korgs and Steinways) are routed to two JLCooper MSB+'s. From there, they go to two Roland MKS-20 piano modules. This setup allows for maximum flexibility as any controller can access the MKS-20's at any time. In addition to MIDI, each Steinway also has a microphone and a Helpinstill pickup. This allows house engineer Brian Ruggles to combine any of the three elements into the FOH mix.

Although David can take the credit for designing his own rig, he's quick to point out the importance of his tech Jerry Pratt. Jerry's been with him for three and a half years and has seen the evolution of his keyboard rig. As soon as the rig is designed, Jerry puts it together, does all of the soldering, wiring, and grounding. And off they go — wherever that may be.

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IN A BAND VAN

THE *EQ* GUIDE TO LIVE EQ

Photo by Steve Jennings

NO, WE HAVEN'T STUTTERED. HERE ARE SOME NEW TAKES ON THAT EVER-PRESENT TOOL ENGINEERS LOVE TO HATE.

BY ROBERT SCOVILL

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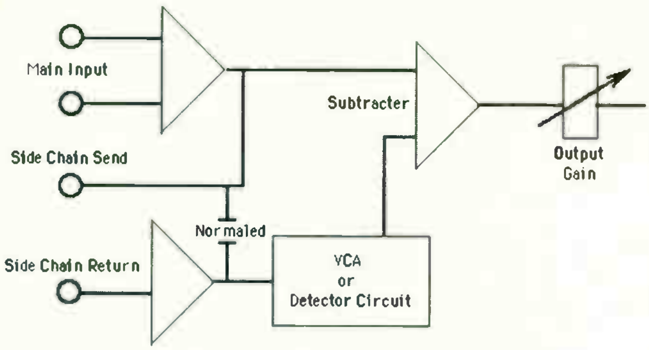

Figure 1

low on the fader (below halfway) you can be making changes of 20 to 30 dB with a small fader movement, whereas if you are at the top of the scale, you are only making 2 and 3 dB changes with the same amount of movement. You can, however, use this to your advantage, depending upon how you set up the gain structure of that fader and the effect you are trying to achieve. If you patch in an equalizer after the fader or even use the channel equalization that we mentioned above, you now have a frequency-dependent, fader-controlled compressor. With a little practice you can come up with some pretty spectacular results with this setup. Remember, though, that as you pull the drive fader down, the instrument or vocal will get louder.

Okay, it is now time to apply the Robert Scovill credo: "Anything worth doing is worth overdoing" (well, at least once). If you substitute programmable equalizers (such as the RANE MPE28, for example) into the sidechain inserts mentioned above, via MIDI program change you can then set up different frequency compression for a given song or part of a song. Also, if you use the MPE28's input level programmability, you can even dictate threshold changes through MIDI program change. It should be noted though, that you work in pretty low resolution when adjusting threshold this way (±12 dB in 2 dB steps). Alternatively, if you are lucky enough to have some spare tc electronics model 1128 programmable 1/3 octaves lying around, you can use these in the same way but have the equalizers chase SMPTE (provided they are outfitted with the SMPTE option).

Also, for all of you automation groupies, if your console is outfitted with automation you can automate your drive fader. Those of you without automation can print the moves of your equalized compression drive fader to tape, then patch the tape output into the sidechain return of your compressor to dictate the compression. This has the added benefit of always being a part of your slave or working reel, so you can carry that compression drive with you wherever you work. These principles could also be applied when using a programmable attenuator such as the Niche. By using a sequencer in conjunction with its continuous controller faders, it could be considered an additional source of automation. The possibilities are really as endless as the breadth of your imagination.

Let's switch modes now and take a look at using these techniques on noise gates. A potential application here would be in attempting to replace recorded drums with

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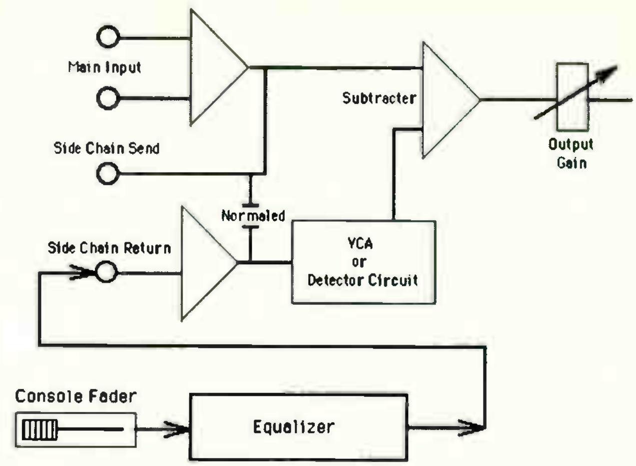

Figure 2

samples. Not an enviable position to be in, but if you must, the use of equalizers patched into the key input can make your life a bit more bearable. A frequent problem in this situation is that there is a lot of bleed between microphones, and that can result in inconsistent gating. This can make triggering very difficult because minute gate openings can cause phantom triggers of your sample. Achieving a threshold that works 100 percent of the time without missed gate openings or phantom triggers can sometimes be virtually impossible. By taking advantage of the programmability of the equalizers and by being selective in your frequency boosting and cutting, you can adjust it to the point where the snare, the kick, or whatever you want is the only thing opening the gate. I used this technique once to pull four toms and a cowbell out of a poorly recorded stereo group of toms and cymbals. It was purgatory maximus (pure hell) I assure you, but in the end the task was accomplished.

The majority of contemporary noise gates do offer a high- and a low-pass filter in an attempt to achieve the results mentioned above, but rarely are they adequate for such a situation, as I am sure anyone who has tried it will attest. I should note, though, that at the AES this year in New York, I saw a new dual gate by Aphex (Model 622) that addresses this problem by offering variable Q on those filters. I must admit that the gate performed marvelously.

This approach also works very well if you are using a MIDI noise gate to develop a note-on/note-off from a recorded source to trigger your samplers. Of course, you have to allow for the MIDI delay if you are combining sounds, but we will leave that for another article, okay? Thank you.

These techniques can be a lot of fun to work with and develop. Don't be afraid to experiment. They can be used with duckers, expanders, compressors, noise gates — literally anything that gives you access to the detector circuits of a given device. In the past I have used them on everything from selectively gating recordings of drums for sample replacement, to selectively compressing sections of a final mix, to creating really bizarre compression effects. You can get a lot of bang for your buck, and their effects are only limited by your imagination. And just think, now you can get some good out of that dusty old equalizer and at the same time improve the performance of some of your favorite compressors and gates. What a deal!

Robert Scovill is currently gearing up for a world tour with Rush. You can reach him on America Online, screen name "SCOVI."

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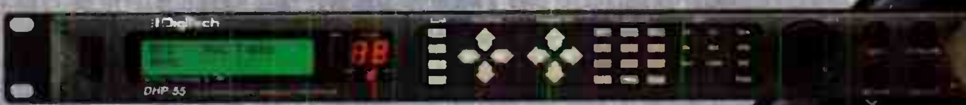
The ultimate in instrument harmonizers, the new DigiTech DHP-55 is the first five-part oversampled intelligent harmony processor designed especially for studio and instrument applications. Based on an innovative dual-micro/dual-DSP architecture, the DHP-55 can operate either as a true stereo or mono signal



processor. Proprietary technology allows the DHP-55 to provide a host of functions including intelligent five-part harmonies multi-octave pitch-shifting and, for the first time ever, true polyphonic (chordal) harmonizing.

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CIRCLE 21 ON FREE INFO CARD

Studio Recording on Tour

STUDIO NAME: Goin' Mobile
BASE LOCATION: Boston, MA
TOURING RADIUS: Mostly Northeast corridor, with occasional jaunts to Florida and Mississippi
MAIN MAN: Lonnie Bedell, owner/operator
CREDITS: Bill Bruford's Earthworks North American Tour, Throwing Muses (Sire), Bim Skala Bim, T.H. & the Wreckage, Rambo McGuire Ministries, Worpraise Records, TAANG! Records, Videocraft, Drew Townson, Robin Danar, Sidney Burton, New Alliance Productions, Bebop Productions, Fort Apache Studios
CONSOLES: Seck 1882 (main); Hill Multimix (sub)
MONITORS: Yamaha NS10M; Realistic Minimus 7; AKG K141 & K240 headphones
RECORDERS: Tascam MS16 1-inch 16-track with dbx noise reduction; Sony TCD-D3 DAT; 24-track Alesis ADAT
OUTBOARD: Ashly SC-66 stereo parametric EQ; Yamaha SPX90 digital reverb/processor; Deltalab Delay; Ashly SG33 stereo gates; Symetrix CL150 compressor/limiter; Tristech stereo gates; Yamaha GC2020 stereo compressor/limiter; dbx 166 stereo compressor/limiter
MICROPHONES: Shure SM57, 515, SM58, and SM81; AKG D112; Audio-Technica ATM11 and PZM; Sony ECM-23F; Sennheiser 421; C-Ducer Contact Pickup; Countryman DI; DOD 265 DI
CABLES: 400 ft. 16-pair Snake in 100 ft. lengths; 32 channels of transformer splits
VIDEO: Color monitor and video camera (also accepts 75-ohm composite video)
POWER: Tripplite Power Stabilizor on all AC lines; truck requires 120 VAC 5 amps single-phase for technical; separate 15-amp circuit for HVAC
EQUIPMENT NOTES: Bedell states: Goin' Mobile fills the void between 24-track semi-trailers and setting up a backstage control room on your own. We prefer to keep the equipment list and prices moderate, and rent for the occasional huge job rather than penalize the smaller client. Most of our work is multitrack, and the NS10M's serve well and are recognized by our independent engineer clients. The recent addition of an ADAT to the local rental scene gives us the ability to offer 24-track digital at an extremely competitive price.

A STUDIO ON WHEELS THAT MAKES THE BAND IN A VAN FEEL RIGHT AT HOME

Photos by Lonnie Bedell

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Eric Marienthal – Musician
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Tom Coster – Musician
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Brian Wheat – Musician
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Tom Size – Recording Engineer
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CIRCLE 82 ON FREE INFO CARD

Get the Most Out of Wireless Mics

As wireless prices plummet, even the band in a van is getting into the act

By Steve La Cerra

■NOT TOO LONG ago, wireless microphone systems were used exclusively by successful touring bands with fat budgets. But over the last several years the cost of this technology has decreased, and as a result, its use has become more widespread. But a wireless mic system involves a sensitive type of technology that works best under *controlled* conditions. Unfortunately, the live concert situation is anything but controlled. The musician or engineer cannot manipulate such variables as weather, crowd noise, or radio frequency interference. But they do have power over their equipment, and by knowing how to get the maximum performance from that equipment, can avoid problems.

A basic understanding of how wireless technology operates can help us exploit the capabilities of any given wireless system. A wireless microphone system consists of three basic components: the microphone, the transmitter, and the receiver. The transmitter is usually quite small and is housed in the body of the microphone, making the mic larger than the body of a wired mic (a wireless lapel or headset mic will usually have a wallet-sized belt pack that contains the transmitter). The system works much like a low-powered radio station. The signal picked up by the mic element is sent to the transmitter, where it is converted to radio waves and then broadcast to the receiver. The receiver picks up the signal and converts it back into audio (just as your radio does), which will ultimately be sent to a mixing console.

This sounds simple enough, but in practice is not so easy. There are many sources of interference to a wireless system, just as there are many sources of interference to your television and radio reception. In crowded city areas the airwaves are cluttered with police band communication, aircraft signals, satellite signals, and CB radios, plus all the television and radio station broadcasts. These sources of interference are generically referred to as RFI or Radio Frequency Interference, and all represent possible obstacles to clear reception in a wireless system. While you may not be able to stop all these phenomena, you can minimize their effect.

POWER UP

Probably the most important thing that can be done to help a wireless system is to make sure that the battery inside the transmitter is fresh. Together, the mic and transmitter make up the first stage of the signal chain; if the signal is not strong and clean at this point, it never will be anywhere else down the line.

Vega's VX-20 (top) and Samson's Synth Series Wireless Systems

Although different systems will eat up batteries at different rates, you can expect approximately three to four hours of quality transmission when using an alkaline-type battery.

A system where the mic element is a condenser will use the battery to power both the transmitter and the mic element itself, shortening battery life. But if the mic element is dynamic, the battery will be used only for the transmitter, and will last longer. If the unit has a "standby" mode, the life of the battery can be increased a bit by using this mode when no one is actually singing into the mic.

So when do you change the battery? It depends upon how critical the situation is. I recommend installing a fresh battery before every show. If this sounds like too expensive a proposition, get three rechargeable batteries and rotate their use. During the show, you can have one powering the mic while a second charges, and still have the third ready in case the first goes belly-up during the performance. Do not forget to physically mark the batteries with numbers or letters so that you know which is which.

The next step in avoiding problems is to do a preshow check of the stage for "dead spots," which are locations on the stage where the receiver will not pick up the transmitter. They can be caused by visible interference or metal obstructions between the transmitter and receiver, but also by multipath reflections of the transmitted signal within the area of use. Sometimes, housing the wireless receiver in an all-metal rack will produce dead spots,

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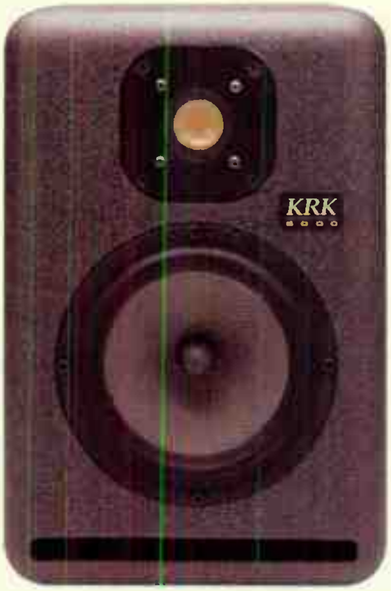

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CIRCLE 37 ON FREE INFO CARD

depending upon the location of the antenna(s).

To identify dead spots, a person walks around the stage area while the engineer monitors the mic signal for "dropouts" (interruptions in reception of the signal). If the engineer consistently hears a dropout from a particular stage location, the person speaking into the mic marks that spot on the stage with an "X" and directs the performer NOT to stand in that spot.

Locating the wireless receiver as close as possible to the stage helps reduce dropouts caused by "out of range" conditions. In an arena-type situation it might be possible to locate the receiver as far as two hundred feet away from the mic/transmitter combination, but this is asking for trouble. Instead, position the receiver close to the performer, and then run a cable to the destination of the signal (e.g., a mixing board). Some wireless microphone receivers have both a microphone level output and a line level output. Take advantage of this and use the line level output. Since it is a much stronger signal than the mic level output, it will be less subject to interference and can be sent down a longer length of cable without degradation.

RECEIVER RHAPSODY

Where possible, I like to use a diversity receiver. This type of receiver has two separate antennas plus a detector circuit that constantly monitors the signal strength at each of them. Once the detector has found the stronger signal it instantaneously switches this antenna to the receiver, thus maintaining a clear signal even when the performer is moving around. For best results with a diversity system, locate the two antennas at least several feet apart. Note that just because a wireless receiver has two antennas does not mean that it operates on the diversity principle.

While you are dealing with the receiver, make sure that the antennas are fully extended. Do not cut them shorter or modify them in any way. Wireless systems are designed with specific antenna lengths for optimum broadcast reception. Give the antennas a clear "line of sight" to the transmitter. Also make sure they are not sitting within a rack enclosure, as this can also cause interference. If there are cables connecting the receiver to the antennas, be certain that they are of the coaxial type (RG-58 or RG-59), as these are less subject to RF interference. *Do not* use standard audio cable to connect the antennas to the receiver. Check to see that the ends are securely terminated with high-quality connectors and be aware of the quality of the cable: some cables are more thoroughly shielded than others, and this helps to reject RFI.

If there are several wireless systems in use simultaneously, check the broadcasting frequency of every system. Each must have a different frequency, or the systems will interfere with each other. Most manufacturers label their transmitters and receivers with the actual broadcast frequencies, or may provide color coded dots on the components of the system. In the latter case, be certain that only one system of each color is in use at a given time. If there is a backup wireless mic on the same frequency, it should be turned off until it is actually being used for singing. Systems from different manufacturers will usually be incompatible, and even different systems from the same manufacturer are

IN A BAND VAN

likely to be incompatible due to varying broadcast frequencies or the noise-reduction techniques employed.

Be aware of within which frequency band the wireless system is operating. Most wireless microphones broadcast in either the VHF (Very High Frequency) band or the UHF (Ultra High Frequency) band. As you might recognize from your television set, the VHF band is more crowded than the UHF band. Also, UHF mics operate within a frequency range that has been set aside by the FCC for this specific purpose. As a result, the UHF systems are much less subject to interference from outside sources. Now here is the bad news: While VHF systems cost anywhere from about $200 and up, UHF systems start at around the $3000 mark. There is no such thing as a free lunch.

On the receiver, there will usually be a knob labeled Squelch. This is a type of muting circuit applied to the audio output. Adjust the squelch so that the receiver mutes the audio output in the absence of signal from the mic. Be careful of adjusting the squelch too high or the receiver may mute the output when there is signal coming from the mic (thus cutting off parts of the audio).

When shutting the system down, always shut off the receiver first and then the transmitter. If you turn off the transmitter first, the receiver will be left searching for a signal and will emit a loud, harsh type of sound much like that you hear when tuning a radio to a frequency where there is no station.

One of the most recent developments in wireless technology is the "multi-range" system. This type can have up to four transmitters broadcasting (on separate frequencies) to four receivers that are housed within a single-rackmount enclosure. Each of the four receivers has an exclusive audio output so that separation of the channels can be maintained. Multirange systems greatly reduce the amount of hardware needed to simultaneously run several wireless systems.

Remember, a wireless mic can only sound as good as (never better than) the wired version of the same microphone. So in choosing a wireless mic it is important to start with a model that has high-quality audio in the wired version. Once a clean start to the signal chain has been established, careful planning and attention to detail will make the use of wireless microphone systems consistently successful.

Sennheiser BF1051's Wireless System

New Gear For Your Next Your Gig

Mating Season

The new Monitor Mate Personal Monitoring System from Pertek Engineering provides you with a personal mix of mic/line sources in either a recording or live environment. The Monitor mate will drive a "Hot Spot" monitor of up to 30 watts and also has a stereo headphone output for both live and recording applications. Monitor Mate provides adjustment of both balanced mic and speaker line inputs. It provides you with control of the balanced mic level output, as well as a stereo line output of the mix. And the compact device clips easily to a mic stand. The controls consist of Mic/Line Mix; bass and treble (±20 dB), Mic Out Level, Direct/Processed, and Line/Speaker Input Level Select. Frequency response is 20 Hz–20 kHz, signal-to-noise is around 70 dB, maximum speaker output power is up to 15 watts RMS into 4 ohms, and the unit weighs in at just four pounds. For more information, contact Pertek Engineering at 22431-B160 Antonio #459, Rancho Santa Margarita, CA 92688. Tel: 714-858-1685. Circle EQ free lit. #116.

Pertek's Monitor Mate

MORE EQUIPMENT THAT'LL HELP YOU BRING OUT THE BEST IN ANY BAND

Shape Of Things

The low-profile Elite EX-350M stage monitor from Yorkville features a specially angled horn for better onstage throw, while allowing you to show off your boot tops. The 350-watt Elite offers an efficient 102 dB sensitivity from a 12-inch driver in a horn enclosure. The unusual angled cabinet design makes it possible to add more EX-350M's for a tight floor-monitoring array. The Elite 350 features a built-in stand mount, allowing it to be used for other PA applications as well. A specially designed P-350 Elite processor can be used for optimum full-range sound or as a crossover with Elite subwoofers. The transferrable Yorkville two-year unlimited "even if you break it" warranty is also featured. For more details, contact Yorkville Sound, 4600 Witmer Industrial Estate, Unit #1, Niagara Falls, NY 14305. Tel: 716-297-2920. Circle EQ free lit. #117.

Yorkville EX-350M

The Next Stage

Stage 33 from Samson features an exclusive new microprocessor true diversity circuitry employing computer-based technology that consistently scans incoming signals from two front-mounted antennas at ultrahigh speeds and selects

Samson's Stage 33

the optimum channel for clear, fail-safe reception over maximum distances. The system also features dbx noise reduction. The HT-3 hand-held and CT-3 belt-pack transmitters offer Samson's extended 17-hour battery life plus an array of onboard controls that include power on/off, audio on/off, mute and audio sensitivity, plus a selection of popular mic elements. For complete details, contact Samson Technologies, P.O. Box 9068, Hicksville, NY 11801. Tel: 800-328-2882. Circle EQ free lit. #118

You Got The Power

Whitenton Industries has added the onePower SL-20A to its S-Series product line. The SL-20A is a low-cost 20-amp power relay box. It connects to a signal output from any onePower CQ Series product via six-wire phone cable, and turns on or off in sequenced coordination with the CQ system. The SL-20A is also able to work with any voltage input from 12V DC to 24V DC, so it can operate with other manufacturers' controllers and power supplies. Up to five SL-20A's can be controlled from a onePower CQ-15A by connecting power supplies to the power Pod outputs on the back of the CQ-15A. For additional information, contact Whitenton Industries, 7320 Ashcroft, #104, Houston, TX 77081. Tel: 713-772-1404. Circle EQ free lit. #119.


Whitenton Ind. SL-20A

POWER TO GO

BGW has introduced the second model in its Performance Series of power amplifiers — the Performance Series 1. This general-purpose amplifier is a lightweight, two-rack-space unit that provides 1500 watts per channel. Features include quiet, forced-air cooling, a full complement of LED indicators, detented front panel gain controls, and XLR and 1/4-inch inputs. Five-way binding posts are provided for reliable speaker connections. The amplifier circuitry offers exceptionally wide bandwidth and the entire unit's basic circuitry is designed as one field-replaceable module. The Performance Series 1 retails for $799. For more information, contact BGW Systems, 13130 Yukon Ave., Hawthorne, CA 90250. Tel: 310-973-8090. Circle EQ free lit. #120.

VALLEY-ABLE PROCESSOR

The Model 730 digital dynamics processor from Valley Audio allows complex, multithreshold, multiratio dynamics functions to be combined. It performs as a digital stereo compressor/limiter, expander, sibilance controller, and gate, and accepts every form of analog and digital interface currently in use. Parameters added to traditional threshold, attack, release, ratio, and gain controls include setpoint, slope, range, predelay, delay, delay hold, mix, and stereo spread. For more detailed information, contact Valley Audio Products, 9020 West 51st Street, Merriam, KS 66203. Tel: 800-432-9412 Circle EQ free lit. #121.

E-V/Dynacord DRP 10 Reverb

■DYNACORD'S DRP 10 is a reverb and multieffects processor specifically designed to suit the musician who uses digital effects on-stage or in the studio. The front panel offers direct access to presets, arranged in eight groups according to the musical instrument — vocals, keys, guitars, bass, drums, brass, string, "special" — to be processed. This makes navigation of the 240 factory presets and 259 user presets very quick. Those who process sounds from a more technical view may prefer the DRP 15, which uses similar hardware but has the algorithms arranged according to function — reverb, delay, and so on.

FACE VALUE

The DRP 10's display is the same as that of many other current processors: a large LED for preset numbers and a two-line LCD display for names and descriptions. A single control to the left of the displays adjusts the stereo input level and is located alongside a single eight-segment level indicator. On the right of the display are the nine buttons for selecting groups of presets, each with an LED to indicate the current selection. Above the ninth button are three additional LEDs that indicate edit mode, MIDI data reception, and effect bypass. The large PARAMeter knob is a rotary encoder that also incorporates a multifunction push-switch. On the right side of the unit are six dedicated function switches and the power switch. The layout is clear and easy to learn. The 18 front panel controls are very functional and multifunction switches are kept to a minimum.

The rear panel includes the stereo inputs and outputs, separate level switches for input and output level, the usual trio of MIDI connectors, remote footswitch and pedal connectors, a ground-lift switch, and the IEC power connector. The inputs can be switched between nominal levels of +4 dBu/10k ohms and -6 dBu/500k ohms to accommodate pro-audio equipment connection or direct input from an instrument.

The output is selectable between +10 dBu and -0.5 dBu to interface with any pro-audio or project-studio mixer. The internal power supply is truly universal, capable of running on any 50 to 60 Hz AC voltage between 90 and 250 volts. This means the DRP 10 won't quit during an outdoor gig at a festival where everything is powered by a tired old diesel generator.

Pressing the PARAM knob (or double-pressing the group-selector button) recalls the selected preset, and presets are serially recalled in each group by rotating the PARAMeter knob or pressing the Select buttons. Storing a modified preset is simply a matter of pressing the Store button, selecting a destination preset-number with the PARAM knob, and pressing the Store button again. Pressing any other button will abort the edit.

The PARAM knob also goes to an X10-step mode when turned while depressed. This really speeds selection of presets or parameters and becomes second nature almost immediately. The Edit button also jumps between primary parameters, while the two Select (and) buttons scroll through all of them. A dual footswitch can be used in place of the Select buttons to really speed up your parameter editing by keeping one hand on the mixer and the other on the PARAM knob. The bypass button and LED indicator are labeled EFF OFF (this probably doesn't sound as rude in Germany). This mutes the output if the Original (direct) signal level is set to zero.

MIDI functions are comprehensive, if not extensive. An LED on the front panel is illuminated when MIDI signals are received and the DRP 10 includes an Option mode that displays the (accurate, if not friendly) hexadecimal code for incoming MIDI messages. Up to four parameters can be assigned to a MIDI controller. (A fifth external control can be assigned to a foot pedal.) The actual value of a parameter is not displayed — perhaps because Dynacord expects musicians to listen and not care if the reverb is 1.5 or 5 seconds long, as long as it sounds right.

The unit's sound quality is very good, offering full, rich reverbs; subtle to excessive flanging, phasing and chorusing; clean delays; and a few lesser effects. The parameters are well thought out and appropriately named, with a range of control that runs right to the edge of excessive.

The reverb parameters can be kept simple or be extended as required. The control over details such as early reflections, reflected/reverberant ratio, and room proportions are impressive in a low-cost processor. This allows the user to create reverb effects that include the impression of sitting front and center or near the back corner of a concert hall. There are ten different reverb types — room; chamber; halls 1, 2 and 3; plate; spring; reverse; left > right; and expand — but I

Road Test

MANUFACTURER: Electro-Voice, 600 Cecil Street, Buchanan, MI 49107. Tel: 616-695-6831.

APPLICATIONS: Reverb and multieffects for project studios and sound reinforcement.

SUMMARY: Easy to use reverb processor with low noise and a practical range of effects.

STRENGTHS: Excellent construction; good range of control over effects; three year warranty.

WEAKNESSES: Restrictive multieffects architecture and an inadequate distortion algorithm.

PRICE: $1250 **EQ FREE LIT. #:** 122

By Wade McGregor

BETA *Bio*

ROGER LINDSAY, ***SOUND ENGINEER***

HOMETOWN:
Liverpool, England

CLIENTS:
Sade, Mark Cohn, Basia, B.B. King, David Gilmour, Frank Sinatra, Frank Zappa, Joe Jackson, et al.

ON GETTING STARTED:
"As a keen young roadie in Liverpool in 1967, my family thought I would do it until I was 21, then 'get a haircut and a proper job.' Thanks to my wife's patience, some talented artists and a little luck, the former is infrequent and the latter has evolved into a long and enjoyable career."

ON BETA MICROPHONES:
"Sade's voice is unique — very subtle, very difficult to capture. When the Beta 58 came out a few years ago, it established a new industry standard. We're currently using the new Beta 87 Wireless. Its response is amazing — studio quality in live performance."

OTHER SHURE FAVORITES:
SM98A ("Great for drums."), SM91A ("Outstanding for kick drums and piano.") VP88 Stereo ("I use it for cymbals and percussion racks, and it's great for house tapes."), Beta 57 ("Superb for snare, bongos and congas.").

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CIRCLE 52 ON FREE INFO CARD

found the room simulations to be a bit too similar in their impression of room shape.

Small room ambiences are very good, with a range of control over the coloration of the sound that allows "reverb" to be used to increase the apparent size of an instrument without reducing its presence. Reverb times can be misleading, however, because the room size will vary the RT60 without changing the displayed value. The reverb algorithms share an unnatural cyclical "oing" on longer decays, but this is only noticeable when the reverb tails off after the music stops. Changing reverb-time parameters is quiet and immediate, but changing other parameters, such as predelay, can be very noisy.

The delay algorithms are clean and have the added advantage of using time or beats-per-minute as the measure of delay time. Changing delay time doesn't cause any clicks or distortion in the audio output, just a bit of pitch shifting while turning the knob.

The Long Delay algorithm includes regeneration settings, but the left and right output delays are only switchable between mono, ping-pong, or two fixed relationships of stereo that are not very mono compatible. I found this to be a major restriction on the way I use stereo delays — for instance, when creating specific rhythmic effects between the two channels.

There is also a Delay Line algorithm that could be used as an architectural delay in many sound-reinforcement applications. This algorithm allows the user to select between settings in milliseconds, feet, inches, meters, or centimeters. The distance units are only accurate at 71 degrees Fahrenheit, as there is no provision for setting the ambient air temperature (which temperatures directly affects the speed of sound in air). Changing units also changes the steps (resolution) of delay time (i.e., one centimeter is equal to 0.029 ms at 71 degrees F). The centimeter-unit setting offers time resolution nearing that of dedicated architectural delays!

The modulation effects (phasing, flanging, and chorusing) are usable and include a range of control right up to the edge of feedback. This puts the unit on a par with many of the processors in this price range. The pitch shifting is also typical of lower priced units, usable only in small increments for solo applications that are intended to remain realistic. The distortion algorithm may be useful in a few situations, where it may fatten/fuzz up an anemic instrument, but generally sounds worse than a cheap effect pedal.

There are six single effect structures: Direct Only (a digital volume control); High Quality Reverb; High Quality Modulation; Pitch Shift; Long Delay; and Delay Line. There are also four multieffect structures: Delay + Reverb; Pitch+ Delay + Reverb; Modulation + Delay + Reverb; and Distortion + Modulation + Delay + Reverb. Chaining of multiple effects is fixed, and the user merely selects a preset that includes the appropriate effects and modifies the parameters to suit the application. This simplifies the process but may restrict the more adventurous user. The algorithms, however, are chained in very practical patterns. They include useful feedback and paralleling but do not, for instance, allow equalization of the feedback loop.

The DRP 10 is supplied with a user manual that includes a listing of all the presets. The manual also includes a complete service manual with schematics, diagnostics, specifications, and a parts list. The algorithms are briefly explained, but excellent diagrams of the internal signal flow are included, which is especially useful for the

The Cool Idea Factory

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Technically, Jam Man is a Digital Delay/Sampler (with MIDI too). Just play a line and then get crazy with multiple sound loops – 8 secs worth (up to 32 secs optional). Grab riffs on the fly and create instant loops and then switch between 'em. Jam Man even lets you lock your loops to your tempo. You'll be doing grooves so cool they just might be illegal. Of course, MIDI heads can sync-up a sequencer too. You can also s-s-sample s-s-s-sounds and play 'em backwards.

Jam Man is from Lexicon, the same techno-gurus who make Alex and those serious studio processors. Wanna' know more good news? Jam Man doesn't cost big bucks. Get into a Lexicon dealer today 'n CHECK. IT. OUT. Ye-e-ow!

lexicon

HEARD IN ALL THE RIGHT PLACES

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chained effects. The manual (dated 5/15/93) is marred by a mediocre translation from German, and included one page that was still completely in German. [Editor's note: E-V informs us that this initial production run manual has been revised to completely remedy any translation errors.]

Currently, the user must scroll through presets until he or she finds a chain that is suitable. It would be very helpful if the algorithms used in each preset were listed, because to build a new preset the user must start with an existing chain and modify its parameters as required.

The internal construction, exterior fit, and dark gray finish are to a very high standard. These details, in combination with a full three-year warranty, are likely to make the DRP 10 a reliable part of any audio rig on the road or in the studio. Handy features, like the ground-lift switch on the rear panel to help avoid ground loops, are part of the unit's well-engineered design.

The DRP 10 is a capable effects processor, offering a wide range of effects, sensible parameters and excellent audio quality. The limitations of a few algorithms could be addressed in firmware upgrades to the unit (the reviewed unit was Version 1.01), but E-V hasn't made any promises about this.

The straightforward interface and ease of programming could make the DRP 10 highly sought after for live performance work, and the arrangement of effects into groups of aforementioned sound sources will be appreciated by those who want the sounds without the acoustical jargon. The few deficiencies of the DRP 10 may not be a problem in your application, so try one out at your local E-V dealer next time you're shopping for effects. Dynacord may have created just what you need.

IN REVIEW

Alesis BRC Big Remote Control

EQ LAB REPORT

MANUFACTURER: Alesis Corp., 3630 Holdrege Ave., Los Angeles, CA 90016. Tel: 310-558-4530.

APPLICATIONS: For engineers who rely heavily upon their ADATs.

SUMMARY: Increases the number of functions able to be performed by the Alesis ADAT and provides a convenitent way to link multiple units together.

STRENGTHS. Resolves many synchronization problems; seamless punching in and out (also can be done by ADAT alone); digital track delays.

WEAKNESSES: Uses an open-loop system, which means that the BRC doesn't know if the commands given to the ADAT are actually carried out.

PRICE: $1995 **EQ FREE LIT. #:** 123

I FINALLY GOT MY PAWS on a BRC from Alesis. As you probably know by now, the BRC allows you to add functions to the ADAT system. Some of these additional functions are simultaneous control of multiple ADATs; pitch control (vso); autopunch, preroll, postroll, and looping; SMPTE, MIDI, and external time control; storage of setup information; transferal of digital tracks from one deck to another; and sample-accurate offset recording [pitch control and looping can also be performed by ADAT alone]. Nice piece of gear.

I couldn't wait to hook it up, so it is a good thing that it only took a few minutes to get everything going. A long 9-pin sync cable goes from the BRC to the first ADAT. If you have more than one ADAT, short 9-inch-long 9-pin cables then connect the rest of the decks in series. After the sync cables are connected, the fiber-optic cables are connected. While the sync connections end at the last ADAT in the chain, one more fiber-optic cable is connected from the output of the last machine back to the input of the first machine, completing a loop for the digital audio.

MULTI-ADAT CONTROL

The BRC allows you to have individual control over all tracks on up to 16 ADAT machines. This gives you 128 tracks of digital audio. The track control portion of the BRC displays 32 tracks at a time. To allow control of the full 128 tracks, the

display "pages" through the other tracks. Pressing one button lets you view the status of tracks 1–32, tracks 33–64, tracks 65–96, and tracks 97–128. Functions such as All Input, Auto Input, All Clear, and All Safe affect all machines simultaneously.

AUTO LOCATE AND AUTO PUNCH

Besides locating to zero, you can locate to 22 other points that can be stored in memory. These sample-accurate locate points can have names instead of numbers, so that you can locate to "Verse A1" or "Chorus 2." These names can be stored on the ADAT tape so that you don't have to re-enter them next time you work on the same tape.

One of the great features of digital recording is its seamless punching capabilities. There is actually a cross-fade performed between what used to be on tape and what you are about to record on tape. During the punch out, the cross-fade process reverses and you have a smooth transition back to the signal on tape. The BRC will let you change the cross-fade time (between 10.67 ms and 42.67 ms) to fit your recording requirements. You can also store the punch-in and punch-out points in location memory and allow the BRC to perform the chore over and over again perfectly. If you don't like the spot you picked, it can be changed in increments as small as one sample. Preroll and Postroll times can be set in 1/10-second increments. Looping allows you to play over a section of tape as many times as you want, automatically.

Rehearse lets you see if the punch will work well before you actually erase what is on tape. When the part works well, just select Auto Punch and do it for real.

Group record is a function that works in conjunction with Looping and Auto Punch. It lets you loop between two locate points and automatically record each pass on a different sets of tracks, as well as individual tracks. This is a function that has drawn many people to hard-disk recording systems. Now you can have it with a tape system.

PITCH CONTROL, DIGITAL BOUNCING & SYNC

Pitch control determines the speed (sample frequency) at which the tape plays back. There is a display area dedicated to pitch change so you can tell what is going on at a glance without paging through other displays. The range is –300 cents up to +100 cents. If you want to record at 44.1 kHz, just set the pitch control at –147 cents.

With the addition of the BRC, you can copy any track from one machine to any track on another machine. You can fill a tape with vocals or solos and then assemble a composite track by copying sections digitally to one master track with no generation loss.

With the BRC, you have several ways to synchronize your ADAT with the outside world. If you are using SMPTE as your basic sync method, you no longer have to lose a track on the ADAT to SMPTE. With the BRC, you can generate SMPTE that will be in sync with the absolute time present on the ADAT tape. If you don't want the SMPTE to be the same as the absolute time, you can add an offset to the generated SMPTE. I used the BRC as the master and synchronized a Fostex G24-S. The Fostex had the built-in synchronizer and was set up to chase a "code only" master.

The BRC generates SMPTE while it is fast winding. The timecode rate (frequency) remains the same, but the frame number jumps to reflect the correct tape position. It works much the same as VITC (Vertical Interval Timecode) coming from a video machine. Because of this feature, the Fostex would chase right along with the fast wind of the ADAT. When the ADAT got to the locate point, the Fostex would get there at about the same time. [*Editor's note: Alesis reports that with v1.04, the BRC can generate configurable (on/off) SMPTE and MTC in fast-wind mode.*]

With the same choice of machines, running the BRC as the slave was not as efficient. The Fostex did not spit (technical word) out SMPTE during fast wind operations, so the ADAT would not go anywhere until the Fostex reached the locate point and spit (there it is again) out the SMPTE of its current location. This sync hookup took twice as long as using the BRC as the master.

With the BRC operating as a slave, you have the choice of continuous sync, where the ADATs will synchronize with the SMPTE and change speed as necessary to stay locked, or of having the machine release to word clock or video sync once SMPTE lock has been achieved.

If you are using a sequencer that you want to run coincident with your multitrack, you have three choices. One: use the SMPTE out of the BRC and feed it to the SMPTE in of your sequencer interface and let the interface do the SMPTE to MTC (MIDI Timecode) conversion. Two: let the BRC directly generate MTC that can be fed into a MIDI interface that does not support SMPTE. There is a MIDI connector on the BRC for this purpose. Three: build a tempo map in the BRC (much as you would in a sequencer) and use MIDI Start and MIDI Clock to control your sequence. I tried SMPTE and MTC and they both work perfectly.

TEMPO MAPS

Tempo maps are necessary if you have an old sequencer that doesn't recognize MTC (MIDI Timecode) or if your sequencer does not have tempo-mapping capabilities of its own.

Before Mac-based sequencers, I used a Roland MC-500 for sequences. It wouldn't lock to SMPTE or MTC, so I would send SMPTE to a Roland SBX-80 sync box and do my tempo mapping in there. The SBX-80 would then send MIDI song pointer and MIDI clock to the MC-500 sequencer. When I switched over to Mac-based sequencing, I stayed with the SBX-80 for most synchronizing tasks. When working on a project that used different sequencers for different parts, the SBX-80 became the common denominator. The tempo-mapping capability of the BRC replaces the SBX-80 in this regard.

The tempo map allows 100 tempo or time-signature changes. Each song can have up to 9999 bars. Tempos can vary between 40 and 240 beats per minute. All reasonable time signatures are supported, as well. Tempo maps start at the Song Start offset point that you designate.

TRACK DELAY TIME

Here is one of those features whose purpose you can't figure out at first, but once you have it, you don't know how you got along without it. Here you have a digital delay on each track. You can delay up to 170 ms in one-sample increments. This would be perfect for Donald (Mr. Increment) Fagen. If a track feels a little "rushy," then delay it a little. If you want something to come earlier, then delay all of the other tracks. With the delay built in to the ADAT, there is absolutely no fidelity loss. Much better than having to run your favorite solo through a junky-sounding digital delay.

STORAGE OF SETUPS

All the settings of the BRC can be stored at the beginning of each tape. The

ALESIS AI-1 REVIEW

The Alesis AI-1 allows bidirectional transfer of two channels of digital audio between an ADAT and an AES or S/PDIF (RCA and optical) device. Sample rate conversion can be performed in either direction.

The AI-1 is a one-rack space add-on to the ADAT system. It connects after the BRC and the last ADAT in the chain (although the BRC is not necessary). Both the 9-pin sync cable and the optical cable need to be connected. Additionally, a supplied BNC cable must be connected between the AI-1 and the BRC to provide clock information when transferring to ADAT from an external source.

To operate the AI-1, all you have to do is select the source, the destination, and the destination sample rate. You can select any pair of adjacent ADAT tracks, the AES interface, or the S/PDIF interface. If the input you selected shows a solidly lit LED, then the input signal is good and you are ready to go.

According to the Lexicon LFI-10 (used to check digital I/O format and compatibility), both the AES/EBU and S/PDIF outputs are perfectly behaved. Sample rate flags, CRC, parity, emphasis flag, and word length are all sent and received correctly. Digital transfers were made successfully between ADAT and Panasonic 3700 DAT, Technics SV-DA10 DAT, Fostex D-20B DAT, Sony 7030 DAT, Sony 3324S 24-track, Mitsubishi X-86 2-track, Akai DD-1000 optical-disc recorder, Marantz CDR-600 CD recorder, Digidesign Sound Tools and Pro Tools, Apogee A/D and D/A converters, Wadia A/D converters, Roland 660 digital EQ, Roland 880 reverb, and the Lexicon 300 reverb. I also transferred a Sony F-1 tape to the ADAT using a Sony PCM-601 with digital out. The F-1 format's sample rate is 44.056 kHz and the AI-1 tracked it perfectly. I would have tried more, but that is all I had lying around.

Now let's talk about sample rate conversion performance. I first compared the AI-1 against a sample rate converter that cost $12,000. The AI-1 performed better. The input sample rate was automatically tracked from 50.8 kHz to 40.4 kHz. The output cutoff frequency was –2 dB at 20,703 Hz. The SIM machine said the frequency response was flat as a pancake and signal to noise was better than 93 dB.

The way I view it, you don't really have an ADAT system without including an AI-1. [Editor's note: For information on using the AI-1 to turn your ADAT into a data storage device, check out Craig Anderton's MI Insider in this issue.]

—Roger Nichols

"data" area you see at the top of each tape is reserved for that purpose. The data stored includes the 20 songs with their 20 locate points, names, tempo maps, pitch mode and pitch value, punch points, loop points, preroll/postroll times, track delays, tape offsets, generate sync type, external sync type, digital I/O configuration, track groups, and MIDI/UTILITY settings. The data is called the TOC or Table of Contents. It is stored on all writable tapes currently loaded. This is so that you can send a tape to a friend to work on and he can recall your settings to ensure a common work mode. Only one TOC can be stored, but one TOC can hold the setups for many songs.

SUMMARY

Well, I guess that about covers it. The BRC was well worth waiting for and should solve a lot of synchronizing problems in your project studio.

One word of caution. The BRC communicates with the ADAT machines as an "Open-Loop System." This means that the BRC tells the ADAT what to do, and assumes that the ADAT will do it. There is no "handshaking" that tells the BRC that the operation was successfully performed. This is the way normal MIDI communication works. The sequencer tells the synthesizer what note to play and never knows whether the note was played. In musical MIDI applications this is just fine, but in a multitrack/controller environment you have to be more cautious.

When a track is record-enabled at the machine instead of at the BRC, the track will not be indicated as record-ready on the BRC. If the machine is placed in record, this "phantom channel" will record without your knowing about it. Also, if an error stops one of the ADATs while you are recording, the BRC won't know it. The display on the BRC will keep advancing as though the tape were moving and the BRC will show the tracks as being in record, but the ADAT will be stopped in its tracks. [*Editor's note: Alesis says that the BRC's current software, v1.04, has some communication between BRC and unit.*]

This does not mean that you can't still use the BRC. You just have to be aware of this "single-ended" control mode, and operate your studio accordingly. Alesis may implement "Closed-Loop" communications in the future, but it may take another software modification to do it.

All in all, the pluses more than outweigh the minuses. Even with only one ADAT machine, the BRC is a "must have" item in the Alesis arsenal.

—Roger Nichols

D&R Orion Console

MANUFACTURER: D&R U.S.A. Route 3, Box 184-A, Montgomery, TX 77356. Tel: 409-588-3411

APPLICATION: Multichannel mixer for both acoustic recording and MIDI project studios.

STRENGTHS: Extremely low noise; versatile mixer.

WEAKNESSES: Insufficient front-panel silk screening to indicate full functionality of switches; lack of status indicators.

PRICE: $25,795 [32 inputs; 12 stereo effects returns; patchbay; 88 inputs in mix mode]

EQ FREE LIT #: 124

SHOPPING FOR A MIXER? There are plenty of manufacturers and models to choose from and certainly no shortage of hype. If you are currently in the market, you could be overwhelmed by the number of possibilities. This review takes a look at a legitimate contender in the field.

D&R manufactures its recording consoles in Holland and sells them directly to American end users via its offices in Nashville, L.A., and Texas. Its products carry a two-year labor and limited parts warranty; 90 days on faders, pots, and switches (smoking being the leading cause of these component failures).

THE GOOD NEWS

The ORION is very serviceable, has a substantial power supply, and (drum roll, please) boasts a noise floor that (spec sheet to spec sheet) exceeds that of the Neve V Series. This last bit of info was given to me by Paul Westbrook, the U.S.president for D&R. If I hadn't already heard the lack of noise for myself, I probably would've hung up on him.

The ORION is one console that really surprised me. With all modules assigned to the mix bus, unmuted and with module faders down and master fader(s) full up, the signal-to-noise

World Radio History

ratio is 95 dB, unweighted. That is quite remarkable for a console in this price range. If noise is your nemesis, you might just have less shopping to do.

The ORION was too large to bring to the shop for testing, so the board's owner, Michael Rubin, allowed me to open the hood at his project studio. This means that the tests were made while the ORION was connected to a complete system in Manhattan (RF capital of the world), on an upper floor with windows. All of these factors combined to increase the possibility of radio and television interference (RFI/TVI). Michael, who is not a technician, installed wiring made to his dimensions (and D&R's specs) by Clark Wire & Cable, Northbrook, Illinois (Tel: 800-222-5348).

GETTING FRAMED

The ORION Series comes in three frame sizes — 30, 38, and 51 — which allows for many, many options. For example, the 30 and 38 frames will hold 32 and 40 standard (mono) input modules, respectively. All frames come with six dual-stereo effects returns — a total of twelve stereo inputs. D&R will configure to your specs at no extra charge.

There is an assortment of configuration options for the ORION. Both the master modules and the patchbay (which is optional) can be located wherever most convenient. The patchbay option, however, must be configured at the time you place the order. The input modules can then be placed accordingly.

The four master modules feature separate control room, studio, and headphone level controls. The control room monitoring options are: 2-track A, B, and C, plus aux 1 & 2 (the likely choice for headphone mixes). And yes, Virginia, there is a mono switch and an alternate speaker switch. Both the studio and headphone sections have a headphone jack on the module. Each section has three monitor switches: One follows the control room monitor switch, and the other two select either aux 1 or aux 2.

There are eight aux submasters, a solo level control, a switch to select either solo in-place or PFL, a three-frequency oscillator, and a communications section. There's even a remote talkback-enable switch.

The rear of the console, behind the modules, is a mixture of XLR and 1/4-inch connectors. The patchbay interface is via DB-25, a 25-pin, gold-plated, "D"-shaped connector that is more common in Macs and PCs, where it is used as a data port. It is becoming

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D&R ORION

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very practical for audio applications because of its compact size and its ability to handle up to eight balanced channels. I've seen Neve use this connector as its patchbay interface, and it can also be found on both the Fostex RD-8 and the Tascam DA-88.

KICKING THE TIRES

The ORION features separate mic, line, and tape inputs. (Some manufacturers put a 30 dB pad in front of the mic input and call it a line input. This is not only a waste of gain, but can also be a real drag if the mic preamp quality leaves something to be desired.) Both the line and the tape inputs are balanced (yeah!).

At the front end of the microphone preamp is the SSM 2017, a custom IC from Analog Devices that's especially suited for high-gain, low-noise applications. It's the same chip that is used in the dbx 760X mic preamp (reviewed last month) and in the Amek BIG by Langley mic preamp.

The ORION is extremely versatile. It will be equally at home in both the traditional recording environment and the tapeless studio. By traditional, I mean a studio large enough to accommodate at least two dozen musicians with setups requiring at least that many microphones. For this application, a tape monitor section is key. Hence, the input module can serve two functions. The most common application is to route a microphone to the multitrack via busses, using the record fader to control level. For multitrack playback, a monitor fader can be routed to the mix bus. (These faders — one linear, one rotary — can be swapped.)

Lately, mixers with this split/dual-input feature are over hyped by advertising types claiming "twice as many inputs on mixdown." This may be the case, but it is certainly not a new concept. Keep in mind that the coveted "eight aux sends" become four sends for each half of the split.

There are three pairs of aux send pots. Both aux 1/2 and 3/4 can be driven by either the record or monitor path, pre- or postfader. The third pair of controls can be routed to either aux 5/6 or aux 7/8. These are postfader only.

The HI/LO shelf section of the four-band equalizer can be separated from the HMF/LMF bell-curve bands. All four are sweepable and, once split, can be inserted into the record and monitor path, respectively.

The EQ seemed to be very gentle to recorded tracks, even when at maximum boost. The most important quality that an equalizer in this price range can have for me is to be stable. This is accomplished primarily by using good pots. All the rotary pots and switches in the ORION are made by Alps. D&R makes the linear faders, but offers either Alps or P&G as options.

Considering this console's versatility, I found the lack of status indicator LEDs and the minimal silk-screened graphics to be very frustrating. For example, switches labeled "tape," "mix," and "group" do not indicate in which position this function occurs, or what the other option is. This can be a problem when an unfamiliar idiot/engineer like myself attempts to use the console for the first time. Hey, when was the last time an engineer walked into your studio and asked to read the manual?

Note that the LED metering is to World standards. A sine wave at a –6 reading on the peak meter produces +4 dBU at the output connectors.

FLOATING SUBGROUP SYSTEM

There are eight busses. In addition, each module has a direct output when none of the assignment switches are depressed. The floating subgroup system (FSS) is a novel idea that allows access to multitrack formats with more than eight tracks without patching. It appears on each module as two switches labeled "From Sub 1-2" and "From Sub 3-4." If, for example, you've got a great MIDI mix that needs to get to tracks 21 and 22 of the multitrack, simply assign all channels to busses 1 and 2, then, at modules 21 and 22, press "From Sub 1-2." It's that easy.

The return modules provide the maximum number of inputs with minimum of real estate. There are two stereo returns on each module. A 60 mm stereo fader adjusts the level for both channels. Each return has balanced inputs and a rotary gain control, plus HI and LOW shelving EQ. There are also three pairs of aux sends (1/2, 3/4, or 5/6), two pairs of busses (1/2 and 3/4), a mix switch, a balance control, and solo and mute switches.

The manual describes the function of each and every switch and control very clearly. The first time that I opened it was to check mic preamp circuitry. Unfortunately, the channel module schematic was reduced to fit 8x10 paper stock. It should have at least been a fold-out. There were no kinks in the translation, aside from a few spelling errors.

I did get to see how D&R got the noise floor so low. A pair of transistors is placed at the front end of a 5534AN guaranteed-low-noise op-amp. One transistor is used to sum the incoming signals and the other to cancel incoming ground-bus noise. It is an especially clever application of a common circuit used in higher gain applications as a transformerless mic preamp. Very resourceful.

There are a number of wiring diagrams detailing the optimum interconnection method. These are large and clear. I don't particularly care for the way the DB-25 connectors are wired to and from the multitrack. The connector is split, four inputs and four outputs, while both Tascam and Fostex keep inputs and outputs separate so that all eight inputs or outputs are on a single connector.

Channel modules are fitted with a connector to facilitate interfacing with automation systems from JLCooper, Optifile (France), and C-mix (Germany). D&R will soon offer three automation options: Power MUTE, Power VCA (level, muting, and dynamics), and Power Fader (very sexy faders that move). The last two should be ready by the first and second quarter of '94. A 64-channel Power MUTE is available now for under $1600.

SUMMARY

The ORION is made extremely flexible by switches strategically placed in the input modules. During the listening test, Michael provided me with eight tracks of a cappella vocals that had been recorded directly to ProTools. At one point, I had both the control room and mix bus levels at maximum. The input faders were barely up from Off when I pressed Play. Noise levels were so quiet that the vocals seemed to come from out of nowhere, smacking me in the face before I could turn them down. ORION's great noise spec is the direct result of concise and clever circuit design: as much as is needed, and no more. It will adapt to almost any situation you might encounter, but newcomers to your facility could need some hand-holding. I would like to see the features it has be enhanced, say with status LEDs, but that would, of course, raise the price. (More detailed silk screening might help.) I'm often called upon to reduce noise, but the ORION is one piece of gear that won't have EC tinkering with its gain structure. *—Eddie Ciletti*

REVIEW SHORT

Emerald Music Thru Box Three

"This isn't a product for everybody, but it can be an excellent solution to some basic MIDI problems."

A $199 MIDI thru box may seem like overkill for some situations, but for others the Thru Box Three (TBT) could be exactly what the MIDI doctor ordered because it saves the hassle of having to use a computer or patchbay to do routing and data distribution.

The single-rack space TBT houses two independent thru box sections. The 1x10 section has one input that feeds ten MIDI thrus. The 4x2 section's input can switch among any one of four different MIDI sources and drives two MIDI thrus. All 17 MIDI connectors are on the rear panel, as is the socket for a standard detachable power cord (included).

The front panel is uncomplicated: power switch, MIDI input data activity indicator for each thru section, pushbutton switch for the 4x2 section input selection, and a single-digit LED readout to show the selected input.

Here are some typical applications:

• When running a universal MIDI librarian on a computer, patch the computer's MIDI out to the 1x10 thru section to drive the instruments. Patch one 4x2 thru to the computer interface's MIDI in, and the four switchable inputs to instrument MIDI outputs. Switch between these when you need to send sys ex from a particular instrument to the computer.

• With a computer-based sequencer, distribute the MIDI out via the 1 x 10 thru section and use the 4x2 to switch among different MIDI controllers when recording tracks into the sequencer.

• Use the 1x10 outs to distribute MIDI control signals to the signal processors in a rack. Patch one of the switchable 4x2 outputs to the 1x10 input. Switch between various controllers for the processors — sequencer, "fadermaster"-type device, control pedals, and so on.

• For live performance, use the 1x10 section to send the master controller MIDI signals to offstage rack gear. The switchable 4x2 can select different controllers or sync sources.

For most applications, this box does either too much (if you don't need the switcher and rugged construction) or too little (if you need merging inputs and greater patchbay capabilities). However, plenty of live and studio MIDI setups require only something like the TBT to do all necessary switching and routing. For these applications, the TBT is less expensive than a MIDI patchbay or multiport interface, and because it has no microprocessor or software, seems particularly well suited for live use.

If you need a thru box, the TBT offers the benefits of rack-mount packaging and simple switching functions to go along with a generous number of MIDI thrus. This isn't a product for everybody, but it can be an excellent solution to some basic MIDI problems. *—Craig Anderton*

For more information, use EQ free lit. #125 or contact the Emerald Music Company, 392 Oak Knoll Rd., Ukiah, CA 95482. Tel: 800-828-3837.

First Aid for Loudspeakers, Part 1

How important is the lowly loudspeaker? To paraphrase a leading cultural icon: "If they don't sound good, you don't sound good." Trouble is, for many people loudspeakers seem complex and mysterious beasts, when in fact they're as simple to maintain as your average crew cut. All you need to do is follow these simple rules of thumb:

Photo and diagrams courtesy of JBL

Don't Let a Two-Year-Old Play With Them. I should know; I'm currently going through my second two-year-old. They especially like poking at that shiny silver dot in the middle of the woofer. (They also like seeing how many times your demo tape will wrap around the chair when pulled from the shell, but that's another article.)

So you say you don't have a two-year-old handy? Then how about a guitar player? Just kidding. But most clubs come equipped with drunks who make you long to be playing those *Romper Room* gigs. Beers poured into the drivers, cigs tossed in the ports, bottles thrown, stacks tipped over by clumsy feet — these are just a few of the regular hazards faced by the working speaker.

Loudspeakers aren't made of the most durable of materials. The cabinets and driver frames are usually strong enough, but the moving parts can be easily damaged. There are a number of different types of drivers out there, but since they all do exactly the same thing — turn electrical energy into mechanical energy and push air around — let's look at how a basic driver is built and how it can be damaged. (Though generally interchangeable, "driver" usually refers to the individual components, while "speaker" refers to the whole unit.)

First, let's discuss the architecture of a standard woofer. The coil in the middle of the speaker is where the conversion from electricity to sound takes place. A fine, varnished wire is wrapped around a form (usually a

Wait! Don't throw out that beloved, broken-down speaker yet. It may be able to live again.

BY CALIX LEWIS RENEAU

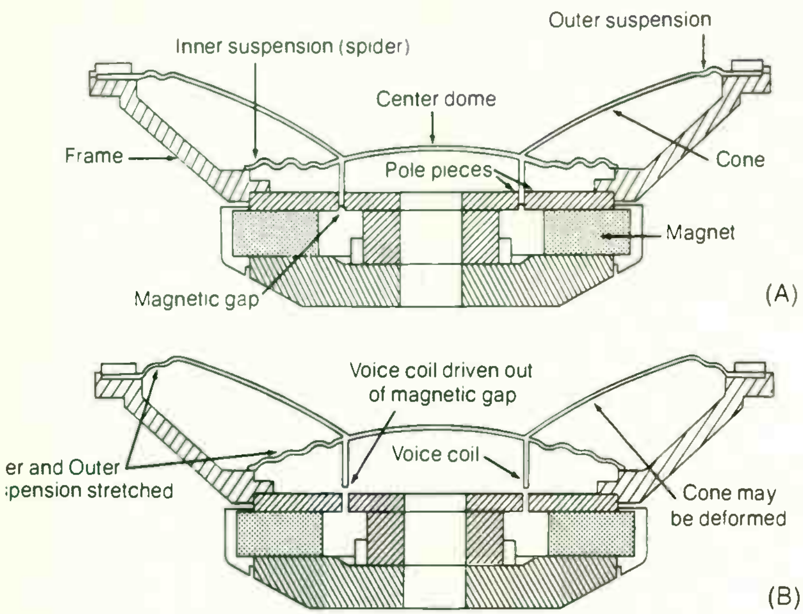
lightweight cardboard tube) and set in a fixed magnetic field. In most drivers this field is established across a gap in a metal plate with a tolerance measured in thousandths of an inch; the windings of the voice coil are held centered in this gap.

When an alternating current (your audio signal) passes through the coil, a magnetic field is created that moves the coil in relation to the fixed field. A dynamic microphone is built the same way, but works in reverse with the coil moving in the magnetic field to produce alternating current. That's why DJs can use headphones for mics in a pinch and you can use smaller drivers for miking — I've had interesting results with a 5-inch driver taped to a bass head. You can also turn a dynamic mic into a driver, but you'll most likely burn it out before you hear the sound; we don't suggest it but, hey, it's your party.

The coil is held in place by a flat or saucer-shaped piece of stiff cloth, called a spider, and by the cone itself, so that the coil is free to move in and out of the gap without rubbing. The center of the driver is protected by a dust cover (that round silver dot in the middle). The outside of the cone is attached to the frame by a flexible medium called a surround. In most professional PA speakers the surround is fabric or paper, while in most monitors or home speakers it is made of rubber or foam. A tweeter or horn driver is built much the same way, except the radiant surface attached to the coil changes from a cone to a diaphragm of some sort (the round dome part that costs so much more in horn drivers).

SO WHAT CAN GO WRONG?

The first thing that usually happens to a driver is that the dust cover gets dented. Purists will argue that this hurts the sound, but unless the listening environment is extremely crucial I wouldn't worry about it. With soft dust covers (such as felt and some kinds of paper) you can lightly press a piece of masking tape on the dust cover and pull the dent out. Failing that, you can hook a bent straight-pin



A standard (top) and overdriven speaker.

in the center of the dent and pull — gently, to avoid tearing.

With an aluminum dust cover, it's a bit more problematic. Aluminum, once bent, is not easily straightened and is more likely to crack than felt or paper, at which point the aluminum will start buzzing. In a pinch, a small amount of epoxy on a cracked dust cover will stop the rattle. Dust covers can be replaced fairly cheaply by a speaker reconing shop, although frequent replacements will weaken the joint between the voice coil and the cone (the highest point of mechanical stress in the driver).

If the dust cover is missing, you should have it replaced. The gap in the magnetic plate where the coil fits is extremely tight, and any small piece of flotsam that finds its way in will destroy your driver in a hurry. (I did have a friend dump a can of cut green beans into an open voice coil and still have the driver work, but I don't recommend it.)

The next most common result of physical abuse is a torn cone. This is not necessarily a bad thing. In the early days of rock and roll some artists would play guitar through speakers intentionally torn to get a very warm, mellow fuzz-physical distortion. I've even heard of speakers that had the cones soaked in water and dried to make them more brittle and "buzzy." Not a bad effect if that's the sound you're looking for. Many modern guitar drivers, in fact, are made with thin, hard cones that distort easily (such as the Celestions found in classic Marshall cabinets).

If that's not the sound you want from your main PA speakers, you still have a couple of options. If the tear is in the surround (the outside of the cone that connects to the frame) and is small enough, you should reinforce around the hole to keep it from spreading. Same with small tears in the cone. How? Pull out that epoxy again, mix some up, and spread it lightly on the cone around the tear. Don't epoxy on the surround, as epoxy dries stiff and the surround has to flex, but anywhere else on the cone is fair game.

In a pinch, a torn surround can be patched with watered-down white-paper glue, which remains somewhat flexible after drying. Just be careful not to put any pressure on the cone while it's drying, otherwise you might distort the cone, which can shift the voice coil into rubbing the gap. If you're worried this might damage the driver, remember — it's already broken! If you end up having to recone the driver, all of the guts (except for the frame and magnet assembly) are going to have to be replaced, anyway.

A third physical problem is less common but more deadly. A driver is glued together from the gaskets on the front to the magnet assembly. Under extreme conditions (such as loud extended jams), the glue will degenerate with heat and time. The most common glue joint to fail is also the most fatal: the glue that holds the magnet in place between the back plate and the voice-coil gap. When this happens, the magnet will slip over and pin the voice coil in the gap. This driver is trashed. (It can be rebuilt by demagnetizing the magnet, resetting the gap, remagnetizing the magnet, and then reconing, but the process is more expensive than just buying a new driver.) One easy way to separate a magnet is to drop a speaker, especially after you've been playing through it and the drivers are still warm, so be careful whom you let load out at the end of a hard night in the trenches.

Another point to watch out for is the spider becoming separated from the frame. When this happens the speaker will chatter horribly as the voice coil, free from constraint, bounces around inside the gap for a bit before burning out. Usually this also means a trip to the recone shop, but if the problem is caught before the spider comes completely loose and the voice coil is damaged, you can try to glue it back into place yourself. Once again, epoxy is your friend.

With high-frequency drivers, there's less you can do in the way of emergency repairs. For soft-domed tweeters that get dented, you can use masking tape (although I suggest avoiding the pin trick except for a last resort), but other than that, a broken high-frequency driver has parts that just plain need replacing. In many horn drivers, the diaphragm may be replaced by removing the driver section from the back of the horn and opening it up, but don't be sticker-shocked when you go to replace them. Horn diaphragms are amazingly expensive.

The only other driver you're likely to encounter is the piezo. Used mostly in PA applications, piezos are very-high-frequency drivers that vibrate a piece of electrostatically charged plastic instead of a magnetic coil. Piezos are expensive, shrill, and bulletproof. It's hard to blow one out, but when you do just throw it away and buy a new one. The piezo is the hamster of speaker drivers.

That's it for now. Tune in next issue for more speaker repair advice. EQ

UNPLUGGED

continued from page 63

ing" (in spite of the two boxes being connected serially). More common is recompression, where a track recorded through a limiter is recompressed on mixdown. Yes, each additional device does degrade the overall signal-to-noise ratio; so use these techniques judiciously.

Recently some new gear has appeared on the marketplace that allows you to selectively apply varying amounts of compression in different frequency ranges. Mastering engineers have been using "banded" compression for years. Actually, so have you; whenever you used a de-esser. You can create this effect by establishing a sidechain in which you follow an EQ with a limiter and mix the product back with the original signal. But please be warned: this approach is guaranteed to produce some sort of phase shift. You might like the result, but it's definitely an "effect."

THE LISTENING ENVIRONMENT

An acoustic recording can only be as good as your ability to hear it.

Recording an unplugged instrument gives you the chance to compare what the thing really sounds like with what you're making out of it. You never had that blessing/curse when recording your synths. And your listening environment had better be up to the task. If you're looking for an excuse to buy some more toys (and aren't we all?), the contemplation of an acoustic project is an opportunity born in the dreams of an equipment dealer. If you haven't done so already, go out and get yourself the most accurate monitors and the best power amp your credit card can handle. [Hint: do not buy either of these items from a "stereo store."] When shopping for amplifiers, remember that when used with common sense, high-powered amps do not blow out speakers. It's low-powered amps that do, by feeding ugly, squared-off waveforms into nice round cones and voice coils.

A number of years ago, Charles Benanty of Soundworks Digital Recording Studios in New York City coined the term "mothership" to describe the relationship of high-end recording facilities to project studios. His prophecy has proven to be right on. It is now possible to conduct live, real-time sessions in a number of locations simultaneously. Not only is it

possible, the biggest of the big are doing it. If you get your acoustic chops together, there is no reason why you can't find yourself jamming (and jam-syncing) with the major players.

Oh, and not to shatter your illusions, but it's fairly common to find artists retracking their parts — particularly vocals — in preparing mixes for video releases of "live to tape" concerts. May they do so in your studio at exorbitant rates.

In the next installment of our *EQ* Acoustic Clinic, we'll focus on the four main ways acoustic instruments produce sound — which will tell us how to go about capturing it. 'Til we meet again, keep your heads clean and your timecode synchronous. EQ

How Quiet Should Studios Be?

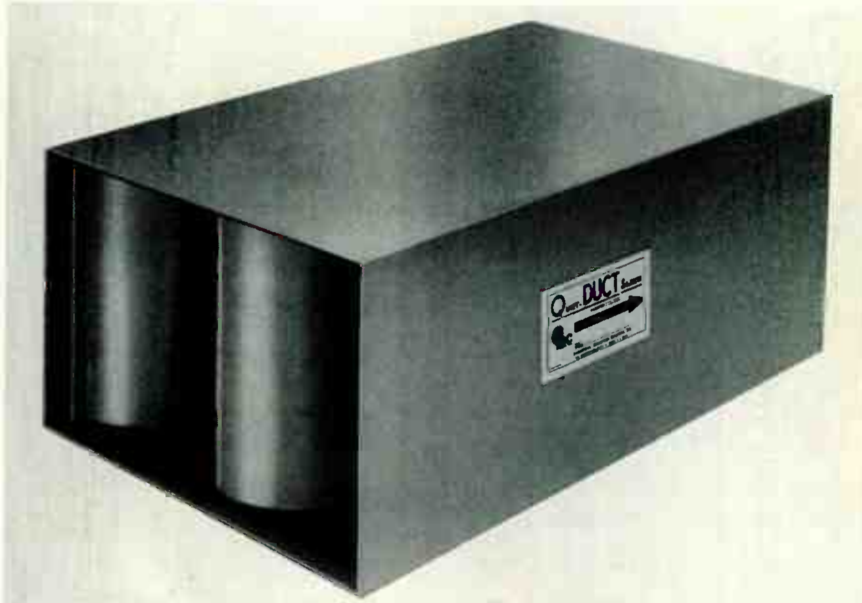


Figure 3. Typical Duct Silencer

Keep your sounds from bugging your neighbors, and keep your neighbors' sound from bugging you

BY JOHN STORYK AND BETH WALTERS

Every studio project starts with a look at the potential site for the project. Often there is more than one site to choose from and sooner or later the same questions come up:

How quiet should my studio be? How do I keep unwanted noises out of the studio? Where should I locate the studio? Or, in the case of an existing building shell, is this space big enough? Is it high enough? How do I start laying the studio out? And so on.

Although there is usually not an easy one sentence answer to these questions, there are certainly established guidelines.

It's simple enough to recognize that studios have to be quiet. You cannot successfully record live vocals or music on microphones that will also pick-up disturbing noise from outside sources, noisy air conditioning, neighbors, etc.

So how do we define quiet? As in most acoustic issues, measuring and defining quietness is a function of the frequency we are concerned with. The human ear does not hear all frequencies with the same efficiency. The most demonstrative graph of this are the famous Fletcher Munson curves that show equal loudness as a function of measured SPL values as function of frequency (see fig. 1).

It is interesting to note that the human ear does not hear very well at lower frequencies (or at very high frequencies). These curves, essentially, are the frequency response of the human hear. Equal sound pressure level (SPL) values (dB- unweighted) will be heard as equal loudness levels. SPL is a logarithm of ratios — a numerical value directly relating to a measured variation in pressure.

Loudness is a subjective value. To

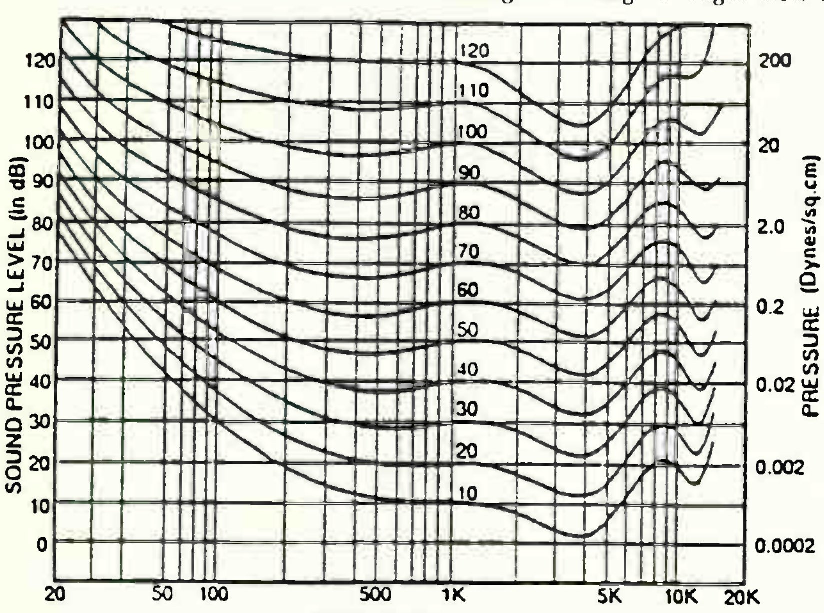


Figure 1. Fletcher - Munson Equal Loudness Contour Curves

accommodate this, we have developed various weighted curves for one value SPL — full frequency readings from sound level meters. These weighted values more closely relate to the human ear and its characteristics.

The more precise way to define quietness is not with a single value, but with octave SPL values measured (or defined) in a space. These values can be plotted on weighted curves (essentially reverse curves of Fletcher Munson) to give us more precise single values for quietness called NC values. An NC value is the curve which is completely above all measured octave band noise values in a space. NC values are 5 dB separated (see fig. 2), although it is often used to give extrapolated numbers in between the 5 dB curves.

Now that we know how to measure and define quietness, where do we go? The quietness of our studio is important in defining the necessary transmission loss (TL) values for all the boundaries. Selecting the walls, floors, and ceilings with the most efficient TL value will be discussed in the next column. It is similar in concept to Studio Doors (see August '93 issue of *EQ*).

The quieter the studio requirements, the higher TL values are required for your studio, which usually result in thicker and more expensive walls. Conversely, quieter surrounding spaces are beneficial. Modern commercial audio recording studios, where open microphones are used, typically need NC values in the low 20s. We are regularly seeing studios installed with NC values below 20 — and these are *very* quiet.

On the other hand, a small project studio with an isolated iso/vocal booth (low 20s NC value) could have a NC value in the high 20s or even low 30s and work perfectly satisfactorily. Although these numbers seem very close to each other, this difference in requirements can easily result in $10,000 of savings — particularly in HVAC costs (see below). To this day there are several very famous control rooms in New York City in which you can hear the slight rumbling of New York's subway system — yet, this noise never gets on tape in the more isolated recording rooms.

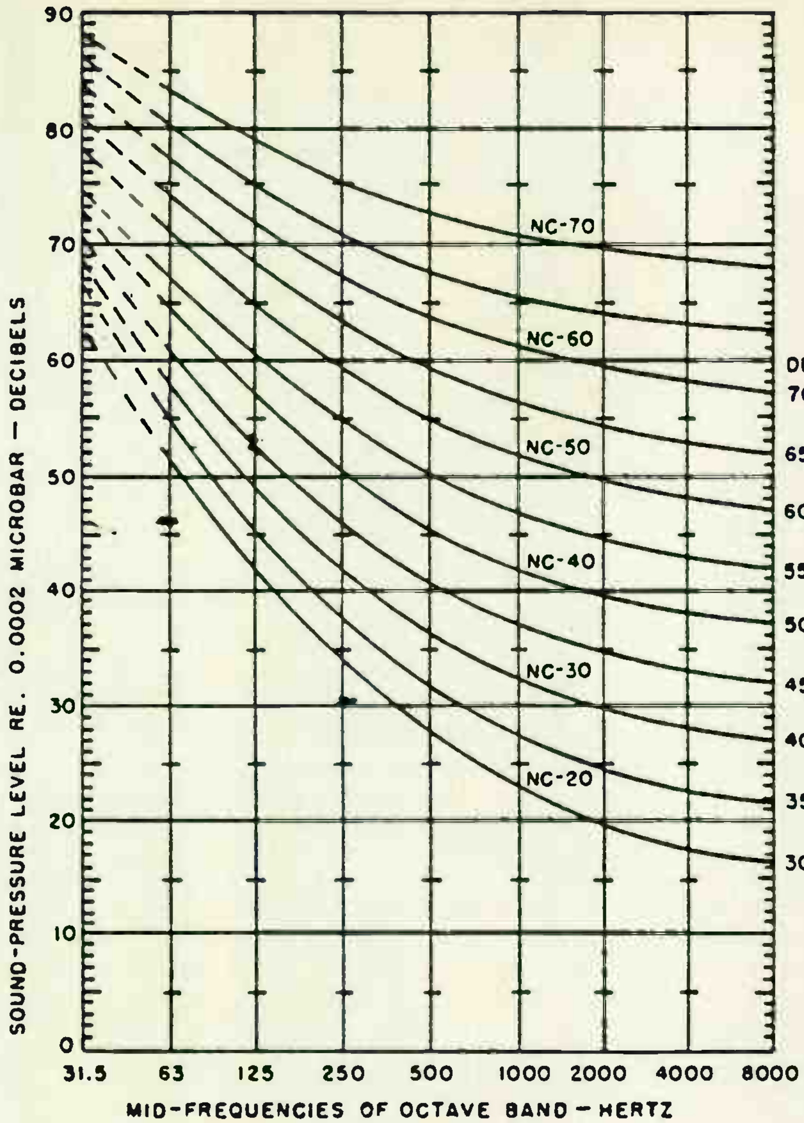


Figure 2. Noise Criteria Curves

HVAC SYSTEMS

Within the studio environment, it is quite common that the worst noise offender is not outside neighbors, but the studio's own HVAC (air conditioning) system. These systems will typically make noise for three types of reasons:

1. Low-frequency and structural-borne noise from vibrating equipment (typically 100 Hz and lower). Correcting this type of problem is usually accomplished by isolating large equipment on springs or away from the studio completely.

2. Mid-frequency noise that is typically generated from the equipment noise being "passed down" through the ductwork system. Lining the ducts, both supply and return as well as the use of duct silencers again in the supply and return) will usually solve this problem (see fig. 3). These silencers work similarly to mufflers in an automobile and, in fact, one type actually looks a little bit like one.

3. Higher frequency (1-2k) noises are typically air velocity noise — air passing through supply diffusor grills at velocities that are too high. Final exit velocities in recording spaces should not exceed 250 ft./min. in order to accomplish the required 20-25 NC levels. These are very low speeds, but must be adhered to. This is a guideline. Things change with height of the grilles off the floor as well as the type of grille.

Again, having a clear definition of the required NC value is important. The air conditioning industry has particularly well-documented information concerning noise generated by equipment as well as NC values for virtually every type of grille and diffusor made — corresponding to NC as well as octave based frequency information.

TIPS

1. The first step towards having a

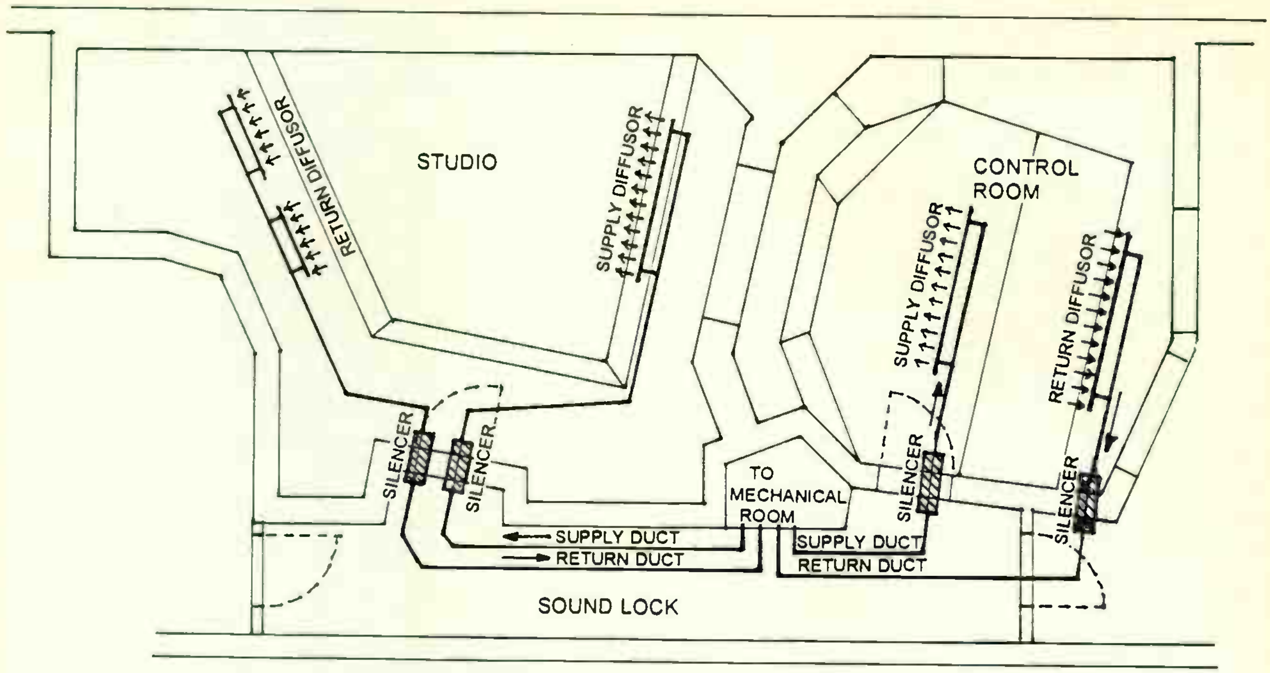


Figure 4. Duct Silencer Layout in ceiling of Studio Sound Lock

quiet studio environment is to select a quiet site. This seems obvious, yet is too often overlooked. Next suggestion, try not to put two "noisy" spaces next to each other in your facility layout. Again, this should be obvious, but is too often overlooked. High STC walls (necessary to deliver two low NC valued spaces that are adjacent) are possible - but *very* expensive.

2. In general, air conditioning ductwork will have to be larger to accommodate studio air velocities, typically 50 percent lower than those in normal commercial construction. Plan ahead for the necessary space for these.

3. Lining ductwork on both the supply and return systems is very important. It has been suggested that only the supply will deliver noise to and from the system due to the direction of the air. This is simply not true. Compare air velocities of 250 ft./min. with the speed of sound at 1130 ft./sec. or 67,000 ft./min.! The direction of the air flow is minor.

4. When installing duct silencers, try to install them in the middle of a wall, typically through a sound lock. Remember, ducts are acoustic holes in walls, and silencers are much heavier construction than normal ducts and will assist in acoustically "patching" these holes (see Figure 5).

As usual, good luck in design and construction. EQ

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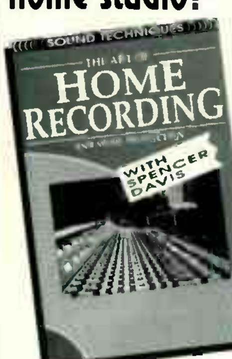

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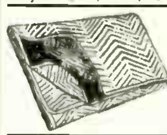

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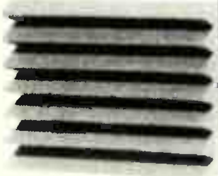

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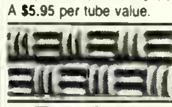

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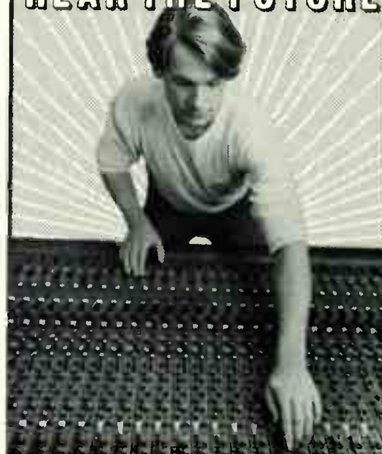

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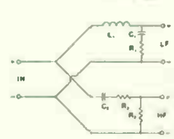

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Manic Compression, Part 2

More on squeezing your sounds in order to get the most out of your disk space

BY MARTIN POLON

The audio industry is facing a crisis of sorts in the '90s as various schemes for coding and compression of audio data are utilized to maximize space on recording media or in transmission systems. As we have seen, the push to use technology to eliminate "nonrelevant" signal information for space-limited applications has been more or less successful when used alone. Industry production centers are abuzz, however, about the possibility of problems when coding/compression systems are layered via multiple passes or when they are used in combination with other signal processing technology. These problems usually refer to digital, but some reports have come in involving analog technology as well.

The presence of psychoacoustic processing seems to be one of the key factors, but other elements are also present. The most difficult aspect of these interaction conflicts is that there are no absolute factors that could be isolated and corrected — at least not at this time. For the majority of users, the systems and technologies work flawlessly. But in those instances where there is an interaction problem, the audio quality is degraded — in some cases seriously.

In all the cases of compression/coding interference with the integrity of the digitally recorded signal or the creation of artifacts in the signal path, the efforts of many separate testers with various equipment options and hookups were utilized. Yet the recognition of impairment is usually confirmed aurally. The human ear remains the most useful tool for exploring the impact of compression tools, since the nonlinear characteristics of coding render conventional test equipment less than useful for the purpose.

Modeling the number of variables in everyday audio use requires testing of a nearly infinite number of combinations of coding schemes and different equipment. Some possible areas of concern would be any interaction or any combination of the following:

1. Single or repeated passes through the same compression/coding system.
2. Sequential passes through several different coding systems.
3. The use of equalization and analog signal processing before using a compressed digital recording scheme.
4. The use of equalization and analog signal processing after using a compressed digital recording scheme.
5. The impact of digital delay and signal phase shifting.
6. Digital-to-analog conversion, and the obverse analog-to-digital process, when the conversions occur repeatedly in the signal chain.
7. Digital-to-analog conversion and the obverse analog-to-digital process; especially when the conversions occur repeatedly in and around the signal-compressed digital recorder.
8. Audio delivered for project insertion, or other studio use via satellite, T1 telephone company data carrier, or other transmission carriers where compression coding schemes are used to provide extended bandwidth.
9. The use in completed mixes of various psychoacoustic enhancements that add depth, warmth, or three-dimensional imaging, and so on.

What must now be done by the audio industry is to develop a single high-quality compatible compression coding standard that will work for all equipment makers and all equipment users without any noticeable degradation of the audio signal or if not that, to at least fully identify the current problems and create a clear agenda for data compression usage.

It is equally important to recognize that the nonlinearity of small systems and speakers seems to accentuate the possible degradations of cascaded coding errors and shifts the perceived effect of the spectral masking and interactions with other equipment. The public listens to recorded audio via a range of small and inexpensive listening devices over 75 percent of the time. Even when a music consumer actually owns and uses a high-quality music system in the living room, most of his or her listening time will still be spent in a car or bedroom with small speakers and limited amplifiers.

That leaves us with the bottom line question for today: Do we compress? The answer is yes — if we really need the space saved by the compression process. Many of the compression schemes now in use are capable of very high-quality results if employed properly and if several different compression or coding algorithms are not superimposed on each other. But with potential users testing the impact on the entire audio chain rather than on just a single signal pathway, the byword has to be caution. With the same care taken that any facility should take in adding any new technology to its lineup, there should be no problems with compression hardware or software.

Martin Polon is the principal of Boston-based Polon Research International (PRI). PRI forecasts the electronic entertainment industry for the financial community.

Four (and More) on the Floor

The basics on Dolby Surround Sound, Pro Logic, and other systems that circle you with sonics

BY LEN FELDMAN

This time I'd like to talk about all (or at least some) of the stereo processing techniques used in master recording that result in extra channel outputs, or at least in stereo enhancements that trick our ears into believing that sounds are coming from beyond the pair of stereo speakers. If you can think back to when you took a high school algebra course (was it that long ago?), you may recall a teacher telling you that to solve for more than one unknown (such as x or y) you needed at least two different "simultaneous equations." If you wanted to solve for three unknowns you would need three equations; to solve for four you would need four equations; and so forth.

Yet here we are, when we want to solve for four channels of surround sound, faced with only two stereo-encoded signals. We pass those two signals through a magic "black box" and out come four different signals derived from the original two. Has the recording industry discovered a secret formula that contradicts our algebra teacher's hard and fast rule? Don't bet on it!

The truth is that the resulting four signals (derived from two stereo-encoded signals) never exactly duplicate the four original discrete signals that were encoded into two signals in the first place. This is true whether you're talking about quadraphonic sound reproduction (popular in the late 1960s and early 1970s) or its reincarnated modern versions such as Dolby Pro Logic (used for sound tracks intended for home theater playback) or the many "surround sound" soundstages often found in A/V receivers used by consumers. The essential difference between the latter two is that Dolby Pro Logic decoding is derived from specially encoded signals that must be created at the recording studio, whereas most of the soundstage surround sound effects can work with ordinary stereo signals.

PRO LOGIC PROCESS

Let's start by taking a look at the encode/decode process used for Dolby Pro Logic. For more than two decades, motion pictures shown in theaters around the country have included in their newspaper ads a trademark of two back-to-back "D's" that are by now instantly recognized as the Dolby Stereo logo. When you see that mark you know that the sounds for that motion picture were recorded with at least four separate sound tracks, and sometimes as many as six: a center front (for actors' dialog and other on-screen sounds), left-front, right-front, a pair of side channels and a pair of rear channels.

For the home environment, Dolby Labs figured out a way to mathematically combine the four original sound tracks used for the theater version of a film into two encoded tracks. Once in that form, the sound tracks can be recorded on a videotape or videodisc as if they were ordinary 2-channel stereo. In fact, when you play back a Dolby Surround-encoded videotape or videodisc using only a 2-channel stereo system, everything seems quite normal. The sound stage is still spread out in front of you from left to right. If consumers equip their playback systems with one of the many available Dolby surround decoders and one or more properly positioned extra speakers and amplifiers to drive them, however, they can recreate much the same sound environment that was present when the movie was shown in a theater equipped with Dolby stereo.

Don't confuse this type of home surround sound installation with so-called "4-channel" or "quadraphonic" stereo mentioned earlier. In those early quadraphonic installations, four loudspeakers were used in a "square" pattern — left and right front speakers and left and right rear speakers. In today's Dolby Surround sound installations, the basic configuration is "diamond" shaped. The system employs left and right front speakers, a center channel speaker, and a rear speaker.

While most people use a pair of speakers at the rear, rather than one, both rear speakers are fed the same program material. The Dolby decoding scheme delivers signals for front left and right speakers, a monophonic equivalent signal for the front center-channel speaker (vital for keeping actors' dialog "on-screen"), and a separate signal for the rear or surround speaker or speakers, often referred to as the "effects" speakers. If consumers want to economize, they can omit the center-channel speaker and use what is referred to as the "phantom channel" mode of the Dolby decoding circuitry. In doing so, they will find that their listening location becomes more critical. They will need to sit directly in front of the TV screen, which must then be positioned midway between the front two speakers. If they move off center, spoken dialog will tend to shift "offscreen."

Depending upon how much money you are prepared to spend, there are various levels of Dolby Surround decoding available. The simplest type involves the use of only three loudspeakers and amplifier channels. In such setups, you would flank your TV set with a pair of speakers, while a single speaker would be placed behind you. All the "surround" effects would be reproduced by that single rear speaker, while on-screen sounds would be reproduced by the two speakers positioned on either side of the TV screen.

Another three-speaker version of Dolby Surround uses all three speakers up front.

The preferred arrangement, however, involves the use of a fourth speaker as a center-channel sound source. Better decoders make a provision for such a center-channel speaker. Still other decoders even have an output intended for a low-frequency subwoofer. Adding a subwoofer enables the system to reproduce those thundering ultralow bass sounds that sometimes lend excitement to an action film and enhance the overall effect.

As I mentioned at the outset, it's impossible to solve for four unknowns with only two equations. In other words, when the two encoded Dolby Surround tracks on a tape or disc are decoded by the simplest decoders, there is no way to recover an exact replica of the original four discrete sound tracks. In fact, the separation between adjacent channels when using such simple decoders is only about 3 dB. Three decibels is a very minimal degree of audible channel separation. That's why it's so important to sit at the right spot (equidistant between and directly in front of the front pair of speakers) when using this type of simple Dolby Surround decoder.

There are ways to enhance the separation and spatial effects electronically, however. Decoders that can increase separation in this way are called Dolby Pro Logic Decoders and, as you might have guessed, they cost more than the simpler types. Simply stated, Dolby Pro Logic decoders sense which channel is currently producing the loudest sounds (albeit only 3 dB or less louder than the other adjacent channels) and electronically make that sound even louder while simultaneously attenuating the sound levels coming from the remaining speakers. The net effect is greater apparent channel separation.

While Dolby Surround decoding is what consumers will want when watching movies, there are times when they just want to listen to music. That's when other schemes for surround sound come into play. Several companies have come up with surround sound products that are able to create the approximate ambience of specific halls. A technique known as digital signal processing, or DSP, has enabled audio equipment designers to come up with products that alter audio signals in such a way that when they are reproduced from multiple speakers in a listening room, sounds can be tailored to simulate everything from an intimate jazz club to a cavernous cathedral or enclosed stadium.

Next time we'll examine some of these other approaches to surround sound and stereo enhancement. EQ

ACROSS THE BOARD

continued from page 122

Meanwhile, the 230 MB internal drive was chugging along just fine. I tried to install Studio Vision v1.51. Half way through the installation, an error message said, "Because of an ERROR this installation cannot be completed." No further information was available. Because the software is copy protected and compressed, it must be installed correctly before it will work. I cheated. I installed Studio Vision on the IIci and then, over the AppleTalk network, copied all of the files to the 840AV. I was then able to "authorize" the installation from the original master disk, and Studio Vision was up and running.

Next it was time to install Galaxy+ Editors v1.2.5, OMS+Patches v1.2.2, Sound Designer II v2.6, Sample Cell Editor v2.0, and Pro Tools v2.03. There were a few minor problems here, but I just count them as features that keep you from falling asleep during the installation process. I configured OMS to recognize all of Walter's synth modules and got it talking to the Studio 5.

All seemed well. I went back and started Studio Vision. The little window said, "The wrong version of DAE is present. You will not have access to audio services." (DAE stands for Digital Audio Engine, and is a software bridge that lets other software access the digital audio services of the Sound Tools DSP board.) What? It ran perfectly a few minutes (hours) ago. After some tail-chasing, it turns out that Pro Tools installed an older version of DAE over the top of the one that Studio Vision installed. I reinstalled the DAE that came with Studio Vision, and was back in business.

Well, there you have it. Tomorrow the correct formatting program should show up so I can get the 2 GB drives running. Check in next month to see if we ever got it all running. By then I should be in the middle of an upgrade to Vision 2.0.

And you thought you had upgrade headaches.

STRIKES AT THE BAND

I have a little company called Wild & Woolly Productions that is a signatory with the Musicians Union so that I can hire players and pay them through the union. All record companies and production companies just got a copy of the new union contract. It says that anyone who programs a sequencer or drum machine or computer or any device that produces music, or anyone who starts any device that causes any sequencer or drum machine to start playing music must belong to the Musicians Union. Does this mean that engineers now have to belong to the Musicians Union? If I start a tape machine that produces the SMPTE that makes the sequencer play the synthesizer, all of the musicians on the session could be fined because I'm not in the union.

This is the same Musicians Union that called me up in 1978 because of rumors that I was replacing a musician with a machine (a Wendel). Then they were going to fine the record company because there was no drummer listed on the tracking session. Now they will fine the musicians if I don't pay union dues to be able to use a drum machine. Why don't they require all the tape machine manufacturers and SMPTE generator makers and sequencer company employees to join the Musicians Union? It seems to me that making the machines is what really causes the music to play. Oops, maybe I should have kept my mouth shut. EQ

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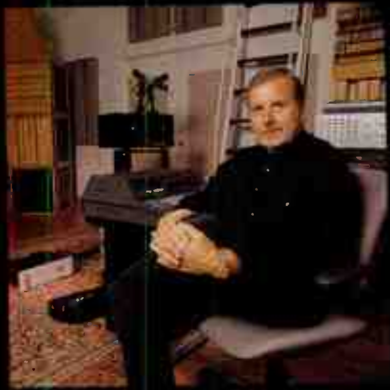

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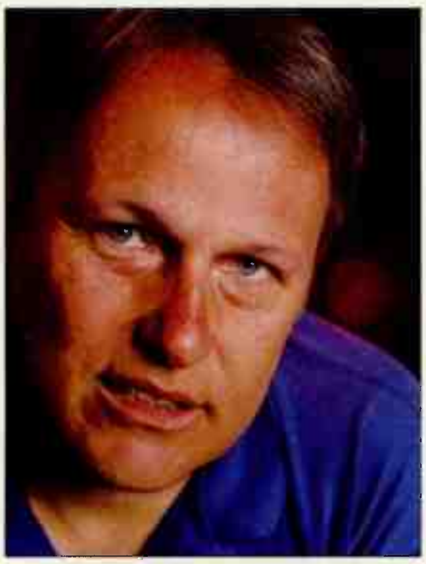

You think upgrading is easy? Guess again.

BY ROGER NICHOLS

I finally figured out why software and software-based equipment are so reasonably priced. They make up for it by charging for upgrades — a never-ending flow of upgrades.

In his Hawaii studio, Walter (Becker) has been using an old Macintosh IIx with 100 MB of hard-disk storage, 8 MB of memory, a Sample Cell board, and a Sound Accelerator system. The software he was using consisted of Opcode Studio Vision v1.32, Opcode Galaxy+ Editors, Digidesign Sample Cell editor v1.3, Digidesign Sound Tools v2.0, Opcode Studio 3 MIDI interface, and a KMX MIDI patchbay. A nice system that got the job done.

Down at Walter's house there is a little MIDI room where he can mess around with sequences without driving an hour to the studio. The computer there is a Macintosh IIci with a 200 MB hard disk, 16 MB of memory, a Sample Cell board, and an external 1.2 GB hard disk. There is no MIDI interface other than a Pocket MIDI to Mac box. The software complement is the same as at the studio, or at least it was to start with.

We tried to upgrade the old IIx to allow 32-bit applications. No such luck. For some unknown reason, whenever this computer would boot up in 32-bit mode it would crash, taking no prisoners in the process. It must have had something to do with the fact that it had originally been a Mac II, upgraded to a IIx. We gave up.

The external 1.2 GB drive was used to store digital audio samples and items that would be carried back and forth between the house and the studio. The software to run the sequences could not be run from the external drive because some of the software pieces must reside on the computer's internal drive to operate properly. Now we had a situation where the two different computer configurations required different software versions and the sequences saved in one version would not run in the other version. This situation had to change.

Walter decided to go "all the way" — he bought two new identical computers with the latest version of all of the software and the newest revisions of the hardware. The choice was a matched pair of Macintosh Quadra 840AV computers, each with an internal 230 MB drive and an external 2 GB drive, 24 MB of memory, Sample Cell II boards with 32 MB of memory, new Pro Tools 4-channel DSP boards with Pro Tools Audio Interface, and an Opcode Studio 5XL for outside world MIDI and SMPTE communication.

Here is where the fun started. The Micropolis 2217 2 GB drives are only 1.6 GB after you format them. We decided that in the Gear Slut tradition, we would continue to call them the "2 GB drives." We started loading samples and sequences and more samples onto one of the 2 GB drives. The samples would be much easier to sort out if they were all in the same place at the same time.

Just as we got to around 1 GB of information stored on the drive, *crash*. The disk icon disappeared from the desktop and no amount of SCSI software could find it. The only thing we could do was to reformat the entire drive. Apple has upgraded its format program to support drives of up to 2 GB, so we decided to try it. No such luck. The Apple formatter will not even recognize add-on drives. We called Apple, and they said we would have to use a third-party formatting program. They made it work that way on purpose. They said that the drive shouldn't have crashed, but drives over 1 GB should be formatted with a driver that includes the "new" SCSI 4.3 driver.

I got on the phone and tried to locate a formatting program that would help us get out of this new mess. In the past I have used FWB Hard Disk Toolkit to format all of my drives, including Sony optical drives. FWB was the only formatter that would let me use 1024 byte/sector disks with the Mac (I wanted to use the same disks for the Mac and the Akai DD-1000 optical disk recorder). It turns out that the versions I have do not support the SCSI 4.3 driver. I would have to upgrade. They said that the new version would be out some time in December. Three weeks after I needed it (which was immediately).

Micronet has a formatter that supports SCSI 4.3, but it will only work for drives that you buy from them. That lets us out.

I then called CharisMac, which makes the Anubis Utilities. The company has a new version that should be available soon, but everybody who could help me was at Comdex, a big computer show in Las Vegas. This is where all of the hardware and software companies go to spend all the money that Gear Slut members have paid them for upgrades since this time last year. A nice gentleman in the CharisMac office contacted the people who were at Comdex and got permission to send me a prerelease version of the new formatter. I think we will again barely escape the "bleeding edge of technology."

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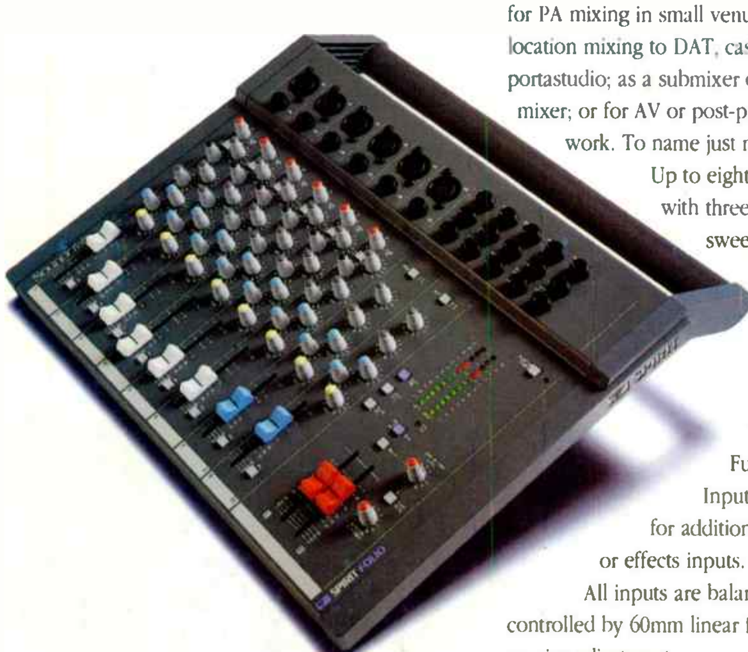

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