PROJECT RECORDING & SOUND TECHNIQUES

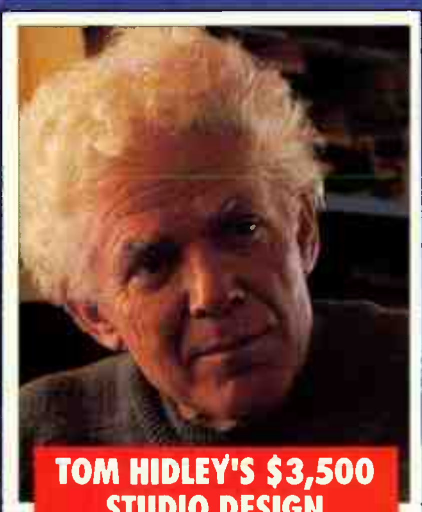
EQ

FEBRUARY 1991 · A PSN PUBLICATION

GNOME AWAY FROM HOME

DAT GETS IN TIME
By Roger Nichols

- BAND IN A VAN
- PRODUCT REVIEWS:
  A.R.T. POWER PLANT
  FOSTEX G-16
- BUILD YOUR OWN MIDI SWITCHER



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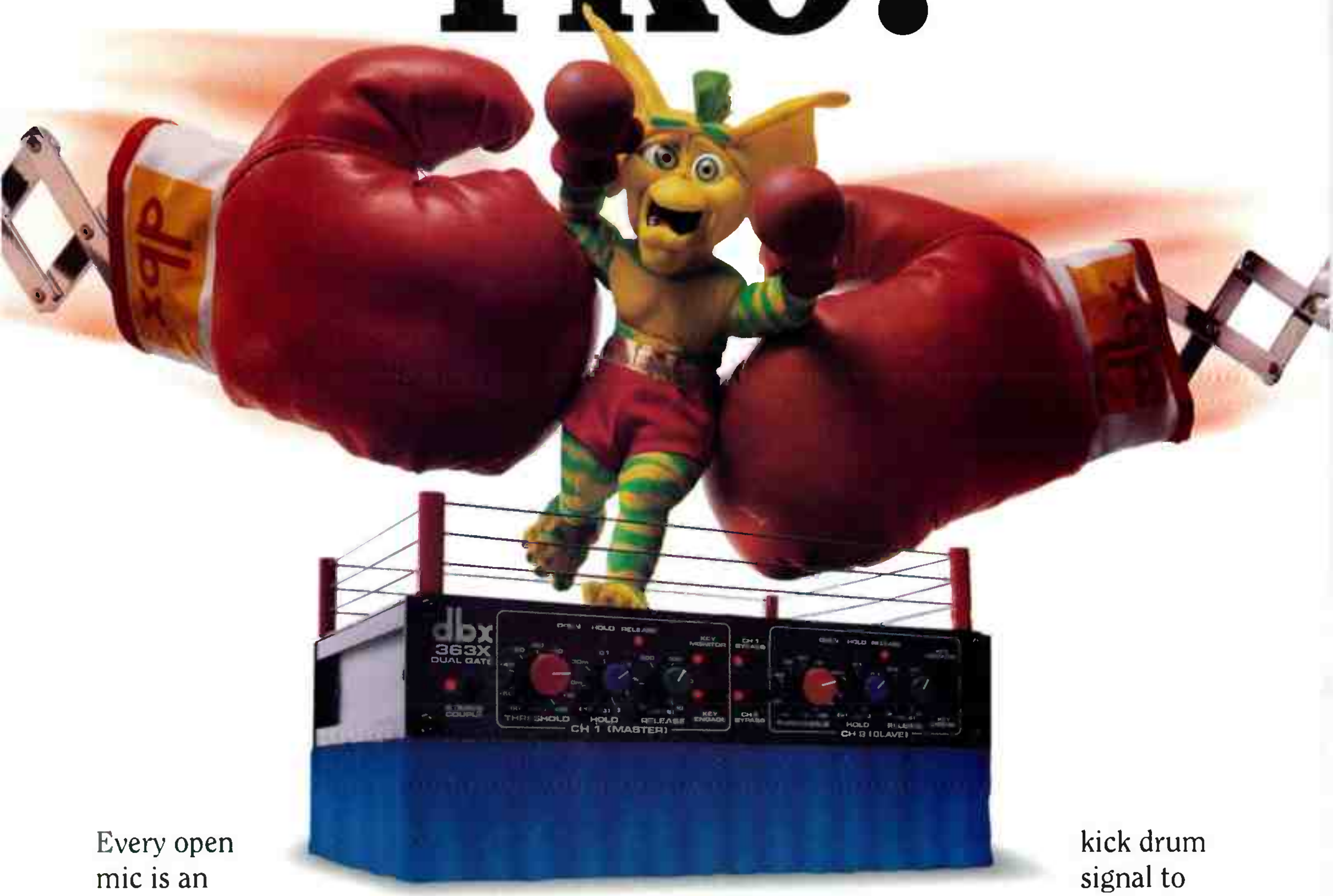

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PROJECT RECORDING
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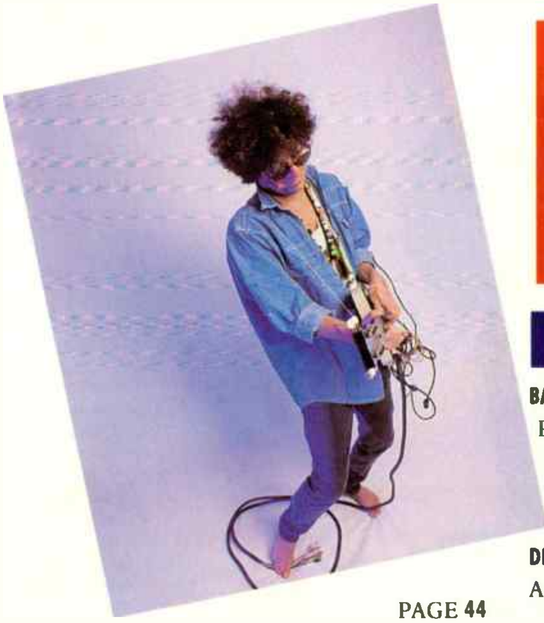
JANUARY/FEBRUARY 1991



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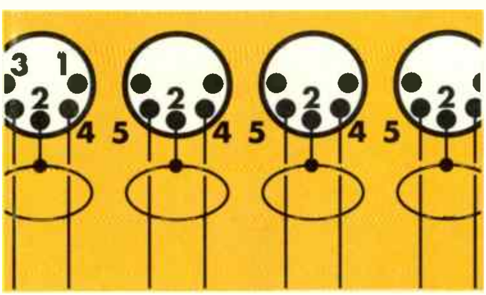

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DEPARTMENTS

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Our new professional DAT re
you'll have to look at DAT in a

Time Code. It's about time someone perfected time code for DAT. And Sony's new PCM-7000 DAT recorders have done just that. They make time code based editing easier than you ever thought possible. They allow you to read and generate SMPTE, EBU or film rates which can be prestriped, poststriped or recorded simultaneously with your audio. They even let you translate from one time code rate to another during playback. All of which means if you're not using a Sony professional time code DAT recorder, you may want to take a closer look at what you're missing.

Speed and Size. When speed counts the PCM-7000 recorders are the tools you want. They have a shuttle speed 175 times play speed, which lets you locate cues or lock to other equipment faster than with open-reel systems. They also come with helpful menus and self-diagnostics for fast set-up and easier maintenance. But speed isn't the only issue. Size is also important. Unlike reel-to-reel recorders, you can fit any of our new DAT recorders into just 5¼" of rack space. In addition, each DAT tape can fit two hours of stereo digital audio into a package smaller than a standard audio cassette, saving you plenty of storage space. And since DAT tape costs about one-third of analog open-reel tape, most facilities could save enough in the first year to pay for the recorder.

Instant Start. If you need "On-the-Air" or "On-the-Fly" cueing, you can equip our new DAT recorders with Instant Start. It's simple to use. Just mark and trim the starting point, then press PLAY. You'll get instantaneous audio output with absolutely no start-up "wow." To make it even more convenient, you can initiate Instant Start with a fader-start or GPI.

Chase Synchronization. With the internal Chase Synchronization option of the PCM-7050 and 7030, you can press a single button to lock to any time code based equipment—whether it's a VTR, ATR or a second Sony PCM-7000. You can also enter or capture an offset instantly and maintain synchronization with the time code data or from an external reference.

The System. The PCM-7050, 7030 and 7010 time code DAT recorders are all designed and constructed for professional use. They each offer clear advantages in sound quality and operating economy over conventional analog recorders. They're smaller and faster than analog recorders. And when you add the RM-D7300 edit controller to the PCM-7050 and 7030, they create a powerful electronic editing system. No other DAT manufacturer offers you this kind of system approach because no other manufacturer looks at DAT quite the same way as Sony. For more information, call the Sony Professional Audio Group at 1-800-635-SONY.

Professional Inputs and Outputs. It goes without saying that you can connect these DAT recorders with a variety of I/O's—like balanced XLR analog I/O's or optional digital I/O's, including AES/EBU. This enables you to transfer audio to digital VTR's and just about any other professional audio equipment you desire.

corders have so many features, whole new way.

External Control. Our new DAT recorders offer you several external control options—an RS-232C port for computers, parallel connectors for external synchronizers and a 9-pin serial port on the PCM-7050 and 7030 for compatible video editors. So you can control our DAT recorders from just about any source you choose.

Off-Tape Monitoring. A sophisticated four head design lets you monitor off-tape as you record. So you'll always be confident of the quality of the recorded signal.

Electronic Editing. Plug the PCM-7050 and 7030 into the RM-D7300 edit controller and you'll be able to capture edit points in RAM memory, trim with the jog wheel and preview edit points before committing them to tape. You'll also be able to set the crossfade time from 0 to 999 ms and digitally adjust the audio level for the smoothest transition possible. It's definitely the best way to edit DAT.

Audio Quality. We've minimized phase distortion in our new DAT recorders by giving them 18-bit D-to-A converters with 8 times oversampling—and A-to-D converters with 64 times oversampling. And because they're digital, frequency response is extremely flat from 20Hz-20kHz, dynamic range exceeds 90dB and "wow and flutter" levels are so low they can't be measured. Which sounds pretty good here, but sounds even better on DAT.

SONY.

BUSINESS AND PROFESSIONAL GROUP

EQ

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LETTERS TO EQ

DEJA VU?

I'd like to compliment you for putting together a book that, in my opinion, is one of the better looks at our industry. I hope you will be able to continue the quality of fresh *real* material that I have seen so far.

The December issue included an article I found particularly entertaining and practical. The practical part is the live, hands-on, "how I do it" approach of the author. The entertaining part is a little more obtuse.

It never ceases to amaze me how we in this communication-based industry communicate so little. Having been in the field for over 30 years I may have lost sight of how they do it in other businesses but this one seems unique. Each succeeding generation seems obligated to reinvent the wheel. From modern fiberglass horns that look suspiciously like what Paul Klipsch whittled out of wood in 1932 to today's focus of my attention, Wade McGregor's fine and practical piece on ringing out a room using a metronome.

He's right, it works. We know it works because Jurgen Wahl and the original United Recording Electronics Industries gave us the Sonipulse in the mid '70s. At this point I cringe because probably Mr. McGregor and two-thirds of your readership are now trying to remember what elementary school grade they were in then. Anyway, the Sonipulse was a "high tech" version of just what Wade McGregor described and a whole lot more. The product had a short life because the LED-based real time analyzers became affordable and the Sonipulse was time consuming and less flashy to the end user. But it did the special tests Mr. McGregor described, something the real time analyzer could never accomplish and the TEF costs thousands to perform.

We old timers can't let go.

I still use my Sonipulse when no one is looking.

Keep up the good publication.

Peter Horsman, Product Training Engineer, Yamaha Professional Audio Division

JAVA READY

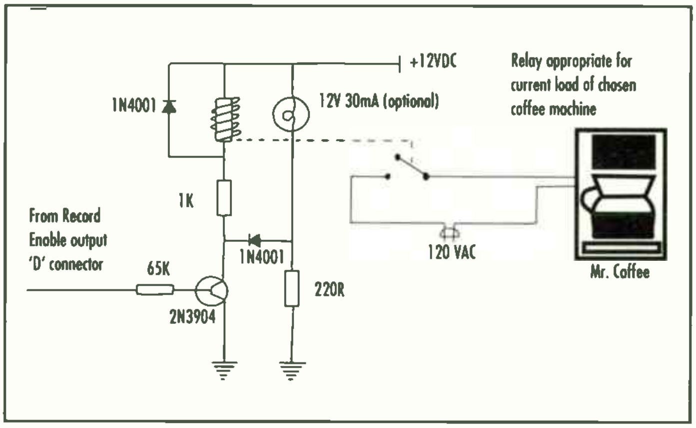
Regarding Ira Rubnitz's "Mostly Mozart," (December 1990) we'd like to point out two things:

1) The size and positioning of the buss switches [on Amek's Mozart] are dictated by the module size and console ergonomics. The aim of the designers was to keep the length of the module to a minimum, reducing any risk of strain on the engineer,

2) The coffee maker can easily be controlled from the console.

Steve Harvey, National Sales Manager, Amek/TAC

Diagram follows:



CORRECTIONS

There were two editing errors in Craig Anderton's sidebar to December's "Going Direct" feature. We said that "some (DI signal processors) are actually miniature guitar amps *without* output transformers and speaker box emulators..." We really meant *with*. Also, there is no such thing as a megaohm. The correct term is megohm.

WRITE TO US

Write to: Letters to the Editor, EQ, 939 Port Washington Blvd., Port Washington, NY 11050. EQ welcomes letters but reserves the right to edit them for clarity and space.

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

There may be digital effects processors that rival some of the 300's features, but you'll never find one with them all. The 300 delivers precise delay and stereo pitch shifting, as well as stunning reverb and ambience. It automates sound changes with SMPTE/EBU time code. And it inputs and outputs analog, as well as consumer and professional digital formats — in any combination. All with magnificent Lexicon Sound.

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The Art and Science of Sound

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EQ&A

Q I've always thought SMPTE sync was the best way to handle synchronization, but recently some friends told me that sync systems that record MIDI Song Position Pointer (SPP) onto tape — because they don't have to do so much computation — will actually provide more accurate results during playback. What's the deal?

Albert Pisano, Baltimore, MD

A This is really a three-part question: 1) Which sync system is more accurate: SMPTE sync or MIDI Song Position Pointer? 2) Which uses more computation time? 3) How much bandwidth does your system need?

MIDI Time Code (MTC) translates SMPTE to MIDI. It transmits its updated time to the sequencer more frequently than SPP, and so is more accurate per beat. However, MTC uses more computation time than SPP, because it calculates tempo and adjusts itself in relation to the overall tempo.

SPP is less accurate than MTC because it runs off the clock of the tape (which warbles), and not off the internal clock of the computer, and because it adjusts itself to the incoming sync signal less frequently. It uses less computation time than MTC because it does not calculate a relative, overall tempo. Computation time does, however, increase if the tempo increases.

For example, at 120 BPM one beat in MIDI timecode has an accuracy of 60 quarter frames (for 30 frame SMPTE which has 120 quarter frames per second), while one beat in SPP at 120 BPM has an accuracy of 24 clocks, which is 48 clocks per second. However, if the tempo is increased to 300 BPM, SPP will have about the same number of parts (clocks compared to quarter frames) as MIDI timecode. But who records at such brisk tempos?

Another point to keep in mind is how much bandwidth your system needs. SPP takes up less bandwidth in the MIDI data stream, so if your system uses a merged MIDI input (a single input carrying MIDI performance data synchronization) it would be better to use SPP, because a large multinote chord or pitch bend or modulation could cause the MIDI data stream to overload and so upset timing. Most systems today, however, don't require a merged input, and it is not recommended. It is much more common to use separate inputs, one for MIDI data and the other for synchronization. So, say you're using Opcode's Vision sequencer and Studio MIDI/SMPTE interface, and you have a MIDI controller keyboard on one port and leave the other port for synchronization — MTC would be the best choice because the MIDI data streams are separate. In addition to being more accurate and the best suited for the most systems, SMPTE (MTC) is also the only way to really sync to video.

If the timing of your music is important to you, think ahead so you can use a method most appropriate to your system.

Paul de Benedictis,
Opcode Systems, Inc.

Q I've read about the Meyer SIM-CAD, and I'm wondering what the effect of using a delay line on my RTA (Real-Time Analyzer) would be. Can I measure the direct sound from my speaker stack?

Angela Seymour,
Shreveport, LA

A SIM is an extension of two-channel FFT (Fast Fourier Transform) analysis. In SIM, the two analyzer inputs are: (1) the house console's electrical output signal; and (2) the acoustical output of the loudspeaker system in the hall, as sampled by a switched array of measurement microphones.

Using a function known as "cross spectrum computation," the FFT analyzer computes the difference between these two signals. The result reveals frequency and phase response variations introduced by the hall acoustics in interaction with the loudspeakers. By equalizing out these variations, we can closely match the actual sound in the hall to the console's electrical output.

The delay line to which you refer is inserted between the console output and the analyzer input. Its function is to compensate for the "propagation delay" caused by the time that it takes for sound to get from the speakers to the measurement microphone. This synchronizes the two analyzer input signals, assuring accurate frequency and phase measurements.

The greatest advantage of SIM is that, by taking the system input signal as the measurement reference, it allows us to measure a loudspeaker system using music as the test signal. This means that the system can be equalized unobtrusively during a concert, in the presence of an audience.

Real-Time Analyzers do not require any sort of time synchronization, and so would not benefit from a delay line. RTA's gather what is called the "steady state" frequency response of the loudspeaker/room system. This includes reverberation components as well as the direct sound from the speaker, and the RTA has no means of separating the two.

To get the best results with your RTA, feed the speaker system from a pink noise source and use averaging to obtain a stable, consistent display. Take measurements at several places in the hall, and equalize only those peaks or dips that appear consistently in all locations.

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

When adjusting your equalizer, don't go for a flat display on the analyzer: it makes the sound too bright.

Because the response gathered by the RTA includes reverberant energy, which is composed of mostly low frequencies, you should equalize to a "house curve" that tilts downward with increasing frequency. The degree of tilt will depend on how reverberant the hall is.

Ralph Jones, Marketing Manager, Meyer Sound Labs

Q I am in the process of buying a new console for my project studio. What is the difference between a "split" type and an "inline" design?

Paul Accardi, Clearwater, FL

A The recording mixer can actually be thought of as two consoles. One console (let's call it the input console) takes the input from microphones and other sources and routes them (via the busses) to tracks of the multitrack recorder. This is where the engineer exercises control over what goes to tape during tracking and over-dubbing operations.

Of course, the performers need to hear what's going on too. But what they need to hear may differ from what the engineer listens to in the control room. A big part of the engineer's job is giving the musician a headphone mix that lets them feel comfortable (and hopefully, inspired) during tracking and over-dubs. This cue mix (sometimes called "headphone mix" or "studio mix") comes from the monitor section of the recording console. In addition, the engineer (who also wants to "listen off tape") will set up a control room mix using the monitor section.

The multitrack recorder (MTR) is usually the source for the inputs to the monitor mixer section. During a recording session, the output of the MTR is routed to the monitor section of the console. When the multitrack is in RECORD, input sources can be heard through the multitrack at the monitor section of the console. In mixdown the MTR output will be re-routed to the "input console" with its more capable EQ and auxiliary bussing.

So what are the relative merits of inline and split console configurations? During tracking and overdubs, auxiliary busses in the monitor section are used to send headphone mixes. In the inline console, the monitor section and the input section are physically in the same module, so in mixdown, these same auxiliary busses can be switched to send from input

EQ&A

channels to effects. Since additional busses are a major component in console cost, some savings are realized by this dual-use arrangement. On the minus side, there may be less options for custom channel configuration. Since every input channel also carries with it a monitor channel, a 32-input console will also have 32 monitor channels — even if you only have a 24-track recorder.

On the other hand, if you want a 56-input console with 24 tape returns, a split console is probably the only way to get it. Some engineers also feel that physically separating the input section from the cue mix section results in clearer, more intuitive operations. The separation also makes it possible to assign a second engineer the task of running the cue mix while the first engineer devotes full attention to getting the music onto tape. At the same time, a split design will usually be considerably wider than an inline console, taking up more control-room real-estate.

All else being equal, neither design offers any sonic advantages over the other. These days, in both inline and split console configurations, the monitor section is becoming more fully-featured, so it can be used to provide additional inputs in mixdown.

The essential difference between split and inline recording consoles is the location of the tape monitor section. Inline console designs integrate input channel and tape monitoring functions into a single module. A split console will have a dedicated input section (usually on the left side of the master section) and a separate cue mix section (typically on the right side of the master section).

Bob Davis, Peter Chaikin, Yamaha

Q What is the purpose of the "keying," or "trigger" input on a noise gate? How can I use this input in my recording or live sound work?

Timothy Asher, Springfield, MA

A A noise gate is basically an automatic switch that "turns on" when the noise signal rises above some predetermined level. Many noise gates have several parameters that can be adjusted to control the behavior of the "switch." Some of these gate behavior variables are: what minimum level is needed to turn the gate on, how fast the gate turns on, how long it remains on, how long it takes to gradually turn off, etc. These adjustable parameters give the user great flexibility on just how to turn the signal on and off. A fundamental limitation, however, is that the signal being gated can only be turned on by itself.

Many professional gates have an external KEY or TRIGGER input and a switch to select either internal or external triggering, much like an oscilloscope. Keying or triggering a gate is the process of activating the gate ON with some signal other than the signal which is being gated. The key input signal is an analog audio signal as opposed to a digital or MIDI signal. The ability to key a noise gate allows greater flexibility because the signal can be gated ON by a processed version of itself or even by an unrelated signal.

One problem which may be encountered when using noise gates with several microphones in sound reinforcement or recording is unintentional triggering of a gate from another nearby source. The KEY input can be used to solve this problem by triggering the gate with a band-limited version of the signal, thus making the gate insensitive to all other sound sources which have little energy in the bandpass region.

This works well when miking tomtoms, for example. Simply split the signal before it is fed into the gate, and feed the tapped signal into a bandpass filter or equalizer, then return the bandpassed signal into the KEY input. Some noise gates have this frequency-sensitive keying feature built-in.

The key input is also useful when gating effects. You can get more accuracy by keying the gate with the unprocessed or dry signal. This helps control the ON time of a gated reverb from a snare or limits the number of echo regenerations from a vocal channel.

Finally, the key input has applications in controlling the dynamics of one instrument with another. An example would be one keyboard channel triggering one or more other keyboard channels to achieve a unique voicing or perhaps a thicker sound. Also the attack of one instrument such as a kick drum could key the gating of another instrument such as a bass guitar to tighten a mixdown.

Gene Goff, Chief Engineer, Ashly Audio, Inc.

EQ PRODUCT VIEWS

POWER HOUSE

Crown's Macro Reference amplifier sets new standards for studio amplification. Offering better than 20 bit dynamic range, it includes a multiple plug-in input configuration, Output Device Emulation Protection circuitry, LED displays for Crown's Input/Output Comparitor, and Signal Process Indicator Circuits, as well as front panel level controls and two input sensitivity settings. Operable in bridged/mono or parallel/mono modes, its bandwidth lies between 3 Hz and 100,000 Hz, and its ultimate damping capabilities are rated at more than 20,000 at 8 ohms. It's priced at $3,500. Contact **Crown International**, P.O. Box 1000, Elkhart, IN 45617; (219) 294-8000. Circle EQ Free Literature Card number 100.

FIGURE EIGHTS

Soundcraft has added the **Delta 8** — a compact recording console designed with the project studio operator in mind — to its line of mixing consoles. Its all-modular split format offers eight groups with a choice of 20, 28, or 36 input channels. Mono input facilities include six independent auxiliary sends, a four-band EQ section with two swept mid-bands as well as High Pass Filter and Phase Reverse switches. Its stereo input module features a three-band sweep EQ section and six auxiliary sends. Solo trim control and extra control room monitor speaker outputs are included for switching between main and nearfield monitors. Prices range from $9,900 to $15,900. For more information contact **Soundcraft,** 8500 Balboa Blvd., Northridge, CA 91329, (818) 893-3639. Circle EQ Free Literature Card number 101.

THE WIRELESS SIDE

Sporting a dynamic range approaching 120 dB, the **Nady 1200** VHF wireless microphone system claims that hardwire has nothing over its wireless technology. The receiver features an unbalanced line level output, a balanced output switchable between line and mic levels, a headphone monitoring output, three LED bar graphs that display audio and signal strength on both front ends and a front panel switch to select channel A, B or True Diversity mode. The 1200 HT handheld mic/transmitter is a metal, wireless unit with no protruding antenna, offering user-switchable mic elements, input level controls, and audio on/off for silent muting. The system also features the 1200 LT transmitter for wireless lavalier operation, and the 1200 GT for wireless instrument use. The HT system runs $1,699 and the LT and GT systems are $1,599. Contact **Nady Systems,** 6701 Bay Street, Emeryville, CA 94608; (415) 652-2411. Circle EQ Free Literature Card number 102.

FLOORED BY SOUND

The **Audix SCX-1** studio condenser microphone is suitable for numerous applications including studio, sampling, and live sound. It achieves a noise floor measurement of 13 dB and a signal-to-noise ratio of 129 dB by incorporating surface mount technology along with close-tolerance matched components. This transformerless, high output/low noise unit features a gold-plated APC board and interchangeable capsules with selectable polar patterns. Optional accessories include a 10 dB attenuator, windscreens, a pop filter screen, and a shock mount. It retails for $699. Contact **Audix**, 5653 Stoneridge Drive, Pleasanton, CA 94588, (415) 463-1112. Circle EQ Free Literature Card number 103.

DAT'S IN TIME

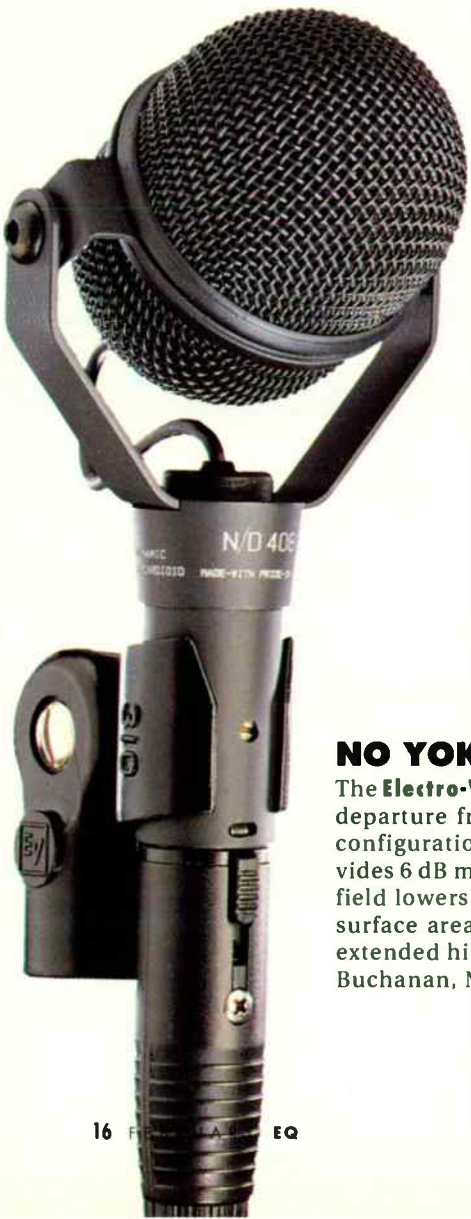
Sony Professional Audio has introduced a range of fully professional DAT recorders and an edit controller that are timecode-capable, and permit flexible system configurations to meet the requirements for a range of production applications: the PCM-7050, PCM-7030, and PCM-7010 recorders and RM-D7300 professional DAT edit controller. Up to two hours of 16-bit stereo digital audio can be recorded on each cassette. Designed with SMPTE, EBU or film timecode, each of the Professional DAT series sports specs that include 90 dB dynamic range, frequency response from 20 Hz to 20 kHz, and immeasurable wow and flutter. For prices and more information, contact **Sony Pro Audio,** 3 Paragon Drive, Montvale, NJ 07645, (201) 358-4197.

MAGNETIC MAESTRO

Superior magnetic performance, ultra-low noise and a dual-layer anti-resonance mechanism are the highlights of **TDK's Sound Master** audio cassette intended for studio and demo applications. The high bias Type II cassette features ultra-fine, uniform Super Avilyn particles, resulting in a coercivity that's a hefty 600 Oersteds and an impressive 1700 Gauss remanence. SM comes in demo-convenient 10-, 20-, 30-, and 60-minute tape lengths. For more information, contact **TDK Electronics,** 1411 W. 190th St., Gardena, CA 90248, (800) 752-9835. Circle EQ Free Literature Card number 104.

ROAD MUSIC

Ready for some road work? **ENSONIQ's SQ-1** Personal Music Studio is a compact keyboard with 24-bit effects, a 16-track sequencer, and mixdown capabilities. It gives mobile maestros up to 180 internal sounds and adds digital effects processing for a full range of effects including various reverb, chorusing, flanging, delay, distortion, and roto-speaker programs. Recording can be handled in real time or step-entry, and autolocate and range editing options center in on a specific bar, beat, or individual note. Sound and sequencer data for the SQ-1 can be stored on credit card-style memory cards, and the sequencer memory can be expanded from 9,000 to 58,000 notes with an optional SQX-70 kit. The SQ-1 is priced at $1,595, and the SQX-70 memory expander is $275, including installation. For more information, contact **Ensoniq,** 155 Great Valley Parkway, Malvern, PA 19355, 1-800-553-5151. Circle EQ Free Literature Card number 106.



OVER EASY?

Cook your sounds the way you like 'em with the **dbx 160XT** compressor/limiter. It's the only box of its breed that lets you pick between classic hard-knee compression and dbx's trademarked OverEasy sound — no matter what compression ratio you're working at. The ratio, by the way, is continuously variable from 1 to 1 through infinity to 1 to -1 to 1. The result is an incredibly wide palette of settings. The 160XT also features a dual RMS display system that monitors input or output with a 19 LED display while monitoring gain reduction over a 40 dB range at the same time. Access to the RMS detector input offers both frequency and time-dependent compression, and when stereo-coupled, the 160XT provides true RMS power summing for precise tracking. Price: $459. Call **dbx,** 1525 Alvarado Street, San Leandro, CA 94577, (415) 351-3500. Circle EQ Free Literature Card number 107.

NO YOKES

The **Electro-Voice N/D408A** supercardioid-dynamic instrument microphone represents a radical departure from conventional instrument microphone designs. It has a unique pivoting yoke configuration that allows maximum sound source positioning flexibility. The N/D408A provides 6 dB more output sensitivity over conventional designs, while the more uniform magnetic field lowers distortion during peak sound pressure levels. The large diaphragm has twice the surface area of conventional designs and is reinforced to prevent breakup — the result is an extended high-frequency response. It's priced at $258. Contact **Electro-Voice,** 600 Cecil Street, Buchanan, MI 49107; (616) 695-6831. Circle EQ Free Literature Card number 108.

TALE OF THE TAPE

3M boldly claims that their new **996 formula** is the first analog audio mastering tape to come close to digital sound, capable of recording at an operating level of +9 with virtually no distortion, and with a signal-to-noise ratio of 79.5 dB, reportedly the highest on the market. It is packaged in a library box that seals out dust and humidity, which is particularly useful for archiving. For more information, contact **3M Professional A/V,** 223-5N-01, 3M Center, St. Paul, MN 55144, (612) 733-3888. Circle EQ Free Literature Card number 109.

MUSCLE MUSIC

Carver's PM-600 magnetic field power amplifier kicks out 300+300 watts into four ohms, incorporating a highly-regulated magnetic field power supply with remote/manual sequential power on/off and a linear tracking output design. Other features include Clipping Eliminator circuitry, 7-segment LED power displays, heavy-gauge chassis and detented input gain controls. The unit is constructed with barrier strips as well as TRS and XLR inputs, and may be used for 70V direct operation or with 70-volt transformers, each with its own rear "card slot" for use with Carver plug-in modules. The PM-600 is priced at $820. Contact **Carver Corporation,** P.O. Box 1237, Lynwood, WA 98046, (206) 775-1202. Circle EQ Free Literature Card number 110.

MAY THE QUARTZ BE WITH YOU

Soundtracs' QUARTZ is a high-quality, 24 bus inline console with computer mute automation of all channels and auxiliary masters. It stores up to 100 mute patches in memory and can recall them from a MIDI sequencer, letting you lock mute playbacks to timecode. Frame sizes accommodate 32 and 48 I/O modules both standard with TT patchbay. Its features include 6 auxiliaries and a four band all sweepable equalizer. Options include stereo input modules, effects return modules and P&G faders. Prices range from $43,000 to $55,000. Contact **Samson Technologies,** P.O. Box 9068, Hicksville, NY 11802, (516) 932-3810. Circle EQ Free Literature Card number 111.

UP DATE

News & Notes From The World of Modern Recording & Sound

Torn, Tech & Texture

David Torn's recently released "Door X", **Windham Hill's** first-ever rock offering, was built with a unique recipe of signal processing, innovative looping techniques and an individual approach to studio recording philosophy.

Torn, well-known for years on the jazz, fusion and new music scene, has built a reputation for his unique electric guitar style and innovative signal processing, and has worked with the likes of trumpeter Don Cherry, saxophonist Jan Garbarek, David Sylvian of Japan, trumpeter/synthesist Mark Isham, The Everyman Band and others. Joining him on this latest release are long-time collaborators Bill Bruford (Yes, King Crimson), bassist Mick Karn (Japan, Dali's Car), Antony Widoff, and engineer/co-producer Stephen Krause.

The texture of Torn's music comes from a wide variety of studio techniques. He generates long, multiple delay loops by "improvising into my modified (20 second) Lexicon PCM 42, processing it and sometimes playing more guitar into it," he says. Through this technique, Torn creates an extremely rich and singular texture.

Torn's rack also boasts a **Lexicon PCM 70**, with which he usually applies the 'space' to his loops. Combined with a masterful use of the volume pedal, Torn creates rich musical backdrops (or soundscapes) for his compositions — sounds he calls 'hypnodrones.' "I decided I had to give it some kind of stupid name," he says.

Torn uses **Klein** electric guitars, built by Steve Klein, a Sonoma, California luthier who combines Steinberger-patented **Trans-Trem** bridges and **EMG** pickups, mounted in his ergonomically designed neck-body configuration. He sticks exclusively to **Pearce** all solid-state amplifiers, and despite an anti-tube stance, he develops both warm, singing and also crisp, biting tones that are unmistakably his own.

"I'm trying to get it to be a unified, single-instrument thing," says Torn of his guitar effects processing technique, which stresses the integration of

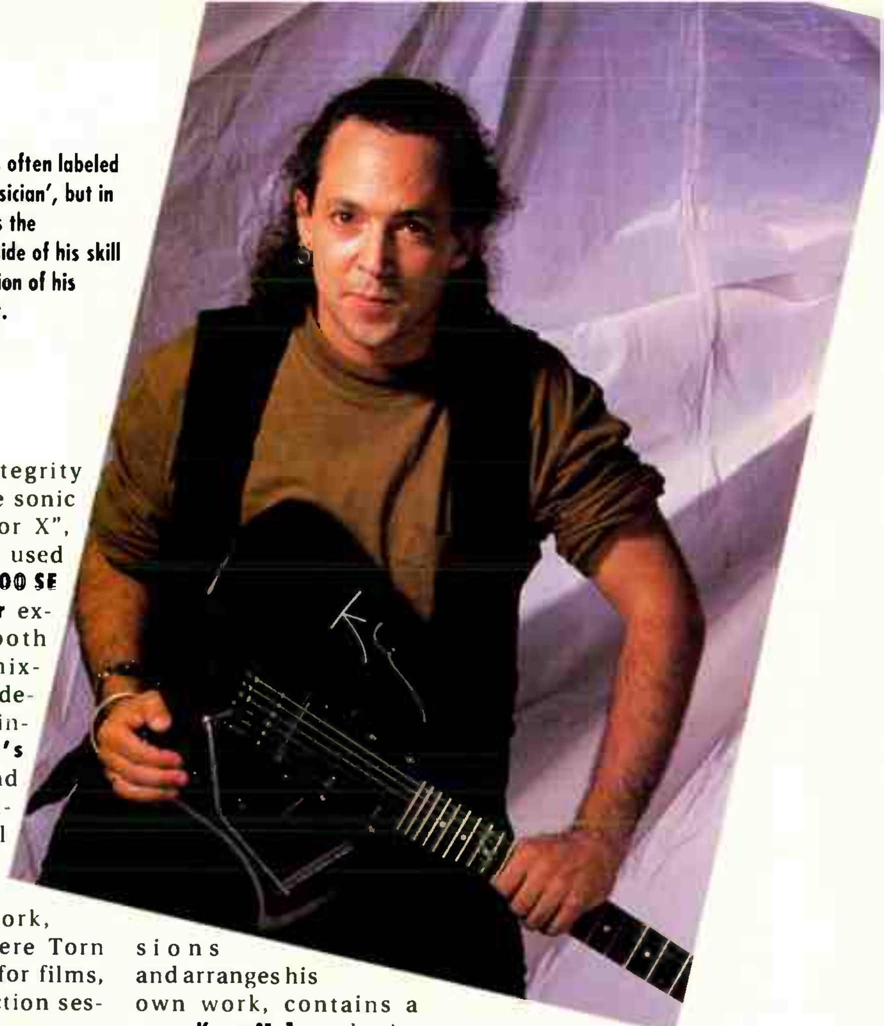
Gloria Estefan and her Audio-Technica ATW-1032

LIMELIGHT MICS

A look at the first links in the sound chain — a stage and studio sampling.

LIT #	BRAND	MODEL	ARTIST	USE
112	Audio-Technica	ATW-1032	Gloria Estefan	vocals
113	Audix	OM-1	Beach Boys	vocals
		OM-7	Toto	vocals
114	Beyer	M-88	Phil Collins	drums
		TGX-580	Eric Clapton	vocals
115	Bruel & Kjaer	4006,4007 4011	Pretenders	n/a
			Rolling Stones	n/a
116	C-T Audio	CX Series	Duran Duran	drums
		SXI Series	Pink Floyd	saxophone
117	Countryman	Isomax Headset	Def Leppard	vocals
			Peter Gabriel	vocals
118	Electro-Voice	DS35	Smithereens	vocals
		N/D408A	Aerosmith	drums
			Santana	drums
		N/D757A	Smithereens	drums
119	Fostex	M-51	Rush	vocals
		M-88	Julian Lennon	vocals
102	Nady	1200 VHF Wireless	Billy Idol	bass, vocals

David Torn is often labeled a 'techno musician', but in fact he views the engineering side of his skill as an extension of his overall talent.



player, instrument and outboard device.

Torn's uncanny ability to get a great sound right out of the rack makes long-time engineer/technician Krause's job a little easier. For melodies and solos, Krause generally mics both of Torn's Pearce 1 x 12" speaker enclosures with the dynamic **Shure SM57** and/or a **Sennheiser 421**. For ambience, loops and textural sounds that are looser and require less attack and more 'air', **AKG 414's** and **Milab VIP-50** condenser mics are used at distances of between six and 12 feet. In virtually all cases, the signal is sent directly to tape via his **Focusrite** microphone preamp/equalizer modules, maintaining as well as possible the integrity and purity of the sonic source. On "Door X", Torn and Krause used an **Eventide H-3000 SE Ultra-Harmonizer** extensively, for both tracking and mixdown. Torn has developed a real affinity for **Eventide's** 'Swept Reverb' and 'Comb Filter' algorithms, as well as others.

The Woodstock, New York, home studio where Torn generates music for films, holds pre-production sessions and arranges his own work, contains a **Korg M-1** synthesizer, an **E-mu Systems EMAX** sampler, two **Casio CZ-101** synths, **Alesis HR-16** and **Roland R-8** drum machines, two **Yamaha TX81Z** tone generators and **Lexicon MRC** and **Yamaha MCS2** MIDI controllers all driven by his Macintosh-based **Performer, Upbeat** and **M** software. But in contrast to the perfection commonly associated with sequencers and MIDI production, Torn prefers to let people know that his music is conjured up fairly improvisationally, or, as he puts it, "on the edge of creation."

Ultimately, modern technology aside, it is this "edge" that allows David Torn to pursue what he calls "the joy of being able to capture a cool sound on tape." His technological fluency is only one piece of the picture.

—Michael Birnbaum

LIT #	BRAND	MODEL	ARTIST	USE
102	Nady	1200 VHF Wireless	INXS	guitar
			Mick Jagger	vocals
121	Panasonic	WMS2	Miles Davis	trumpet
		WMS5	Tom Petty	drums
122	Peavey	PVM 380N	Michael Bolton	vocals
123	Samson	Broadcast Beltpack	Madonna	vocals
		Broadcast Wireless	MC Hammer	vocals
124	Sennheiser	MD 421	Paul McCartney	drums
		MKE48	Janet Jackson	vocals
		SK2012	Doobie Brothers	guitar
125	Shure	VP88	Little Feat	cymbals
			Paul Simon	percussion
126	Sony	WRT-27A	David Bowie	vocals
		WRT-28A	Billy Joel	guitar
		WRT-67A	Madonna	vocals
127	TOA	HY-3	Psychedelic Furs	keyboards
		K-4	Billy Ocean	vocals

Chart research by Paula Heer. All information provided by manufacturers; no celebrity endorsement or exclusive use implied.

AN EXPLOSIVE PERFORMANCE

Several musicians seeking stardom reportedly saw stars when a flash pot at a Holiday Inn cocktail lounge gig in Punta Gorda, Florida malfunctioned and blew up on December 1, damaging the dance floor and sending seven people to the hospital — one in critical condition. The local band, In and Out, had been using the homemade pyrotechnic device for the past two years with no ill effects. The incident, which injured 24, is currently under investigation.

While performers may be concerned with the welfare of their brand-new MIDI mixer, a band should never overlook the overall safety of its members, crew and audience. A thorough pre-show safety check of all systems — sound, lighting, rigging and special effects — is an essential element to the success of every show, regardless of scheduling conflicts. Remember: safety first. Or else. *— Joseph Spiegel*

EQ On-Line

Need help programming a new synth? Want to check out the latest sequencer software or get some quick advice on setting up your multitrack? You can probably find the help you need from The MIDI/Music Forum, a telecommunications service on the CompuServe network.

This 11,000 member-strong resource network provides authoritative answers to MIDI-based questions an average of 300 times a day. Topics covered include everything from new products to general music to highly technical MIDI applications.

The 24-hour-a-day Forum supports all computer formats and also offers some helpful services including a segment that allows the user to download sequencing software for demonstration, and a "Song Collaboration" Library where members upload songs, tracks or melodies to get feedback from other members.

The service will also carry synopses of various EQ stories allowing its readers to leave questions on-line for the editors and other subscribers. And by joining through EQ, a subscriber will get a free membership to the service and $15 free connect time. There are no long distance charges, since all connections to the Forum are local. Watch for this special offer in EQ or write to the magazine for more details.

Anatomy of a Remix

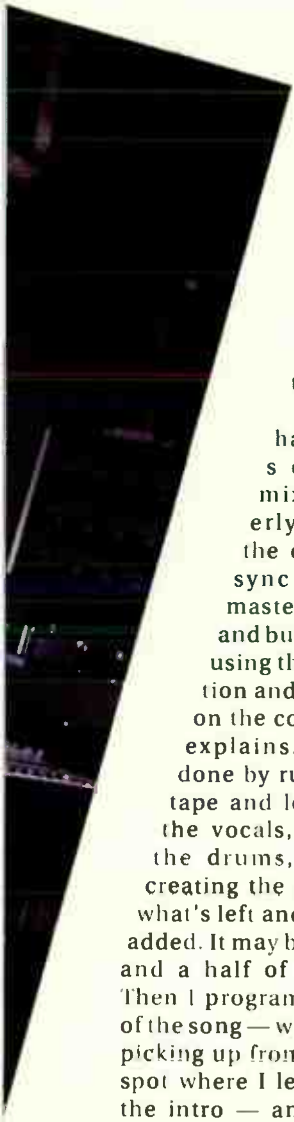
Junior Vasquez takes a multitrack project and, by dissecting and rearranging it, creates all-new remixes that explore the different grooves that lie dormant in a song. The result is what DJ's love to spin and single-buyers send up the charts (Photo by Peter Monroe).

By the time **Junior Vasquez** finished remixing **Sheena Easton's** new single "Come Back 2 Me," a **Prince**-penned and produced track from her upcoming album, the song bore little resemblance to the 16-track master that left **Paisley Park Studios.**

In fact, of the 48 tracks Vasquez ultimately used on the project, only the vocals, Prince's guitar parts and some percussion tracks from the original remained. "It's like I wrote the song again in house form," he explains.

Vasquez, who previously worked with Prince on ten cuts off the soundtrack for his new film, **Graffiti Bridge,** created three versions of "Come Back 2 Me" — one for the album, one for a 12-inch "club" release and a third "underground" mix. Upon receiving the master, Vasquez analyzed the tracks and decided which to keep and which to discard. He then booked studio time with drum machine programmer and keyboardist Joey Moskowitz, and filled up the remaining tracks on the 24-track tape with drum samples and keyboard parts using **Roland's 909** and **808** drum machines, **Akai S-1000** samplers, a **Korg M1** synthesizer and **Juno 106** keyboard — all of which he used to mix the album version of the song. For the club and underground remixes, Vasquez added a wide variety of tape loops, percussion sounds and tape samples. "I use a lot of samples — recycled stuff that you get from other songs or other places or drum machines," he notes. "I also try to use a lot of peculiar things like pots and pans sounds, weird synth noises and so on. But instead of programming a tambourine from a drum machine, we'll just loop a tambourine sound that we like. It works because house music is just made up of four- and eight-bar measures. There are no bridges or chord changes to worry about."

Tapes in hand, Vasquez headed to New York's **Right Track Recording**, where he mixed all three versions of "Come Back 2 Me" on the facility's **SSL G Series** console. Before he began mixing, however, Vasquez employed a range of effects — such as **Massenburg** preamps and **Pultec** EQs, an **Eventide H3000**, a **Lexicon 480L** digital effects processor, a **Yamaha SPX-90** and an **Akai MPC 60** composer — to beef up



his additional tracks.

"After I have all my sounds mixed properly through the effects, I'll sync up the master to a slave and build an intro using the automation and the faders on the console," he explains. "This is done by running the tape and leaving out the vocals, the bass, the drums, and just creating the intro from what's left and what I've added. It may be a minute and a half of just that. Then I program the body of the song — which is just picking up from the same spot where I left off with the intro — and I run it with the vocals and everything else. I copy that onto DAT and then to half-inch. This is where I begin editing and making it longer. You get to a point in the song — let's say three minutes in — where you want the break to be. I develop that in the automation, too. So, I print that piece and edit it to the body of the song. I'll also have an automated piece of the continuing vamp on the ending. Once it's done, you punch up the version you want to lay down to tape, hit the button and the computer just dumps it out. Then it's on to the next one."

— *Steve Schwartz*

EQ HOT PICKS

☐ **Audio Logic** has bowed a line of six new signal processors, including four new crossovers (X 23 Stereo 2-Way, Mono 3-Way; X 34 Stereo 3-Way, Mono 4-Way; X 22 Stereo 2-Way; X 32 Stereo 3-Way), a four-channel noise gate (440 Quad), and a gate compressor/limiter (Model 266). Circle EQ Free Literature number 128.

☐ **Shure Brothers** recently introduced a new line of video production microphones, and the first model to be unveiled is the VP88 single-point stereo condenser. This model uses two independent mic elements to produce a classic Mid-Side stereo signal, and is expected to be particularly useful for video field production. Its two condenser mic cartridges are mounted in a coincident fashion to produce a stereo signal that is mono compatible. The VP88 is also suited for stereo sampling, DAT recording, and on-stage instrument miking, and carries a net price including accessories of $995. Circle EQ Free Literature number 125.

☐ Japanese manufacturer **Sanken** has boldly entered the microphone melee with its CMS-7 and CU-44X models. The CMS-7 is a portable M-S stereo condenser with a dynamic range of 108 dB and an almost flat audio frequency response range. The transformerless CU-44X is billed as being the world's only dual-capsule condenser mic. It offers transparent characteristics for digital recording. Circle EQ Free Literature number 131.

Information Society Live

If producing electronic music is difficult in the studio, it's one hell of a job to perform it live. But **Information Society** aims to do just that.

The electronic dance group is packing up its samplers and sequencers for an upcoming 1991 tour to promote their MIDI-made album, "Hack."

Lead singer **Kurt Harland Valaquen** explains that the technical side of the stage show will center around four to six **Akai S-1000** samplers that the three-man band will use to load most necessary sound effects into sampler formats. "We're going to make a patch on our samplers for everything," Valaquen says, adding that "the Akai S-1000 is the piece of equipment I value most."

For their first tour in winter '88, the band used a sequencer on stage a lot. But today, a Voyadrift SP3 sequencer's only use will be as a clock and output for repetitious parts. Backstage, however, the sequencer is a major player. Each member composes on the SP3 rather than paper.

Information Society has spent a lot of time preparing for the tour due to the enormous amount of responsibility placed on each member. "I don't believe there are any limitations to performing our music on stage...except that you have to be willing to do all of the work," Valaquen says.

— Jon Ruzan

TBS TEST (TROUBLESHOOTING BAD SOUND)

So you think you know how to handle any technical problem you might face in your project studio. Well, tinkering with tape machines is one thing, but correcting bad sound is quite another. Here's a quiz to tell if you know how to solve sound problems.

Match the sound problem in column A with its possible solution in column B:

1. Dull sound or dropouts in playback
2. Muddy sound or excessive leakage
3. Excessive reverberation
4. Dull sound
5. Distortion in mic preomp
6. Sibilonce
7. Noise (low frequency)
8. Bass Boost close to mic
9. Sound lacks clarity
10. Excessive noise (hiss)

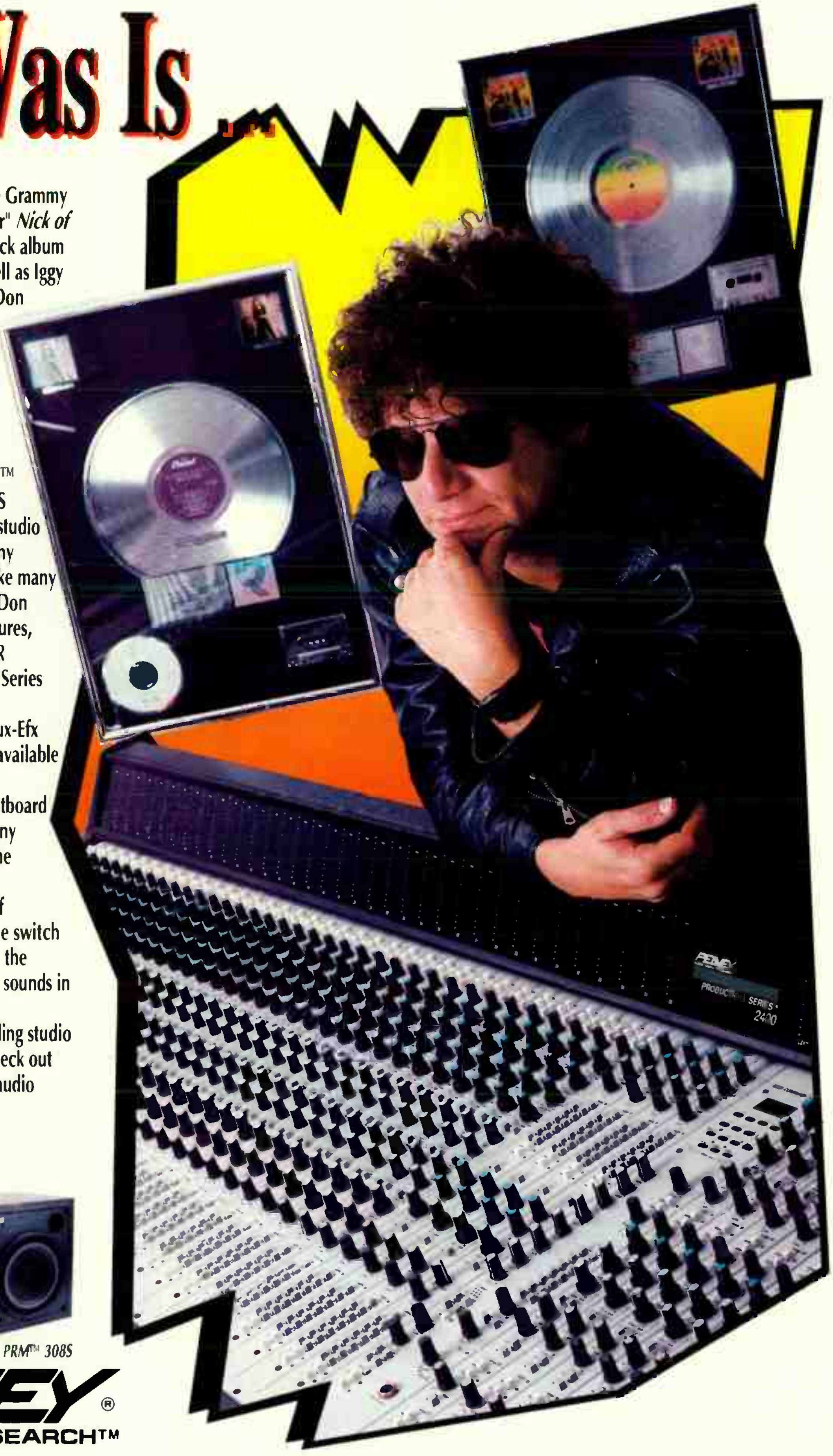
A. Use a high pass (low cut) filter set around 40 to 80 Hz
B. Reduce aux-send or aux-return levels
C. Use omnidirectional microphones
D. Increase input attenuation or plug in pad between mic and mic input
E. Use a de-esser signol processor
F. Use exciter signal processors
G. Record electric instruments direct
H. Clean and demagnetize tape path
I. Scrap your console
J. Delay the reverberation send signol by about 30 to 70 ms
K. Check for noisy guitar omps or keyboords

Questions and onswers were derived from *Recording Demo Tapes at Home* by Bruce Bartlett another book in the John Woram Audio Series from Howard W. Sams & Company. To find where you can get a copy or order directly from the publisher call 800 428 SAMS

ANSWERS 1 H 2 G 3 B 4 F 5 D 6 E 7 A 8 C 9 I 10-K

Don Was Is ...

...hot! Producer of the Grammy Award-winning "Album of the Year" *Nick of Time* by Bonnie Raitt, the comeback album *Cosmic Thing* by the B-52's, as well as Iggy Pop's new album *Brick by Brick*, Don also collaborated on the critically acclaimed Was (Not Was) album *What Up, Dog?* and the hit single "Walk the Dinosaur."



Don chose the Peavey Audio Media Research Production Series™ 2400 Console and the PRM™ 308S Phase Reference Monitors for his studio after searching for and testing many different brands on the market. Like many other professionals of his caliber, Don appreciates the performance, features, quality, and creative freedom AMR equipment offers. The Production Series 2400 Console features 4-band full sweepable EQ on each Input, 8 Aux-Efx Send/Returns, 92 Inputs with EQ available for mix-down, an on-board MIDI command center for control of outboard devices, low-noise design, and many additional production facilities. The PRM 308S Monitors have uniform frequency response, a minimum of coloration, and feature a selectable switch for reference and EQ playback for the option of hearing how your music sounds in an alternate environment.

If you're designing a new recording studio or upgrading your present one, check out the complete line of professional audio equipment from Peavey . . .

DON WAS DID.

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CIRCLE 30 ON FREE INFO CARD

Gnome at Home...

Guitar and bass player Bruce Nazarian rocks and reels in his L.A. project room.

STUDIO NAME: Gnome Studio, within Studio 55, Los Angeles, CA.
OWNER/OPERATOR: Bruce Nazarian
CREDITS: Guitarist and bassist for Was (Not Was), engineered and played on Anita Baker's hit "Giving You the Best That I've Got," original score for the films "Air America" and "Turner & Hooch," member of Brownsville Station.
STUDIO DESIGNERS: Bruce Nazarian, Todd Wilson.
DESIGN NOTES: Conventional room-within-a-room construction. Isolated concrete slab floor. Wall forming is alternated codes of gypsum wall board. Multi-layer construction. 14 x 18 control room space. Designed as a personal workspace for a variety of projects including composing, film scoring, and post production.
CONSOLE: Trident Series 24 with 44 inputs and an additional full 24-track monitoring section.
TAPE MACHINES: Studer 827. Access to a variety of tape machines in Studio 55 including Sony and Mitsubishi digital multitracks.
ANALOG TAPE: Ampex 456.
MONITORS: Tannoy LGM-12s, PBM 6.5, Yamaha NS-10.
OUTBOARD GEAR: Focusrite 115 preamp, AMS delay and reverb, Eventide H3000 Ultra Harmonizer, Lexicon 224XL digital reverberator, Lexicon LXP 1 digital reverberator and LXP 5 digital reverberator, Yamaha SPX90 digital effects and reverberator, Yamaha SPX1000 16-bit digital effects and reverberator, Drawmer gates, Yamaha Rev 7 and Yamaha Rev 5 digital reverb and effects.
KEYBOARDS/SYNTHESIZERS: NED Synclavier expanded to 32 MB with 32 voices. Also Akai S-950, Midi Moog, Roland D50, Proteus, Oberheim Expander, Kurzweil Module.
MICROPHONES: Neumann U87, Neumann TLM170, AKG C414, AKG 2, Sennheiser 441 and 421, Shure SM47.
MIDI: Apple Macintosh II, Performer, Digidesign Sound Tools with external hard drive, Mark of the Unicorn Midi Time Piece, NED MidiNet, J.L. Cooper MSB 16/20.
OTHER STUDIO EQUIPMENT: PSTech DirectBox, Fostex D-20 DAT, JVC CR-8250 3/4-inch with Lynx synchronizing system.
EQUIPMENT NOTES: "The Trident console is physically the smallest 44-input console out there, and the thing I like best about it is its large number of inputs and full function monitoring section. The console's sound is very familiar to me and it sounds great for the price. It's also the perfect size for this studio, and it lets me run 24 tracks while routing through live MIDI. The monitoring section, in particular, is extremely useful because it also provides me with equalization. The Synclavier is my sketch pad. The most detrimental thing is losing the magic that you've created on the demo when you're putting it onto a master. Going from demo to master is where you can lose all the life and vitality of an idea. When I come up with that magic — an idea for a song or a film score — I can give it to the Synclavier and come up with a master-quality demo. It lets me program music that feels very real and allows me to be a multi-instrumentalist at the same time."
PROJECT TIPS: "The key is to try to bridge the gap between personal comfort and technical capabilities. You can have a wonderful, high tech play room, but if you can't live in it, your creative vibe will be off kilter. You really need to establish a comfortable environment so that the technology sparks your creativity and doesn't stand in the way. If your creativity is compromised in any way, you've totally negated the advantage of having your own project studio in the first place."

Photo courtesy of Trident


MIDInet Master
MIDInet Expander
Roland
SOUND DESIGNER II

CIRCLE 09 ON FREE INFO CARD

The Barber Shop

STUDIO NAME: Billy Barber Productions, Greenwich, CT.

OWNER/OPERATOR: Billy Barber, songwriter/producer.

CREDITS: Wrote Number 1 hit "Little Things" by Oak Ridge Boys, themes for TV series "American Chronicles" and soap opera "All My Children," member of Flim & the BBs, wrote and produced for a variety of TV commercials.

STUDIO DESIGNER: Billy Barber.

CONSOLES: 3 Yamaha DMP7s,Alesis 1622.

TAPE MACHINES: Sony 100 DAT recorder, Fostex E3 2-track with SMPTE, Revox B215 cassette deck, Akai AKM14D 12-track.

MONITORS: Westlake BBSM6, Yamaha NS-10M.

WIRING: Monster Cable.

KEYBOARDS/SAMPLERS/SYNTHESIZERS: Korg M1 synthesizer, Roland D50 synthesizer, E-Mu Proteus, Kurzweil StringExpander with woodwind card, Roland R8 drum machine, Yamaha RX5 drum machine, Prophet VS synthesizer, Akai 900 sampler, Prophet 2002 sampler, Kurzweil 250 sampler, Prophet 5 analog synthesizer, Kawai PH50 synthesizer, Yamaha KX5 keyboard controller.

COMPUTERS & SOFTWARE: Macintosh IIcx computer with Vision sequencer software, Wise 286 computer with Texture program, Atari 1040ST with Hybrid Arts ADAP software.

OUTBOARD GEAR: 360 Systems Matrix, dbx 160 limiter, Symetrix 501 limiter, Sony 501ES D/A converter, Acoustat TNP pre-amp and power amp, Sony VO9800 3/4-inch video recorder with Address Timecode.

MICROPHONES: Neumann TLM-170, Bruel & Kjaer 4011, Sennheiser 421 and Shure condenser models.

EQUIPMENT NOTES: "The three Yamaha DMP7s comprise the most important part of my entire project studio setup," Billy Barber says. "After I hire a recording engineer to come in for the final mixdown, I can save all of his work, including all the complex EQ and fader moves, and it is always possible to call it back up later if for some reason I need to make changes on a few select tracks. I also find that it's much, much better working with the Yamaha's than dealing with a large, beast-like analog console. They allow me the same flexibility with eight inputs each and 24 total tracks, and I can simply tie them into the Alesis 1622 when I need to generate more tracks.

PROJECT TIPS: "Take a good look at the studio-in-a-box concept; the less equipment the better," he suggests. "And be sure to keep your focus on creativity more than on technology. It's human nature to get totally caught up in all the technology, but try not to lose your perspective."

Photos by Peter Monroe

Make A Room Sing — For $3,500

We've all heard someone say (if only in jest), "I've got a barn. Let's put on a play." For better or worse, it's an epidemic attitude in recording these days, with a new generation of high-quality, low-cost gear enabling sound pros to build rooms in their own offices, homes, garages — and yes, barns.

One of the world's leading acousticians and studio designers tells how to prepare a low-cost studio that sounds like a million bucks.

TOM HIDLEY

But no matter how good the equipment and gifted the musicians, it all comes to naught if the acoustic environment isn't tailored to deliver true and honest sounds, free of distortion and wild-flying sound reflections that will fool the engineer into making bad musical decisions.

Acoustic design is an exacting — and to some, almost mystical — science, and it is probably the most neglected aspect of sound control in today's modern studio. Time and again, I've seen people build rooms with the best of everything, only to be stumped in the end when it all sounds non-musical. And that's sad, because the principles needed to make a decent basic and clear acoustic (musical) sound space are fairly simple and inexpensive.

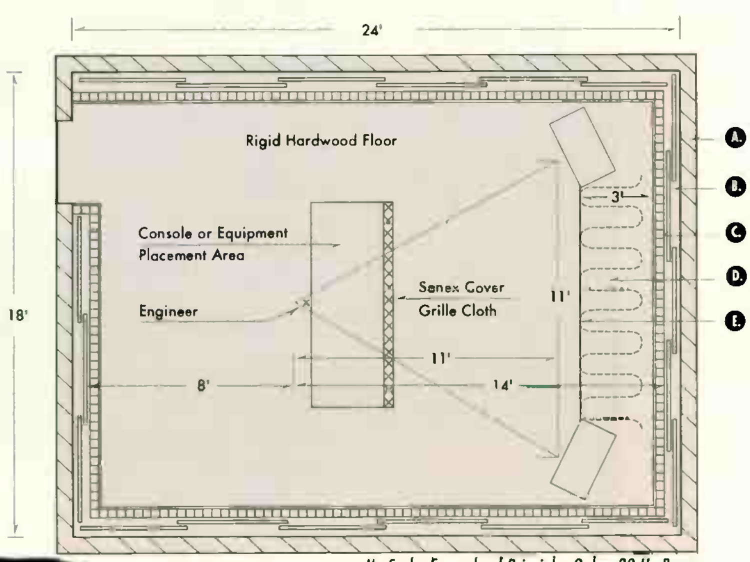
THE RIGHT ROOM

So how do you do it? Well, in the first place, we have to assume that you have a room to begin with. It shouldn't be a warehouse or an airplane hangar, or some other enormous structure; your sound will reflect uncontrolled — everywhere. So assume you already have a room with a floor, ceiling and four walls — ideally, it's symmetrical. For our example, I've picked an 18 x 24 foot room (see the accompanying diagram).

The room itself should be reasonably isolated from the neighbors, for your and their peace of mind. The best situation is to have a concrete floor on ground level.

Preferably the floor isn't a wood joist system, and preferably it has no windows (if so, let's hope they're double glazed). Try your best to avoid using a room with a tin or "live" roof, and in the best case scenario, there's an attic overhead or some such isolation space. If you don't have a ground floor site, then putting your room on the top floor is a better choice than using a middle-floor room where people will be walking overhead.

The size of the room is an important consideration



No Scale: Example of Principles Only - 30 Hz Room

A. Exterior Wall
B. Flank Blanket Cover 1" Fiberglass
C. Open Frame Wall Cover Grille Cloth
D. 2' Minimum Dense Rockwool
E. Rockwool Cover Grille Cloth

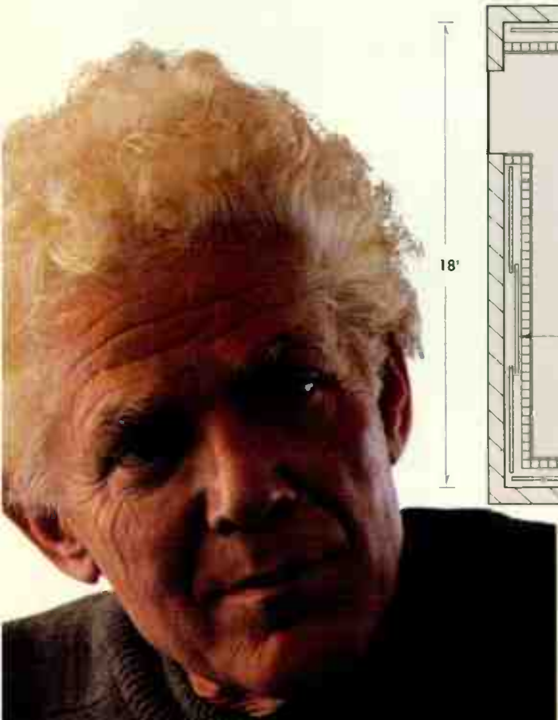


Photo of Tom Hidley courtesy of Studer Revox

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

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

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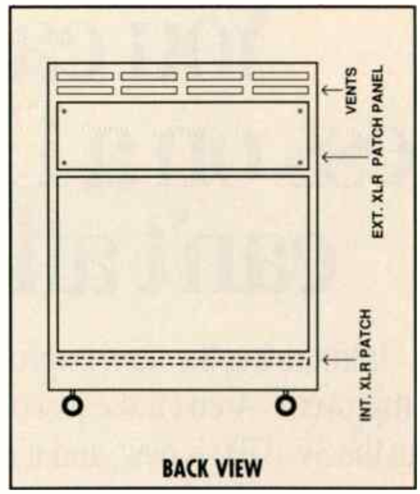
CIRCLE 29 ON FREE INFO CARD

— and it isn't always a factor over which you have absolute control. Look for a room that would complement the lowest frequency of the monitoring system that you intend to use.

Basically, you need to look for a room with a physical long dimension of half the wavelength of the lowest frequency your monitor can produce— so for a 50 Hz monitor, you need 14 feet of room front to back dimension. A rule of thumb to remember is that every time the frequency reduces an octave, the distance needed doubles. Thus, for 25 Hz, you need a 28-foot room. Here's a table to help you (in half wavelength — monitor low frequency to room):

60 Hz — 12 ft
40 Hz — 16 ft
30 Hz — 24 ft
20 Hz — 32 ft

So let's assume that before you even start spending any money on acoustic construction, you already have a space and a set of monitors that match. Set up the longest dimension of the room as the length dimension from the monitors to the rear wall. The side-to-side room distance should ideally be the shortest, and higher ceilings are better than lower ones.



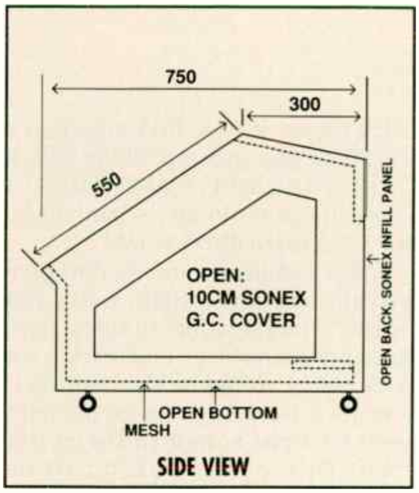
For essential "in room" gear, I suggest a wooden "low boy" portable rack construction whose front mounting panel geometry reflects arriving sound away from the engineer/producer. Dim. given in cm. Scale-1:10

Now, since we're working on a limited budget, we have to get our priorities straight from the beginning. Believe me, the largest potential problem in any listening room is uncontrolled and certain unwanted sound reflections. Obviously, reflections are needed in any human listening indoor environment, but if they fly loose, the result is a dishonest room — and you'll end up making incorrect musical decisions.

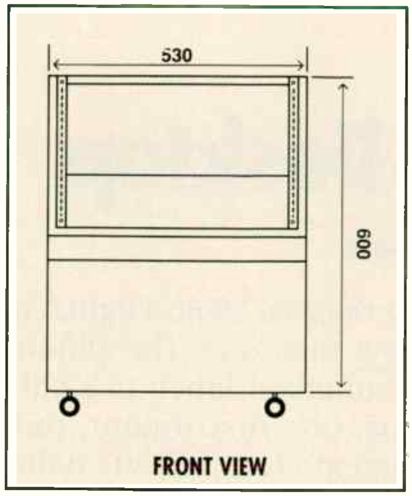
SOUND FLOOR

If your priority is to control the reflections (and it should be), start with the floor. This will be the prime source of room reflection in our room. In the final analysis, you want structural stability in your floor. When completed you want to end up walking on a rigid concrete, hardwood finished floor. Ideally, its finish is butt-jointed, without ridges and no beveled edges, and polished to a sparkle. I use very stable laminate woods manufactured in Scandinavia by Kährs and Tarkett. Both floorings are about 15mm thick. An inexpensive alternative is using a prefabricated, 1/8-inch hardwood parquet available at most flooring stores.

Of course, what's underneath that wood surface is important as well. First, you use mastic to fill in any pits



SIDE VIEW



FRONT VIEW

and holes you may have in your base cement floor, making sure it's perfectly flat. Cover it all with glue, and lay down a 3/4-inch tongue-and-groove moisture-proof sub-flooring of something like chipboard or particle board) over the entire area. It's best to shoot this into the concrete with Hilti — the trade name for a tool that shoots special Hilti-hardened nails into concrete or stone. When the sub-floor is secure, glue and blind tack your finish hardwood floor onto it.

Don't lose all hope if you don't have a concrete floor. In many cases, it's possible to lay 3-inch dense Rockwool and 1-inch shutter plywood on top of the existing wood floor and pour concrete (at least four inches) on top of the ply to achieve a float slab condition. I would advise consultation with a structural engineer first, or you may end up with a collapsed building. If you can't pour a new floor, another alternative is to put the monitors up on a coil-type spring whose resonant frequency is at least one octave lower than the lowest frequency the monitor is capable of. But this is not a cheap way out — purchasing 1 Hz to 10 Hz industrial viscous damped springs can be costly.

One U.S. company, Kinetics, makes neoprene isolators that can get down to about 10 to 12 Hz. If you provide Kinetics with the needed data, they can tell you what kind of base to build and how to place the pads (and which pads to use) for best results. Air pressure from the monitor will still create unwanted motion in a wooden frame floor, but the right monitor base can go a long way in minimizing impact transmission to the internal wood floor structure of the room.

WALL HANGING

The next step is to deal with the walls. Basically, it is necessary to construct and hang four walls of 1/2-inch flanking blankets and then visually hide them with acoustically transparent grille cloth. These "flanks" help negate the horizontal standing wave pressure buildup and keep the room sounding smooth. They are hung 6 to 12 inches into the room parallel to the building walls and hung from the ceiling with eye hooks and wire, allowing them to swing free to the floor, clearing it by a

continued on page 72

From DAT To Desktop

LAURENCE JUBER

The title of my first CD release, "Solo Flight," is applicable in more ways than one. The album, on the Beachwood/Chameleon label, is a collection of acoustic guitar solos: one instrument, one player, and for the most part, one modest project studio located in my home.

You see, I recorded the bulk of "Solo Flight" in my Sign of The Scorpion project studio, taping straight to DAT and assembling the cuts on Digidesign's Sound Tools through an Atari Mega ST 4 computer.

TOOLING UP

An Aiwa consumer DAT deck was my primary recording machine for this project. Though you'll hear advice to use only a professional quality DAT recorder, this unit was totally sufficient for my needs. I improvised for hours, and then streamed all the data into Sound Tools. Using the program's block define and playlist functions, I assembled what would eventually be the final release.

Sound Tools assembles your music by defining portions and then letting you put together a playlist. I'm particularly impressed by its advanced cross fades. It allows you to assemble in a compositional sense and, as such, it is a very important creative tool.

Sound Tools is configured in several modules, allowing users to design custom-tailored systems. The three main components are the Sound Accelerator DSP (digital signal processing card), the Sound Designer II software package, and an analog, pro analog or digital interface.

Using the Sound Designer II software, I was able to see "zoomable" waveform graphic representations of my improvisations, and could shape, edit and record them note-by-note to fit the final arrangement. Meanwhile, Sound Tools acted as a recording device, and offers scrolling much like a tape transport. It is equipped with high resolution VU meters with clipping indicators and user-selectable sampling rates from 8 to 48 kHz.

MAKING TRACKS

The guitar tracks were laid down with very little EQ or compression. Using a pair of B&K 4011 mics and a TubeTech mic preamp, along with a CAD Equitech and an AKG "TUBE," I miked the guitar in an X-Y and modified X-Y pattern direct to DAT.

This technical combination gave the solo album its bright, warm and organic sound — achieved with a minimum of manpower and within an extremely reasonable budget. Actually, I have to give a lot of credit for the bright sound to the guitar itself. This steel-stringed custom instrument was built by luthier John Le Voi from a Brazilian hardwood called imbouya — similar to rosewood but with a somewhat more effervescent reverberation and timbre.

An "analog-like" warmth was provided by the TubeTech mic pre, while I also used vintage Neve outboard EQ on several cuts, streaming the audio from Sound Tools back to DAT using very light compression (also Neve). The old Neve modules gave the music an added texture and subdued them just the right amount.

On one of the album's most animated cuts, "This Process (Is a Process)," I also used an Eventide Harmonizer to add an upper and lower octave with a slight delay for even more enhancement.

BRINGING IT HOME

I should credit my co-engineer, Avi Kipper, a successful L.A. scoring engineer, for helping to tweak the signals to sonic perfection before they were ultimately mastered at Ocean View Digital Recording by Joe Gastwirt. We mastered with Sonic Solutions running on a Macintosh IIcx, which is largely responsible for giving the work its overall acoustic consistency.

The bottom line is that with the emergence of computerized digital music assembly — Sound Tools being a prime example — it is possible to achieve the sonic excellence of a top studio in virtually any environment. I was able to play uninterrupted by technology and then I could assemble and edit to perfection at my desktop PC. Give the album a listen. The sound speaks for itself.

Juber, *former Wings lead guitarist, is also a well-known composer, studio musician and film scorer.*

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CIRCLE 35 ON FREE INFO CARD

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The Five Day Demo

Making a video demo at home: the agony & the ecstacy. An A/V nightmare in five days.

PAUL D. LEHRMAN

If you want to get film scoring work, you need more than just a great audio demo cassette. Producers and studios get hundreds of them a week and invariably dump them in a pile labeled "Listen to these some day." To stand out, you need to show not only how great your music is, but how well it supports someone else's images. In short, you need to make a video demo.

I've done a few video and film projects, so I figured it was time for me to put together a videotaped demo. My goals were to get the best possible audio quality without sacrificing video quality at the lowest possible price. It had to be a knockout — in under 12 minutes.

Getting good-quality 3/4-inch video dubs from the various producers I had worked with was no problem. Trouble was, all the audio mixes were done to serve the purpose of the film, which meant mono, highly compressed, and with the music well underneath voices and effects, at levels ranging from acceptable to nearly inaudible. I would have to remix everything.

Since I composed and recorded all the music in my home studio, using a computer, SMPTE-to-MIDI Time Code converter, and various sequencing programs — I still had all the files — I could have edited and remixed at a video studio with a Macintosh-based MIDI room, but it would have cost a fortune. I decided to do the remixes at home. I would first assemble the various video segments together on 3/4-inch tape in a real editing studio, and then would rent a 3/4-inch machine, lock up my computer to it, and lay my audio tracks, in stereo, back onto the same tape. A friend with an editing studio that wasn't particularly busy offered me cheap time, and I found a rental house that would let me have a Sony BVU-150, a cute, portable, stereo 3/4-inch deck, over a weekend and just charge the one-day rate.

Day 1: After a few days of going over the material and deciding which segments to include, I go to a friend's studio with a stack of 3/4-inch cassettes. We use a character generator to title each segment, and then assemble the six parts. We leave the audio in place, because I'll need to refer to it later, and lay SMPTE timecode on the address track.

One segment gives us trouble: we're splicing three sections from different parts of the film, and if the music is going to run through the splices, the cuts have to happen on the beat. Unfortunately, some of the scenes stop before the beat, so we have to grab a few frames from elsewhere in the film to fill in the gaps.

We finish in a few hours, and make a workprint of the master on a VHS Hi-Fi deck I brought (very few video studios seem to own VHS Hi-Fi equipment), putting the dialogue, effects, and music on one channel, and the timecode on the other. This way, I won't have to keep shuttling the master tape, possibly wearing it out, while I'm editing the music. It also allows us to make a "window burn" — putting the timecode number for every frame on the screen — an immeasurable help when editing.

Day 2: I pick up the Sony 3/4-inch deck from the rental house. I want to play back the video, sending the timecode to my sequencer, and as the sequence plays, lay the mix back to the two audio tracks on the 3/4-inch tape. Everyone assures me this will work just fine.

When I get home, I discover that my sequencer won't sync to the VHS workprint. Every 15 seconds the "lock" light on my SMPTE-to-MTC converter goes out. I play with levels, try different converters and a timecode re-shaper, but nothing works. Fortunately, I have a beta version of Mark of the Unicorn's new Video Time Piece, which can do window burns. So I cue up the 3/4-inch tape, run it through the Video Time Piece, and make a new window-burn dub on my VHS Hi-Fi deck. It goes smoothly, and when I play back the new workprint, the sequencer locks perfectly. So much for "professional" SMPTE gear.

Re-creating the musical score for these segments isn't going to be hard. But there are also sound effects, voiceovers and on-screen dialogue that I can't re-create, because I don't have the source tapes for any of them. What I have to do is strip that non-musical audio from the source tapes and store it in a format from which it can be played back mixed, and in sync, with the sequenced music.

To make things more complex,

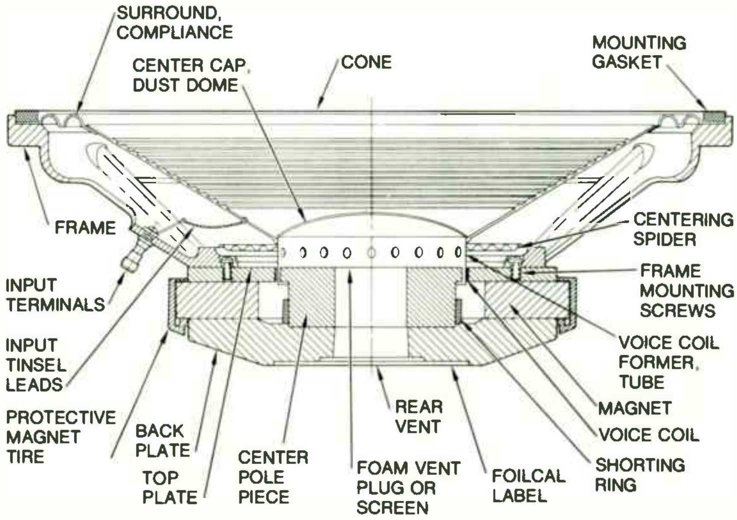
OUT OF THE BOX

If the speaker is out of the box, look at the spider-(inner surround)-to-voice coil tube glue joint. Check to see if it is attached all the way around (a little light finger pressure is the test method here.). This is a terminal failure if it is left untreated.

Look at the dust dome itself to ensure that it is intact. It keeps the crud out of the gap. If it is compromised, all sorts of trash, especially little tiny bits of ferrous particles, will crawl into the gap, causing rasping and buzzing. This is a critical place to watch for in coaxial speakers, as the surround on the high frequency horn is often not as damaging to the dust dome when it is damaged.

Take a look at the flexible wires, or 'flyleads,' that run from the speaker terminals to the cone. If they are burnt or discolored, check to see if they have become limp. If they have lost integrity, they will fray and fail. These can be replaced without reconing the loudspeaker.

Look at the voice coil leads where they run up the cone from the voice coil. Look for signs of cracking in the glue that holds them down at the cone apex, where the cone is glued to the voice coil tube. If there are cracks, the copper wire may break under the fractured glue.



Cutaway drawing of typical low frequency transducer (courtesy of JBL)

NUTS AND BOLTS

Check the mechanical bits too. Make sure the bolts that hold the magnets onto the frame are tight. Aluminum tends to flow, and the area around the bolts will often relax and let the magnet move on the frame. This will often cause mystery misalignments, usually found when the cone is sliced out for a recone. Check the bolts on any bolt-in cone assemblies; they tend to come loose.

Heat is a big killer of loudspeakers, and if you want to find out just how hot your speakers get, find an industrial supplier of those little stick-on liquid crystal thermal indicators. Apply them to the aluminum dust dome, at the center and the edge, and to the magnet. They will change color with changing temperature, and give you a good visual indicator of how hot things are getting.

Make sure the magnet vents are clean and free of obstructions, since the magnet acts as a heat sink. The voice coil dumps heat into the magnet until the magnet reaches a thermal equilibrium with the surrounding air. If the magnet is prevented from dissipating heat, its temperature continues to rise. Because heat only flows from hot objects to colder ones, the voice coil temperature can rise to the melting point of the glue.

For this reason, always keep the fiberglass away from the magnet assembly. Fiberglass is a wonderful insulator; that's why they stuff it in walls. If your magnet ends up wrapped in fiberglass, it can't get rid of the heat, and your voice coil will melt down.

TESTS AND MEASURES

If you can do full bandwidth impedance measurements, you have a powerful tool for inspecting your loudspeakers. An RTA can be set up to do this, and the method should be outlined in the owner's manual. Doing baseline measurements of the devices you use allows you to look for changes in their state without looking at them physically. A sudden drop in loudspeaker impedance can be the result of a voice coil having some turns bypassed by shorting and arcing.

Changes in the shape of the impedance curve down below 250 Hz, where the motional component of the loudspeaker makes a big impact, can mean a mechanical problem such as voice coil rub or insulation blocking free movement of the cone.

If you're diagnosing failures and have the cone and voice coil out of the speaker frame, look for signs of burning on the windings. If you remove the voice coil former but the wire is missing, it's probably still in the gap somewhere. These are sure signs of excess heat failure. Also look for folding of the bottom edge of the voice coil tube. This indicates a centering problem near the limit of

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RUB 'N BUZZ

To use an oscillator to check for rub or buzz, you will need a sweepable oscillator that will produce 20 Hz to about 200 Hz. Connect this to a small amplifier (20-50 watts is plenty) and then connect the speaker to the oscillator. Starting at 40 Hz for most 15-inch bass drivers, turn up the level 'til you get approximately 1/4-inch to 3/8-inch, peak to peak, of cone travel. If the speaker is still clean, sweep it down to 20 Hz. The cone travel should increase and large excursion rubs or buzzes should be audible. Don't put so much power into the device to cause stretching of the surround or bottoming of the voice coil. This should be at just an adequate level to see reasonable excursion. The reason you need as much as a 20-watt amp is to ensure a reasonably clean signal at 20 Hz. Now sweep it up to 200 Hz. The upper region is where you will find most cone and dome buzzes from bad glue joints.

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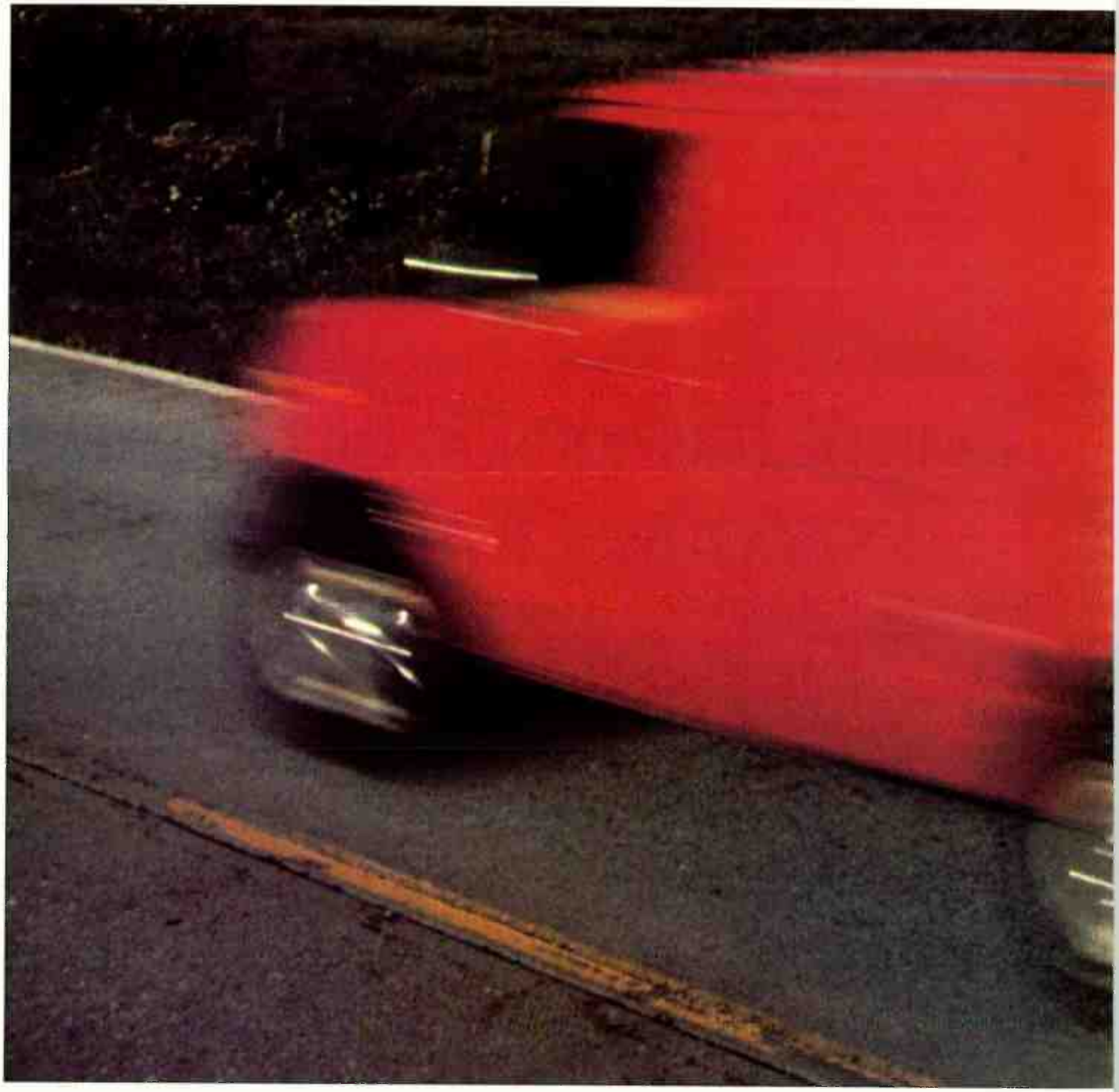


BAND IN A VAN

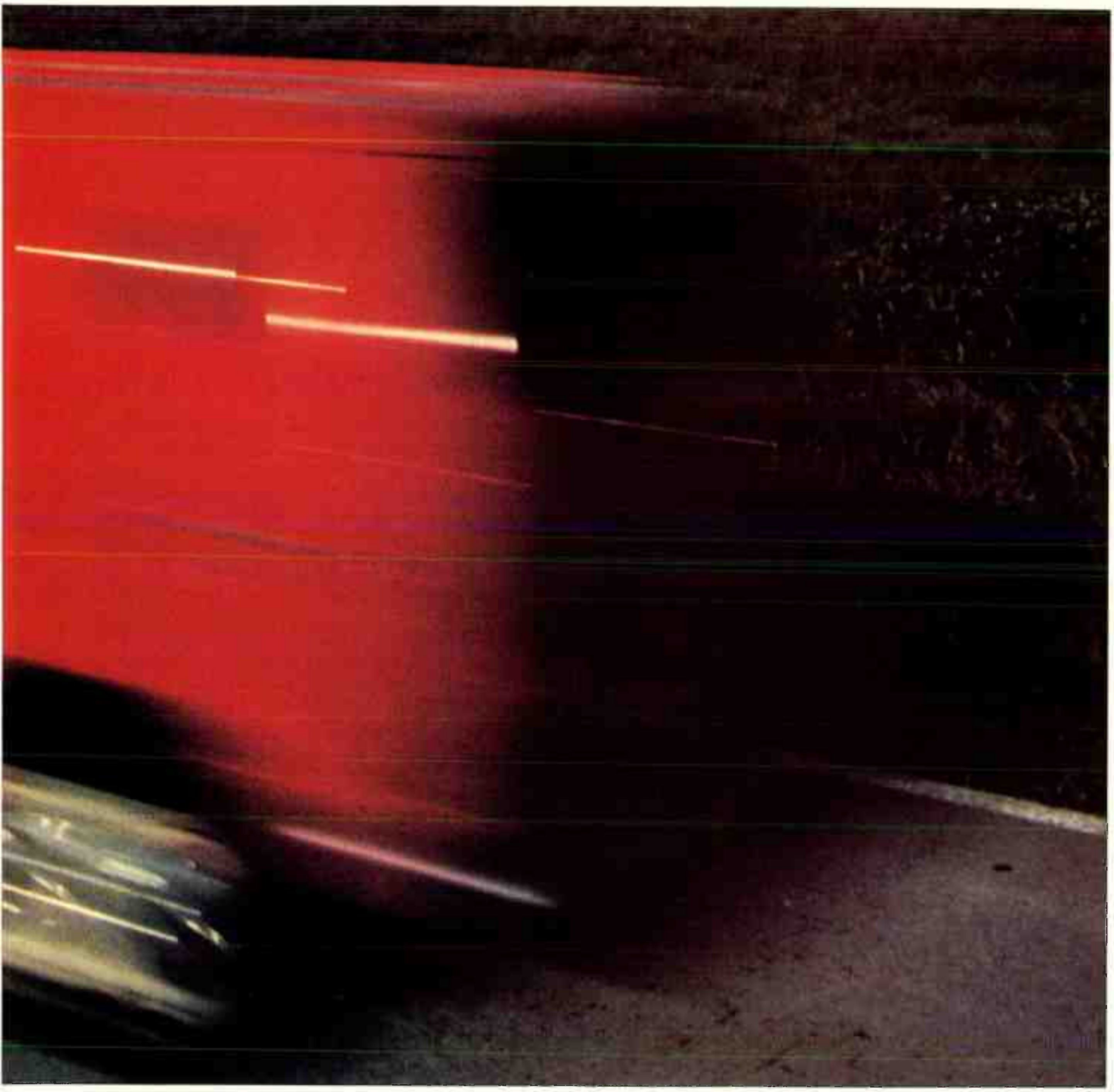
How to pack your live sound system into limited cargo space and still make it to the gig on time.

RICK CHINN

You got a van, you got a gig, you got a lotta gear, and you gotta get the gear to the gig. Simple, right?

Wrong. What's simple is stuffing your mixer, P.A., instruments, mics and cabling in haphazardly, hoping that you won't have to strap any hardware leftovers onto the roof at the last minute. What's harder is trimming the load to perfection so that nothing gets broken on the way to the show.

A good packing job fits everything



in, doesn't damage anything in the load, and distributes the majority of the weight over the rear wheels. No packing job is worth anything if the contents get damaged in the process. Similarly, if you stop suddenly, you don't need a case turning into an unguided missile. Finally, distributing the weight sensibly lets the truck's suspension do its job, improving handling and making the van safer to drive.

PACK THE PAYLOAD

The van we're talking about is a small truck (Econoline, Chevy S10, etc.) equipped with two seats, and the remainder of the truck is cargo space.

Start by taking stock of what you've got to load. Speakers are the bulkiest item in any sound system. Fortunately, they're not the heaviest. On the other hand, power amplifiers, though small, pack a lot of weight. If you can, try packing the speakers on the extreme left and right of the cargo area, then fill the space between them with smaller items. Arrange the weight so that the heaviest items fall more or less over the rear wheels.

Pack to fill the space. This makes for a tight load that doesn't shift when you stop, or when you take a turn a bit fast. Aside from possible damage to your equipment, a shifting load can alter your van's balance during a turn.

Packing to fill the cargo space can be tricky, thanks to several interior irregularities that may not jibe with the shape and size of your sound reinforcement system. Wheel wells, for example, are a problem since they intrude into the cargo area. Minimize their effect by packing all the way up to the obstruction, then pack around it, using the space above them for mic stands, power cables, your snake, etc. The wheel wells keep everything from being crushed.

A good way to see if everything fits is to make a diagram of the van to scale on grid paper, and create correctly-sized cutouts of your gear. Mix and match sample arrangements until you have a game plan.

BOXING MATCH

One of the first decisions you're going to have to make is what to put in road cases. Sometimes, this can be a sticky problem. Take mic stands, for example. Moving microphone stands is easily my least favorite chore. Traditional ones (round iron base with chrome tube) are heavy and time consuming to move singly. Take them apart (put the tubes in one place, the bases in another) and now you have a box that requires a Sumo wrestler to move. Still, this is the most space efficient way to move this type of stand around.

Of course, you can always think ahead and buy gear that won't need this special treatment. Sticking with mic stands as an example, there are special tripod-based stands (KM, AKG, etc.) that cost a bit more money, but fold up into a nice, compact package. Indeed, they are lighter and easier to move as a result. When you pack up, be sure that all the hardware is tightened down. If not, the screws and other miniature hardware tend to get lost. Since they all use metric thread, replacements are difficult, but not impossible, to find. Pack them over the wheel wells.

Do you need road cases? They're expensive, but they save your equipment, pure and simple. If you load and unload your gear a lot, then road cases are an excellent investment. There are many different brands (Anvil, Calzone, Jan-Al, etc.). There are several grades of protection available, and you pay accordingly. Of course, you can always roll your own (see sidebar).

Ignoring the financial aspect, there's another angle to road cases: weight. They're heavy. They're also bulky. This means that you're going to give up perhaps a fourth of your cargo space to just holding the cases.

Speaker boxes are another thing you'll have to think about. If you're building new speaker enclosures, then use the width of the truck to determine the width or depth of the box. Don't forget the height, as you've still got to get the boxes in and out. If the boxes are too tall, then you've got to slide them in through the door then tip them up to a standing position. When pivoting a box on its corner, remember that the edge describes a circle whose radius is the diagonal measurement of the box plus a wee bit for clearance.

RACK IT UP

Equipment racks are another consideration, and they can save you or your roadies considerable setup time. The obvious savings is moving several pieces of gear at once. The less obvious savings is wiring time since you can leave the rack partially wired during transport.

There are two basic styles of rack rail (the metal rail that the equipment mounting screws attach to). One is a piece of sheet metal with a right-angle bend in it, and tapped holes. The other is a piece of heavy sheet metal with a right angle bend in it and punched but untapped holes in it. A spring-steel clip holds a nut, and slides over the hole. The advantage is that you have a nut-and-bolt fastening to the rail. In order to pull the bolt out, you've got to strip the threads in the nut...not at all easy.

SAFE & SOUND

Once you're boxed and racked, it's time to think seriously about sound system security. A truck loaded with tens of thousands of dollars worth of instruments and gear is a prime target — and a great hit — for the thieves of this world.

The windows in your truck are a mixed blessing. On the plus side, you can see out. On the minus side, others can see in. Sure, you can paint your windows out, but you lose a great deal of visibility in the process. One thing that really does work is window tinting. Tinting makes the windows difficult to see through from the outside, but really doesn't affect your ability to see outside while driving. The "tint" is a self-adhesive plastic film applied to the insides of your windows. You can damage the film by scraping the inside of the window, but otherwise it is quite rugged. Look in the Yellow Pages for a window tinter near you.

If you live in or will be traveling in urban areas, consider investing in a good alarm system. It helps to deter all but the most skilled rip-off artists. The only problem is that the things tend to go off periodically for no apparent reason, and if you're going to be mixing or playing during the gig,

make sure someone with free hands has the keys and knows how to deactivate the thing before it ruins your life.

Also, check with your insurance agent to see what will be covered in the event of an accident or theft. It may be necessary to purchase extra coverage. Don't lie and say you won't be transporting the gear from place to place — it's worth the extra money to be covered to the hilt.

MONEY-SAVING TIPS

Something near and dear to all of us is saving money. Here's a few tips for making your bucks stretch further:

■ Build your own cases. You can make simple plywood boxes, or ATA cases. You need to have access to a medium-well equipped woodshop to pull this off effectively.

■ Build your own racks. The plywood and steel racks described in the sidebar are cheap and rugged. They're also easy to build without a woodshop.

■ Use odd boxes. Plastic milk carriers (you know, the ones the milkman uses to deliver milk cartons in that are illegal to use) or their equivalent make great cable cases. Cut a piece of plywood for the bottom of the carton to keep everything inside.

Find the best way to pack and you'll make moving your gear easier, safer, and more efficient. You'll save your energy for where it really counts — performing on stage, that is.

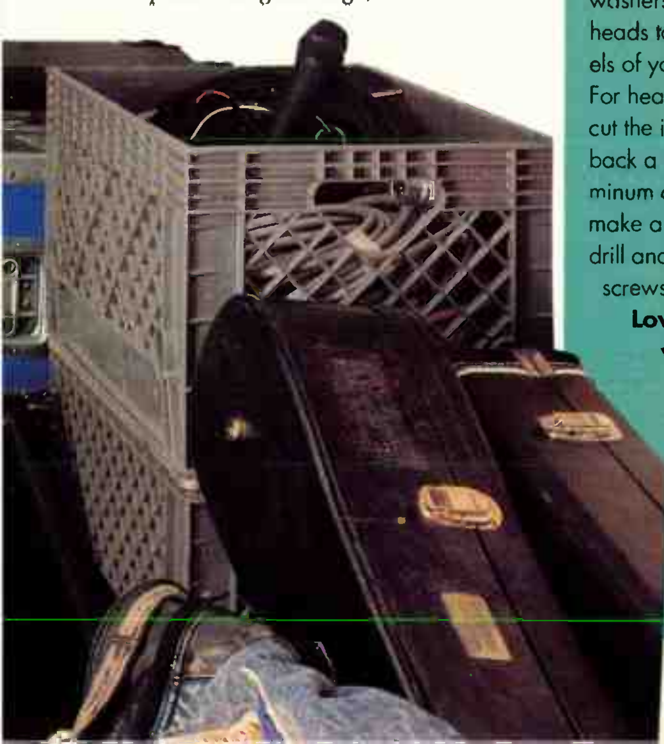


Photo courtesy of Yamaha

BUILD YOUR OWN CASES & RACKS

How do you build your own cases and racks?

Lowest cost: Plywood

You can build a durable, but relatively low cost rack from plywood.

Use 1/2-inch or 5/8-inch ACX plywood. The sides are double thickness and glued together. This helps make the hand-holes more comfortable for whoever gets to move the rack as well as strengthening the top/side joints. If the equipment isn't very heavy, you can simply screw it into the side pieces with long sheet metal screws. Use flat washers under the screw heads to protect the panels of your equipment. For heavier equipment, cut the inside side piece back a bit and use aluminum or steel angle to make a rack rail. Then drill and tap it for 10-32 screws.

ATA CASE: Handles (2 per side) and catches not shown. Allow 1/4" height and 1/2" width allowance. Be sure to compensate for width of rack rails.

Low cost: Plywood and Steel Angle

Power amplifiers, because of their weight, take a slightly different tack.

Use 5/8-inch or 3/4-inch ACX plywood. Have a metal shop make you a rectangular frame out of 3/4-inch steel angle. Weld all seams. Bolt the frame into the front of the box with 1/4-20 bolts. Put washers under the bolt heads to prevent splitting the wood under the bolt head. Be sure to put a lockwasher under the nut. Drill and tap the rail for 10-32 machine screws to match your equipment.

Best: Road Cases

If you're crafty, you can build your own ATA-style road cases. All of the parts used by commercial case makers are available for sale. Generally speaking, the suppliers have minimum orders, but you won't have any trouble meeting this requirement.

In highly-simplified form, here's what you have to do. You'll need the following parts: aluminum valence extrusion (for lids); aluminum edge extrusion (for edges); plywood; Kydex or ABS plastic sheet (for case sides); ball corners; angle braces; dished handles; dished latches; contact cement (3M water based); pop rivets with backup washers.

You'll also need the following tools: table saw; miter saw with metal-cutting blade; pop rivet gun; sabre (jig) saw.

Begin by purchasing the plywood. Cut off a piece of the valence extrusion and the plastic sheet used for the sides and take them with you to the lumber yard. Pick a thickness that, when combined with the thickness of the plastic sheet, fits snugly into the valence. Laminate the plastic sheet onto the plywood with the 3M contact cement.

Lay out the plywood pieces needed. Remember to consider the additional width/length created by adding the extrusion to the plywood. There are many similar-sized pieces needed and setting up cutting jigs will be worth your time. Cut wood. Measure and cut the aluminum extrusion using the miter saw.

Attach the extrusion to two of the sides. Insert the other two sides into the extrusion. Your box should be taking shape now. Now is the time to locate and make cutouts for the latches and handles. Rack cases generally need two handles per side for balance when being carried.

Rivet the latches into place using pop rivets and backup washers. Now rivet the corners into place. Finally rivet the valence to the case sides.

Your case should now be complete. If you haven't figured it out, it is a fair amount of work. If you don't charge for your time, then building your own ATA cases can be a big savings.

Suppliers: *Penn Fabrication, (201) 423-4404 or (805) 499-5932: Case Component Network, (714) 432-6630 or (800) 387-5122*


EPP▷5
EPP▷

THE LUNATIC ROOM

"I call architecture frozen music."
— Goëthe

Music, painting, architecture — it all comes out of your guts. The basic things, the primal things, are really the most important.

To me, that's what record producing is all about. A series of primal decisions made against a backdrop of inspiration.

That's not to say that every decision a producer makes is "monumental." What a producer does, as the builder of a recording, is make a series of yes/no decisions. Is this cymbal too bright? Yes. Should I use a Telecaster instead of a Stratocaster? No. You make all these little intuitive decisions that are like on/off switches, and you weave a fabric from them. At the end, you either have an album or a very peculiar tablecloth. The sum of those decisions develops a blueprint, assembles a framework, and culminates in an overall architectural design that adds another audio frozen yogurt store to the shopping mall of today's musical landscape.

That brings me to the notion that music making is visual, with colors and shapes, and that it fills the dimensions of a non-existent room. I always visualize what I hear in the recording studio. What we're doing at the console is creating a false image, whether we accomplish that through overdubs or through making the musician stand in a different spot in the room to achieve the perfect acoustic imaging. Then we mix with delays and other effects to invent space. In this sense, mixing is an incredibly visual experience.

In essence, you're creating imaginary walls that sounds are bouncing off of. When someone puts on headphones and falls into the middle of the concoction, hopefully you've built a room that's amazing for him to sit in. Oh yes, and while he's sprawled on the floor, you'd better provide him with some great songs performed by eloquent communicators. That pretty much sums up my mission as a record producer.

Besides the artist, it takes the right producer/engineer team to serve as the architect and contractor of the construction project, if you will. My dear friend Ed Cherney, for example, always builds a comfortable, intimate room to sit in. The room he built for Bonnie Raitt's "Nick of Time" was perfect. You just wanted to sit on a nice comfortable sofa and have her sing right at you. Michael Brauer builds rooms with incredible precision, using delays in an inspired and almost mystical way. His work is extremely geometrical. Then take Keith Cohen, who does a lot of remixing. I've worked with him on quite a few projects in the past. He manages to combine the slamming low end and roughness of street music with a pop music sheen that strokes the ears and complements the vocals.

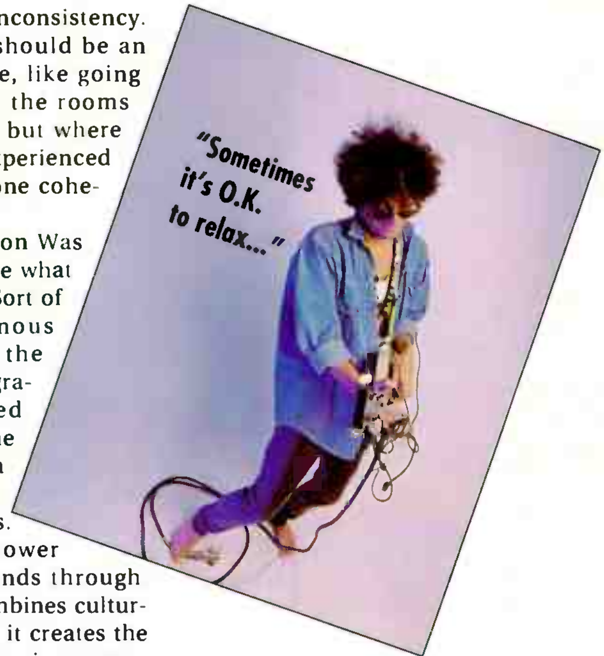
BY DON WAS

The producer should know the basic shape of the room he wants before he gets started. When my brother David and I were producing the Bob Dylan album, "Under a Red Sky," our idea was to evoke the feeling of Sun Records or Chess Records, but in a modern, rock and roll context. In this case, I visualized the room as being gray and primitive. Like Motown: sort of a concrete color — something like a garage with live walls. And I think the record carried that feeling from start to finish.

I had a similar room in mind with the Iggy Pop album, which appropriately is entitled "Brick by Brick." We made a grainy, black-and-white studio recording. In fact, after hearing monochrome for three months, I was initially startled to see the bright colors of the cover art.

THE LUNATIC ROOM

I like the consistency in my own Was (Not Was) work to come from controlled and calculated inconsistency. Listening to a record should be an entertaining experience, like going into a funhouse where the rooms don't quite fit together but where you come out having experienced the sensory jumble as one cohesive whole.



When I'm working on Was (Not Was), I want it to be what I call a "lunatic room." Sort of like Gaudi's famous Barcelona cathedral, the Sagrada Familia. The Sagrada Familia, or "Sacred Family" church mixes the baroque power of a Gothic cathedral with modernistic folk motifs. The church seems to tower and melt at once. It bends through time, and because it combines culturally dissonant elements, it creates the sensation that you're seeing something entirely new but totally familiar at the same time.

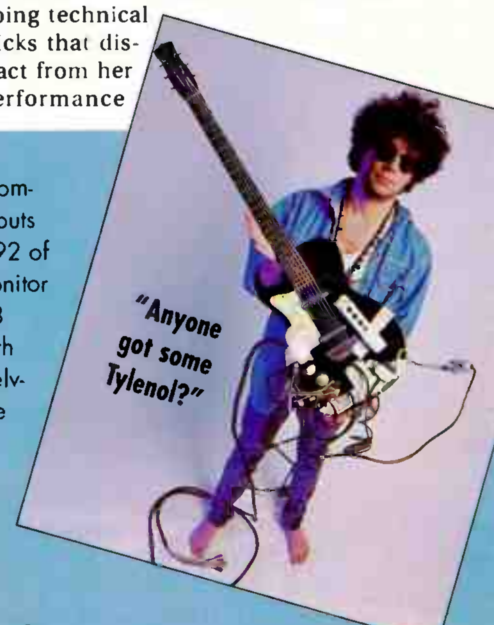
That's what I'm trying to accomplish in my recordings — a new feeling of controlled impossibility. Taking this particular approach is almost like working in animation as opposed to film; you have unconstrained control over the color, the texture, the temporal flow.

No matter what approach you prefer to take in your own recording, you have to have that overall structure in mind before you begin. That's not to say that you can't change course halfway through the recording (because you certainly will) but simply that it helps to keep sight of where you'd like to end up. Is it going to be the funhouse, the Louvre, or the 7-11?

I guess the next logical question is how much of a role technology plays in making all this architecture fit together — and how heavy a hand the producer should wield in the process.

TECHNICAL ANTI-MATTER

Sometimes it's O.K. to relax. Bonnie Raitt's album is a great example. I felt it was presumptuous for me to go in and "play the console"...the artist is the artist, and listeners buy Bonnie's work because they want to be intimate with her, and not with the producer. The work I did was almost anti-production, and we stripped it all away to make room for the emotional content of her vocals. Bonnie can consistently send chills through you — to start doing technical tricks that distract from her performance



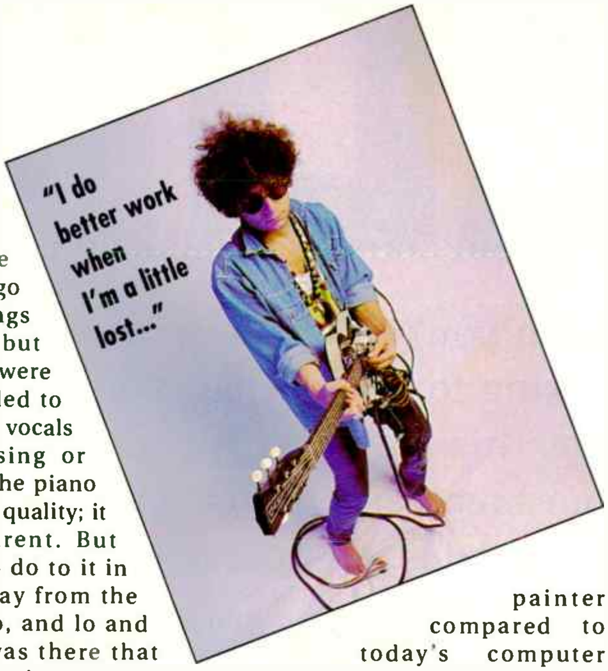
Was At Home

While Don Was is busy working the commercial studio circuit in Los Angeles, he is also in the process of assembling his own project studio in his multi-level home — an architecturally unique structure with a characteristically random array of floors and floor layouts.

The studio construction in progress overlooks a rear wooded area and is being designed for composition and songwriting instead of tracking and mixing. Built around the Peavey AMR Series 2400 console, the project room also features two Sony/MCI JH-24 analog multitracks, a Peavey DPM3 keyboard workstation and PMA 200 amplifier, and the new Peavey Audio Media Research PRM 308S nearfield studio monitors.

The Series 2400 is an input-abundant console designed for the MIDI composer, featuring 116 inputs in the mixdown mode, 92 of which have EQ. The monitor section has a total of 48 monitor inputs, each with active high- and low-shelving EQ, over and above the main 36 channels which have four band, sweepable EQ. Some engineers have said that the voicing on the four band sweepable EQ is reminiscent of the API is simply disrespectful.

Think of the "Nick of Time" cut. Here's a really powerful song. Almost everybody can relate to it. When we cut the track, we figured we'd go back and perfect things later on in overdubs, but the tracks themselves were so great that we decided to keep the live piano and vocals without any processing or overdubs at all. Sure, the piano didn't have a full tonal quality; it was kind of transparent. But everything we tried to do to it in overdubbing took away from the vocals. So we let it go, and lo and behold: everything was there that needed to be, even though some ears might reckon that there wasn't a tonal base to hang your hat on. That's what I call "deliberately not using the studio" as a medium.



THE VALUE OF IGNORANCE

If not using the studio as a medium is some great goal to achieve, then what is the point of all the high technology equipment in the first place? Besides making your music sound better and putting new sounds on your palette, I think one of its most valuable side effects is keeping you ignorant. Yes, I really mean that.

New equipment continually drops you on unfamiliar turf and gives you an uncertainty and naivete that I think comes across as something fresh in the music.

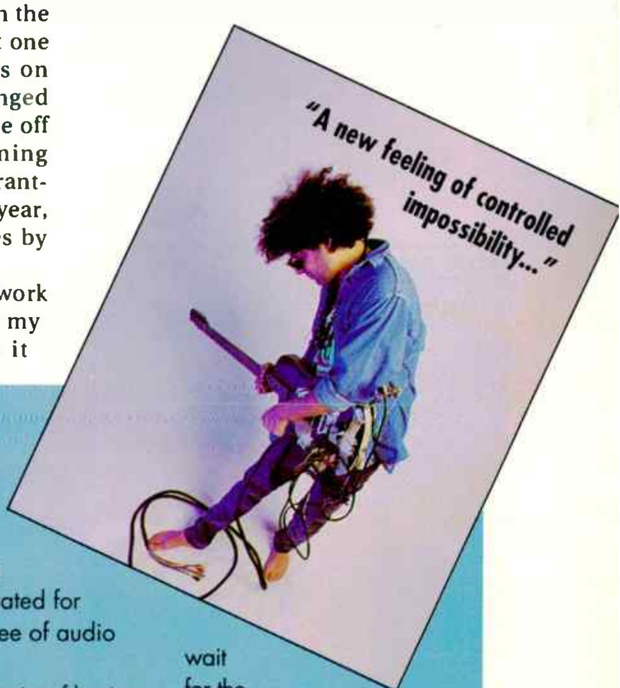
Take a moment to remember Brian Wilson's early work, from back in the days when we really didn't have all this crap to play with. In purely technological terms, he was like a cave painter compared to today's computer artist. But listening to his stuff is like walking into a cave and seeing Da Vinci's "Last Supper" on the wall. It was complex and inspired despite his relatively low-tech approach.

It's a great time to be making records today with all the science at our disposal, but I think I do better when I approach a recording project from a naive standpoint. I especially like to find a new piece of gear that I don't know how to use. I just mess around with it until I happen upon the right effect. For example, I bought one of the first Linn drum machines on the market. David and I just banged away on the thing with the volume off and our eyes closed, programming beats that were totally random. Granted, we sold very few records that year, but we were hailed as visionaries by two percussionists in Canada.

The point is that I do better work when I'm a little lost, when I park my car in a pool of quicksand, as it were. The first Was (Not Was) album, "Born to Laugh at Tornados," was made from total ignorance. I believe that Motown happened that way. They tried to copy Atlantic's New York R&B style of recording, but in the process they stumbled upon something far greater than what they were imitating. Dylan was attempting to capture a Gordon Lightfoot vibe when he cut "John Wesley Harding." It ended up sounding nothing like it, but the end result was something revolutionary.

For me the mystery is out of the recording process, which frankly makes it more difficult to come up with something totally unique. I know how to make an album, so I have to build traps for myself to keep the naivete and the ignorance. Technology provides that spark — it keeps giving me new spices to cook with. I just leap in, push a few buttons, and wake up on the surface of Alpha-Centauri. Anyone got some Tylenol?

Was, *as in Was (Not Was), is currently working with Paula Abdul. He has publicly expressed his interest in producing the real Milli Vanilli.*



550, except for its ±15 dB range and its quieter performance and better noise floor. The board also features a modular fader bay which offers users the option of adding fader automation (from J.L. Cooper to GML) or MIDI muting industry standard packages.

The dual-voicing PRM 308S monitors feature two, switchable discrete crossover networks, making them flexible for either mixdown or tracking. One EQ reviewer found that they had particularly good bass extension for such a small nearfield package. Was' project studio is powered by the Peavey AMR PMA 200 reference 100 watt per channel amp, with its patented soft-limited circuitry, which proves to be a valuable option for the high transient activity of the MIDI keyboard environment. When composing, users find it is useful to have the circuit engaged to avoid audible clipping or driver failures. When mixing, the circuit can be easily defeated for a higher degree of audio accuracy.

"The whole point of having a home studio is that you ought to be able to capture your inspiration when it hits," Was says. "The ability to work at your own pace, to keep the continuity...it's like surfing. You wait for the wave, and when you catch it, you ride it to the end. It's the greatest thrill. They don't make drugs that can make you feel that way."

—*Martin Porter*

Photos by Ed Colver

DEALING

When you're shopping for pro audio, you're looking for more than a box and a bargain. Hey, this isn't a VCR you're trading for. It's your bread and butter. You're looking for service, advice, compassion, expertise — call it all...HELP! So what should you do?

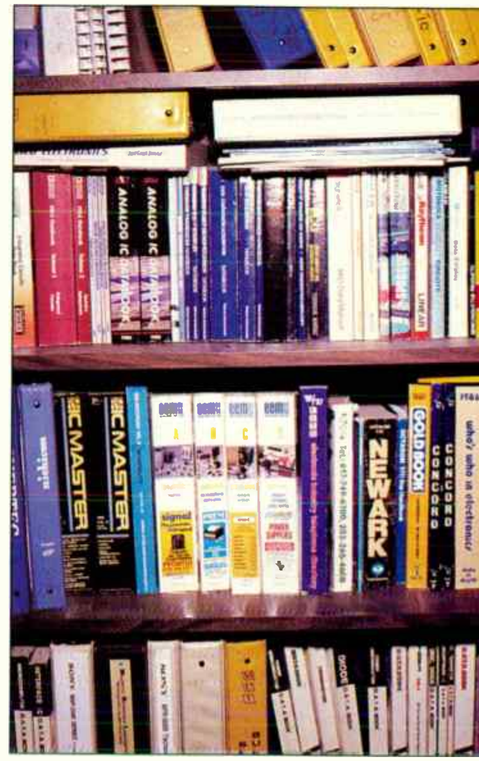
SHOP FOR A DEALER AS CAREFULLY AS YOU SHOP FOR YOUR PRO AUDIO GEAR.

J.T. WAY

■ Take a peek at the service department. If there's no dual trace oscilloscope (or better yet, an Audio Precision test set), they're not properly equipped. ■ Ask what the company's service policy is. How fast do they usually turn around repairs and how extensive is their parts inventory? Most importantly, find out if the dealer has a loaner program. ■ Expect to get a warranty of at least 90 days and up to a year on new equipment, and be sure to inquire whether or not the dealer can tag on an extra warranty plan of his own. ■ Ask for a copy of the customer list so you can find out how well your dealer has

YOUR

WITH



supported previous sales. Or just check around by word of mouth with studios in your area. ■ Look for a range of financing options. Better dealers offer a variety of leasing programs for you to choose from. ■ Judge your dealer by the questions he asks. He should ask about your application, experience and expected projects. If he's not asking questions, he's not trying to fit the equipment to your particular needs. ■ Locate a vendor who'll arrange private A/B demos — particularly between consoles. ■ Call the manufacturer to check if a dealer is factory-authorized. ■ Beware of gray goods — those pieces of gear designed to be sold in other countries but that have been re-outfitted for use in the U.S. Make sure the dealer is going to give you instructions and help when your exotic unit needs repair. ■ Look for a full-service dealer. Snoop around for major brands (e.g. Otari, E-V, Tascam, JBL, Peavey, Yamaha, etc.); they regulate their carriers and insure they're providing optimum service and support.

EQ would like to thank Audio Techniques, Sam Ash Music, Sweetwater Sound, Pyramid Audio, Peavey, and Electro-Voice for their input and suggestions.

Photos courtesy of Audio Techniques, New York, NY

DEALER

AUTOM

THE AGE OF

ATION

SMART CONSOLES GET AFFORDABLE

ALAN DI PERNA

Not too long ago, console automation was an amenity that ranked up there with catered meals, jacuzzis and private lounges. It was a luxury only to be found at major SSL or Neve studios. But like many other areas of computer-based technology, less powerful automation has now come home to the project studio.

The innovation that has largely made this possible is MIDI, which most of us associate more with keyboards than with studio equipment. Not all low-cost automation systems are MIDI-based. But MIDI does provide an ideal link between consoles and the computers used to automate them. It's a widely recognized protocol that syncs easily with SMPTE and works with lots of other gear in a typical project studio.

Otari's Series 34 with DISKMIX 3.

The market is currently awash with options for the studio owner who wants a moderately automated console for around $50,000 or less. Configurationwise, there are two basic choices. You can buy a console that has its own built-in automation facilities. Or you can purchase an add-on system that will impart automation capabilities to just about any console. Either way, it's important to understand the different types of automation functions these systems can perform.

At the most basic level, there's snapshot automation, which is typically used for automating mutes. The term comes from the old studio trick of taking a photograph of the mixing board at the end of a session as an aid in recalling console settings at later sessions. In snapshot automation, though, computer memory takes the place of camera film.

A range of parameters — say the mute status of every channel — is stored as a scene. Scenes can then be recalled and changed via MIDI program change commands, which can be recorded onto any MIDI sequencer. When the sequencer plays back — synchronized to tape via SMPTE or some other sync code — the snapshot scenes will change at appropriate points in the program material.

The Soundtracs PC MIDI Series is a range of consoles with built-in snapshot capabilities, while the DDA MUSIC (Mute and Solo Interface Computer) is an add-on device that brings snapshot automation to DDA's DMR 12 console, although it can be used with other boards as well. In addition to changing scenes via program change commands, the MUSIC system can also do real time muting, unmuting and soloing on individual channels and lets you assign channels to any of 10 mute groups.

PC-based automation systems often include cue-list-style, off-line editing, punch in/out, snapshot memory, MIDI sends and more. In other words, they can be used as the basis of a total studio automation setup.

At the next level of sophistication are systems that provide real-time automation of channel levels as well as mute. VCAs (Voltage Controlled Amplifiers) are most commonly used to achieve this task in automation systems that fall in the middle price range. Although VCAs can cause some signal degradation, high-quality VCAs from companies like dbx, VCA Associates, Aphex and B&B Audio produce widely-admired results.

THE VCA WAY

VCA-based systems consist of three basic parts. First, there are the VCAs themselves, which are either hard-wired into the console's signal path or plugged in via the board's insert points. Secondly, there's a device that stores control data for the VCAs. It's the part of the system that syncs to tape and tells the VCAs when and how much to raise or lower the signal level. A good many systems use popular personal computers, like the Macintosh, IBM PC or Atari ST, for this job. And last, there's a device for inputting mix moves to the storage device. Ideally, this will be the console faders themselves, but on more affordable automation rigs, an external fader module or a computer mouse are used to input data.

There are quite a few VCA-based add-on automation systems on the market, including, J.L. Cooper's MAGI II, MegaMix from Musically Intelligent Devices, Steinberg's Mimix and the Jellinghaus C-Mix system. All of these automation packages are personal-computer-based and use MIDI as the connecting protocol between the computer and the VCA units.

Additionally, there's the DISKMIX 3 system from Otari, which is based on the IBM-AT, but uses its own communications protocol instead of SMPTE. The Optifile IIID is another IBM-based system that's currently available in Europe and soon to be distributed in the U.S.

With add-on automation systems, you typically start with a 16-channel "starter" package. The average for this is around $6,000, although the MegaMix starter system is even more affordable ($2,595 for a 16-channel system and $1,095 for an eight channel configuration), and DISKMIX 3 is significantly more expensive (about $25,000 for a starter VCA setup). Systems are expanded by buying extra VCA "packs." Many add-on rigs

J.L. Cooper's MAGI II

give you the option of installing the VCAs right in the console rather than having an external rackmount VCA box, which can be prey to unwanted noise and interference. Also, many systems provide an option that lets you switch the VCAs out of the signal path when you're not using the automation.

TO THE BOARDS

Many of the above-mentioned add-on automation systems also appear as factory-installed, OEM options on many popular mixing consoles. In cases like this the only "extra" you need to buy is the personal computer needed to run the automation software. You can buy an Allen and Heath Sigma, Sabre or Spectrum console, or a Biamp Legend board with a factory-installed Magi II system. CAD has recently begun offering MegaMix in their Maxcon range of consoles. DISKMIX 3 comes as an option in Otari's Sound Workshop consoles. And the Soundcraft 6000 board has an optional automation system that's akin to Steinberg's Mimix. Both systems are based on an earlier automation package called Twister. In the lower ($33,000 - $55,000) end of Amek's price range, the company installs the Jellinghaus system as an option on their TAC and Magnum series.

The major strength of PC-based automation systems is their editing capabilities. These typically include cue-list-style, off-line editing and the ability to punch in and out on fader moves. Most systems provide snapshot memory as well as real-time automation. And most can send MIDI note commands, program changes and/or continuous controller messages out to external MIDI gear, such as signal processors, synths and samplers. In other words, they can be used as the basis of a total studio automation setup.

COMPANY CONTACTS

Manufacturers	*Phone*	*Lit #*
Allen and Heath	(203) 795-3594	135
Amek	(818) 508-9788	136
Aphex	(818) 767-2929	137
Audio Kinetics	(44-081) 953-8118	138
CAD	(216) 593-1111	139
dbx	(415) 351-3500	140
DDA	(516) 249-3660	141
J.L. Cooper	(213) 306-4131	142
Niche Audio	(818) 993-4091	143
Otari	(415) 341-5900	144
Soundcraft	(714) 556-6191	145
Soundmaster	(805) 494-4545	146
Soundtracs	(516) 932-3810	147
Tascam	(213) 726-0303	148
Uptown	(303) 443-1171	149
Yamaha	(714) 522-9353	150

But what if personal computers just plain give you the willies? Is there any way to get around having to buy a PC? Yes. The breakthrough Tascam M3700 is an under $15,000 automated console that prides itself on being completely self-contained, with an onboard computer, SMPTE reader/writer and disk drive which can store up to six real-time mixes on a disk, or up to 63 files of 99 snapshots each. And moving upmarket ($55,000 and up) Soundcraft's TS-12 has its own integral computer, disk drive and frame automation system.

Tascam M3700

AND THEN AGAIN...

VCA's and MIDI snapshots aren't the only ways to automate affordably, though. Consider that for $479, the Niche Audio Control Module (ACM) gives you an eight-channel unit that uses a large-scale integrated resistor network rather than VCAs to do level changes. Via MIDI, level changes can be recorded to and played back from any MIDI sequencer.

And of course, moving fader automation has long been the high-end way to give VCAs the slip. DISKMIX 3 comes in a moving fader version as well as a VCA version. Starting at $38,000, DISKMIX 3/Moving Faders is hardly a budget item, but it is among the most affordable moving fader packages out there. Meanwhile, Audio Kinetics in the U.K. has bowed a system not unlike DISKMIX 3, MotorMix — a moving fader option that can be substituted for an AK2 VCA fader, allowing existing users to upgrade. It's only available in the U.S. through S.G. Audio in Chicago. Similarly, Audiomation from Uptown Automation Systems is an upscale way to add up to 96 moving faders to most consoles without any big internal modifications.

Fully digital automated mixing boards are the other non-VCA option. All level changes and other signal manipulations are done in the digital domain, so there's no need to mess with VCAs. To date, Yamaha is the only company to offer fully digital mixers on the affordable end of the market. The first was the DMP7, a $4,235, eight-channel unit that could be cascaded with other DMP7s for more channels. One of the great strengths of a digital device like the DMP7 is that all mix parameters can be automated in real time. Not just mutes and levels, but EQ moves, panning, buss assignments...everything. And because the DMP7 has three onboard effects processors, that includes effects automation as well. All of these moves can be recorded to any MIDI sequencer and then played back in sync with tape. Recently, Yamaha upped the ante with the DMC1000, a 22-input, eight-buss fully digital console that sells for about $32,000 and has a built-in sequencer and synchronizer for automation.

In today's recording environment, automation is rapidly becoming a necessity rather than a luxury. Luckily, affordable automation for the project studio is an idea whose time has also arrived...just in time.

Build a MIDI Output Switcher

Clean up your rack by assembling your own MIDI switcher.

CRAIG ANDERTON

YOUR MIDI STUDIO is really getting together, and these days you're driving a rack of nifty MIDI-controlled signal processing and tone generating gear with your sequencer. Clever person that you are, you've even installed a MIDI thru box in the rack frame so you need send only one MIDI out cable from the sequencer to the rack, with the thru box distributing the MIDI signal to the various MIDI inputs.

But now you've just finished a demanding session where you came up with a lot of custom patches for your rack gear. Since you know from past experience never to trust anything with a microprocessor in it, it's time to send all that patch data through the various MIDI outs and to a system-exclusive storage device, such as a computer running librarian software or a suitable sequencer or keyboard. These can save the data on disk for a more permanent form of storage than just sitting in battery-backed RAM.

However, patching to the appropriate MIDI output may be easier said than done. Unless you've invested in a MIDI switcher (with appropriate cabling), you've probably found yourself crawling around the back of a rack looking for the right MIDI out connector and patching into each unit, one at a time. Low-tech? You bet.

There is an easier solution. It takes only about $30 of disposable income, some electronic construction chops, and an evening of your time to build a custom output switcher that can really help civilize your rack.

It takes only about $30 of disposable income, some electronic construction chops, and an evening of your time to build a custom output switcher that can really help civilize your rack.

HOW TO DO IT

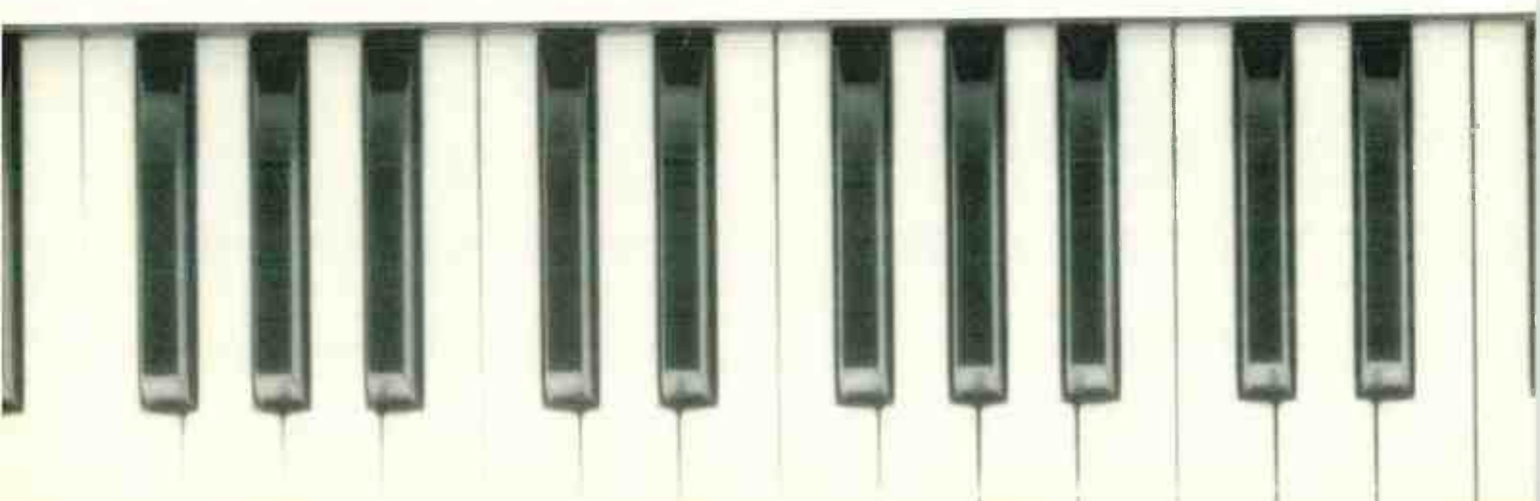
I'd recommend building the MIDI Output Switcher in a single-space rack panel. The front panel has a master MIDI output jack that connects to a 6-position rotary switch. This switch selects between six different cables that plug into six different MIDI outs. Turning the switch's knob selects which output gets routed to the master MIDI out. There are no power supply considerations, since this is a passive circuit.

Don't worry about finding parts — they're all commonly available. The parts list on page 55 indicates Radio Shack part numbers for most, as well as list prices from their 1991 catalog. For rack panels, check the electronic parts supply houses located in major metropolitan areas.

When it's time to start constructing, refer to the schematic. The only "active" pins in a MIDI connector are pins 4 and 5; pin 2 is used for shielding. The MIDI plugs (each of which plugs into a particular MIDI output in your rack) connect via two-conductor shielded cables to switch S1. The shield for each cable should connect to pin 2 of the related plug, to prevent ground loops, but the other end of the shield should not connect anywhere.

In the schematic, these plugs are viewed from the rear (i.e., where you do the soldering). Note that all pin 5 connections from the plugs connect to S1A (one-half of the two-pole rotary switch) and all pin 4 connections terminate at S1B, the other half of the switch.

The master MIDI out jack is viewed from the rear (again, where you do the soldering). In case there's any confusion, note that the plugs and jacks should have tiny pin numbers embossed somewhere on the connector. The main point is to make sure that pin 5 on the master out jack switches to pin 5 on the plugs, and pin 4 on the master out jack switches to pin 4 on the plugs. After construction, use a continuity tester or ohmmeter to

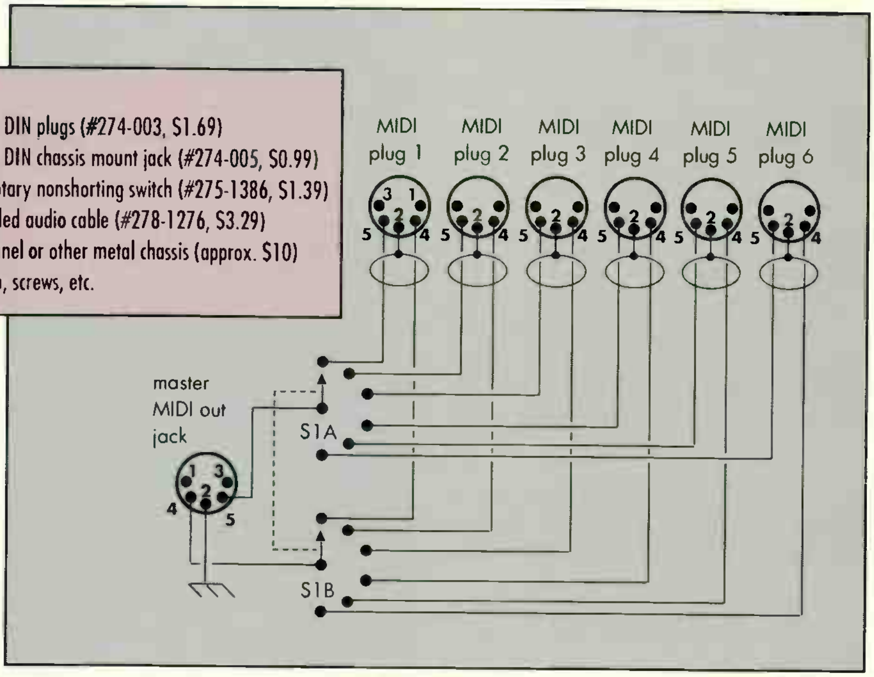


make sure this is the case.

Note that in keeping with the MIDI spec, pin 2 of the master MIDI out should connect to a chassis ground so that cables plugged into this jack will be properly shielded.

Parts List

6	5-pin 180-degree DIN plugs (#274-003, $1.69)
1	5-pin 180-degree DIN chassis mount jack (#274-005, $0.99)
1	2 pole, 6 throw rotary nonshorting switch (#275-1386, $1.39)
1 roll	2-conductor shielded audio cable (#278-1276, $3.29)
1	1U rack mount panel or other metal chassis (approx. $10)
Misc.	Wire, solder, knob, screws, etc.



MODIFICATIONS

Two-pole rotary switches are available in up to 12-throw configurations if you need to switch between more than six MIDI plugs. Also, note that this circuit can serve as an input switcher. The plugs could connect to various keyboard outputs, with the switch connecting a particular keyboard out to the master output. Finally, you can save construction time if instead of wiring up individual cables to each plug from the switch, you simply cut three MIDI cables in half, strip a few wires, and solder them to the rotary switch terminals.

And that's all there is to it. Don't fumble with the wires behind your rack. If you'd rather switch than fight, warm up that soldering iron and start building!

Craig Anderton *is the author of several books on musical electronics, including* Electronic Projects for Musicians. *His latest album,* Forward Motion, *is on the Sona Gaia label (distributed by MCA).*

Are You Remix Ready?

Photo courtesy of SSL Black Book.

Before you fork over for a five-figure remix, make sure your roughs are ready.

KEITH COHEN

PROJECT AND PLATINUM studios are now working closer together than ever before. For the most part, I have the pleasure of remixing tapes that are perfectly prepped and nearly finished except for that critical final mixdown, requiring an experienced mixer and a fully automated console.

However, this emerging studio relationship demands that the basics of prepping tapes be closely followed. If you want a kick-ass remix, you have to prep your tapes properly so the engineer can concentrate on the mix and not on asking a lot of questions.

■ Be thorough. Notate your track sheets as well as possible so the mixing session doesn't turn out to be a guessing game.

■ Print tones. Give the mixer a reference to work by — print 20- to 30-seconds of 1 kHz, 10 kHz, 15 kHz, 100 Hz, and 50 Hz tones each on the tape.

■ Make sure to align your tape machines. This will assure that tapes will play back as well as they were recorded. And make sure to have a technician come in for a routine maintenance check.

■ Think clicks. Tapes with drum machine tracks, in particular, should be printed with a click track or sync track from a sequencer. Without a click track, it's difficult for the mixer to sync tracks.

■ Composite your vocals. If you run out of tracks, don't just erase the click track. Composite your lead vocal tracks first, since figuring out which parts of which vocal tracks you want used can get really hairy anyway.

■ Watch out for track debris. It's understood that you want that guitar or sax solo just right, but always clean up the unwanted tracks underneath. In cases where you have a lot of extra material, proper track notation is a lifesaver. And if you can't decide which parts you want used, you may have to sit in on the session or — gasp— leave it up to the engineer.

■ Leave the EQ up to the mixer — record tracks flat. Bringing up the high end or making some other EQ adjustments may seem smart, but it limits the mixer's adjustment options.

■ When recording vocals in particular, watch your levels closely. Pinning will cause distortion that's almost impossible to repair. The best a mixer can do is hide the distortion and we all know how horrible that can sound.

■ Give 'em some breathing room. Allow enough pre-roll (as much as 20 seconds if possible) for the tape machines to lock up. And when you're locking up two 24-track machines, make sure that the SMPTE timecode is properly striped and that all the tracks play back correctly.

Finally, if you're as careful about getting your masters ready for a mixdown as you are about laying down your tracks, you will be saving us mixing engineers a lot of time and energy. This will eventually translate into a balanced budget as well as a better quality mix. Keep in mind that even though the master leaves your hands, the final mixdown is really as much your responsibility as it is that of the mixing engineer.

***Keith Cohen**, producer, engineer, and remixer, has worked with artists including Paula Abdul, Prince and Elton John. He's often found at L.A.'s Larrabee Sound.*

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CIRCLE 31 ON FREE INFO CARD

The Wild and The Wireless

Miles Davis with RAMSA's WM-S2 customized for wireless transmission.

Read these miking tips and cut the cord — without losing your cool.

RICHARD J. GRULA

UNPLUGGING YOUR AX during a solo, yanking an amp off a speaker cabinet, tripping over a cable and landing flat on your faces — these Kodak moments might be amusing to sadistic audience members, but they're no fun for the band on stage. In fact, they can be dangerous (ever been hit with a falling Marshall stack?). The obvious answer is wireless, but many users still don't know how to maximize performance from their wireless mic systems. Here are some ideas:

1 Buy fresh batteries by the box. Change them before every performance. Your power-hungry transmitter will drain a nine-volt battery in four to eight hours. Dying batteries weaken transmission strength, which permits RF interference to crowd out your signal. Crap sound is the end result. Most wireless problems stem from weak batteries.

2 Place the receiver properly. If your receiver has built-in antennas, place it in line-of-sight with the transmitter and as close as possible to the action. For vocalists, this means on the side of the stage by the monitor board or actually on stage, near the snake's multi-box (not in the back of the room on the house mix board). Guitarists should put the receiver in their racks or on top of their amps. At the foot of the stage, near your stompboxes is O.K., but you risk theft or accidental destruction from stepping on it. Avoid receiver placement near metal structures (i.e. metal ceiling supports), which can cause reception problems. Dual antennas should be set at 45 degree angles.

3 Separate the two remote antennas of a diversity system by at least 10 feet for optimum reception. This allows each antenna to "see" different sets of signals from the transmitter (placing antennas on opposite sides of the stage doesn't improve reception any more than 10-foot separation). Also, vary the height of the antennas by a foot or two. This prevents out-of-phase dropouts caused when two antennas see the exact same set of signals.

4 Splurge on cables. The antenna cable supplied by manufacturers is O.K., but for best results, replace it with RG8-grade cable (about 50 cents per foot). Yeah, it's heavy, ugly and a bitch to work with, but it will cut down RF interference and improve quality.

5 Check transmission frequencies. Be sure no one else in your band or on a nearby stage is using a wireless with the same transmission frequency. A transmission frequency is a number such as "183.56 MHz" — not "Channel 3." Transmission frequencies are usually printed on the receiver, the transmitter and the original packing box.

6 Do a sound check. Walk all around the performing area to discover any dead spots. Mark those dead spots with reflective tape, and then stay away.

7 Don't trust experience. It may have sounded great last night, but each stage offers different RF problems. A system which was dead-quiet last night might suddenly pick up a billion pops and buzzes during tonight's sound check. Traveling bands should look into wireless systems with switchable frequencies or carry a few different wireless setups.

8 Think metal detector. The RF from your transmitter can be affected by pieces of metal coming in and out of contact with each other on your instrument or anywhere on stage. The result is annoying clicks, pops and scrapes. To prevent this, solder a 220 pf capacitor across the hot and ground leads of your guitar's volume and tone pots, use a glass slide instead of a metal slide and bridge rattling metal anywhere on stage with a small piece of wire (this might include case handles, loose bits of the floor or ceiling and even your drummer's kit).

9 Adjust the transmitter. Adjust a wireless system's gain control for the individual user. This is a three-man operation — the vocalist sings as loud as he can while the sound man listens from the house and tells a tech with a screwdriver how to adjust the transmitter. Badly adjusted transmitters will produce either distorted sound (from being set too low) or a noisy signal (from being set too high). Guitarists should have a separate transmitter for each instrument. If not, adjust one generally for all and pray.

10 Always power down. Never turn off the transmitter without turning off the receiver. If you do, the receiver will look for a signal that's not there and often lock into ear-shattering VHF white noise. Use the transmitter Mute switch, which cuts your audio and still sends an RF signal to the receiver.

11 Skip the squelch. This is a useless carryover from CB and ham radio. It's supposed to cut noise when the transmitter is turned off — but if you follow the above rule, you don't need it. It also decreases the operational range of a wireless. Just leave it adjusted to minimum and forget about it.

12 Think mic technique. Wireless microphones without built-in antennas coming out the end use the mic handle itself as a two-piece antenna (a plastic or rubber band on the mic indicates the antenna division). Sweaty vocalists who wrap their hands around the entire mic will cover both sections, drastically reducing the antenna function and causing dropouts. Hold the mic above or below the dividing line — never hold both parts.

13 Follow the "more than four" rule. Inexpensive wireless systems are fine, but if you're going to use more than four on one stage, you'll need the better filtering provided by mid- or upper-grade systems. The reason is that although transmitters use one frequency, they also beam out partials on other frequencies (akin to the harmonics of a musical note). When you use more than four wireless systems, these partials are picked up by other receivers. Better systems have filters which control partials.

14 Bring backup systems. On stage, during a performance, is not the place for troubleshooting. If you can, carry an extra transmitter. They tend to be the component that breaks most often. You should also have a cable or hardwired mic nearby. For guitarists, one trick is to use a standard 20-foot cable to connect the receiver to the amp or effects. If something goes wrong, it takes about three seconds to unplug the transmitter and the receiver and plug the cable into your guitar.

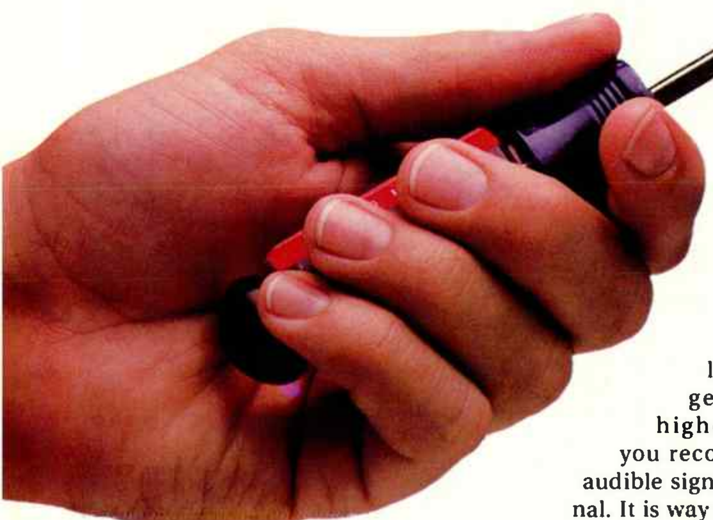
15 Consider wireless monitors. Does your sax player complain he can't hear? A wireless monitor uses a belt-pack receiver and special invisible headphones custom-made by an audiologist. Your hard-of-hearing bandmate gets his own mix in his ears and everybody else can get down to the business of playing.

How To Get Level Headed

The continuing saga of puttin' on the tweak: bias and record alignment.

RICHIE MOORE, PH.D.

NOW THAT YOU'VE had a couple of months to mull over the reproduce alignment of your tape recorder, it's time to deal with the critical record alignment. The reproduce alignment is important because it sets up the basis for all the levels on the tape yet to come. Proper reproduce alignment can play back a tape accurately, but the record alignment is what makes a tape precisely reproduce the signal put into the machine. You knew that, didn't you?



For this part of tweaking your machine, you will need a couple of tools. First is a sine wave signal generator capable of a stable signal at 1 kHz, 10 kHz and 100 Hz. It would be good if the generator had a full sweep available from 20 Hz to 20 kHz for proof-of-performance checks.

TEAC makes a small generator that plugs into a 1/4-inch jack, but the most general purpose device for the studio is the Loftech TS-1 test set. This is a great generator, frequency counter, and a dBm meter all rolled into one small box for about $340. It is really invaluable in any size studio. Of course, most consoles have built-in oscillators for tone generating.

The only other tool required is a small tweaker to adjust the trimmers (pots) in the machine. This can be a small flathead screwdriver, but better yet, there are tweakers made by Spectrol that have a small flathead at one end and a recessed flathead at the other for centering over trimmers. With these tools in hand, we are ready to align.

SETTING THE BIAS

The very first thing to do in the record alignment is set the record bias. Bias is a high frequency (roughly 180 kHz to 240 kHz — generally 10 times the highest audible frequency you record) that mixes with the audible signal and linearizes the signal. It is way too much rocket science to go into here. Briefly, without bias you have a very noisy and distorted signal; with bias applied in the proper amount you have a good clean signal.

> The proper bias setting for a machine corresponds to the point where third harmonic distortion and modulation noise are minimized. If all this sounds crazy, it really is.

Bias has absolutely nothing to do with the playback of magnetic tape; it is only a function of recording. We set a tape machine's bias in RECORD and REPRO on blank tape. You have to be able to see the effect of bias on the signal.

Bias depends on tape speed and head-gap width, and can vary slightly for different kinds of tape. Narrower head gaps require more bias than wider gaps. Although finding the maximum sensitivity while in RECORD at 1 kHz can work well, bias is usually set using a short wavelength (i.e., a 10 kHz sine wave signal).

It is also customary to set the bias for a certain amount of "overbias" so that a reduction in the sensitivity occurs. The amount of overbias required is dependent on the gap width of the record head. Most models of tape machines in use today use 0.25 mil record head gap length. Most Ampex tape machines and some older Otari's use 0.50 mil head gaps. The proper bias setting for a machine corresponds to the point where third harmonic distortion and modulation noise are minimized.

If all this sounds crazy, it really is. You should ask your dealer or call the factory for the overbias for each speed and tape that your machine will use. It usually is in the manual that comes with the machine.

A word of caution: machines with two heads have to set bias by trial and error and should only be done by a qualified technician.

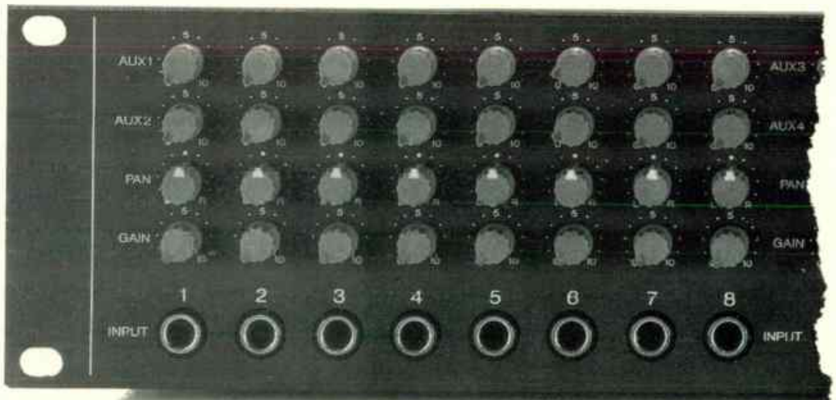
RECORD ALIGNMENT

Now that we have a complete knowledge of bias (sure), we can start the record alignment procedure. Follow along with these simple steps:

(1) Hook up the test signal generator (osc.) to all channels of the tape machine (you could do one at a time if you have a lot of patience). Put the machine in INPUT. Verify that all the channels read 0VU. If not, adjust the INPUT TRIM. This really should have been done prior to reproduce alignment.

(2) Set the oscillator to 10 kHz at -3VU. Put blank tape on the machine. Put the machine into RECORD and

continued on page 71



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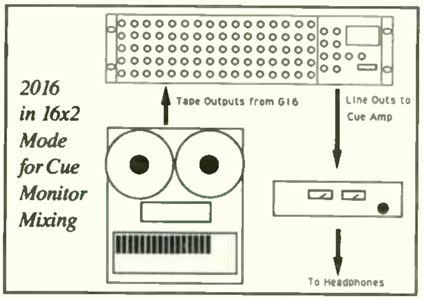
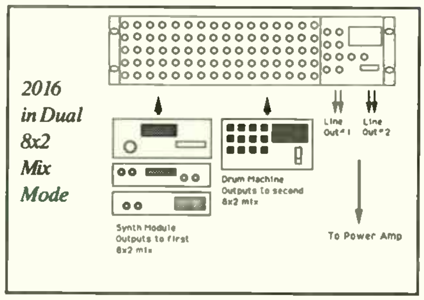


Sometimes it's a perfectly straight-forward 16 x 2 submixer with Gain, Pan and two Effects Sends on each input.

Then it turns around and becomes two independent 8 x 2 sub-mixers, each with a stereo Effects Return and Bus Inputs. That's when it acts like an active patch bay and eliminates a bunch of patching hassles.

The 2016 handles keyboards and drum machines with equal ability and it processes almost any line source with unequaled flexibility (like front panel insert points and foldbacks on all inputs).

It is absolutely eager to please sophisticated MIDI users and synth programmers, yet it feels right at home in a tight recording rig or on the road in a PA rack--great for extra monitor feeds (the sonic equivalent of duct tape!).



In fact, the 2016 is so versatile, so quietly helpful in so many different situations, no wonder it's always wondering what it's going to be next.

P.S.
In its own schizo way, the 2016 thinks it ought to cost a lot more than the mere $400* price tag it carries.
(Maybe you're crazy if you don't buy one.)

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CIRCLE 18 ON FREE INFO CARD

The Total Jingle Studio

How to put together a fully-loaded jingle studio for $35,000 or less.

J.D. SHARP

OUR MISSION: CREATE a jingle studio that's ideal for production of ad music beds. Budget: $35,000 or less. Is it possible? Yes.

We'll need a video playback system that integrates into an audio-for-video synchronization system. We'll be MIDI- and computer-based, so we've gotta have both software and sound generation hardware. Our console should be loaded with inputs to handle both virtual tracks as well as a multitrack recorder. Our mixdown format needs to be useful to video houses, since they'll lay back our music bed to their video master. To round out our recording we'll need effects, signal processing and microphones.

With these considerations in mind, it's time to put together our package. The console and tape recorder are at the heart of it all. The Tascam M3700 console meets all our criteria; it's priced moderately, provides 32 dual inputs plus four effects returns for a total of 68 mix-times line in's, and best of all, it's automated. The automation covers the main input faders, mutes for main and aux inputs, sends and returns and even EQ Bypass. All changes are transmitted and can be received over MIDI. A sequencer program can record all the changes; multiple mixes can then be "cut" and "pasted" together for a final mix. This indispensable capability has previously cost thousands more. For ease in working we'd add on the optional full meter bridge (MU-3532).

The tape recorder should be half-inch 16-track for budgetary reasons; their performance, boosted by on-board noise reduction, is more than adequate. Because of the convenience of a built-in synchronization option, we've elected to use the Fostex G-16, with the 8330 Synchronizer Card installed. This not only takes care of our lockup with our video deck, but also neatly ties in with the sequencer, providing tape sync. We require a Fostex 8755 video interface to the JVC BRS-610 half-inch video deck; this advanced VHS unit has everything we need to play back a working dub of the video master and it provides a timecode dub of this type.

On the software front, those who favor the Mac could do equally well with either Performer from

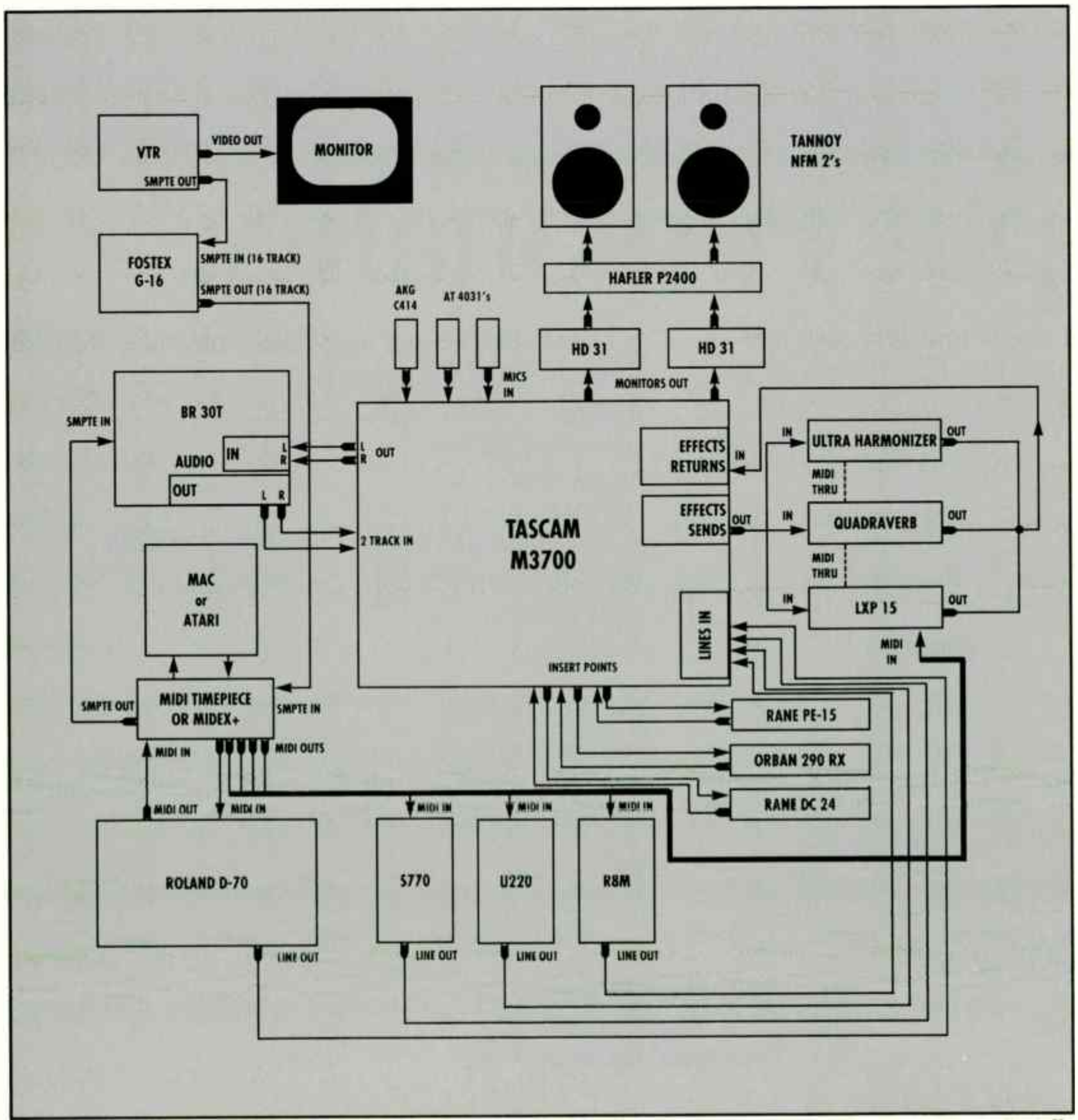

Diagram by Mark Alhadeff

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The "sound" you were looking for, but could not find! "Punch, slap, fat, pump, or squeeze" ... now you can get it *fast* ... get it *clean* ... and get it at *low cost* with the Aphex Expressor. It gives you greater flexibility to express yourself without the sonic degradation associated with other compressor/limiters.

The exclusive High Frequency Expander* allows higher ratios without the "dullness" of other wideband devices. Spectral Phase Refractor™ restores bass clarity and punch without any increase in amplitude.

*Patent Pending

The Expressor works with any system, balanced or unbalanced because of its servo-balanced input and output. It's loaded with other features including adjustable input, threshold, attack, release, output and ratio, plus link, slave and hard or soft knee compression. And, like all Aphex products, the Expressor features the highest quality signal path.

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CIRCLE 41 ON FREE INFO CARD

All Aphex products are designed and manufactured in the U.S.A.

Mark of the Unicorn or Vision by Opcode. Since we're going to load up with MIDI, let's try Performer combined with Unicorn's MIDI Time Piece. For the Atari, we've found nothing finer than CuBase 2.0 from Steinberg/Jones, combined with their MIDEX+ interface. The interface provides four MIDI outputs for 64 channels, and SMPTE read/write.

Ideally we'd use DAT for mixdown, but only a handful of production facilities use the synchronizable Fostex D-20. So we'll go for half-track, with a center track carrying SMPTE. Tascam's inexpensive BR30T neatly fits the bill. It's a basic 2-track with both 15 and 7.5 ips speeds, and built-in editing features (even down to an integral splicing block.) It also offers future synchronization if needed.

Now it's time to add the effects, and we even have some money left. The Ultra Harmonizer from Eventide offers a dual pitch changer useful for everything from time compression to correcting pitches that are as much as a quarter-tone off. Meanwhile, Lexicon's LXP-15 can combine up to five effects at once and also includes pitch change and complete MIDI control over every adjustable parameter. An Alesis Quadraverb Plus can handle just about anything you can throw at it, and is very reasonably priced. For an audio enhancer, the 290RX from Orban is ideal. It combines single-ended and sliding-filters noise reduction with both harmonic restoration enhancement and multiband dynamic enhancement, basically doing the work of three separate units. Even though our console has very powerful EQ, it's nice to be able to patch in a parametric for thorny problems. A Rane PE-15 will neatly address this need. There are a host of great compressors; we're particularly impressed with Rane's DC-24 Stereo Dynamic Controller, which includes a crossover so that the two channels can be used to "split process" a signal. The DC-24's noise gate will also come in handy. A pair of A.R.T. HD-31 High Definition 1/3rd octave equalizers will serve as room equalizers.

In synths, let's pick up a Roland D-70; it has plenty of firepower as a 76-note controller and offers a rich synthesizer — and you don't have to pay for a redundant sequencer. Roland's U-220 comes in handy, with 30 voices and the ability to handle up to six parts PLUS a rhythm part. To spice up our percussive arsenal a Roland R-8M module is in order. There are many strong alternative modules; Korg's M1REX, Yamaha's TG-77 and both the Proteus/1 and Proteus/2 from E-mu could easily be substituted for any of our selections. Finally, no MIDI studio could truly be complete without a sampler. It's a tough choice, with strong entries from Roland, Akai and E-mu. The Roland S-770 is our favorite, with the recent introduction of an extensive sound library available on Syquest 45MB removable cartridges. The S-770 has the ability to drive a monitor with its on-board sample editing software.

An AKG C414 ULS takes care of our need for a large-diaphragm vocal microphone. This we'd supplement with a pair of Audio Technica's AT-4031's, a small-element condenser mic that combines the toughness of an electret with the sensitivity and definition traditionally associated with capacitor designs.

Now all we need is a pair of reference monitors to complete the picture. Tannoy's System 2 NFM speakers will do the trick. They feature shielded magnets so they can flank our video monitor without distorting or discoloring the image. These will be powered with a Hafler Pro 2400 amplifier.

Our assembled studio will be ready to tackle just about anything that could arise during the production and editing of a musical track for radio or television advertising purposes. Just add creativity and start producing!

Our assembled studio will be ready to tackle just about anything that could arise during the production and editing of a musical track for radio or television advertising purposes. Just add creativity and start producing!

You're looking at two of the most respected names in sound.


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DSP 256 Digital-Effects Processor
Delivers vocal and instrumental multi-effects, up to four simultaneous effects from 24 available.


IPS 33B Super Harmony Machine To generate vocal and instrument harmony effects, two and three-part chromatic, intelligent and user-defined harmonies.


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Lost in the Timecode Zone

The mystery and the history of SMPTE timecode and the intricacies of synchronization are unveiled at last.

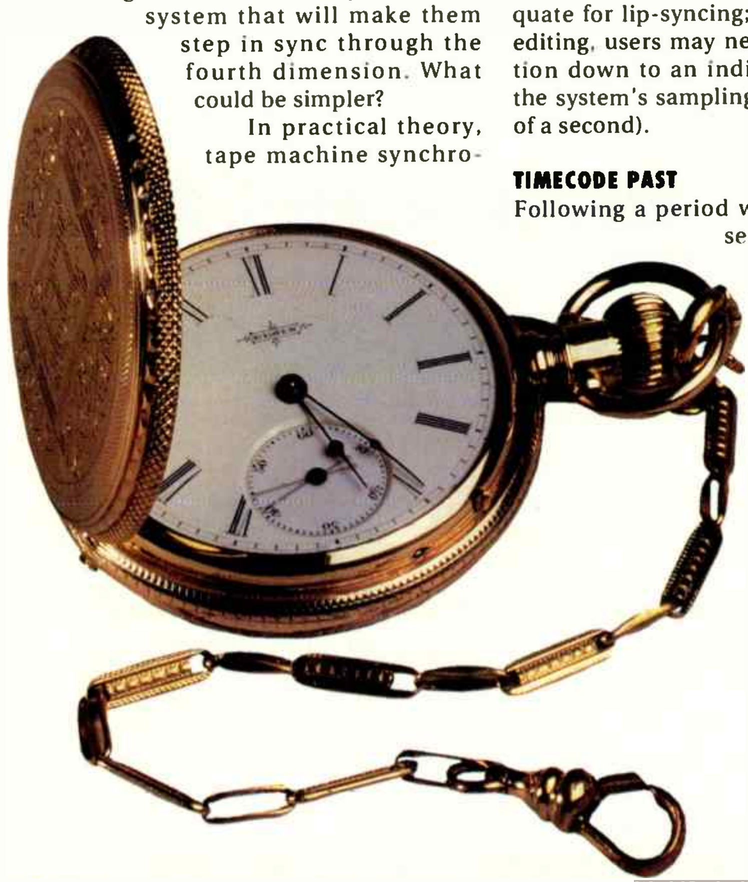
FRED RIDDER

YOU'VE GOT TWO tape machines. You need them to work together, linking up elements, for example, like sights and sounds. So you invent a system that will make them step in sync through the fourth dimension. What could be simpler?

In practical theory, tape machine synchronization via timecode should be pretty straightforward. But in practice, it can be downright confusing, if not a veritable nightmare. The problems arise in the practical and mechanical implementation of what at heart is a relatively simple idea.

Consider what challenges face those geniuses who invent timecode systems. The only absolute requirements for a timecode signal are that it contain some form of position data (e.g., a "frame number") and that the signal be recordable on the chosen medium.

The first complication is that in mixed media synchronization (e.g., audio-video or audio-film) the frame rate of the timecode on the ATR (audio tape recorder) has to match an existing frame structure. In addition, the resolution of the position data must be appropriate to the application; but as we know, audio-for-visual projects vary drastically in their requirements. When working with film, for example, accuracy of one film frame (1/24 second) is usually adequate for lip-syncing; in digital audio editing, users may need to read position down to an individual "tick" of the system's sampling clock (1/48,000 of a second).



TIMECODE PAST

Following a period when there were several competing audio tape machine synchronization systems using non-compatible timecode signals, synchronizer manufacturers finally adopted the SMPTE/EBU edit code as their timecode standard. This choice had the advantage of making ATR/VTR synchronization easier (thus opening the door for sophisticated audio-for-video post production), but it had the disadvantage of assuming the limitations of a timecode that was originally designed for a different application.

BOARD FORMERS

During the development of electronic video editing (to go along with the young videotape recording technology), the video industry had to address the question of non-compatible "edit code" formats. A committee was formed with representatives from the U.S.-based Society of Motion Picture and Television Engineers (SMPTE) and the European Broadcast Union (EBU) to establish an edit code standard. The result was a standardization format for encoding hours/minutes/seconds/frames (plus optional data) in a form that could be recorded on any audio channel.

Unfortunately, there had to be two different types of timecode, even though they shared a common frame format, because American TV broadcasts black and white images at 30 frames per second (fps) while European TV broadcasts at 25 fps. The result: SMPTE timecode, which runs at a nominal 30 fps and uses frame numbers from 0 to 29, and EBU timecode, which runs at 25 fps and numbers the frames 0 to 24. Because the two timecodes share the same data format, the same reader hardware can decode either type.

The next complication was the result of the NTSC committee's standard for color TV signals in America. To prevent problems that might have been caused by certain limitations of practical electronic technology (as it existed then), the NTSC committee shifted the video frame rate slightly (0.1 percent) away from 30 Hz, which happens to be 1/2 the AC line frequency. The definition of the NTSC color frame rate at 29.97 fps is a burden that all timecode users have had to bear ever since.

NTSC AS THEY WANNA BE

One consequence of the NTSC frame rate is that a standard timecode generator won't keep time any more. If we take a standard SMPTE timecode generator and run it at NTSC frame rate, so that precisely one frame of SMPTE timecode is generated for each color video frame, the hours/minutes/seconds portion of the timecode will keep falling behind the clock on the wall because the SMPTE clock is ticking 0.1 percent slower.

For every hour of real time, the timecode will fall 108 frames (3.6 seconds) behind. To eliminate this unacceptable accumulating error, the engineers came up with a thing called Drop Frame.

The basic idea of Drop Frame timecode is to make the timecode generator periodically skip over (or drop) certain frame numbers so that it never falls too far behind real time. If frames are skipped frequently enough, the error never gets very large, but 108 frames is an inherently awkward number to distribute evenly through 60 minutes. So the engineers used another trick: they drop two frames at a time, but only in nine out of every 10 minutes (i.e., 54 times an hour rather than 60 times an hour). The formula for Drop Frame specifies that frames 00 and 01 will be dropped from the first second of each minute unless the minute ends in zero.

In other words, the generator will skip from 00:03:59:29 to 00:04:00:02, but at 00:09:59:29 it will count normally to 00:10:00:00. Because the format of Drop Frame timecode only deviates from standard SMPTE timecode once a minute, a timecode reader can't distinguish the two by observing a few seconds of timecode as it can with SMPTE vs. EBU. Instead, the engineers had to include a flag bit in the format of each Drop Frame timecode word to identify it to the reader.

Drop Frame timecode is at the root of several common misconceptions. First among these is that NTSC frame rate (29.97 fps) is totally synonymous with Drop Frame timecode. In reality, *either* Drop Frame or Non-Drop Frame code can run at NTSC frame rate. Many facilities, particularly audio-oriented ones, prefer to use Non-Drop code even when running at 29.97 to lock with video. The only time it is essential to use Drop Frame is when it is essential that one hour of timecode equal one hour of real time.

The definition of the NTSC color frame rate at 29.97 fps is a burden that all timecode users have had to bear ever since.

TIME WARP

Another common misconception is that one can identify whether the SMPTE stripe recorded on an ATR is 30 fps or 29.97 fps. On playback, there is no way to be certain which frame rate the timecode generator was running at. A timecode reader can identify Non-Drop Frame SMPTE vs. Drop Frame SMPTE vs EBU (25 frames); but it can't identify the original frame rate. The one assumption you can make with 98 percent certainty is that if the timecode is Drop Frame, it should be running at 29.97 fps. There is no good reason to generate Drop Frame code at 30 fps — but there is nothing to prevent it, either. Don't worry; it generally doesn't matter whether the original frame rate of the timecode on an ATR was 30 or 29.97. An inherent part of synchronization is to run an ATR at the right speed to reproduce the timecode at the appropriate frame rate for the present circumstance. If the audio tape was originally striped at 30 fps and is later locked to video at 29.97 fps, it is true that the machine will run slightly slower, and that all audio pitches will be shifted down by 0.1 percent — a very small amount.

The timing of cues will never be affected, however; an audio event that occurs at a particular timecode will always occur at that timecode regardless of the frame rate, which is what we were trying to accomplish in the first place.

A/V NIGHTMARE

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sample by sending it some pitchbend, but then I would lose lip-sync with the woman talking on screen. So I fool around with the sequencer tempo until the rhythm locks in. The pitch discrepancy sounds like a chorusing effect, and it's not terribly noticeable. Rather than trying to match the pitch by putting pitchbend on all of the sequencer tracks (and then having to worry about how jarring they'll sound against the segments immediately before and after), I decide to leave them alone.

Day 5: I'm ready for layback. I cue up my 3/4-inch master, re-route my audio lines, and discover to my horror that none of this is going to work.

Although the BVU-150 does record in stereo and has an "Audio Dub" function, you can only lay back audio to an already-recorded videotape on channel 1, and *not* on channel 2. Why? Only God and Sony know — the rental house sure didn't, and the information was buried in the manual. Someone later tells me that Sony never considered that anyone might want to do this, because these portable machines are generally used for field recording, and the only reason anyone would ever want to do an audio dub would be to replace a voiceover, and that, of course, only needs one channel.

Yes, you can do a video demo yourself at home. But don't trust anything anyone "video" tells you about audio. None of them have a clue.

A few phone calls, and I find that a friend in another studio has a 3/4-inch machine that really will do stereo overdubs, but he can't let it out because it weighs 90 pounds and is worth $19,000, and maybe there's some way I could bring all of my equipment — synths, samplers, reverbs, compressors, mixer, synchronizer, and computer — over there?

After the blood returns to my brain, I devise a solution: master the thing on VHS. I have an SVHS deck, and I can dub the video signal from the 3/4-inch to it, at the same time recording the audio from my sequencer, while locked to the 3/4-inch deck's timecode. It means an extra generation of video (and I can't really take advantage of SVHS's high quality, because I'm not feeding it a composite video signal), but on the other hand, the audio quality will be a lot better than it would be on 3/4-inch, because I can record it in Hi-Fi.

In addition, because I have a second VHS Hi-Fi deck, when I'm all done I can make my own dubs for potential clients, instead of giving the 3/4-inch master to a duplication house. This will save me money, and will also give me control over the dubs' audio quality.

Two passes and I'm done. All the synchronization works perfectly, so I can concentrate on the audio mix. I like the results, and the first few people I show it to are impressed. They're even more impressed when I tell them how much I spent on the project.

Epilogue: Yes, you can do a video demo in your home studio, if you have the patience, and you know your gear well. Just be prepared for Murphy's Law to manifest itself every few minutes. And don't trust anything a video dealer, serviceman, manufacturer, or instruction manual tells you about how their equipment handles audio. None of them have a clue. Try it yourself, and if it works, do it.

THEY DO MORE THAN EXPAND THE CAPABILITIES OF YOUR AMPLIFIER.

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P.I.P.-AMC is the newest of our P.I.P. modules, and combines all the capabilites you need most in one high-performance package. DIP programmable, the AMC features a 4th-order Linkwitz-Riley crossover, "constant-directivity" horn equalization, filter-assisted B_6 vented box equalization and a signal-driven/error-driven variable-threshold compressor. In addition, the AMC can be used for bi-amping, tri-amping and "daisy-chaining" of amplifiers.

P.I.P.-FTE has balanced 1:1 transformers, 12 dB/octave RFI filter, variable 18 dB/octave high-pass filter and a 6 dB/octave 3 kHz shelving network for "constant-directivity" horn equalization. Barrier block connectors.

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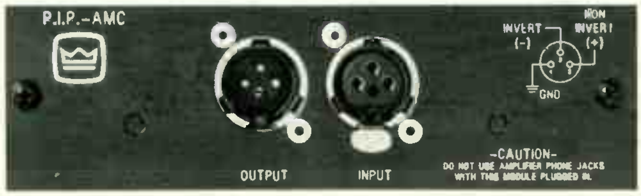
P.I.P.-FXT uses balanced 1:1 transformers to isolate the source from the inputs. It comes with balanced Female 3-pin XLR connectors



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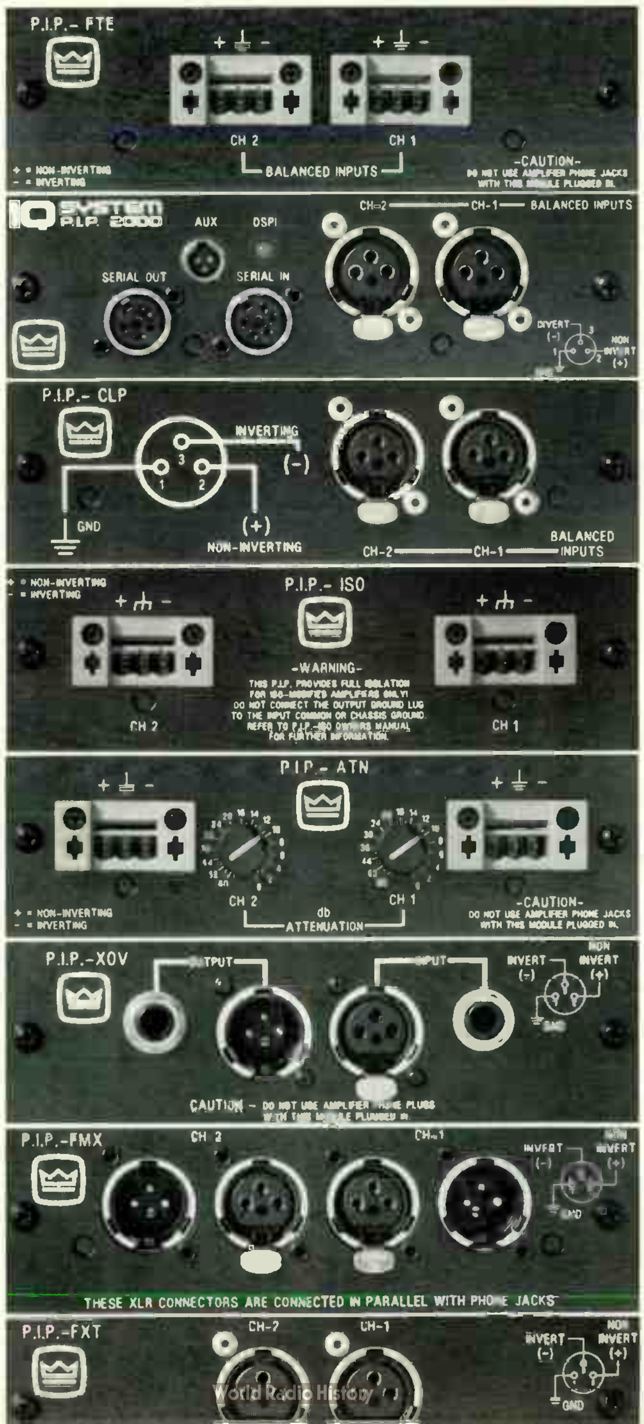

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CIRCLE 12 ON FREE INFO CARD

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excursion or a high speed meeting of voice coil and bottom or magnet top plate. Either way, it's a sign of too much power at too low a frequency.

HEAVY METALS

High frequency compression drivers take their fair share of abuse. The expense of replacement high frequency diaphragms make most people pale, so it pays to keep an eye on them.

The crud that accumulates in the throat can reduce driver output. If the screen is punctured, the high flux magnet gap will suck metal particles in like a Hoover.

Many diaphragms are stamped or formed from single pieces of thin metal. Almost all metals harden and become stiffer as they are flexed back and forth, leading to fatigue failure.

A quick visual inspection of the back side of the diaphragm will usually tell you if you should pull it out. Take a look at the surround; are there any signs of a tiny crack? If it is made from a different material than the dome itself, look to see if the glue bond between it and the dome are intact all the way around.

Additional problems include:

- The flex leads between the terminal and the voice coil, and especially the often troublesome glue joints that hold them down.
- Separation or cracks that may lead to failures.
- Punctures in the diaphragm caused by wayward screwdrivers or fingers.
- The phase plug imprint on the dome, a definite indicator of excessive low frequency material (I've actually seen it).
- Uneven width of the voice coil gap around its circumference.
- Dirt and metal in the gap; loose bolts and connections.

CLEAN SCREEN

Make sure the fine screen in the throat is undamaged and clean. The crud that accumulates in the throat can reduce driver output. If the screen is punctured, the high flux magnet gap will suck metal particles in like a Hoover. The solution? Buy some replacement screens and glue them on the front of the driver, under the throat gasket.

Most aluminum diaphragms are not anodized, so if there is significant moisture, cathodic oxidation occurs between the steel driver pieces and the aluminum diaphragm. I've pulled drivers apart where the diaphragm had completely dissolved into aluminum oxide. Keep them clean and dry. If high frequency devices get rained on, pull them apart as soon as possible and dry them thoroughly.

If you go through these simple checks, you should be able to rest easy that your speakers will keep on cranking throughout the gig — if there's not a blowout, that is. If there is a problem, you should be able to identify what it is and take care of it.

By the way, it never hurts to run through these diagnostics when you're in the market for a used speaker. They can help prevent you from buying a bargain box that will blow up in your face. Go ahead and try to explain that to a crowd full of crazed heavy metal fans.

MAINTENANCE

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REPRO, and start recording.

(3) Adjust the bias trim pot fully counter-clockwise. Then turn clockwise until a peak is achieved. Continue turning clockwise until it decreases to the proper amount of overbias. This is usually 1.5 dB below the peak for 30 ips, and -3 dB below the peak for 15 ips for a 0.50 mil head gap. Adjust all controls the same.

If there is anything that you do not understand about the technical operation of this device, don't touch it...

(4) Set the oscillator to 1 kHz at 0VU. In RECORD and REPRO, adjust the REC GAIN control for a 0VU reading on all channels.

(5) Set the oscillator to 10 kHz at 0VU. In RECORD and REPRO, adjust the RECORD GAIN, or TREBLE, or HF for a 0VU reading on all channels.

(6) Set the oscillator to 100 Hz at 0VU. In RECORD and REPRO, adjust the LF or BASS trim pot for a reading of 0VU on all channels. Since the LF adjustment is a function of the REPRO section, you can record a section of tone at 100 Hz and play it back for further adjustment.

You should now go on to printing reference tones on your tape in preparation for the session. Then you can finally put some complex waves (a.k.a. music) on tape. Next time we will talk about reference tones and other tape machine quirks.

Before I leave you, though, one word of warning. These machines contain few user-serviceable parts. So please refer to a qualified tweaker for servicing. If there is anything that you do not understand about the technical operation of this device, don't touch anything. Your sanity and the financial security of your project studio might be at stake. Let the tweaker beware.

Richie Moore, *Ph.D., is an independent recording engineer, studio designer and technician. Address questions to EQ Editorial Offices, 939 Port Washington Boulevard, Port Washington, NY 11050.*

HIDLEY

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couple of inches. They should be free at the top a little bit as well. Suspending the flanking blankets lets them dissipate pressure build up.

Construct the flanking blankets of 1/2-inch particle board, which usually comes in 4 x 8 foot sheets. Let the boards overlap at the edges, and bolt all edges together for the full wall length (and height). Then staple 1-inch fiberglass on the insides (room-facing surface) of the 1/2-inch board, keeping the shiny aluminum foil or paper backing next to the board and letting the fluffy fiberglass face the room. The fiberglass will keep mid- and high-frequency reflections down to a reasonable room content. Finally, build an open frame 2 x 4 wall in front of the flank blankets (which are now covered with 1-inch soft fiberglass) and cover the open frame 2 x 4 wall with acoustically transparent, synthetic, 100 percent all-polyester open-weave grille cloth to hide the fiberglass from view.

MONITORS & EQUIPMENT

The monitors need to look down the length of the room — leave about 2 feet between the monitor and the wall behind it. Put the monitors on an appropriate center commensurate with the width of your finished room (see room diagram page 28) and focus them horizontally at the engineer. Meanwhile, pack the space between the monitors solid with at least 2 feet minimum (and preferably 3 feet) of Rockwool or dense fiberglass.

The monitors should be sitting on solid concrete blocks, keeping a piece of sound board underneath them to prevent scratches. In my listening room I sit on the floor, so the bottom of the monitors are about 8 inches from the floor. If you're in a chair for listening, keep the bottom of the monitors about 3 to 4 feet off the floor. Never put grille cloth over the front of the speakers. Now you're ready to bring in the gear and fit it into the acoustic environment. Clearly, you want it to interfere as little as possible with the sound field. It's always better if the console is open-bottomed so that the low frequencies won't be obstructed as they pass to the back of the room. Of course, not all consoles are built this way, so you do what you can.

Cover the back of the console with a 3-inch soft fiberglass or 3-inch Sonex. Sonex, a porous and spongy eggcrate-like material is nonreflective to mid and high frequencies (as is 3-inch soft fiberglass). Cover the fiberglass or Sonex with grille cloth. If you can't put the fiberglass or Sonex against the back of the board because of heat dissipation, keep it a few inches away on standoffs.

As much as possible, keep the outboard gear out of the room. For essential "in-room" gear, I recommend a "lowboy" (2 feet high) wooden housing with an equipment mounting front having a 45 to 55 degree angle that deflects sound upward and away from the engineer (see diagrams pp. 30 and 31).

Now you're ready to go. The room should have a clean, honest sound, and the bill should be between $3,500 and $5,000. That investment in acoustic treatment can save you thousands in buying new processors to compensate for a room that doesn't sing.

Acoustician and studio designer **Tom Hidley** *has designed and constructed hundreds of the world's leading music recording studios.*

Voice of a Generation

The Shure 55 Series gave lip-service to an entire generation of warblers.

MARTIN PORTER

If any piece of audio gear carries the same strong and cultural identification of a 1957 Chevy it would be the Shure 55 Series mic. Elvis...Buddy Holly...the nostalgia hormones in your mind conjure visions of a generation of rockers with a shift to their hips, a guitar slumped to their side, and a plated silver birdcage of a unidirectional mic clasped in their hand.

Oddly though, this entire generation of mics, actually known as the 55S Series, was only the most public evolution of a long line of single-element unidirectional vocal mics that were originally introduced in 1937. Back then it was a much larger birdcage-like model that introduced the name Unidyne to the Shure lexicon and accomplished directional sensitivity with a single element where two elements were required in the past.

It was diecast zinc, polished and chrome plated, with a unique, surround grill that gave sound access to the element from all directions. The mic was originally designed at Shure by Ben Bauer, a celebrity of sorts in audio engineering circles, who ended his career at CBS Labs and who accumulated hundreds of microphone and loudspeaker patents along the way.

The product evolved slightly throughout the wartime years.

It became a military mainstay during World War II — from broadcast locations to U.S.O. tours. Astute mic collectors know that the 55A, 55B and 55C were the way dealers sold the models by the mid-forties, with shock mounted versions (featuring bonded isolation and cannon plugs) during that period called the 556A, 556B, and 556C.

However, by the late-forties it was apparent that the venerable voice mic was becoming too unwieldy. Besides, there was a new generation of musicians on the scene who wanted a sturdy but smaller vocal microphone, still characterized by that certain smoothness and rich low frequency response (everybody knew that the high frequency response — like everything else during those pre hi-fi days — left a bit to be desired).

So, in a decade of chrome fenders and chrome drive-in diners, a smaller Shure birdcage mic evolved that would grace the lips of everyone from the doo wop scene to Bob Dylan. These microphones were so popular in the fifties and early sixties that used mic dealers now claim that, even in mint condition, they're only worth about $75 a piece. The 55S Series came to an end by the early 1980s. Microphone technology had left it an antique and, besides, the handheld microphone that didn't hide your lips was the vogue of the day. Oddly though, the 55S refused to die.

Dinah Shore and one of the older generation birdcage microphones.

With the emergence of fifties and sixties groups on the scene, and a high profile in movies like "Good Morning Vietnam," Shure decided to reissue this microphone classic in the form of the 55SH Series II. Its sound quality was improved courtesy of a newer cartridge (the same as the SM48), along with conveniences like an on/off switch placed on the lower part of the swivel base. Update kits were offered for the existing multitudes of vintage 55 Series mics still out there being used.

Microphone collectors still get plenty of requests for the "real thing." Robert Pacquette, owner of Select Sound in Milwaukee, Wisconsin, and curator of one of the country's largest private collection of vintage microphones, readily admits that the 55S Series doesn't compare in value or in demand to the much rarer RCA ribbon mics. However, he recalls, that didn't matter to a recent Japanese caller who wanted to locate and buy 50 vintage Shure 55S Series mics at a premium price.

What for? One for every table at his fifties karaoke bar in Tokyo.

IN REVIEW

A.R.T. Power Plant Guitar Preamp

PRODUCT: Power Plant
MANUFACTURER: A.R.T., 215 Tremont Street, Rochester, NY 14608; Tel: 716-436-2720, Fax: 716-436-3942
EQ FREE LIT #: 155.
APPLICATION: A task-specific single rack space guitar preamp designed for clean and overdriven amp simulation, the Power Plant adds kick to direct-recorded guitar tracks.
SUMMARY: This model performs only one task, but performs it brilliantly. It offers complete inputs and outputs, including effects loop, and puts out an extremely clean, musical — and controllable — sound. It cannot, however, store different sounds or respond to program changes — a drawback for guitarists who trigger program changes from MIDI footswitches or sequencers.
PRICE: $329 suggested retail.

Electric guitars are difficult to record directly into a console because for many guitarists, the amp serves as a signal processor in its own right. Leaving that particular "effect" out of the signal chain usually results in a flat, uninspiring sound (although there is some compensation in that you don't have to deal with the hum and noise typical of many vintage amps).

Still, you can't always mike a stack of Marshalls cranked up to 11, especially in smaller studios. Knowing a product niche when they see one, several manufacturers now offer both solid-state and tube preamps designed to simulate a "real" amp sound so that guitarists don't have to lug around a huge amp from session to session, and engineers can cut a guitar track directly to the console without mic hassles. (Editor's Note: See "Going Direct," EQ, November/December).

A.R.T.'s Power Plant (Model 410, single rack space) is designed for clean and overdriven amp simulation — nothing else. In today's era of everything-in-one multieffects boxes, that may seem anachronistic; there's no MIDI, microprocessors, or memory. Yet there are some compelling compensations for such dedication. By zeroing in on one specific task, the Power Plant performs that task brilliantly. On a more pragmatic level, in many of today's studios a multieffects unit can actually be redundant, so there's no point in paying for features that may not be needed. There are lots of excellent signal processors that produce multiple effects (like the Lexicon LXP-1, Alesis Quadraverb, A.R.T.'s Multiverb, etc.) but don't offer distortion. Well, stick a Power Plant in front of one of these digital wonder boxes, and guitars suddenly come alive.

Muscle tone. The Power Plant offers a good deal of tonal flexibility. There's a clean channel with defeatable EQ (low, mid, and high, boost and cut), and an overdrive channel, with one or the other selected by a front panel control or remote footswitch.

The overdrive channel has six controls. Input Drive sets the level going to the distortion stage; turn up to crunch, down to clean up. The "Harmonic Superdrive" control is also pre-distortion. Boosting it feeds more midrange (around 1 kHz) into the distortion, which imparts more definition to chorded notes. Spinal Tap fans, take note: both controls go up to 13.

The three post-distortion, boost/cut EQ controls are bass (called "Thrust" and centered at 130 Hz), midrange (450 Hz), and high (called "Edge" — around 5 kHz).

Ins and Outs. The Power Plant is well endowed with multiple outputs, each of which is equalized and level-matched for a particular application. If you're using the Power Plant with a standard guitar amp because you don't like the stock overdrive sound, connect one or both of the two paralleled "Instrument Level Out to Guitar Amp" 1/4-inch phone jacks to the guitar amp inputs. If the Plant serves as a guitar preamp driving a clean power amp/guitar speaker cabinet combo, patch the "Equalized Output to Power Amp" 1/4-inch phone jack to the power amp input. This is a stereo jack with out-of-phase outputs, making it usable in balanced line applications as well. When doing direct injection into a console or recorder, there are two "Equalized Lo-Impedance" jacks (one XLR and the other 1/4-inch phone; the latter can also drive headphones). Both are balanced 600Ω outputs, although they work with unbalanced console inputs too. A level-shift switch selects -10 or +4 output levels for the power amp and console jacks.

All outputs are available simultaneously should you want to, for example, send a direct feed to a console *and* drive a guitar amp for an ambient room mic to pick up. It's important to use the specified output for the intended application, even if you're using other effects with the Plant; using the wrong output won't break anything, but you won't get the full benefits of the unit.

The input scene is much simpler, with paralleled jacks on the front and rear panel. The front panel jack takes priority over whatever is plugged into the rear panel. Finally, an effects loop rounds out the complement of jacks.

Sounding off. The sound is, in a word, magnificent — but most importantly, it's controllable. Turn down the guitar's volume, and only the character, not the quality, of the distortion changes. The notes within a power chord always remain well-articulated, even with lots of drive and sustain; notes don't "wash out." Pick attacks are well-defined and they retain the character of the guitar being played. As expected, there is considerable intermodulation distortion when playing several notes at once, but it's a "musical" effect compared to the aliasing-based distortion artifacts present in all too many digital designs.

The Power Plant also excels at creating warm, crunchy rhythm guitar overdrive sounds. These are traditionally difficult, if not impossible, to obtain with solid-state distortion circuits. If, on the other hand, you want smooth and sweet lead sounds, just crank up the Drive and Harmonic Superdrive controls. The noise remains surprisingly low under these conditions.

My only disappointment is the Power Plant's inability to store different sounds. This box is capable of generating lots of great timbres, and it's a shame they can't be loaded into memory for later recall. Guitarists who trigger program changes from MIDI footswitches or sequencers will have to factor in the cost of a Scholz Octopus or some other MIDI switching device to control the Power Plant, and even then, you won't be able to call up different settings — just in/out, and the clean or distorted channel. Nonetheless, the Power Plant sounds so good it's definitely worth an audition.

There's one caution: with so many controls and outputs, it may be a few minutes before you find the sound that's perfect for a given guitar. (Check out the suggested settings on pages 10, 11, and 13 of the manual to get started.)

So is the Power Plant just another overdrive box? Not at all. This is one amp simulator that definitely makes all the right moves; it puts "*that* sound" in your project studio with a minimum of complication and expense.

— Craig Anderton

Fostex G-16 Multitrack

When I set out to review the Fostex G-16, I thought it was going to be just another ordinary 16-track analog tape recorder. Was I in for a surprise. Fostex has somehow managed to take affordable multitrack technology to new heights. This two-head, 1/2-inch G-16 has all the ingredients of the higher-priced reel-to-reels, and then some. It is both a sonic and technically crafted instrument that delivers a great deal of flexibility.

Overview. Everything is located in one unit: the tape transport, the record/reproduce electronics, a detachable control panel, and built-in Dolby C noise reduction. Among its other features, the G-16 incorporates a new 16-bit serial transmission CPU that was developed for multitrack recording, and a new cam drive system in the transport for increased stability.

The transport only operates at 15 ips (38 cm/s). Two DC servo motors are employed in the transport for use in reel control; one phase-locked DC servo is used as the capstan motor. The other, a 'loading motor', is part of the cam

IN REVIEW

drive system. The transport uses a guide-roller-roller-guide design that keeps the tape stable and eliminates the need for a scrape-flutter idler and ensures constant tape tension. The transport handles a maximum reel size of 10.5 inches.

The G-16 is designed to operate at an unbalanced line level of -10 dBV (317 mV), and has 16 line input and output connectors on the back panel. Two additional RCA jacks on the back panel deal with track 16. One, the 'Loop Out-16 In' jack, is connected in parallel with input 16 for sending timecode to another device when you've connected the timecode generator to input 16. The other, a '16 parallel Out' jack, is good for playing back timecode on track 16 and sending the code to two other pieces of equipment. Why hasn't someone thought of this before? Besides an Accessory 1 jack, there are two 1/4-inch jacks for plugging a footswitch to punch in/out, and also to play/locate. There's also a receptacle to plug the controller into the back, and a switch to select which place the controller is plugged in (front/back).

The removable front panel controller unit, divided into two areas, is the heart of the G-16 system. On the front left, control panel 'A' contains the bargraph meter display for the 16 channels, the individual READY/SAFE switches, the Option key area for the Model 8330 synchronizer card, and an extensive numeric keypad for entering/viewing data. Switches on the 'A' side include the INPUT MON, METER, ZONE LIMIT, RESET, PITCH CONTROL and the pitch control knob. Control panel 'B', on the right side, contains the transport control buttons for STOP, PLAY, RECORD, etc. It also has the LOCATE and LOCATE ZERO; as well as

PRODUCT: G-16
MANUFACTURER: Fostex Corp. of America, 15431 Blackburn Avenue, Norwalk, CA 90650; Tel: 213-921-1112.
EQ USER LIT #: 156
APPLICATIONS: This 16-track analog tape recorder fits into virtually any studio environment, and with the addition of the optional 8330 Synchronizer card, it acts as a slave or in tandem with a wide range of MIDI and other studio gear.
SUMMARY: The G-16 packs an enormous amount of power in a compact and streamlined unit. It operates only at 15 ips, but through the use of innovative feature splitting, it offers an almost uncountable array of functions for you to choose from.
PRICE: $8,000 alone, $9,000 with the 8330 (Suggested Retail).

the AUTO RETURN and AUTO PLAY buttons. The tape and memory LED display are also located in this area.

Kitchen Sync. The unit I tested was equipped with the optional internal Model 8330 Synchronizer card, and I think it's an absolute necessity. It gives the G-16 the same features as the Fostex 4030 synchronizer unit and the MTC-1 that is used with the Fostex R-8. Suffice it to say that the SMPTE and MIDI options it adds are well worth the price. For that matter, the MIDI control of the transport from your Macintosh or Atari alone is well worth it. The 8330's MIDI interface will output information on operating conditions and tape position to your computer and sequencer. The G-16/8330 is designed to operate as a slave machine, to either another ATR (analog tape recorder), VTR (video tape recorder), or your sequencer program. The 8330 supports SMPTE longitudinal time code (LTC) and MIDI time code (MTC).

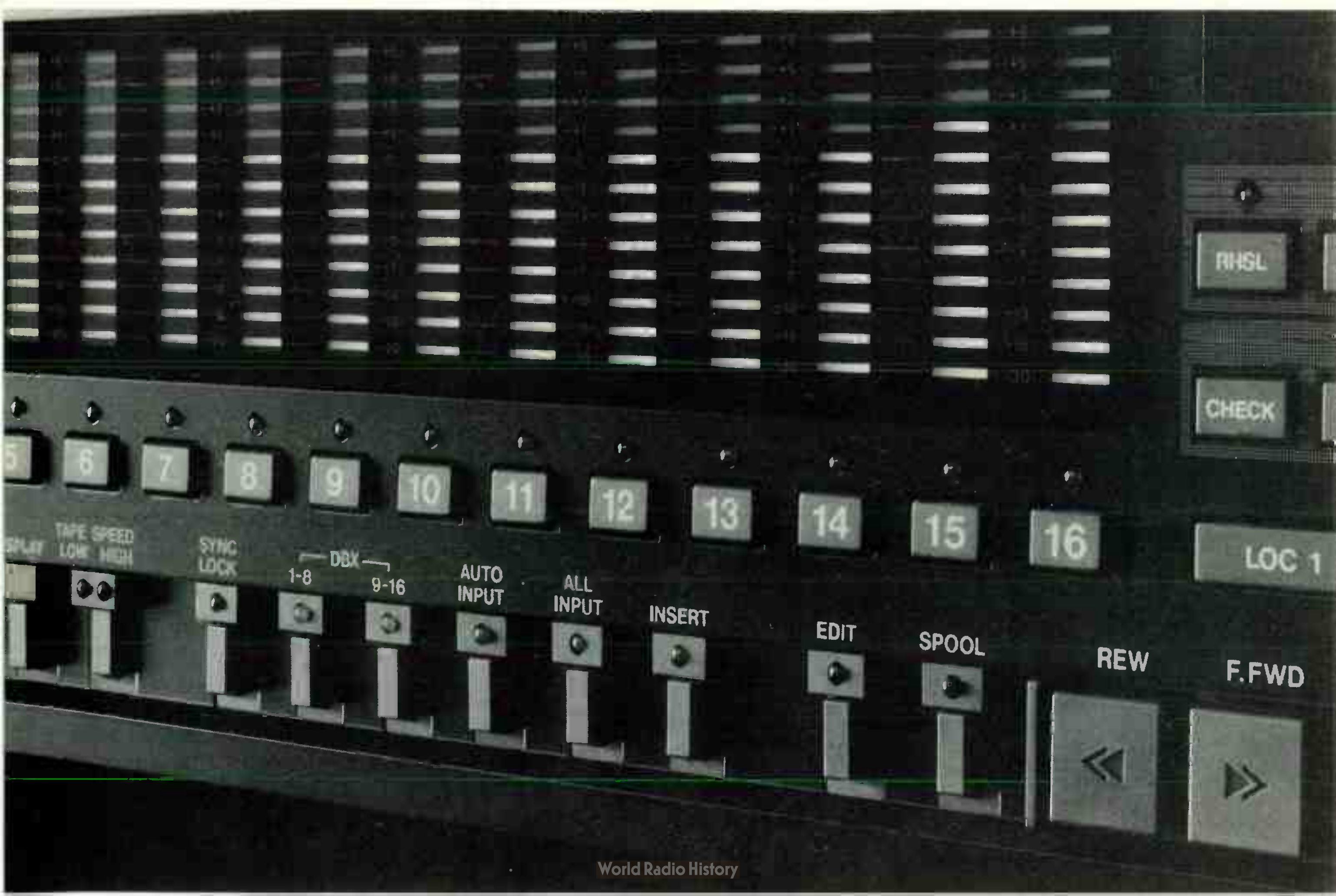
Since the G-16 is a two-head machine, it is not really a field alignable device. The product is aligned at the factory and it is set up for the industry standard Ampex 456 tape.

All the controls are well laid out, and the switches have a nice tactile feedback. The best thing about the control panel is that you can disconnect it from the machine and move it up to 16.5 feet away for remote operation.

The EDIT, SPOT ERASE, and the EDIT JOG wheel are located on the transport, as they should be. If you are going to edit or spot erase, you will want to be right there. The only drawback to the spot erase is that the bias ramping of the erase head can leave a pop on the tape if you're not careful.

Importantly, you can really customize the G-16/8330 to match your individual recording requirements by using the RCL (recall) key on the keypad in combination with another key. For example, you can set the INPUT MON to two different modes — one so a track in ready will stay in INPUT all the time, and the other so it stays in INPUT only when the tape is not in play. You can set up the thickness of the tape so that the ZONE LIMIT will function correctly. Other parameters include setting the pre-roll time, the last play point, setting the AUTO RETURN and AUTO PLAY points, and the meter resolution.

The second mode function lets you set numerous parameters, like the lock-up



frame rate, the reel diameter, and the reel control so that the tape slows down before winding off (a fabulous feature). There are some 38 adjustable settings in the second mode.

Thanks to Dolby C, the machine is very quiet. The frequency response is well within spec, and you can hit the machine very hard with signal before it distorts. Vocals are very clean and silky sounding; percussion is very sharp and distinct. There is a little looseness in the low frequency, but this is one area where a narrow format machine cannot compete with wide format. It still sounds very good.

The machine does not feature seamless punching. There always seems to be a little 'burp' when you punch in the middle of a sustained chord — but no audible 'pops'.

The G-16/8330 is a lot of machine for the money. It would have been nice to describe everything that you can do with it, but that would be impossible here. Fostex itself is still discovering new creative possibilities that the deck's designers originally built into it.

— Richie Moore, Ph.D.

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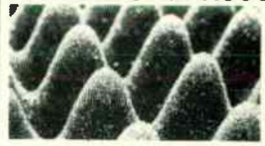

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need for SMPTE timecode.

I recently ran out of tracks on a 32-track digital machine. After we were all done, we wanted to add just one more instrument. I could have mixed down six tracks of background vocals to two tracks, but didn't have the open tracks to do it. I recalled my conversation with Gary Katz 16 years earlier and decided that there must be a way.

There was. The plan was to move a pair of keyboard tracks to a normal DAT machine, mixdown the six vocal tracks to those two tracks, put the keyboards back onto two of the old background tracks, and have four tracks open for further overdubs. I pre-rolled the digital multitrack to about 30 seconds before the start of the tune, and copied about 20 seconds of SMPTE from the timecode track onto one of the keyboard tracks I was going to copy to the DAT machine. I then rewound the multitrack an additional 30 seconds before the start of the tune and made a digital transfer of these two tracks onto the DAT machine.

At this point what I had on the DAT machine was 30 seconds of nothing, then 20 seconds of SMPTE, then 10 seconds of nothing, then the two tracks of keyboards for the duration of the tune. I mixed down the background vocals to the old keyboard tracks, and then erased the originals, leaving six empty tracks.

Now comes the cool part — putting the keyboards back on the multitrack. The digital outputs of the DAT machine were routed back to the digital multitrack. The analog output was patched to the SMPTE input on the synchronizer. The DAT was the master, and the digital multitrack was the slave. The synchronizer was in video mode so that upon SMPTE lock, it would release the machine to external sync. In this case, it would be word clock generated by the digital audio from the DAT. I located the DAT to the start of the SMPTE and pushed PLAY. The multitrack was locked up within six seconds and the synchronizer released the multitrack to external word clock. During the 10-second space between the end of SMPTE and the start of the track, I punched into record on what was to be the new keyboard tracks. Bingo, it worked. The keyboard tracks were being transferred digitally back onto the multitrack without a hitch.

Using this method, it is possible to record audio in the field and sync it up later. Try a switch box on the analog audio inputs of the DAT machine. Feed the SMPTE from the common generator going to the cameras and analog recorders. Start the DAT machine way before you would normally and record the SMPTE from the generator. A few seconds before the take, switch over to your normal audio feed. It's a real piece of cake, am I right?

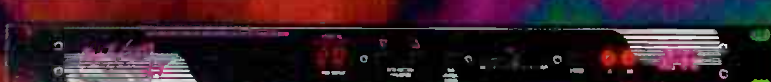
If there's a choice, use a normal SMPTE DAT for these types of applications, but Murphy's Law says that the very time you need it most, you won't have it. Having alternatives that work can often save a session.

So next time Gary Katz asks why I can't use these extra knobs on the board to store a track for just a few minutes, I promise not to yell at him. You know, he was right after all.

THE SOUND OF PERFECTION

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

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



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

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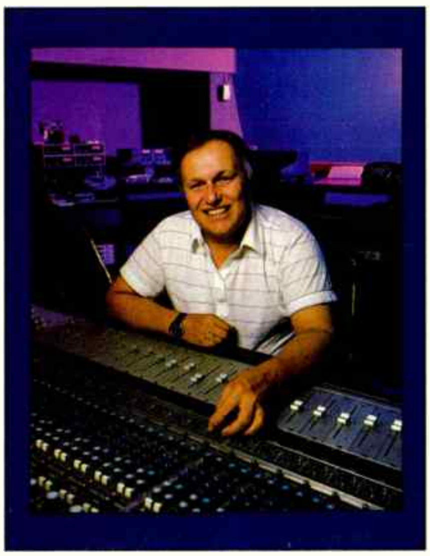
As Time Passes By...

A timely look at timecode, trickery and that ever-so talented DAT.

ROGER NICHOLS



Courtesy of Rupert Neve

I remember some early Steely Dan sessions, around 1974, when I began realizing that 24-track just wasn't enough. Gary Katz, Steely Dan's producer, would often look at the extra positions on the console — modules 25, 26, 27, etc.— and ask why we couldn't store sound on one of these unused tracks. Just for a second — just long enough to combine some tracks so we'd have more room to record, and then we'd put it back on the tape.

I guess we could have synchronized a second multitrack, bounced the track over there for a while and then put it back after the tracks were opened up on the master, but that track would be two generations old. "Nope, won't do it," I said. "We'll have to wait until they invent SMPTE synchronizable DAT machines."

DAT TIME

On the surface, SMPTE timecode and DAT go together like military and intelligence. The DAT format was never designed to accept timecode. Sony's first portable machine, the PCM 2000, recorded timecode on a longitudinal track, but it couldn't synchronize with external systems. To lock up your digital audio you had to transfer the audio and timecode to another format.

Recently, however, a few ingenious engineers figured out how to cram the data into the subcode area of the DAT tape. The first to use the technique was the Fostex D-20. It could record and play SMPTE at any frame rate and also work as a slave machine locked up with an external synchronizer. I personally have one of these machines and have clocked around a billion hours on it, and there probably isn't an audio-for-video studio in existence that doesn't have at least one of these machines.

At the 1990 AES show, a few companies showed their new SMPTE-based DAT machines. Sony's new 7000 Series can be built up with whatever features you want — sort of like a car. The base price is around $7,000, and it can more than double with all the available add-on features.

JVC's DS-DT900N SMPTE-based DAT machine (around $4,500) has only two heads, meaning you must record timecode at the same time you record audio. You can't pre-stripe the tape with SMPTE or add timecode to a pre-recorded tape. But it is as good a sounding digital audio machine as you could ever ask for. With Delta Sigma A-to-D conversion and oversampling DACs, it could be used for just its converters. Its multi-pin connector on the back panel lets it talk to an external synchronizer, so you should be able to incorporate it into any existing lock-up situations.

Otherwise, Otari showed a prototype of their SMPTE DAT machine that will be available later this year. And several other manufacturers said that they are looking seriously at SMPTE DAT machines to add to their pro audio arsenal.

HEAD COUNT

How many heads do you really need? With four, you can pre-stripe the tape with SMPTE and record audio while the machine is locked to an external source. The four-head design also allows for punch-ins with the appropriate cross fade to insure glitch-free edits. The caveat is that when you add SMPTE to an existing tape, you are erasing the audio data already on it. One set of heads plays back the digital information already on the tape and stores it in short-term memory. The incoming SMPTE data is added to the sub-code area and the new digital information is written back onto the tape. If an error occurs, the original audio data can be lost. I know I would be hard pressed to explain how I erased a client's tape. So make sure to stripe a digital copy, not the original master.

The same sort of thing can happen while adding audio to a tape pre-striped with SMPTE. While audio is being recorded, the SMPTE information is continually being re-recorded. If you are doing a lot of punch-ins, then there is a high probability of drop-outs on the SMPTE track.

FUDGING IT

For those of you who only occasionally require SMPTE DAT, but can't justify the extra expense, it is possible to synchronize a non-SMPTE DAT with timecode. That's because when you transfer audio in the digital domain from one machine to the other, the two machines are actually synchronized by virtue of the common word clock. When using a synchronizer with two video decks, after the initial lock, the synchronizer releases the machines to the video sync signal common to both machines, and they stay frame locked from that point on without further

A few ingenious engineers have figured out how to cram timecode into the subcode area of a DAT tape

continued on page 80

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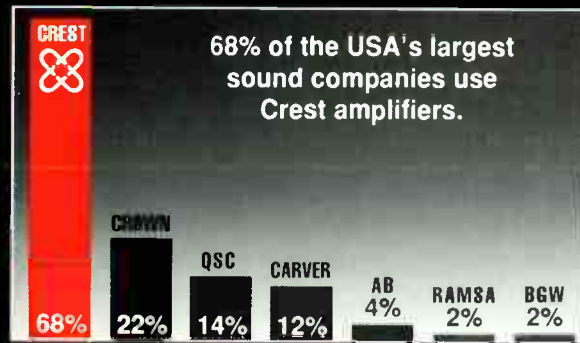

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