PROJECT RECORDING & SOUND TECHNIQUES

EQ

JUNE 1992
PSN PUBLICATIONS

CROMARTIE AT HOME BASE

MIC SPLITTERS · IN A BAND VAN · SELF POWERED MIXERS

LETHAL WEAPON 3, ERIC CLAPTON, ROBIN HOOD & (OH YEAH) MICHAEL KAMEN by Michael Kamen

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- Drawmer DS301 Gate/Processor
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MIDI 2.0 BY CRAIG ANDERTON

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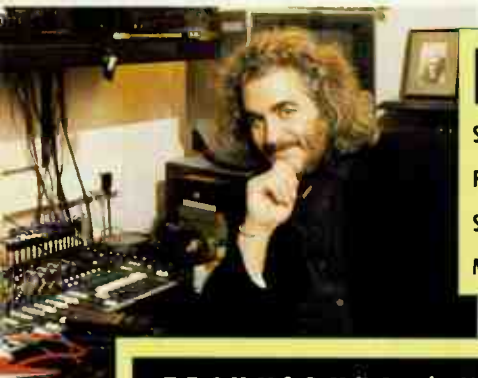
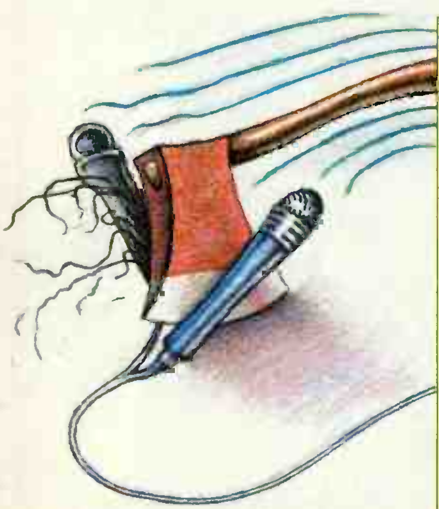
VOLUME 3, ISSUE 2
JUNE 1992

FEATURES



BAND IN A VAN / SPECIAL SECTION

TECHNIQUES / WORKSHOPS



COLUMNS/DEPARTMENTS

On the cover: Kamen by Stephanie Rushton; Cromartie by Deborah Gray Mitchell; Band In a Van logo by Mark Alhadeff

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Delay

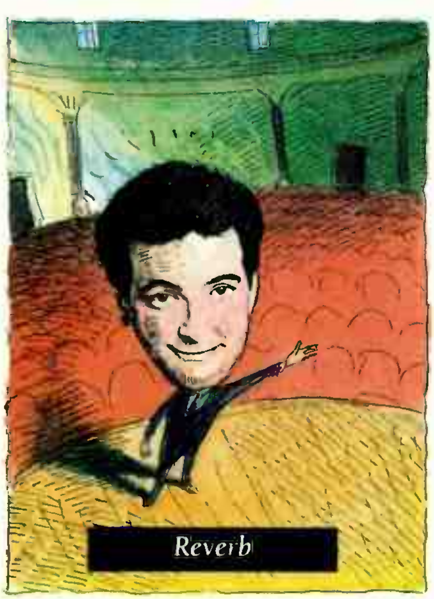

Stereo Flanger


Ambience


Rotary


Multi-Band Chorus


Phase Shifter


Reverb


Vocoder


Stereo Pitch Shifter

Now, just about anyone can become a schizophrenic.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

No we're not. Yes, we are.

Roland®

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CIRCLE 51 ON FREE INFO CARD

EQ

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LETTERS TO EQ

LOVE LETTER

EQ Magazine. Great publication. Logo stinks. Looks too much like a cigarette sticking out of a fat, red face. Change logo. Thank you.

(unsigned)
Denver, Colorado

Excuse us, *but we'd like to correct you. Our logo is not* just *a cigarette sticking out of a fat, red face. It is sometimes a cigarette sticking out of a fat* blue *face. [See the April/May issue for proof.] However, this does bring up an interesting point: What do* you *think our logo resembles? Psychiatrists for years have been using Rorschach tests as a means of discerning their patients' sanity. We here at* EQ *want to discern our readers' sanity (or lack thereof). Drop us a line describing what our logo looks like to you. The best responses will be the winners of our brand new, never before seen (and as of yet non-existent) Band In A Van T-Shirts. Send your responses to EQ Rorschach Test, 939 Port Washington Blvd., Port Washington, NY 11050. And don't worry, we'll only send the severely deranged ones on to a well-trained professional psychiatrist.*

COLD WAR POINT

Recently a lot of press attention has been given to Microtech Gefell microphones. Much of the claims reported demand and deserve clarification. Misleading insinuations and factual errors have left the impression that these microphones are somehow associated with (i.e., manufactured/distributed and/or endorsed by) Georg Neumann GmbH. As a result, Neumann/USA has been asked frequently to respond to inquiries concerning the Microtech Gefell "Perestroika" microphones.

Virtually any mention of Microtech Gefell products is accompanied by a fanciful report about a fortunate discovery of a long lost Neumann relation behind the now crumpled Iron Curtain by the former U.S. representative for Georg Neumann GmbH. A brief review of the history will put this relationship, as it were, into proper perspective.

In the late years of World War II, production of precision equipment became extremely difficult in Berlin as it was one of the main target cities for allied bombings. Georg Neumann moved his manufacturing operations to a small farming town half way between Berlin and Munich, named Gefell, where production soon resumed. In the aftermath of the war, the deteriorating political situation in post-war Germany prompted Neumann to return most viable manufacturing equipment and operations to Berlin. Soon thereafter, all business and technological ties with the Gefell plant were severed. The East German Government expropriated the plant and renamed it Mikrofon Bau, Gefell, abbreviated as M/B Gefell. Decades passed with virtually no contact or exchange of technical information between the two companies. With Germany's recent reunification, M/B Gefell microphones became available in the West. With the coincident Neumann restructuring, the former U.S. representative picked up a replacement line in Microtech Gefell.

Examination of the construction, circuits, components and quality of the Gefell microphones reveal that they bear little resemblance to the Neumann line of microphones. They would be more aptly described as being the product of decades of technological developments by the communist East German Government. Georg Neumann GmbH was not involved in any way with the design and/or construction of these M/B Gefell products, as might be inferred by the erroneous statements printed in some recent publications.

In response to some of these erroneous statements here are some facts:

The capsules used in the UM70 (and the slightly less noisy UM70S) are a *copy* of a capsule originally developed by Georg Neumann in 1947. It is not produced by Neumann and any inference suggesting that these are Neumann capsules is erroneous. Neumann capsules are incorporated only in genuine Neumann microphones. Furthermore, to our knowledge, Gotham Technology Group had not gotten hold of any cap-

WRITE TO US

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sule, Neumann or otherwise, leading to the development of any product. The Gefell products were a fully developed line.

There was no Neumann involvement of any kind in the design of these microphones or any other Microtech Gefell microphones.

And there is neither a Neumann "Perestroika" mic line, nor any involvement by Gotham Technology Group with Neumann.

The decision to purchase a high quality studio microphone should be based on careful subjective listening tests as well as a thorough examination of technical specifications (i.e., self generated noise figures, immunity to RF interference, etc.) from a manufacturer with a proven track record of reliability and performance. Neumann/USA is proud to represent those microphones which have served as *the* reference in virtually all comparisons in this field for the past four decades.

Jeff Alexander
Product Manager
Neumann/USA

COLD WAR COUNTERPOINT

This is in response to the letter being circulated by Neumann/USA that calls into question Gotham's credibility on its marketing efforts on behalf of the UM 70 Perestroika microphone manufactured by Microtech Gefell GmbH.

We have made only one claim with regard to the UM 70's pedigree. Its M 7 capsule was originally designed by Mr. Georg Neumann (in 1932 not 1947) and used in the U 47, U 48, M 49 and UM 57 microphones. This is fact. Gotham has never inferred, implied, insinuated or published any statement to the contrary.

Microtech Gefell's current product line is indeed "the product of decades of technological developments by the communist, East German Government. Georg Neumann GmbH was not involved in any way with the design and/or construction of these M/B Gefell [sic] products..." As our promotional literature states: "Much of the equipment which Microtech Gefell GmbH has...is far more advanced technology than what Neumann, Berlin [Georg Neumann GmbH] owns."

Our published version of Microtech Gefell GmbH's history is almost identical to the one related in this letter, with one exception. In 1972, the East German Government renamed the company VEB Mikrofontechnik Gefell, not Mikrofon Bau Gefell. This is another company entirely, which did not exist until the mid-1960s.

Gotham's Perestroika advertising campaign was indeed designed to evoke romantic images. A company launching a campaign for a product in a market deluged with similar products must focus its creative efforts in capturing attention. We are proud to say that our Perestroika campaign has done exactly that. Perhaps too successfully since it has engendered the "sour grapes" tactics that are being employed to discredit it.

The references made in various articles linking "Neumann" and the M 7 capsule are not inaccurate. First, we did not have editorial control over

these articles and could not and would not exert any influence over their contents. However, these references were not to Georg Neumann GmbH but to Mr. Neumann himself. We would like to point out, again, that this capsule is Mr. Georg Neumann's design. Strictly speaking, the capsules "incorporated" today by Georg Neumann GmbH in its microphones are not produced by him either. In the 1960s, Georg Neumann GmbH revised the M 7 capsule to make it easier to manufacture. In the process, the "sound" of the capsule was changed and in our opinion, its integrity was compromised. Mr. Neumann taught the workers at Microtech Gefell GmbH how to produce the M 7 capsule and it has been in continual production since 1943 by these same workers exactly as he taught them. We feel Microtech, in its strict adherence to Mr. Neumann's original specifications, is closer to being his ideological heir than Sennheiser or Neumann/USA will ever be.

Every consumer should base his buying decisions on criteria (including price) reflecting his priorities and needs. The Perestroika microphone, in a very short time, has become a viable alternative to its high-priced competition and for good reason! We know it's gained marketshare and has caused some scrambling among the competition. Some have responded by sharpening their sales and marketing techniques. Others have responded with distasteful and unprofessional tactics. To those in the latter category, we would like to explain our marketing strategy by quoting a past Soviet leader: "We will bury you."

R. Wm. Wannamaker
Director, Publicity & Advertising
Gotham Audio

ROGER AND ME

First off, let me say I always enjoy Roger Nichols' column. I find it informative, interesting and amusing. After reading his "More Dis About Dat" in the January issue, I was curious as to the modification made on his wife's Technics DA-10 DAT recorder. This modification allows the machine to record analog in at 44.1 kHz.

I would like to have the modification done to my DA-10 or, better yet, I'd like to do it myself. I'm an electrical engineer, as well as musician/studio owner so with the right info, I wouldn't hesitate to proceed.

Doug White
46 Central Street
West Boylston, MA

The modification was performed by Marc at F.E.T. Electronics, 17306 Saticoy Street, Van Nuys, CA 91406; 818-881-2656.

CLEANING UP CRUDE

Regarding the "Maintenance With Moore" column in the January issue.

Generally, the statements are correct, but the mention of acetone as a cleaning agent is a partial boo-boo. [*Mr. Moore mentions acetone only with respect to cleaning a machine's brakes — The Editors*]

To simply mention acetone as a

tape recorder cleaner may cause some unwitting readers to use it on other parts of the machine. In this day and age, when we are looking into every aspect of everything we do, it is imperative that the downside of everything should always be mentioned.

Acetone is a good cleaner. Unfortunately it also attacks most plastics, which is why it is used as nail polish remover. If it is used to polish the "glass" on the VU meters, we usually find the "glass" is/was plastic, and is now smeared beyond repair. Some pressure rollers will also react badly, as will some of the sleeve bearings on various shafts and posts in current equipment.

If a commercial cleaning fluid is used, always read the label. It should warn against bad reactions. It will also warn against any toxic effects that may exist. HEED THE WARNINGS! They are there for our protection, and for those around us who may be forced to breath the fumes, or contact surplus liquids and powders.

Other than denatured alcohol, most of the chemicals used today are hazardous in at least one aspect. They can attack the parts they are being used on, can give off toxic or noxious fumes and can burn eyes or skin. They can also cause later damage through breathing or direct skin contact.

One quick point to be made of a later paragraph. Using a pencil eraser is still using "sandpaper" on delicate contacts. If this is done, a soft, fine eraser must be used, and then only sparingly. The gold plating is very thin, and can be worn off or "trenched," causing even more problems than the corrosion may have.

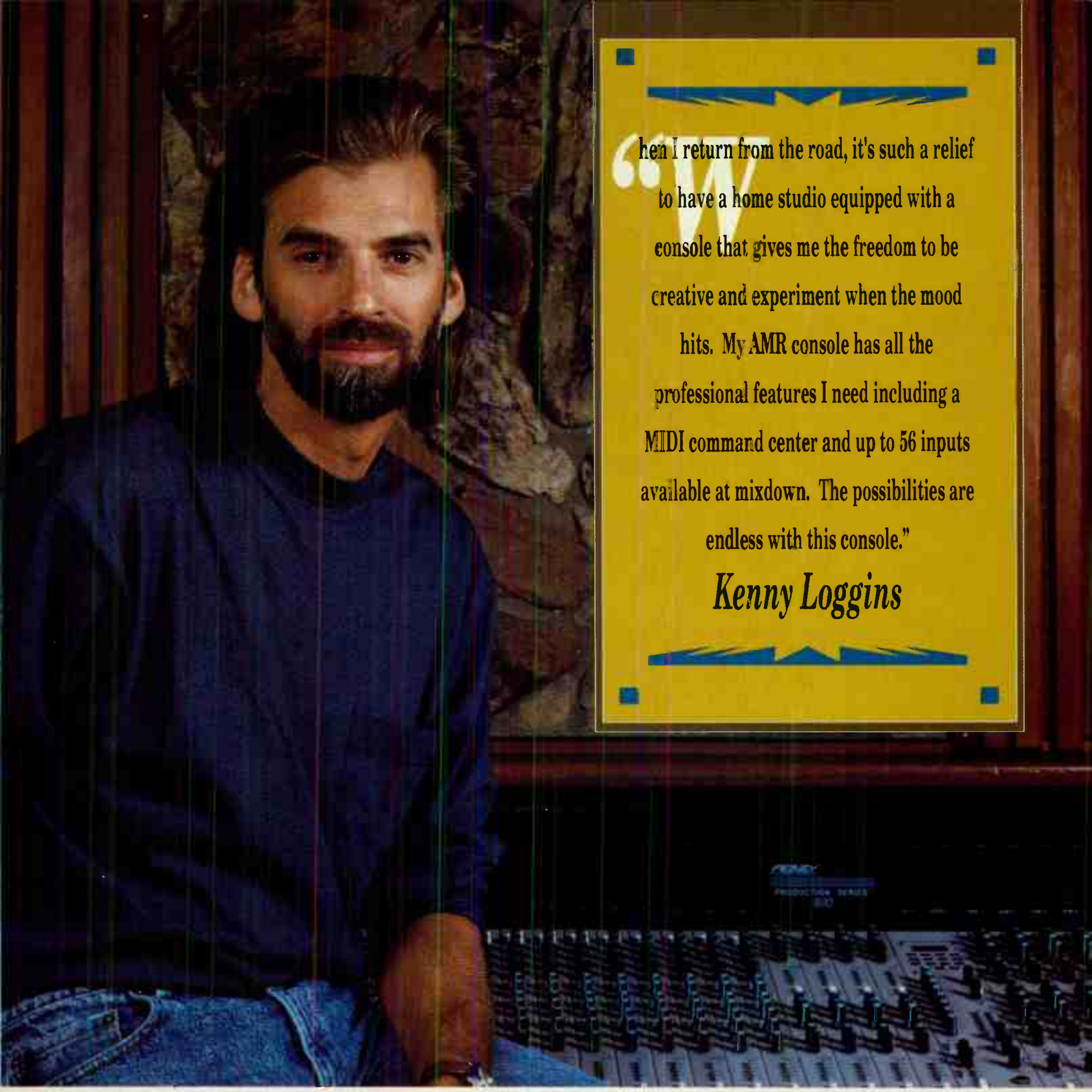
Sequoia Electronics
Los Gatos, CA

CAESAR SQUEEZER

This is in response to Marvin Caesar's letter, "Getting Wired," in the April issue of *EQ*.

He states "The common mode rejection ratio (CMRR) is the number of dB the input stage attenuates the noise and is a function of how well the input stage is balanced." This defines common mode *Attenuation*, not CMRR. CMRR is the ratio, usually expressed in dB, of the *normal or differential mode, voltage gain divided by the common mode differential gain.*

continued on page 84



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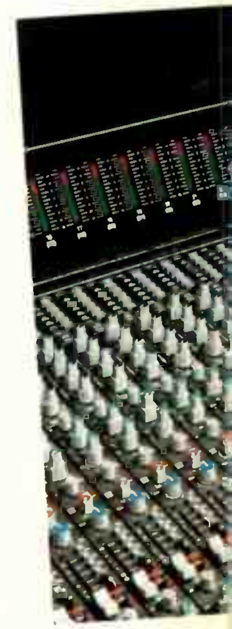
EQ PRODUCT VIEWS

COMPACT DISKS

You can cram a cornucopia of information into a single rack space with Pacific Coast Technologies' optical disk drives. Regardless of disk size, any two of their drives — such as the two 688 megabyte drives pictured — can be combined in one rack-mountable case. These systems are popular with Roland DM80, ProTools by Digidesign and all other Direct-to-Disk recording systems. Both drives feature a five-year warranty, thermal controlled fans, front-mounted address switches and access times as fast as 3.9 milliseconds. 1.0 gigabyte opticals, CD ROMs and DAT drives are also available. For further info contact Pacific Coast Technologies, 7940 Silverton Avenue, Suite 205, San Diego, CA 92126. Circle EQ free lit. #101

STRONG SENNHEISER

Sennheiser's MD 409 is a large diaphragm dynamic mic in a compact casing that is designed primarily for musical instruments. It is capable of handling extremely high SPL (sound pressure levels). It boasts a gritty low midrange and mild high frequency peak. The 409's low profile design makes it ideal for close miking of drums, for example, and still gives a large diaphragm sound. Its tight pick-up pattern provides low leakage which is important for studio as well as live applications. Due to its mid-band frequency response, the 409 is also suited for male vocals, and it gets "that Celestion sound" of guitar cabinets, according to its manufacturer. In addition, the 409 is ruggedly constructed for road use and comes with an unbreakable mic clip. The entire story on the MD409 can be obtained from Sennheiser Electronics Corporation, 6 Vista Drive, P.O. Box 987, Old Lyme, CT 06371; 203-434-9190. Circle EQ free lit. #102



POP (LESS) MUSIC

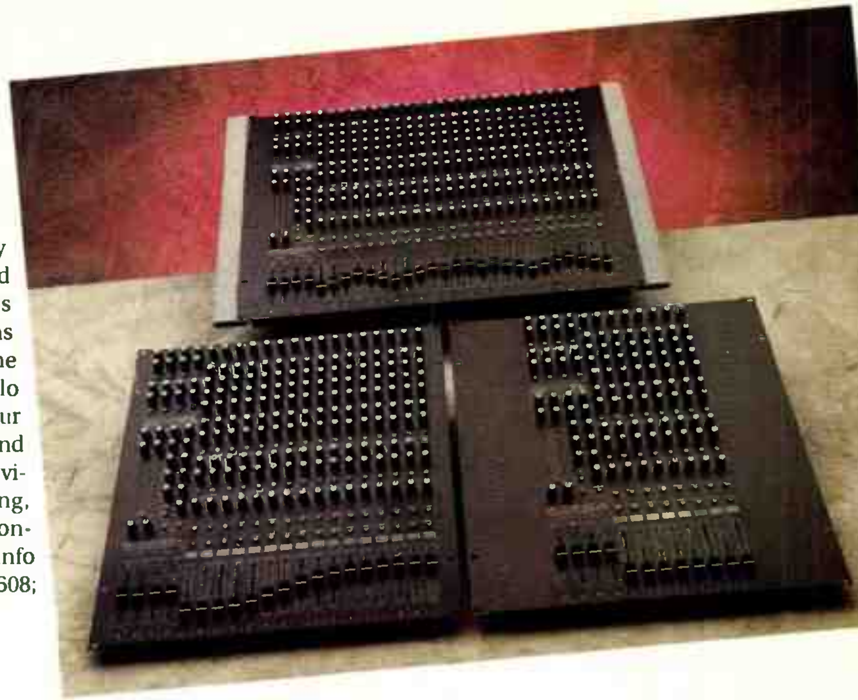
Poplesss Voice Screens (PVS) has introduced the Acoustic Compressor — a high-quality pop filter ideal for studio applications. The device employs two spaced layers of specially selected acoustic material, designed to greatly reduce popping and sibilance. The PVS-6wc and PVS-4wc (six- and four-inch diameter screens, respectively) clamp onto the mic stand and feature a versatile adjusting system that allows for easy and precise screen positioning. Models PVS-6c and PVS-4c offer gooseneck and mic stand clamp without the adjustable isolation mount. The PVS-6wc and 4wc sell for $89 each and the PVS-6c and PVS-4c sell for $63 each. For more information, contact Poplesss Voice Screens, 716 Pennington Street, Elizabeth, NJ 07202. Circle EQ free lit. #103

YOUNG EINSTEIN

When Einstein formulated his now famous hypothesis, $E=MC^2$, he may have been divulging the secrets to the newest Amek console designed specifically for the project studio market. In addition to offering up to 64 inputs (each with fader and 4 band EQ) and 24 balanced group outputs, this budget-wise mixing brain boasts comprehensive metering and monitoring facilities that are usually found on only the most expensive mixers. A new virtual dynamics option provides software-based gates, autopanner, compressors, limiters and expanders for each channel. Also, a single dynamic "device" can be assigned to each channel and set-up, adjusted and switched on and off in real time. All mix information generated is fully interchangeable with the Supertrue systems on Amek's Mozart and Hendrix consoles. For more information, contact Amek U.S. Operations, 10815 Burbank Blvd., North Hollywood, CA 91601. Circle EQ free lit. #104.

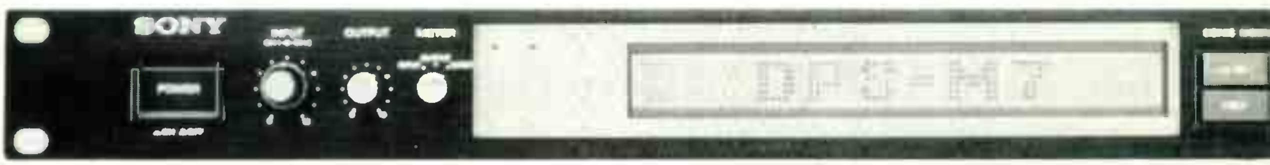
THREE FOR ALL

Symetrix has developed a new, affordable, dual-channel dynamics processor, the 425. The 425 ($579) features a downward expander, a compressor and a peak limiter, giving the user total control over all dynamic parameters of any audio signal. Multiple LED displays indicate exactly what each processing section is doing for maximum ease of use. Both the downward-expander and compressor have controls that adjust the release-time of the program-controlled attack/release functions to suit different source requirements. Finally, the two channels can operate independently or stereo-slaved and the 425 accommodates both balanced (XLR) and unbalanced (1/4") systems. For more information, contact Symetrix, Inc., 4211 24th Avenue West, Seattle, WA 98199. Circle EQ free lit. #105.



CATCH THE PHANTOM

Applied Research & Technology (A.R.T.) recently introduced the Phantom Series (1608, 2408, and 3208) consoles. The 2408 boasts 16 XLR channels plus eight additional line channels that can serve as dedicated tape returns with panning and soloing. The Phantom also features multi-function metering, solo and muting on every channel, four monitor sends, four post fader auxiliary sends, four master subgroups, and XLR outputs. It is configured to work in a recording environment with eight dedicated line returns, panning, assignable and separate outputs for a two-channel control room output or two-track group tape submix. For info contact A.R.T., 215 Tremont Street, Rochester, NY 14608; 716-436-2720. Circle EQ free lit. #106.



INTO THE DARC AGE

Denmark's t.c. electronic has introduced the M5000 Digital Audio Mainframe — a user expandable digital signal processor. The M5000 uses t.c.'s DARC™ (Digital Audio Reverb Co-processor) technology. Programs include reverb, ambience and pitch shift effects. The M5000 may be configured in the digital domain with up to four stereo channels of digital processing, all in a two-unit rack with AES/EBU, SPDIF and optional I/O. Standard interfaces include MIDI, RAM Card and SMPTE (in). For more info, contact Virtual Designs, Ltd., 717 Larkfield Road, Westlake Village, CA 91361; 805-373-1828. Circle EQ free lit. #107.

FULL EFFECT

Sony's DPS-M7 Digital Sonic Modulator, the latest addition to the DPS Series line of sound processors, offers a great deal of signal processing in a 1/2-rack sized package. The unit's fast 32-bit signal processor and advanced 20-bit H.D.L.C. ™(High Density Linear Converter) pulse D/A converters provide a wide array of advanced effects at a suggested retail of $1500. The DPS-M7 offers such effects as Haas effect panning, ensemble, and spiral modulation. Haas panning utilizes delay and phase to create a more natural panning. Ensemble can transform individual instruments into orchestras, and Spiral Modulation combines pitch shifting with revolving spatial effects to create the illusion of sound spiraling around the listener's head. Additional effects include Doppler, rotary speaker, chorus, pitch and reverse shifting, phaser, flanger, vibrato and tremolo, and ring modulation. For the rest of the story, contact Sony Corporation of America, 3 Paragon Drive, Montvale, NJ 07645; 201-358-4197. Circle EQ free lit. #108.

EQ TIPS

EDITED BY DAVE BRODY

BALANCED VOCAL BLEND OF BACKGROUND SINGERS

So, ya say the background singers are bitchin' at you because the level of their combined tracks in the cans changes each time you do an additional overdub? Here's how to fix it. Plug the output of the last device in your background vocal signal processing chain into a mult. Patch out of that mult into each track you'll be recording on. Set up your monitor gains, pans and cue sends for all those tracks. [In other words, make believe you've got all the background vocals recorded already.] When it's time to record the first pass, record on all the tracks (or at least run them all in "input" mode). For the next pass, drop the Record-Ready from the first track you recorded. Note that the overall level remains the same but the sonority changes. Next pass, "safe-out" both the first and second tracks...and so on. Using this method, you'll be giving the performers (and yourself) a much more accurate perspective of how their composite efforts will sound in the final mix; plus it's a whole lot less knob twisting for you! That should translate into a more efficient — and therefore more spontaneous — session. It's a real "vibes improver." Try this approach with guitars, brass or any other doubled (or multiply-recorded) source. It even works well with pairs of stereo sources (provided you have two independent mults).

TOAST THAT WHITEBREAD

You stayed up late last night fine-tuning the MIDI sequence for that guitar or lead synth part. Now, in the cold light of day, you're tracking it. But, much to your chagrin, it sounds like just what it is; a mostly lifeless and altogether too-clean waveform from a silicon circuit. And all the effects boxes in your rack only seem to magnify the problem. The answer: route it out to that old tube amplifier and mic that mold-encrusted speaker cabinet. The (shall we say) "non-linearity" of amp and speaker will impart a wide range of coloration choices for you to mess with — to say nothing of what you can do with your room mics.

VIVE LA DIFFERENCE!

And speaking of non-linearity, if you need a special vocal sound that really calls attention to itself, try two reasonably high quality condensor mics of different models or different manufacturers with their capsules placed as close as possible to each other and with their diaphragms aligned in the same plane. The idea is to overlap the pickup patterns of the two instruments to the extent they can be. Bring the signals of both mics up and match their levels as best as you can. Now for the magic. Reverse the phase on one of these mics (either on your console's input module, at the mic pre-amp, or by using a special well-labeled phase reversing cable). What you are now hearing (mostly) is the difference between your microphones. Certain frequencies — especially in the upper range — will be accentuated, yielding an unusual sound. As the performer moves, the effect will subtly change, giving you a kind of performance-related flanging effect (as the frequency spectra of the mics' waveforms constructively and destructively interfere). Raising or lowering the level of the "phase-flopped" mic gives you a means of fine-tuning the effect. [Needless to say, you should perform any signal processing thereafter on the combined ("bussed") feed.] Yes, several recent hit records have been cut this way. And, no, *EQ* magazine will not be awarding prizes for guessing which ones... EQ

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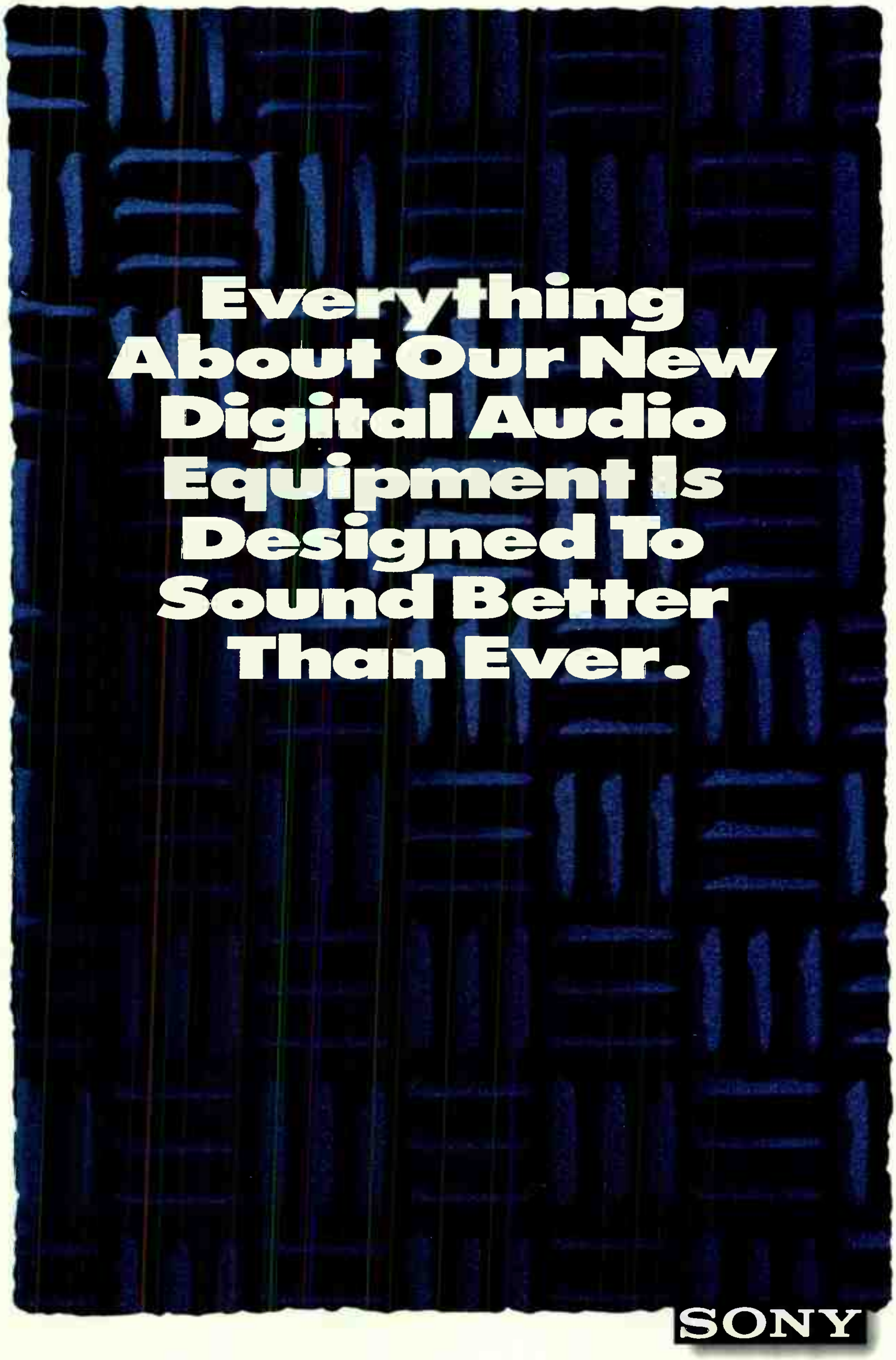
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SONY

TCD-D10 Pro II
Portable DAT Recorder

The TCD-D10 Pro II is the smallest professional DAT recorder from Sony. Yet, while it weighs only 4 lbs. 7 oz., the TCD-D10 Pro II is no lightweight when it comes to performance.

Built to withstand the rigorous demands of field work, the TCD-D10 Pro II allows you to stay in the digital domain from acquisition to studio. It also features absolute time (A-time) recording/playback which places a continuous time code on tape, allowing you to locate recorded segments faster and more easily.

Plus, A-time is compatible with SMPTE time code DAT recorders like our PCM-7000 Series. There's even an improved digital I/O and LCD multi-display with a combination of safety/warning indicators to help insure fail-safe operation. And when combined with one of Sony's high quality microphones, you're fully equipped to meet the most demanding challenges in the field.

KEY SPECIFICATIONS

DYNAMIC RANGE: MORE THAN 85 dB
FREQUENCY RESPONSE: 20 Hz-22 kHz
THD: 0.06%
I/O: ANALOG—MIC/LINE BAL, DIGITAL—AES/EBU
ACCESSORIES: BATTERY (X2), CHARGER, REMOTE, AC SUPPLY, CASE

▼ SHOWN WITH OPTIONAL SONY ECM-MS5 STEREO MICROPHONE.

PCM-2700
Studio DAT Recorder

Taking advantage of Sony's latest innovations in digital technology, the PCM-2700 is the first affordable professional 4-head DAT recorder.

Featuring Sony's advanced HDLC (High Density Linear Converter™) System, the PCM-2700 delivers superior sound quality. The PCM-2700 also employs a 4-motor direct drive transport to insure tape stability, accuracy and reliability. Its 4-head design provides off-tape monitoring to verify your recordings.

There's even a duration adjustable digital auto fader for fade-in and fade-out times as well as an A-time search function for rapid access to any recorded A-time location—all giving you the utmost in professional performance.

KEY SPECIFICATIONS

SIGNAL TO NOISE RATIO: MORE THAN 90 dB
FREQUENCY RESPONSE: 20 Hz-22 kHz
THD: <0.045%
ANALOG I/O: +4 dBs (+24 dBs MAX.) ADJUSTABLE
PARALLEL REMOTE: TTL COMPATIBLE, D-SUB 37

DPS-D7
Digital Hyper Delay

If you want to take your creativity in exciting new directions, Sony's DPS-D7 Digital Hyper Delay is the way to go.

Featuring seven sophisticated algorithms, there's virtually no limit to the number of unique and complex digital delay effects you can create. The DPS-D7 incorporates an 18-bit over-sampling A/D and 1-bit HDLC D/A converter system with digital filters for excellent linearity, ultra low noise and wide dynamic range.

Second generation LSI's allow for high-speed 32-bit digital signal processing. And with its large graphic display and help button for assistance on any function, the DPS-D7 is always simple to use.

KEY SPECIFICATIONS

DYNAMIC RANGE: MORE THAN 94 dB
FREQUENCY RESPONSE: 10 Hz-22 kHz
THD: <0.0035%
ANALOG I/O: BALANCED +4 dBs (+24 dBs MAX.), UNBALANCED −10 dBs (+10 dBs MAX.)
MEMORY CAPACITY: 100 FACTORY PRESETS, 256 USER LOCATIONS

DPS-R7
Digital Reverb

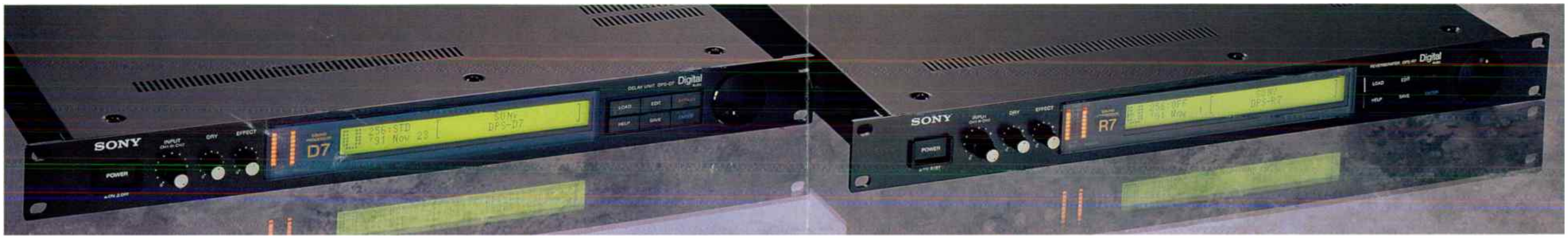
If you want to add even more power and versatility to your audio system, Sony's DPS-R7 is right on the money.

Offering two discreet channels of advanced digital reverb effects, the DPS-R7 is an invaluable tool for the audio professional. As with the DPS-D7, the DPS-R7 employs HDLC D/A converters for superior sound reproduction as well as high-speed 32-bit digital signal processing, which deliver sophisticated, multiple reverb effects.

It also includes 100 factory presets as well as 256 memory locations for your own presets. In addition, the DPS-R7 features an ingenious "data wheel" and large graphic display for easy operation.

KEY SPECIFICATIONS

DYNAMIC RANGE: MORE THAN 90 dB
FREQUENCY RESPONSE: 10 Hz-18 kHz
THD: <0.004%
ANALOG I/O: BALANCED +4 dBs (+24 dBs MAX.), UNBALANCED −10 dBs (+10 dBs MAX).
MAXIMUM SIMULTANEOUS EFFECTS (TEN): 4 PRE EFFECTS (2 PER CHANNEL), 2 REVERB (1 PER CHANNEL), 4 POST EFFECTS (2 PER CHANNEL)



PCM-2300
Studio DAT Recorder

As Sony's most affordable professional DAT recorder, the PCM-2300 is ideally suited for a wide variety of applications where high quality recording and playback are necessary.

Like the PCM-2700, the PCM-2300 incorporates the latest conversion devices—1-bit delta Σ A/D converter and HDLC 1-bit D/A converter—for outstanding sound quality. The PCM-2300 also incorporates a sophisticated 3-motor transport design for solid reliability. And in 32kHz long-play mode, it delivers twice the normal recording and playback time—a full four hours.

Plus, its analog and digital I/O's provide a wide range of flexible interfacing possiblities.

KEY SPECIFICATIONS

SIGNAL TO NOISE RATIO: MORE THAN 86 dB
FREQUENCY RESPONSE: 20 Hz-20 kHz
THD: <0.05%
ANALOG I/O: +4 dBs (+24 dBs MAX.) ADJUSTABLE
SUPPLIED ACCESSORIES: WIRED/WIRELESS REMOTE, 19" RACK MOUNT, POWER AND REMOTE CABLES

CDP-2700
Compact Disc Player

The CDP-2700 compact disc player delivers a multitude of professional features for a very compact price.

Like all Pro Standard equipment, the CDP-2700 is rugged and reliable while delivering superb sound quality. Ideal for on-air applications in radio broadcasting and sound sweetening in video post, the CDP-2700 includes important features such as variable speed playback, fader stop/start control from a mixing console and an auto cue function for instant start.

And because its digital output conforms to both the AES/EBU and IEC-958 formats, the CDP-2700 directly interfaces with other professional equipment for flexible system expandability.

KEY SPECIFICATIONS

DYNAMIC RANGE: MORE THAN 110 dB
CROSSTALK: 100 dB
THD: 0.04%
VARI-SPEED RANGE: ± 12.7% (0.1% STEPS)
D/A CONVERSION: DUAL 18-BIT 8X OVERSAMPLING

World Radio History

EQ&A

DIFFERENT DRUMMER

Q *Why does one need to assign multiple tracks to a drum kit or machine, and feed the toms, snare, etc., different types of reverb effects? Why not use a stereo drum mix and a global reverb, since in an actual performance the entire drum kit shares the same acoustical environment?*

John Lim
Pihang, Malaysia

A The simple answer to your question is that you only need to because you can. It's all a matter of creative choice. Early jazz, country and pop recording sessions created outstanding recordings doing exactly what you suggest. Modern rock, however, with its higher volume levels, has taken advantage of multiple tracks as a means to isolate and control the sound of each instrument. The technique provides producers with greater control of each individual drum (or other instruments) with regards to EQ, panning, isolation and effects. When you have complete control over each drum, you can then treat it individually in any manner you choose. You can, for example, create a signature sound, big-as-a-house snare drum without having to change the "size" of the other drums. Or you can build the illusion of a huge kit with effects and by spreading the drums across the entire stereo spectrum.

Your desire to recreate an actual performance has merit because it means you want the music to remain the most important element. However, modern music, such as rock, rap, dance, and metal, often depends on illusion. The music industry and its consumers have grown to expect larger-than-life recordings. If acts, engineers and producers want to play the game, they often must play (or get trapped) by the rules they developed. You'll have to decide which music genre and production style make you happiest. The availability of multiple tracks shouldn't be viewed in a negative light, but rather as a further means of advancing your creativity.

Hector G. La Torre
Executive Director

TAKE IT TO THE LIMIT

Q *I usually keep the faders on my Tascam M-3500 console at 0 dB when working, but lately whenever I raise two or more faders up past the 0 dB point I hear a low-frequency noise. What's wrong?*

Scott Sanders
Gary, IL

A One of the features of the M-3500, and all Tascam mixers, is that the input channels have up to 10 dB of gain. When you move a fader beyond the 0 dB point you are increasing not only signal level, but also any noise that could be found in the signal. If the noise you are hearing is on all the channels, regardless of the input source, there could be a ground loop problem in the studio. However, without more information it is very difficult to determine where the noise is coming from.

Ken Hirata
Marketing Communications Manager
Tascam

OHM SWEET OHM

Q *Why do my professional headphones sound so wimpy whenever I use them in a cassette multitrack unit?*

Dave Sanderson
Elmont, NY

A Your "pro" headphones are probably rated at 600 ohms impedance, while virtually all cassette multitrack recorders have 8-ohm impedance headphone outputs. Impedance refers to the amount of electrical resistance or "push" at which a circuit is designed to operate. Therefore, an 8-ohm output has its level severely reduced when attached to 600-ohm headphones because the resistance is too great. Conversely, you risk frying an 8-ohm headphone by connecting it to a 600-ohm headphone amp. Too little resistance equals too much power, and so long headphones.

Jimmy Yamagishi, Bill Stevens
Tascam

DIODE DILEMA

Q *My Dokorder 7140 open-reel 4-track runs fine for five to eight minutes and then the solenoid that pulls the pinch roller into contact with the capstan starts kicking out, disengaging the pinch roller. Thinking the problem might be due to the solenoid becoming magnetized, I use a bulk demagnetizer on it and it works fine for another 10 minutes. Could the diode across the contacts of the coil be causing the problem? Would replacing the solenoid fix it? Help!*

Ron Carlson
Ogden, IA

A While I don't have a schematic, I have a few tips on how to solve your problem. If the problem were a magnetized plunger (the moving part of the solenoid), the plunger would not release. The diode across the solenoid is supposed to suppress the "kick back" voltage that normally occurs when a solenoid is de-energized. This protects either the transistor or the switch which applies power to the solenoid.

Typically, a high voltage pulls the solenoid plunger in; then a lower voltage is all that's required to hold it. These voltages may be 24 and 12 volts,

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
Fax: 516-767-1745

respectively. Check the voltage across the solenoid during the time the problem occurs. The duration of the high voltage will be a couple of seconds. Since the plunger is pulling in, it would appear that the voltages are good. It seems to me that the solenoid is not physically properly positioned. As the solenoid coil heats up, the resistance goes down, as does its ability to hold.

The plunger must be fully seated upon initialization of the Play mode. If not, adjust the solenoid position. It may be necessary to enter the Play mode several times from Stop until the solenoid is properly adjusted. Keep in mind that too little tension will cause speed problems.

Now, unless the pinch roller pressure spring is fatigued, there should be enough pressure. It's possible that in an attempt to get more pressure (possibly to correct a tape path or speed problem) someone replaced the spring with one that may be too heavy.

Ed Ciletti
Manhattan Sound Technicians
New York, NY

WRONG NUMBER?

Q *I control my Fostex R8 multitrack from my computer and MIDI keyboard. Occasionally, the R8's control panel stops functioning. The only way to "correct" this is by switching the deck off and then on again. Why does this happen? Also, what are the different SMPTE timecode formats used for?*

Eduardo Panza
Seville, Spain

A It sounds as if you're sending a "local off" MIDI message destined for your keyboard that's also being picked up by the R8/MTC-1. While "local" control separates your keyboard from the sound-generating circuitry, thus preventing MIDI feedback loops in a sequencing environment, it also prohibits front panel access to the R8. The remedy is either to keep your MIDI channels separate or use a different MIDI port for the R8.

The different timecodes are used in the following capacities: 24 fps (frames per second) is used for motion picture films; 25 fps is used in European TV and video; 29.97 fps is for American color TV and video; 30 fps is used in music recording and American black & white TV and video [NTSC standard].

Roger Maycock
Product Specialist
Fostex

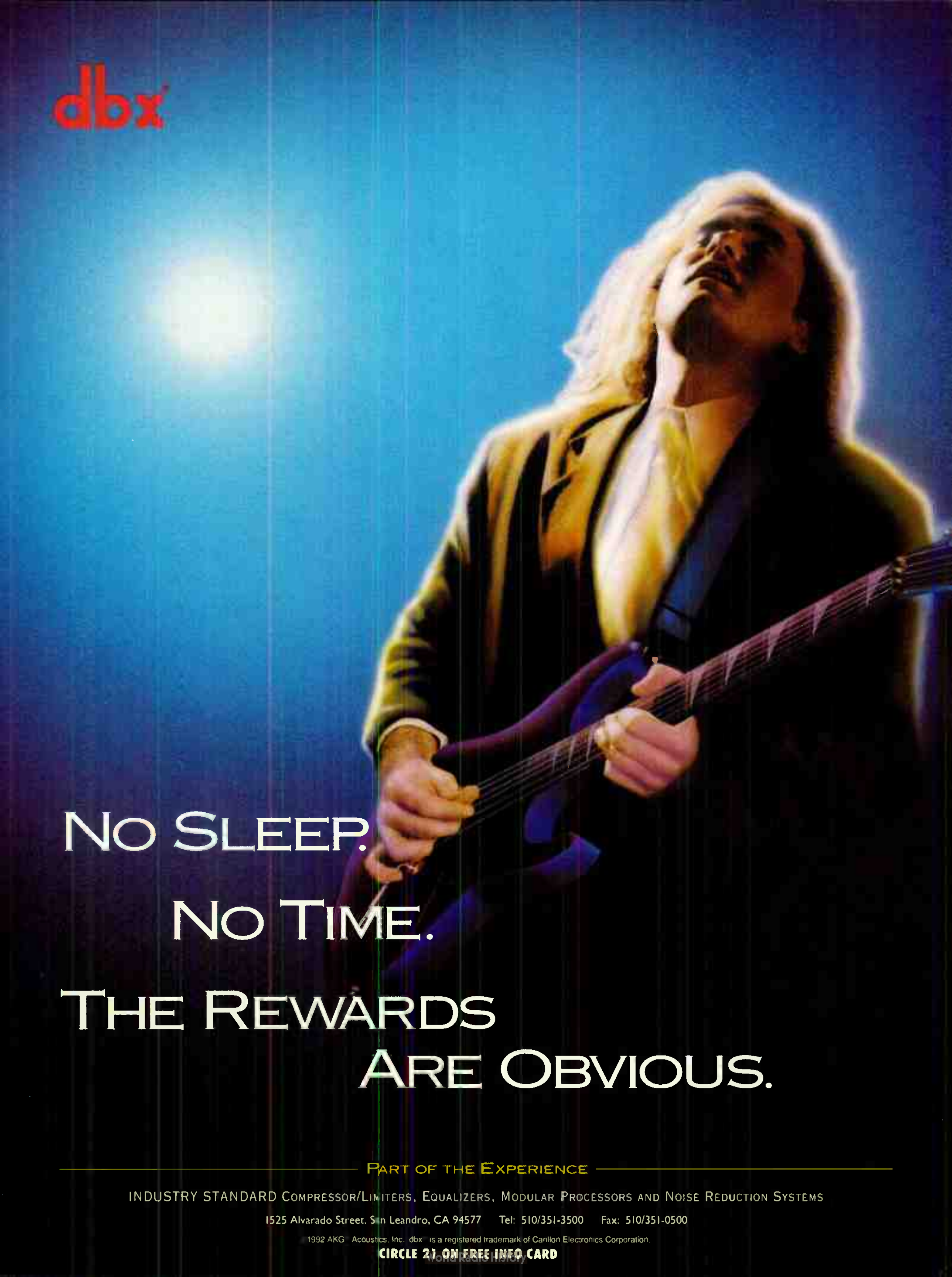
FADE(R)ING FAST

Q *I am a recording engineer looking to add automation to a Tascam M-3500/32 console in our 24-track studio. Could you make some viable (i.e., not too expensive) suggestions?*

Peron Rarez
Brazil

A Many people have asked about adding automation to the M-3500. Essentially they're asking, "Can I turn my M-3500 into an M-3700?" Unfortunately, because of great internal differences, the M-3500 cannot be modified to be an M-3700. However, there are quite a few VCA automation packages

continued on page 82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Mackie Attackie

Tom Mgrdichian puts together a system the quality of a Rolls...

MAIN MAN: Tom Mgrdichian
TO HIS CREDIT: Films/TV: *Die Hard II, Friday The 13th Part VIII, ABC Into The Night, Totally Hidden Video, Night Court, Life Goes On, Married People, Shattered Dreams, The Fisher King.* Recordings/Live Performances: Olivia Newton-John, Air Supply, Debbie Gibson, Ronnie Milsap, Englebert Humperdink, Seals & Crofts, Richie Havens, many others.
RACK 1: Voce DMI-64 MarkII, E-mu Proteus 2, E-mu Proteus 1XR with InVision Protologic card, two Yamaha TX802, Korg M1R, Roland R8M, Roland MKS-20, Yamaha TX7, E-mu Proformance+, Roland D550, and two Mark of the Unicorn MIDI Time Pieces.
RACK 2: Four Mackie 1604 mixers, two Mackie Mixer Mixers, two Symetrix SX-201 parametric EQs.
RACK 3: Dynatek 44 Mb removable drive/CD ROM combo (presently upgrading to a Dynatek 520 Mb fixed, 600 Mb Magneto Optical, 1.2 gig DAT back-up and CD ROM), Macintosh IIci with two Digidesign Sample Cells and Sound Tools DSP card, Crest FA 901 power amp, Sound Tools DAT I/O, Aphex 612 Expander/Gate, Yamaha Rev 5, Lexicon LXP 15, two Yamaha SPX 90IIs, Tascam CD401, Panasonic SV3700 DAT, Aiwa AD-F800, Symetrix SX205 Meters, two Aphex 10/4 boxes and a TT patchbay.
NON-RACKED GEAR: Roland A-80, Roland JD-800, Memory Moog+, Prophet V and Yamaha NS-10M near field monitors.
EQUIPMENT NOTES: I can't praise the Mackies enough. In my opinion they blow away all the competition anywhere near their price range. With the four Mackies, I have 64 faders. They are quiet and they don't color the sound. They also have tons of headroom. The EQ is clean but a bit limiting, although Mackie will be coming out with a module that will provide more flexible sweepable EQ per channel in addition to a MIDI automation upgrade for the 1604.

The Mackies have a mute switch on every channel that has a really cool feature, which I use quite frequently. When muting a channel, it assigns that channel to another pair of outputs. So it gives you an additional set of outputs rather than just a single stereo bus. And, by the way, you've *got* to get hip to Dynatek drives. I can honestly say I've never had any problems with them. They can take the abuse of being transported frequently in addition to being extremely reliable and quiet.

My whole system is wired with ELCO connectors. Each connector, mounted on the back of the racks, has 90 pins in it so I don't have all these patch cords everywhere. I also have a TT patch bay. The bay makes things easier for engineers because I can just push the rack up against the patch bays of the recording consoles and go right into the input of the tape machines.
WHAT WORKS FOR HIM: I sample a lot of my own sounds. For instance, I'll take the Proteuses, Sample Cells, TX802, D550, Memory Moog+ and the Prophet to make one great brass sound. Then I take all of those sounds and send them, via the Mute, through the extra third and fourth buses on the Mackies to the SV-3700 DAT. The SV-3700 converts the signal to digital. I then send that into Sound Tools and sample it. I monitor everything by bringing Sound Tools back through the Mackies on the main stereo bus. So while I sample, I'm monitoring the sampled output — not the input. In doing pre-production for records (and song demos) we'll use the Mackies for all the keyboards running virtual tracks while mixing in recorded guitars and vocals from an Akai 1214. The keyboards and recorded tracks go down live to DAT locked via SMPTE timecode.
PRO TIPS: Think long term when making your equipment purchases. Experiment to find the best combination of gear that will suit your present, and more importantly, future production needs. Remember, you don't have to buy the most expensive or esoteric product on the market to make great recordings. EQ

Photo by Ed Colver

Major League Mixer

Baseball slugger Warren Cromartie trades in the bat and glove for a new console and a room with a VU

STUDIO: High Five Studios, North Miami, Florida.
MAIN MEN: Warren Cromartie (owner/writer/producer), Eric Schilling (chief engineer).
PRODUCER AND ARTISTS CREDITS: "Take A Chance" (Toshiba/EMI), a collection of rock songs Cromartie wrote as a baseball player for the Expos and, later in his career, The Tokyo Giants. He currently writes and produces songs that are heavily influenced by the vibes of world music, reggae and Brazilian rhythms. One of his new tunes features lead vocals by Geddy Lee, the frontman for Rush.
CONSOLE: 56-input Amek Mozart with Supertrue Automation.
RECORDERS: Otari MTR90-II 24, Otari MTR20 2/4 (14-inch reel capability).
DAT: Sony DTC1000, two Panasonic SV-3700s.
MONITORS: Meyer 833, Westlake BBSM6.
OUTBOARD GEAR: Eventide H3000SE, Focusrite ISA115, Sontec Parametric, Lexicon 480L, Eventide 2016, three Yamaha Rev 5s, two Drawmer DS201s, Aphex Compellor, BSS DPR402, two dbx 165As, Valley Dynamite, two Lexicon PCM 42 w/MEOs, two UREI 1176LNs, Pultec Mavic Tube Mic Pre.
KEYBOARDS: Yamaha DX7, Emulator EMAX II.
SEQUENCER/COMPUTER: Macintosh IIfx w/Opcode Vision.
MICS: Neumann U87, Sennheiser 421/441, Shure SM57/58, AKG D12-414-451-460, Countryman Isomax.
SYNCHRONIZER: TimeLine Micro Lynx.
EQUIPMENT NOTES: The Amek Mozart console perfectly suited my need for a fully-functional, easy-to-use board. Besides delivering crisp, clear sound, the Mozart provides 24 tracks, 56 inputs and fabulous automation. When I'm looking for that clean world sound, I score big-time on the Big Mo'. As far as keyboards go, the Yamaha DX-7 and Emulator EMAX II provide some excellent vintage sounds. The Emulator has some incredibly beautiful ethnic creations, including an African flute that'll really soothe the senses.

My initial songwriting is done on the Macintosh with Opcode Vision. It's versatile enough to handle everything from the studio budget to my son's homework while still serving as a vital songwriting tool. Basically, the Mac serves as an engineer, saving lyrical and musical notes until we reach the second phase of the production process.
PRODUCTION NOTES: Baseball is the fire of my music and music is the fire of my baseball! Although I love the game of baseball, I'm basically a percussionist at heart. As I see it, whether you have one stick in your hand or two, it's all a matter of style and rythm (or polyrythm, as the case may be). Throughout my many years in the big leagues, I used to write music and envision beats incessantly. Back then rock and roll was my musical taste of choice. However, it was during my tenure with the Tokyo Giants in Japan that my musical tastes really began to shift, and my senses became infiltrated by the sounds of African, Brazilian and world music.

When I built High Five Studios, I set up an isolation booth to record all the vocals, drums and horns in a live environment. To me, world music is best captured in a live-band setting, a setting which vibrantly captures the unique feel of the multi-cultural rythms. By treating myself to some top-notch equipment, and staying true to the principles of my sound philosophy, I've been able to live out another one of my life-long dreams and own my own studio.
PREDICTIONS: The Twins take it in '92. EQ

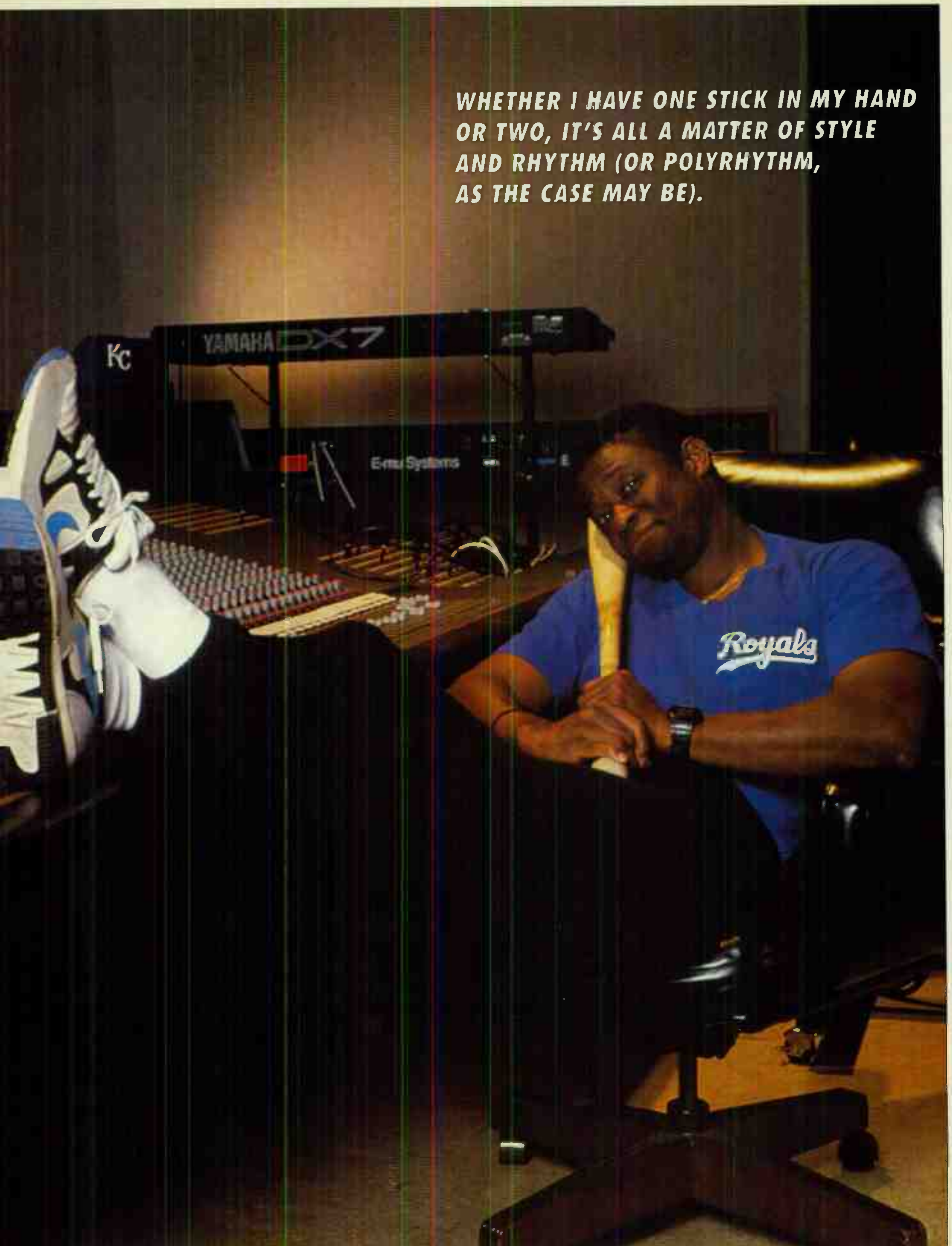

Photo by Deborah Gray Mitchell

MIDI 2.0 Is Here!

10 years after its introduction and more popular than ever, an updated (evolutionary) MIDI enters the scene

BY CRAIG ANDERTON

Photo by Ed Aiona

Things are really starting to get out of hand. It was improbable in 1983 that manufacturers from competing companies and cultures could get together and agree on a specification. It was miraculous that companies actually adopted the spec and created gear that all worked together. And it's almost beyond belief that MIDI is getting rediscovered as something seemingly brand new by a generation of multimedia PC enthusiasts.

But what really boggles the mind is that, almost 10 years after MIDI was introduced, it is gaining momentum — not losing it. This is a testimony to those who conceived of the original spec and wisely left room for extensions; but it's also a testimony to an industry that has kept the trust of the music buying public by maintaining downward compatibility between new and old MIDI gear.

Changes to the spec have been made over the years in an orderly, deliberate fashion via the MMA (MIDI Manufacturers Association), which always seems to be headed up at any given moment by someone hip like Jeff Rona or Chris Meyer. The IMA Bulletin, edited by Lachlan Westfall, disseminates information to the public and also serves as a forum for MIDI ideas. Credit too must also go to PAN, the Performing Artists Network BBS by Perry Leopold, which links much of the MIDI community together via electronic mail.

EXTENSION CORDS

And now, just a few months before the eve of MIDI's first decade, we have "MIDI 2.0" to celebrate. Well, not really; it's still called MIDI 1.0. But in truth, the only similarity between the MIDI of 1992 and 1983 is that instruments built in 1983 still work with the 1992 stuff — that's about it.

There are so many extensions to the MIDI spec that if MIDI was under the aegis of someone's marketing department, it would probably be up to MIDI 5.0 by now. All the people who have been asking for "MIDI 2.0" essentially have it — something with vastly more applications, speed, and useability than anyone ever dreamed possible back in the days of MIDI's creation. We'll get into the goodies a bit later, but first...

HIT AND MYTH

MIDI has been a big hit, but it has also been the subject of nasty rumors (i.e., talk of a delay through the MIDI thru port — *not*). And a lot of people still say that "MIDI's too slow." But the fact is that it's the sound generating equipment, more than MIDI, that causes perceptible delays. I've been using Sound Tools to test the timing delays of various pieces of gear, and found typical delays to be in the 3 ms to 10 ms range to process a single MIDI note-on command, which only lasts 1 ms. Therefore, with a 6 ms delay (about average), MIDI is only accounting for 16 percent of the delay; the rest is the fault of the synthesizer. The more notes you play, the more the delays add up.

Manufacturers are doing what they can to speed up synth responses (matters are improving; the slowest synths often turn out to be the oldest ones). But there are certain practical and financial limits. You could double the speed of an instrument — not a trivial undertaking — and with a typical instrument MIDI would still only account for less than a third of the delay. So, let's not worry about whether MIDI's fast enough until we have instruments that can handle the higher speed.

Not enough channels? True, the original MIDI spec only had 16 channels. But there are so many ways to add channels — from syncing sequencers together to using multiport computer interfaces — that there really isn't any need to run out of channels either.

For those who truly need more than just higher speeds and lots of channels, Lone Wolf has produced a LAN chip that allows MIDI gear to be part of a high-speed local area network. The transfer rate of MIDI data becomes a bottleneck only at the server level, which translates the high-speed network data into something individual pieces of MIDI gear can handle.

The only other real complaint about MIDI is that it still does not allow for bi-directional communication (except in very primitive ways). However, a single line of communication can provide enough that we probably shouldn't get too upset.

THE WAY WE WERE

It's hard to believe, but many things we take for granted nowadays were not in the original MIDI spec.

• *Standard MIDI Files.* Being able to transfer MIDI files between different computer platforms via disk or modem turned out to be a breakthrough. Although implemented in a lot of gear, unfortunately not all elements of the spec have been embraced. For example, SMFs include provisions to standardize patch data into a standard file format but this part is woefully underused.

THESE CONSOLES SO MANY FEATUR COULDN'T FIT THE ALL ON THIS PAGE

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SOLO MIDI RECORDING. Available in 16, 24 and 32 input frame sizes. Automated MIDI Muting on all channel inputs, monitor inputs, group outputs, stereo effect returns and auxiliary masters. Four band EQ with two swept Mids, assignable to monitor inputs. Six auxiliary sends—four assignable to monitor inputs. Four stereo effect returns with two band EQ, balance and level controls. Raised meter bridge.

We wanted to list *all* of the features on SOLO consoles but we ran out of space. If you want to find out more about eve

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CIRCLE 54 ON FREE INFO CARD

• *The Sample Dump Standard.* It's slower than molasses, but at least you can get digital audio data interchanged between various samplers. Some people complain because only the samples and loop points are transferred back and forth, not the signal processing parameters (filters, envelopes, etc.; this does a sort of automatic copy protection since people can't take a polished sound). But considering the chaos in digital audio standards, I'm just as glad there's at least some way we can get the really important stuff through. Although not all samplers implement this, it has had a reasonable impact.

• *MIDI Time Code.* All professional Macintosh sequencers rely on MTC for synchronization, and instruments like the Emulator III let you create cue lists triggered by MTC timing. A boon to video and audio-for-video people, MTC was slow to take off but is now everywhere.

There have also been numerous other small enhancements. Recently, though, there have been two more biggies.

THE WAY WE ARE

• *MIDI Machine Control.* MMC is the culmination of a quest to merge MIDI sequencing and audio tape so that tape recorders become, essentially, MIDI system peripherals. MIDI Machine Control got off the ground at the October 1987 AES convention, largely through the work of Gerry Lester (formerly with Adams-Smith) but with extensive input from other industry people. Lester revised the proposal five times himself, but TEAC's Shoji Fujiwara, chairman of the Japan MIDI Standards Committee working group for the MMC, revised the spec from 1989 to 1991. MIDI Machine Control has the potential to extend MIDI's reach into the project and pro studio even further than it is now. Also of note: it's video/CD-ROM/multimedia-ready.

• *MIDI Show Control.* Originally proposed by Charlie Richmond (of Richmond Sound Designs, Vancouver, BC), MSC uses MIDI messages to control theatrical, live performance, and similar equipment — lights, strobes, lasers, CD players, rigging, special audio effects, animation, video switchers, projectors, and even such things as compressed air, natural gas, hydraulic oil, pyrotechnics, etc. Other manufacturers know a good thing when they see one and have jumped on the bandwagon, so it appears that MSC is going to be another popular MIDI extension.

• *General MIDI.* Roland and Passport worked hard to get this one across, because it will be difficult for consumer/multimedia MIDI to take off unless people composing MIDI "clip music" have some assurance about what the sequence will be playing back through. Among other things, General MIDI (Level 1) correlates specific sounds to specific program numbers, creates a standard drum note map, and defines a GM instrument as having at least 24 dynamically allocatable voices, 16 MIDI channel responses, drum/percussion sounds on channel 10, and at least 128 programs.

And that's still not all. Some of the other recently-adopted or about to-be-adopted goodies include:

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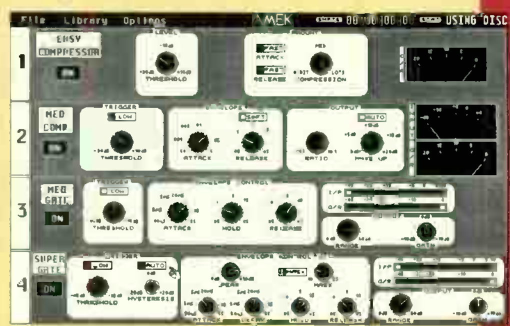
A decade ago, a revolution in console design was made by building Dynamics control into the hardware of the channel strip. Ten years later, developments in console technology by AMEK now give you Dynamics control from the computer screen – Virtual Dynamics.

Unfettered by the constraints of hardware, we give you a choice of Dynamics controllers. These software templates, complete with virtual knobs, switches and meters, perform all of the required control functions. Easy, Advanced, Dual Slope and Stereo Compressors; Easy, Medium, Sound-Shaper and Stereo Gates Autopanner, Limiter and expander are all included. For ease of use up to 128 user-defined configurations may be stored in the library. All these devices occupy no rack space, use no additional VCAs, and when you save the mix, the Dynamics device you selected for each channel is saved with the mix data.



In addition to the Dynamics the system package provides a MIDI in and out for every 8 channels, leading to new possibilities in MIDI gating and control. Furthermore, each Virtual Dynamics unit can be switched in or out of circuit against time code via the automation master Cue List.

AMEK/Steinberg SUPERTRUE automation provided a new level of sophistication in automated mixing. The first major enhancement was a complete machine control system, Superloc. Now Virtual Dynamics is an even more revolutionary step forward which brings the possibilities of console automation into a new era

AMEK Virtual Dynamics is available on SUPERTRUE automation systems fitted to AMEK MOZART and HENDRIX consoles.

AMEK

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• *All Sound Off.* Unlike All Notes Off, this command also sets envelopes to 0 as rapidly as possible.

• *Sound Controllers.* Controllers 70-79 are defined as "sound controllers" that affect the overall timbre of a sound. There are already five recommended defaults: 70 creates sound variations, 71 changes harmonic content, 72 edits release time, 73 attack time, and 74 brightness.

• *Legato footswitch.* This turns legato mono response on or off, which has several uses but would be of particular interest to guitar synthesists who want to rapidly switch a sound module to match mono or poly mode string output.

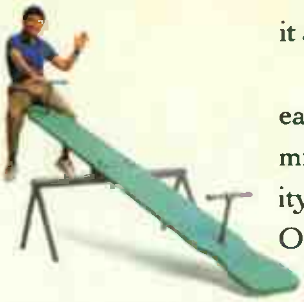
• *Bar Marker and Time Signature.* This Universal Real Time sys ex message indicates that the next clock received indicates a new bar. The time signature message can either take effect with the next clock or next bar marker.

• *Tuning Standard.* Not only does this define alternate tuning tables that spell out the exact pitches for each MIDI note, but also allows for on-the-fly retuning of individual notes to allow for simplified modulation without having to load new tables.

• *File Dump.* The biggest problem with Standard MIDI Files is that they work when transferring files via disk or modem, but can't be transmitted over the MIDI line. This extension fixes that problem by providing a way to get SMFs, as well as other MIDI file types, into a format that can travel down the MIDI cable. Finally, we can get sequences from one place to another without having to play them in real time and record them!

Is this the end of the line? Absolutely not. It's just the tip of the iceberg, and we might have additions like SMDI (SCSI Musical Data Interchange, for high-speed SCSI sample transfers) before too long.

MIDI's vitality is nothing short of remarkable in today's fast-paced world; it seems only fitting that musicians, who are often known for not fitting in with the "real world," went ahead and created their own standard world that works just fine. Kinda makes you wonder if the MIDI Manufacturers Association people should become an elite political negotiating team should MIDI ever go out of style. EQ



They say you can't have it all. Don't believe it.

On one side, you need an easy-to-operate powered mixer with great sound quality and just the right features. On the other side, you need it at a reasonable price. It's the proverbial balancing act.

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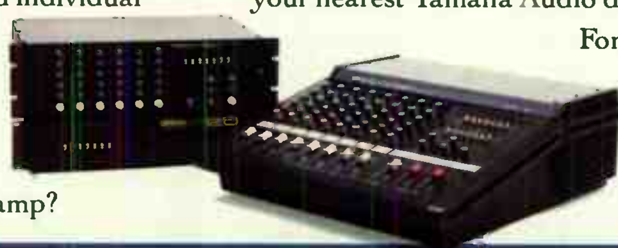
You only need a stage monitor amp? The Yamaha P120 has the same power amplifier and graphic equalization you find in the mixers. It's the easy way to power stage monitors. And as your needs and budget grow, it's an ideal addition to an EM2820 or EM1620 system.

All three of the units have a built-in limiter to protect the power amp from being over driven. They have forced-air cooling for reliability under the most extreme conditions. And both the P120 and EM1620 are rack mountable in Yamaha's RK319 rack case or any 19-inch equipment rack.

Contact Yamaha at 1-800-937-7171, ext. 300 for more information about the perfectly-balanced EM2820, EM1620 and P120. And the location of your nearest Yamaha Audio dealer.



Forget conventional wisdom. Remember to call.

YAMAHA®

According to conventional wisdom, these should cost more. Or sound cheap.

CIRCLE 64 ON FREE INFO CARD

Ultimate DAT Handlers Handbook

This article's seed was planted when a well known New York studio supplied me with DAT tapes to master that were such a mess I actually complained about it. (I have the best job in the world so I don't complain much!) The owner of the studio called me and asked if I would be interested in speaking to all their assistants some morning about how to properly prepare a DAT tape. I did and this was what we talked about:

Are you entrusted with the studio DAT machines? Then read on!

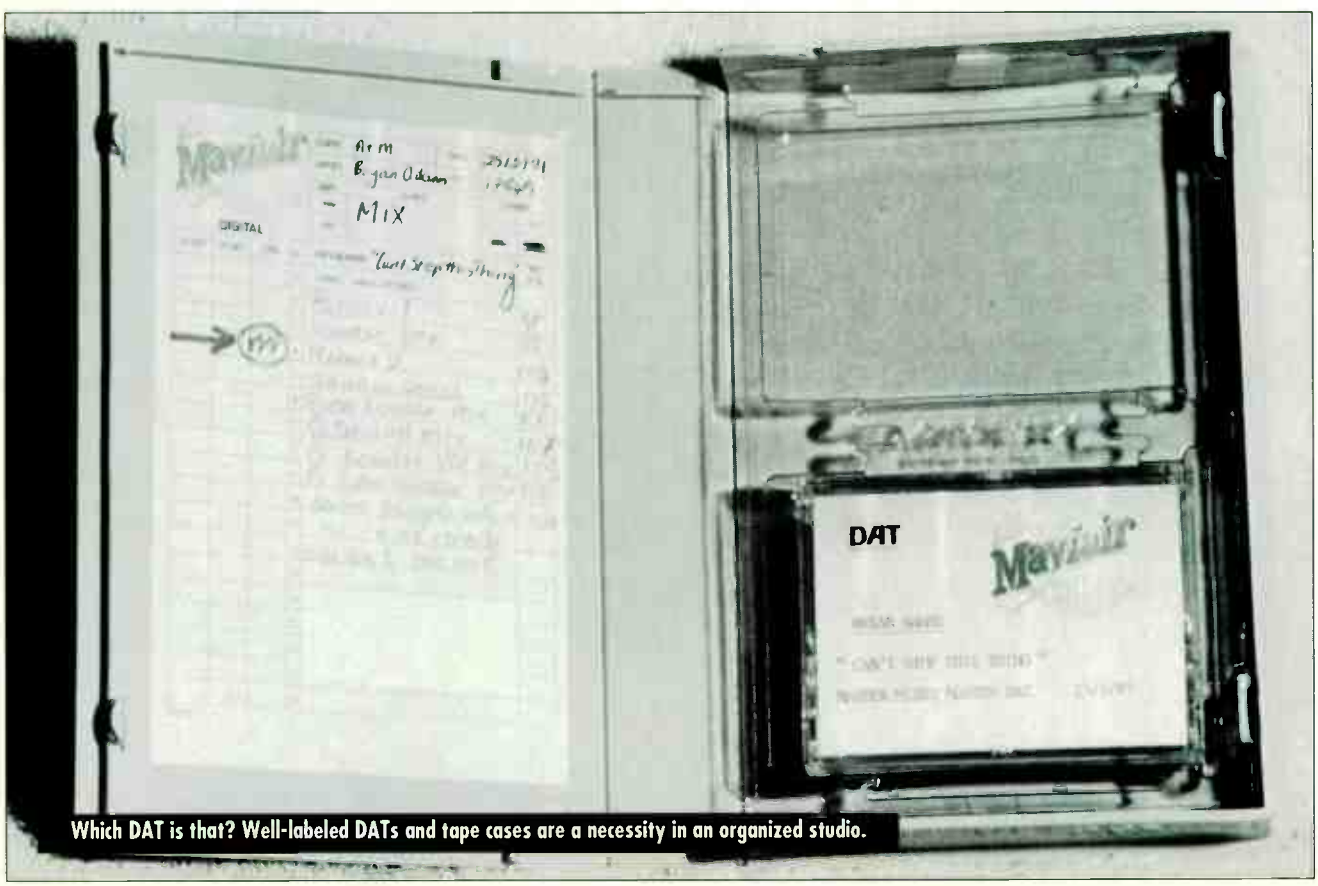
BY BOB LUDWIG

The first rule is to be consistent! Obviously if Mr. or Ms. four-figures-a-mix engineer asks you to make a DAT go backwards, start trying to do it. But there are certain procedures that you should follow on your own initiative.

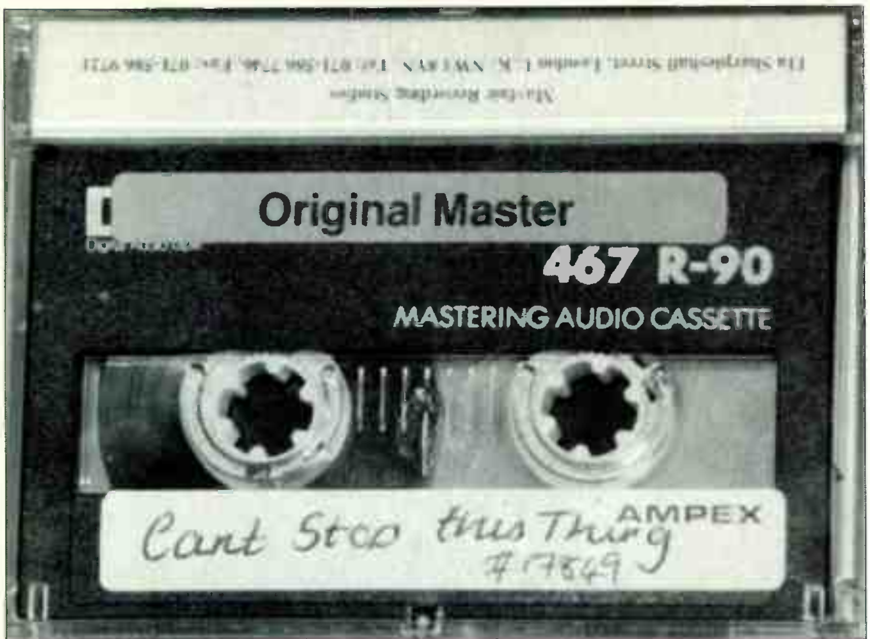
If you can, record all the session DATs at the same sampling frequency. Having some at 48 kHz and others at 44.1 kHz can really be a pain. The DATs should be recorded at 44.1 kHz if possible. The reason for this is that CDs, and soon the MC (Mini Disc) and the DCC digital cassette, are all sampling at 44.1 kHz. To record at 48 kHz will, in fact, enable one to record up to almost 24,000 Hz and might sound better to some golden ears than a 44.1 kHz tape, but it will either have to be transferred via the analog world or through a sample frequency converter to 44.1 kHz. After this mess it most likely won't sound as good as recording at 44.1 to begin with. Sample frequency converters do not clone (make perfect copies of the original) but employ digital filters and sometimes very nasty sounding algorithms to accomplish their task. Some sampling frequency converters sound quite excellent, some quite horrible. In this case, some cheaper ones sound better than others costing almost four times as much! So avoid this problem by recording at 44.1 kHz in the first place.

GETTING TO KNOW YOU

The next rule is to be familiar with the DAT machine you are using. Do you



Which DAT is that? Well-labeled DATs and tape cases are a necessity in an organized studio.



Always pretend that all the DATs have been taken out of their cases and lumped together. Could you tell which was which from the DAT alone? You must be able to do this!

know what DAT PACs (areas of subcode information on the DAT tape) it supports, its output level, metering, the quality of sound from its converters etc.? If your studio cares about creating the most accurate sounding DATs, buy a dedicated analog-to-digital converter. My current A-to-D favorites are the Wadia WA 4000 Pro from Wisconsin and the Pygmy made in Florida. Does it surprise you that the little op amp D/A of a $1,400 DAT machine may not sound as good as a $4,000 dedicated analog-to-digital converter? While the average DAT machine sounds a heck of a lot better than even the most costly professional gear of a few years ago, there have been dramatic advances in the art of analog-to-digital and digital-to-analog conversions in the past few months. Discrete analog amplifier stages feeding true 20-bit converters with noise shaped dither going to 16-bits and reproduced through D/As with huge power supplies feeding true Class A output amplifiers yield digital sound that make A-B comparisons futile even on high resolution monitoring systems. It has come a long way. Now it is "perfect sound forever" (just kidding!!).

IF YOU'VE GOT THE TIME...

Can your machine record timecode? Truly professional DAT machines can record SMPTE timecode on the subcode part of the tape. Machines like the Sony 7000 series can record SMPTE or EBU timecode in any flavor, starting at any timecode number you like. Many machines record Absolute time which is a reference time that begins only at 0 minutes 0 seconds (frames are not supported) and increments up as the tape is recorded. This timecode must begin at the very top of the tape for it to work.

Other machines only show a counter. If you are making reference DATs for a producer it may not matter much what kind of time the machine can record, but if the tape is going to be used on a DAT editor or if it is going to a mastering facility please use a machine that records SMPTE timecode or at least one that records Absolute time. It can add hours to a mastering or editing session to stripe the code on afterwards.

Machines like the Sony 7050 can read Absolute time and output it as though it were SMPTE, making life very easy for the editors and automated digital consoles. Please be sure that the Absolute time or SMPTE is continuous with no breaks in the time. Some machines with Absolute time are tricky in this regard. When you continue recording on a tape that you have started, be sure that you see good time displayed on the machine's read-out. If you have even a one second gap in the code, the Absolute time will no longer record.

One weird thing: The Panasonic SV-3500 DAT machines will read out as "Absolute Time" on a Sony 7000 series machine but the resulting "SMPTE" the machine tries to generate is not pure — it gets slightly messed up every 1.5 seconds. On the other hand, the Absolute time created on the Panasonic SV-3700 will read out as "SMPTE" on a Sony 7000 series and the generated "SMPTE" output is fine!

Panasonic explains that the SV-3500 writes only "A" Time and, as a result, the Sony 7000 series translation of the "A" time to SMPTE data is not pure, resulting in its every 1.5-seconds of inconsistency. The DAT machine's internal frame rate is 100/3 or 33.333, the SMPTE 29.97 or 30. The likelihood of any frame start coinciding with these radically different frame rates is thus very small. The SV-3700 and SV-3900 DATs write both "A" Time and "Pro-R-Time" which is read as SMPTE timecode, to the sub data areas. Pro-R-Time carries a piece of data specifically to give the offset between each "A" time frame start and the actual Pro-R-Time frame start. Although the Panasonic DATs are not able to access the Pro-R-Time/SMPTE timecode they write, this function was added to these machines to allow use of tapes recorded on these machines on full timecode DATs. The SMPTE time written to tape on the Panasonic DATs starts at 0:00:00:00 and records for the time of the tape.

One last item here — be sure that you have at least 10 seconds of servo-locked timecode (Absolute or SMPTE) before the beginning of a master take. The digital editors need this time for pre-roll. This also applies to anything regarding timecode and some situations will require more than 10 seconds...but at least that for DAT.

ESSENTIAL DATA

Does your DAT machine record with pre-emphasis? If it does you may not want to use it if it can be switched off. For pop recording with bright, closely miked cymbals and sibilant vocalists, the high frequency boost employed by the pre-emphasis curve will make the average level of the master recording drop in order to keep the high peaks from over-recording. Use 44.1 kHz; don't use pre-emphasis!

Are the meters on the DAT machine you are using true digital meters or are they analog-driven with perhaps only the "Over" indication truly digital. Many manufacturers don't tell you, so you have to look at the schematic in the service manual to tell. One way to know for sure is to record a short burst, such as the pulse created by an absolute phase tester. If one recorded this pulse to "0," or just below clipping, machines with true digital meters will deflect to a full scale "0" while analog driven meters such as the Sony 2500 will only deflect to about minus 30! Again, if you claim to be a professional studio, you should have a portable professional digital meter such as the Sony DMU-30 hooked-up to the digital output of your DAT machine. I have found this meter to be the best for my applications.

Speaking of the "Over" indicator, even if it is a digitally driven indicator, they differ from machine-to-machine. Some are set to light up if a single sample of the 44,100 cycles a second goes into clipping. Some are set up for the CD tape-checker standard of four consecutive samples clipping. The point is to know which is which. The Sony 7000 series is set by the user. I recommend a setting of 1 word= "Over," as the Panasonic SV-3700 has. I assure you that any "Over" is distorting the signal. The question is if it is noticeable. If you get "Over"s, check them out. My rule is, don't record any "Over"s! By the way, most record companies in the US do not allow CD master tapes with "Over"s recorded on them. In Europe they often do allow it. If in doubt, check it out.

I suggest recording at least a 1 kHz tone at the beginning of the first DAT tape of the session to confirm that the balance between left and right is correct and that you know where your input and output levels stand. If this is the first time you are using a DAT machine with the console, due to the fact that many of them are unbalanced non-professional devices, it is a good idea to record a whole set of tones to be sure there are no gross impedance mismatches between the console and the DAT machine. If you are making a compiled album master to send to a mastering house, don't put an index on the tones! That way index #1 will equal song #1, etc. It makes life just a bit easier for the mastering engineer.

WHICH DAT IS THAT?

Labeling the DAT tapes is actually more important than labeling big 14-inch reels of 1/2-inch analog tape. It is difficult for a 14-inch reel to "get lost" in a studio, while DATs can slip behind a cabinet or get tucked underneath a DAT machine!

It gets scary seeing master tape DATs of projects that cost over a million dollars that can seemingly disappear by themselves. So write down everything — and then some! As with analog tapes, each take should be carefully noted as to what differentiates it from any other take. Be specific! Make doubly sure that the index number on the tape matches the index number on the label.

Often an assistant will re-number the DAT indexes to be sure everything is sequential (which is a good idea). Sometimes the indexes can get confused however, so double check everything! It is common for an A&R person to hand us a DAT and say "use ID# 16 for the master." ID# 16 had better be right!

If the session results in many DAT tapes, it is best to store them in a 10.5-inch tape box or larger to keep them together. I have seen some assistants actually tape the outer DAT case to the inside of the box, which is nice. While you can be good at writing down all the mix data, you must also be sure to write some identifying information on the DAT itself. Write DAT #1 on the take sheet and DAT #1 on the DAT itself. Always pretend that all the DATs have been taken out of their cases and lumped together.

4 inputs...

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Could you tell which was which from the DAT alone? You must be able to do this! Also, if a mix is re-called at a later date, don't forget to dig out the DATs of the previous mixes and write "see re-call of (date)" on the take sheet and on the DAT itself if possible.

OBEY THE SPEED LIMIT

Regarding vari-speeding a tape: If you wish to remain in the digital domain from the DAT recording to the finished CD be careful of vari-speeds! Changing the speed of the DAT tape (as is easily possible on a Sony 7000 series machine) requires the use of a variable-frequency sampling rate converter in the digital domain. A small speed-up may require the sampling rate converter to convert from 44,313 to 44,100 Hz. Try doing that math conversion in your head!

Most sampling rate converters are fixed frequency (i.e., only go from 48 kHz to 44.1 kHz). Again, vari-speeding requires a variable-rate converter. Three of the four variable rate ones that I have heard don't sound that great. So be cautious about changing speeds after the fact. Suggest to the producer to print several versions on DAT by using the multitrack's vari-speed instead. In the future, when we mix digital multitracks in the digital domain, vari-speeding will become a very expensive problem.

Good digital recording technique has always required constant cloning for safety's sake. Any piece of tape can jam at a moments notice even on a "good" machine. I suggest recording on two DAT machines at once. They don't cost much compared to the cost of a session. At the least, make a safety of all your good takes at once. Drop-outs can always happen even though DAT is a very robust recording medium.

It is fine to daisy-chain the digital output of one machine into another when recording on two machines. AES distribution amps are available and are probably a good idea if you want to record on a bunch of DATs at once. Some of them employ de-jitter circuits that make the final playback sound a bit better. Some have switches that will let you interface consumer IEC ports and professional AES/EBU ports when they would not work together before.

THAT'S MY DAT

Good Luck! And although you never asked, I'd like to leave you with my personal definition of a truly professional DAT machine:

1. It has four-heads for read-after-write record confidence.
2. It has adjustable balanced +4dBm inputs and outputs.
3. It can record and play SMPTE timecode.
4. It has good sounding converters.
5. It is serviceable, it can be repaired and is worth repairing.

Bob Ludwig is one of the world's leading mastering engineers. He has been involved in thousands of projects and has won numerous awards. His most recent work includes Bruce Springsteen's two new albums, Lucky Town and Human Touch.

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The Return of the EMT

You've probably heard of it. You've definitely *heard* it. Maybe you've even seen it out-of-the-way in a studio closet, store-room or maintenance shop. The historic EMT 140 echo plate was one of the most important devices in record making throughout the 1960s and 70s. But due to the impact of digital technology, plate echo was almost rendered obsolete. Now, due to an ingenious but simple modification developed by Jim Cunningham of Chicago, it's ready for a comeback.

For those of you who are too young to remember this studio classic, an echo plate is actually a large flat thin sheet of steel suspended from its four corners within a protective frame. A transducer (the driver element) transfers the incoming (echo send) signal to the plate, exciting it and causing it to ripple and "wave" imperceptibly to the eye. These "waves" are analogous to sound waves bouncing off the surfaces of a room. Two transducers pick up the echo signal created by the plate's movements and send it back (the echo return).

You've heard the result of this sound process thousands of times. It can be heard on just about every quality popular recording made between 1960 and 1980 — until suddenly it almost disappeared overnight, another victim of the digitization of the modern recording scene. The output of the plates was simply too low by today's standards, requiring a great deal of amp gain, and thus adding too much noise to the mix.

The old accelerometers that served as the pick-ups on the EMTs had to touch the plate at a very small point in order to maintain high frequency response and this minute point of contact resulted in the low output. Jim Cunningham's improvement consists of a new "Unimorph" pick-up design that covers a large area of the plate while still maintaining sonic integrity. This larger mass provides a much stronger output simply due to its contact with enough of the surface to "feel" the waves, rather than a tiny portion of them, requiring less amp gain and improving the signal-to-noise ratio substantially. So even the modification that brings the beautiful plate sound up-to-date is analog in nature.

The modification itself was relatively easy to do. Within a couple of hours, I was able to replace the original pick-up with the Unimorph design. I simply shaved off the old pick-ups with a razor blade and

Tom Jung's office and studio

The famed EMT Plate is ready for the all-digital audio environment thanks to a new modification

BY TOM JUNG

BLAST FROM THE PAST: With a modification, the EMT is ready to take on today's digital sounds as well as it did anaolg sounds in the past.

attached the new ones with the supplied cement. Then I changed two resistors in the amplifier, lowering the gain and there it was — a 20dB increase in signal-to-noise ratio.

The next step was to interface this newly-modified gem with my Yamaha DMR8 all digital console. Of course, the signal coming from the console has to be converted from the digital domain into analog, and the analog signal from the plate needs to be converted back to digital. It may look like a lot of effort but believe me, it's worth it.

A LITTLE HISTORY

EMT plates were invented in Germany by Dr. Walter Kuhl of the Institute of Broadcasting Technology in Hamburg back in the 1950s. When the earliest plates were introduced to America they sounded terrible. Harvey Radio was the original importer and sold a dozen of them to Columbia Records, which couldn't get them to sound right. At the same time, Phil Ramone and Don Frey were in the process of opening A&R Recording in New York, which was destined to become one of the most famous and successful studios of all time. Harvey asked Phil and Don to see if they could figure out how to improve their sound and, in return, they could have one for A&R, while Harvey could maintain the Columbia sale. It was Phil Ramone who discovered that the EMTs had to be "tuned" by adjusting the tension of the steel plate. The rest is recording history.

BACK TO THE FUTURE

The record label I founded, Digital Music Products (DMP) of Stamford, Connecticut, specializes in presenting live performances of modern jazz with audiophile sound quality. Celebrating our 10th anniversary with 50 releases currently in our catalog, we're an all-digital label and we always have been.

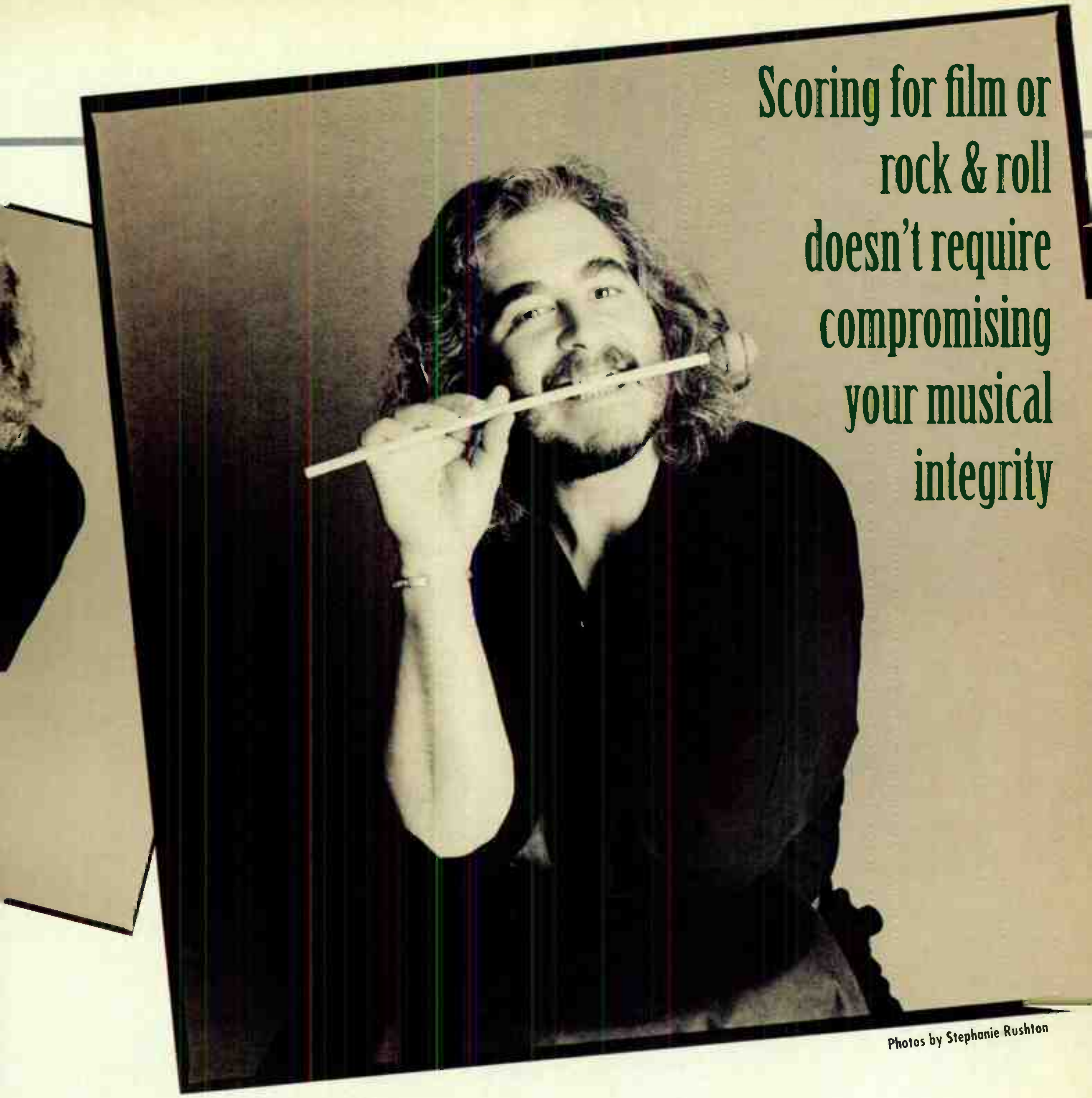
I am known in the industry as somewhat of a digital guru, my studio is digital, but there are still some wonderful analog things that have that warm, round sound that you just can't duplicate with digital equipment. The EMT plate had a sound that no digital device can imitate. But what about the noise level? Until recently, it simply didn't make the grade. But thanks to Cunningham's inexpensive modification, the noise level of the old plates can be reduced by around 20 dB. And I've been using an Ecoplate with the modification ever since with great results.

Cunningham, who in the late-70s designed and manufactured the Ecoplate (an improvement on EMT's design), has come to the rescue and EMTs and Ecoplates can now do their stuff in the digital environment. His modification has reduced their inherent noise to an acceptable level for nitpickers like myself. And don't get the impression that these things were nothing but bundles of noise in the first place — they weren't. They just didn't have the noise floor of today's digital devices. In fact, some facilities still use them in their original state

continued on page 83

Knowing The Score

BY MICHAEL KAMEN



Scoring for film or rock & roll doesn't require compromising your musical integrity

Photos by Stephanie Rushton

Manos Hadjidakis once said to me, "You mustn't write at the keyboard because you become a prisoner of your technique." You can also become a prisoner of the technology. This is especially true with film scoring, where it's best to learn the mechanics when you get the gig. There's no need to preoccupy yourself with all the functions of a piece of equipment or the politics of meeting producers and directors, because when you finally do get an assignment, you might have stopped concentrating on something far more important — making beautiful music.

As is the case with most film composers, I actually backed into writing film music. I specifically set out to compose for ballets, but there wasn't much of a call for it. (I was damn lucky to be able to do a few.) My background has always involved writing or playing orchestral and chamber music and I have refused to let go of my personal style — whether it's for films or rock and roll.

NEVER SURRENDER

Perhaps the most difficult aspect of film scoring is having to surrender some personal style to a film score's needs. You can't write western music if you're working on a space movie, right? (Or can you?) Merely because a piece of music is intended for film does not mean that it should be shoved into the background and not have an identity of its own. Film music has a more noble notion than simply filling a void. Internal or self references can be injected into a music score's theme, giving a composition a consistent feel throughout a film and causing the audience to easily acquaint the music with either a feeling in the movie or the entire movie itself. Such themes can be most easily identified by little pieces of recognizable melody. And when you have a melody to rely on, scoring becomes an arranger's function. All you have to do from that point on is ask yourself, "What version of the melody can I play now?"

For example, the score I wrote for *Robin Hood: Prince of Thieves* had a boldly obvious melody. The hook in the theme song "Everything I Do" became the theme for the movie, and the melody itself actually became a "character" cropping up throughout the film. And that is the best function a film score can perform. The other functions are basically cosmetic, designed to help get around tough bits of filmmaking such as when a scene isn't scary enough or when a scene is frightfully too long. However, a film composer is not a miracle worker. Film music can help a good movie, but it can't save a bad one.

The rest of the scoring process is really just a series of cinematic problems that require musical solutions. Any scene can have as many as a dozen musical solutions, and everyone's approach to a similar scene is different. My approach is to take a reactive position, as if I were watching the scene as a member of the audience or as if I were an actor trying to relay the feelings of the scene to the audience. You want the viewer to feel a specific emotion. Fear, love, hate, exhaustion — whatever the emotion, you need to get it across musically.

Of course, there are some basic things all film scorers must do. For example, a motion picture always exists within a time frame and a location. Establishing the crucial elements of time and place musically is the second most important goal of film scoring. The most important goal is getting the score done on time. And that's where the right pieces of equipment — properly used — turn out to be critical.

EQUIPMENT OF NOTE

When I first heard that there was an instrument that provided notation, I immediately went and bought one. The original Kurzweil 250 had an intelligent design that represented a more solid investment pound-for-pound than any other single piece of music technology. And though its notation system wasn't quite there, in my opinion, there was still a lot to offer. Considering it was and still is an instant 12-track MIDI sequencer, the 250 is endlessly fascinating because I can play a reasonable facsimile of an idea into it and then come up with something that sticks. It has become part and parcel of why I've been able to score seven or eight films a year.

My project studio in my London home is where I work most of the time, and while it has just an average run of equipment, it's functional and easy to work in. I use a Macintosh computer with Performer software to access files from the Kurzweil, and to also edit sequences. There have been times where I've pasted an assortment of ideas together using Performer, and have come up with wonderful sounding cues. I also have an Akai A-DAM digital recorder and an old Seck mixer that I run to the Akai. I use a pair of Yamaha NS-10-monitors (because they won't blow up). My video equipment is all multi-standard since I work on movies made in the U.S. and the U.K. I also use an old Sony monitor and a U-Matic tape machine. A Sanyo LCD video projector is the most recent addition. Not exactly your major high

Bach and Roll

Michael Kamen has always felt that combining classical music with elements of rock and roll was a viable means of composing beautiful music. Music that would be right for the modern era. Music that has taken him a long way.

After dropping out of Juilliard to play oboe with a rock band in the mid-60s, Kamen wrote ballet music and moved on to work in such diverse arenas as musical director for David Bowie's "Diamond Dogs" Tour, scoring feature films, working with Pink Floyd, and directing a 100-piece East German choir. In the process, he has also worked with David Sanborn, Eric Clapton, The Eurythmics, George Harrison, Queen, Kate Bush, Queensryche and Tim Curry. He wrote the score of *Robin Hood, Prince of Thieves* and the film's hit song "(Everything I Do) I Do It For You," with lyrics by Bryan Adams and R.J. Lange, which is presently the second largest selling single record of all time, having gone triple platinum. The song won him *Billboard*'s Song of the Year and was nominated for the Golden Globe Awards' Best Score and Best Song. The song was also nominated for an Academy Award, where Adams and Kamen played it live for the Oscars' audience.

Kamen's gifts find him performing many roles including, musical director, producer and arranger, as well as composer and scorer. His other feature film credits include *Shining Through, Die Hard I* and *II, Lethal Weapon I, II* and *III, The Krays, License To Kill* and others. Scores for England's Channel 4 and the BBC, Steven Spielberg's *Amazing Stories, Christina's World* (an Andrew Wyeth documentary), and the ABC series *Animals, Animals, Animals*, are among his TV credits

How does he get so many gigs, and in such varied musical fields and styles? Kamen feels that all forms of music have something for everyone. "I speak the language of music, which I feel is the language of human emotion," he says.

That belief has certainly worked quite well.

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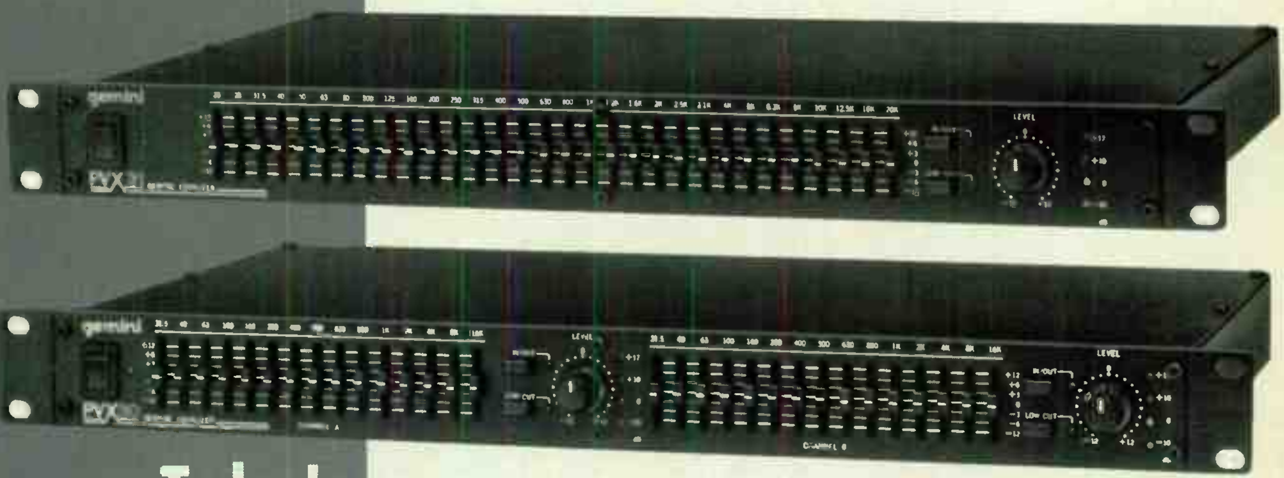

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tech environment but it gets the job done.

BACH OFF

The other side of my studio is devoted to state-of-the-art 16th century technology: pencils, paper, rulers and brains. Listening to Bach, I sometimes wonder if his brain worked so much better than ours simply because he kept so many of his ideas in his mind. He certainly worked a lot faster than most of us, his music is painfully better than anything else written today, and he did it all with a quill and a piece of paper.

But then again, Bach was a musician and all musicians are, to some extent, tweak freaks who love to play with the latest gadgets. You never know when that one piece of equipment that will make a colossal difference may come along. Some time ago, I thought that piece was the NED Synclavier, but now I feel as though a few layers of Synclavier can sound like a giant theater organ. I just don't like its harsh tonalities. But, when its sounds are kept simple, it can cover a fairly wide emotional range. Nevertheless, I still feel a more immediate relationship to the Kurzweil than to the Synclavier.

Good film music can help a movie, but it can't save a bad one.

Photo by Peter Osbourne

When composing, I prefer to use as few electronic devices as possible. I almost never write to timecode, and I've become accustomed to working without a click track. In fact, I hate to use a click track when creating an emotional soundtrack although sometimes it's unavoidable. The time frame for *Robin Hood* was nearly impossible (about three and a half weeks), so I had to use a click track just to be able to get more than two hours of music done in time. Scoring *Shining Through* on the other hand, I had almost six months to write and virtually none of that was done to a click.

NO SMALL HALLS

Of course, sooner or later I have to leave home to record. When I'm

continued on page 107

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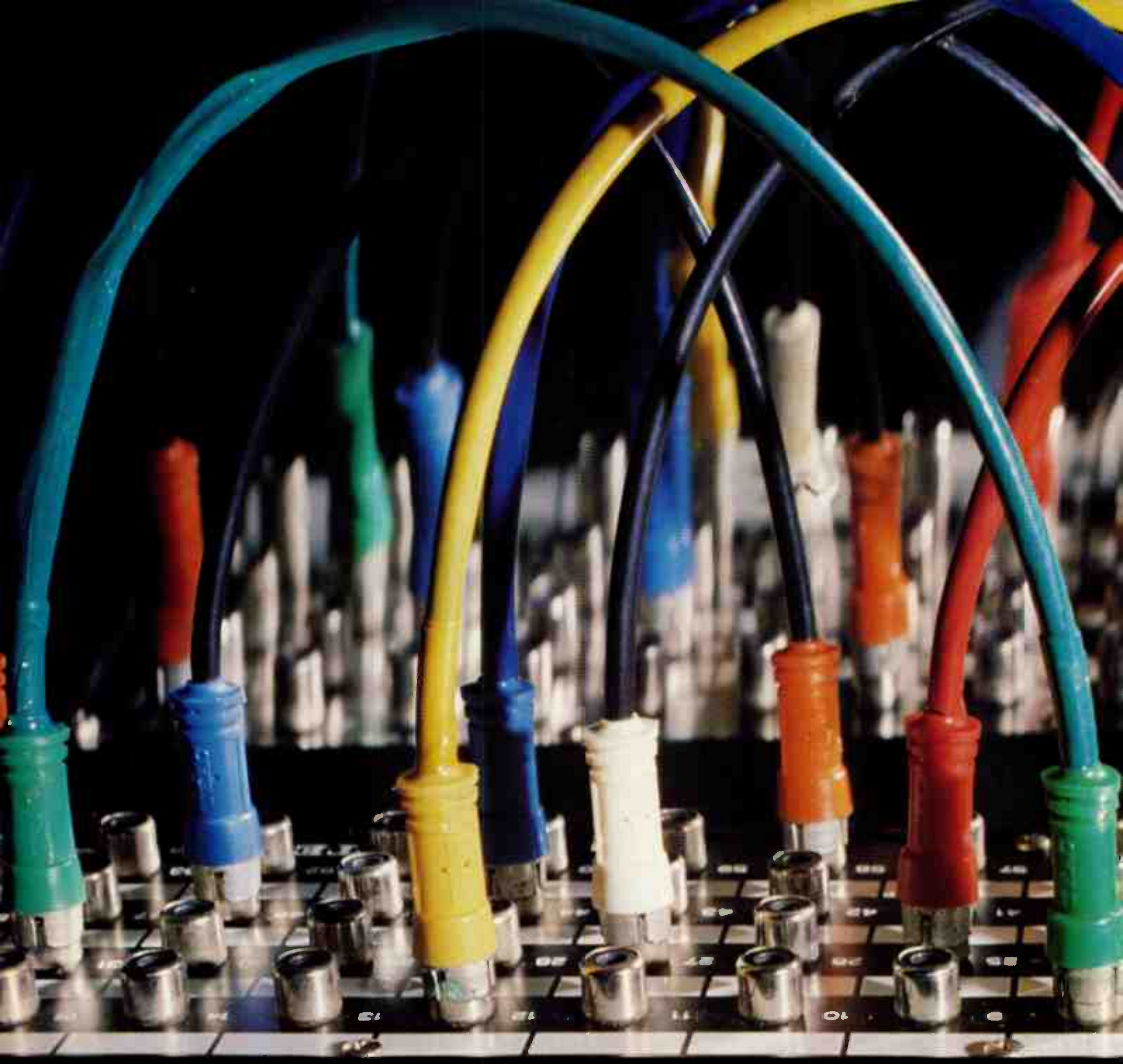
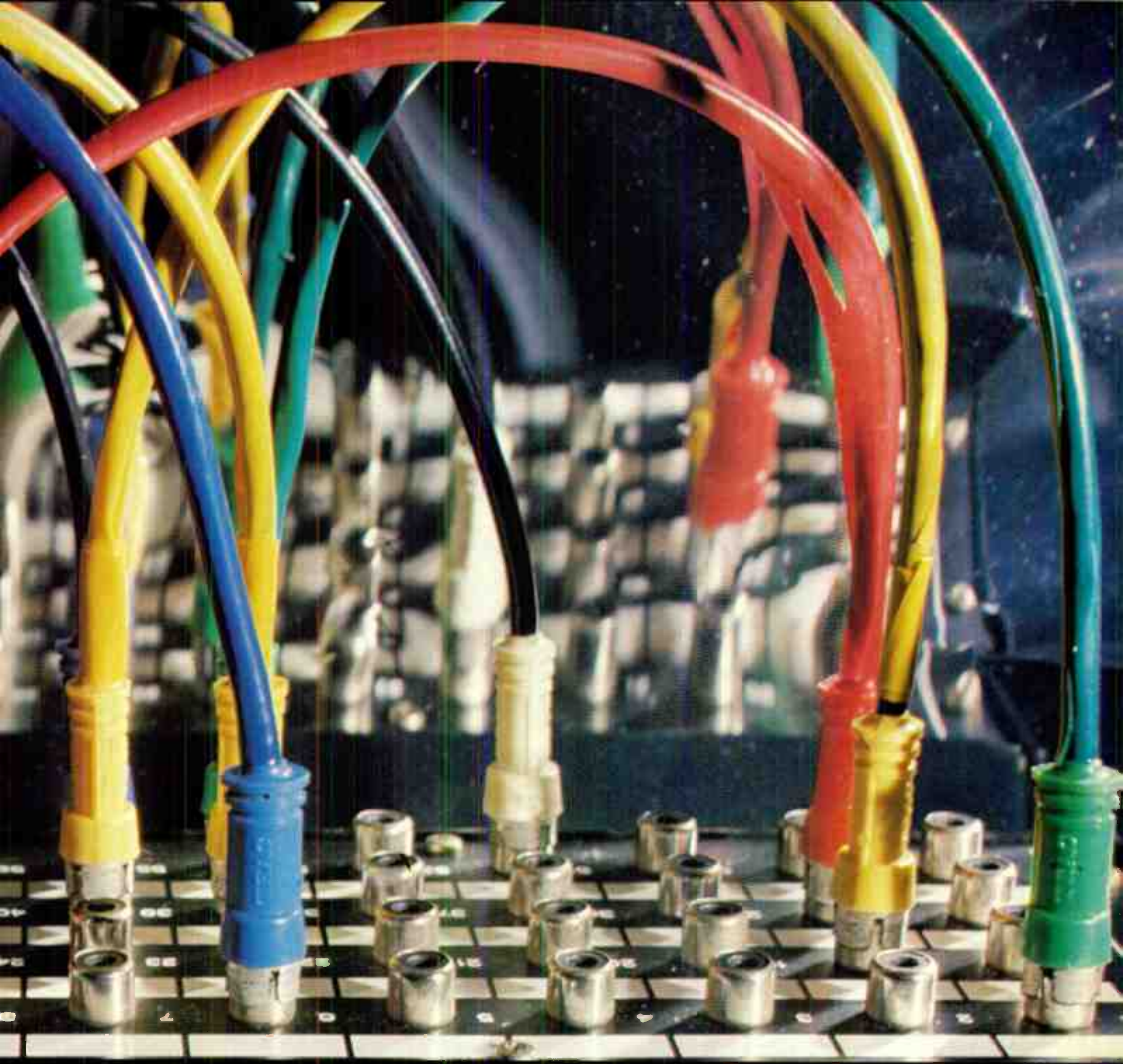


PATCHBAY PERFECTION

Everything you need to know about planning and building the brain of the studio

By Dave Brody



We all know him. The guy down the street who dives behind his PortaStudio each time he needs to change-over his one reverb from riding on the monitor mix to printing to tape. Meanwhile, down on Studio Row that big-time video place has an automated patching system, called a routing matrix, to access any device in the house.

Somewhere in between: you and me and our project studio.

There probably isn't any piece of equipment that is more important to a working studio than its patchbay. The patchbay (or "jackfield") is the nerve center of your studio; it lets everything in your workplace interconnect.

When you plan your patchbay layout, you are planning your future growth potential. The strategy here is to preserve your options. What do you own now; what do you see yourself adding? Is your equipment electronically balanced, un-balanced or a mixture of the two? Does the architecture of your present (or future) mixer include an "insert point" (ie, the ability to drop the signal from an individual channel through an outboard processing loop)?

Do you work live, sequenced, both? Will you be mixing virtual tracks with tracks recorded on a multitrack? Do you use timecode; will you in the future?

If your studio is composed solely of unbalanced, "semi-pro" gear you can use jackfields of the RCA, 1/4-inch mono or mono mini-plug type. These are wired with a single conductor plus a ground. As unbalanced bays are generally for smaller or younger, start-up operations, they can be configured as "jacks wired to jacks" — allowing you to use off-the-shelf cables. If you'll eventually acquire a major device that has balanced inputs and/or outputs, you will need a TRS ("tip, ring, sleeve") type patchbay. The standard for this is the "Bantam" TT ("Tiny-Tel"). TRS bays interface using wire that contains two conductors plus a ground.

Outlay? The cost of patchbays can be astronomical. And you should definitely buy more patchpoints then you think you need. So you'll be thinking: "A lot of coin for something that makes no sound of its own, adds no signal processing, records no audio and stores no data." But having a good interconnect system will make your life easier and more productive.

TT patchbays can be purchased in a variety of configurations: from naked-jacks to fully-cabled and terminated in multi-pin connectors. They are generally one rack unit high and contain two, 48 point rows. If you're stuck for space, you can get three rows — 144 points — in the same 1U of height, but it may make wiring them and reading their labels more difficult. Short-loaded units are available (in multiples of 24) but can waste space.

Lately many pros have been showing up in studios with their toys in road racks cabled to TT patchbays on snakes via large, multi-pin block connectors (usually of the DL or ELCO type). These patchbays can sit on the console's meter bridge, adjacent to the desk's jackfield. This allows for a speedy, standardized, "use it where you need it" interface.

SO MANY HOLES, SO LITTLE TIME

To understand how much patchbay you'll need, make yourself a Master Wiring List. If there's a computer in

SOME SPECIAL STUFF:

You'll probably figure out your own ways to juggle the ins and outs of your jackfield; after all, this is your workspace. Consider these special cases:

Expander Gates

Few of us can afford mixers with dynamics modules in every input, so we buy external gates. The standard way of arranging gates places the output above the audio input and the key input adjacent thereto. Thus we might have something like this on our patchbay:

Top:	Bottom:
Gate 1 Out	Gate 1 In
Gate 1 Key	Gate 2 Key
Gate 2 Out	Gate 2 In
Gate 3 Out	Gate 3 In

etc...

Mults

Try to arrange things so that somewhere on the bay you'll have at least four unused jacks adjacent to each other. Wire these together: "High to High," "Low to Low," Ground to Ground, and nothing normalled. These jacks will now form a mult/split. Having two such mults is a great advantage; it lets you multi-route a stereo signal.

Copy-Ready Decks

If your mixer does not have a built in facility to accommodate extra machines (like cassette decks), you may wish to extract a feed from the output of your mixdown recorder to drive them. You do this by normalling the output of the mix machine at its top row patchpoint (Stereo Tape, 2 Track Out, etc.) down to the blades of the input patchpoint of the supplementary deck, e.g., your cassette deck. You can mult this around to additional machines (which comes in handy for making those multiple dubs). If your mixdown machine is +4dBu out, you'll have no problem hanging these additional recorders on its output.

Auxiliary Sends as Cue Sends

If you work with live mics and headphones at all, you'll probably want to normal at least one of your "potted" sends to your cue amp input. Full normalization will probably work here, as you will most likely never want to send the same mix to both headphones and external equipment (like reverb). Of course you will want the option of folding effects back to the cans without printing it (use your other send(s) for this). When it comes time to do the mix, just patch the send you used for cue into the effect you want to drive — this will break the normal to the cue amp and give you another "aux send" to play with.

Leaky Timecode

If you are running an unbalanced patchbay, and/or a lot of semi-pro gear, you might not want to bring timecode up into your patchbay; nasty little squarewaves contaminating your pristine audio! If this proves to be a problem, dedicate a standard track for timecode and cable your synchronization devices directly to and from that track, avoiding the patchbay altogether.

IN Trouble

You may also want to think twice about bringing the inputs for your monitor amp(s) up on the patchbay. There's nothing technically wrong with doing so, but the first time your new assistant mistakenly plugs a high level source (or, perhaps even worse, an open ended patchcord) directly into your wide open 500 watt amp may be the last time that those particular tweeters tweet properly (or at all).

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the house, open up a virgin spreadsheet (or else get a printed multi-column ledger). The Master Wiring List I use was developed using Microsoft Excel™ Ver. 1 running on a Mac Plus. No field formulae are used, so you should be able to do yours in just about any specialist program; on nearly any computer. The idea here is to number and document every connection in the place — before actually making the first one.

Enter everything that has audio going through it except mics and speakers. Be sure to include any musical instrument that you want to keep "plugged-up and ready to play." Also include mic snakes or mic panels.

Now try to envision your studio as it shall be. If there's a realistic chance of owning a particular piece of gear, find out all about its ins and outs (literally). Enter each of its connections into your list and designate them as "TENTATIVE." If you're not sure what you'll be acquiring, include some extra lines in each category of device and mark them "SPARE." If you're a MIDI-module intensive operation, you'll probably want the flexibility to sell old gear and buy new stuff, as well as to run borrowed, rented or other devices. The best way to do this is to designate a number of patchpoints as "instrument tie lines." These may end up as snakes to your keyboard rack(s), or in DI's (direct injection boxes), or hooked up to an external patch panel located near where the instrumental action is going to happen.

Notice that there are natural groupings of wires that go to similar destinations: inputs to the multitrack, lines from the keyboards, outputs from your mixer, etc. Think of the aggregate of wires in your studio as a tree. This tree will have its roots in your patchbay and branches reaching out to your gear. By creating a Master Wiring List, each wire of your studio-tree now has a unique identity in the form of a number. For this to do you any good, you'll have to physically attach that number to the actual wire — but more on this later.

Your patchbay should be organized so that the roots of your tree that most often need to talk to each other are located close to one another. Standard TRS jackfields contain a way for connections to conditionally communicate — without even using a patchcord.

WHAT IS NORMAL ANYWAY?

An important technical decision you must make concerns the "normalization" of certain outputs and inputs. Here's the key: On most professional consoles, the upper row of each vertical pair of patch points is an output; the lower row is usually an input associated therewith.

Within the patchbay itself, we can wire the audio going into the top jack down to the bottom jack for additional use. So, as an example, we can hook the output of one track of a multitrack recorder (say Track 6) to the top jack, hook the wire going to the line input of the corresponding channel of our mixer (Input 6) to the bottom jack and "normal" them together at the patchbay.

How? Look at the back of a patchbay and you'll see that each jack has five connection points: One ground (correspond-

ing to the Sleeve of the plug), one "high" Outer (corresponding to the plug's Tip), one "low" Outer (corresponding to the plug's Ring), one "high" Inner or Blade (which is in contact with the "high" Outer) and one "low" Inner or Blade (which is in contact with the "low" Outer).

Now insert a plug into the jack you've been looking at. Notice that the action of the plug physically lifts the outers off of the inners. Therefore, if we wire the outers of a jack in the top row to the inners of the jack below it, we create a situation in which the feed to the top jack will also go to the device connected to the bottom, *unless we interrupt it* by sticking a patchcord into the bottom jack. This is known as a "Half-Normalled" condition.

Notice that a plug inserted into the top jack of a half-normalled pair will not break the connection to the lower jack. Thus we have a perfect way to split any signal fed to the top jack of a half-normalled pair. We could, for example, pick up the feed of our multitrack Track 6, send it through our new Digital Time Transmogrifier and plug it into, say, Input 7 of our mixer, letting us mix it with the un-effected feed reaching Input 6.

Full Normalization (Inners of the top jack wired to Inners of the bottom) legislates a condition in which a plug inserted into *either* jack "breaks the normal" between them. It's therefore reserved for those situations where we want to *preclude* multipatching. Let's say you bring your microphone feeds (or mic-level feeds from DI's) up to the top row of patchpoints, and then connect the inputs going into your mixer's mic pre-amps to the associated lower-row patchpoints. You probably won't ever want to split an un-amplified mic feed to two different inputs.

You may not want *any* normalling between certain pairs. There's no reason to short the outputs of your outboard gear (compressors, gates, DDL's, etc.) to the inputs. Or your top row may have nothing to do with your bottom row, as in the case of instrument tie lines. It's common practice for outputs to continue to appear above inputs even if no normals are drawn between them.

This business of normalling (Half, Full and None) gives us a guiding principle for the layout of our entire patchbay. Think about the flow of audio during your sessions. Perhaps it's virtual tracks from MIDI boxes through console inputs mixed with effects and composited on DAT. Or maybe you have various audio sources layered to a SMPTE-locked multitrack ending up on videotape. Your patchbay logic should reflect the way you want your audio tree to grow.

To see what the options are, let's look at the top to bottom patchbay logic which most large studios use:

Bay #1: Studio Microphone Feeds/
Mic-Level Instruments (DI's)
Console Microphone Pre-Amp Inputs

Bay #2: Outputs from Multitrack Recorders
Console Line Inputs

Bay #3: Insert Point Sends from Input Channel
Insert Returns to Channels

Bay #4: Outputs from Routing Busses (Groups)
Inputs to Multitrack Recorders

Bay #5: Outputs from Multitrack Recorders
(Repeated)
Inputs to Console Monitor Channels or
Monitor Section

continued on page 83

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BAND IN A VAN

CBGB SOUND
PAGE 59

A SPECIAL SUPPLEMENT TO EQ MAGAZINE

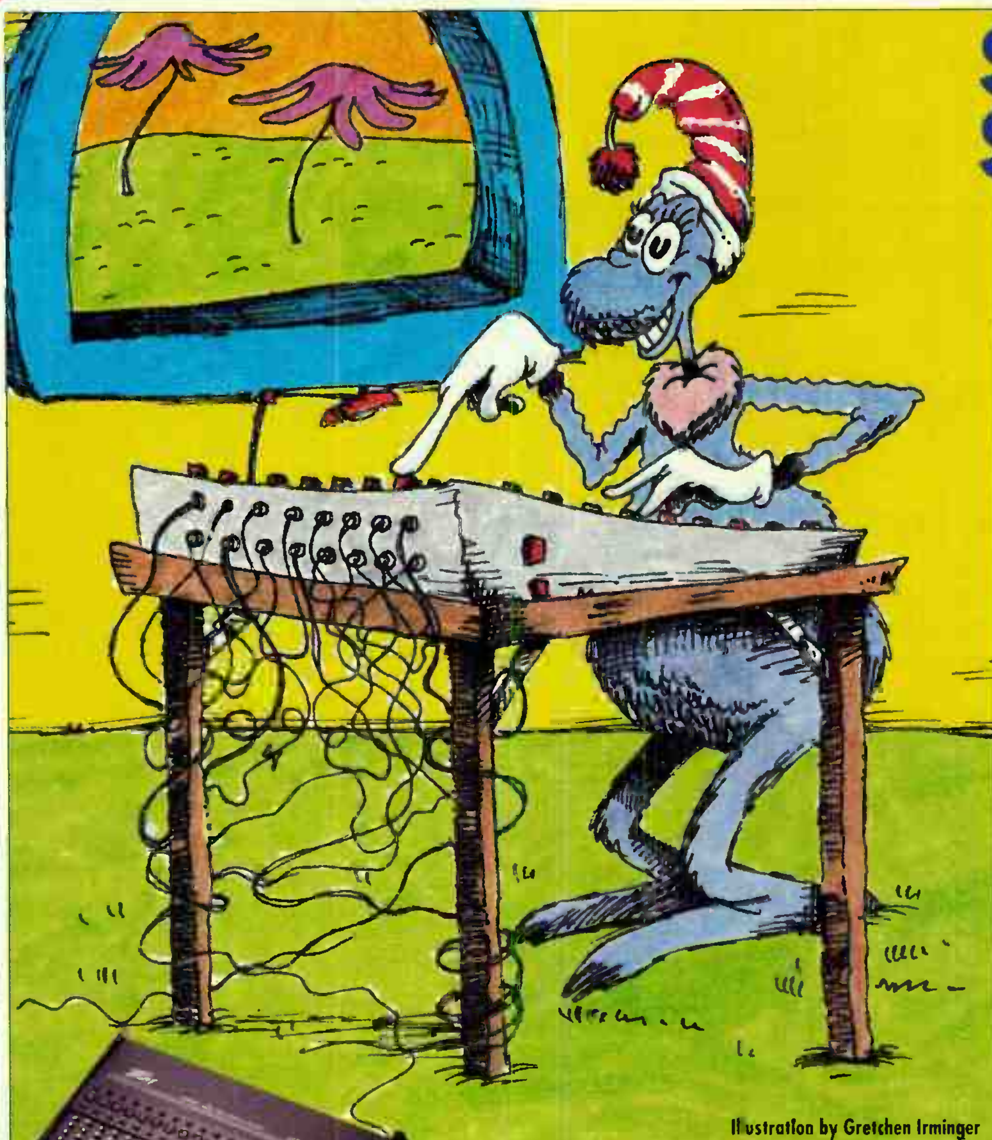

Illustration by Gretchen Irminger

SOUNDMAN SAM, I AM

HIRING THE RIGHT SOUNDMAN IS AN ACQUIRED TASTE LIKE (WELL) GREEN EGGS AND HAM

BY JIM PAUL

IN THE COURSE of a band's growth, one goal should be to get bookings into larger venues and clubs. Eventually there may come a festival, large club, or opening act spot with a big professional sound system under the control of a sound company or house mixing person. Before that time comes, however, it makes a lot of sense for a band to hook up with a person who will take responsibility for the sound of the band. Otherwise, you will be at the mercy of whoever is running the sound system.

Finding the right person is not an easy task and should be taken as seriously as adding a new guitar player. It's difficult for the band

(continues...)

INSIDE: POWERED MIXERS BUYER'S GUIDE, MIC SPLITTERS & MORE

members to give up control of their final sound — and it is also difficult to hook up with someone willing to put in the hours necessary to do the job right. But what a difference it makes in the long run.

I remember how I felt with my first soundman. I wanted a sound guy in the band about as much as I wanted a root canal.This particular incarnation of my many bands was called Cornerstone. We played some pop covers and some originals and we sounded pretty darn good (I'm being totally objective here). Our venues included parties, wedding receptions, festivals and supper clubs. It had occurred to us that maybe we needed to find a soundman, but I had always managed to derail the idea before. This was a real tough concept for me because I had always been the operating sound guy and I really enjoyed the control it gave me over our final product. It was, however, getting more difficult all the time for me to be Mr. Guitar Player, Mr. Singer and Mr. Sound Guy all at the same time.

FINDING THE RIGHT SOUND PERSON IS NOT AN EASY TASK AND SHOULD BE TAKEN AS SERIOUSLY AS ADDING A NEW GUITAR PLAYER.

JAMMING WITH SAM

About this time, we received an invitation to play at a large concert as opening act for the High Rollers, a fairly popular all original band (don't ask me where they are today!). They were publicizing the show well and it was to be at a large hall which, if I remember correctly, seated around two thousand people. They were expecting a sellout and this would be the largest crowd we had ever played in front of. The High Rollers had their own sound guy and when they asked us who ours was, we hemmed and hawed and skillfully avoided the question. This led to a big discussion in our band about hiring a sound guy and despite my objections, and no matter how I protested or how threatened I felt, I was out-voted and we placed an ad for an engineer.

We finally ended up with a fellow who I will call Sam. Sam was willing to come to rehearsals and learn the music, not get paid until we did, and then take a slightly smaller share of the proceeds than the rest of the band when we did get paid. What a guy. I was rather suspicious and (I guess) I gave the guy quite a hard time. But little did I know that Sam was about to teach me a truly important lesson. Like how to let go and learn to trust our sound to someone besides myself. In almost no time, Sam became like another member of the group and, in spite of my meddling efforts, he proved himself to be easy to work with, loyal and dependable.

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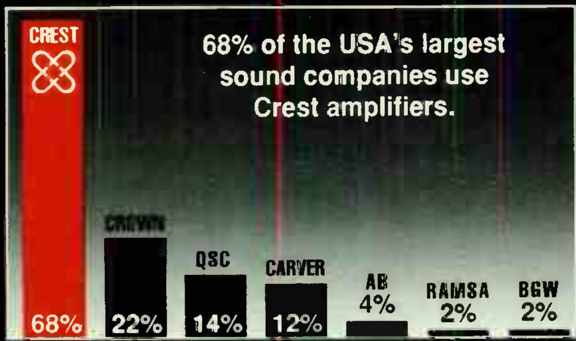

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ODE TO SAM

I will never forget what he did on the day of the concert. We had a totally intense rehearsal that day and were trying to get everything really tight. I was fulfilling my usual role of sweating the details and when Sam arrived at rehearsal I started to raise some of my concerns about the show. To my utter amazement, he had already been to the hall to check out the acoustics, had been in contact with the sound company, and had borrowed a flatbed truck for hauling our equipment. With a new respect for this man, I started to relax and concentrate on my parts.

When it was time to go, I was to receive another surprise. Sam had also recruited one of his buddies to help him, so now we had two roadies! We fully expected to cart our own equipment as usual, so this was like being in heaven. They tore down our rehearsal set-up, loaded all of our equipment on the truck, and said they would see us at the concert hall. Never before had I been able to arrive at a show without off-loading equipment, setting up the sound system, and worrying about all the little details. Instead by the time we got to the place, Sam and his buddy had already loaded in, and had most of our stuff set up and ready to go. He had a stage diagram and set list for the stage manager and the sound company, had met the guy running the system, and had already familiarized himself with the board, a Yamaha PM-2000. The soundcheck went very smoothly, and I began to appreciate just how lucky we were to have a guy like Sam. He knew what we liked in the monitors. He knew how each song went. And best of all, he really seemed to like our music.

> ***I WANTED A SOUND GUY IN THE BAND ABOUT AS MUCH AS I WANTED A ROOT CANAL.***

Showtime came and the High Rollers were right, it was a sellout. I was nervous as hell about how we'd sound, but my drummer started whacking me on the head and told me to relax and trust Sam. Finally we were introduced and ran onstage. During the first song I began to relax and we played our butts off to a very enthusiastic audience. Our part of the show was definitely a great success. Upon listening to the recording from the show, I realized we had sounded better than we ever had before.

What's the point of this little story? It's just to let all you control freaks (like me) out there know that if you find the right people, they can add a lot to your band. I have never forgotten Sam and what he did for us that night. I learned that I could trust our sound to another person and still sound great. And I learned what a competent sound engineer can do for a band.

Thanks Sam, wherever you are (am?).

IN A BAND VAN

From Punk to Perfection

CBGB, the Big Apple's foremost band showcase, sounds much better than it looks

By Robert Goldsmith

AT 4:00 in the afternoon, the interior of CBGB is as dark as Dr. Frankenstein's castle. The walls are covered floor to ceiling in multiple layers of graffiti that may one day become an archaeologists delight. A tacky red velvet sectional and a few forlorn tables face a stage now empty except for memories and a few rangy mikes. Some of these look like they've been flattened by a truck and smell like a city street after a ten-day garbage strike.

"It's home," says soundman Ronnie Ardito.

For tons of up-and-coming (and many going nowhere) bands, CBGB may not be home, but it's a place they'd sure like to visit. Despite the hectic pace of soundchecks and undesirable set starting times (such as 8:30 PM or 2:30 AM), CBGB remains a premier place to play for unsigned bands in New York.

Some of this is due to the club's history as the focal point of the city's legendary late-70's punk scene. The original stomping ground of The Ramones,

Soundman Ronnie Ardito at his home away from home.

Photo by David Llewellyn

Blondie, The Talking Heads, Television and many more. But equally important is the club's reputation for quality sound. For this, Ardito credits the club's PA, created by sound specialist Norman Dunn. "It was designed especially for this room," revealed Ardito. "Audio experts came in here and pink noised the place, and the cabinets were custom built for this room. Nobody went to Sam Ash and just said 'give us some bass bins.'"

The PA includes a Soundcraft Series II 16-channel mixer from 1978. Ardito is especially proud of this old work horse that, by now, has probably had enough beer spilled on it to intoxicate every long-haired guitarist in the country. "It has the best sounding EQ. It's used seven days a week, 16 hours a day for live mixing and recording. When it finally goes, we'll put it in a glass case and send it to The Rock 'n Roll Hall Of Fame." The club also uses eight mighty Crown DC 300 amplifiers, JBL 12-inch and 18-inch speakers and JBL monitors. "There's no fancy stuff here," Ardito says, "but to get it to sound great all you have to do is listen."

After five years with the same equipment, mixing is effortless for Ardito. "I could die," he suggests, "you could prop my corpse up there, and I could still mix the band." Soundchecks begin at 4PM and last 20 minutes for each band. Ardito does the drums first and takes only about three-minutes with them. Next he polishes off the bass, guitars and vocals. "I know just by hearing people speak how I'm going to EQ their voice. I know whether I'm going to have to roll off some top because their speaking voice is a little squeaky." Then the band performs two songs as he straightens out the monitors and then they get off the stage.

Occasionally a band's unusual line-up makes things a bit more complicated. For instance, how do you EQ an oil can? Boost the hi-test, uh...high end? Set the low end at 10-40? The oil can in question belonged to Cop Shoot Cop, a creative noise band. "They're challenging to mix, but it's a pleasure," Ardito says.

Not all the bands are worth this much effort. Given that he's probably seen about 10,000 groups over the years, he's bound to have suffered through a battalion of clinkers. "A bad 35-minute set can seem like 50 years. You can't make a bad band sound great," he sighs. "All you can do is define the sound so you can listen to the mistakes better."

But one of the veteran soundman's most endearing traits is his sympathy for bands. "Playing in a band and trying to make a life of it is a really tough proposition, and you know that almost everybody isn't going to make it. So it's kind of sad right from the start because the chances are way against you — even if you're talented," he says.

He knows this from personal experience. Like most sound engineers, he started as a musician. In fact, he even had his very own "15- minutes" of fame as a guitarist with the pop band The Shirts. The group had three albums

"I COULD DIE, YOU COULD PROP MY CORPSE UP THERE, AND I COULD STILL MIX THE BAND."

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IN A BAND VAN

out on Capitol and a couple of Top 10 European hits, including "Tell Me Your Plans" and "Laugh And Walk Away."

That's when he first arrived at CBGB, which, like most clubs then, had no PA and no stage monitors — just some stereo speakers hanging above the stage. "We brought our own Community Light & Sound horns and a Shure mixer," Ardito recalls. "Hilly (Krystal, the owner of CBGB) said, 'If you play too loud, I'm pulling the plug on you guys.' I had an Acoustic amp and I faced it to the back wall, away from the audience, because I was so afraid of playing too loud."

The Shirts' career had a number of ups and downs, including playing a dive in Bristol, England where Peter Gabriel saw them and asked them to open for him on a six-week European swing. After a year as a roadie for the B-52's and a songwriter, Ardito wound up back at CBGB, where he estimates that he and The Shirts played about 500 times. Now, in addition to live sound, he engineers live recordings for bands and for CBGB's video program. He records onto the club's venerable Autotech 16-track ("built in the year of the flood") and mixes down to a Panasonic SV-3500 DAT.

HOW DO YOU EQ AN OIL CAN? BOOST THE HI-TEST, UH...HIGH END?

After his near brush with stardom, you might expect Ardito to be disappointed with his current soundman role, but not a chance. He works 70 to 80 hours a week, and says, "It doesn't seem like work. When rock 'n roll becomes a drag, pack it in baby."

Powered Mixer Buyer's Guide

Choosing a self-powered mixer can be a complicated process. To make things easier, *EQ* presents an easy-to-use comparision chart detailing some of the hottest systems around. These powerful tools certainly have their place. Just remember — in comparison to a separate mixer and power amplifier, a high-performance self-powered mixer is tough to design. There are four inherent trouble areas: heat, magnetic fields, high voltages and excess gain.

A skilled designer can usually deal with these problems. Nevertheless, before purchasing, make sure to conduct a careful inspection. Pay extra atten-

Brand	Model	Price	Inputs Bal/Unbal	RMS Watts @ 4 Ω/@ 8 Ω	EQ Bands (input)	FX Sends	Metering/ Type	Weight (lbs)	Dim. W x H x D (in.)	Comments
Carvin	FX1244P	$1679	12/12	500W/340W	4	6	Yes/VU	64	25 x 7 x 25	3 stereo returns and 1 mono return; 4 subgroups and 1 independent mic and line level controls.
	MX640	$529	6/6	300W/150W	2-band	1	No	41	21 x 12 x 12	3 returns
	MX842	$779	8/8	400W/300W	2-band	1	No	45	21 x 12 x 12	3 stereo returns
Dynacord	PSX802	$2895	8/8	2 x 250W @ 4 Ω	3-band	2	Yes/LED	42	20 x 8 x 20	48V Phantom Power, Alps Faders, 2 seperate 16-bit reverb and delay modules. 1203 and 1603 feature 1 x 200W Monitor Amp
	PSX1203	$4495	12/12	2 x 250W @ 4 Ω	4-band	3	Yes/LED	57	29 x 8 x 23	
	PSX1603	$5495	16/16	2 x 400W @ 4 Ω	4-band	3	Yes/LED	70	33 x 8 x 23	
Electro-Voice	61PMX	$1000	6B	200W/125W	2-band	1	No/Clip	25	17.75 x 6.9 x 15.5	All boards have 2 sends per channel (1 monitor, 1 FX) and phantom power to all channels
	81PMX	$1180	8B	200W/125W	2-band	1	No/Clip	27	17.75 x 6.9 x 15.5	
	100M Entertainer	$1980	88	150W/100W	3-band	1	Yes/LED	36	20 x 8 x 19	100M and 200M have two additionol line level inputs for tape, CD, etc.
	200M Entertainer	$2798	88	300W/200W	3-band	1	Yes/LED	38	19.5 x 8 x 18.25	200M has 3-band EQ with sweepable mids and Lexicon® digital effects processor.
Peavey	XR-1200D	$1800	12/12	300W/160W	4-band	2	Yes/LED	55	28.5 x 6.5 x 26.125	16-bit digital stereo effects processor w/128 presets.
	XM-6	$430	6/6	150W/90W	2-band	1	No	33	24.5 x 9 x 11.75	
	XRD680	$800	8/8	150W @ 4 ohms	3-band	1	No/Overload	N/A	N/A	16-bit digital sterea effects processor w/128 presets.
	XR600C	$600	6/6	200W @ 4 ohms	3-band	1	No	41	24.6 x 11 x 11.375	
Roland	PA-200	$1495	6/8	100W x 2/65W x 2	3-band	3	Yes/LED	32	18.125 x 9.25 x 19.72	PA-200 and PA-400 have built-in digital reverb and delay; 2 of the inputs have 3 selectable aux inputs for cassette, CD, etc.
	PA-400	$1795	8/10	200W x 2/130W x 2	3-band	3	Yes/LED	39	19.25 x 9.25 x 19.8	
	CPM-120II	$1195	8U	60W x 2 @ 4 ohms	2-band	1	Yes/LED	19	14.3 x 6.7 x 10.8	Inputs 7 and 8 selectable between 1/4" and RCA

tion to hum, noise, crosstalk and acoustic noise from fans. Be sure the mixer performs to your satisfaction.

Even if the powered mixer's performance is acceptable, consider its applications before making your final choice. If your loudspeakers will be located more than 50 feet away, consider a separate mixer and amplifier. Separates allow you to place the amplifier near the loudspeakers, reducing the need for lots of large-gauge, high-cost speaker cable. Separates can also improve the system's damping factor.

If you decide to use a self-powered mixer, try to keep microphone and loudspeaker cables separated at least a foot apart as they run to and from the stage. Or purchase a special snake that shields the microphone and speaker cables from each other. This will prevent crosstalk and oscillation and will keep your system performance as high as possible.

—Chris Foreman

Chris Foreman is Manager of Kearny Operations at JBL Professional.

Brand	Model	Price	Inputs Bal/Unbal	RMS Watts @ 4 Ω/@ 8 Ω	EQ Bands (input)	FX Sends	Metering/ Type	Weight (lbs)	Dim. W x H x D (in.)	Comments
Ross Systems	PC6400B	$660	6B	400W/230W	2-band	1	Yes/LED	40	21 x 11 x 11	All Ross Systems powered mixers feature built-in monitor sends, effects loops, graphic equalizers, short circuit and thermal protection.
	PC7250	$730	7U	250W/150W	3-band	1	Yes/Bar Graph	48	20 x 7 x 21	
	PC8400	$1250	8B	200W x 2/135W x 2	3-band	1	Yes/Bar Graph	75	21 x 7 x 27	
	PC4110	$390	4B	150W/100W	2 band	1	No	30	17 x 10 x 12	
SoundTech	PC830	$1500	8/8	300 @ 4 ohms	3-band	3	Yes/LCD	60	18.25 x 7 x 20	Features 99 digital effects on board and phantom power.
	PC1250	$2000	12/12	500W @ 4 ohms	3-band	3	Yes/LCD	68	22.75 x 7 x 20	
Studiomaster	82PH	$1495	8B	250W/140W	3-band	2	Yes/LED	39.7	20.7 x 6.6 x 17.5	Both units have on-board, 127 program Digital Stereo Reverb plus Dual 7-band Graphic EQ on the master O/P's.
	122PH	$1795	12B	250W/140W	3-band	2	Yes/LED	60	25.4 x 6.6 x 17.5	
TOA	MX-101	$391	4/4	75W/57W	Hi-Lo	0	No	16.5	18.1 x 6.7 x 8.9	Built-in compressor/limiter, 3-spring reverb, power auto-shutdown and protective circuitry on all models. MX-401 and 601 feature patch bay and allow independent operation of EQ and power amp.
	MX-401	$557	4/4	150W/100W	2-band	0	Yes/LED	22	19.4 x 7.09 x 12	
	MX-601	$971	6/6	200W @ 4 ohms	4-band	6	Yes/LED	40.7	23.03 x 13.1 x 11.89	
Yamaha	EM1620	$795	6/6	200W/120W	3-band	1	No	26	18.9 x 5.2 x 13.6	One monitor send, built-in single limiter
	EM2820	$1195	8/8	200W/120W	3-band	1	No	43	19.7 x 6.75 x 21.6	One monitor send, built-in dual limiters
	EMX2200	$1795	8/4	250W/160W	3-band	2	Yes/VU	57	24.13 x 7.9 x 23.6	16-bit digital processing
	EMX2300	$1995	12/6	250W/160W	3-band	2	Yes/VU	66	30.4 x 7.8 x 23.6	16-bit digital processing.
Yorkville	SP-8	$1799	8B	250W/150W	3-band	2	Yes/VU	44	17.7 x 5.3 x 16.3	
	SP-12	$1999	12B	250W/150W	3-band	2	Yes/VU	51	22.6 x 5.3 x 16.3	
	MP-6	$699	6/7	300W/170W	2-band	1	No	35	21.2 x 9.5 x 9.8	Phantom power, gold plated RCA tape input jacks.
	MP-8	$899	8/8	300W/180W	2-band	1	No	35	21.3 x 9.5 x 9.8	Same as above. Also has a dual amplifier which can be assigned to either Main A + B, Main + Monitor or Monitor.

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IN A BAND VAN

MIC SPLITTER MASSACRE

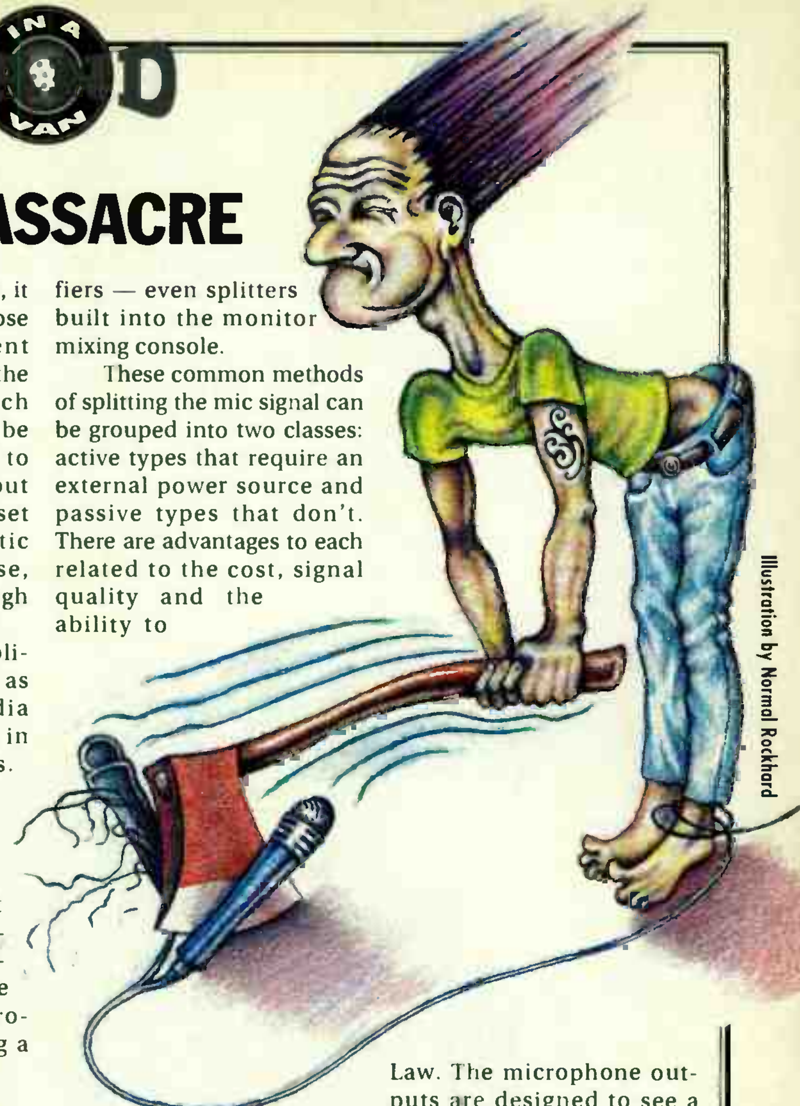

Illustration by Normal Rockhard

MOST OF US started mixing live sound solo. You were in charge of the equipment, system design, setup, soundcheck, mixing during the show, tearing down, loading out the system and sometimes even fixing anything with wires. You were in complete control (well, nearly) of the sound. As the weeks (or years) went by, you became involved in more complex shows and eventually had to include a separate onstage monitor mixer into your system. Now the setup and show had to include another person in the sound crew. You were forced to yield some of your autonomy and even share the mics with another mixer. How can you remain in control of your mix in this situation? Use a mic splitter.

The common application of a mic splitter in live performance is to provide a separate feed of each microphone on stage for: the front-of-house mixer, the onstage monitor mixer and occasionally for a television, radio, satellite broadcast or remote recording truck.

There are situations where a mic splitter is handy even if only one mixer is in use. For example, when mixing a show where there are numerous performers, each with his own equalization, monitor and effect sends and balance. If the same mics or mic lines must be used for many performers, it may be worth splitting those mics and using different mixer input channels for the same mic. This way each performer's settings can be kept from the rehearsal to the performance, without having to repatch or reset controls during a hectic show. (This is, of course, assuming you have enough channels in the mixer!)

WHETHER PASSIVE OR ACTIVE, MIC SPLITTERS WILL BE MORE EFFICIENTLY CARRYING THE WAVES OF THE FUTURE

BY WADE MCGREGOR

There are other applications for mic splitters as well, such as the Media Feed, a common device in many A/V departments. This unit may contain a small mic mixer (or just a mic input) that's then split to a panel with many output connectors. At press conferences, all the journalists can plug their tape recorders into one microphone instead of having a forest of microphones.

SPLIT DECISIONS

The splitter is a device that yields two or more outputs from each input. Each mic cable is plugged into the input of the splitter and the output from the mic splitter will be plugged into a mixer's mic input. An ideal mic splitter will give each mixing console the impression that it's the only console connected to a mic. This will allow each person to create the mix they need for their application without having to rely on someone else's levels, equalization or balance.

The mic splitter can take many forms: a simple Y cable with a female XLR-type connector on one end and male XLR-type connectors on the other ends; a stage box with chassis XLR-type connectors mounted on the box or on a cable fan-out; rack mounted splitter units or distribution amplifiers — even splitters built into the monitor mixing console.

These common methods of splitting the mic signal can be grouped into two classes: active types that require an external power source and passive types that don't. There are advantages to each related to the cost, signal quality and the ability to drive long cables between the mic and the mixer. Let's start with the passive splitter types.

THE LAID-BACK SPLIT

The parallel method is a common form of passive splitter created by simply connecting each pin on a female XLR type connector to each pin (pin one to pin one, pin two to pin two, pin three to pin three) on two or more male XLR connectors. It's inexpensive and will work in some situations but...The mic output sees two inputs (probably halving the load impedance), and also is grounded in two places forming a noise inducing ground-loop. Adding a ground lift switch to each output of the splitter may solve the grounding problem, but not the loading problem. The loading problem is based in Ohm's Law. The microphone outputs are designed to see a load (at the mixer input stage) of approximately ten times their output impedance. This means that if the mic has an output impedance of 150 ohms, the mixer mic input should be at least 1500 ohms. If there are two similar mixer inputs fed directly from one mic output, then the mic will now see a load of only 750 ohms (1/2). This is even worse if more mixers are added: four similar mixers would load the mic with 375 ohms. Some mics may not have a problem with this, but others will change frequency response and headroom before clipping when driving lower impedances. Each mixer input will receive less level from the mic than if the other console(s) were not there, and so there's also a decrease in the signal-to-noise ratio.

Parallel splits won't protect you from cascading catastrophies — a short circuit in one mixer's input cable will result in loss of that mic to all mixers; phantom power from one of the mixers causing problems when it appears at the input of the other mixer; inadvertent crosspatching to a mixer's output, instead of the input, etc. The interaction between mixing consoles will also cause more subtle changes in the mic's output signal. When the gain or input pad is adjusted at one mixer, it's possible that levels at the other mixers will change. This happens because some mixer input designs change impedance at different input pad and gain settings. It's for these reasons (and more) that some form of isolation between the splitters is a better approach. The isolation can be achieved in two ways — a transformer and/or an active circuit can be placed between the mic and each mixer.

Transformers provide a better method of giving each mixer an independent signal from the mic. Some splitters have one output directly connected to one of the mixers, bypassing the transformer, which allows that mixer to provide phantom power to condenser mics and active DI boxes. Splitters that do not have a "direct" output must provide phantom power if 48V power is required.

A transformer is, basically, a device which will reflect an impedance at a ratio determined by the design. This means that the transformer can fool the mic into believing there's only one mixing console, where there are many. Although if one splitter output is shorted out, then there will be level changes, on that splitter channel, in the other mixers. In fact, any change in impedance on an output (as I've mentioned, a possibility in normal operation of some mixers) may have effects on the signal reaching the other outputs of that splitter channel. Transformers do, however, provide very good common mode rejection of the noise induced into the mic cable.

THE AGGRESSIVE SPLIT

The active mic splitter can take two basic forms; analog and digital. Active splitter designs allow complete isolation between each output of the splitter. This can be important. For example, a worst case scenario: the mic snake at the recording truck gets driven over and shorts out to an AC line. Only the recording truck is in trouble; the other consoles have not gone up in smoke (although you might want to check that splitter output)!

Analog splitter designs have improved remarkably over the past decade. There are now models commercially available that provide high quality mic pre-amps and amplify the signals to line level. This allows very long lines to be driven, such as a 500 foot snake in an arena, with less cable induced signal degradation at the mixer. The more expensive mic splitters provide onboard phantom

continued on page 83

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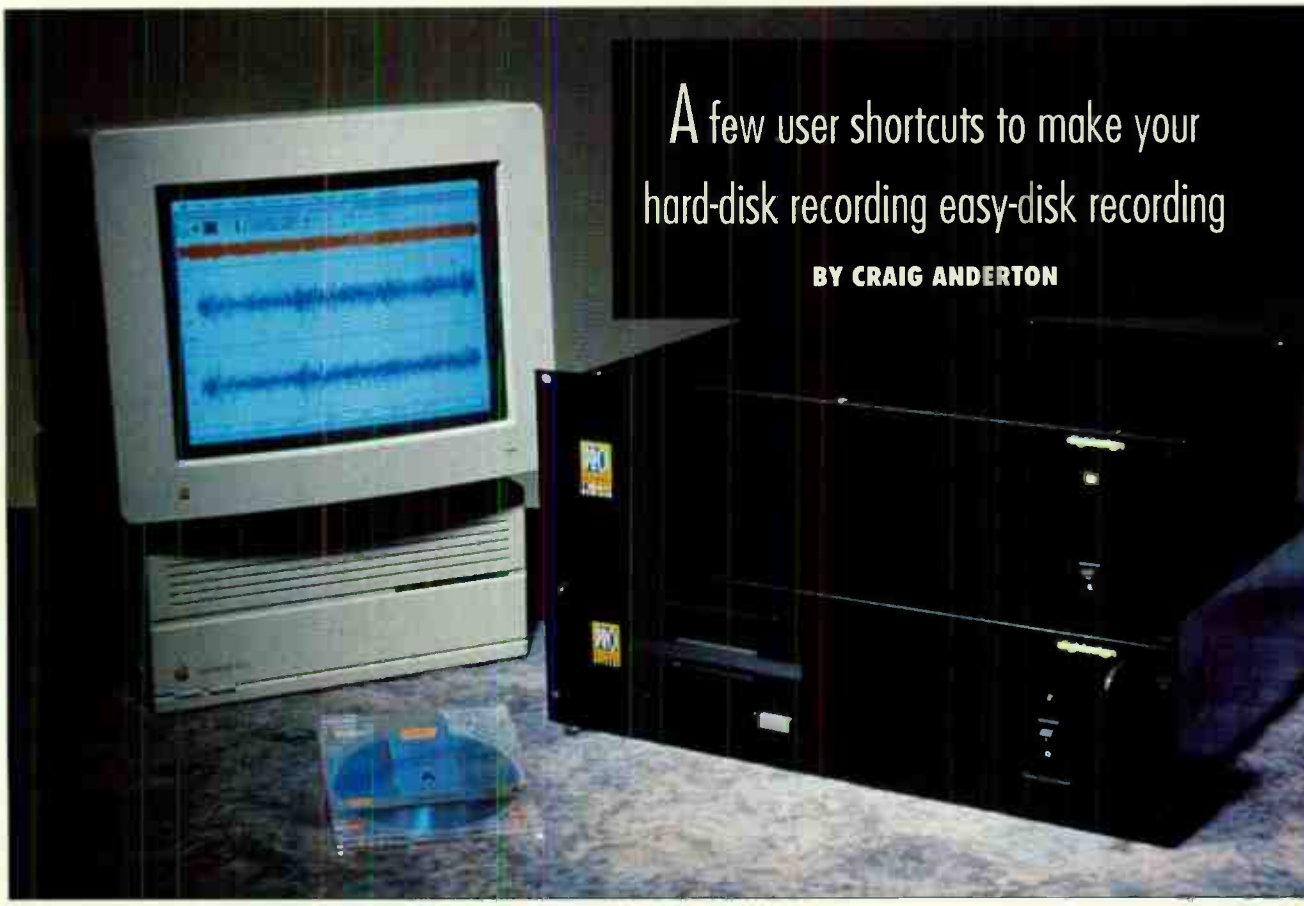

A few user shortcuts to make your hard-disk recording easy-disk recording

BY CRAIG ANDERTON

Photo courtesy of Digidesign

Hard disk recording is great: you can manipulate audio with the same kind of finesse as MIDI data. You can copy good notes and paste them over bad notes, adjust note attacks to fall into a particular rhythmic pocket, and even do quantization and special effects.

If you're just getting into hard-disk recording, the following tips should help you scale the learning curve a little quicker. They're oriented to Sound Tools because of its pervasiveness, but are applicable to other systems too. (Note that Opcode's Studio Vision and Mark of the Unicorn's Digital Performer can do many of the tricks described below, but they are not available for platforms other than the Macintosh II family of computers.)

First, though, it's crucial to understand the concept of a playlist. Unlike tape, which is linear, hard disks allow random access. For example, Sound Tools can trigger audio segments of any length from a hard disk, at the SMPTE times of your choice. Because of random access, you can play segments in any order. The first verse could follow the third verse if you wanted. The playlist is a set of directions for Sound Tools that describes which segments of audio to play, and their SMPTE start times. If Sound Tools is synced to SMPTE along with a sequencer or tape recorder, the playlist makes it easy to fly in chunks of digital audio as needed.

1. FOLLOW THE BOUNCING PART

I use Sound Tools not just as a high-fidelity recording medium, but because of the audio processing power. Unfortunately, it's a two-channel system (the multitrack Pro Tools costs considerably more) and I often want to digitally record and process several tracks of acoustic sounds. Bouncing Sound Tools parts over to multitrack tape is one solution:

First, record a guide track on the tape, if needed. (Since I generally use at least some sequenced MIDI tracks synced to tape, recording a rough mix of the sequence usually serves as the guide.) Record a part into Sound Tools while listening to the guide track, edit the part, then on playback set a SMPTE start time for the part (or assemble a longer playlist from multiple takes) and roll tape. As the Sound Tools part plays back, record it on tape. If it's not quite in sync, set a new start point and try again until everything works together.

Record the next part into Sound Tools, process it, bounce that over to tape, record the next part, and so on. Bouncing to digital tape gives the least degradation, but bouncing to analog lets you apply tape saturation to digital recordings. (If you have limited hard disk capacity, use DAT I/O to back up parts to DAT before erasing the hard disk to make way for new parts.)

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

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

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

Pat Weber, Technical Engineer
Record Plant, Capitol Records,
MCA Records

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IRCLE 34 ON FREE INFO CARD

II. MOVING EXPERIENCE

Destructive edits (e.g., cutting) can take a while in Sound Tools because it has to rewrite the file to "close up" around the cut section. (I used to complain about this until I saw how long it takes a Mac to process video files.) Fortunately, operations that don't change the overall file length work much faster. To cut a piece of data and move it somewhere else:

a) Copy (don't cut) the piece to be moved.

b) Click on the destination start point and select Replace. This overwrites any data that's already there.

c) Return to the original chunk of audio, select it, then choose the Silence command. This erases the data without changing the sound file length.

III. DIGITAL DUET

DATs are great, but if you don't have a second DAT, how do you make a digital copy of a DAT tape? Simple: Use DAT I/O to transfer the DAT contents to Sound Tools (your DAT must have digital outs, of course). Load in a new DAT tape, then transfer the Sound Tools data over to the copy tape.

If your DAT doesn't have digital outputs, don't worry — playing the DAT into Sound Tools through the analog inputs, then bouncing back to a second DAT tape from Sound Tools' analog outputs, is still going to sound pretty good.

IV. CHAPTER AND VERSE

I often record vocals in mono, which leaves one channel of Sound Tools free for nifty effects. For echo, copy parts of the vocal and replace silence on the other track (replace works faster than pasting, and doesn't alter the sound file length). These echoes can then be reversed, EQ'd, set to strange polyrhythms, have levels adjusted, etc.

Vocal doubling is easy to do but, best of all, the effect doesn't have to be consistent throughout a piece. For example, the verse could have a different flavor of doubling (either due to EQ changes, different amounts of delay, etc.) compared to the chorus.

V. CLICKING YOUR CHOPS

When syncing both a sequencer and Sound Tools to SMPTE and creating playlists from chunks of audio on various parts of the hard disk, it's not easy to find a SMPTE start point that triggers the audio chunk at the exact right instant. Trial and error will do the job eventually, but here's an easier way if you're recording mono parts into Sound Tools:

Program a click track into the sequencer and have it trigger a fast-attack sound like a clave. As you listen to the sequence and record your part into one track of Sound Tools, record the clave sound into the other track. (Figure 1 shows what the tracks look like; the click is on the bottom.)

When assembling the playlist, monitor the click coming from Sound Tools and the one triggered by the sequencer, then adjust the playlist segment start time so that the clicks coincide. To hear large differences, it helps to pan the clicks to opposite channels; when the timing gets tighter,

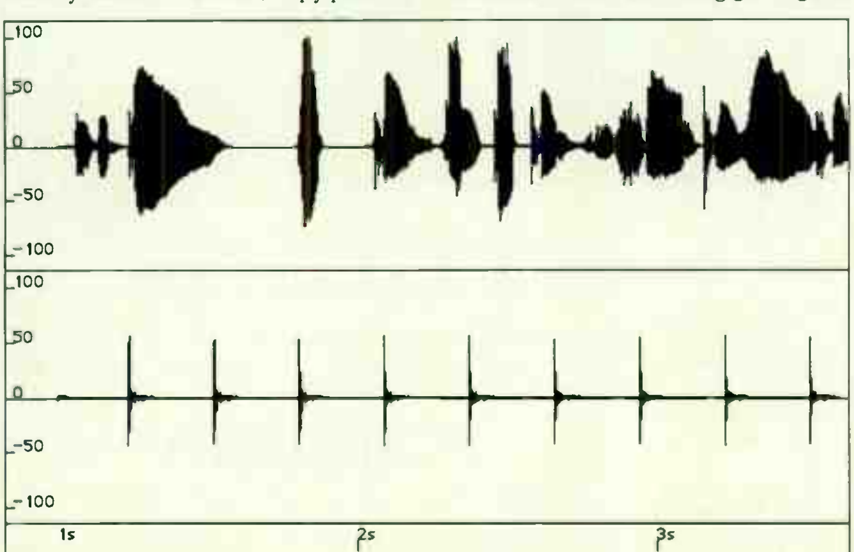


Figure 1: The top line represents the clave sound, while the bottom line shows the click track.

pan the clicks to center so that they flange if they're off. After creating a playlist, mute (or erase) the click track.

If both the sequencer and Sound Tools share the same computer, this technique can still work if you have a tape recorder striped with SMPTE. Sync the sequencer to tape, and record the MIDI instruments and the click on separate tape tracks. Monitor the tape; record your part into one Sound Tools track and the taped click into the other track. On playback, trigger Sound Tools from the tape's SMPTE, and compare the Sound Tools click track along with the taped click track. Adjust the playlist start time so the clicks coincide. If necessary, fine-tune the delay by entering a SMPTE Offset to compensate for delays caused by the SMPTE-to-MIDI converter.

When it's time to mix, use the instrument tracks recorded on tape, and the audio from Sound Tools. Or, record the Sound Tools output into one of the tape tracks, then treat the sequenced instruments as virtual tracks to be mixed along with the tape tracks.

If you have a hard time hearing which track is ahead or behind, the Russian Dragon (from Jeanius Electronics) can show the offset visually, making it easier to "jog" the playlist start point to the proper SMPTE time.

VI. LOOP DE LOOP

One of the best MIDI sequencer features is loop recording, where you can repeat part of a sequence over and over again. As you record a solo, each pass of the loop puts the solo on a different track, so you can record several solos without stopping. When you have enough solos, you can then create an ideal "composite" track from the best bits of the individual solos. Wouldn't it be nice if you could do that with hard disk recording?

Here's how to do it. Suppose you have a MIDI sequencer and Sound Tools, each synced to SMPTE timecode, and now want to record a lead guitar into Sound Tools. Although many sequencers cannot do loop recording when synced to SMPTE, there is a way around that limitation if you have a DAT recorder.

a) Create a temporary sequence that repeats the part of the MIDI sequence that plays behind the solo, and start playing the sequence. As your MIDI instruments play the sequence, record it onto DAT so you have a reference to play the solo against. There are two very important cautions:

i. Sync the MIDI sequencer to the same SMPTE signal that will synchronize Sound Tools and the sequencer during the final mix. This will prevent timing errors later on.

ii. You can no longer change the sequence tempo during the part where you're playing the solo, so make sure it's right.

b) Set up to monitor from DAT (not the sequencer).

c) Put the DAT in play mode and, as you listen to it, start recording the solo into Sound Tools. The DAT will play the "looped" part, allowing you to record several overdubs into Sound Tools. Remember, you're only *listening* to DAT, not sending any of it into Sound Tools. Sync is not a problem if you record and play back on the same machine because the DAT speed shouldn't drift to any significant degree.

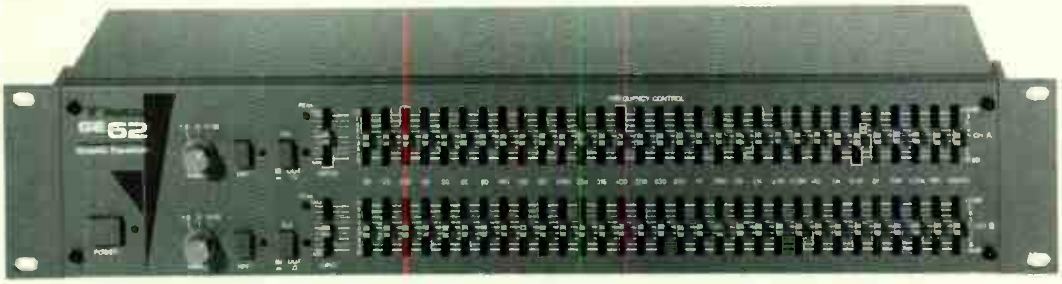
continued on page 107

What is the definition of E.Q.? To us it means...

Extra Quiet

GE-30
Dual 15-band
Graphic Equalizer

GE-31
31-band
Graphic Equalizer



GE-62
Dual 31-band
Graphic Equalizer

VESTAX introduces the GE series of rackmount graphic equalizers. VESTAX offers 3 different models to help musicians create the perfect tonality for their music. A unique 3-position range switch allows you to get even more detailed equalization. Perfectly matched to the VESTAX MR Series of multitrack recorders, the VESTAX GE Series will deliver Extra Quiet, un-rivaled quality.

Designed with recording interests in mind, the GE series also features XLR balanced outputs, enabling them to be used in any live performance applications.

	GE-30	GE-31	GE-62
Frequency Response	20-20KHz	20-20KHz	20-20KHz
T.H.D	less than 0.1%	less than 0.1%	less than 0.1%
S/N Ratio	88dB	88dB	88dB
Total gain	22dB	22dB	22dB
Control freq.	15 Band Stereo	31 Band Mono.	31 Band Stereo
Power supply	AC Adapter DC-15	AC Adapter DC-15	AC Adapter DC-15
Dimensions	19 x 1.8 x 7	19 x 1.8 x 7	19 x 3.6 x 7
Weight	5 lbs.	5 lbs.	9 lbs.

Vestax®

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Fairfield, CA 94585
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CIRCLE 68 ON FREE INFO CARD

Someone Say, "Digital Midio?"

The trials and joys of using MIDI sequencers with digital audio systems

BY HOWARD MASSEY

In just ten short years, MIDI has become a staple in every kind of recording environment. But that's not the only major change to take place, as yesterday's analog tape recorders are slowly but surely being replaced by today's digital recording systems. How do they work together and how can you prevent these dual digital audio revolutions from dueling?

A COMMON GROUND

The key to integrating MIDI sequencing and digital audio recording is the common ground of SMPTE timecode. If your sequencer and digital audio recorder are physically separate devices, you'll need to route SMPTE to both simultaneously. I've had good success with the technique of splitting the timecode audio signal by passive means, such as a patchbay mult or even a Y-cable (don't split it with console bus assignments, since the active circuitry involved can cause degradation of the SMPTE square wave). Alternatively, you can instruct the first SMPTE reader in your system to regenerate (jam sync) the incoming timecode before sending it on to the second device.

Usually, the incoming SMPTE is converted to MTC, which is then fed directly to the sequencer and digital audio recorder. If your MIDI sequencer can't directly lock to MTC, you'll need to use the SMPTE-to-MIDI song pointer capabilities of your computer interface or utilize a dedicated converter such as the Roland SBX-80. Most MIDI sequencers and digital audio systems are capable of punching into record while locked up, so you can even use timecode to automate your punch-in and punch-out points.

DEALING WITH SPEED FLUCTUATIONS

A SMPTE-to-MTC converter will normally expect to receive frame changes every 1/30 of a second. What happens if it doesn't, due to speed fluctuations in the SMPTE playback? Well, as far as the MIDI sequencer goes, there's no big problem; playback will simply speed up or slow down accordingly. Since this causes only a (usually slight) tempo change and no pitch change, you probably won't notice it at all. But digital audio is another beast altogether, and even though they provide two different ways of dealing with the problem, the bad news is that neither solution is completely satisfactory. The first option is simply to ignore any fluctuations by instructing the software to begin playback at the designated frame number and "freewheel" from that point on. This will work okay if your digital audio file or region is relatively short, but if you're playing back a long file (such as a complete vocal, for example), you'll hear it begin to slip out of sync with the continuously adjusting MIDI tracks — and this effect will increase cumulatively as playback continues.

The second option is to use a function called continuous resynchronization (or "locking to tape"), where the playback speed of the audio data is constantly altered to follow changes in the speed of the incoming timecode. This way, even long audio files will remain in sync with changing MIDI playback — but the problem is that, as you change the playback speed of digital audio, its pitch changes also. Depending upon the program material, this pitch change will range from barely noticeable (if you're playing back percussive sounds) to very obvious (if you're playing back sustained notes).

To minimize the problem of speed fluctuation in SMPTE playback, use the most stable

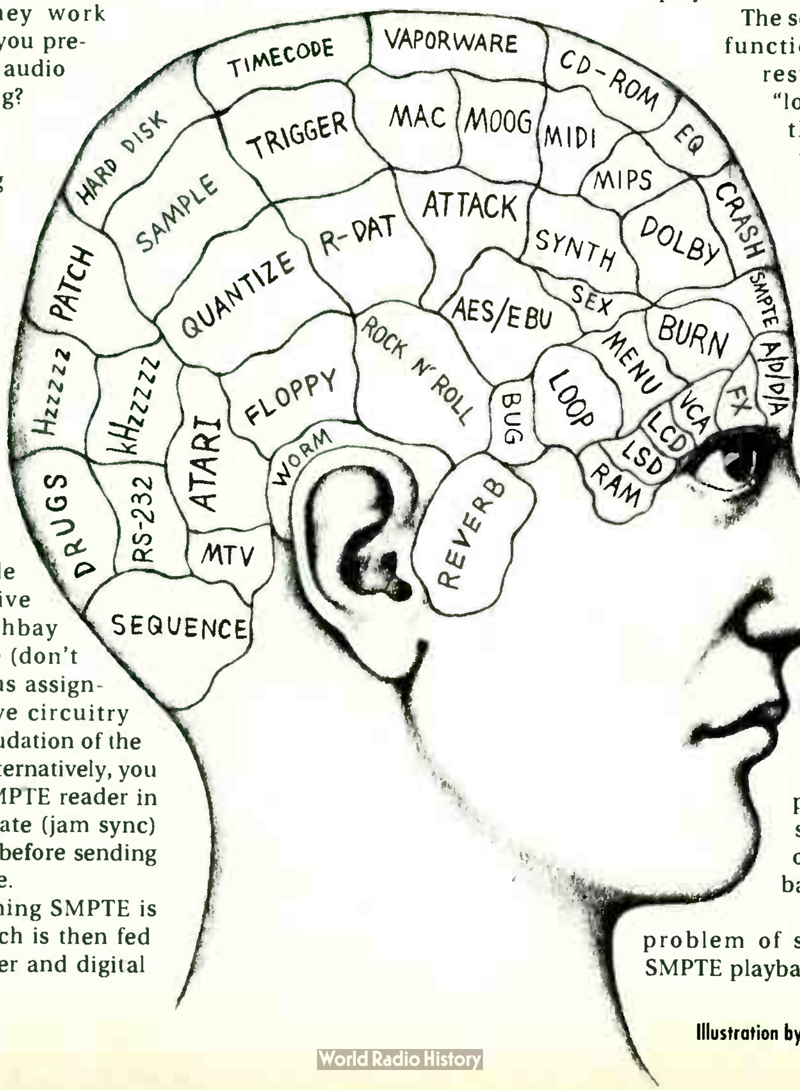

Illustration by Normal Rockhard

tape medium you have for striping — DATs are particularly good for this purpose. If you're fortunate enough to own two DAT recorders, use one for striping and mix down to the other one. If you're using a multitrack digital audio system, use your DAT for timecode and mix down directly to two hard-disk tracks (this is actually preferable to mixing to DAT, because of the error correction used by DAT); from there, you can dub master copies of the final mix to DAT afterwards. If you don't have a DAT (or are working with a two-track digital audio system with a single DAT), you can use a track of a multitrack or two-track tape recorder or even a cassette.

You should also get in the habit of dividing all your audio files into as many brief regions as you can. For a vocal, designate each verse and chorus as a separate region; for an instrument sound, make each four or eight-bar section a separate region.

TWO MINDS SHARING ONE BODY

You can also opt to have your MIDI sequencer and digital audio system be separate components sharing the same computer platform. For example, you might run a Macintosh sequencing program along with a Mac-based digital audio system. Again, the key here is to have both applications receiving the same timecode signal simultaneously. In the Mac, you can use Apple's MIDI Manager to "patch" incoming MTC to more than one destination (most high-level sequencers, including Vision, Performer, and Pro 4 support this). However, there are some drawbacks: First, MIDI Manager eats up a lot of memory and tends to slow down computer performance. Second, running MIDI Manager concurrently with two (or more) professional-level applications requires Multifinder and lots of RAM (4 megs, minimum, in most situations). Finally, MIDI Manager currently does not support the color non-drop video standard of 29.97 frames per second, which will result in slight but noticeable timing errors if you're working with a videotape striped at 29.97 fps and your converter is working at 30 fps.

In computers that do true multitasking (such as the Amiga and the NeXT), you don't need to use a software link like MIDI Manager, but there's still relatively little in the way of MIDI and/or digital audio software for these systems. And, as of this writing, there are no non-proprietary ways of distributing MIDI and timecode signal among multiple applications in IBM systems (even operating under Windows) or the Atari ST.

EDITING IN TWO PLACES AT ONCE

The basic problem with having your MIDI data and digital audio data living in two separate devices — or even within two separate programs in the same computer platform — comes when you want to tinker with your song.

Obviously, whenever you make a change to the timing of your MIDI data (i.e., deleting or adding a period of time), you'll almost always need to make the same change to your digital audio, and vice versa. Most MIDI sequencers allow you to view your data in terms of its absolute time (in addition to the standard bars/beats display), and this display is very helpful, particularly since all digital audio recorders have the same capability. Conversely, most digital audio systems allow you to designate a region of time as being a bar or beat and can extrapolate from that data to show you the bar/beat value for each moment of digital audio. However, unless the original audio was recorded to the sequencer's click track, this can only give you an approximation. It's far better to work with the absolute time display in both the sequencer and digital audio editor, adding or deleting the same size regions in each.

There will probably be times when you need to speed up or slow down the tempo of part or all of a MIDI sequence; this is often required when producing audio for video, where cues have to be precisely timed. Compensating for this in the accompanying digital audio can be tricky, but, if you've only gone up or down by a factor of 10 percent or less, you can successfully use time compression or expansion without significantly degrading the audio quality. Many digital audio recorders allow you to apply this as a percentage of the original time, so you can simply apply the same time compression value to the equivalent digital audio region. Adding ritardi or accelerandi to digital audio, however, is tougher than applying it to MIDI data. Your best bet is to divide your digital audio track into several small regions and apply increasing amounts of time compression or expansion to each until it matches the sequence. Again, use the final percentage value of the MIDI sequencer's tempo change as a reference.

If tracks in your sequencer are being looped, you may need to do the same to accompanying digital audio events by constructing a playlist that has the same region play repeatedly. If you're using a controller to fade MIDI tracks in or out, you may also be able to apply it to the volume of your digital audio events or you may need to construct a fade envelope. If you use absolute time to measure the length of your MIDI fade-out or fade-in, it shouldn't be difficult to match its length in the digital audio.

If time is being added to or subtracted from areas where there is only MIDI and no digital audio, the only editing you'll need to do in your digital audio recorder is to alter the SMPTE start times of the regions in your playlist. Many systems allow you to use a "shuffle" mode, where all later regions are moved forward or backward by the amount of the offset you enter.

ONE MIND, ONE BODY

If you get the sense that editing MIDI and digital audio in two separate systems is not straightforward, you're absolutely correct. Fortunately, all trends point towards the integration of MIDI and digital audio in one system. In the past year, we've seen releases of Macintosh programs such as Opcode's Studio Vision and MOTU's Digital Performer (each of which integrates a professional-level MIDI sequencer with 16-bit digital audio playback and concurrent editing) as well as Passport's less expensive Audio Trax, which works with 8-bit digital audio. Other Mac products include Digidesign's Q-Sheet A/V (which provides a playlist for mix automation as well as playback of MIDI and/or digital audio events) and their new multitrack digital audio system, Pro Tools (which allows for the importation of standard MIDI files and even offers some rudimentary MIDI editing functions). And some dedicated systems like Korg's SoundLink are including a MIDI sequencer right in the same package as the multitrack digital audio recorder.

The main disadvantage to using an integrated system is that you lose the freedom to choose the optimum MIDI sequencer or digital audio recorder for a particular application; instead, you're pretty much locked into using just one program and one computer platform. But there are enormous advantages that far outweigh this one negative.

For one thing, you won't need to route timecode to more than one place. More importantly, by doing all your editing in one environment, you won't have to keep going back and forth between two different programs or devices every time you want to cut, copy, or paste a segment of music.

Studio Vision, for example, treats audio events (which can be entire digital audio files or just specified regions of files) just like MIDI events; they can be placed in the same tracks as MIDI events and appear in both event list and graphic editing displays. Just like MIDI events, they can be "grabbed" with the mouse and moved to different points in time. Each audio event can be assigned volume and pan values, which can be altered in real time with an onscreen fader and/or MIDI controller. They can be quantized, just like MIDI events, and you can even substitute one kind of event for the other. One of my favorite Studio Vision tricks is to build a drum track by using MIDI notes to trigger any handy percussion sounds from the nearest synth or drum machine. Once I've got the rhythms I want, I use the "substitute" command to replace each MIDI voice with custom drum samples which have been ported to the Mac from various samplers via Alchemy or Sound Designer.

Today, we've got the MIDI standard, the SMPTE standard, the CD Red Book standard, even a multimedia standard. Perhaps it's only a matter of time before a standard for the integration of digital audio and MIDI is developed. Did anyone out there say, "digital midio?"

Howard Massey is an author and educator who heads up On The Right Wavelength, a MIDI consulting company. He also runs Workaday World Productions, doing freelance scoring and production work.

Building On A Budget

How to build a powerful project studio for under $20,000

BY PHIL RUBIN

Photo by Peter Monroe

How does a composer on a limited budget start a studio that can compete with the big guys? I knew that the sound is only as good as the writing — you could have all the best and most expensive equipment in the world but the most important thing is the music itself. By putting together good, clean sounds with original writing, you can give your clients the best quality music at an affordable price. That was my goal in setting up Phil Rubin Music. Research, careful planning and a game plan that includes adding equipment on an as-needed basis made it all possible.

I started the studio in my apartment for under $20,000 less than two years ago. Six months later, I was busy enough to move to larger quarters at 157 West 57th Street in NYC — just across the street from Carnegie Hall.

In building the studio I concentrated on the equipment I knew I needed right away and looked for the best way to get it. My first purchase was the board. I chose the Seck 1882 because it's quiet, has a warm EQ, and is very inexpensive — at the time it was just under $2,000.

My first computer was a MacPlus with a 40 meg hard drive ($1,450) and installed Performer software ($350). I've since upgraded to a Mac IIci.

For amplifiers I chose the Adcom GFA 535 (approximately $350) because it is accurate and very affordable. I also purchased a pair of Tannoy PBM 6.5 monitors ($265) for their compactness and great presence. Two dbx 563 dual-channel noise gates ($200 each) were bought to provide cleanliness. We use a Yamaha SPX900 effect processor ($700) and a Mark of the Unicorn MIDI Time Piece MIDI interface ($350).

I have a Proteus 2 which is great for orchestral sounds. It cost about $2,000 at the time. We also use a Roland R8-M ($695), a Yamaha TX81Z ($200 used), a Roland U-220 ($600) and a Korg M3R ($800). Our first sampler was a Roland S-550 ($1,700), since upgraded to an S-750. Since I am a pianist by training, a realistic piano sound was very important to me. I chose a Yamaha CLP-560 digital piano ($1,800 mail order) because it has the sound and realistic action I want.

For mastering, I have a Tascam DA-30 DAT (approximately $1,150) and a Tascam 112 cassette deck ($600). I bought patchbays and wired them up with my engineer. To round out the picture I bought a Sony Trinitron TV (approximately $400) and Sony SLV-686 VCR ($450).

In looking for the best buys I turned to mail order purchasing because many of the prices were substantially lower and I was able to save on sales tax. I also found some of the equipment used — the Prophet 5 and Jupiter 8 — which were added to the DX-7 I already owned.

I trained as a classical pianist but have also toured with the Village People. So although we use a lot of classical-style themes, we also can provide anything from funk to dance to jazz.

Our first clients at the studio were producers of medical films. Soon we got into more complicated commercials which required frame-by-frame scoring so I added a Mark of the Unicorn Video Time Piece ($1,000) which has been a great help.

We share our studio space with L.A. Bruell, Inc., a film and video production company headed by my wife Lucy Bruell Rubin. Through this association we have been working on a lot of health and science projects.

We just completed a tape which is part of a piece for Turner Broadcasting's new cartoon channel. I was given the original recordings of some of the most popular cartoon themes, including the Flintstones and the Jetsons, and was asked to put together a 40-second montage using eight of the themes.

The montage was our first major project to use Digidesign's Sound Tools ($2,000). Sound Tools makes it possible to transpose pieces or parts of a piece without changing the rhythm, making tonal transitions a lot smoother. Our next purchase will be Digital Performer by Mark of the Unicorn which will enable us to add live instruments to our work, storing and editing them in the computer.

If the first two years are any indication, putting together a top notch studio without breaking the bank is an idea whose time has come.

Phil Rubin is the owner of Phil Rubin Music as well as an accomplished session pianist.

PERESTROIKA

50 Years Ago . . .

Allied war planes laid siege to Berlin forcing Georg Neumann and his company to flee for their lives. They set up shop in a small town far from the bombing and, once again, started production. But history would again overtake them. They were inside the Eastern Block and the Iron Curtain was soon to fall.

Time Passed . . .

We knew little about their fate. But with the dawn of Perestroika, we decided to trace this branch of the NEUMANN family tree. To our surprise, the company had flourished, supplying high quality condenser microphones to Soviet Bloc broadcasters.

But more to our surprise, they are still manufacturing the M 7 capsule, the capsule that made the original U 47 the glorious sounding instrument it was.

Now . . . Thanks to Perestroika, you can buy MG microphones from your favorite dealer. You'll be amazed by the sound — and by the affordable price.

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CIRCLE 79 ON FREE INFO CARD

EQ&A

continued from page 20

available from third party companies that will interface with the M-3500. These can be easily connected through the insert jack on the channels. [We invite any third party companies offering such automation packages for the Tascam M-3500 to drop us a note in care of this column.—H.G.L.]

Ken Hirata
Marketing Communications Manager
Tascam

HARMONIZIN'

Q *I enjoyed the Digitech "Vocalist" review [EQ, February 1992]. But I have had a problem using mine since I bought it in November 1991. I send the device into channel 12 of my Tascam M-520 mixer, using channels 14 and 15 for processed vocal sound. Vocalist settings are as such: Input=75 percent; Vocal=off; Effect=100 percent. All I get is garbled sound and signal breakup. Please help!*

Rick Alberts
Virginia Beach, VA

A To achieve the best results from the DigiTech Vocalist, a few guidelines should be followed during the set-up.

Setting the input level is the first, and probably the most important, step. Too much input can cause "grainy" or distorted harmonies. Not enough input signal can cause the harmonies to cut in and out. The Vocalist can receive its input directly from a microphone, or from an auxiliary or effects output on a mixing console. The Input Sensitivity switch on the rear panel should be set to -10 with a microphone, or +4 when used in an effect loop. The Vocalist has a "Signal Lock" LED which lights when it is receiving enough input signal to harmonize with. The input level should be adjusted so that the Signal Lock LEDs light whenever you are singing. The unit also has four headroom LEDs. Only the first one or two LEDs should light while you are singing. If more than two light, the harmonies may begin to sound distorted. In most cases, the input slider should be almost all the way down.

Nect connect the outputs from the Vocalist to your mixing console. The Vocalist has three outputs: the Left, Right, and Line Out. For best results all three should be used. Harmonies one and two appear at the Left out, and three and four at the Right. Connect the Left and Right outputs to two separate channels on your mixing console. The Line Out contains your voice with nothing added to it (dry signal). It is not necessary to use this output if the Vocalist is receiving its signal from an effects send as your voice will be controlled by the mixer channel that your microphone is connected to. If your Vocalist is receiving its signal direct from a microphone, the line out should be connected to a third channel on your mixer.

The Vocalist has two sliders to control the output level. The slider labeled "Vocal" controls the mix of your voice in the Left and Right outputs. In the set-up mentioned above, the Vocal slider should be all the way down because your voice will already be at the line output. The "Effect"slider is the volume of the harmonies at the right and left outputs. This should be set as high as possible.

At this point you're Vocalist is ready to add up to four clones of yourself to your song. Adjust the level and EQ of all three channels, you will hear your vocals come to life.

Bruce Holt
Customer Service
DOD/DigiTech

EMT RETURNS

continued from page 41

quite frequently. But now even my recordings can take on the plate's dense reverb sound.

Can you obtain a reverberation plate or two for your facility and still keep that digital cleanliness you depend on? Should you? The answer to the first question is yes. There are many used plates available out there. At least 4,000 were up and running in this country at one time. The answer to the second is — it depends on your needs and budget. If digital reverb systems, which can offer many types of echo sounds in one box, are suitable for the sounds you require, cool. But if you want one echo that stands out on its own like no other, check out a plate. Until Jim Cunningham's mod catches on, used EMTs are affordable, selling for as little as $1,000 on the current market. And the modification is available for under $75 simply by contacting Jim Cunningham at 708-831-5628.

May your music resound like never before! EQ

PATCHBAY

continued from page 53

Bay #6: Mix and Monitor Buss Utility Outputs and Echo Returns
Mix and Monitor Buss Utility Inputs and Mix Machine Ins
Bay #7: Cue Sends, Echo Sends, Mixdown Machine Outs (Replay)
Monitor and Cue Amp Ins, Copy Machine Ins, Echo Ins
Bay #8: External Equipment (Outboard Gear) Outputs
External Equipment (Outboard Gear) Inputs
Bay #9: External Equipment Outputs and Tie Lines
External Equipment Inputs and Tie Lines

Remember, this array is for 48-track studios. For most project studios, it's perfectly practical to subdivide patchbays — two rows of 48 points become four rows of 24, etc. The present and future size of your multitrack format is a critical issue in determining how wide your jackfield should be. If you're strictly a virtual track operation, the number of console inputs is a good number on which to base the horizontal spread.

Take your time with the planning. Try alternatives. Enter your ideas on your Master Wiring List. And always leave yourself room to grow! EQ

MIC SPLITTERS

continued from page 68

power and even remote gain switching.

Digital audio snakes are now being made that can provide further advantages. Digitized in a form similar to that used for professional DAT machines, the audio may then be multiplexed so that combinations of audio, communication, MIDI and control signals can be sent down a single cable. This cable can be either conventional copper or fiber optic. Digital signals can be controlled, divided (split), combined and monitored through the use of microprocessors dedicated to the system.

These digital systems require smaller cables to carry the signal greater distances — typically, a cable the size of a single mic cord to transmit up to 64 channels of audio. Digital signals also have the advantage of being immune to many of the noise sources that currently plague audio in the analogue domain. We are going to see more of these digital snakes as the audio signal chain becomes digitized from beginning to end. EQ

LETTERS TO EQ

continued from page 12

For example, if the input stage exhibits a common mode gain (attenuation) of -40 dB and a normal mode gain of 26 dB, the CMRR is 66 dB.

In general, we endorse his advice regarding the "pseudo-balanced" connection of unbalanced outputs to balanced inputs. We recommend the scheme for connecting to *transformer* input circuits. For transformer*less* balanced inputs, however, the ground system *does* become a noise contributor to the signal path, via the circuit's common mode impedance.

Common IC differential amplifiers, like the SSM-2141, can handle common mode noise only up to about ±10 V peak. Spikes beyond this, which are quite common in real systems, will "break through," often producing full scale output. This voltage limit can be extended with resistive input attenuators, but signal-to-noise ratio will suffer. Because of the low common mode impedance, these circuits are also exquisitely sensitive to source impedance imbalances. Even a 5 Ω imbalance will degrade CMRR by 20 dB. Think about this for the case of the "pseudo-balanced" connection where the source imbalance can range from 30 Ω for professional gear to over 1000 Ω in consumer gear! The "real world" CMRR will likely prove inadequate to suppress much hum, buzz, or click noise.

A high quality input transformer can match even the claimed "laboratory" CMRR of these circuits, and will do so for voltages to at least ±250 volts. Because the common mode input impedance of a transformer is extremely high, it is much less sensitive to source impedance imbalances. As a bonus, the signal-to-noise performance of the input amplifier is almost always improved as well.

We think our transformers are "the best money can buy" and can challenge any active circuitry for overall sonic performance. For example, the Jensen JT-16-B input transformer has a -3 dB bandwidth of 0.35 Hz to 200 kHz, phase shift of +2 degrees at 20 Hz and a deviation from linear phase of under 1 degree at 20 kHz, THD of 0.003 percent at 1 kHz and 0.036 percent at 20 Hz, and CMRR of over 140 dB at 60 Hz and over 80 dB at 10 kHz. Audio transformers are designed to perform with certain operating conditions controlled, just as ICs are, and well designed transformers, properly applied, are very tolerant of source load variations.

We agree that high quality balanced input stages are more important than balanced output stages. We also think that a transformerless output stage that can turn off the grounded output is a great idea. However, recently popularized balanced line drivers that use crossed feedback attempting to "imitate" a transformer are a very BAD idea. They "dump" distorted currents into the grounded line, causing not only "hot" side distortion, but injection of these "nasty" currents into the entire ground system as well.

We have received numerous calls from confused people who say they read that audio transformers "should be avoided whenever possible," but genuine professional equipment always seems to have them.

The bottom line is that they neatly solve real world problems, without tweaking or excuses.

Bill Whitlock
president
Jensen Transformers, Inc.

We're sure that Marvin will have some more comments on this one. Stay tuned to "Letters To EQ" for the inevitable next installment. EQ

Studio Design at Warp Speed

Captain Kirk's favorite sound effects designer tells how to keep a handle on the design and construction of a high-end project studio

BY ALAN HOWARTH

I've had my home studio, Electric Melody Studio, since 1983, but my workload has grown so much from that time that I had to expand. I felt that I needed more than just add on to my current studio, it was finally time to move into a commercial area.

The Lantana Center in Santa Monica, CA. seemed like the perfect place to relocate to. My studio would be right above Skywalker Sound and adjacent to Digital Magic, a large video post facility. Historically, guys who move near video facilities do pretty well for themselves. Besides, all of my work involves audio for film and video so these companies seemed like good neighbors to have.

With a new location already in mind, it was time to find a design consultant. With a design consultant, you pick up their experience — the lessons they learned on the 300 or 400 rooms they did before yours assure that mistakes won't happen. Through social contact, I met John Storyk, the man who designed Jimi Hendrix's Electric Lady Studios. He had a lot of ideas about what could be done with the space and he seemed really interested in the project.

I originally thought of an office-type floor plan to fill up my 2,200 square feet of space, which I drew out and handed over to John. He transformed my original sketches into five acoustically proper rooms. Three of the five are edit rooms, one is for multitrack mixing and the last is a deluxe sound design/scoring room. I also have two central areas just for machines, one for the edit suite, which contains a Synclavier 3200 and two 16 track New England Digital Post Pro's, and one for the main pre-lay rooms which has a digital 32 track, a 24 track with Dolby SR, 2 1/2-inch four and eight track, a Synclavier 9600 and a 16 track Post Pro.

GETTING IN GEAR

The new studio design is really just an outgrowth of my home studio — with a few improvements. I now have an isolation room, something I always wanted in my home studio. Luckily, all the gear I owned was standard studio equipment, so it wasn't necessary to change any of the designs to accommodate it into the new set-up.

Also, I've always been a big fan of the cockpit theory, I often don't work with an engineer and I like to operate everything myself. I told this to John, and we worked out a studio where everything could be reached easily without needing assistance.

Other plans included small CAD (CTI Audio) 16 x 16 consoles for the editing suite. As an editor, you are mainly just monitoring. If you track, 16 tracks is all that you will ever need because everything is at line level. Also, there is a Neve V-3-48 with Flying Faders in the main room and a Soundcraft 2400 with VCA automation in the pre-lay room.

BUILDING BLOCKS

When construction began, it was necessary to go down there every morning to check on how everything was going. John had drawn the plans, but it was our job to manage the actual building of the facility. Fortunately, construction workers start early in the morning, so I could go down to the site before I started in on any of my own projects.

The construction went pretty smoothly, although I did have to keep

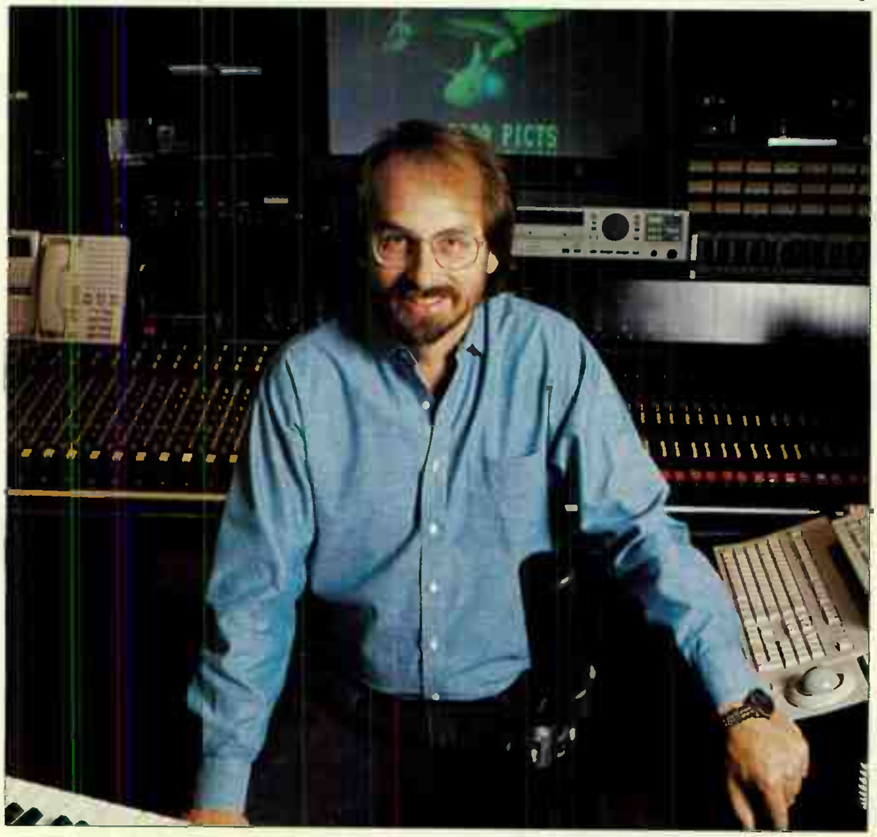
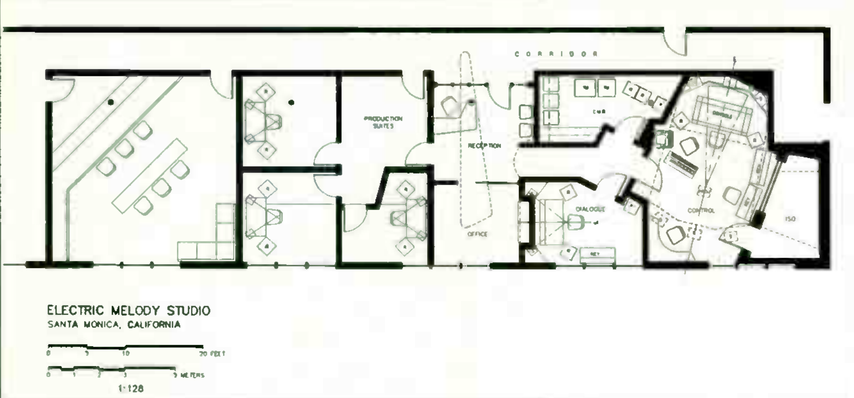


Photo by Jacki Sallow



Floor plan courtesy of Walters-Storyk Design Group

an eye on the workers because I was the one who knew what I really wanted, and they just wanted to get the job done. I remember near Christmas, they had to tear down three days of previous work because they were in a hurry and did it wrong.

Renovations to the plans were ongoing throughout the construction. For example, one of the exterior windows faced the sun for most of the day, so we decided to install double glass to cut down on cooling costs.

During construction, I discovered another big advantage to moving upstairs from a video facility. We were able to run tie-lines between the two facilities, so now we can send our layback over to D-2 or one-inch video all over the tie-lines. This saves a lot of running around for both us and the clients.

By the time you read this, I'm sure I will be working away in the new location of Electric Melody Studios (it feels good to add that "s"). My first few projects will be two feature films for Universal, *Army of Darkness* and *Fortress*, as well as continuing work on the sound for the CBS-TV series "Northern Exposure" and ABC's "Dinosaurs."

One final tip for anyone who wants to have a project studio constructed. I've used my experience to develop a formula that tells you how long and how much money it will take to get the job done. First, double the time you think it will take and then double the money that you expect to pay. Add them together and there you have it — a brand new studio.

Alan Howarth is a sound effects designer who has worked on the sound effects for all of the Star Trek movies, including the latest, Star Trek VI: The Unchartered Territory. He has also worked on HBO's "Tales From the Crypt" and the feature film Grand Canyon. Cresendo Records will be releasing a compact disc of Howarth's sound effects from the Star Trek films early next year.

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Less is More

MASTER MINIMALIST: Pierre M. Sprey

An audiophile recordist takes an unorthodox approach to reproducing the excitement of live music

BY PIERRE M. SPREY

Mapleshade Studio is a colonial-era mansion in Upper Marlboro, Maryland where we make live-to-two-track audiophile-quality recordings of jazz, gospel, blues and chamber music. We record with purist techniques: a single stereo pair of microphones, real room reverberation and minimalist, highly advanced electronics. Mapleshade is also a label. We currently have ten CDs out and over 80 in the can. Artists we've recorded include jazz and blues greats such as Clifford Jordan, Shirley Horn, David Murray, Randy Weston, Walter Davis Jr., Larry Willis, Woody Shaw, Gary Bartz and Sunnyland Slim.

Musicians, consumers and reviewers have all noticed that there's something different about the sound of Mapleshade CDs. The music editor of *The Sensible Sound*, a well-established audiophile journal put it this way: "...an audiophile's dream...this recording has all the qualities a sound buff could ask for — clarity, definition, air, ambience. It almost literally puts the listener in the room with the musicians."

THERE'S NO PLACE LIKE HOME

In running the studio for almost eight years, I've learned that musicians may like good sound, but it's not nearly as important to them as the recording environment. So we've worked hard to make Mapleshade a place where artists feel inspired to play and create. We've eliminated time pressure — no session in this studio has ever been scheduled or billed by the hour. Sessions end when musicians are too tired to play. Unlike most others, this studio has windows and a view of woods and tobacco fields. The room is acoustically live — everyone can hear each other. There are no booths, so everyone can see each other. During a typical two- or three-day session, musicians live and eat with us here at the studio. The result? The musicians are relaxed, take chances and play with real fire. You can hear the difference in the music right away.

I chose this beautiful old house for its acoustics and its studio layout. It has high ceilings and heavy sliding doors that open into a large reverberant hall. Opening and closing the doors meters the reverb in the studio. To get a lot of reverb I put the singer out in the hall and it sounds great.

WEDGED IN

I use a pressure zone microphone (PZM) wedge for essentially all my recordings. My basic design is a V-shaped stereo array of two plexiglass panels, two-feet square and carefully damped to eliminate resonances. On each panel I've mounted a highly-modified Crown PZM 31S. I bypass the transformer by going directly from the mic capsule to my ultra-wide bandwidth minimalist mic preamps. There's one volume control per channel. I avoid mixing consoles like the plague — their long, tortuous signal path can't help but to degrade the sound. My high-end Audioquest mic cables are deliberately under 25 feet long. The difference in sound compared to the usual 100 feet or more of commercial mic cable is startling.

The PZMs reproduce piano and drums better than any mics I know. The percussive impact of the hammers hitting the piano strings really comes through. The studio has a 1911 Steinway Model O piano restored to perfect condition by Potomac Keyboards. This instrument has a fabulously rich overtone structure. It's so responsive that the tone changes more with the touch of different players than any piano I've worked with.

Drummers are knocked out by the accuracy of the PZM sound. The mics are so fast you get the real impact of the sticks on drum heads or the airiness of brushes on cymbals without any audible time smear.

For the horn soloist, I often use a Sound Grabber PZM, much modified with the helpful advice of Crown engineers. You'd be amazed at how much more of the horn's breathiness and overtone detail you get compared with even the best of conventional big diaphragm condenser mics.

Frequently I use Josephson microphones for vocalists and strings. They're the best sounding omnis I've heard, warm yet quick and detailed. David Josephson built me a "purist" mic capsule, removing the protective grille and the resonant chamber over

MAPLESHADE STUDIO: Home Sweet (Sounding) Home

the diaphragm (intended to extend sine wave amplitude response to 18 kHz). Even though this "purist" capsule no longer measures flat to 18 kHz, it is by far David's best sounding, fastest mic — particularly in the high treble. Just another example of the uselessness of sine wave frequency response specs.

STUDIO DEPARTMENT

I use a thoroughly reworked Sony TC-880 two-track tape recorder running at 15 ips without noise reduction. I've applied constrained-layer damping to every resonant element in the chassis. George Weber of P.I.E. has upgraded the tape path and electronics, with more mods in the works. I'm desperately trying to find another 880 for back-up. Although it's an analog recorder, by recording hot there's no hiss problem.

I use headphones for monitoring the live mix, because if I used a booth and monitors, I'd have to run long, sound-degrading mic cables and balancing transformers. During sessions I engineer from the doorway to the studio. That let's me remove the headphones periodically to check the sound of the live music against my mic feed. You can't do that in a glassed-in booth.

We do playback in a separate listening room. The playback system uses a high-end tube/MOSFET amplifier (custom-built and designed by Foster Blair and Jim Nestell) plus Martin Logan CLS II electrostatic audiophile speakers that I've modified and then mated to two Rohrer subwoofers. The Rohrers are seven foot tall cylinders good down to 16 Hz. All the speaker cabling and interconnects were designed by Ron Baumann and me, based on two years of listening tests. We are currently producing them for the professional and audiophile market.

If you want clean, analytical sound to let you really improve mic placement and electronic component selection, electrostatics are wonderful. They're far better than any studio monitors around.

To make digital masters for our own Mapleshade CDs, we do our A/D transfers at Bob Katz's Digital Domain in New York, using his custom-designed, 128x oversampling A/D converter fed from our Sony tape recorder, using only first generation master tapes. These analog tape to digital transfers sound better to me, with more warmth, ambience and detail, than any direct-to-digital recordings I've tried or heard.

ON THE JOB

I usually position the musicians roughly in a semi-circle around the stereo mic wedge. The piano (without lid) is in the middle, with the wedge pointed straight into it. Because the string bass is relatively quiet, it's placed very close to the wedge. Drums are usually five to six feet away. By miking drums at a natural listening distance, the engineer's preferences aren't imposed on the drum mix. The master tape mirrors what the drummer played and the wedge gives the drums a very nice stereo spread.

Some musicians are placed closer to the mic wedge, some farther away, because I get the mix by adjusting physical distance, not by knob twiddling. We start sessions by spending an hour or two setting up, experimenting with positions. We do a trial take, then we all move into the listening room. The musicians critique the mix. We go back, tweak the positions of the instruments, do another test take. Another two or three iterations and then we've got a mix. From then on, I monitor on headphones.

The sound of live-to-two-track analog is far cleaner, more transparent and more dynamic than a multi-track mixdown.

At times, the horn soloist or vocalist will need their own mic. For saxophones, I now usually use a pressure zone mic of my own design mounted on a single two foot square plexiglass panel. A good deal of the beauty of the sax comes from the mouthpiece and the body, so I place the panel vertically in front of the sax to pick up all that. When I use the small Sound Grabber PZM for singers, I usually point the mic's flat plate at the singer to pick up the bass in the voice. But if a singer has too much huskiness at the bottom, I turn the mic plate perpendicular to the singer to roll off the bass — that's natural acoustic EQ and, to my ear, sounds better than any mixer EQ.

Recording in this unconventional, purist way takes no more total time than recording multi-track and then mixing down. But the sound of live-to-two-track analog is far cleaner, more transparent and more dynamic than a multi-track mixdown, whether analog or digital. And sticking to a single stereo pair of mics gives a terrific sense of real depth — real three-dimensionality.

In addition to helping design the F-16 and A-10 jet fighters, Pierre Sprey is currently working on high-end turntable design and the design and manufacture of audiophile electronics, speakers and cables/interconnects. He has engineered over 90 albums, both for Mapleshade and for four other labels.

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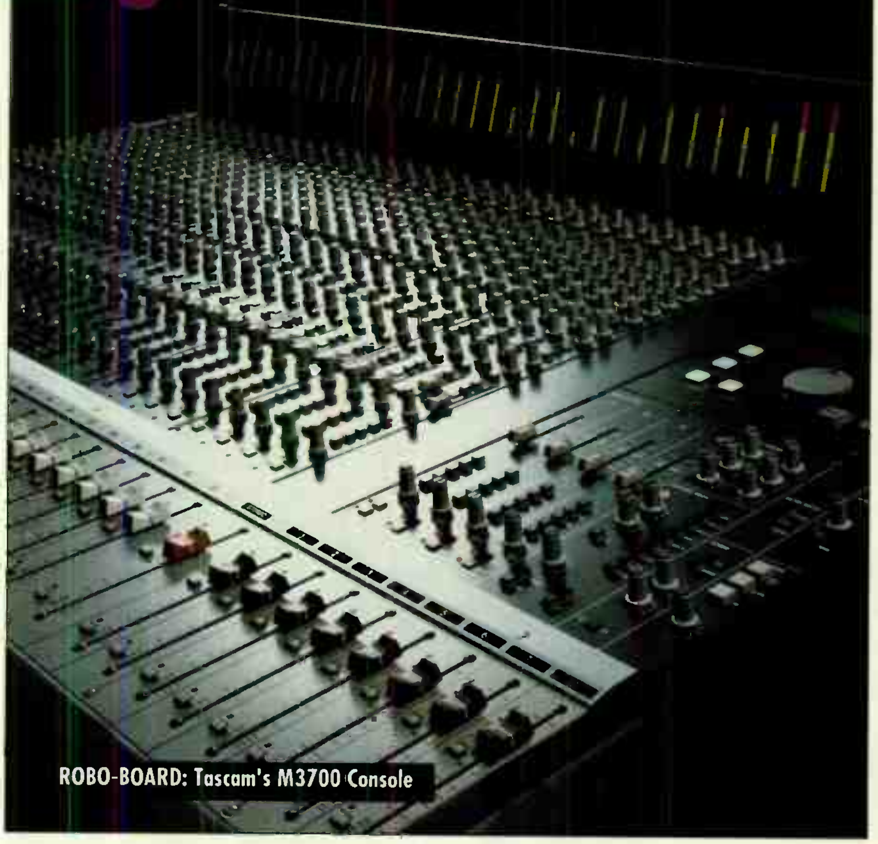
Robo-Recording

Welcome to the age of budget console automation

BY J.D. SHARP



ROBO-BOARD: Tascam's M3700 Console

You can keep your Flying Faders. And your Total Recall too. For that studio on a budget, or that B room with down-and-dirty rates, lower-cost console automation options are currently available (or will soon be offered) that can assist your wasted brain cells during that mega-marathon mix.

Once a luxury onboard SSL and Neve desks only, console automation now belongs to the people, taking a variety of forms that include MIDI muting, outboard fader-controlled units, computer-controlled devices, and built-in fader and mute automation.

The most basic and cost-effective form of budget automation is mute control, which is simply the mechanized turning off and on of channels. The primary benefit of this type of system is that tape tracks of instruments that are not active at a particular moment can be cut off, eliminating the residual noise they would otherwise contribute. But there are other tricks that make muting systems beneficial; you can "mult" a signal into two or more channels and have different effects or EQ set up for each, and use automated muting to apply and remove changes at the appropriate place in a mix, completely hands-free. This works great if you have enough spare inputs to pull it off.

Mute automation has been incorporated into quite a few mixers lately. Tascam offers a muting system that incorporates 99 separate "snapshots" of mute settings, accessible by MIDI patch changes, as well as direct MIDI control over individual mutes. You'll find it in their MM-1 keyboard mixer, as well as the recording-oriented M2516 and M2524 consoles, and the 688 MIDIStudio. Studiomaster was one of the pioneers in this area, and MIDI mute is now standard on their Proline Gold and Mixdown Gold four- and eight-bus recording consoles. Soundtracs has included MIDI automation for some time in several ranges of consoles, and it is prominently featured in their new Solo MIDI series of eight-bus multitrack consoles, covering not only main and monitor inputs for each channel, but also all subgroups as well as effect sends and returns. You'll find a similarly comprehensive system built into Allen and Heath's new GS3 console. And the Alesis recording console shown in prototype form at this year's winter NAMM show will fully implement MIDI muting — but don't expect it for some time.

Ah, but mute control doesn't quite satisfy those who crave automation for continuous and subtle level changes. Read on.

OFF THE BOARD

Moving up the scale of complexity we arrive at systems designed to be controlled either by a computer, an external set of faders, or both. Four products fall neatly into this category. The Niche ACM Audio Control Module ($495) can be used with any mixer. It is a one-rack-space module containing eight independent gain elements that can be addressed by MIDI continuous control information. External fader units such as J.L. Cooper's Fadermaster or Lexicon's MRC can do the job, or sequencer programs that offer graphic "faders" are equally suitable (and nearly all top sequencers now have this feature). Roland offers a set of eight multi-function faders as part of many synthesis and sequencing products including; the SC-155 Sound Canvas, the JV80 synth, and the MV30 sequencer/synth.

Perhaps most intriguing is a product that Niche has announced: the Mix Automation Station — also known as Fader Monster. This system provides 16 faders, each with a

COME FLY WITH ME: The FaderMaster from J.L. Cooper

mute/solo/write-safe button, and system functions that include Master/Safe, Group, Snapshot, System and Select. These double as Rewind, Stop, Play, Fast Forward and Record selectors. A Jog Wheel and "soft" data entry keys are also furnished. These functions cover all aspects of a complete fader and mute automation system, as well as providing a control panel for computer-based recording and sequencing.

Niche's ACM doesn't use VCAs, but rather resistor arrays with very small increments between steps (many more than the 128 available levels provided by MIDI; the "in-between" steps are interpolated by the ACM). J.L. Cooper's new MixMaster MIDI Controlled Mixer ($495) can also be used with any mixer and is similar to the ACM, but employs VCAs which you can see as an advantage or disadvantage according to your own preferences and prejudices:

Advantage: no "steps" between levels, thus no "zippering" when faders are moved rapidly and radically.

Disadvantage: VCAs provide some noise, distortion and, arguably, coloration (although J.L. Cooper's specifications are most impressive). At any rate, both units contain eight pairs of 1/4-inch input jacks, and are usually patched across the insert points of the mixer for quietest operation.

A newcomer to the automation field is CM AUTOmation. Their MX-816 unit is quite similar to Cooper's in overall concept (it uses dbx VCAs) and it provides the same level and mute functions, but with some extras. It's available in both eight- and 16-channel versions (both fit in one rack space), and the eight-in version can be inexpensively expanded to 16 via an expansion board. On the feature side, there are 100 "snapshot" memories that can be recalled with MIDI patch changes, and 32 different fade time variations (zero to 30 seconds) between patches. A joystick or mod wheel can also be used to manually fade between two patch settings, a neat trick for live applications. Like the other units, continuous controller and note number information is used to activate level changes and mute info. Published spec's are notable, with signal-to-noise ratio better than 95dB and 98dB of attenuation provided at 1 kHz.

Greg Mackie has come up with a similar solution for his CR1604 mixer. He's announced an add-on module that fits right into the board and provides 16 MIDI-controlled VCAs that will allow full level and mute automation via MIDI (but not from the built-in faders). It's like having two J.L. Cooper MixMasters built in. Projected price is $700. You can look forward to some sort of similar system as an optional offering when Mackie's eight-bus recording mixers come to market later this year.

The last product in this category is Mark of the Unicorn's Mixer 7S. It features seven stereo inputs each with a stereo noise gate, two stereo effect sends and returns and programmable two-band EQ per unit. A desk accessory offers a graphic representation of

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FOUR ON THE FLY: Lexicon's MRC External Fader System

all functions, and all parameters can be varied by MIDI info. Multiple mixers can be used together. While not as quiet as the Niche and J.L. Cooper units, the additional features can be very useful, especially in an environment where there is no need for an elaborate mixing console and you want the computer to do *everything*!

YOU CAN (ALMOST) HAVE IT ALL

The dream for many engineers is to have all these functions neatly built into the console. After all, the simplest interface is to grab faders and start moving them, without having to refer to a computer screen or alternative set of faders. This is where things start to get expensive, but help is on the way.

Currently on the market is Tascam's M3700 console, available in both 24- and 32-input versions. It has a comprehensive VCA-based fader and mute automation system built in, along with a 3.5-inch floppy disk drive for storage and any mix can be updated indefinitely. If there's a shortcoming to the M3700, it's that Tascam didn't implement MIDI communication completely; the console uses 256 levels internally, but MIDI can only handle 128 (or so it seems; read on!). So the MIDI out is just to provide information for a graphic display of fader levels, not for off-line merging and cutting/pasting of mixes. But these consoles represent a breakthrough, and have the advantage that no additional (expensive) computer equipment is required to enjoy the benefits of automation.

Soundcraft has put together an automated version of its Spirit Studio mixer that should be available by late summer or early fall of this year. Like the Tascam M3700, the interface to the system is via the built-in channel faders and mute buttons, but the "brains" are external; the console transmits its changes over MIDI. If you dedicate the MIDI output to its own port, a full eight bits (and thus 256 levels) of information are sent and received, so Soundcraft has managed to put one over on the MIDI spec! Utilization of this extra bit requires the use of either Soundcraft's proprietary software, or the Macintosh or Atari version of Steinberg's CuBase sequencing software. Because CuBase also offers complete support of MIDI Machine Control (MMC) protocol as well, it is also possible to autolocate and set track record/play statuses on a connected tape recorder (if it has an MMC interface) *and* completely automate levels and mutes on the Spirit console — all from the computer. Since mix information is stored as MIDI sequence data, parts of different mixes can be cut and pasted or copied. This provides the "mix merging" functions that have been the exclusive domain of high-end automation systems and big budget studios.

For the future we can look for built-in level and mute automation to become commonplace and ever-more affordable, while the power of automation will be extended to monitor inputs, equalization, signal processing and all sends and returns at the upper end of the market. Meanwhile, there's nothing to keep you from jumping in today, especially because computer-based systems have brought the price of automation down to the level of a half-decent effect. EQ

Tech Talk (Blah, Blah, Blah)

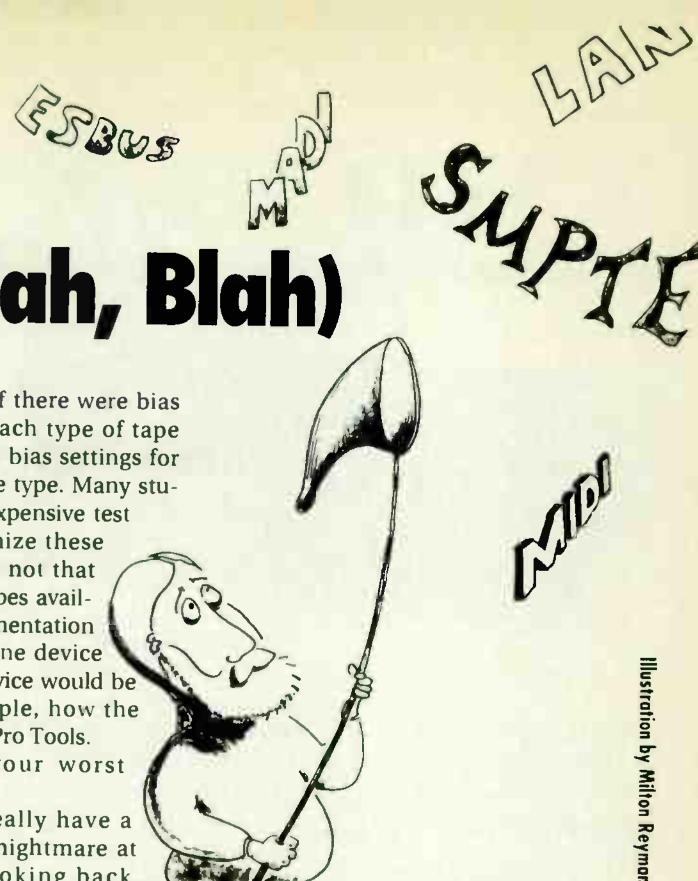

Illustration by Milton Reyman

Two old tech friends talk turkey, trends and troubleshooting

BY RICHIE MOORE, PH.D.

I recently ran into an old friend, Mick Higgins, at a technical seminar at the Studer offices in L.A. on the A820 and A827. Mick is an expatriated Brit who originally came over with Trident consoles back in the late-70s. Today, he is the technical director at Michael Omartian's Sound House in Los Angeles. He and I worked together at the Plant Studios in Sausalito, California for a few years and I learned a lot from him during our time together. I'll never forget troubleshooting my first Trident mic/line module. I asked him what potential trouble sources might be among the 1,100 components on the module and he answered, "There is nothing much on there that can go wrong." I love English wit.

Getting together we mused about old times. I put on my reporter's hat and did some interviewing. At some points, unfortunately, I couldn't keep my own mouth shut. The pow-wow between Mick and Rick (that's me) was preserved for posterity on audiotape (not multitrack Studer, I might add). And I began by asking the first question that came to mind: What major problems of a technical nature are you facing at this time?"

MICK: Probably if I have a 'major problem' at this time, it has been the mechanical inconsistency of some of the new high bias tapes available (no names mentioned!!). Solutions range from 'go with it' to 'try another batch.' I sometimes wonder how much tape manufacturers talk to the machine manufacturers when they design new products.

RICK: Communication and documentation are my major problems. I agree that there should be communication between the manufacturers. It would be wonderful if there were bias charts available for each type of tape showing the optimum bias settings for tape type and machine type. Many studios do not own the expensive test gear it takes to optimize these settings and there are not that many tape machine types available. Real world documentation would be nice. How one device works with another device would be wonderful. For example, how the MicroLynx works with Pro Tools.

RICK: What's your worst technical nightmare?

MICK: I don't really have a worst case technical nightmare at the moment, but looking back, both here in L.A. and in San Francisco, I would probably say the quality of the AC power would have to come under that heading. Here, at Sound House, I'm lucky. It's clean and consistent. We checked before we built the place! But in this age of MIDI, SMPTE, Digi-this and Digi-that, it is imperative, like never before for a facility to have good clean power everywhere, not just in the production areas.

RICK: I'm still having a repeat nightmare left over from the San Francisco Earthquake of '89. This is having a major technical disaster of Biblical proportions and not being able to get any support from the manufacturers. Clients want to keep working no matter what. But when everything goes down for one reason or another, and everyone is off at the AES in New York and it's Friday night, you are a little helpless. My point is that when night falls and the weekend is here, studios don't stop working. When you have megabucks invested in a studio, 24-hour support and parts on the weekends is real important.

RICK: What solutions of a day-to-day nature would you most like to see?

MICK: Other than the clean power issue, I'm glad to see manufacturers are attempting to implement MIDI and E/BUS and all other digital interfaces that allow extremely high tech expensive equipment to talk to one another. Now, if we could just get all this MIDI stuff to come out analog at +4!!

RICK: I am in favor of the +4 dBm standard also. I really would like to see all the synths standardized at one line level that is repeatable from manufacturer to manufacturer. I would also like a firm wiring convention for polarity. I don't care if it is Pin 2 or Pin 3, but that everybody does it the same. I would also like to see a standard multi-pin connector of the ELCO or DL type, with a standard pin configuration. This may be asking too much.

RICK: What items of manufacturer support do you like the best, and the least?

MICK: What I like best is the communication. A company like Studer lets you know what's happening with your equipment, what the options are, when the new software will arrive, if it's free or will be charged for, what known bugs exist, etc., etc., etc. — i.e., COMMUNICATION. What I like the least are manufacturers that sell you a 'new hard disk driven, automated, flying, integrated, self-aligning, digital audio stuff that costs various parts of the studio owner's anatomy. They tell you that all the software updates are free, and then define 'free' as the time it takes to bank the check for your new acquisition.

RICK: I agree and I like to see manufacturers try to go to any lengths to support a product. Studer and SSL are the two that come to my mind. What I like least is promises of fixes or future enhancements that never materialize, and when they do show up, they are not what was represented. I currently have a piece of gear that still does not have properly operating software. And the software's promises were why I purchased it in the first place. Now the warranty is about to expire.

RICK: So, where do we go from here?

MICK: To a degree, we sometimes forget that we are in the communications business, no matter how high tech it gets. If you get the chance to have some input on a device, do it nicely. I'd like to see less carts before the horse, (i.e., equipment doing what it is supposed to do, not what it might do someday). Also a more 'back-to-basics' attitude. By that I mean good tape labeling, proper tones, labeled record pads, correct leadering, etc. The ability for second engineers (and some first) to write things down in a legible manner.

In a day and age of fast high-tech development, where today's hot toy is tomorrow's door stop, where LAN's, DAT, SMPTE, MIDI, MADI, ESBUS and PASC are flying around, the ONLY person who is expected to know everything is the maintenance tech. Since it is just about impossible to keep up with every new development every day, the time taken to just read up on everything would prohibit you from getting much done. Dealing with all this requires not just a technical approach, but a human one as well.

We forget that we don't just fix equipment, we interface the equipment with the user. In the coming technological eras, people skills will become, to my mind, more important than ever before. You can't know everything, but with a little understanding and knowledge of a particular situation, any problems or difficulties that might occur can be alleviated. If that fails, I'm taking my two cats and my Miata and I'm going to move to Oregon and make beer.

RICK: Any closing words for the *EQ* audience?

MICK: As a mixer, turned tech, I appreciate the technical side of things; but being a studio technician is not a glamorous job. When everything works, you're invisible and not very well paid. When something breaks, it's your fault, and you become an overpaid low-life. When a studio is in financial trouble, the first thing that goes is the maintenance. Yet a poorly functioning facility has little chance for success. This is one of the little ironies in studio life. EQ

Blues for Michelangelo?

Can computer viruses affect production facilities? You bet

BY MARTIN POLON

Who or what is Michelangelo? Is he the Michelangelo Buonarroti — the renaissance artistic genius who painted the ceiling of the Sistine Chapel? Is he one of the stalwart Teenage Mutant Ninja Turtles making the world safe for pizza? Or is Michelangelo the name of an especially dangerous computer virus, written by a Scandinavian hacker for the thrill of infecting the world? The answers are yes, yes, yes, and all of the above.

Now, the next question is what does this have to do with personal and project recording studios? Since this "excellent!" audio publication rarely addresses the issues of Italianate church art or the consumption of cheese, tomato sauce and bread by large green slang-speaking reptilians, one can only be left with the much media-hyped threat to computer data worldwide, posed by computer viruses in general and Michelangelo in particular.

Now many of you may ask "What does this have to do with music and recording? After all, aren't computer viruses a threat to white collar businesses or government agencies and their important (boring) information? What possible value could anyone find in destroying electronic music and disrupting recording environments. And why would they?" A partial answer can be found in the story told by John A. who had a particularly unpleasant interaction with our boy "Mikey."

A TAXING LESSON

"I had taken Friday off from work to finish editing my electronic music composition, which I had recorded via MIDI over the previous couple of months. I had everything stored on hard disk on my PC and I was using this digital audio record-and-edit software. I also use the computer for other things and had recently done my income taxes using a new software that had been loaned to me by a friend. I turned the PC on as I always do, but then all hell broke loose. It was a disaster. Everything on my hard disk was trashed and the drive itself crashed. A computer-savvy friend came over and tried to restore some of the hard disk memory but it was too late. He said I probably 'picked up' the virus from the tax disk."

Lesson One — Computer viruses are totally without discrimination as to which machines and software will be wiped out and/or wiped clean. If a virus manages to get onboard your computer, it will reside silently in some part of your system until it is triggered by the clock on your computer or by a function you perform or the number of times you do it.

Lesson Two — There is precious little that you can do that will give a 100 percent guarantee of freedom from your personal computer virus infection. There are an estimated 1,000 software-borne computer viruses in circulation at present for MS-DOS IBM or compatible systems and a fair number for the Macintosh as well.

Lesson Three — Michelangelo is only one of these many viruses, yet it accounted for about 25 percent of all virus crash incidents reported during the past 12 months. It is designed to destroy data and gains access to your machine when a "foreign" floppy disk is introduced. Other dangerous viruses can be passed inside of files from on-line bulletin board systems.

When you consider that a recent survey concluded that 80 percent of all project studios own or have access to a personal computer of some kind, and at least 25 percent of these systems are being used to store, edit, compose or analyze music — the danger from computer viruses becomes very real indeed. One form of protection is achieved by a minimal use of disks from outside or "foreign" sources. This is only partially effective because Michelangelo was spread in some cases via so-called "safe" sources, such as software maker's program disks or by being pre-loaded on blank disks.

HEALTH INSURANCE

A more definitive mode of protection is the purchase of virus exterminating software. These are capable of identifying the presence of a virus and then eradicating the offending bits and bytes. All your old disks must be checked and every time a new disk is introduced to your system it has to be "scanned" too.

With these precautions most project studios can operate without too much fear of the viruses. But it does give pause to consider the form of punishment the Saudi government is rumored to be considering for the hacker who created Michelangelo. The Saudi's have always favored severing various body parts to match the crime. Pickpockets lose their hand, rapists lose their...well, you know. And a virus hacker could lose his "software'" and "hardware" in one fell swoop!

Martin Polon is the principal of Boston-based Polon Research International (PRI). PRI forecasts the electronic entertainment industry for the financial community, with a special focus on the audio business at all levels. Polon is a 14-year veteran of service to the Audio Engineering Society (AES) as a national officer.

More Than Meets The Ear!

Photo by Peter Monroe

With distortion, knowing your THDs isn't enough.

BY LEN FELDMAN

Last time, I talked about total harmonic distortion — the kind of audio distortion that most manufacturers of pro audio gear like to specify in their brochures because the numbers always look so good. But there are other kinds of distortion that are far more obtrusive and annoying than total harmonic distortion, or THD.

INTERMODULATION DISTORTION

This type of distortion, usually abbreviated as IMD, is a lot more audible for a given percentage than THD. After all, harmonic distortion consists of multiples of the frequencies contained in the original music or speech. And since music and speech already has its own natural harmonic components (that's why a trumpet playing middle C doesn't sound like a piano playing the same note), adding a few extraneous harmonic components may change the timbre of the instrument or instruments, but it's not all that unpleasant unless present in horrendously large amounts. Quite a different situation arises when we talk about IMD.

When several musical instruments are playing together they produce all kinds of frequencies at the same time. A bass fiddle may be emitting a 100 Hz note while, at the same time, a keyboard player in the band may be playing a note seven octaves higher, at around 6400 Hz. No problem as long as both notes are reproduced properly with no new frequencies added. But, unfortunately, when you have a non-linear transfer system what you may get are some quantities of unwanted frequencies which are, in fact, the sum and difference frequencies between the two desired tones — in this case 6400-100, or 6300 and 6400+100, or 6500. If those extra frequencies are present, even in small amounts, the audible effect can be very annoying because those frequencies make the whole thing sound "off tune."

Test instruments designed to measure IMD work somewhat like THD meters. In the case of IMD measurement, though, the test signal consists of a low frequency (often 60 Hz) mixed with a relatively high frequency (usually 7000 Hz) in a ratio of 4:1. The test instrument uses the composite waveform to set up the reference 100 percent level and then eliminates the two fundamental frequencies and reads only the percentage of those unwanted sum-and-difference components. This type of IM distortion is often referred to as SMPTE-IM Distortion. (SMPTE standardized the tests for this particular form of distortion.) Let me tell you that if 1 percent THD is only barely audible, the same percentage of IM distortion will be audible even to tin-eared listeners.

TWIN TONE DISTORTION

A form of distortion that's even more insidiously annoying than the SMPTE-IM form of distortion is the so-called twin-tone IM distortion. Again, the non-linearity of the offending piece of audio equipment causes a "beating" of two or more frequencies, but this time the two frequencies may be close together, so the resulting unwanted "beat" frequency may be a low- or mid-frequency.

For example, suppose two instruments were producing tones as 11,000 Hz and 12,000 Hz, or 5000 Hz and 6000 Hz. If certain kinds of non-linearities exist in the electronic circuitry that's processing such signals, a sum and difference signal will once more be generated. In the case of a pair of high frequencies such as 11 kHz and 12 kHz, the sum signal (at 23,000 Hz) is not likely to bother anyone. But what about the difference frequency? It will be at a very audible 1,000 Hz (12,000-11,000) and boy, you can bet that you and any other listeners will certainly be bothered by that form of distortion!

Generally, to measure for twin distortion, the two higher frequency signals are used. Unlike the tests performed to measure SMPTE-IM, in twin-tone IM measurements, the amplitude of both signals is the same, and the combined amplitude of these signals is established as the 100 percent reference level. The test equipment then introduces a 1000 Hz band pass filter, by means of which the amount of the "beat" frequency is isolated and measured. Again, much lower percentages of twin-tone IM distortion can be heard by most listeners than can the same percentages of the more commonly quoted THD. Incidentally, if you ever do see a specification listed as CCIF-IM distortion, that's the same as the twin-tone IM I've just described.

TRANSIENT INTERMODULATION DIST.

A few years ago, transient intermodulation distortion (TID or, as it was sometimes called, TIM) was all the rage in esoteric and professional audio circles. This kind of distortion, also referred to as Dynamic Intermodulation Distortion, or DIM (are you still with me?) is a direct result of the use of large amounts of negative feedback in amplifiers, preamplifiers, mixers and the like. That's kind of ironic, since negative feedback, in use for many decades, was designed to reduce harmonic distortion in audio electronic equipment. However, too much of it adds this dynamic form of distortion, which arises because it takes a finite amount of time for the out-of-phase feedback signal to come back to the input to do its distortion reducing job.

This is seen particularly in the case of high-frequency signals, while

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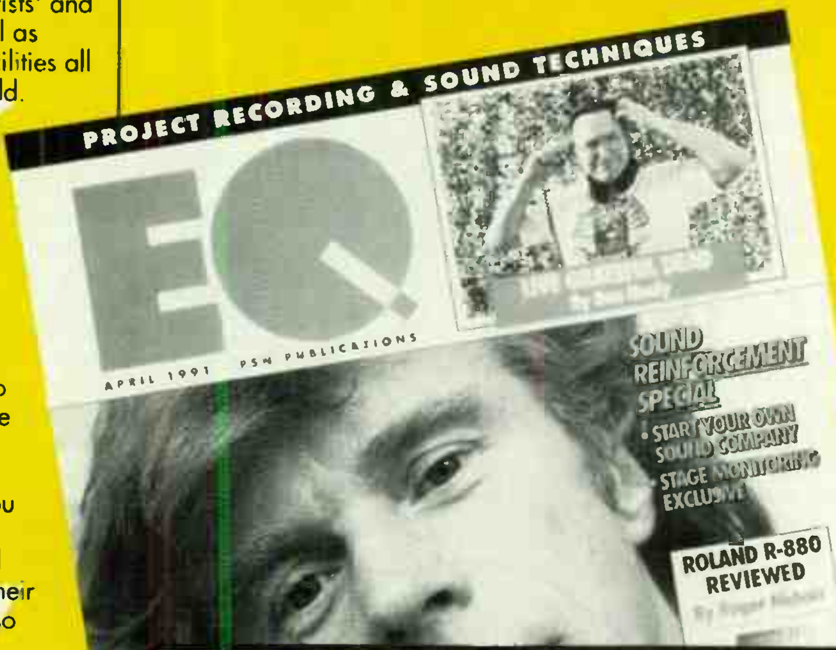

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the feedback signal is making its way back to the input, the output is uncontrolled and just goes off wildly in the direction the input tells it to. Once the correcting feedback signal arrives, the poor amplifier has to wait again to see how accurate the correction signal really is and this goes on and on, seesawing back and forth until the amplifier output settles down.

Why don't we hear more about TID or TIM these days? Simply because as solid state devices have gotten faster and faster, so dynamic settling times have become relatively insignificant compared with the signal transients they have to cope with.

PHASE DISTORTION

The last form of distortion I want to talk about is phase distortion. There are those who maintain that human beings can't even detect phase distortion, while others maintain that it is one of the worst forms of audio distortion going.

Musical signals are made up of complex waveforms in which many frequency components add together to form the sounds we hear. Now, suppose a sound consists of a fundamental tone, say, at 440 Hz and a bunch of lower amplitude harmonics at 880 Hz, 1320 Hz, 1760 Hz and so forth. Those harmonic components have not only specific amplitudes, compared to the fundamental 440 Hz tone, but they also have specific time relationships to that tone and to each other. What's described as phase distortion occurs if those time relationships are upset during the electronic signal processing or amplification and reproduction of the signals. In other words, all the harmonics are there and in their correct relative proportions, but some or all of them are displaced or delayed in time compared to their place in the original sounds being recorded or processed.

As I said earlier, there are those who maintain that this time displacement that we call phase distortion doesn't matter, audibly, while others insist that it does. My own experience suggests that phase distortion can definitely affect stereo imaging and localization, but I'm not sure it matters all that much if we're talking about a single channel or monophonic audio. Since this is a very controversial topic, I'll stop here and wait for the mail to come in, criticizing that last personal opinion. Next time we'll talk about other audio basics. EQ

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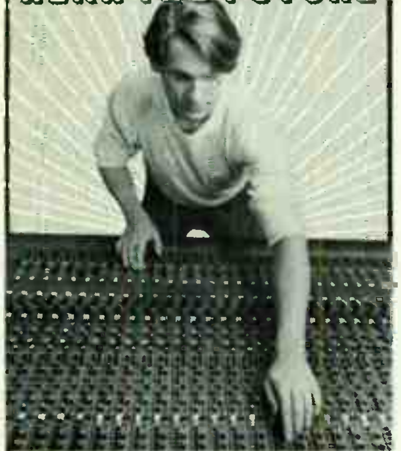

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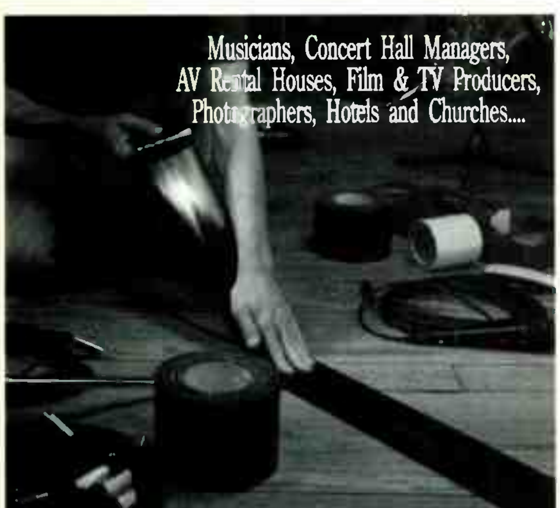

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WISHFUL THINKING

continued from page 114

"Let's print another one, I think the 1000 millisecond re-entrant phase-inverted pre-delay on the vocal answers may be a little much!" The assistant engineer, who I have grown to trust, assumes that I want to go over the previously printed mix. I don't ask him, and he doesn't tell me. We print the new version, listen back and then go on to the next song. After we leave that night, the assistant runs off a safety copy of the mixes.

A week after finishing the album, the artist says that he prefers the original version of the mix (he had a DAT of it) and we agree that it should be the version included in the album. When we gather the tapes for sequencing we realize what happened and there is no previous version. We made safeties — but after the erasure. I call the artist and he says he doesn't have the DAT any more, he was out of blanks and went over it for something else. True Story.

SEE SPOT RUN

Well this will no longer happen to me. I now mix to CD. Yup, *directly* to write-once CD. The breakthrough that allows me to do this is a feature called "SPOT" or incremental recording. Until now, a CD could only be recorded all the way through without stopping and the table of contents had to be recorded before the music was put on the disk. Not any more, PQ subcode breath.

The way it works now is that you can record audio on to the CD in chunks (another technical term) and add to them as often as you like until the CD is full. Once you are done recording, you press a button that assembles all of the start ID information and then goes back and writes the TOC (Table Of Contents) area of the disc. At this point the CD becomes a legal Red Book CD and will play on any CD player. Before the CD is finished off, they can be played in the recorder, but not on a regular CD player.

Versions of the CD recorder are available from Marantz, Gotham Audio, Studer and others. Yamaha has its own two-piece version (YPD R601) that allows you to play the partially recorded CD on any player. They do this by recording a TOC that has 99 Start IDs at 30 second intervals. When you record your mix, it uses multiple 30 second blocks for the incoming audio. When you stop recording, it fills in blank data to the end of the current 30 second block. This method allows you to print a mix to CD and let the producer take it home to listen on his normal CD player. The next day he can bring the CD back to the studio and add the next tune on to the same CD. The only problem with this method is that because the blocks are 30 seconds and the maximum number is 99, the last start ID is at 49:30. This means that the last tune will fill the remainder of the disc wasting the last 13 minutes. Glenn Meadows at Masterfonics and I are trying to talk Yamaha into also providing a mode that uses 60 second blocks. This seems to be more realistic for mixing, although the 30 second blocks will work out fine for radio spots and station IDs.

I think that the life span of the reel-to-reel digital two track is growing shorter. Besides, with all of these un-erasable outtakes in the hands of the record companies, there will be plenty of "never before released" box sets available in the coming years. I know I can't wait.

TTFN (Ta Ta For Now) EQ

MICHAEL KAMEN

continued from page 46

working with an orchestra, I prefer to use the biggest hall available. In a small studio you're forced to listen to an amplified sound effect that compromises the music, which doesn't allow it to sound the way you imagined it. Likewise, electrical interference is kept to a minimum. Miking depends on what we want to hear and it's basically done through trial and error.

Once it's all down on tape, the mixdown is where I turn it over to my mixer, Steve McLaughlin, and music director, Chris Brook. The three of us co-produce the soundtracks. Steve and Chris handle the mixing and make sure that the soundtrack does what it's supposed to. If a scene needs changing, they take care of it and also take my suggestions along the way.

Carefully used, electronic equipment may be a saving grace. But don't let electronics lure you away from your musical goals. The concept of music is inside your heart and not inside your equipment. EQ

HARD DISK

continued from page 74

(I have, however, heard reports that all DATs aren't created equal in this respect.) If you're playing along with the DAT, your part will be in sync as long as you give it the right start time.

The result will be several solos on the hard disk, one right after the other. Put the DAT tape away, and sync the sequencer and Sound Tools up to SMPTE.

You now have two main options:

1. Choose one take as the best take, then use the Trim command to define that take as a region and erase all other takes. Create a playlist that triggers this region at the appropriate time to be in sync with the sequence.

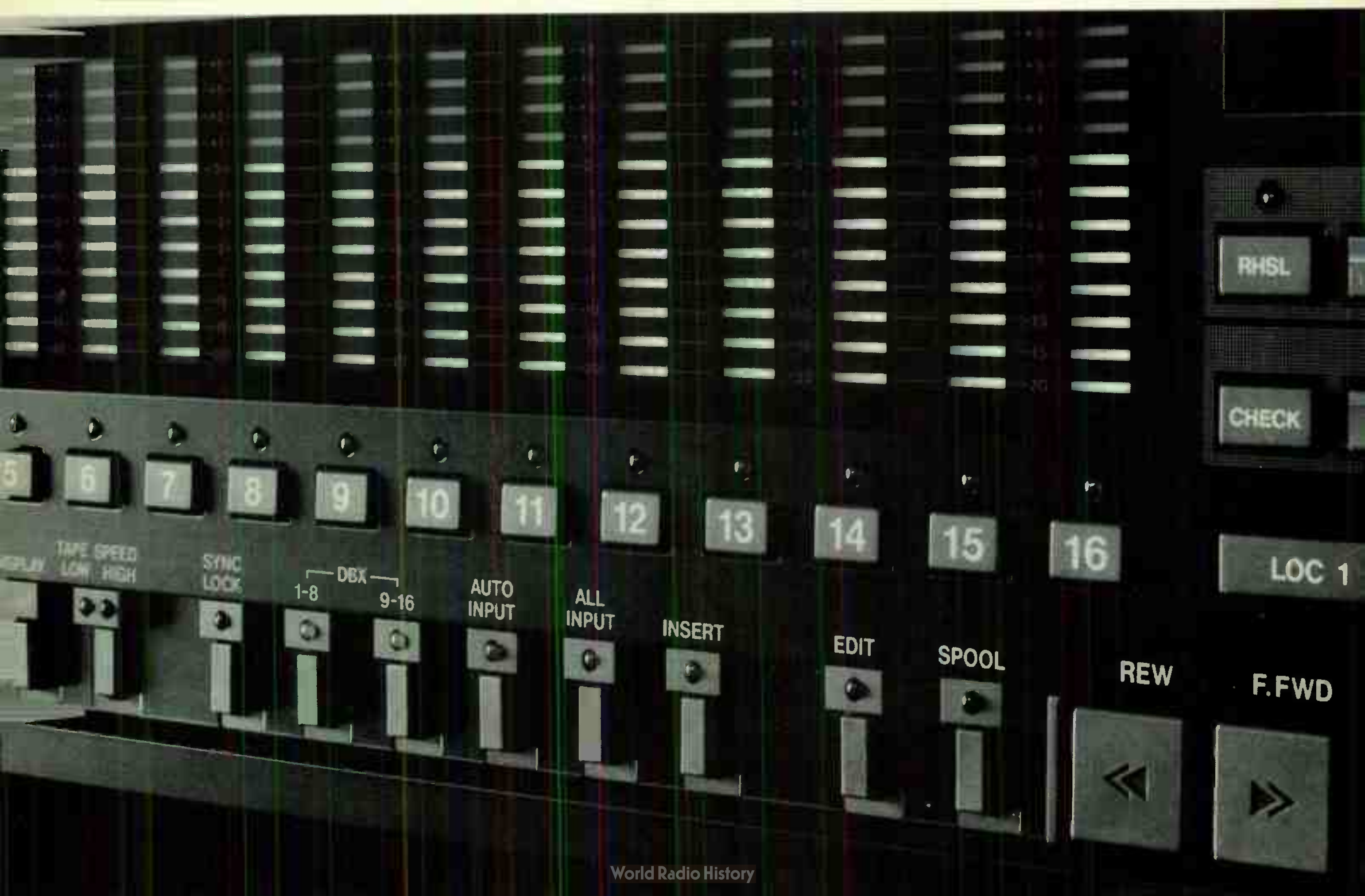
2. Create a playlist that plays different segments from different solos, thus forming a composite take. After creating the composite solo, you can save the playlist as its own sound file. Keep that file, and erase the other versions to free up memory.

VII. TAKING VI ONE STEP FURTHER

The above technique requires a fair amount of hard disk space if you want to do lots of consecutive takes. If you don't have a big hard disk, an option is to monitor the repeated MIDI sequence and play not into Sound Tools but into DAT. This lets you record up to two hours (that should be enough!) of solos, vocals or whatever in your quest for The Perfect Take. Once you've attained it, use the Sound Tools DAT I/O option to digitally transfer the good bits over to Sound Tools for further processing and/or playlist creation.

VIII. HARD DISKS AND HARD CASH

As budget digital multitrack recorders start to appear, hard-disk recording looks more — not less — attractive. You can use a basic hard-disk system to do editing on one or two tracks at a time, but instead of storing the results on expensive hard disks, you can store them on inexpensive digital tape and clear out the hard disk for the next tracks (after backing up, of course). And once you've mixed everything down to DAT, you can shoot the master over to hard disk and do not only your own mastering, but CD prep. Pretty cool. Perhaps best of all, hard-disk recording can be a creative tool as well as a utilitarian one. It's not so hard after all. EQ



IN REVIEW

Drawmer DS301 Noise Gate

EQ LAB REPORT

MANUFACTURER: Drawmer Distribution Ltd., Charlotte St. Business Center, Charlotte St., Wakefield, W. Yorkshire WF1 1UH, United Kingdom. Tel: (011-44) 92 437 8669. U.S. Distributor: QMI, 15 Strathmore Road, Natick, MA 01760. Tel: (508) 650-9444.

APPLICATION: Provides noise gating and downward expansion, but also provides for creative signal processing thanks to sidechains and keying inputs that accept MIDI commands.

SUMMARY: The DS301 costs more than the average noise gate, but also provides more. The MIDI implementation is very useful and the specs are top notch.

STRENGTHS: MIDI control gives more options than standard noise gates. Professional "feel" and operation. Real knobs and switches allow for easy setup.

WEAKNESSES: Comparatively expensive. No provision for 1/4-inch phone jack inputs or outputs. Only recognizes poly and channel pressure data; only transmits channel pressure (no controller 7).

PRICE: $1,499.00 **EQ FREE LIT. #:** 109

A NOISE GATE REVIEW — a real yawner, right? Well, there are a few surprises to the DS301 that add novel creative options to the standard gating functions, and make this unit much more than just a noise-killing workhorse.

FACTS AND SPECS

The DS301 is a two-channel, 1-rack space unit featuring gating (the signal squelches below a particular threshold) and expansion, along with keying inputs and side chain connections for each channel. (With expansion, the signal decreases at a rate set by the expansion ratio; for example, with a 3:1 ratio, every 1 dB drop in input signal causes a 3 dB signal drop at the output.)

Like most pro units, the inputs and outputs are balanced XLR connectors (pin 2 hot). Each channel also has a side chain jack (tip/ring/sleeve), and there are MIDI in and out jacks (the output merges the input signal with any MIDI signals generated by the DS301). The MIDI options are what make the DS301 truly different from the norm, and will be covered later under applications.

The complement of controls includes (all controls, switches, and the 6-step LED bar graph meter are duplicated for each channel): threshold (the level where the gating/expansion action occurs), attack (sets how long it takes for the gate/expander to open once triggered, from 5 μs to 100 ms), hold (keeps the gate open from 20 ms to 12 secs to minimize chattering), release (determines the fade time, from 5 ms to 2 secs, that occurs once hold is over), retrig/ratio (ratio applies to expansion; retrig sets the amount of time, from 5 ms to 5 secs, that must elapse before the unit can retrigger), and range (sets the amount of attenuation, from 0 to -100 dB, in either gate or expansion mode).

Switches include output (normal, bypass, or — thank you! — keying signal monitor), peak (gives a slight boost to percussive transients as they open the gate for "punchier" sounds), expand/gate, auto attack (disables the attack control and automatically adjusts the attack time for the fastest possible attack time consistent with the signal being processed), and expand-gate/duck-limit (brings in the side chain functions if desired).

The keying section offers a choice of internal, external, or MIDI triggering, as well as independent high frequency (200 Hz to 35 kHz) and low frequency (25 Hz to 10 kHz) filters, with a bypass switch that enables or disables the filters. There's also a test button that simulates opening the gate, a trigger LED to verify that a trigger has occurred, and a stereo link switch that allows the DS301 to operate as two independent mono devices or a single stereo device.

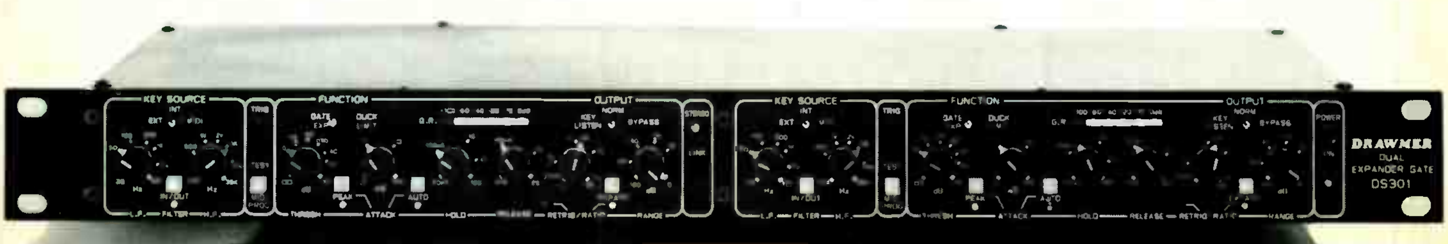
COOL STUFF AND APPLICATIONS

There are some unexpected but interesting features, such as the attack starting from the level set by the range control rather than full off to let through low-level attacks. Another trick: in the expand mode, setting the expand-gate/duck-limit switch to duck turns the DS301 into a pretty decent limiter.

The MIDI options, though, are the slickest part. Each audio channel can be keyed by MIDI notes and/or generate MIDI notes in response to audio signals passing through the unit, and have its own associated MIDI channel.

In gate mode, a MIDI note appearing on the designated channel serves as a trigger. You can also use a sequencer (synced to tape, of course) to do programmable muting of particular sections of a track.

In expander mode, the unit



responds not just to the note but also to its velocity and any subsequent pressure data (poly or channel). This means you can open up the gate to a particular level with a note, then increase the level with pressure commands. This provides expressive options for a variety of instruments. Even better, ratio control values tailor the response to MIDI note velocities. And remember, all these changes can be recorded in a sequencer.

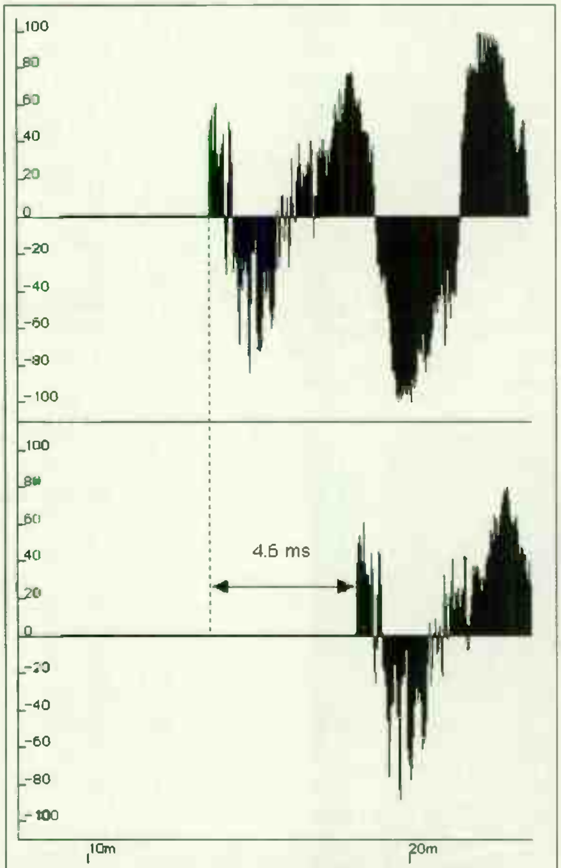
Generating MIDI notes in response to audio triggers is perfect for replacing or augmenting acoustic drum parts with MIDI-triggered drum sounds. Velocity is determined by the attack control setting; fast attacks give high velocities, whereas slow attacks give a lower initial velocity followed by channel pressure data that ramps up to maximum over the duration set by the attack control.

To measure the Drawmer's MIDI processing time, I sent a snare drum into the Drawmer and used the MIDI note to trigger an identical snare sound. Then both were recorded into Sound Tools on opposite channels. Fig. 1 shows the results — a cumulative delay of 4.6 ms. However, the drum machine itself has a 1.7 ms processing delay, and a MIDI note-on command requires 1 ms. The MIDI interface also adds processing time. As a result, the total processing time ends up as less than 1.9 ms — more than fast enough for critical drum replacement applications, and faster than any synthesizer I've tested.

In expander mode, the note velocity can be varied according to how the expander opens (as determined by the ratio and threshold settings). The side-chain controls also affect the MIDI data. Of course, if you have a sequencer synced to MIDI, the DS301 MIDI trigger output can be recorded in multiple passes for future editing or modification instead of being used exclusively in real time.

OPINIONS

$1,499 is a lot to pay for a dual noise gate, but this is no ordinary device. The specs are very good (±1 dB response from 20 Hz-22 kHz, with RMS noise in that range spec'ed at -96 dB), the controls are easy to learn and operate, and the MIDI options turn this into a creative signal processing tool. If all you need are basic noise gate/expansion functions, the DS301 is probably overkill. But if you want a noise gate that lets you get creative, you may want to commune with your bank account and see if you can swing the extra bucks. *—Craig Anderton*



MIDI output processing delay. Although the total delay shown is 4.6 ms, when you subtract the processing time of the sound being triggered and the MIDI command itself, the MIDI delay clocks in at under 1.9 ms — very speedy.

IN REVIEW

Turtle Beach 56K Digital System

MANUFACTURER: P.O. Box 5074, York, PA 17405, 717 - 843-6916.

APPLICATION: Inexpensive two-track digital editing system that allows DAT to be freely edited on a PC or compatible.

SUMMARY: This system is simple to operate and easy to set up and get running. Basically operates by zoning the selected cuts on the master.

STRENGTHS: Now even the smaller project facility can have digital audio editing capabilities at a very affordable cost. And the system is feature rich.

WEAKNESSES: Limited to two-tracks due to the fact it is only compatible with with DAT.

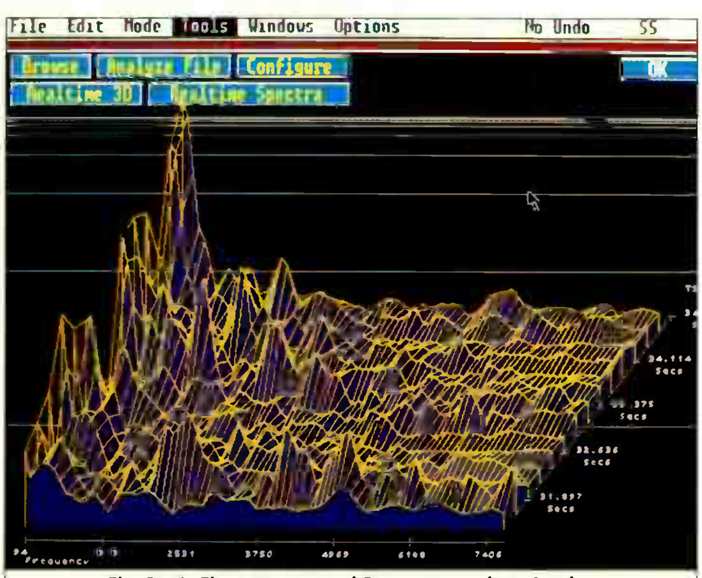
PRICE: $1995. **EQ FREE LIT. #:** 111

Edit DAT recordings on a PC? Thanks to a new system from Turtle Beach, the time is here. Check out their 56K editor for IBM compatibles. At $1995 list, it's one of the least-expensive 2-track digital editors. And it's fast, fun and effective.

The 56K system includes: a DSP circuit card that plugs into a slot in your computer, an interface box with connectors for SMPTE, MIDI and digital audio, pro (AES/EBU) and consumer (S/PDIF), and SoundStage software for digital audio editing.

To run the Turtle Beach 56K you'll need: an IBM-compatible 286, 386, or 486 computer running at 12 mHz or faster with DOS 4.01 or later, a Microsoft-compatible mouse, a hard disk with 28 ms or less random access time, 500 Kb/sec or faster data transfer rate, at least 11 megabytes of disk space per minute of stereo audio and a DAT recorder with digital ins and outs or an A/D converter.

According to Turtle Beach, the 56K system works with most DATs. [Turtle Beach approves the following DATs for use with the 56K: Aiwa HD-X1; Casio DA-7; Denon DTR-2000; Fostex D-20; JVC DSDT900; Nakamichi 1000; Panasonic SV-3500/3700—/3900/255; Sony D10 Pro/DTC-700/PCM-2500/PCM-2000/TCD-D3/DATman, plus others; Tascam DA-30 w/software update; Technics SV-DA10; Yamaha DA-202.] 56K does not support AES/EBU on the Panasonic SV-3700, but does support the S/PDIF.



The 56K's Three Dimensional Frequency Analysis Display

An important thing to know about the 56K is that you can't change the sample rate. The 56K system uses the same sample rate that your DAT machine sends out (32, 44.1, or 48K). Apparently, a sample-rate converter would cost a lot.

The editing program employs a GEM Graphical User Interface, using mouse-driven menus and icons. I found the screens to be uncluttered and colorful — very pleasant to work with. You'll need to read the manual because there's very little on-line help. In fairness, on-line help wastes memory if you don't need it. Some controls are not labeled. Icons are not labeled, but they are well thought-out. Operation becomes obvious after some practice.

A real delight is the 56K instruction manual, which is easy to follow, friendly, and even funny. However, I'd like to see a "Quick Start" sheet for those impatient to get started. A helpful video shows how to install the hardware (it's easy) and how to run the software. I installed and tested the complete system in just 20 minutes with no glitches or special adjustments.

HOW IT WORKS

Let's run through some typical procedures. First, you open up to four soundfile windows for your recordings. Choose one and name it. Then you'll record an audio program — a sample, a song, or several songs — onto your hard disk.

To do this, click the record-button icon and hit Play on your DAT machine. The DAT recording will copy digitally to hard disk.

A stereo soundtrack of your recording appears on screen. Its time axis can be measured in real time, SMPTE time, number of samples, film feet/frames, or beats. If you wish, play the recording using the spacebar or loudspeaker icon. To hear the playback, set your DAT to monitor the digital input.

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Now you're ready to edit the recording. Basically, you define zones (segments of the recording) such as a song, chorus, phrase, or word, assemble these zones into a playlist (a sequence of zones) and play the playlist.

The first step in editing is to look at the overview bar, which represents the entire recording. Find the location of the part you want to work on, and highlight it with your mouse. The SMPTE time of that part shows in a window, and that part's soundtrack appears. To see if it's the part you wanted, play it with the spacebar or loudspeaker icon.

Next you'll select a smaller area within that soundtrack — say, a verse, chorus, phrase or word. To do this, play the program. A cursor line moves left to right across the soundtrack. Watch where the cursor line crosses the area you want to select and highlight this area.

Play the area using the spacebar or loudspeaker icon. Trim the exact start and stop points. To define your selected area as a zone, give it a name.You can zoom in on the selected area to edit it in fine detail — all the way down to individual samples (1/48,000 sec.!).

How do you define an entire song as a zone? First, as you record a song, note its start and stop times from the on-screen counter. Then you go through several windows to get to the zone-edit screen. This process is needlessly complicated — I'd like to be able to define a zone quickly by typing in the zone's name, start time and end time.

In the zone-edit screen, type in the SMPTE time where the song (zone) starts. Now you can trim the zone precisely. With your mouse, move the cursor line to the exact beginning of the song. If necessary, you can scrub the soundtrack — play it slowly forward and backward until you find the exact spot to trim to. After defining the beginning of the zone, go to the end and type in its approximate ending time. Trim the end of the song, and click on OK. The zone is trimmed and defined.

You can cut zones (temporarily remove them), paste them (insert them elsewhere in a soundfile), copy them, or paste fill them over another zone to replace it. These are destructive edits which change the recording on disk (you can undo them, however) and are relatively time-consuming. Playlist edits, discussed next, are non-destructive and much faster.

CREATING A PLAYLIST

After editing and defining several zones, choose Playlist. The names of the zones you defined appear in a window. Now you can create a playlist. Simply click on the names of zones you want to hear and drag them into the playlist window. You can repeat zones, loop them, re-order them, insert silence between them and crossfade between them — all with the click of a mouse. Transitions between zones are seamless.

Want to remove Verse 2 in a song? Define two zones — one before Verse 2 and one after. Put these two zones in a playlist, one after the other, and play them.

Once your playlist is done, play it and record it back onto DAT. There's your edited recording. It might be an edited song, a radio spot, a film soundtrack, or a record album.

The playlist can include zones from various soundfiles. When you load a playlist, the program removes any playlist zones from soundfiles that aren't open. Be sure to open all the necessary soundfiles (one soundfile per window) before loading a playlist. Each time you load the same playlist, crossfades must regenerate, and this takes a long time.

If you have a playlist set up and you load a new recording into the same window you're working in, every other zone in the playlist will disappear. Be careful!

Another thing to bear in mind is that the playlist highlight blinks slowly. So when the zones play rapidly, you can't always tell what zone you're in.

Playlist events can be triggered by SMPTE time, a MIDI note, mouse or spacebar. You can generate MIDI note events at selected times.

You can't save soundfile information files (zone definitions and playlists) to DAT. But you can save them to hard disk or to floppies. Future versions of SoundStage will let you save these files to DAT along with the audio data.

Time compression and SMPTE chase are not yet available, but will be eventually.

OTHER GOODIES

The Turtle Beach 56K comes packed with a host of other features and abilities that make using the unit easy (and even fun). They include: a pencil cursor that lets you redraw a waveform to remove clicks, realtime analyzers (level vs. time, spectrum vs. time), four bands of EQ with adjustable frequency and bandwidth (plus an "Amaze-O-Graph ™" showing the frequency response of the equalizer), playback speed control (this affects both speed and pitch), polarity inversion and zone reverse, SMPTE and MTC generator/reader/converter, automation of a multimedia production or a film soundtrack by triggering sound effects and MIDI music at selected SMPTE times and the mixing of up to three soundfiles or zones to a fourth soundfile. And these are just a sampling of the 56K's many features.

In summary, the 56K is quick to install, easy to use, and gets the job done. It includes all the features you're likely to need to record and edit two tracks of digital audio — all at a reasonable price. It is highly recommended, just be sure to check whether the 56K will be compatible with your DAT.

Turtle Beach Systems also sells complete workstations, including a computer, for $5000 and up.

— Bruce Bartlett and Jenny Bartlett

Ad Index

92	AB International	80	916-783-7800
12	Acoustic Science Corp	01	503-343-9727
41	Acoustical Solutions	02	800-782-5742
2	AKG	03	415-351-3500
10,16-17,95	Alesis	05,06, 07	415-367-3889
29	Amek	08	818-508-9788
3	Ampex	04	415-367-3889
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9	Ashly Audio	11	716-544-5191
96	Audio Inst. Of America	12	415-931-4160
63	Audio World	15	305-379-6013
86	Bananas At Large	16	415-457-7600
68	Beyerdynamic	69	516-935-8000
47	Bruel & Kjaer	73	508-481-7000
51	C-Lab	17	215-647-9390
112	Center for the Media Arts	74	212-807-6670
86	Columbia Magnetics	18	212-445-4791
57	Crest Audio	75	201-423-1300
69	Crown	19	219-294-8000
18	DAT Audio Gallery	20	213-829-3429
21	dbx	21	415-351-3500
115	DK Digital	14	201-224-9344
35	Digital Audio Labs	22	612-559-6104
91	Disc Makers	23	215-232-4140
[illegible]	Dolby	24	415-558-0200
88	Drawmer	[illegible]	508-650-9444
28	Dynatek Automation	26	416-636-3000
112	Electron Artistries	6	513-746-4283
37	Ensoniq	28	215-647-3930
30	Fishman Transducers	27	617-938-8850
5,87	Fostex	29,45	213-921-1112
74	Furman Sound Inc.	30	415-927-1225
45	Gemini Sound Inc.	31	718-851-6000
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18	Illbruck	55	800-662-0032
38	Industry Sourcebook	66	310-841-2700
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90	KABA	37	800-231-8273
96	Klarity Kassette	38	207-873-3911
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78	KRK	39	714-841-1600
54	Lexicon	40	617-736-0300
33	Mackie Designs	41	206-487-4333
12	Microtechnology Unlimited	42	919-870-0344
36	Oktal International	43	514-738-3000
98	Pacific Coast Tech.	44	619-693-0209
20	Passport	46	415-726-0280
13	Peavey	47	601-483-5365
73	Polyline	48	708-390-7758
56	Poplesss Voicem Screens	49	908-527-0767
12	QSC	33	714-645-2540
98	Rad Rock	71	805-379-4252
59	Rocktron Corporation	50	313-853-3055
6,7	Roland	51	213-685-5141
46	Sennheiser	52	203-434-9190
39	Shure Brothers	77	708-866-2527
52,53	Solid State Logic	53	212-315-1111
8	Sony	—	201-930-1000
27	Soundtracs	54	516-932-3810
77	Symetrix	55	206-282-2555
82,83,106,107	Tascam	56,57	213-726-0303
79	Taxi	70	818-888-2111
30	Tech 21	58	212-315-1116
34	The DAT Store	59	310-828-6487
84	The John Hardy Company	60	708-864-8060
94	The Recording Workshop	61	614-663-2510
60	Thought Processors	78	718-857-2860
116	UREI	62	818-893-8411
75	Vestax Corporation	68	707-427-1920
63	Walters-Storyk	67	914-255-2255
73	Whisper Room	63	410-997-1440
31	Yamaha Pro Audio Division	64	714-522-9011
58	Yorkville Sound Inc.	65	716-297-2920

Wishful Thinking?

Photo by Ed Colver

What you want is not necessarily what you get

BY ROGER NICHOLS

I always seem to wait until the last minute to write my column. I like to blame it on the fact that all of the neat equipment comes out just before my deadline. Because of the lead time at *EQ* and the fact that the magazine comes out every other month, a piece of equipment could be introduced and become obsolete before you even read about it. On the other hand, I still stop at my local Radio Shack once a week to find out if the $400 Tandy recordable CD they promised five years ago is out yet.

ADAT ADULATION

I haven't received my Alesis ADAT digital eight-track yet, but already the field is getting thick with announcements of alternatives. Tascam is coming out with an 8mm videotape-based eight track. Fostex also has one in the works, but is waiting to decide whether it should use VHS or 8mm tape, and (ugly) rumor has it that there will be many Japanese versions of this format. I'm not sure about compatibility, but as you know from the DAT mess, even compatible units aren't completely compatible. Another VHS vs. Betamax war to be sure.

There are two features that I love about the Alesis — I hope all of the other versions include them too. One is the fact that you can transfer all eight tracks at once through a fiber optic cable, the other is that you can slip any track in single sample increments up to about 80 milliseconds.

WISH LIST

The multitrack transfer capability would become very significant if all of the digital eight track manufacturers get together and agree (yuk yuk yuk, sorry) on a data exchange protocol similar to, if not actually, the multitrack AES/EBU format. Having a common interface would add credibility to multitrack hard disk recorders and editors. Being able to transfer eight tracks at a time between your archival storage medium and your hard disk recorder would make the up-load and down-load time manageable. If you own one brand of machine and your buddy owns another, you could still pool your resources to back-up your recordings. I can't believe how much time I have spent backing up Akai ADAM tapes onto DAT two tracks at a time.

Track slipping capabilities are a necessity. I would kill (or at least severely wound) to be able to individually shift tracks to make up for the way a musician plays or a sequencer slops. Imagine being able to make up for all the MIDI delays you've had to put up with all these years. All the times that you wanted to shift a part a little, but the increments in the sequencer were too coarse. Now you can do it in jumps of around 1/50th of a millisecond.

GETTING YOUR FEET WET

The ability to shift tracks can easily be designed into any digital recorder, so demand it. All rotary-head digital machines and hard disk-based recorders must record and play back the digital data in spurts. This is because, in the hard disk-based machines, the data is stored in blocks of data on sectors of the hard disk that may not necessarily be in sequential order. In the rotary-head machines the data is stored on the videotape (or DAT tape) in little packets of data. In both cases, this data is read from the medium and stored in buffer memory.

The buffer works exactly like a bucket of water with a hole in the bottom. We can put water in the bucket a glass-full at a time. As long as we don't let the water level get too low, we will always have a continuous stream of water through the hole. If we don't put water in often enough, we will run out of water in the bucket and the output stream will mute. If we put in too much water, the bucket will overflow and some of our data will be lost over the top. As long as we keep the bucket water level somewhere in the middle, everything will be okay. There is a set amount of time between the time we pour a glass of water in the bucket until that exact glob (technical term) of water comes out the hole on the bottom. If we want the data sooner, we can just drill a new hole further up the side of the bucket. This works the same way as moving a pointer in the memory buffer of the digital machine. This water analogy is exactly the reason why the operating manuals that come with your DAT machines tell you not to use them in the shower or bathtub.

NEW FUN TOYS

I haven't mixed a project (including home projects) to analog tape since June of 1980. In the studio it was 3M four track, then Sony 1610, Mitsubishi X-80, Sony 1630, Sony 3324, Mitsubishi X-86HS, DAT and MO (Magneto Optical) disk. At home it was Sony F-1, Mitsubishi X-80 (okay, I borrowed it a couple of times), DAT and MO disk.

The biggest similarity among these formats is that sooner or later, on every one of them, I have erased something that later I wish I had kept. It wasn't so bad when I consciously made the decision to go over a piece of tape. But every once in a while it would happen accidentally. There is no excuse for that.

More than once in the studio I have printed a mix to the reigning mix medium, listened back to it, then decided that I could "do one better."

continued on page 106

BACKED BY POPULAR DEMAND.

After more than twenty years, UREI Compressor/Limiters remain the of choice of audio professionals, having earned a reputation for excellence as hard working tools of the trade. It is difficult to find a recording or broadcast studio that does not own at least one UREI Compressor/Limiter. And engineers in the Sound Reinforcement and Installed Sound industries have long considered our products as vital links of any high quality audio equipment chain. With experience as our teacher, and modern technology as our guide, we are proud to offer three LA Series Compressor/Limiters, beginning with the LA-22.

The LA-22, a dual channel unit, contains three Gain Reduction circuits, can be used as a Dynamic Expander, and is equipped with a Full Parametric Filter on each channel. Its unmatched versatility sets it apart as a truly unique multi-function tool. Designed with innovative "spectral agility," the user has the option to reduce or expand gain across the total audio bandwidth or at a chosen center frequency with variable "Q" of 1/6 octave to 2-1/2 octaves. With proper settings in the expansion mode, you can use the LA-22 to "lift" vocals in a live or studio mix or increase intelligibility in paging systems or radio broadcasts. Conversely, in the gain reduction mode, the compression can be frequency focused to control levels to prevent feedback, for De-essing, De-popping or to creatively "fatten" the sonic character of particular instruments and vocals. The parametric filter circuit, completely accessible via the rear panel barrier strip, can be accessed and routed to the Side Chain, thus making the LA-22 a frequency dependent gain reduction or expander system.

Full Parametric EQ Section.

As a pure Compressor/Limiter, the LA-22, along with the LA-10 single channel and LA-12 dual channel models, offers unparalleled performance and seamless transition, employing proprietary Smart-Slope™ compression ratios. All three models feature transformer isolated output stages, optimized by a patented active circuit to assure the elimination of distortion and saturation frequently associated with transformers. Active Balanced Bridging Inputs easily handle amplitudes in excess of +24 dB and both input and output connections can be achieved by your choice of XLR-type, 1/4 inch phone or Barrier Strip connectors. Exceptional signal integrity, low distortion and superb dynamic range is common to all models.

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