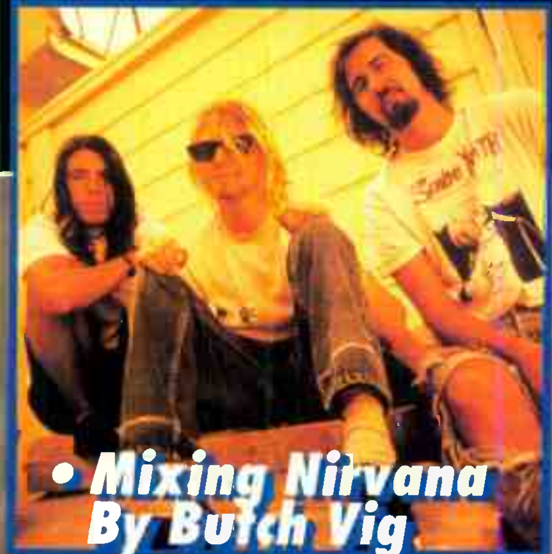
PROJECT RECORDING
& SOUND TECHNIQUES

EQ



• *Mixing Nirvana By Butch Vig*

• *Peter Gabriel's Studio Hideaway*

• *Anderton's NAMM Hot Picks*

LIBERATING NASHVILLE
by Jimmy Bowen

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APRIL 1992
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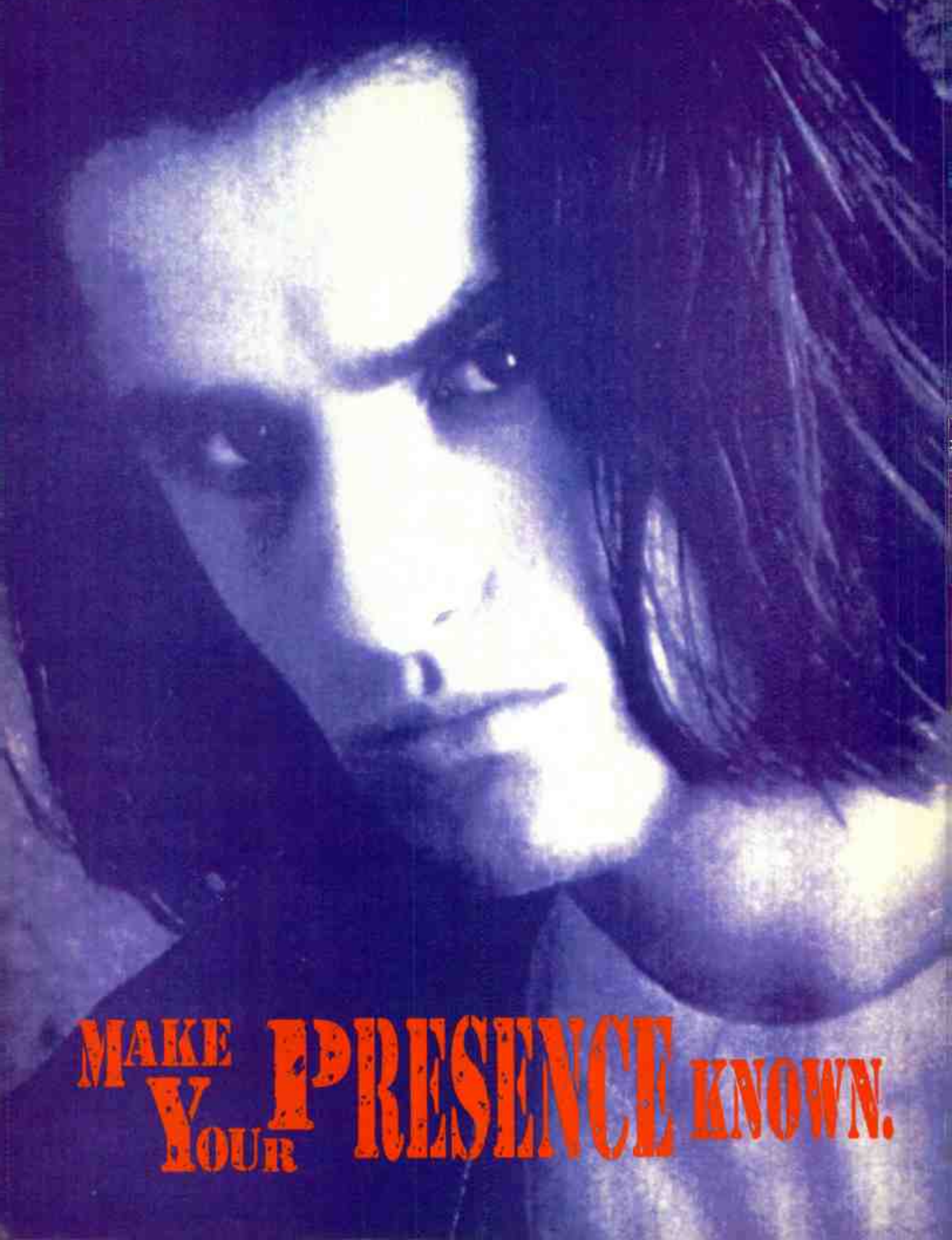
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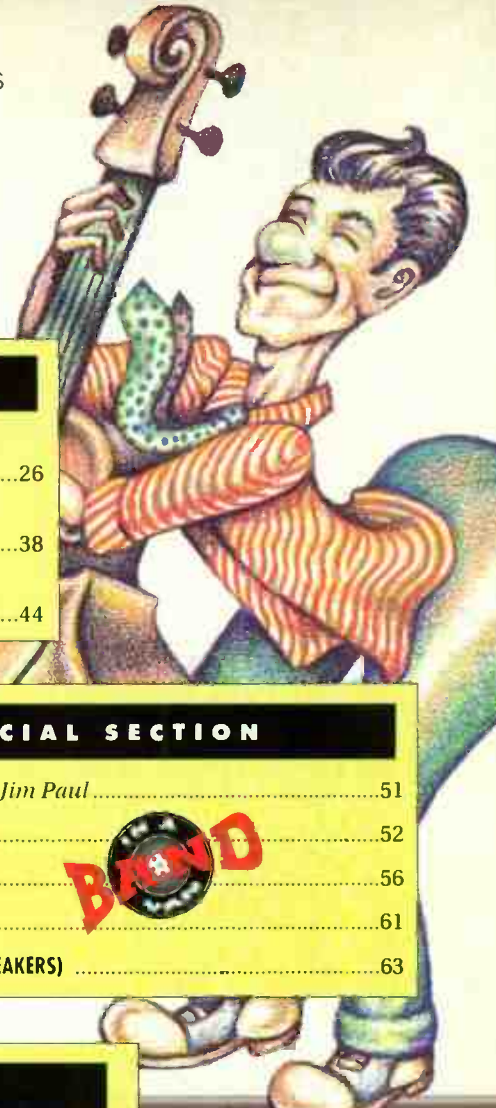

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BAND IN A VAN / SPECIAL SECTION

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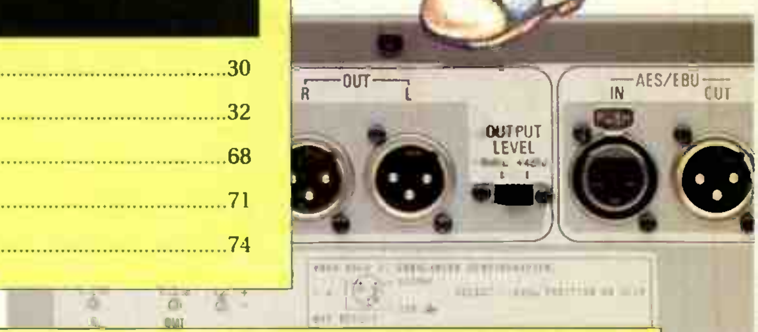

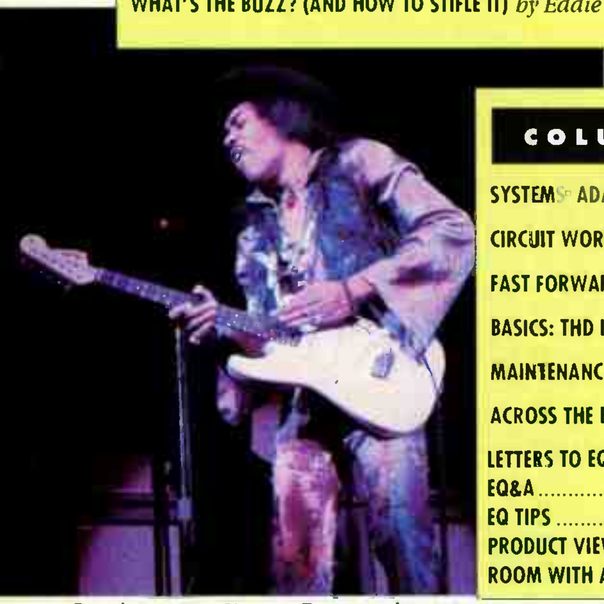

COLUMNS/DEPARTMENTS

On the cover: Jimmy Bowen photo by Beth Gwinn; Band In a Van logo by Mark Alhadeff

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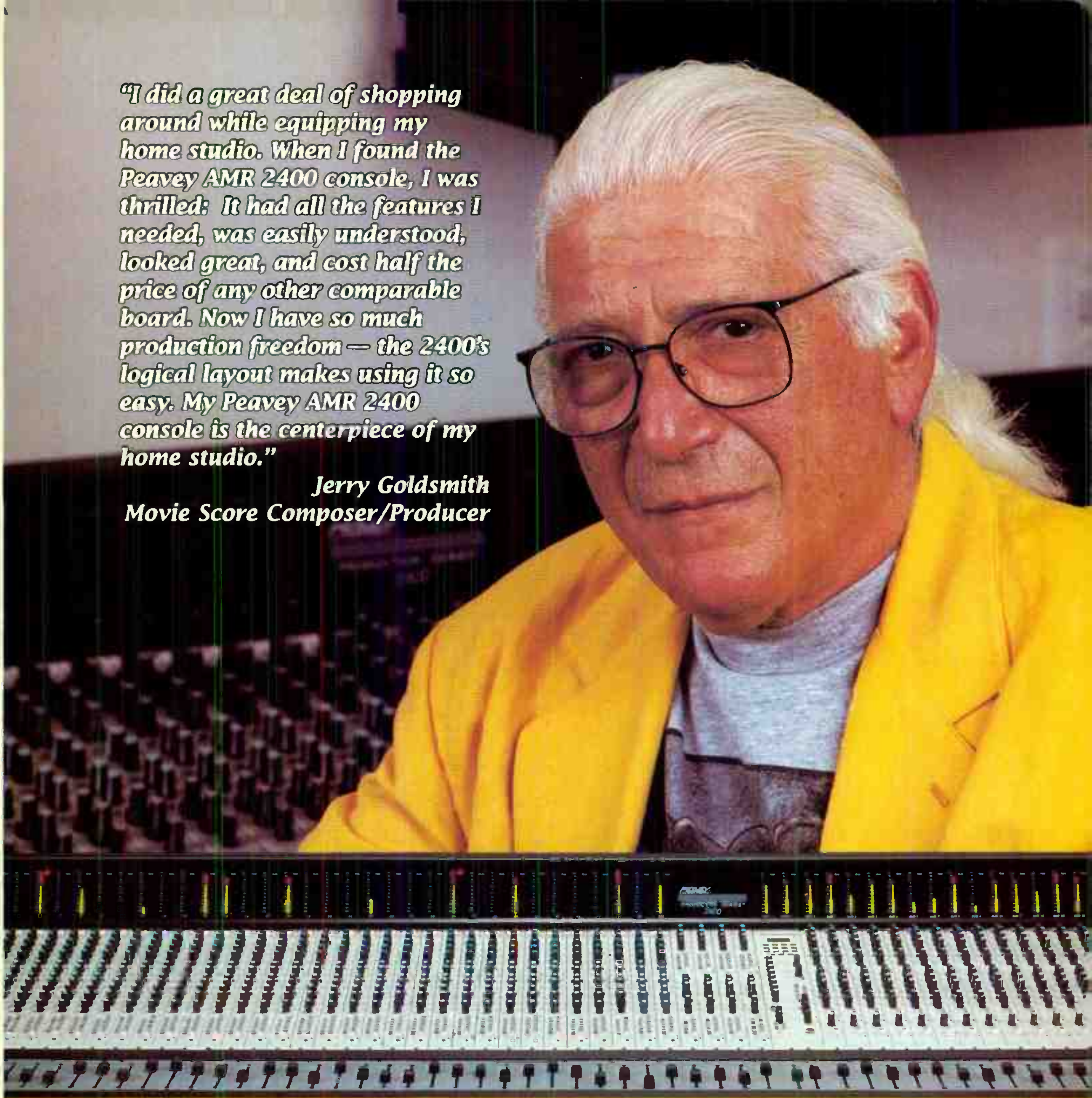

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Editorial Offices
939 Port Washington Blvd.
Port Washington, NY 11050
Tel: (516) 944-5940 Fax: (516) 767-1745

Administrative/Sales Offices
2 Park Avenue, Suite 1820
New York, NY 10016
Tel: (212) 213-3444 Fax: (212) 213-3484

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LETTERS TO EQ

PARTICLE THEORY

Imagine, if you will (Twilight Zone music in the background), a world. A world in which magnetic particles, driven by mysterious forces, migrate randomly across your hard disk. A world in which an opposing party of more patriotic tape-based particles squat in their established homes.

Strange? Impossible, you say?

Nevertheless, this fantastic scenario is what Tom Jung described in his article on the Yamaha DMR8 in your December issue.

All seriousness aside, you do your readers a disservice by publishing false information like "Hard disks . . . don't always give back the exact information you put in. With tape you always get back the exact information you put in. With tape you're always reading off the exact magnetic particles you recorded onto."

NO magnetic media always gives back the exact information recorded. It's a technical falsehood that hard disks are somehow inherently unreliable because they don't give back the same information that was recorded. For example, when loading computer programs, a single missed bit can crash the computer system. Elaborate disk-reading error-detection schemes exist on the most common computers. There's even greater error correction in disk-based audio recording.

Anyone who has experienced tape drop-out knows that you don't "always get back the exact information you put in." Both digital tape and disk record discreet numbers which are played back. Both have predictable error rates. I'm sure that one media is more reliable than the other, but not for the reason Mr. Jung wrote of.

And, in any case, you read from the "exact magnetic particles you recorded onto" in both media. Shame on you for not catching this one before publication!

Sincerely (but playfully)
Richard Berman
Toothpaste Music
Los Angeles

PHIL IN THE BLANKS

I found the Phil Spector article [Jan./Feb.'92 EQ] interesting. However, Sonny Bono's comments are quite inaccurate. Phil Spector did not "invent the technique called bouncing." That honor must be given to a man by the name of Les Paul [July/Aug. '91 EQ]. I thought everyone knew that. The year was 1948, but the recording industry did not begin to utilize the technology until the early Sixties, with the splendid exception of rock genius Buddy Holly, who used overdubbing on many of his classic recordings in the mid-to-late Fifties. Phil Spector was a great producer, but let's give credit where it's due. Maybe Sonny Bono should study the facts before inserting his foot in his mouth, even if he is a politician now.

Musically
Jerry Vanek
Rathdrum, ID

MANIC COMPRESSION

Your fact sheet on compressors in the January issue of *EQ* presents some incorrect information about dbx products. The dbx 903, 160XT, 166, and the 163X (which was not included in the chart) compressor/limiters offer variable attack and release times in the sense that the units will respond to the energy of the input signal and automatically adjust attack and release times accordingly. This is referred to as "Program Dependent" attack and release times. Among the advantages are shortened set-up time, the elimination of less than optimal sound which commonly occurs as a result of the misadjustment of these parameters, and a more musical interaction with the changing dynamics of the program material than is possible with static attack and release settings. Manual adjustment of attack and release is not offered on these products. The dbx 165A does offer variable attack and release times, but not an enhancer as your chart suggests. (PeakStop may enhance the longevity of an HF driver or a mix engineer's ears but it's not messin' with the sound.)

WRITE TO US

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

I know you want your readership to have only the most accurate and timely information and appreciate the opportunity to set the record straight.

Jawxillion Loeb
Product Manager
dbx Professional Products

TAKE DIS ABOUT DAT

I was shocked and disappointed to read Roger Nichols' comments in his column "More Dis about DAT!" published in *EQ*'s January issue. While I appreciate Roger's frustration at not being able to do everything he wants with the various digital audio devices, his examination of digital interface compatibility between the SV-3700 Pro-DAT and certain digital audio workstations, and his conclusions, were inaccurate in almost every respect.

Professional audio is in transition between analog and digital worlds. During such times, it is unfortunate, but to be expected, that there will occasionally be confusion. One can find hardware within today's studio environment that is equipped with one of maybe a dozen different interfaces. For example, we'll find equipment and systems that offer SDIF-2, Mitsubishi A/B/C Dub, Yamaha Cascade and other proprietary formats, along with the more familiar AES/EBU, S/P DIF, EIAJ CP-340J and contemporary IEC-958 ports.

Some of these interfaces, on first examination, appear to be compatible because they sometimes work together. Under certain circumstances, however, they will not behave as we might wish, with frustrating results. As Roger Nichols and many professional users have discovered, frustration will almost certainly result from a partial or incomplete understanding of the uses for which these interfaces were designed, and the physical incompatibilities they exhibit.

While I do not wish to dwell too long on the subject, I feel it is important that the facts are clearly stated on a number of vital operational points.

The SV-3700 and SV-3900 Pro-DATs are equipped with AES/EBU digital I/O ports that accurately follow the AES3-1985 Recommended Practice, a document that specifies, amongst other things, professional grade 3-pin XL connectors. Again, following the AES3 Recommended Practice, the input examines byte 0/bit 0 of the Channel Status information to confirm that the data is indeed of an AES3 professional format. If the digital data being input to the AES3 ports is being output by a device that fails to see the correct Channel Status flags, including the consumer/professional ID bit, then the DAT machine will not enter record mode. This is intentional and important to the recording of material that is to be used and interchanged within the professional community.

The sample-rate flag carried within the Channel Status is designed to relay to receiving equipment the sample rate that was used to digitize the original material. Both the professional AES3 and IEC 958 Type II I/Os contain sample-rate flags: the former uses bits 6 and 7, while the latter (dependent upon the type of consumer device) reserves bits 24 thru 27.

AES and IEC interfaces exhibit some degree of compatibility. Given their common ancestry, it is not surprising that both AES3 and IEC958 utilize a common 32-bit structure with very similar bit and sync group patterns. It is unfortunate that, in many situations, a consumer-grade input will connect to an AES3 output and data will sometimes pass across the interface. It is quite conceivable that many of today's problems would not exist if, from the start, these interfaces had not been compatible. If this had been the case, it is more than likely many manufacturers would have been obliged to develop correct implementations of the professional interface, instead of simply using low-cost consumer interface chips with XL connectors. In reality, we must consider AES3 and IEC 958 "consumer-use" interfaces totally different, any electrical compatibility being purely incidental.

Again, confusion between AES3 and IEC 958 I/Os exists when we raise the thorny subject of SCMS copy protection. AES3 interfaces are not intended to recognize nor carry SCMS information. IEC 958 type II interfaces, however, must honor the presence of SCMS data even if, in the case of professional machines such as the SV-3700/3900 (and others), they do not respond to a data stream that is coded to an SCMS level of "10." (A situation that would prevent a suitably equipped consumer-grade machine from entering record-ready mode.) The AES3 interface is not designed to carry information specific to any type of hardware. For example, it will not recognize the existence of DAT specific "PNOs" where consumer interfaces often do.

It would be extremely valuable if this magazine presented its readership with an overview of the technical differences between equipment and made greater efforts to avoid printing inaccurate and inflammatory articles. *EQ* has impressed me in the past with the high ground position it has taken on editorial issues. I hope it will maintain this charter in the future.

To that end I would like to propose two actions:

1. That a responsible trade magazine (*EQ* perhaps?) measures the digital I/Os of many popular DAT machines, CD players, workstations, and other digital system components, and publishes the results for all to see. This critical information, accessible to professional users, would help avoid costly and frustrating problems.

2. That our industry forms an Association of Digital Audio Manufacturers, to act as a forum for the exchange of information, to develop interfaces/standards, and to inform users of the incompatibilities that might exist between different units.

Our industry has access to some fine standards, documents and recommended practices that, with some care and attention from the manufacturers, can result in compatible digital interfaces for all of us.

Steve Woolley
Nat'l. Sales and Marketing Manager
Panasonic Communications and Systems Company
Cypress, CA

EQ&A

SILLY RUMOR MILL

Q *In relation to DAT machines, is there really a noticeable difference between optical transfers made in the light and in the dark? Or is this a silly rumor? Also, why all the talk about the big difference between optical and coaxial transfers? Isn't it all digital?*

Hans Henrik Nissen
Copenhagen, Denmark

A There should be no difference, noticeable or otherwise, between optical digital transfers made in the light versus the dark. Even if there was a "leak" in the black covering of the optical cable, allowing ambient light to enter, the refractive properties of the optical transmission cable would keep that ambient light from entering or influencing the transmission path itself.

Is there a difference between optical and electronic digital transfers? Not if both transmission paths are optimized. But that's a big "if."

Although I cannot know for sure, I will guess that the optical digital transmission system you're using was developed for consumer product usage. Unfortunately, it's very difficult to make good optical connections using the plastic optical cable common with these consumer systems. The result of a bad optical connection is smeared light transfer and errors in the digital audio transmission.

Those professional studios using optical transfer systems are using true glass optical fiber cable with professional optical connection systems. Professional glass fiber connections, while expensive, can be reliable and virtually error-free. The same cannot be said for plastic optical connections.

Most digital audio connections are made using electronic transmission systems, either coaxial cable and RCA connectors (IEC/SPDIF standards) or balanced, three-wire cable and XLR connectors (AES/EBU standard). Either of these systems can be reliable and virtually error free provided good quality cable and connectors are used.

Always use video-grade coaxial cable and high-quality video-grade RCA connectors for IEC/SPDIF connections. Always use high-quality, low-capacitance three-wire cable (two wires plus shield) and high-quality XLR connectors for AES/EBU connections. A good test of the cable and connections is that they'll pass clear video with no ringing. Some vendors are now offering special cables for these digital audio transfers. Panasonic, for example, ships a digital-quality coaxial cable with all of its professional DATs.

Because most digital audio equipment utilizes one or both of the electronic digital audio transmission standards (IEC/SPDIF or AES/EBU) we recommend that you use these systems for your digital transfers. Remember, however, that none of the current digital audio transmission systems standards, optical or electronic, include the kind of robust error correction that protects DAT tape recordings. Errors in DAT tape playback (caused by tape dropouts, for example) are usually caught and corrected. Errors in digital audio transmission are, in general, not corrected. For this reason, the quality of cable and connections you use are critical to error-free transmission.

Chris Foreman
Marketing Manager
Pro Audio Dept.
Panasonic/RAMSA

PCM PLEASE

Q *I have read numerous articles on "how to make great tapes." These articles often referred to the use of a VCR and a specific PCM converter that I have been trying to locate for some time now without luck. Where can I find a Sony PCM 501 ES (or 601) and what would be the approximate retail cost? Also, are there any used ones on the market? I would rather use a VCR-PCM converter than a DAT to record my original synth pieces and some live classical concerts.*

Danny Ferguson
Springfield, NJ

A My technical grapevine tells me that you may well be in for a long search. It seems that the Sony PCM 601 ES is no longer in production. Officials at Sony noted that the need for outboard converters has been virtually eliminated by the introduction of DAT. I assume your reluctance to try DAT also stems from not wanting to purchase a whole new device when you already own a VCR, but it seems DAT may be your saving grace. Nonetheless, if you are hot on finding a PCM 601 ES, you should try looking through the classified and display ads in *EQ* and contacting the new and used equipment dealers you find there. If there are any readers out there who can help, write and we'll pass the info on.

Hector G. La Torre
Executive Director

WHAT'S THE DIFF?

Q *What's the difference between 2-track, 2-channel and 2-track, 4-channel master recorders? Are the formats compatible if the speed is the same?*

Sandy Bright
Manalapan, NJ

A Two-track, 2-channel recorders are also known as "half-track," while 2-track, 4-channel units are known as "quarter track." The more common mastering format is half-track and is configured as two (left and right) very wide tape tracks. Quar-

This is where your questions get answered. Send your query with your name and address to:
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ter track is the older, hi-fi format and is configured like a big cassette, two tracks in one direction and two tracks in the other. All things being equal (which they seldom are), the half-track tape ought to make a quieter recording. In theory, you can play a half-track tape on a quarter-track machine, but with reduced sound quality. The opposite — quarter-track on a half-track — isn't practical, since you'd be hearing some tracks played backwards.

Jimmy Yamagishi and Bill Stevens
Tascam

THE PHANTOM RETURNS

Q *The following is in response to your December 1991 EQ&A question titled "The Phantom Knows." Mr. Sui's question applies to a broad spectrum of mic inputs not equipped with phantom power. The AKG N62E microphone phantom power supply is internally configured to provide 48 volts to pins 2 and 3 of the unit's XLR-type connectors. This includes both the input and output connectors. There's no DC isolation designed into the N62E, so I believe Mr. Sui's "dull roar" results from the application of 48 volts to the Tascam 688's mic input. A simple modification, which would enable the N62E to be used with any mic preamp, might be a solution. Install a 100 microfarad 50v capacitor in series with each of the leads connected to pins 2 and 3 of the output (male) XLR-type connectors. Make sure the negative end of the capacitor connects to the pins. This internal mod prevents the 48 volts from reaching the input of the mic preamp into which the output of the N62E is plugged.*

Skipp Tullen
Tullen Sound
Morristown, NJ

A AKG Acoustics offers the following further advice:

The N62E is the simplest type of phantom power supply — one which puts a powering voltage out on every input and output, without blocking the DC from the outputs. When the N62E is used with a console input that's not balanced and floating, the phantom voltage present on the N62E's outputs will be shorted out via the console tying pin 2 or 3 to ground. Once the N62E is shorted out, the microphone won't see its required voltage, and hence no audio. If, rather than no audio, you get a "dull roar" (such as Mr. Sui describes), your console's design may use an FET-balanced input, which isn't happy seeing 48 volts on its input. Either way, as Mr. Tullen suggests, you can add blocking capacitors to the N62E's outputs to cure the problem.

However, since the N62E is capable of putting out 52 volts into certain loads, the suggested 50 volt capacitor value is a bit low to provide a complete block. You would usually use about a 2:1 ratio in a capacitor to block a DC voltage. Therefore, a good quality, electrolytic cap in the 100 microfarad, 100 volt range would be our recommendation.

You will need four 100 microfarad, 100 volt (or larger) capacitors.

Add them in series: unsolder the wire that was soldered to the pin of the XLR connector. Solder the negative (-) side of the capacitor to that pin, and solder the positive (+) side of the capacitor to the wire that you removed from the pin (leading to the circuits of the N62E). Do this four times, on each of pin 2 and pin 3 of the two male (output) connectors of the N62E.

The AKG service department in San Leandro, CA, will provide this modification to N62E owners at a nominal charge.

Dave Ogden
Product Manager
AKG Acoustics Inc.

GETTING WIRED

A I believe the recent question of the differences between "balanced vs. unbalanced" and "transformer vs. transformerless" [EQ&A Feb. '92] requires further analysis.

A balanced line consists of two similar wires which carry a signal current from a source (output) to a load (input) over a finite distance. These two wires are twisted together and encased in foil or braided shield. Neither of the signal wires is connected to ground and the shield should be connected to ground only at the receiving device. Thus the signal current always remains independent of the ground system. The two wires are driven by the same signal voltage but with opposite polarity on each wire. The current therefore flows through both wires equally but in opposite directions. This is called the "normal mode" of transmission.

Because the two signal wires effectively occupy the same space, any hum or noise induced into the line past the shield will be induced equally and with the same polarity into both signal wires. This noise is referred to as "common mode."

A balanced input stage consists of a differential amplifier which responds only to the voltage difference between the two signal wires. Any common mode signals are therefore rejected by the input stage. 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.

The high common mode rejection provided by balanced lines is critical in any situation in which the audio lines may be close to AC power, lighting, TV monitors or computers (e.g., any MIDI gear). Since the ground system is not part of the signal path, balanced lines prevent ground loops caused by the send and receive devices being connected to different grounds.

An unbalanced line consists of a central insulated signal wire ("hot") encased by a shield ("ground"). Since the signal is referenced to ground on both ends a ground loop (hum or buzz) may be caused by an AC power differential between the send and receive devices. The shield does provide adequate protection against induced noise if there are no strong

continued on page 78

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EQ TIPS

EDITED BY DAVE BRODY

Got a special technique you want to share? Or maybe a war story from the front lines (no real names, please)? Write or fax us with your unique discoveries on any aspect of audio work. If we like it, we'll print it and give you a free subscription. (One word of caution before we tip off, though: we're passing these studio tricks along non-lab-tested; so implement them at your own risk.) We'll get the tip ball rolling with these:

THE POOR PERSON'S SLAVE REEL

Running out of tracks for those big background vocal parts? Create a rough instrumental mix on your two-track then dump it back to your multitrack on a different area of the tape (or if you're working with MIDI sequences, print the song twice when you first record it). Record your extra parts; mix them down (audition their blend with the rough mix up, then turn it off when you go to tape). Now "wild-sync" them back onto an unused track (or stereo pair) on your multi, recording each entrance separately. It'll take a while, but you'll get them locked in and it'll be worth it. You'll eventually get used to your machines' ballistics and the whole process will go faster.

THE MORE-AFFLUENT PERSON'S SLAVE REEL

Create the slave reel and record as above but, this time, mix down to a sampler. Trigger the sampled sections with a single MIDI event, offsetting as necessary. If you want to be absolutely sure of time placement, sample a section of your rough mix; offset the MIDI trigger 'til the rough flanges with the original track.

CAPT. HOOK'S SECRET PIRATE MIC

Okay, so Jimi's Ghost has just inhabited the body of the guitar player. You want to capture it on your Walkman-style deck — but you forgot your stereo mic. No problem, simply plug your headphones into the mic input and place them somewhere sensible (isolated from vibration). Be careful setting levels and you'll get a respectably hi-fi tape. Listen back on headphones and notice that the stereo imaging is surprisingly good — what you're doing is almost binaural recording.

MECHANICAL MEGA-CHORUSING

Need a big sound on a rhythm guitar or horn section part? Track two separate performances and pan them out in stereo. Nothing special about this so far; but it will be if you change the speed of the machine very slightly for the second pass. Don't let the musician(s) re-tune for the overdub; if the pitch change bothers them at all, it's probably too radical. You'll find this most effective on longer events, like sustained chords.

DO IT WITH YOUR FEET

Effects (reverb, echo) that sound great during a song in a live set can sometimes muddy the band leader's speaking voice between songs. Instead of playing with faders or knobs, hook up a footswitch to mute the send to the digital effects device. This will allow the effect to decay naturally, while still permitting the effect to be turned on/off as required. Setting up a mute footswitch on the device's input also allows you to create special effects on musical cue, e.g., vocal doubling on every chorus

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EQ PRODUCT VIEWS

LISTEN HEAR

JBL is offering a new line of console-top studio monitors. The 4206 two-way monitor features a 6.5-inch woofer while the 4208 has an 8-incher. The 4206 and 4208 were specifically designed for near field applications. A new 1-inch pure titanium diaphragm high frequency transducer was developed for smooth, extended response. JBL's patented "Diamond Pattern Surround" provides control over secondary resonances for virtually flat axial response beyond the upper limits of the hearing range.

MISSING LINK

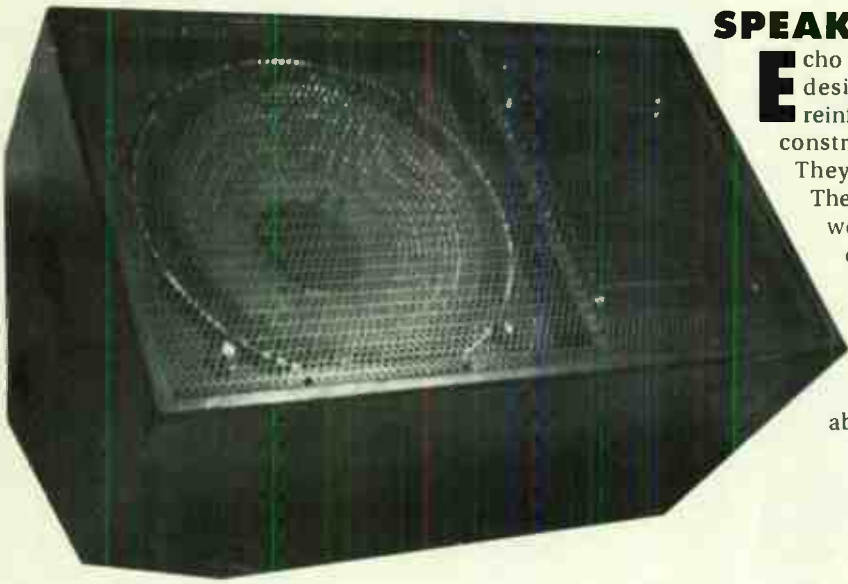
Korg's SoundLink is a comprehensive and versatile random access digital audio multitrack system that's affordably priced. A full repertoire of interfacing capabilities for analog and digital audio, timecode, machine control and other capabilities are included. SoundLink is designed to be at home in any studio environment. Audio production, audio-for-video, video-post and music recording are all tasks that this system was created to handle. From one to eight tracks can be recorded simultaneously. Tracks can be overdubbed, bounced and transferred. A variety of nondestructive editing tasks can be easily accomplished and the system features automation, signal processing that includes three-band EQ for each channel, and complete MIDI capabilities. Contact Korg Professional Audio, 89 Frost Street, Westbury, NY 11590; 516-333-9100. Circle EQ free lit. # 102.

WATT'S UP

This year marks Bryston's 30th anniversary and to honor this milestone, the company has released its 4BNPB 250 watt per channel professional stereo amplifier. The 4BNPB is the fifth generation of Bryston's popular 4B series, and represents a complete redesign. It departs from traditional practice in a number of areas, including multiple smaller filter capacitors for improved high frequency linearity and power delivery, a new input design with a proprietary buffer circuit which reduces distortion in the source by maintaining a completely linear input impedance, and clipping indicators which react to any form of signal degradation, providing more accurate information. For further information circle EQ free lit. # 103.

TRIPLE TREAT

AKG Acoustics' Tri-Power series of live performance mics were developed over several years of intensive R&D. The top-of-the-line D3800 and D3900 vocal mics feature AKG's Moving Magnet Suspension (MMS) system which reduces handling noise by at least 10 dB beyond the latest competitive technology. The D3600 instrument model features careful attention to proximity effect, cardioid pick-up pattern for isolation, and built-in shock-isolated stand adaptor. For more information circle EQ free lit. # 104.



SPEAKER OF THE HOUSE

Echo Audio has developed three speaker enclosures designed to meet the special criteria of sound reinforcement rental companies. The cabinets are constructed from 3/4-inch 13 ply, void free birch. They stand 18" tall, are 30" wide and 18" deep. They feature a 16 gauge perforated steel grille and weigh in at approximately 94 pounds. These enclosures feature a JBL 2226 vented gap 15" speaker that can handle 600 watts. The driver is a JBL 2445 two-incher that handles over 150 watts. Two Neutrik 4 conductor Speckon connectors complete the package. A stage monitor (pictured) is also available. For further info circle EQ free lit. # 105.

POWER PLAY

New for '92 are Yorkville's Audiopro 1212 and 1216 compact self-powered mixers. The 1212 offers 12 channels and the 1216 — you guessed it — 16 channels of comprehensive audio processing. The mixers weigh in at less than 50 pounds each, provide 600 watts per channel, and are loaded with desirable features including: Alesis digital signal processing; phantom power; two monitor sends; two FX sends; fully buffered channel inserts; dual nine-band graphic EQ for mains and monitors; speaker processor with switchable EQ curves; and Yorkville's exclusive Self Correcting Hum Reduction. List price for the 1212 is $2449 and for the 1216, $2699. Contact Yorkville Sound Inc., 4600 Witmer Industrial Estate, Unit #1, Niagara Falls, NY 14305; 716-297-2920. Or, circle EQ free lit. # 106.

SI SR

The new SRP Series of multitrack units from Dolby Laboratories incorporate Dolby SR in a new cost-saving configuration tailored specifically for music studios. The SRP-24 lists for $16,875 — 25 percent less than the XP-2 SR, which (until now) was the least expensive 24-channel unit available. An SRP main frame consists of up to 24 channels, each consisting of a plug-in module with Dolby SR circuitry, input and output amplifiers and level controls, and an LED calibration display and a bypass control. For more info, contact Dolby Laboratories, 100 Portrero Avenue, San Francisco, CA 94103-4813; 415-558-0200. Circle EQ free lit # 107.

CROWN-ING ACHIEVEMENT

Crown's new Power-Tech 1 and Power-Tech 2 amplifiers are specifically designed for professional musicians. Both models provide switchable input sensitivity configurations, which are suitable for applications ranging from sound reinforcement to electronic instruments. They weigh in at less than 35 pounds and require just two rack spaces. The Power-Tech 2 is capable of delivering in excess of 440 watts per channel into 4 ohms and 320 watts per channel at 8 ohms, in addition to delivering more than 900 watts bridged mono into 8 ohms. The Power-Tech 1 delivers more than 300 watts per channel into 4 ohms and 220 watts per channel at 8 ohms. Both models have front panel layouts that are designed for convenience. For further info circle free literature # 108.

British studio performance at a revolutionary American price!

MEGAS Studio is the first 24 buss recording console to combine world-class *British sound* with a wealth of smart, practical features for today's recording situations—*priced from $17,100*!*

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MEGAS STUDIO

CIRCLE 46 ON FREE INFO CARD

*Suggested retail price for the Model 24/24 Megas Studio console. Other prices will vary somewhat based on specific configuration and features.

UK Hideaway

Peter Gabriel's personal studio at the top of a 200-year-old mill building is a perfect blend of rural calm and technological chaos

STUDIO: The Workroom, Real World Studios, Box, Wiltshire, England.
MAIN MAN: Peter Gabriel.
PRODUCER AND ARTIST CREDITS: Founding member of Genesis. Numerous solo albums and film soundtracks, including *Birdy* and *The Last Temptation of Christ*. Collaborations with Youssou N'Dour, Laurie Anderson, Kate Bush, Karl Wallinger and others. Current event: completing the long-awaited followup to his six-year-old multi-platinum album *So*, with co-producer Daniel Lanois.
CONSOLE: SSL 4048E with Total Recall, G Series computer and EQ, integral synchronizer and 8 stereo channels. Modified to provide 2 stereo and 8 mono sends per channel, 4 with MIDI mute. Plus an S series DDA, with 24 inputs, aux send matrixes; it's used as a 48 into 16 MIDI-controlled mixer with Peter's keyboards and samplers going through it.
RECORDER: 2 x 24 track Studer Revox A820s with 48 channels Dolby SR, A820 half-inch/quarter-inch two-track with center timecode, Sony DAT, Studer Revox A700, access to numerous Mitsubishi X850s (and other analog machines in the other three studios on site).
MONITORS: Neil Grant Boxer system, Yamaha NS10s, Meridian M30s, AR18LS.
SIGNAL PROCESSORS: Four Neve 33114 EQs, BSS DPR 402 Compressor Peak Limiter, BSS DPR 502 MIDI Noise Gate, dbx 165 Over Easy Compressor, six Decca Compressors, two Delta Lab DL2s with Memory Module, Drawmer DS201 Dual Gate, Eventide Harmonizer M910, Eventide H3000SE Ultra Harmonizer, AMS RMX16 Digital Reverb, AMS DMX1580S Stereo Digital Delay, Ibanez SDR1000 Stereo Digital Reverb, Korg DRV3000 Dual Digital Effects Processor, Lexicon 480L Digital Effects System, Lexicon PCM70 Digital Effects Processor, Quantec Room Simulator, Rebis RA701 MIDI Gate, Rebis RA401 Parametric EQ, Roland Dimension D SDD320, Roland SDE 1000 Digital Delay, Sycologic M16 Digital MIDI Matrix with remote, two Sycologic M16X MIDI Matrix Expanders, two Urei LA4 Compressor Limiters, 2 Urei 1176 Peak Limiters, Yamaha REV5 Digital Reverb, Yamaha SPX90 II Digital Effects Unit, Yamaha SPX 1000 Digital Effects Unit, TC 1140 Parametric EQs. Plentiful other units available elsewhere on site.
MICS: There were only 103 microphones on site at time of going to press, but more available if you need them! Among the collection are classics and the latest from Neumann, including an M49 and KM185s, Bruel and Kjaer, Electro-Voice, beyerdynamic, AKG, AMS, Soundfields, Sennheiser, Shure, Sony, STC, Tandy PZM, Toa and numerous capsule mics.
THE ROOM: The Workroom really is a room with a view, being in the loft of a tall 200-year-old mill building. Two balconies, one with equipment winch and one leading onto the roof from a den-like mezzanine level, look out over the green English hills and the stone village of Box. The room is a typical loft with polished wooden floors and scattered rugs and during a Peter Gabriel project is a chaotic mass of hi-tech, cable spaghetti and ethnic instruments. Unfortunately, it's so popular for renting clients, he finds it difficult to get in the room. "When we get a booking we just kick him out," says manager Dave Taraskevics. The Workroom is tie-lined for multitrack, talk-back, computers, video and MIDI to two other control rooms and numerous stone and wood acoustic rooms in the building, as well as parts of the adjoining house.
PETS: Maria Lopez's beloved ducks and swans, though they're not allowed in the studio. EQ

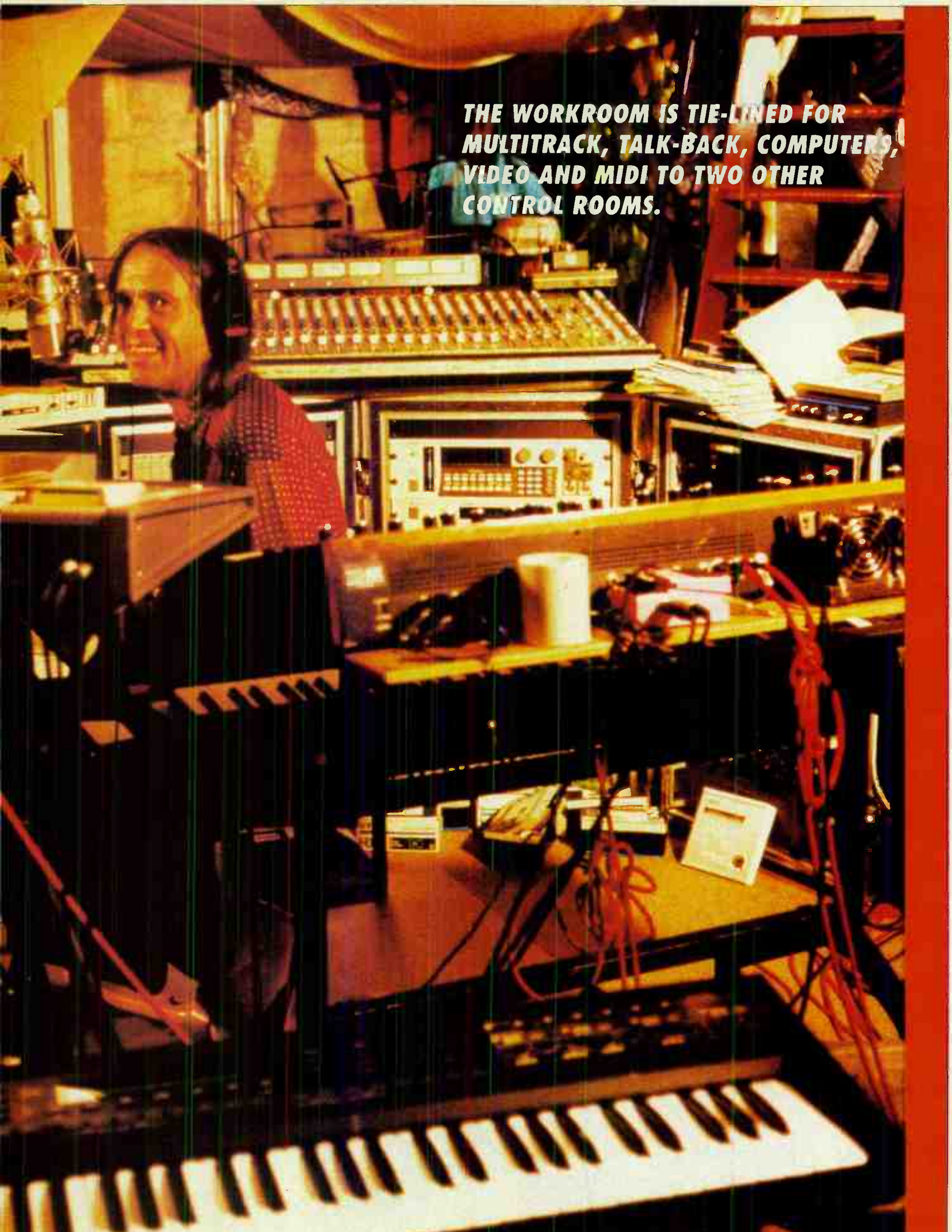
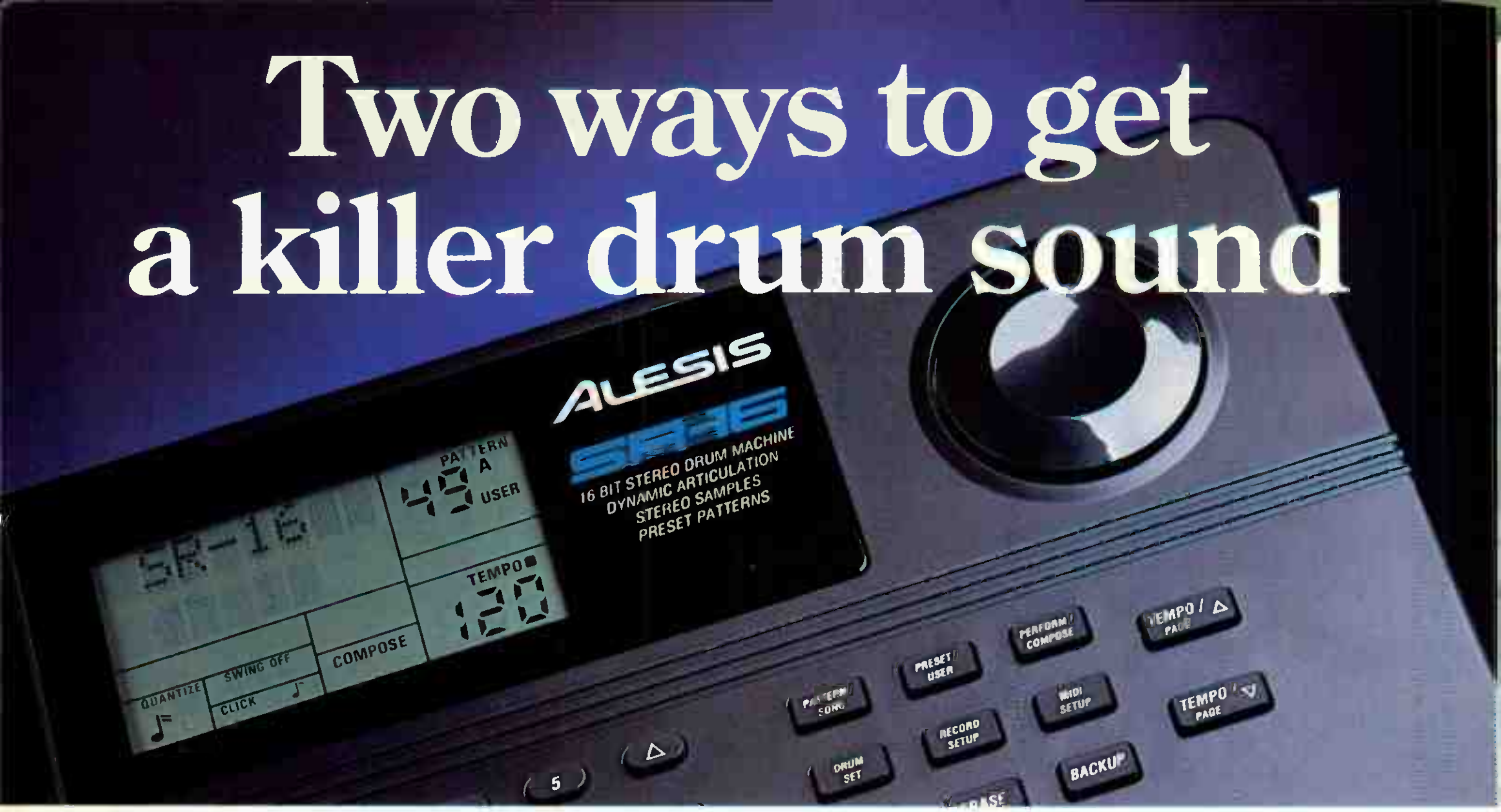


Photo by Peter Williams

Two ways to get a killer drum sound



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ALESIS STUDIO ELECTRONICS

CIRCLE 08 ON FREE INFO CARD

Rush Service

Photo by Andrew MacNaughton

Robert Scovill may be masterminding Rush's "Roll the Bones" tour, but he doesn't throw dice in setting up

TOP GUN: Robert Allen Scovill
CREDITS: Rush, Def Leppard, Tesla, Alice Cooper, Air Supply, Laurie Anderson, Rick Springfield, The Go Go's, etc. etc.
CURRENT TOUR: Rush, "Roll the Bones" 91-92
RACKS: (I currently have two racks aside from the actual system drive. One houses all channel or group insert types of equipment; the other houses "FX" or buss-driven equipment and the desk mix recording equipment.)
Insert Rack, top left and working our way down: two Tube Tech compressors linked with two Klark-Teknik 1/3 octave EQs. These are used on Geddy's vocal channels (love em!) Next, two Summit tube compressors used on bass DI and mic channels, followed by two stereo BSS DPR402 compressor/limiters used on guitar solo channels and acoustic guitar. Lastly two Rane programmable EQs. These are used to drive the side chain inputs on the Tube Tech and Summit compressors to give me the ability to do MIDI programmable frequency selective compression. On the right half of the same rack: 4 channels of Drawmer compressor/gates followed by 16 channels of Drawmer gates all used for gating of drum mics, etc. Next we have four RANE programmable equalizers. These are used for MIDI programmable level and EQ changes on four sampler outputs that carry things like chimes, bells, wood blocks, etc., so I can have preset levels for a given song. Below those are two Urei 1176s used on Alex's guitar subgroup. And, finally, I have a dbx 900 frame carrying eight channels of BBE exciters and four dbx gates.
Second Rack, from the top left: One 2-channel Aphex gate that provides ducking of four audience mics for Geddy's micro monitors and also for desk reference mixers. Next, Lexicon PCM-60 and PCM-70, which are primarily drum reverbs. Followed by three SPX 90s, used on those sampler outputs mentioned above. Under these is a dbx 120X, which is basically a "boom box" by description (great on bass drum, toms and keys). Next I have two TC 2290s for vocal echoes and chorus/flanging. Next in line is the Eventide H3000 for keyboard FX, etc., followed by the Sony DRE2000, used primarily for vocal reverb (and which also contains beautiful sounding delays). Last is the Lexicon 480L for snare plates and room sounds for the rear speaker system. Right side: Starts with the MEP-4s. These are used in conjunction with a Yamaha MPC-1 and MFC-1 to map out all program changes for the show. It also gives me the ability to dump and store my mapping for a given tour to be recalled later, which I do going from tour to tour. Next is a BBE 802 and a Urei 1178, which is part of the insert chain on the main stereo bus. Under that a Tascam CD player and 2 SV-3700s, used for walk-in music and recording reference mixes. Last we have an old Dolby 301 unit with the lo process cards pulled. This is used as a type of exciter, mainly on Geddy's vocals and acoustic guitar (sounds wonderful!).
CONSOLES: two Gamble EX 56 channel consoles with selective interlink to create one 112 input console.
MAIN LOUDSPEAKER SYSTEM: Electrotec Productions Lab-Q system (JBL loaded).
QUAD LOUDSPEAKER SYSTEM: Electrotec Productions Lab-Q on a 360 degree beam (JBL loaded).
PRO TIPS: Become a musician, mentally that is; learn to think and communicate on a musical level. If you're an actual musician, no matter what your skill level, it will make your approach to tone and mixing a more musical undertaking as opposed to a technical one. Also, if you understand pitch and tuning it will influence your perspective on frequency layering and instrument and voice positioning in a blend. No matter how many bells and whistles you have sitting in front of you, it's still music you're trying to communicate to the audience not technology.

(The gear in the rack not mentioned here is set aside for any opening act. Rack assembled and maintained by Ted Leamy.) EQ

Apocalypse NAMM Revisited

The showdown at showtime in Anaheim showed more new trends than technologies, stay tuned

BY CRAIG ANDERTON

How did the recession affect NAMM? Well, those who count on NAMM to get clothed each year probably came away disappointed by the fact that far fewer free T-shirts were being given out. Those looking for new high-tech breakthroughs were probably a bit miffed as well, since the recession seems to have cut into research and development. Despite the economic downturn, though, the dealers appeared quite happy. With few significant new products being introduced, they won't have to sell off all their old stock at a ridiculous discount in order to order up what's new.

Even so, the NAMM show had its own fair share of significant developments. Maybe nothing can be characterized as a breakthrough. Still, several music technology trends bear watching in the months ahead.

MIDI MACHINE CONTROL

It started at the 1987 AES, and reached fruition at the 1992 NAMM: finally, tape, hard disk and video recorders can be considered peripherals in a MIDI system. MIDI machine control defines commands for autolocation, punching, transport functions, and much more, with all commands traveling over the MIDI bus. This will revolutionize not just traditional audio studios, but post-pro as well.

• Fostex got a jump on the idea by implementing machine control with their own proprietary code, and showed Atari/Fostex systems where C-Lab's Notator, Dr. T's Omega and Steinberg's Cubase controlled various Fostex recorders. But Fostex is now on the MMC bandwagon, and will update their MTC-1 controller to speak MMC.

• Other software companies are working MMC into their products. Mark of the Unicorn, Opcode, Passport, Twelve Tone Systems and many others have expressed interest; and Opcode's Studio AV, a 1U rack video deck transport controller for all major video decks and some audio decks, supports MIDI Machine Control messages. Roland's DM-80 is also slated to respond to MMC.

• The Alesis ADAT handles machine sync via MMC messages, as does their companion BRC (big remote control). JLCooper Electronics supports ADAT with the dataSYNC, a device that extracts MIDI Time Code from the ADAT 9-pin sync connector to provide master timecode for driving MIDI sequencers, thus making it unnecessary to give up an ADAT track for sync. The dataSYNC also converts ADAT transport functions into MMC messages. Cooper's Media Control Station, a hardware interface with tape transport-style controls and a jog/shuttle wheel for controlling MIDI sequencers, generates MMC messages too.

• Tascam's MMC-100 Interface Unit translates MMC data to or from Tascam's 238, 644, 688, TSR-8, MSR-16/MSR-16S, and MSR-24/MSR-24S recorders; it responds to commands generated from sequencers, including fast forward, rewind, stop, play, record, record enable (up to 24 tracks) and up to eight autolocation registers. The MMC-100 also can generate SMPTE and provide SMPTE-to-MTC conversion.

Illustration by Alex O'Neal

COOPERATION IS THE NAMM OF THE GAME

Maybe it's the end of the Cold War, but manufacturers seem more and more interested in working together on projects than ever before. MMC is just one example. Opcode announced that Steinberg, Lone Wolf, Dynaware, and Digidesign will all support OMS, the Opcode MIDI System for MIDI studio instrument definition. The Atari and IBM booths were packed with competing software developers all showing off their wares, generally to packed crowds. You could even go to the Passport booth and not see just the latest version of Pro 5, but also Mark of the Unicorn's Performer sequencer running under Producer, Passport's hot-shot multimedia cue sheet program.

It also seems that if something has a disk drive and can play sequences, it's Standard MIDI File (SMF) compatible. The Kurzweil K2000 can read SMFs; Korg's new 01/Wpro and 01/WproX (upscale versions of the 01/W with more keys, more sounds, more programs, and other new features) can also read and write SMFs, as can the Roland JW-50 Music Workstation (a 16-part multitimbral, 24-voice machine with 16-track sequencer) and Kawai's Q-55 stand-alone sequencer. Peavey's MIDI Streamer, a mass storage disk drive for samples, sys ex, and other MIDI data, can also load, record, and play back Standard MIDI File sequences. And CodeHead's MIDI Spy software for the Atari can not only record MIDI data in the background while you're working with another program (never lose another musical idea again!), but can play SMFs in the background while other programs are in use.

> Maybe it's the end of the Cold War, but manufacturers seem more and more interested in working together.

Another development that seems to be gathering momentum is SMDI, the SCSI Musical Data Interchange. This new protocol essentially specifies how to send MIDI Sample Dump Standard digital audio over SCSI, thus speeding up data transfer by a factor or 50 or greater, compared to transferring digital audio data over MIDI. The Peavey SP already supports SMDI; Kurzweil is rumored to be looking into adding it to the Kurzweil 2000, and a Passport representative confirms that the next version of Alchemy will support SMDI. (Passport is also doing what it can to promote the SMDI concept, since music industry adoption of this standard would lead to much easier design of sample editing software.)

There was consolidation as well as cooperation: Jim Dunlop bought the Herco pick company, BBE bought G&L guitars and in a major surprise, Ensoniq will now be distributing C-Lab's line of sequencers in the U.S. With Mac and IBM versions of the acclaimed Notator program sitting in the wings, Ensoniq may be buying into multi-platform software that's a real contender.

BIG MACKIE ATTACK

The Mackie mixer must be a rousing success, because there were a lot of manufacturers trying to get their piece of the $50/channel or under action. DOD's 1642 16-Channel Mixer, 1222 12-Channel Mixer and 822 8-Channel Mixer are available in rack or tabletop configurations. Roland expanded their line with the 4-rack space, M-160II 16-input line mixer; ART showed the 16-channel Phantom 1608, 24-channel Phantom 2408 and 32-channel Phantom 3208. Just another line of mixers? Well, the 3208's price raised a few eyebrows: $1199 for 32 channels.

Tascam showed the MM100 and MM200 Keyboard Mixers. Both have 8 stereo input channels; the latter includes BBE circuitry for enhanced high frequency definition. There were also some rack mount mixers, such as the Rolls Corporation RM81 MixMax 8-channel mixer.

Big mixers were in evidence, too. The Alesis X-2 Recording Console is a full-blown, 24-channel mixer with three 56-pin connector blocks for three-cable connection of up to three ADAT digital tape recorders. And Mackie wasn't resting on its laurels; they showed three 8-Bus Mixing Consoles (16X8, 24X8 and 32X8), ranging in price from $3000 to $4500.

AUTOMATION RALLY

Inexpensive MIDI automation, and

continued on page 78

CRAIG'S ANNUAL NAMMINATIONS

So those were the big trends. And now, for some awards to products deserving of recognition (sorry, no cash prizes):

• Roland picked up the **I Hate to Read Manuals Award** for the Boss ME-10 Guitar Multiple Effects. It's a standard multiple effects unit with the usual effects, but each parameter has an associated membrane switch. To program, just push the switch, and set the value with an alpha dial — no screens, no scrolling, no LCD, no sweat.

• The **I'm Not Dead Yet Award** is shared by Roland for their GR-1 guitar synthesizer, which takes pitch-to-MIDI conversion to new levels of efficiency and reliability; Quantar, whose MIDI guitar does for ultrasonic scanning what Yamaha's G10 should have done; and Zeta, who figured out how to make fret wiring work. Now that guitar synth technology works, it's too bad that people are retaining prejudices held over from the first generation of products.

• Alesis garnered the **Vaporware of the Show Award** for their S4 QuadraSynth Sound Module, a $995 rack-mount module with 64 voices and up to seven simultaneous effects, independently assignable to four effect busses; and S5 QuadraSynth Master Keyboard, a 76-note keyboard version of the S4. You couldn't hear it or play it, but it still got a helluva buzz.

Runner-up: Oktal's Multitude 2.0 sequencer. Not too many people noticed what was going on at their unassuming, out of the way booth, but once this program hits the real world, it looks like it could carve out a share of the sequencer market for all platforms — Mac, IBM, and Atari ST/TT.

• The **Tubular, Dude! Award** goes to ADA, who is rumored to be designing a brand new vacuum tube in Russia that's optimized specifically for low-noise musical applications.

• The **Talk is Cheap, But What About Recordable CDs? Award** goes to the Marantz CDR600 Compact Disc Recorder. At $7500, it's a lot less expensive than competing units that let you create Philips Red Book and Orange Book compatible test CDs.

• In a related category, the **Vaporware of the Decade Award** goes to Radio Shack for their Thor recordable CD system, which was announced years ago amid great fanfare — so where is it?

• The **Good Fences Make Good Neighbors Award** goes to Demeter Amplification for their Silent Speaker Chamber SSC-1 and Stereo SSC-2; these are completely sealed speaker cabinets that allow for miking noisemakers like guitar amp outputs with minimal leakage and noise.

• The Hammer Jammer was, without a doubt, the prime candidate for the **Weirdest Gadget of the Show Award**. This unusual accessory for guitar positions six small hammers just above the guitar strings. These connect to six buttons that you can tap like typewriter keys, to do tricks like lightning fast multi-string runs, or hitting all six strings at exactly the same time instead of strumming them.

• The **We Haven't Forgotten About You Award** goes to Marion Systems for providing a SCSI hard-disk interface for the Akai MPC60 and MPC60-II.

• The **Coolest Guitar of the Show Award** always has intense competition, but the Parker Guitar, distributed by Korg, was the hands-down favorite. This ultra-lightweight guitar has piezo transducers (for acoustic sounds) and standard magnetic pickups for electric sounds.

• The **Speaker of the House Award** goes to ART, for their Attack Module guitar amps. The cabinets are made of carbon fiber to provide much higher density than wood, at a fraction of the weight. The small size and clean sound makes them good candidates for amplification in the project studio.

• Rane picked up the **Your Martin's Plugged Into What?!? Award** for the MAP 33, a high-tech instrument preamp designed specifically for acoustic instruments.

• The **Pet Brain Award** goes to Atari for the STBOOK, an 8.5 x 11 x 1.5 inch notebook computer with — yes — built-in MIDI ports.

• The **Half a Loaf is Better than None (and Certainly Less Expensive) Award** went to Mark of the Unicorn for their Waveboard, a $1500 NuBus card that provides direct-to-disk, random access 16-bit digital recording. Why half a loaf? You provide the AES/EBU or SPDIF A/D-D/A converters (such as those found in DAT machines, or stand-alone units).

• Finally, the **Wait Until Next Year Award** goes to all the manufacturers who had outstanding products that didn't get into this column because either I didn't see the product in question or there wasn't the space to include everybody.

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Mix It Raw

The first time I heard "Teen Spirit," the first single from Nirvana's number one debut album *Nevermind,* I knew it was an incredible song. And that's saying a lot, considering I was hearing a completely distorted demo recorded in a basement, on a boom-box, through a cheap PA.

I couldn't even hear the vocals on that version, but when we went into rehearsal with it I could hear — everything. It was incredibly intense and loud. The first couple of times they played the song I was up and jumping around the room.

The entire recording of *Nevermind* was a tremendous experience. It was both Nirvana's and my first time recording with a major label, although I had worked with them on some demos for Sub-Pop, an independent label. The band was concerned about "selling-out" and losing their intensity and raw energy. It was my job to capture them live and reproduce their energy and passion as best as I could.

We never expected this kind of reaction to the record, though. I mean, achieving a number one album on the same chart as new releases from Michael Jackson, Guns 'N Roses and U2 — it's still hard for me to believe.

I think one of the reasons that people have responded so well to this record is that it sounds honest and real. It doesn't have a real high-processed sound where everything is perfect and glossy. You have all these high-tech productions all over the radio, and here's this band that's passionate and real and you can hear that they gave everything on each and every track.

Despite the sound, this wasn't some basement recording. In fact, we recorded at Sound City, which is an older studio in Van Nuys, California, using this great old Neve board that gets in the way as little as possible when letting the sound through. I think that board had a lot to do with capturing the live sound. I also didn't use much signal processing when we were recording. No tape on the drums either — I just tried to get the drums to sound as good as they could in Sound City's big room. I was going for the hottest sound I could get from microphone to pre-amp to tape.

The mics I usually use are Sennheiser 421s and Shure 57s and Neumann U87s, but we also use some great old tube mics, like Neumann U47s, U48s and U67s. I like the fatness and warm sound that tubes produce. I used EQ only when it was absolutely necessary. If something needed more bottom or top, I'd EQ it, but I usually don't like to do a whole lot of processing. I try to make it sound as good as the original source.

NIRVANA'S DOWN-TO-EARTH METHOD

Working with the musicians was what really made it happen. In order to keep the band's intensity, I tried not to bore them. Most of the time I was engineering, they weren't even in the studio. I didn't want them to sit around bored, waiting for me while I was trying to get a certain drum sound, or burned out on a song because they kept hearing it again and again.

Even when we were recording we wouldn't stay on one song for too long. Kurt Cobain, lead vocalist, guitarist and songwriter, told me he was very impatient. That meant that whatever was being played, whether a warm up or whatever, was always being recorded. If we didn't get something right away we'd just move on to something else and go back for it later. I think that's why the band always sounds fresh — they kept their spontaneity while recording.

Raw Energizer: Butch Vig

Photo by Brent Nicastro

Capturing the high-intensity sound of Nirvana meant reevaluating the strengths and weaknesses of high technology

BY BUTCH VIG

NIRVANA FOR THE MASSES

Kurt is an amazing songwriter. He has this knack for wonderful pop sensibilities, even though it's amidst all this heavy metal noise and chaos. He writes these really strong melodies with lyrics that are intriguing — filled with rage and mystery. You may not always understand what he's thinking, but it draws you into the songs just the same.

There's a whole audience of young people who haven't heard stuff like this before. Sure, they've heard heavy metal and punk before, but here it is with this commercial-wide appeal to it. And besides, this is how music began and should be — people playing their instruments.

It's refreshing to hear something that honest on the radio and see it achieve mass popularity. I'm sure we'll have more processed bands appearing on the charts, but at least Nirvana has made people turn their heads a little — especially people in the industry.

> I try to make it sound as good as the original source. I know that may sound cliched, but it's certainly true.

It's tough to say what Nirvana will do next. What happens next depends on Kurt. They might do a rawer album, or maybe something acoustic. Perhaps a real slick pop record. Kurt's a good enough songwriter to pull all those off.

Whatever they do, they're going to do it with energy and spontaneity — and, of course, *volume*. EQ

In addition to producing and engineering Nevermind, Butch Vig has just finished an album for L7. He usually records at Smart Studios, in Madison, WI.

Digital In/Out Ups and Downs

In the familiar world of analog, connecting one device to another is easy: just get the right connector and appropriate impedances, make sure that the wiring configurations are the same, and presto! You're in business. Digital, on the other hand, represents something of a double-edged sword.

So what's the problem — a bit's a bit, right?

BY MEL LAMBERT

On the one hand, we have outstanding audio fidelity, courtesy of 16-bit PCM recordings, linear frequency response beyond 20 kHz, and the ability to perform repeated dubs or transfers with no noise build-up . . . the list continues. On the downside, however, we have the inescapable fact that many manufacturers aren't playing fair with digital interfaces. Put succinctly, few of us are aware that, even where standards do exist, it's rare for a digital I/O to be designed to provide the sort of compatibility that we need in the studio.

How did we reach such a strange situation? Primarily, I would suggest, because the majority of early-generation systems — including DAT recorders, CD players and even workstations — were derived from consumer-grade hardware. For domestic use, all we need to ensure is that the CD's output can be connected via phono or an optical connector to, let's say, a DAT recorder, and the digits printed to tape. For professional users (like me and thee), however, that's often just the beginning. We need to make sure that important other information is generated by the sending unit, carried correctly across the digital interface, and then decoded correctly at the receiving unit.

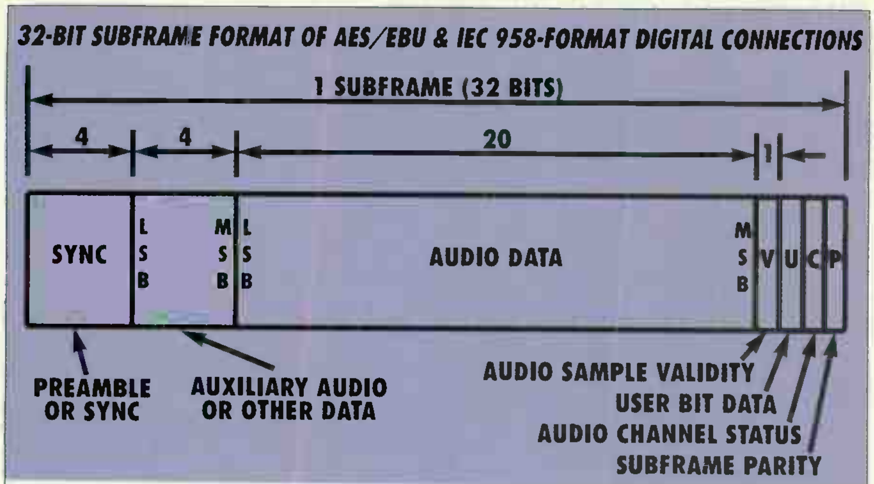
Furthermore, despite the fact that standardized connections and protocol definitions have now existed for a number of years, the reliable and trouble-free exchange of digital data still cannot be taken for granted. Yet this need not be the case. With a small amount of cooperation from hardware manufacturers, who should be more forthcoming with how they've implemented various I/O formats, life for digital-conscious *EQ* readers will be far less complicated!

PRO VS. CONSUMER INTERFACES

Confusion arises, primarily, when we attempt to interface units that are derived from consumer-style hardware, including some (but not all) CD players, signal processors and DAT recorders, with "professional" systems. In the consumer world, most interfaces are configured to the IEC 958 Type II specification. (I'll avoid talking about hardware that was built a couple of years ago and which might

Socket to ya: Panasonic's SV-3700 Pro DAT back panel eases the ins and outs of digital recording.

Figure 1



Of the four ancillary bits, the Validity Bit indicates whether the previous audio sample data bits are secure or error free; the Parity Bit is set to provide even parity over the current subframe (and hence enables simple detection of transmission errors); and the User Bit enables sending equipment to designate hardware- or system-specific information.

feature interfaces labeled, "CD/DAT," "S/P DIF" — standing for "Sony/Philips Digital Interface" — or CP-340 Type II; many of these were configured to older, often radically incompatible standards.)

Most professional hardware, on the other hand, is supplied with "AES/EBU" interfaces, meaning that their designers have (hopefully) read and understood the standards document AES3-1985 "AES Recommended Practice for Digital Audio Engineering — Serial Transmission Format for Linearly Represented Digital Audio Data." (In Europe, the European Broadcasting Union — EBU — has republished the standard document; the result is electrically identical to AES3-1985, apart from the specified use of transformer-coupled inputs and outputs.)

Both of these important standards documents share a common ancestry and, not surprisingly, describe a relatively similar, self-synchronizing digital interface scheme. In essence, IEC 958 Type II and AES3-1985 define a technique for sending two channels of digitized audio from one device to another over conventional shielded cable, for distances up to 300 feet. AES3-1985 specifies 3-pin XLR-type plugs and sockets, carrying balanced, RS422-compatible signals, while IEC 958 Type II describes unbalanced RCA-style phono plugs and sockets carrying lower-level signals. (Actually, IEC 958 also contains a section designated "Broadcast Use," describing a specification identical to AES3-1985.)

The digital data is divided into two subframes, with 32 bits of data being transmitted in one sample. At sampling frequencies of 48, 44.1 or 32 kHz, each subframe comprises a 4-bit preamble or sync group; four bits of auxiliary data (or, in the case of AES3-1985 interfaces, four additional data bits for a possible maximum of 24, if the sample word extends beyond the default 20 bits), 20 bits representing the sampled digital audio, plus four ancillary bits for Validity, User Bits, Channel Status and Parity. The digitized data is sent LSB (least significant bit) first, with alternating subframes for channel no. 1 and then channel no. 2. The Channel Code used to transmit the data is biphase mark from the family of self-clocking Manchester Codes (binary frequency modulation).

CHANNEL STATUS DIFFERENCES

So far so good. Aside from connector type, signal levels and balanced/unbalanced operation, it's the definition and use of Channel Status that delineates use of AES/EBU and consumer interfaces. It's the Channel Status that carries unique and operationally vital data concerning emphasis, sampling frequency, and a host of other information.

For the professional AES3 interface, Channel Status data is sent in blocks of 192 bits as 24, eight-bit bytes. (At a sampling frequency of 48 kHz, these blocks have a repeat interval of 4 mS.) Subframes are separated by a unique 4-bit preamble/sync group, which designates the start of each data sequence. Three unambiguous types of preamble are used to designate the following unique conditions: the start of subframe A (channel no. 1), and hence a frame, but which also marks the start of a new Channel Status data block; the start of subframe A, and hence a frame; and the start of subframe B (channel no. 2).

Although a byte-by-byte explanation of Channel Status would take up more space than I have available here, Chart 1 (*opposite page*) illustrates the primary aspects of the AES3 Recommended Practice. Reading Chart 1:

FOR BYTE 0:

Bit 0 — a "1" implies professional I/O; "0" the consumer interface.

Bit 1 — a "1" implies general data; "0" audio information.

Bit 5 — a "1" implies Source sampling frequency unlocked; "0" Fs locked.

FOR BYTE 4:

Digital Audio References Signal; Grade 1, Grade 2, or unspecified.

FOR BYTE 5:

Reserved, but undefined at present.

FOR BYTES 6 TO 9:

Alphanumeric Channel Origin data, comprising 7-bit ASCII, odd-parity,

byte 6 being the first character.

FOR BYTES 10 TO 13: Alphanumeric Channel Destination data, comprising 7-bit ASCII, odd-parity, byte 10 being the first character.

FOR BYTES 14 TO 17: Local 32-bit Sample Address code, serving as a recording index counter.

FOR BYTES 18 TO 21: A 32-bit Time-of-day Sample Address code, which refers to the time at which the first sample of the current block was originally recorded.

FOR BYTE 22: Channel Status data validity flags.

FOR BYTE 23: Channel Status cyclic redundancy check character.

In contrast, the IEC 958 Type II ("consumer-use") interface defines a Channel Status information whose 192 bits are divided into 12, 16-bit words, as depicted in Chart 2, *right.*

The following 10 bytes (bits 30 thru 191) are currently unspecified, but are often used to carry hardware-specific data. A compact disc player, for example, might output various subcode data, while a DAT recorder might transmit Start ID information across the interface.

Of immediate relevance is the different position within the consumer-use and professional Channel Status information for data containing sampling frequency (bits 6-7 for AES3, and bits 24 thru 27 for IEC 958 Type II), channel mode and other important data. Problems normally arise when such connections work satisfactorily for certain, but not all, instances.

CONSUMER AND PROFESSIONAL: TWO, INCOMPATIBLE WORLDS

Put succinctly: IEC-format I/Os are intended to carry information appropriate to consumer use, and hardware constructed to lower operational tolerances than those designed for day-to-day use in a professional environment. AES-format I/Os take these factors into account, including connector choice, balanced operation, clock stability, Channel Status IDs and a host of other well-documented operational factors. My advice to users and manufacturers alike is very simple: choose and correctly imple-

continued on page 67

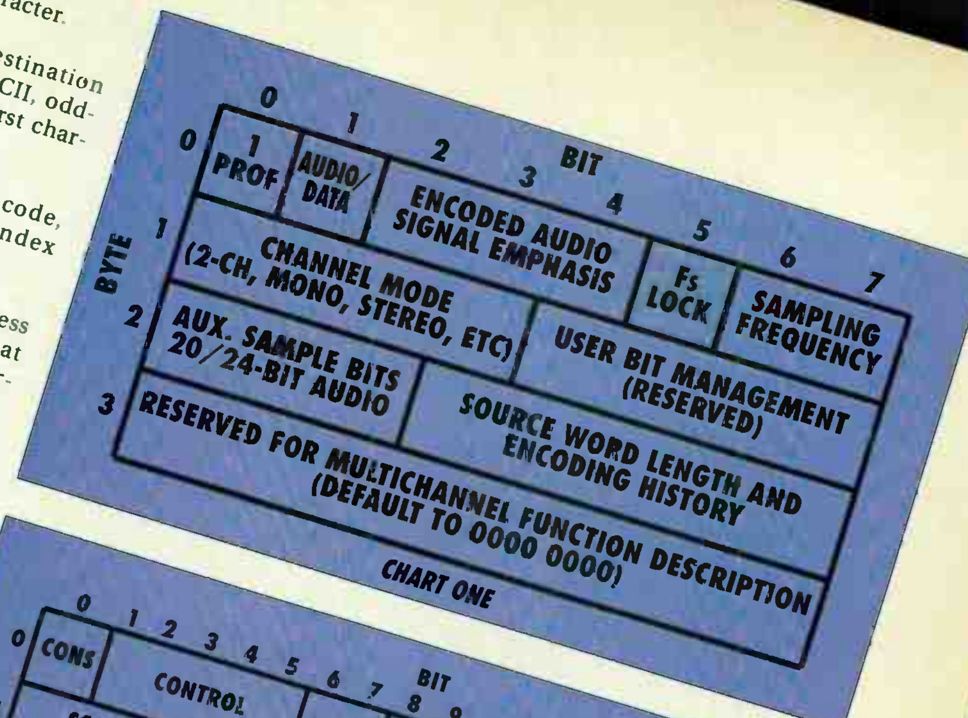


CHART ONE

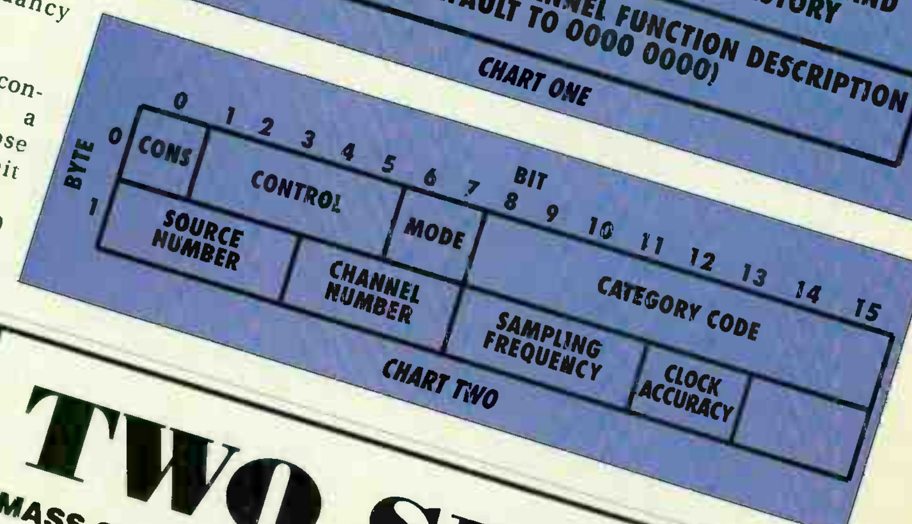


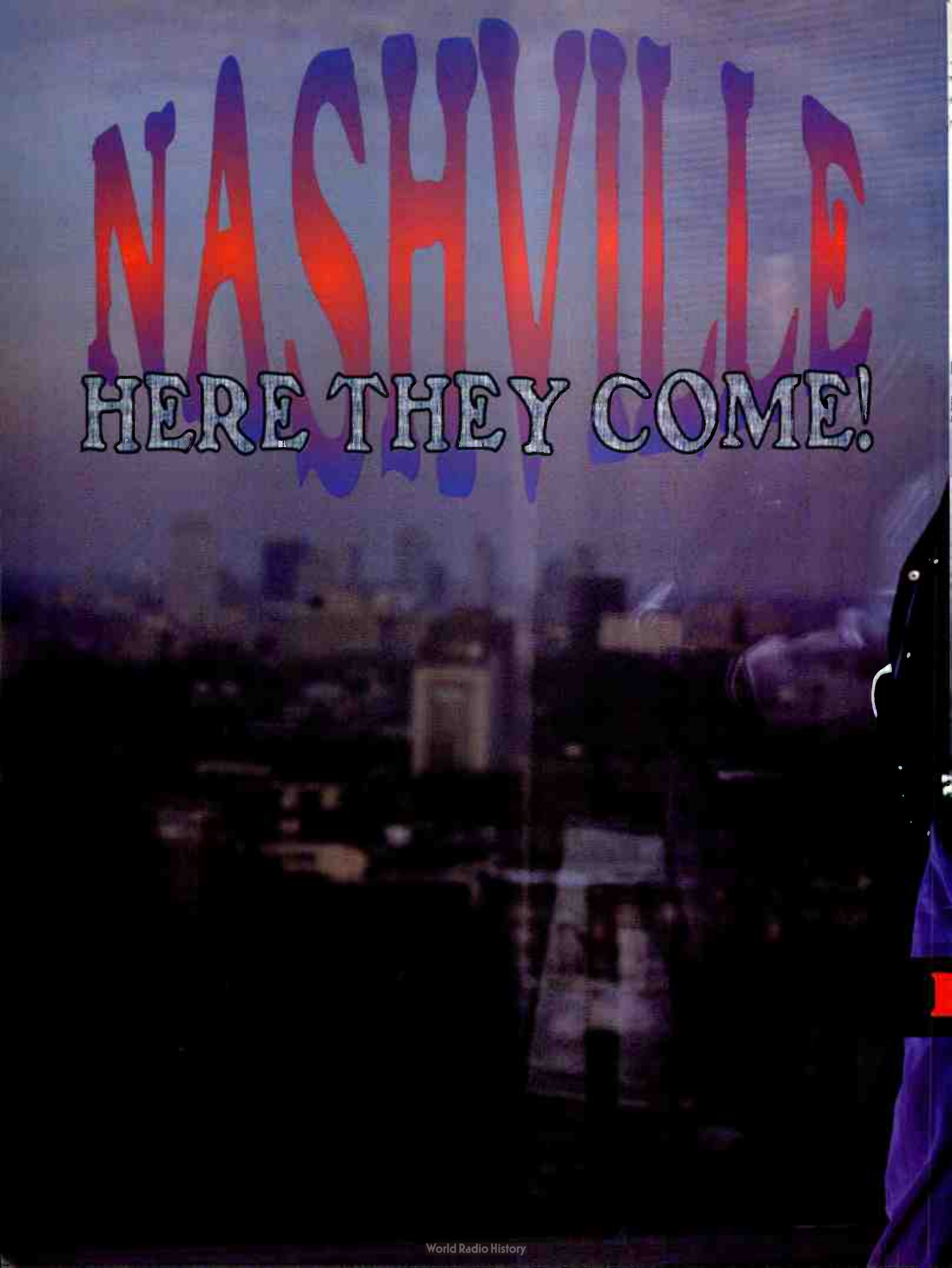
CHART TWO

1. Should you ask a lot of demanding questions before buying a 16-bit sampler?

2. Does it have 20-bit D/A conversion to insure 16-bit fidelity?
3. Does it have multi-mode filters with resonance?
4. Are there multiple Performance locations with "snapshots" of parameters?
5. Is there Truncate Fade-in and Fade-out for "clickless" start and end points?
6. Does it have positional (horizontal) crossfading between Patches?
7. Does it have Time Stretching, and if so, can you see the pitch alteration?
8. Does it have realtime audio digital stereo output versus standard digital I/O only for backup?
9. Does it have Note Number Exclusives for cutting off one sample with another?
10. Does it have an RGB or composite monitor output?
11. Can you combine the velocity-switching, mixing and crossfading of up to four samples on one key?
12. Can you expand the memory with standard Macintosh SIMMs?
13. Does it have an "Undo" or "Recover" function?
14. Is there a mouse port for fast and easy programming and editing?
15. Does it have realtime "Scrubbing" for locating edit points easily?
16. Does it have four-stage rate and level envelopes?
17. Does it have Templates for setting up TVA and TVF envelopes quickly?
18. Is there a comprehensive, world-class sample library available from the manufacturer?
19. Does it have Normalizing?
20. Does it have a Digital Filter with +/- emphasis for permanently "EQing" samples?
21. Is there Auto-looping?
22. Is there realtime aural feedback when looping, and is it easy to do?
23. Are there multiple LFO parameters and waveforms?
24. Is there an Alternating Loop mode?
25. Can you load or save Performances (multi-timbral setups)?
26. Is there matrix modulation in the Patch Control page for flexible control routing?
27. Does it have Analog Feel to make certain Patches sound less "sterile?"
28. Does it have accurate Phase Lock or will it lose stereo imaging when active?
29. Does it have digital Compression and Expansion?
30. Does it have Wave Draw allowing you to actually redraw the waveform?
31. Does it have an Insert function for splicing data into the middle of other data?
32. Does it have an Area Erase function for erasing data but leaving the space (time)?
33. Does it have a sample Mix function with a delay parameter?
34. Does it receive Polyphonic aftertouch?
35. Are there different Velocity Curves available?
36. Is there an Index and Jump function for accessing any page in the sampler easily?
37. Are there "Select" windows for finding and assigning samples patches, etc. quickly?
38. Is there a Volume ID with view field or similar cataloging system?
39. Does it have a 48kHz sampling rate through the digital and analog ins?
40. Is there an "Auto-patch" feature which automatically makes full-blown patches from your samples in a matter of seconds?
41. Does it have resampling, including the ability to resample an entire Performance?
42. Does it have a pre-trigger parameter so as not to lose the attack of your samples?

Yes.

NASHVILLE
HERE THEY COME!



The "Capitol" of Country Is Now the Boom Town For All Pop Music

BY JIMMY BOWEN

I'm already on public record with my prediction that by the year 2000 Nashville will be the music capital of the whole country — not just of country music. Every day more and more young people are moving to Nashville, even transplanting from New York and Los Angeles, because Nashville's such a vibrant, reinforcing music environment. In Nashville you can walk down the street on the way to the store and run into a dozen people in the music industry. You can cut a record or get a deal in no time here because there are a ton of studios, publishers and record companies so close together. It's a tightly knit musical community and that's very nourishing to the creative process. It's also still a small town overall, about a half million people, most of whom are vitally interested in making music and in the music business. Nashville, in other words, is to music what L.A. is to film. Musicians, writers or engineers, especially those with young families, can come here and live relatively cheaply compared to New York or LA. Early in my career, I spent four years or so working in New York and then sixteen in Los Angeles in their respective music industries, and by the end of my time in L.A. I was afraid for my family's safety.

PHOTOS BY BETH GWINN

they thought I was crazy. One of the first acts I produced was Mel Tillis, who speaks with a bit of a stutter to begin with. When he saw the bill for the studio time for the record, he couldn't talk at all. I'd spend a couple of hours getting a drum sound and they'd give me strange looks.

Of course, they didn't really have drums then...or at least none you could hear. They used brushes most of the time, but I knew that a generation grows up with certain sounds and the last couple had grown up with a strong beat. They'd never start taking country seriously until it gave them that sound, and until it caught up with LA and New York music production. The whole town was about ten years behind the times in the late Seventies. Through nobody's fault, the music publishers had become the producers, and usually if a publisher can hear the lyrics, he's satisfied with the production.

About six years ago, we finally got the SSL boards and Mitsubishi 32-track recorders and that's the way we've been recording ever since. We rent the JVC two-track recorders from Glenn Meadows at Masterfonics and, so far, nothing has beat out their sound.

Something I do love from the old analog days are the great tube mics, especially the Neumann U87 and U47s. For me, there's no need to prefer analog over digital equipment. I get the best sound I can for the particular situation, knowing it will be captured forever by the digital tape. I like to recreate the mood of a live performance when possible, and sometimes use a mixture of analog and digital equipment to do that. I really love the new equipment. Some people are afraid to try new technology, but I'm always looking for something new.

One of the areas I still enjoy very much is mastering, which I do at Masterfonics with Glenn Meadows, the owner of the facility. Although I still enjoy live engineering and mixing, my schedule is so busy that I use ten different engineers to work on our product.

TAKING LIBERTIES . . .

Jimmy Bowen has come a long way from his Fifties teen-idol days in Texas (with such hits as "Party Doll" and "I'm Stickin' With You"). He has been a main force in music production and the music business ever since, but he has had a particular impact on Country music since his arrival in Nashville in 1976. (He has almost single-handedly dragged Nashville production into the vanguard of the digital age.) He started off producing acts like Mel Tillis, Roy Head and Red Steagall and by 1978 had become Vice President and General Manager of MCA's Nashville operation. Later that same year he took the same titles at Elektra/Asylum's Nashville office (signing people like Crystal Gayle, Conway Twitty, Hank Williams, Jr., and the Bellamy Brothers). In 1983, Elektra merged with Warner Bros. and Bowen took over that business. A year later, he was on the move again, as President of MCA Records/Nashville. Its record sales tripled and, in 1987, Bowen produced 23 Top 20 singles (11 #1's) and six Top 5 albums. Still not content, in 1988 he started his own independent label, Universal Records. Then in 1989 he was appointed President of Capitol Nashville. Last January, of course, he metamorphosed Capitol Nashville into Liberty. By the time you read this he'll probably be Governor of Tennessee.

—Greg Collins

AND LIBERTY FOR ALL . . .

Thursday, January 23, 1992 was the first official day of Liberty Records. I really just changed the name (from Capitol Nashville), though. We were already a separate label, but because of the name everybody got confused. It was very disconcerting to my people, for instance, to open Billboard and see we had the number one record in the country and it was listed as a Capitol release. Everybody just kept slashing off the "Nashville" part of "Capitol Nashville," because we weren't familiar to them as a separate company. So now it's Liberty, and at last I'm "at" Liberty. Coming up with that name was a pretty simple process. I asked the parent company what names they already owned and when I saw they owned Liberty, which had been an old company run by a good friend of mine, I grabbed it. What's more, every time you open a newspaper these days, some country is gaining its liberty, and so it seemed a perfect concept for these times, and for Nashville. EQ

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

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

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

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

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WHAT FO

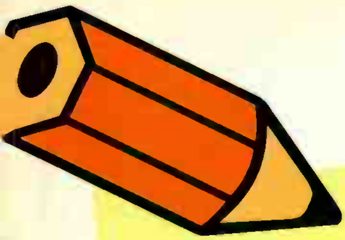
BY DAVID JACOBS

OUR?

Photos by Jon Pickow

So, you're still skeptical of notebook multitracks? You might be surprised at what the new ones can do

What makes today's recording scene so amazing is that there are tools for virtually every job. For times when there's all the time and plenty of budget there's the grand world of the SSL studio, surrounded by all the trimmings. For projects of a more modest means, there's an increasing number of more cost-effective solutions. And for those quick and easy jobs, those demo songs, those preproduction notes, those inspired sessions in a motel room somewhere out on the road, there's a growing number of affordable notebook four-track recorders that do the job — and based on our recent field test, they get the job done real well. Whether you're a wannabe Bob Clearmountain still living at home with your parents, or a don'twannabe Bob Clearmountain who's already making megabucks in the recording biz, these notebook multitracks, as *EQ* magazine likes to call them, have a real and important place.



WHETHER YOU'RE A WANNABE BOB CLEARMOUNTAIN STILL LIVING AT HOME WITH YOUR PARENTS, OR A DON'T WANNABE BOB CLEARMOUNTAIN WHO'S ALREADY MAKING MEGABUCKS IN THE RECORDING BIZ, THESE NOTEBOOK MULTITRACKS HAVE A REAL AND IMPORTANT PLACE.

To help you select a notebook multitrack that has the features you'll need at the right price, *EQ* has tested four new models from Fostex, Tascam, Vestax and Yamaha. All four devices offer four-track capability on a standard cassette and the units cover different price ranges. They all fit in the notebook multitrack category but, as we found out, they're quite different beasts and serve distinctly different purposes.

PLAYING THE FIELD

Basically, our testers fell roughly into two categories. At the lower-priced end were the Fostex X-18 ($399) and the Yamaha MT120 ($520). The upper-priced tier consisted of the Vestax MR-44 ($679) and Tascam 464 ($899). They're as different as night and day. The 464 is the only one of the units to feature EQ on all channels. The X-18 is the smallest and most portable. The MR44 has all controls front mounted and comes with removable rack flanges. The Yamaha features an extremely user-friendly layout. All feature variable pitch and the Tascam, Vestax and Yamaha offer two-speed operation. In addition, MIDI sync capabilities are built into each recorder.

That's the kind of information you can find out by checking the specs on the accompanying chart. But what did we find out about them in the studio?

We tested the machines at Greenhays Records' project studio on Long Island. We first recorded a percussion track from an Alesis HR-16 simultaneously to all four units. Then we overdubbed a live acoustic guitar (Hirade E-9O classical) through a Bruel & Kjaer 4007 studio mic to each machine one at a time. A vocal was then added to each individually (through the same Bruel & Kjaer) and finally a submix of guitar and percussion was added to the remaining track of each unit. Thus the tempo was exactly consistent on each take, but we were able to hear and "feel" each recorder while we added the parts over the percussion. We mixed within the parameters set by each machine using only the pan, EQ, level and submixing capabilities, etc. of each. The final mix was sent through a Yamaha R100 digital reverb unit to a Yamaha MT-1X cassette deck. We monitored the results through a custom-built 75-watt per channel amplifier and Tannoy PBM 6.5 compact monitors. Koss Stereophones were used for cueing.

The results were surprising. First, emphasis must be made that all were capable of delivering a good sound, but tradeoffs for price and features have to be considered when choosing a portable. No one unit can satisfy everyone's needs.

In overall sound quality the Yamaha beat the rest to our ears. Channel separation, presence and high-end capabilities rivaled an analog reel-to-reel. The Tascam offered the most features of the four and has far more sophisticated operations, from its well-appointed input modules that resemble that of a recording console, to its comprehensive LCD display. The Vestax's unique front-panel operation and rack-mountability were big pluses. The Fostex little fella is great as a starter or for the user whose applications demand less sophistication but still need good sound and four-track capability. It's so light and small, the artist can practically carry it anywhere creative juices might start flowing.

So which one is for you? Read on.

SON OF PORTASTUDIO

Want an instant idea of how far these notebook multitracks have evolved since your days as a teenage headbanger? Take a look at the Tascam 464, which has three-band EQ with sweepable midrange on each of its four inputs and two-band EQ on its stereo masters. The four main inputs also feature a trim pot and a pan pot. Two effects busses are featured on all six modules and on the master control module as well. A cue system control module rounds out the picture. The LCD display combines level monitoring with real time and tape distance counting functions and system status. The 464 features Search-to-Zero, Search-to-Cue and precise punch-in and punch-out. Balanced and unbalanced line and mic inputs are right on top of the unit, dbx noise reduction is standard and the solenoid-controlled transport is whisper-quiet, smooth and accurate. Ergonomically, the 464 feels good and its controls are fun to operate. It's also solidly built, yet its light weight belies its appearance.

If your needs point to a sophisticated device that fits into your rack and offers diversity at a lower price point, the Vestax MR-44 features a stereo five-band graphic equalizer at the output stage. The four input modules feature mic trim, assignment, an Aux Send Control (effects send), pan pot, and fader. The LED Vu meters are easy to read and a digital LED readout gives accurate tape count information. The MR-44's solenoid transport is smooth and confident and Zero Return and foot-pedal controlled punch-ins are on the money. We found cue mixing to be variable enough to please all and the machine has a lot of meat to its sound whether through the cue, monitors or playback. The dbx noise reduction improves the unit's S/N ratio to as much as 85 dB, according to the manufacturer.

We found the MR-44 to be easy to operate and fun to work with.

YAMAHA, YEE-HA!

We were simply amazed at how much sound Yamaha has been able to get out of its no-frills, low-cost Yamaha

Roland introduces the Digital Audio Workstation for everyone.

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Model	FOSTEX X-18	TASCAM 464	VESTAX MR44	YAMAHA MT120
Price	$399	$899	$679	$520
Ins/Outs	4/2	12/2	4/2	4/2
EQ	None	3-Band	Stereo Graphic 5-Band	5-band Graphic
Effects Sends/Returns	Yes	Yes	Yes	Yes
Pan Pots	4	4	4	4
Trim Pots	4	4	4	4
Noise Reduction	Dolby B	dbx	dbx	dbx
Punch-in	Yes	Yes	Yes	Yes
Zero Search	No	Yes	Yes	Yes
Cue Search	No	Yes	Yes	Yes
Meters	LED	LCD	LED	LED
Counter	Rotary	LCD	Digital LED	Mechanical
Transport	Top/Mechanical	Top/Solenoid	Front/solenoid	Top/Solenoid
Speed:	1 7/8	1 7/8, 3 3/4	1 7/8, 3 3/4	1 7/8, 3 3/4
Frequency Response (Tape)	40Hz - 12.5kHz	40Hz - 16kHz	NA	40Hz - 18kHz
S/N Ratio	58dB	90dB	85dB	85dB
Distortion	NA	1.5%	1%	1%
Approx. Wt:	2 lbs.	7 lbs	11 lbs	6 lbs.

MT120 ($520). The device features dual speeds, a four-channel ferrite erase head and dual DC servo motors. On the upper deck are four inputs, each featuring fader, pan, Mic/Line trim sliders and aux sends. Aux return (L&R), stereo out (L&R), tape outs 1-4, aux send, monitor out and phones out complete the unit's simple layout.

We found the cue system to be the most comfortable of the lot. While laying the acoustic guitar and vocal down, the sound in the headphones was so pure we found our performance to be superior and the take on the MT120 was the best one we did. The unit also features a microcomputer-controlled transport that seems to operate very well but which we found to be a little noisy; that made us worry about its long term reliability. The MT120 is extremely easy to operate and while it doesn't offer a great deal of mixing capabilities, a more than adequate final mix was achieved.

An interesting trait that the other units don't share is an internal power supply. Thus the danger of losing or damaging an outboard transformer is not a problem. There is, however, a detachable AC cord which could get lost and it has an unusual pin type.

This unit should serve the discriminating artist, such as composers testing out new tunes or musicians working on arrangements for the studio. It's also economically priced enough for some to acquire as a starter unit.

LAPTOP RECORDING

The product that truly pointed to an entirely new category of recording was the Fostex X-18. If your budget is very limited or you need a machine you can take along on fishing trips, the Fostex X-18 is a beauty. At approximately two pounds and measuring 11 1/2 x 6 1/2 inches, it's small enough to stuff in any pouch and take anywhere, without hassle. It's also the least expensive of the four and it's still a basic four-track recorder with sync capabilities. The four inputs simply consist of fader, pan pot and monitor mix pot. Inputs one and two feature high, medium and low trim switches. The device also has a phone level pot, aux send and return and LED meters. In keeping with its reasonable price, the X-18 has a mechanically operated transport that performs just fine. The tape counter is the mechanical type as well. The switchable (Tape or Mix) Vu meters are LED types. We found the cue sound more than adequately comfortable and the final mix was surprisingly clean. And (don't underestimate this point) this unit can be battery operated.

The X-18 is remarkable in that it delivers four-track recording capability at such a low price point and in such a small package. This unit would serve either Eddie Van Halen on tour in one of his four-star hotel rooms or some teenage kid who dreams of being Eddie Van Halen in his suburban bedroom.

SO WHO'S ON DECK FOR YOU?

• Tascam *(circle free lit #109).* We found it suited for the small in-home studio that requires technical sophistication without attempting to compete with larger multi-purpose studios. It's capable of delivering great demos and its broad array of features make it the most complete "studio" of the four. The Greenhays staff felt it would be right at home in their facility for preproduction use. They also expressed interest in the unit for live recording applications.

• Vestax *(circle free lit #110).* Also ideal for the home studio and while its features aren't as vast as the Tascam, it offers a more-than-adequate sophistication for the money. Greenhays liked its rack-mount design and found a spot for it on the wall but, regrettably, had to give it (and the others) back. Musicians might also consider it as a live recording tool installed in one of their stage racks.

• Yamaha *(circle free lit #111).* Is at home in the studio or on the road. Its portability and lower price point make it a good traveler. But its terrific sound qualities make it a useful tool for the most picky user. Cash-poor skeptics would do well to check this unit out first to get an idea what a notebook recorder is capable of sonically. As you can tell, Greenhays was extremely positive about its sound and admitted they were quite surprised.

• Fostex *(circle free lit #112).* A terrific traveler, it accepts both AC or battery power. It would also serve the user on a budget whose needs include home demo recording since its sound is very good and, with the addition of some outboard gear, could give the bigger guys a run for the money.

Some time spent at your local dealer will help you answer which one you'll make the most beautiful music with. And keep in mind that the machines tested here are some of the newest offerings. All four manufacturers involved in the test offer units in various price and feature categories so there's a lot bigger field than just these four units to check out. EQ

MAXIMUM MUSIC, MINIMUM EFFORT.

MIDI was supposed to make life simpler, remember? But until now, you needed a split personality to manage a MIDI sequencer and a tape recorder. Two different systems. Two sets of controls. Two different styles.

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amber 7000

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The 7000 System offers a high degree of configuration flexibility. The basic platform can support a variety of generation and measurement functions to allow the specific requirements of particular users to be satisfied in an efficient and easy to use form. A "soft" front panel avoids the usual compromise of panel understanding, clutter and flexibility. Data can be presented in a variety of user definable forms including any combination of numeric and graphic.

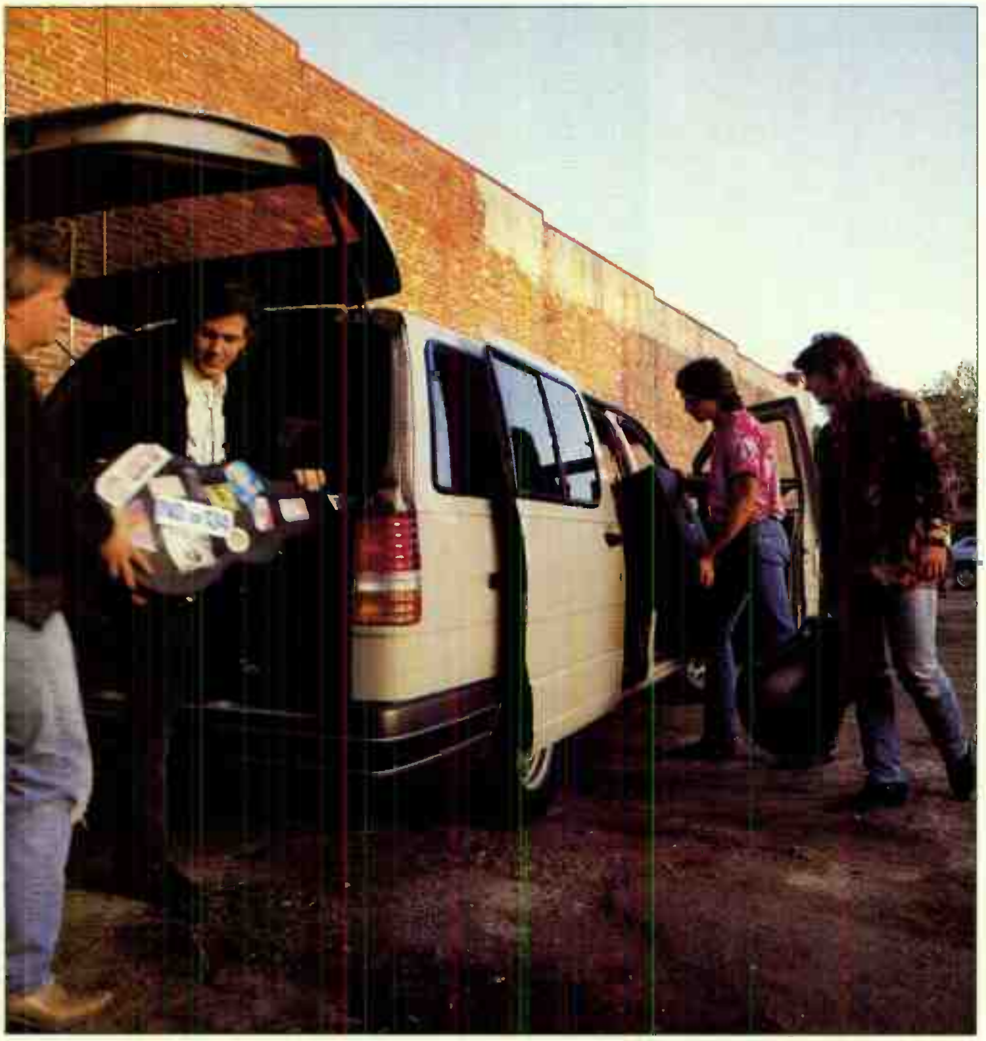
AMBER 7000 was designed to be simple flexible and precise yet state of the art, worth waiting for.

CIRCLE 10 ON FREE INFO CARD

BAND IN A VAN

A SPECIAL SUPPLEMENT TO EQ MAGAZINE

STAGING FRIGHT



BAND ON THE RUN: Buying your first PA can be a hair-raising experience, unless you plan ahead and make sure that you are getting exactly what you paid for (Photo courtesy of Artists Systems).

Photo courtesy of Artists Systems

L.A.'S MEAN STREETS AND THE SYSTEM FROM HELL

BY JIM PAUL

■ BUYING a first PA system can drive a normal person bonkers, not to mention what it can do to an already half-crazed musician. Believe me I know. I've bought and sold more than twenty sound systems over the last fourteen years. And it's true what they say, you know: you always remember your first time. . .

It was back in the late Seventies. I had a lot less knowledge then, and a lot more hair. It was a wonderful time of life. Only 21 years old and sure I was destined to be a rock star, I left my hometown of Albuquerque, NM behind, and headed for rockin' LA with nothing but the clothes on my back and my four guitars. I was betting that my notebook of original songs was filled with number one hits.

After the first three bands never got far enough to need a PA, I finally formed a trio which had the mettle to get our first job at a little

(continues...)

club in West Los Angeles. Suddenly, to my horror, we had only ten days to buy a PA system! Since I was the one always fooling around with electronic gizmos, I was naturally elected to do the shopping.

After much reflection, deep contemplation and a serious count of the band's available dollars, I decided that the way to go was with used equipment, bought from a private party. Maybe somewhere out there, was a little band from Pasadena which only used their system on Sundays at church and wanted to sell it cheap.

"WHERE'S THE AMPLIFIERS?" I DEMANDED TO KNOW. "THEY'RE INSIDE THE SPEAKERS!" HE SHOUTED. WHAT A CONCEPT!

L.A. STORY

Now came the truly fun part. You see, greater Los Angeles is a real big city, and I was still a little wet behind the ears. That week I drove what seemed like thousands of miles through mind-numbing traffic, torrential rain storms and eye-burning smog to look at systems and components that were too small, too big, too old or too ugly.

Days went by. Nervous time had come. Only four days before we opened at the club. I finally found one guy who had everything we needed in one spot and it all worked! I think it was a Biamp 8-channel mixer, a Dynaco power amp, an old NEI reverb and two JBL loaded custom speakers. It sounded like nirvana! Except for the fact that he wanted $1400 for it, everything was perfect. So I went back to the band members and explained the situation, but we really could not budge on our dollar limit which was $333.33 each.

Undaunted, I devised a plan sure to get the guy to sell me the system cheaper. No one can resist cold, hard cash, so I went back and actually counted out $1000 in fives, tens and twenties, right there in front of him. It felt like we were doing this big drug deal and the feds would break in at any moment. I said, "One thousand bucks, take it or leave it!" He stared at that huge pile of money for a long time and finally he looked at me and said, "Sorry man, I've gotta get at least $1250. I'll have to leave it."

Utterly crushed and almost in a panic, I picked up my money and got out of there fast. Two days later, still in depression and still systemless, I made a big mistake. I told my problem to a sympathetic friend, who told me he might be able to help. He had a cousin with a "killer" sound system for sale. Now it was two days till opening night and my back was up against the wall, so like a hungry puppy chasing a bone, I followed him over to his cousin's house. I still have nightmares about that day. In his garage, blasting out disco music, his cousin had an almost new 6-channel Tapco mixer with its own built in reverb, and a pair of huge, 2-way Altec speaker cabinets.

"Where's the amplifiers?", I demanded to know. "They're inside the speakers!" he shouted. What a concept! Inside each speaker, there was a 200 watt mono Altec amp! I was

continued on page 65

Hendrix fed on feedback; you might not.

HOW TO STARVE YOUR FEEDBACK

WITH A FEW PRECAUTIONS (AND TRICKS), YOU CAN AVOID GETTING THAT ANGRY FEEDBACK FROM YOUR AUDIENCE

BY BOB ROSS

■ THE AUDIENCE members at a concert or club may profess to having no knowledge of sound systems, but the instant a strident feedback squeal peals forth from the loudspeakers, it seems everyone's head turns to glare at the poor engineer behind the console. What's a guy to do?

I've listed here some fundamentals, basic procedures, techniques and equipment suggestions that should set you on your way to feedback-free mixing. Dodge these bullets at your own risk.

• Placement of the main speakers is of paramount concern. If the front-fills are behind the edge of the stage, they're probably delivering sound directly into your microphones; feedback will be, by definition, unavoidable. Make sure the main speakers are placed well forward of the downstage microphones. The farther from the mics, the better; increasing the distance between the speakers and microphones

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increases the available gain before feedback. (Remember the inverse square law? Each doubling of distance yields a decrease in sound level of 6 dB.) Become familiar with your speakers' dispersion pattern; if possible, orient them so that the nulls in their polar response point towards the microphones. Be sure the speakers are aimed directly at the audience, and not at reflective surfaces such as walls or ceilings.

• Keep the main speakers off the stage. Stages are little more than large flat slabs with mostly air underneath; i.e., a big resonating chamber. By decoupling the speakers from this resonant platform, you reduce the chance for feedback being transmitted mechanically from the speaker cabinets through stage floor and into the microphones.

• While you're at it, decouple the microphones from the stage as well. Commercially-available microphone shock-mounts don't always fare well in a live environment. They're fragile, expensive and awkward when used by hyperkinetic lead vocalists who like to yank the mic out of its clip and whirl it about their heads. A cheap, effective vibration-absorber can be fashioned from medium-density foam rubber attached to the base of the mic stands.

• The choice of microphone also plays a big part in the elimination of feedback. You'll definitely want directional mics for most stage applications. The narrower the pickup pattern, the more desirable. Several microphones, such as the Electro-Voice RE-15 (for instruments) or the Audio-Technica ATM-41 (for vocals) have a tighter pickup pattern, and hence are less susceptible to feedback than the popular Shure SM57s and SM58s. Whichever mics you use, keep them as close to the source as possible. (Again, that's the inverse square law.)

• Reduce the number of mics onstage. Each additional open mic channel is costing you 3 dB of acoustic gain. Do you really need eight mics on the drum kit? A minimalist miking philosophy has feedback-fighting advantages (besides endearing you to audio purists). Use DIs wherever possible — bass and keyboards being the usual candidates. Lately, guitarists are packing such sophisticated processing in their stage rigs that a direct line output providing "speaker cabinet emulation" may be available. Judicious use of gates can keep excess open mic channels under control, but gates aren't easily adapted to live vocalists, whose mics are, unfortunately, most likely the source of feedback.

• Use side-fills rather than floor wedges for the bulk of your monitor program. Those floor wedges are a high level loudspeaker only four feet away from an open mic; simple physics (that damn inverse square law again!) tells us that gain before feedback will be

continued on page 60

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IN A BAND VAN

Mixing MIDI Live

Live wires: Chick Corea mixing it up live onstage

■ THERE ARE TIMES when you are introduced to new technology in the worst possible way. For example, you're mixing a festival featuring nine music acts in three hours. The acts range from the headline Name Artist to amateur party bands. Only two bands show up for the afternoon sound check. About five acts into the show, The MIDI Duet arrives. The promoter had described their act to you as a lounge duo that sings. No problem, until mid-way through the previous group's set, when a patch list arrives at your mixer. The list includes two vocal mics, and twenty D.I. boxes!

The forty-piece balalaika orchestra leaves the stage and your stage crew quickly sets up the keyboard stands and racks of synthesizers. No time for a line check. The duo walks onstage, greets the audience and begins their first number with a press of the Start button on their MIDI sequencer. Immediately, every one of your mixing console inputs lights up, all the meters pin and somewhere in the middle of all this there are two people trying to sing. Well, at least their lips seem to be moving! You spend the next twenty minutes of their set trimming the mixing console input gains and bouncing around the solo buttons. One moment an input is a bass guitar, the next moment it's a french horn. Your contribution to the music is more like Damage Control than Art and there's still three more bands to mix before this show's over.

How not to get mixed up when mixing MIDI on stage

By Wade McGregor

DEFUSING THE NIGHTMARE

The ideal alternative to the above scenario would have you meeting with the performers well before the show to discuss their requirements. You would show up at the rehearsals, listen to what they're trying to do and discuss how you can help them. This is the time to establish a trusting relationship with the musicians. Take lots of notes.

Find out how the sounds are divided between the musical foreground and background parts. Do all the sounds come from one output jack on each synth or are there separate outputs? When you reach that acoustically difficult venue, will you have control over the snare drum or will boosting it also boost the horn section? Try to understand why the musician has balanced the synth sounds in a particular way. This will help you make the right decision when you must compensate for a strange sounding venue or sound system.

When mixing MIDI live, an audio submixer right at the synth/sampler rack can save using a lot of DI boxes onstage and input channels at the house mixing console. The need for different balances can be solved by creating two independent mixes. One mix, from the main output of the submixer for the instrument's loudspeakers onstage, should allow the faders to be within easy reach of the musician. The second mix can be created on a prefade auxiliary send of that submixer — to feed the house and monitor mixing consoles. This allows changes to be made during the soundcheck to compensate for the different acoustic conditions of each venue and the musician can still make independent adjustments to the onstage mix during the show. This can also allow different levels of reverb or echo effects to be added to each mix, as needed. Direct outputs, from the submixer inputs, can be used for specific instruments that will need to be highlighted or have special effects added in the house mix. The stage monitor system can provide the musician with a reference of the balance being sent to the house sound system.

BAND IN A VAN

MIDI MODS

Some multitimbral synthesizers may make use of MIDI mixing, that is, controlling the audio level of synths and samplers before it reaches their output jack. This means that MIDI volume gain structure must be considered. Decreasing the MIDI volume doesn't necessarily decrease the noise in the output of a sound module. MIDI volume must be set to get the best signal-to-noise and distortion in the same way that input attenuation is set on an audio mixing console (too little gain = noise; too much gain = distortion). It's important to establish an optimum gain structure in the MIDI sound modules and the onstage submixer so that the noise and distortion is minimized before it reaches the house mixer. Where this is still not quiet enough, MIDI-controlled noise gates can help by turning mixer channels on and off under the command of the synth or sequencer.

Once the gain structure and balance are set, mark them on the submixer for quick resetting. Print out a listing of all the important parameters for the sound modules, effect units, MIDI patching, sequencers, computers, etc., to make the pre-show check more reliable. Posting a copy of the patch within each rack will also save time when hunting down faulty cables or devices.

HOT TIP: DECREASING THE MIDI VOLUME DOESN'T NECESSARILY DECREASE THE NOISE IN THE OUTPUT OF A SOUND MODULE.

Creating a MIDI song file made up from brief sections of each important sequencer track and synth/sample voice used during the show can help to identify faults in the MIDI patching and presets, and in the audio submixes and sound reinforcement system. This MIDI file is played back on a sequencer while the systems are checked from the stage, monitor and house mix positions to confirm that everything's normal. This is the MIDI equivalent of playing your favorite CD to check the sound system.

The more that performers rely on racks of MIDI sound modules to create their instruments, the more you'll need to become involved in the process of premixing the show. Developing a good understanding of the MIDI generated sounds can now be just as important as your mic techniques.

(Thanks to Will Alexander, Robert Scovill, Mick Thompson and Scott Willing for their comments and suggestions.)

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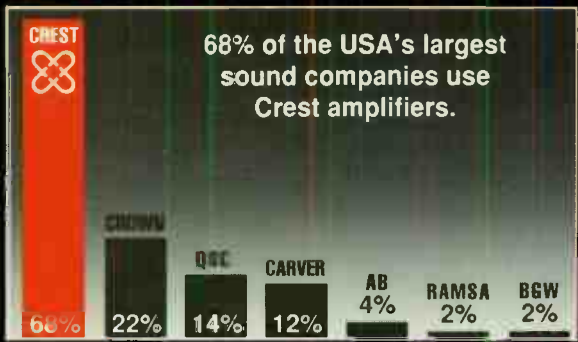
The 50 largest and most prominent touring sound companies in the USA are very serious when it comes to their power amps. Over the years, Crest has earned the number one position as the dominant amplifier supplier for these touring professionals. The results of an independent survey, conducted in July 1990, attest to that. When asked, "What is the main brand of power amplifier your company uses?", these serious businessmen cited Crest Audio by an overwhelming margin.



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ZAPPING FEEDBACK

continued from page 54

appreciably limited by this proximity. With side fill monitors (large full-range speakers twenty or thirty feet away from the mics), considerably more gain is available before the onset of feedback. The wedges can then be run at a much lower level for touching up individual performers' mix preferences.

• It's a luxury, but if you have separate monitor mixes available for each performer, you have the potential to reduce the number of open mics in any given monitor speaker.

• Inexperienced engineers have a tendency to reach for the graphic equalizers at the first sign of feedback. Unfortunately, this usually results in unsatisfactory sound out front, with huge portions of the frequency spectrum drastically attenuated. Feedback modes tend to be only a few Hz wide, whereas one third of an octave (the fixed bandwidth on a typical graphic EQ) can be over 1000 Hz wide in the feedback-prone parts of the spectrum. A parametric equalizer allows you to be more precise when controlling feedback: Create a deep notch, 1/3 to 1/2 octave wide, and sweep it until the feedback mode is found and attenuated. Then, gradually narrow the bandwidth of the notch as much as possible while still suppressing the feedback. You'll remove a significantly narrower portion of the sound spectrum. Whatever type of equalizers you use, be sure they're within reach from the mix location. Feedback actually occurs at a particular wavelength. When the club fills with hundreds of sweaty dancers and the air temperature increases, you may find feedback occurring at different frequencies than it did at soundcheck.

FOXY FEEDBACK FIXES

Here are a few tricks worth investigating in especially stressed-out situations:

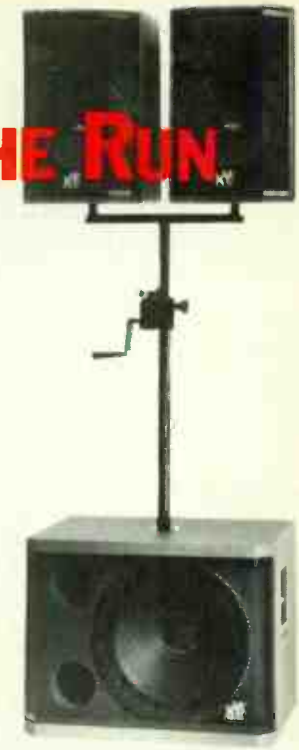
• Patch the entire program mix through a pitch shifter set for a small incremental shift, two or three cents. This effectively prevents the feedback mode from reinforcing itself, while being practically indiscernible.

• Patch the vocal subs through an Aphex Aural Exciter. This won't eliminate feedback, but it can provide increased intelligibility and perceived loudness before the onset of feedback. Be subtle with the amount of the effect; overused, it's pretty harsh.

• Reverse the polarity on one of the monitor speaker leads. (Sometimes this just works.)

• And, not just as a last resort: consider turning it down. Some day you'll thank me for this one.

SYSTEMS FOR BANDS ON THE RUN



Artist System's 4000 Speakers & 5000 Subwoofer

ADAMSON

F318 Loudspeaker System: 8 ohm 3-way passive enclosure; 18" low-frequency driver; 6" mid-range driver; 1" hi-frequency compression driver; 500W RMS continuous; crossover points of 400Hz and 3.5kHz; dim: 38" x 20" x 21".—NA

FR121 Loudspeaker System: 8 ohm two-way passive enclosure; 1-inch compression driver mounted on Acoustic Waveguide; 12" cone loudspeaker; 375W RMS continuous; 60Hz-20kHz freq. response; crossover point 1.5kHz; 63 lbs; dim: 15.5" x 26" x 17".—NA

ARTIST SYSTEMS

System 8000 Speakers: 1 x 1" compression driver; 1 x 10" horn mid-range; 2 x 15" woofers; low 800W; mid/high 300W; 38Hz-17kHz freq. response (±3dB).—$2500

System 4000 and 4500 Speakers: 12" cone driver; fluid-cooled high frequency driver; Uralite horn; 300W cont.; 1200W peak; 55Hz-18kHz (±3dB) freq. response; 128dB SPL max; 68 lbs; dim: 23.5" x 15W x 14.5"; internal stand fitting standard.—$1399

System 3000 & 3500 Speakers: Heavy-duty 10" cone driver; 1" horn driver; 300W cont.; 1200W peak; 65Hz-19kHz (±3dB) freq. response; 126 dB SPL; internal stand mounting.—$899

System 5000 Subwoofer: 18" fluid-cooled driver; 600W cont.; 2400W peak; 36Hz-120Hz freq. response; 88 lbs; dim: 22.5" x 30" x 21"; internal stand mount.—$1399

P-3000/P-4000 Processor: Provides the Electronic System Control for the Artist System speakers. Subwoofer level control; stereo/mono link switch XLR-type left and right inputs; LED signal and limiter indicators; positive amplifier return; Neutrik speaker connectors; Autosense circuit detects current to subwoofer and activates crossover.—$749

P-8000 Processor: Provides the Electronic System Control for the 8000 speaker. Subwoofer level control; stereo/mono link switch XLR-type left and right inputs; LED signal and limiter indicators; positive amplifier return; Neutrik speaker connectors.—$899

BOSE

Professional Acoustimass Loudspeaker System: 12" MB-12 electro-magnetic braking woofer; Six 4 1/2" wide range helical voice-coil drivers; 300W cont.; 200Hz to 18kHz crossover frequency; RIM structural foam; 80 lbs; dim: 16" x 22.25" x 23.25".—$3600

CELESTION

SR3 MEDIUM POWER SYSTEM

SR3 Loudspeaker: 8" concentric cone/dome driver; 16mm edge-wound voice-coil; 175W with SRC2 Controller; 60Hz-20kHz freq. response; 93dB sensitivity; 20 lbs; dim: 10.3" x 12.8" x 9.4".—$299

SRC2 Controller: Dual channel (stereo); 175W with SR3 speakers; switchable crossover at 200Hz; ±1dB freq. response from EQ curve; 7 lbs; dim: 1.75" x 19" x 6".—$330

SR4 Professional Bass Reinforcement Loudspeaker: Pole mount to accept compatible SR3; two 10" drivers; 250W cont.; 50-250Hz frequency response; 94dB sensitivity; Max SPL 118; 55 lbs; dim: 30.8" x 12.8" x 14.8".—$599

COMMUNITY

CSX57 Full Range Speaker: Three-way bass reflex; two 15" ferro-fluid drivers; 1" titanium diaphragm compression driver in the mid-range and the extended response of an ultra high-frequency expanded range transducer; Power Sense DDP protection; 40Hz-18kHz; 300W RMS; 90° x 50° dispersion; 115 lbs; dim: 33" x 26" x 18".—$747

Recommended amplification for CSX57: 500W per channel into 4 ohms.

ELECTRO-VOICE

COMPACT 2-WAY/SUBWOOFER SYSTEM

BK-842 Stereo Mixer: 8-channels; 3-band EQ per input; mid-sweep from 300Hz to 5kHz; 20Hz-20kHz freq. response; mid-sweep from 300Hz to 5kHz; stereo, main and monitor balanced outputs; phantom power.—$1248

7300A Stereo Power Amplifier: 250W/ch. @ 8 ohms; 20Hz-20kHz freq. response; dual/mono bridge mode switch; balanced/unbalanced inputs; Octal accessory sockets.—$1060

APX Stereo Crossover Module: 24dB/octave; 24 switch-selectable crossover frequencies on the 150 one-third octave centers from 50 to 10,000 Hz.—$150

S-152 Compact Two-way Speakers: 90° x 40° constant-directivity horn; 100 dB sensitivity; PRO™-circuit protection; EVG 15" woofer; 200W; 60-20,000Hz freq. response; 55 lbs; dim: 27" x 21" x 16".—$598

S-181 Subwoofer: DL18mt 18-inch woofer; 400W long term; 36-200Hz freq. response; 98dB sensitivity; 79 lbs; dim: 28" x 22.4" x 23.5"—$790

JBL MR Series Loudspeakers

JBL/SOUNDCRAFT

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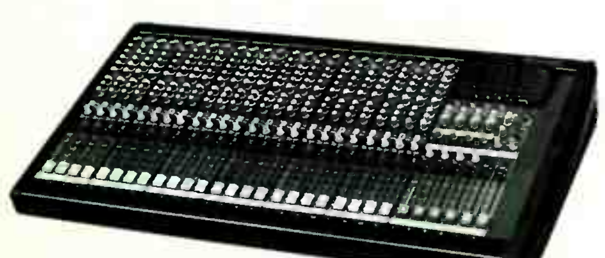
MR825 Loudspeaker: 100° x

80° Flat-Front Bi-Radial® horn; pure titanium compression driver; 15" low frequency transducer; 250W cont.; 1000W peak; 60Hz-20kHz freq. range; 58 lbs; dim: 26" x 19.5" x 13.75".—$595

MR818 Subwoofer: 18" low frequency transducer; 300W cont.; 1200W peak; 35Hz-4.5kHz freq. range; 93 lbs; dim: 40.75" x 28" x 18".—$675

Spirit/Live Mixer: 8x3, 16x3, 24x3 configurations; variable gain control and line input; 3-band EQ; 4 AUX sends; 4 stereo effects returns; mono buss; 20Hz-20kHz freq. response.—$1295, $2395, $3495



RAMSA WR-S4424

M552 Electronic Crossover: 2-way stereo/3-way mono; servo-balanced outputs; 117dB dynamic range; THD + noise less than .004%; 10Hz to 75kHz freq. response; subsonic and RF filtering; XLR input and output connectors.—$430

PANASONIC/RAMSA

WP-1000 Series Power Amplifiers: Class H circuitry (runs cooler and uses less AC power); servo-controlled cooling fans; balanced XLR and TRS phone jack inputs; stereo or mono bridge operation; 5-way binding post.—$620-$780

WR-S4400 Series Mixers: 12-, 16- and 24-channel configurations; 3-band EQ with sweepable mid-range; 4 AUX sends with AUX/group multiplier; 48-volt phantom power.—$1995-$3195

WS-A200 12" Two-way Speakers: Twin-Bessel horn/compression driver; dispersion 60°H x 40°V; 250W program power capacity; 70Hz-20,000Hz frequency range; 98dB sensitivity; 35 lbs.; available in black (WS-A200-K) and white (WS-A200-W)—$590

WS-A500 Series Speakers: Mid-high speaker; 2-way; 200W; 98dB sensitivity. WS-A550 Series speakers: Subwoofer speaker; 400 watts; 91dB sensitivity; combined frequency range 30Hz-20kHz; requires WS-SP2A crossover.— $1000 and $650, respectively

PEAVEY

SP-2Ti Speakers (2): 15" Black Widow woofers; 22T high frequency compression driver; two-way vented enclosure; patch capability; 300W cont.; 60Hz-17kHz freq. response; 93 lbs; dim: 23.75" x 31" x 17".—$499.99 ea.

XR 600C Powered Mixer: 300W RMS; 9-band graphic EQ; DDT compression; aux input; tape output and input; built-in reverb.—$599.99

Addverb II Multi-effects Processor: MIDI In and Out; 50 reverb presets; 40 programmable delay/echo presets.—$319.99

TOA

SL-120, SL-150 Compact Integrated Two-Way Speaker Systems: 12" and 15" woofer, respectively; 90° X 40° constant-directivity horn/piezo driver; 240W RMS cont.; 70Hz-20kHz frequency response; built-in mounting adapter. —$308, $371, respectively

MX-601 Powered Mixer: 6-channels; 10-band graphic equalizer; 300W RMS into 2 ohms; patch bay with buss link; tape input and AUX input w/pan & level controls; spring reverberation unit.—$971

YAMAHA

MID-LEVEL SYSTEM

MC1202 Mixing Console: 12 mic/line inputs; balanced XLR; 3 aux sends; 2 stereo returns; channel patching; cue; 3 band-mid sweep.—$1095

SPX900 Programmable 16-Bit Digital Effects Processor: 20-20kHz freq. response; simultaneous processing of 5 effects; dual FX processing; two-band digital parametric EQ; 50 preset effect programs; 40 memory locations; remote control available.—$895

Q2031A 1/3 Octave Graphic EQ: Dual channel 31 band 1/3-octave equalizer with 6 or 12dB of cut or boost; 20-200Hz variable high pass filter per channel; input transformer sockets; balanced XLR and 1/4 phone jacks.—$695

PN90 Low-Level Crossover: 90Hz passive network; 1/4" phone jacks; designed to operate with amplifiers that have an input impedance of 15,000 ohms; metal pan chassis with flanges for permanent mounting.—$150

P2700 Dual-Channel Power Amplifier: 350W @ 8 ohms/500W @ 4 ohms; 20Hz-20kHz freq. response; stereo and mono configuration; XLR and phone inputs; 53 lbs; dim: 7" x 19" x 18".—$995

P2350 Dual-Channel Power Amplifier: 175W @ 8 ohms/250W @ 4 ohms; 20Hz-20kHz freq. response; stereo and mono configurations; XLR and phono connectors; 42 lbs; dim: 5.2" x 19" x 17".—$795

S115HII Club Series Two-Way Speaker: 15" woofer; high frequency horn and compression driver; 200/400W peak; 45Hz-16kHz; freq. range; 122dB max SPL; dim: 20" x 29" x 13.4".—$325/each

SW215 Club-Sub Dual 15" Sub-woofers (2): Two 15" diameter low-frequency drivers; 200/400W RMS; 97.5dB sensitivity; 40Hz-2kHz freq. range; dim: 34" x 20" x 17".—$395/each

YORKVILLE

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Elite M-600 Two-Way Speaker System: 1" compressor driver on 100° x 60° horn; 600W; 50Hz-16kHz freq. response 2.5kHz crossover frequency; SPL 130dB; 77 lbs; dim: 19.5" x 23.2" x 13.4".—$989

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No Peaking!


Generation gap: When is too much?

■ It's easy to hear your speakers suffer when you've cranked up your amplifier past the point of no return. The distorted sounds cut deep to the bone and send you cringing to lower the volume or switch the unit off — anything to save your equipment before it's too late.

What may be surprising is that playing it too safe, using an amplifier whose power output is lower than the speaker's handling capabilities, can also cause serious amounts of damage to both the loudspeakers and your music. Because of the properties inherent in music, this well-meaning intention often does more harm than good.

SOUNDMEN KNOW THAT TOO MUCH POWER CAN HARM THEIR SPEAKERS, BUT FEW REALIZE THAT TOO LITTLE COULD EVEN BE WORSE

LESS POWER TO THE PEOPLE

The power usage between bass, midrange and tweeter vary by large degrees. The lower registers of music require much more power than the higher. The energy level of the tweeter is typically 10 to 20 dB less than the bass and midrange frequencies. To compensate, most high frequency drivers allow for 10 dB peaks — which is still only one-tenth of the power that the bass and midrange components must endure.

Speaker designers are well-acquainted with this musical property, and generally devote ten percent of a speakers' total power to high frequency components, meaning that a 100-watt speaker will devote 10 watts of power to the tweeter. Some manufacturers offer a higher percentage as a safety precaution.

The problem really stems from the amplifier. Remember that the power output given on most amplifiers is rated with a reference to total harmonic distortion (THD). This means that the amplifier can produce more power than listed, but at these levels you run a high risk of distortion. Sure, you can get your 35-watt amplifier to pump out 70 watts without damaging it, but you probably won't want to hear the way the signal reaches the speakers, even if your speakers can handle a larger power output. Even without cranking the dBs, an amplifier that has too small a power output will be able to produce the lower frequencies without a problem, but it will not be able to handle the high frequency peaks, which will cause distortion — and perhaps damage the speakers.

As mentioned, the distortion from an overextended amplifier hits particularly hard in the higher frequency components. This is because distortion, in this case, is really the multiplication of harmonics [see Len Feldman's column on THD in this issue]. The harmonics themselves are higher frequency multiples of the original signal, so that's the component which bears the brunt.

THINKING BIG

If you choose an amplifier with a higher output, however, you can avoid this problem. On top of getting more power, the extra watts will be able to handle the high frequency's peaking without distorting the signal to the loudspeaker.

SOUND ADVICE: THE DISTORTION FROM AN OVEREXTENDED AMPLIFIER HITS PARTICULARLY HARD IN THE HIGHER FREQUENCY COMPONENTS.

Other ways you can protect your speaker from your amplifier include not disconnecting any component while the amplifier is on. Most of us have done this and we are all familiar with the LOUD, uncontrollable hum. Also, turn down the volume when fast forwarding, rewinding and, if you still own one, moving the stylus of a turntable.

Of course, one of the best ways to avoid damage is to always keep the volume under control. Remember that the volume knob is not the only way to increase decibels. Many of the tone controls are able to provide boosts of up to 15 dB.

There are the caveats, but the best way to handle the amplifier/speaker relationship is with two words — more power. If you follow this philosophy, however, you should make sure that the venue is rooted on steady ground before you turn it up.

[This article was adapted from the JBL Professional Technical Note "Danger: Low Power."]

PA System From Hell

continued from page 52

enthralled with the idea. No speaker cables, no muss, no fuss, and besides they seemed to sound okay. He only wanted $900 for the whole thing, and let's face it, my options were pretty thin. Then I made my second big mistake. I never really tested the system, checked out how he was driving it (direct from a tape deck), or tried to pick up the speakers. Instead, like a fool I counted out the cash, grabbed the mixer and told him I'd be back tomorrow to pick up the speakers.

Coming back the next day with a borrowed truck, we tried to move the speakers, and it began to dawn on me that maybe I had made a mistake. These speakers were meant to be installed somewhere and left there. They were made of 3/4-inch plywood, stood about 4 feet high by 3 feet across and were about 3.5 feet deep. The combined weight of the wood, the drivers and the power amp was over 100 pounds! They would not both fit in the bed of the truck so it took two trips and every bit of energy we had to move the monsters. For some reason, they looked a lot bigger in my living room than they did in his garage. Needless to say, the other band members were a little surprized, but at last we had a system!

Now I don't mean any disrespect to the folks who designed the Tapco or the Altecs, but the sound quality was pretty bad. It was plenty loud, but kind of muddy and metallic sounding. And the built in reverb sounded like someone singing into a metal trash can! But it was our trash can, so off we went to the club with two trucks of equipment and our awesome sound system.

SHOWS MUST GO ON AND ON

Opening night was a sonic disaster and after the show, the club owner came up and the first thing he asked was "Do you think you could leave one of the big speakers home next time?" I was so ashamed and more than a little embarrassed, but we took one giant box home that night and used the other one the rest of the time there, about five weeks.

To avoid further humiliation, I think I'll draw the curtain on this little episode in my life. Even now, nostalgic tears well up as I remember it along with other pleasant "First Time" memories, like my first abscessed tooth or my very first hemorrhoid.

Maybe there is something to be learned from all this. Perhaps some simple lessons like, "become informed before you buy any product." Or how about "give yourself enough time to complete a project so you won't be rushed like I was." And there's always "check out used equipment very thoroughly before you buy it."

Oh yeah, and remember this one; remember it well: "never buy used sound equipment from someone's cousin!"

DIGITAL IN/OUT

continued from page 35

ment an I/O that has been evaluated in a variety of situations, and for which a wealth of hands-on experience now exists.

And refrain from intermixing consumer-grade I/Os with professional formats. Not only are such interconnects "dubious," to say the least, but information is also carried within their respective bitstreams that is designed to be noncompatible. Apart from incompatible Channel Status information carried by both IEC-958 and AES3 ports, synchronizing transmitters and receivers via these I/Os is often achieved differently. The bottom line is simple: AES/EBU and IEC 958 "consumer-use" interfaces should be considered totally different I/Os, any electrical compatibility being purely incidental.

How's this for an example? A while ago, I came across a group of users experiencing problems while transferring two-channel mixes from a well-known workstation into a DAT recorder equipped with both IEC 958 Type II-format "consumer-use" and XLR-equipped AES3-format digital I/Os. Responding to advice that AES connections are usually more reliable, several studio owners were dismayed to discover that the connection either failed to work at all, or was highly intermittent. Sometimes, the DAT would enter record-ready mode — indicating the presence of a digital bitstream — but would then drop into stop mode after a few seconds. Conversely, the IEC-958 input proved 100 percent reliable.

Rather than blaming a faulty AES input circuit, the solution turned out to be rather interesting. The DAW in question is equipped with a programmable digital I/O that generates a variety of different I/O formats. The AES setting produced a totally correct bitstream, apart from two important Channel Status data bits. Instead of setting the Sample Rate flag (bits 6 and 7) to correspond to 48.0, 44.1 or 32.0 kHz, the DAW's I/O simply expects the DAT recorder to determine such information by automatically clocking the subframe sync pulses.

This configuration worked reliably with early-generation recorders. But along comes a new-generation DAT machine that correctly uses Channel Status bits 6 and 7 to set its internal timing clocks. For sampling rates of 48.0 kHz (bits 6/7 = "00"), the interface might be expected to function correctly, because the DAW also set undefined bits to zero; at a sampling rate of 44.1 kHz, however (bits 6/7 = "10"), the DAT was unable to match sync rates and wouldn't enter record mode. An I/O port firmware upgrade that generates the correct Channel Status flags for all sampling rates instantly cured the problem.

FINAL WORDS OF ADVICE:

- Attempt to use nothing but correctly implemented AES/EBU I/Os.
- Determine from the equipment manufacturer which Channel Status bits are included and, for newer AES3-1992x-compatible I/Os, what Level of Implementation is offered
- Don't assume that any particular manufacturer is better than another at correctly implementing AES-format I/Os. To perform quantitative assessments of a suspect interface, you'll need comprehensive test equipment for analyzing the timing accuracy of various sync pulses, for example, as well as Channel Status data. EQ

For seven years Mel Lambert served as editor of REP. He is now principal of Media&Marketing, a consulting service for the professional audio industry.

Now It's a Sync . . .

Most MIDI-philes know about synchronizing a sequencer to a sync track (such as SMPTE time code) recorded on tape. Now there's a new twist: you can have the sequencer use that sync data to control tape functions like punch in and out, track enable, input monitoring, tape transport shuttling, and more. What makes this possible is a collaboration among Fostex, Atari, and three software companies: C-Lab, Dr.T's, and Steinberg-Jones.

The software runs on the computer, and sends commands to the tape recorder via MIDI, which has the "hooks" to respond to these commands. The goal is to treat the recorder as another MIDI peripheral which, once the tape is threaded, doesn't need attention — everything's done from the sequencer/computer.

This article is the first anywhere that focuses on using the Fostex R8 eight track recorder and the MTC-1 synchronization unit with C-Lab's Notator software and Unitor SMPTE synchronizer/multiple MIDI port, running on the Atari ST. While the details of connection and exact operations will differ from system to system (especially if your system uses MIDI Time Code or DTL), the overall experience will be similar to what's described below. Oh yeah, before I forget, as always, read the manuals.

SETUP

Here's how you begin: Connect the MTC-1 to the R8 according to its instructions, but connect the LTC ports and MIDI ports on the MTC-1 as explained in the Notator manual chapter on Synchronization.

A deck's outside track (highest or lowest numbered) is normally used for time code, so connect the Unitor, then, stripe the tape with 30 frame non-drop SMPTE. Before Notator can control the R8, you'll need to set up a sync reference in the Synchronization window.

Set all the DIP switches on the MTC-1 to "0" except no. 7. Set no. 7 to "1." Be sure to set the "Clock Port" on Notator's MIDI Thru window to reflect the MIDI output you connected to the MIDI input of the MTC-1. If possible, use a Notator output you won't use for anything else. Put Notator in SMPTE sync mode. Then press Shift-F on the ST keyboard. Click "On" to enter Fostex mode. (To leave Fostex mode, click "Off.")

One final step: put the R8 in Play for a few seconds on the time code track (be sure you see its level on the meter) before you try to control it from Notator. That tells the MTC-1 where the R8 is on the tape.

ROUTINE OPERATION

The Notator manual chaptor on Synchronization is very clear on how to control the R8's various functions from the sequencer. Consider preroll; it's set on Notator's Count-in window. Depending on the tempo, the process for using Autodrop/cycle mode involves Punch enable and Play, not Record. If you need to practice first, enable Punch and right/click Record, or press R. This will switch the monitor in/out at the punch points, without recording anything. After practicing this way, it seems to be necessary to disable Punch, play the tape for a few seconds, and Stop. Then re-enable Punch and actually do the punch in/out by pressing Play. Preroll is set on Notator's Count-in window. Depending on the tempo, you may need to add a beat or two to the Count-in you'd actually like to hear, because of startup delay.

If you want to have a Count-in (pre-roll) at the beginning (bar 1) of a track, you'll need to set the "Song Start" on the Synchronization window to a point before bar 1. For example, if you want a two bar Count-in, try set-

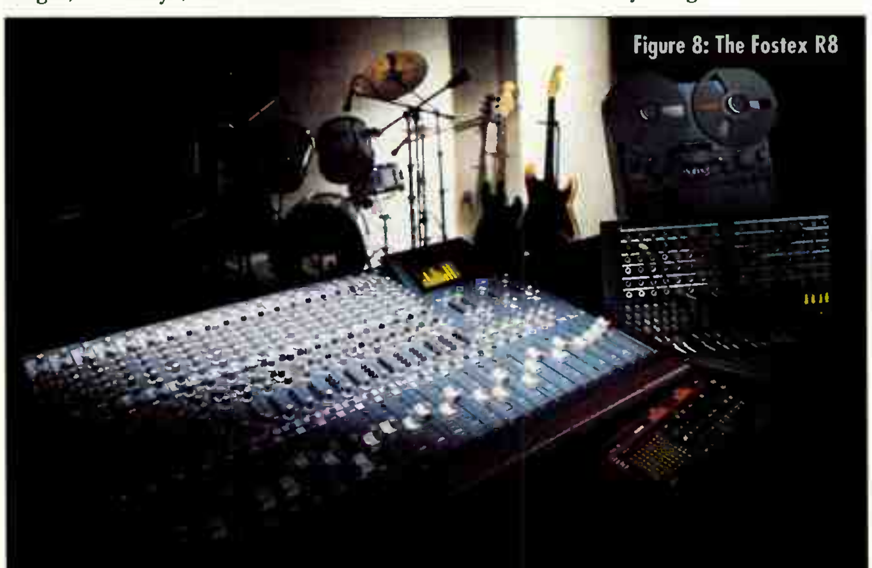
Figure 8: The Fostex R8

. . . To synchronize the tape recorder from the sequencer, using the Fostex R8

BY PHIL SHACKLETON

ting the Count-in to 2 1 0 0 (the extra beat is for the startup delay), and set Song Start to -2 1 1 1. If you're syncing to material already recorded, you'll probably also need to move the "SMPTE Offset" time to a point three bars before your previous SMPTE Offset. (Let's count: bar 0, bar -1, bar -2; that's three bars, okay?) The Fit Time calculator can help you figure this out. Because of Notator's MIDI click, you don't need an R8 click track.

For those times when you want to do sequencer-intensive operations and don't need to run the tape, you can quickly toggle back and forth between SMPTE and internal sync.

Some commands take a moment to happen, and may cause the R8 display to flicker a moment, especially switching the input monitor in/out using Shift-Record. Although automated punching is probably one of the most important justifications for computer-based systems, you may sometimes want to drop in a part at the spur of the moment without having to set punch points. You can do this by simply selecting Record while in Play.

If you're already in Play/Record, you can punch in/out on a given track by just record enabling/disabling it. The R8 control panel is still active, if you need it. In this connection avoid sending a MIDI Local Off message to the R8, because that will turn off the R8 panel controls.

OTHER TECHNIQUES

You can customize the control functions even further by using note-off and system exclusive (Sys Ex) messages. For either method, Notator's faders and buttons on the RMG page can be assigned to control almost any R8 function as a Puser event. Realtime Transform makes it possible to convert almost any income MIDI data into the Puser events that can drive the R8.

Controlling recorder functions via note commands lets you control your deck from a keyboard, drum pad, etc. This is particularly handy if your master keyboard is located out of arm's reach of the computer. If you elect to use MIDI notes, set DIP switch no. 6 to "1." The MTC-1 manual gives MIDI note equivalents for most R8 operations, including individual track record enabling, punching in/out, etc. If the MTC-1 must be part of a MIDI

Controlling the MTC-1 in this way makes Notator into the most powerfully programmable remote ever.

Thru chain, then set DIP switch no. 5 to "1." The MTC-1 manual covers this in more detail, but if you set DIPs 1 to 4 all to "1," the MTC-1 will respond only to certain MIDI note messages on MIDI channel 16, as well as Sys Ex messages for "device number" 16. (It will still be controllable via Sys Ex by Notator in Fostex mode, because Notator uses the "universal device number" 128, and so is unaffected by DIPs 1 to 6. Other softwares may be affected by DIPs 1 to 6.) Avoid sending Omni on/off messages to the MTC-1, as these can change the settings of DIPs 1-5. DIPs 7 and 8 are used to select other LTC frame rates, as explained in the MTC-1 manual.

While for the most part only note-offs are responded to by the MTC-1, it works perfectly well to send notes to it from your keyboard, sequencer or both. Pay particular attention to the concept of "shifting," where you press and hold one note, then press and release another note, then release the first note. Many of the more powerful commands are "shifted" in this way.

Pay particular attention to the "shift" sequence that enables MIDI note remote control. If the right mode isn't selected, none of the note control options above will function.

Controlling the MTC-1 in this way makes Notator into just about the most powerfully programmable remote control any tape recorder ever had. You can program multiple punch ins and outs on multiple or individual tracks, construct complex loop behaviors, and so on.

All of these capabilities (and more) can be accessed via Sys Ex messages by advanced (and determined) users, based on the sys ex code documentation included with the package. Note that the MTC-1 can also be updated to add MIDI Machine Control, so you can use that set of commands as well (the 8330 card for the G-series Fostex recorders is also slated to support MMC). One hint: to see what Sys Ex data Notator is actually sending to the MTC-1, connect the MIDI Thru of the R8 back to a MIDI input on Notator. (*Disable MIDI Thru!*) On an empty Notator track, press the space bar, and then press Play, record enable an R8 track, press Record, etc., then Stop. Notator will record all these messages into the track, allowing you to view them, compare them to the Fostex manual, etc. You can also use them as source material to enter into Puser events on the RMG screen. Remember to use Puser switches for the on/off type of events, and set the maximum/minimum values for VAL to be what you see as the variable number on the Edit screen, for the Sys Ex messages you recorded. For example, there appears to be only one byte of difference between Play and Stop, so one Puser button switch can do both! EQ

Phil Shackleton is a technical writer as well as a lecturer at Azusa Pacific Univ., Azusa, CA.

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Boom-Boom Rooms

A few facts and fallacies about your (and everyone else's) bottom end

BY FRANCIS DANIEL

There are a number of misunderstandings about room acoustics. They resurface regularly, are promulgated by the current crop of gurus, and lead to the inevitable wasted money or unsatisfactory results.

I would like to shed some light on one of the more enduring confusions that I have been hearing for over three decades — and I am sure it was around before my time. It's particularly relevant to the small project studio environment. And it takes the form of a statement something like, "You can't get bass in a small room."

The current pundits making this assertion will produce some formulas showing what the supposed lower limit of bass response is going to be for a given maximum room dimension. Or they will repeat the conventional wisdom that it's necessary to have "a room with a physical, long dimension of half the wavelength of the lowest frequency your monitor can reproduce," with the implication that you cannot get (good? any?) bass response below this limit.

Unfortunately, the gurus sometimes seem to be out of touch with reality, especially what might be called technical reality, or engineering as we sometimes call it. Fortunately, the situation is far more optimistic than these predictions would lead you to believe. So before you go looking for more real estate, read on and take cheer — you can get good bass in smaller rooms, and in fact you are already doing it in some cases, as we will now discuss.

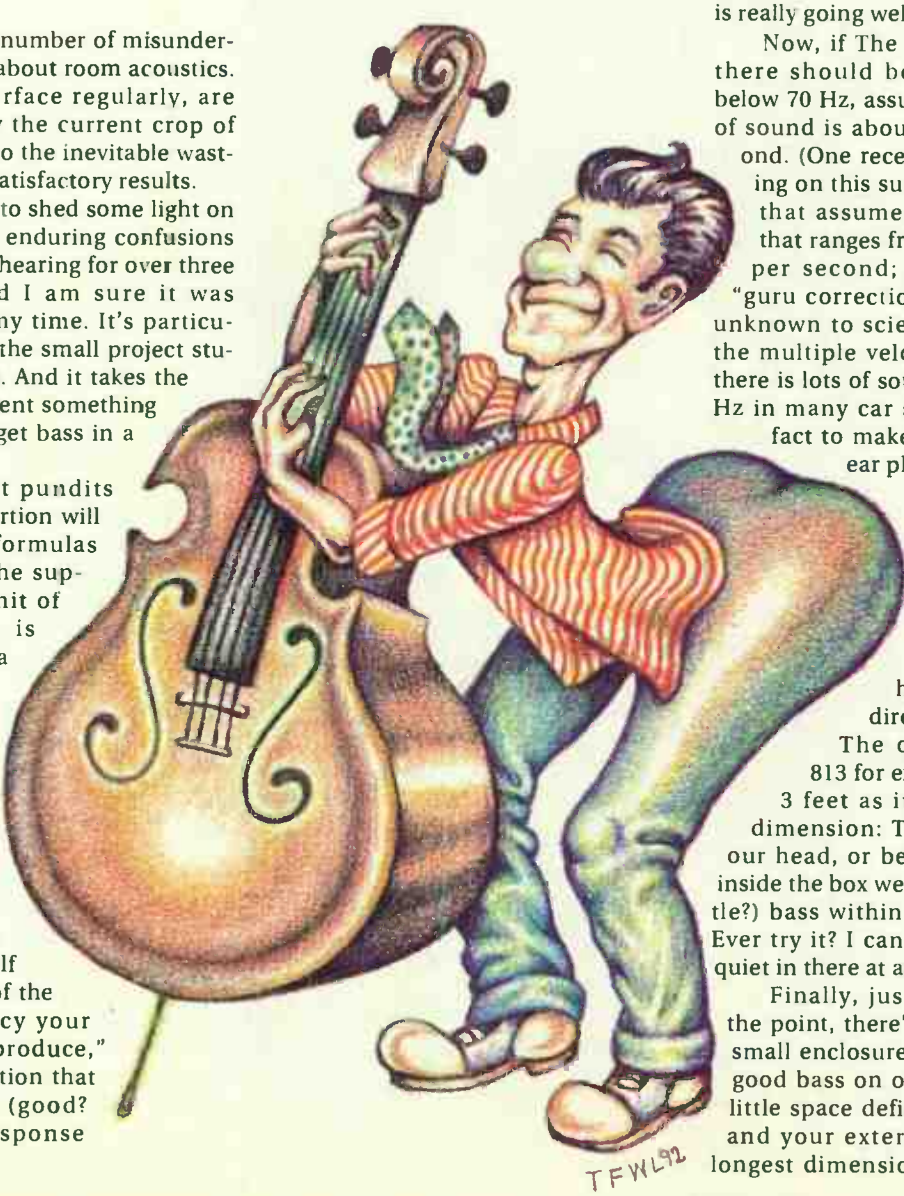

Illustration by Normal Rockhard

LET'S GET SMALL

There's one type of "small room" that nearly all of us are quite familiar with. It has a sound system in it, frequently with very powerful low frequency response. It goes past my window, and yours, every day. It's called an automobile. My rattling glass panes will testify to the bass response of these "rooms," as will your ears if you are unlucky enough to be caught in one of them. The longest dimension of most of them is 6 or 7 feet; let's call it 8 feet to include your favorite vintage gas guzzler, or limousine, if business is really going well.

Now, if The Guru's Rule is true, there should be (no? poor?) bass below 70 Hz, assuming that the speed of sound is about 1130 feet per second. (One recent article pronouncing on this subject included tables that assumed a speed of sound that ranges from 1280 to 1440 feet per second; perhaps there's a "guru correction factor" that's yet unknown to science accounting for the multiple velocities!) Manifestly, there is lots of sound output below 70 Hz in many car systems. Enough in fact to make you reach for your ear plugs.

Next, let us consider another, even smaller, enclosure that by definition has a woofer in it — the loudspeaker enclosure that houses most of your direct radiator systems. The omnipresent UREI 813 for example has less than 3 feet as its longest internal dimension: Therefore, if we put our head, or better a microphone, inside the box we should hear (no? little?) bass within this small "room." Ever try it? I can assure you it's not quiet in there at any frequency.

Finally, just to hammer home the point, there's another extremely small enclosure that produces very good bass on occasions; that's the little space defined by headphones and your external ear canal. The longest dimension here is 2 inches.

This should, according to the conventional wisdom, lead to the conclusion that we cannot hear (any? much? good?) bass below 3400 Hz from headphones!

Okay, these are enough common sense examples that we all know about. They should give the lie to this very common fallacy that you cannot have extended low frequency response within a small space. No advanced technical knowledge required — you have all heard at least two of these examples yourself.

Is that it? Room size has no effect on low frequencies? Not at all. But reality is a little more complex, as we will explain.

GETTING TO THE BOTTOM OF IT ALL

What can be correctly predicted by the half wavelength formula (especially if you use the correct speed of sound) is the frequency of the lowest room mode, or resonance, for a rectangular room. Rooms, all of them, whatever their shape, will have these natural resonances, dependent upon their dimensions and shape. It's a function of enclosing air with reflecting surfaces, no more, no less. Below this limit, a loudspeaker will simply "pressurize the room," rather than excite its resonant modes. But it won't suddenly fall silent. It will simply have to survive without the "modal support" that most rooms provide, however unevenly.

Nailing another popular fallacy: Creating irregular shapes, for rooms or loudspeaker enclosures, doesn't get rid of these modes, or even markedly change the number of them. All it does is move around (unpredictably, unless you go to a great deal of very complex computer analysis) the frequencies or locations at which the modes occur. They're still there, except we'll have a harder time predicting how they'll be distributed along the frequency spectrum and throughout the space.

Notice that I didn't say, in the last sentence above, "along the low frequency spectrum." That's because these room modes occur from the lower limit set more or less by the longest room dimensions, right on up to ultrasonic bat territory, if you have a loudspeaker that can generate them.

However, what happens is that there are more of them as the frequency goes up. The wavelength gets shorter and shorter and hence there are more "wavelength multiples" within each room dimension. As they say in the high brow journals, the modal density increases as the frequency increases. At low frequencies there are fewer modes with larger gaps between them.

RUNNING OUT OF ROOM?

This leads to what we all have experienced: If you slowly sweep a sine wave through the low frequency part of the spectrum, even a perfect loudspeaker has a ragged seeming response, because the room modes reinforce at some frequencies and cancel at others. And this is the crux of the issue about room dimensions and bass response. Most loudspeaker designers are assuming, correctly, that these modes will be there to boost the low end output of their products, even if the boost is irregular. There's nothing wrong with this — rooms generally do have modes. Yes, there are ways around this, but that's another, lengthy matter, and doesn't lead to any immediately easy solutions.

However, if we have a loudspeaker that can produce frequencies below the lowest mode of the room it's in, we'll certainly hear them. The problem for the loudspeaker is that it's now operating in what is called "the pressure zone" and doesn't get any (erratic) help from the room resonances to support/amplify/distort its output. This puts a big load on the woofer, which is having to work increasingly harder anyway as the frequency drops. Many loudspeakers cannot keep up with the demands, and this leads to the erroneous conclusion that the room is the problem.

Most studio loudspeakers are not designed to produce anything below 40 or 50 Hz anyway, wherever they're placed, no matter what the ads say (I can visualize the angry letters coming in now). Using the half wavelength formula, a rectangular room with only a 14-foot length would produce a reinforcing mode at 40 Hz and make most listeners hear "great bass."

In fact, the smoothest bass response will be found below the lowest room resonance, in the pressure zone, if the loudspeaker can hack it. The amplitude will be controlled only by the woofer output and the room absorption, which can be, but is not always controlled.

HOME BASS

To summarize: The size of the room/enclosure/whatever only determines where the lowest resonance will occur. The room resonances will sporadically reinforce (and equally sporadically cancel) the bass output above this frequency. Below this frequency, in the pressure zone, the woofer is largely on its own and often falls short, as would be clear in an anechoic/outdoor environment.

There's a great deal more to room design at low frequencies than simply ignoring its dimensions, but you don't have to be automatically deprived of good bass in a project studio — not if your loudspeaker can survive the pressure zone! EQ

Francis Daniel is a Senior Associate in Architectural Acoustics at Shen, Milsom & Wilke, NYC.

Buzz Words

A quick tutorial on power distribution, grounding, ground loops and the advantages of balanced over unbalanced

BY EDDIE CILETTI

Illustration by Scot Halpin

Did you ever get shocked or hear of someone getting shocked while playing guitar and singing into a microphone? Everyone knows that the reason for the shock is the presence of "voltage." In electronic circles, the word "potential" is used to describe the difference between two points and is measured in volts. These "two points" could be in a power supply or between two pieces of equipment, or anywhere. When a life form comes across enough potential, the end result could be death, or just a shock. But what effect do lesser amounts of "potential" create? The answer is annoyance — by creating HUM and BUZZ. You know, those subtle differences between a clean, quiet mix and one that doesn't rock the house.

SCENE: Two audio devices are connected to power. For whatever reason, their chassis are not at the same potential. What happens when the audio cables are connected?

ANSWER: In addition to carrying the signal, audio wiring also includes a shield to keep the signal free of radiated interference. (Power supplies and their AC power cables can radiate hum and buzz.) When the shield connects two devices not at the same potential, current flows through the shield. There's now no potential difference between the two devices. The shield and internal wiring "absorb" the difference and this seriously degrades both the shield's effectiveness and the device's performance. The audible end result is hum and buzz. Witness the birth of a ground loop.

Consider how most equipment is mounted in a rack. I have solved major hum problems simply by removing one piece from the rack. Why? And how do we get that piece back into the rack without disturbing the others? Read on, if you dare.

THE SIMPLE SOLUTION

If all gear had at least balanced inputs and, if possible, balanced outputs, audio wiring would be nearly effortless. The balanced system rejects most electronically radiated noise. "Balanced" electronics uses two wires to carry signal plus a third wire for a shield. The shield is connected to the electronic ground, which may or may not be connected to the case, or "chassis." "Unbalanced" electronics require only two wires: one for signal and the other doubling as both signal return and shield. The unbalanced system doesn't reject induced noises. Balanced wiring has an additional advantage over unbalanced when attempting to overcome ground loops. With balanced wiring, the shield can be "lifted" (disconnected) at one end. This eliminates current-in-the-shield (the ground loop) without decreasing the shield's effectiveness.

GETTING STARTED

Find a power outlet that has its own circuit breaker or fuse and dedicate it to your equipment. If possible, try not to use any other outlets on that circuit, or do so only with additional sound related equipment. This especially applies to lights with dimmers and air conditioners! As shown in Figure 1, a line conditioner creates four outlets from the wall. Fan out additional power strips/line conditioners from here. Dedicate each one of these for a particular group of devices. (See examples one and two below for equipment grouping suggestions.) This will ensure that each group has the same power phase and ground. For most personal production systems, the power requirements are below 15 amps. If your outlet can't deliver 15 amps, or if 15 amps isn't enough, it may be necessary to find an additional outlet on the same phase. Don't daisy-chain power boxes! (When setting up, it's a good idea to inspect all outlets, whether in the wall or on power strips. Make sure all of the mounting screws are tight. The

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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2860 Cordelia Road, Suite 120
Fairfield, CA 94585
Tel: (707) 427-1920
Fax: (707) 427-2023

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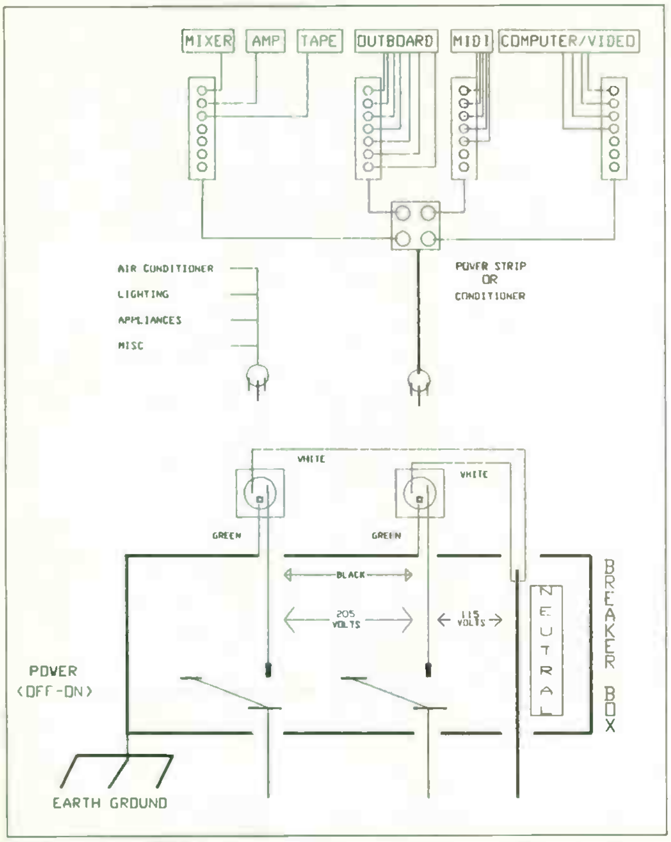


FIGURE 1 **Power Distribution: Often, a good arrangement of equipment and power connections in the studio can quell a good many of those pesky noisemakers.**

screws on the outlet make the connection to ground.) See Figure 1 at left.

EQUIPMENT GROUPING SUGGESTIONS

Example 1: 1) mixer and power amp; 2) outboard gear; 3) tape machines; 4) MIDI gear. *Example 2*: 1) mixer, tape machines, power amp; 2) outboard gear; 3) computer, video monitor, video deck; 4) MIDI gear.

A more sophisticated system requires that the wiring be in the wall, making it less easy to visualize. Hopefully, you see the concept.

Try to establish the same power and ground reference for all of the equipment. Remember to keep the line conditioner and all power wiring separated from the audio wiring.

When trying to track down buzzes and intermittent crackling noises, the source may be: 1) that cheesy light dimmer; 2) a fluorescent light fixture (especially one that's flickering); 3) a neon sign.

Let's deal with the dimmer first. Most are the SCR (Silicon Controlled Rectifier) type. Don't use them. "Auto-transformer" type dimmers are much more effective, albeit more expensive. If you think you're saving money by using the SCR dimmer, *Think again!* When you hire someone to track down your noise problems it will cost more than the better dimmer. On two occasions I traced an intermittent crackle outside of the "studio" environment. In one case the problem was a fluorescent fixture on the floor below, and the other, an outdoor neon sign. These can generate powerful (read: annoying) noises.

PROPER GROUNDING

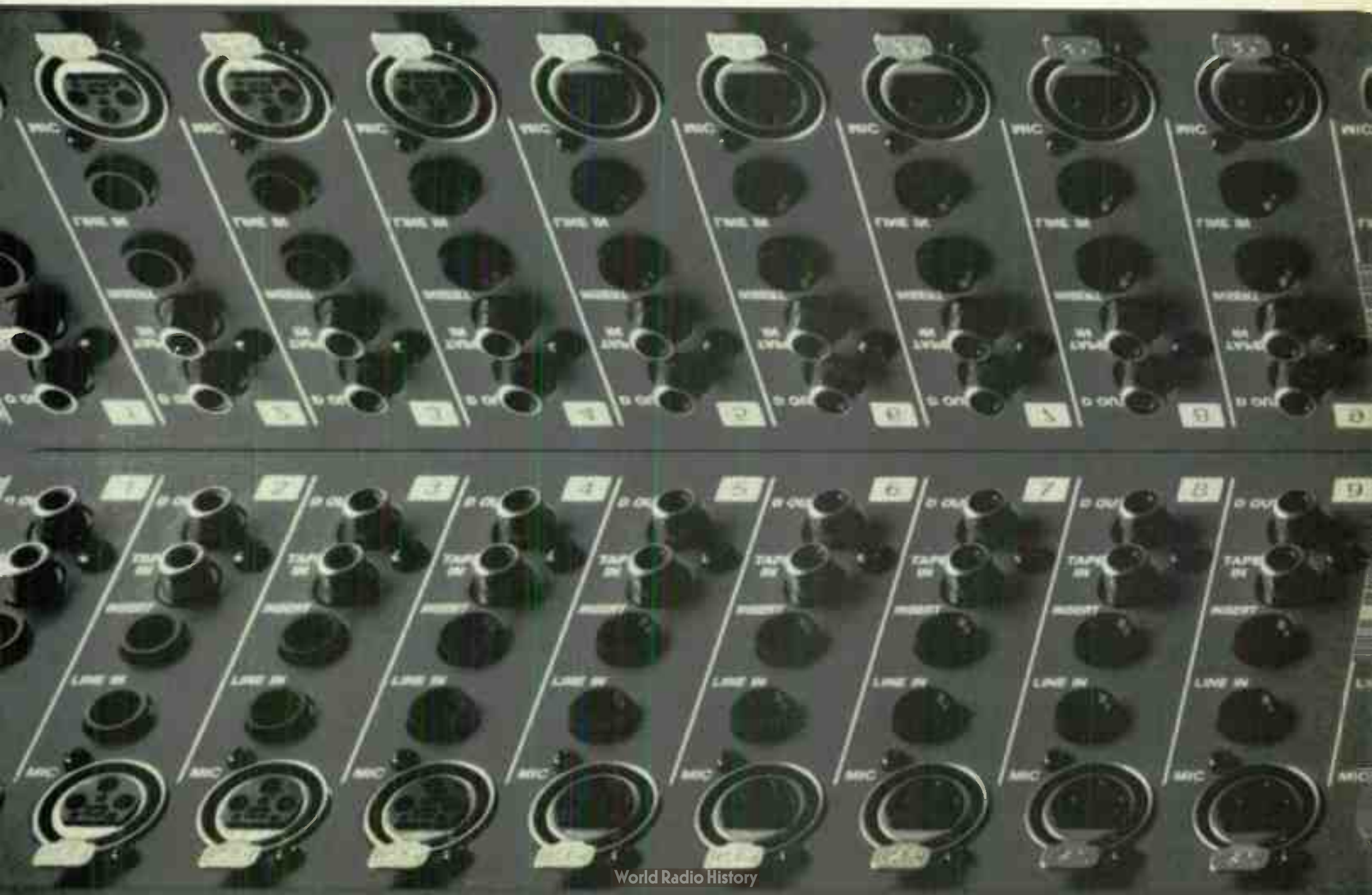
The proper way to ground your system is to use the "Star Ground Technique." This requires that all three-pin power cables be lifted. The optimum ground source is either a cold water pipe or a copper spike/plate that's inserted into damp ground. A large diameter cable (approximately 1/2 inch) brings ground to a central distribution point. From here, individual ground wires (not smaller than 12 gauge) are connected to each piece of equipment. This is the most effective way to produce consistent, noise-free results. It's expensive, it takes time, and it works when followed to the letter. It may not be for you at this time. The larger your system becomes the greater the importance of having a ground scheme.

It's nearly impossible to avoid ground loops. What's labeled as such is quite often the combined result of improper wire dress, inadequate internal grounding plus a ground contaminated piece of equipment. Proper attention must be paid to the "ground scheme" and it can be a time consuming process.

It's common practice to use ground lift adapters to "float," or separate, an audio device from the third-pin AC ground. This may only temporarily eliminate system noises such as hum and buzz. "Floating" the ground pin on all devices means that the "system ground" is now somewhere above ground, greatly increasing the risk of a shock hazard. I have seen many cases where ground lift adapters were used incorrectly. For example, on two pin AC plugs. *Don't do it!* Another misuse is to lift a power strip ground. The gear should be lifted individually instead.

For now, consider the ground lift adapter primarily as a tool used to determine either the source of a ground related problem, or to isolate a ground-sensitive piece of equipment. When used in this way, the wire/tab should be left disconnected. Start without any lifters. Plug in only the essential devices such as the mixer and power amp. Additional pieces can be added in groups, such as effects, signal processors (compressors, EQ, etc.), MIDI, etc. Many of you will be working backwards from a fully connected system.

While the causes of hum and buzz are complex, the solutions can often be simple. Neatness and attention-to-detail are key when trying to minimize power related system noise. Any one piece of gear can inject hum into the whole system.



NAMM

continued from page 28

automation add-ons, made a splash. Mackie introduced a user-installable automation retrofit for the 1604 mixer, AUDIOmation showed their MX816 MIDI Mixing Automation unit (8 channels, expandable to 16) that stores fader moves as well as "snapshots," and JLCooper's MixMaster is a VCA-based, 8 input MIDI-controlled mixer.

EXPANDABILITY (NOT EXPENDABILITY!)
Consumers are tired of the gear du jour syndrome, and manufacturers have taken note. Roland's JV-80 is not only an impressive synthesizer, but has provisions for adding an 8 Meg expansion board of ROM samples. Kurzweil claims that two 8 Meg expansion modules will be available for the K2000 before the end of the year, giving a total of 24 Megs of onboard ROM sounds (you can also have up to 64 Megs of sample RAM). Peavey showed another operating system update to the DPM 3, bringing it up to version 3.0.

So there were the trends. So where are the winners? Check out my personal best list on page 28. But, no, I can't get 'em for you wholesale. EQ

EQ&A

continued from page 15

sources of interference. The advantage of unbalanced lines is low cost and minimal circuitry. Unbalanced lines can only be safely used where all the equipment is powered from the same AC outlet and there are no noise generators around.

An unbalanced output may be "pseudo-balanced" by connecting the "hot" and "ground" to the signal wires and leaving the shield floating at the output. Connecting this configuration to a balanced input provides greater common mode rejection than a standard unbalanced line, but not as much as true balancing.

Transformers, even the best that money can buy, have limited bandwidth, poor phase linearity, and relatively high harmonic distortion. Some of these limitations are minimized in the more expensive transformers but only when the transformers are sourced from and loaded into specific impedances. In other words, different inputs and outputs may cause different responses from the transformer. The bottom line is that transformers affect the sound and should be avoided whenever possible.

One very good attribute of the transformer outputs is that if one output leg is grounded, the full voltage is "swung" to the other leg. In less sophisticated active balanced circuits there is a loss of 6dB when one leg is grounded. Many recent designs do swing the voltage to the ungrounded leg (some designs require that only one of the legs may be grounded); however, problems may arise when the grounded output amplifier runs into a dead short. The Aphex output stage swings full voltage to the ungrounded leg (either one) and turns off the grounded amplifier.

If proper wiring and grounding practices are followed and high quality input and output stages are used, there's no reason to accept the sonic degradation caused by transformers except in the case of extreme RF fields. Balanced lines throughout a system are more immune to induced noise. If complete balancing cannot be maintained, it's more important to have high quality balanced input stages than output stages.

Marvin Caesar
President
Aphex Systems Ltd.

Power Trio

Vision

Opcode's Vision and Studio Vision® are the premier sequencers for the Macintosh computer, and are used by professionals on today's top albums, film scores, and TV commercials. Version 1.4 features *revolutionary* real time editing of MIDI and digital audio, on-screen control of audio and video tape transport controls, input quantize, SMPTE locked markers, drum machine-style loop record/editing, auto locate, fast forward and rewind/shuttle, and more.

MIDI for the '90s: powered by the same kind of microprocessor as your Mac, the Studio 5 has more MIDI interface channels (240), more merging, more RAM patches (128), more processing, more live performance features, and more SMPTE synchronization configurations than any other interface available. And visionary software, OMS (Opcode MIDI System), ties it all together: create your custom OMS studio setup so Vision and Galaxy know what your MIDI connections are, and display your devices by name. You can forget about keeping track of all those cables—OMS does it for you. Version 1.2 with networking of multiple Studio 5 interfaces is the most *integrated* system at any price.

"The Studio 5 is a true monster... its immense processing power and ultra-hip Macintosh user interface sets a new standard of excellence."
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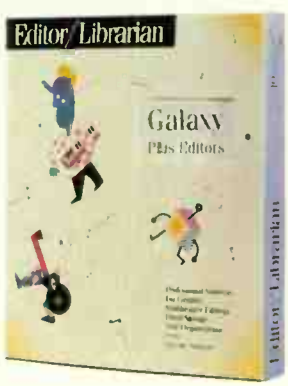

Studios everywhere use and depend on the standard of Opcode patch librarian and editing software. For the first time anywhere, Galaxy Plus Editors offers an integrated package of universal librarian with compatibility for over 140 synths, effects, mixers, modules and *comprehensive* editors for over 50 of them—and we're always adding new ones. Galaxy—The Universal Librarian is also available. Version 1.2 adds a Find command for databasing and searching sounds, plus the Yamaha SY77/TG77 editor.

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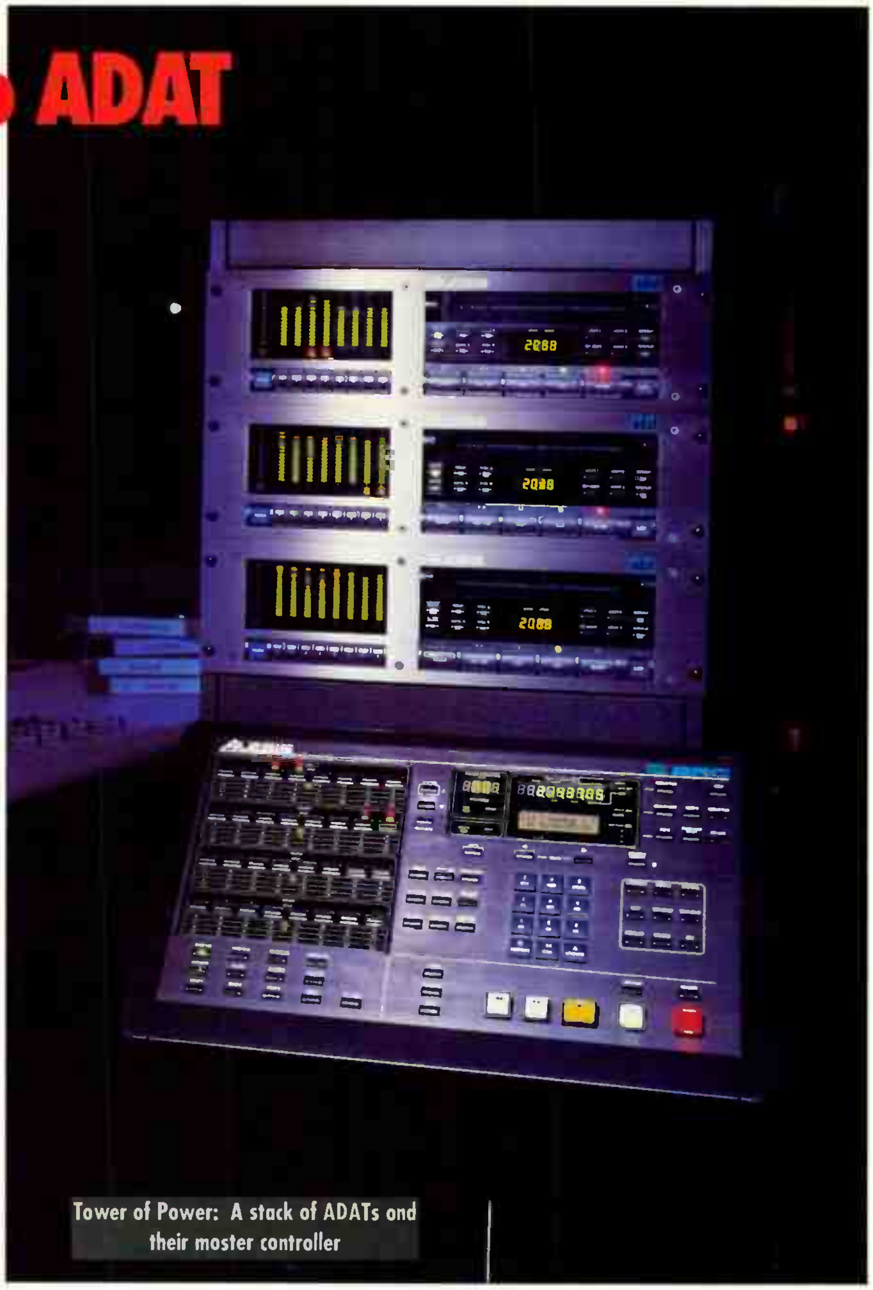
Adapting to ADAT

In the post-ADAT world, everyone will be scrambling to fit in — as quietly as possible

BY J.D. SHARP



Tower of Power: A stack of ADATs and their master controller

The Alesis ADAT is upon us, and it promises to change the way things are done in the world of personal production studios. Its wide dynamic range and the absence of noise normally associated with analog recording will make the shortcomings of other surrounding equipment more obvious. Yes ADAT fans, it's time to start rethinking what you own and what you're going to buy.

If you're going to build a studio around the ADAT or upgrade to accommodate it, the emphasis should be on acquiring equipment that on the one hand can stand up to the signal-to-noise demands of digital recording, and on the other promises to remain current as the state-of-the-art evolves. To this end we've sought mixers, peripherals and instruments that are quiet to begin with, and whenever possible we've looked for digital input and output, which has two advantages. In the near term you'll be able to interface both SPDIF and AES/EBU gear directly to the ADAT, avoiding an extra conversion from digital to analog domains and then back to digital for recording. In the longer run, it's quite likely that all-digital mixing will come to the fore. Gear that incorporates either format of digital I/O will almost certainly interface to whatever comes next.

THE MIXER PICTURE

Perhaps the greatest mystery is which mixer to use. Digital mixing is not a reality, at least not in the cost-effective realm where the ADAT dwells, but this isn't as bad as it might seem. The current generation of analog mixers do a remarkable job when it comes to dynamic range and signal-to-noise performance. The main factor clouding the decision process is that it seems nearly every manufacturer is either already making or plans to introduce a mixer that's "perfect for the ADAT." Over the next eight months or so there will be product introductions from Mackie, Soundtracs, Ross, Alesis, ART and others. Many of these offer significant features. For instance, the Alesis will incorporate comprehensive MIDI muting, eight stereo returns, dual inputs with splittable EQ, stereo Solo in Place and the ability to store muting information directly to the ADAT. It also features three 56-pin connectors that handle all necessary input/output connections for up to three recorders. The targeted list price is $5999, but don't expect to see the mixer before the end of the year or early '93. Mackie's eight-bus mixer is surprisingly complete and promises world-class specifications, dual inputs, stereo Solo in Place and much more, at a projected list price of $3500 for a 24-input version. (There'll also be 16- and 32-in versions.) Expect it midsummer '92. The Soundtracs Solo MIDI ($5999 for a 24 x 8; $8699 for 32 x 8) is much closer to delivery. It incorporates comprehensive MIDI muting that covers the main and monitor inputs, aux sends and returns, and all group and master outputs. Four stereo returns are fitted, and four-band EQ can neatly be divided between main and monitor sources. Solo in Place and PFL are both offered. The Solo MIDI can address up to 24 tracks via either a bus or direct output with no repatching of any kind. Preliminary published specifications look great.

Looking at mixers you can take home today narrows the choice considerably. If you can swing the budget for automation, Tascam's M3700 (24 in: $12,999; 32 in: 14,999) is the only contender below twenty grand. Aside from remembering all fader moves and EQ bypass status for the main channel, mutes are recalled for both main and monitor inputs. For sonic performance, low crosstalk, good mic preamps and excellent signal-to-noise performance, it's hard to beat the Soundcraft Spirit Studio (24 x 8: $5650; 16 x 8: $3995). This 8-bus console is available with either 16 or 24 inputs, and the Spirit contains many features that other manufacturers are scrambling to catch up with. These include splittable EQ (use all four bands in the main channel, or assign treble and bass to the monitor while retaining the two sweep midbands in the main signal path); full 16- or 24-track assignment with no repatching; and direct outputs. About the only things missing are EQ Bypass and Solo in Place; these were deliberately omitted to keep the price down. There's no MIDI muting, either, but many engineers use this feature little or not at all. If MIDI muting is crucial, consider the new GS3 ($4995) from Allen and Heath. This company was recently acquired by Harman, the same folks who bring you JBL, Soundcraft and DOD/Digitech, and if this new mixer is any indication, good things are happening at A&H. Comprehensive MIDI muting is found on board, along with a few twists in the standard feature list. EQ is separate for monitor and main channels, with two-band shelving in the monitor, and three-band EQ for the main with both mid and low bands sweepable. An eight-input expander can be added to either the 16- or 24-input version.

EFFECT-IVENESS

Somewhere in that tangle of mixers you should be able to find a solution that won't muck up the digital dynamics of the ADAT. Signal processing is a bit more rewarding when it comes to keeping the signal completely in the digital domain. Three effects processors contain the requisite digital I/O: the Lexicon 300, Roland R-880 and the Korg A1. The Lexicon 300 ($4795) offers top-quality reverb, ambience, delay and pitch shifting, and includes goodies such as direct MIDI control and timecode synchronization that make it desirable in the high-tech studio. Roland's R-880 ($3995; requires GC-8 graphic controller [$845]) is arguably the most useful all-digital piece, since it delves into parametric EQ and compression as well as dealing comprehensively with reverb, delay and chorus. The Korg A1 ($1995) is probably the best choice when seeking a unit that's equally at home as a studio-grade effect and an instrument processor. Aside from the usual array of effects, it offers overdrive, rotary speaker, compression and gating options, and several of these processes can be used at once. If all these units require excessive deficit spending, choose from among the numerous analog I/O units for something quiet; the new Intelliverb from RSP Technologies (formerly Rocktron) and Roland's RSP-550 are two candidates in the just-over-$1000 arena, while the Alesis Quadraverb and Digitech DSP256XL would make excellent choices in the $500-and-under realm.

THE EQUALIZERS

There are surprisingly few choices in digital I/O equalizers. In fact, Roland's E-660 ($1995) is about it! Luckily it offers two independent completely parametric equalizers, and is fully programmable. A hum canceling feature is incredibly valuable when dealing with single-coil pickups and lighting buzz. In the analog domain, it's hard to beat the SP-15 Studio Parametric ($599) from Rane. Its 120 dB of dynamic range let it handle anything the ADAT can dish out, and then some. Five bands of EQ are offered, and the top and bottom band can switch to shelving characteristic if desired. Bandwidth as narrow as 0.03 octave or as broad as an octave and a half can be dialed up.

COMP/LIM/GATE/ETC.

Compressors and gates offer no all-digital choices other than the aforementioned features of Roland's R-880. But there are some awfully good choices. The **Aphex** Expressor ($495) offers tremendous parameter control, along with a natural-sounding high frequency expander. The **dbx** 160X ($429) continues to be a professional favorite, especially for its intelligent auto-attack and -release mode. Keep your eye on new processors from **Behringer** that are just being introduced; this company has developed a great reputation in Europe for cost-effective, top-grade devices. Also from overseas comes the **Drawmer** DL241 ($749), a compressor/limiter/gate in the dbx 166 mold, but with a bit more refinement and excellent specifications. For a cost-effective yet still very quiet comp-limiter-gate, nothing touches the **Alesis** 3630 ($299). And there are many more viable possibilities from **Audio Logic, JBL, Furman, Peavey/AMR** and others.

ADAT ADDENDUMS

There are a few more points to consider. Your monitor system may find itself dealing with more dynamic range than you've been accustomed to. Invest in an amp with plenty of excess power above what's normally needed (headroom), perhaps 200 watts per side, and speakers that can handle it. There are so many good amps that it will be left to your discretion to select one, but several monitors stand out. **Tannoy's** System 8 NFMs ($1000/pair) are about as large as near-field monitors can get. They handle plenty of power, and their single point source (co-axial) design delivers exceptional clarity and phase precision. **Peavey/AMR** PRM-308s ($599/pair) are incredibly accurate and full-ranged, yet remain very affordable. Near-field monitoring is probably where it's at for most smaller studios, because the room acoustics don't interfere as much with making accurate judgments, but if monitors are placed farther back, consider either System 10s or 12s from **Tannoy** ($1595 and $2195/pair, respectively), or **JBL's** 4425 monitors ($2190/pair), which feature a substantial 12-inch woofer matched with a bi-radial tweeter.

There are only a handful of instruments with digital outputs, although this feature promises to be as ubiquitous as a MIDI jack within a year or two. Digital output is an option for both the **Yamaha** SY99 and the **Kurzweil** K2000. It's a standard feature on **Roland's** flagship S770 and **Akai's** S1100 digital samplers, and can be optionally added to the S1000.

There's little question that the ADAT will change our expectations when it comes to recording, further obliterating the rapidly fading line between professional and personal performance levels. Once the recorder is here, getting the rest of the signal chain up to snuff emerges as the primary project studio challenge. EQ

Phil Ramone photos by Michael Bloom

We asked Phil Ramone to field test our new AT4033 studio condenser microphone.

He wouldn't give it back!

AT4033
Studio Condenser
Microphone

Phil Ramone knows exactly what he wants from a studio microphone. And when he tested a sample of our new AT4033 cardioid condenser microphone, he knew it was right for him and ideal for the artists he records.

He liked being able to concentrate on getting the right music from the musicians, rather than first spending time experimenting with EQ to get the right sound.

What Phil Ramone heard was the result of new condenser technology inside the AT4033. The diaphragm is only 2 microns thick, with a vapor-deposited gold conductive coating just 50 angstroms thick. This sophisticated, very low-mass diaphragm is aged in a five-step process that insures unchanging performance for years.

The high head-room and wide dynamic range, plus low noise floor, make the AT4033 ideal for the most demanding digital recordings. And the maximum input SPL is an awesome 140 dB, so important when recording high-output instruments and very close-up vocals. In addition, transformerless design contributes to overall sonic transparency. The AT4033 also includes a switchable 10 dB pad and lo-cut filter, plus a built-in pop filter and internal shock mounting.

We're not certain we'll ever get the sample AT4033 back from Phil Ramone, but no matter. We're busy making your AT4033 right now. For more details on this impressive new microphone, ask your A-T sound specialist to schedule a test of the AT4033 today.

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CIRCLE 32 ON FREE INFO CARD

Chipping at the Sounds Barriers

Chip technology today is rapidly creating the personal recording studio of tomorrow

BY MARTIN POLON

To see where "chips" will take us in the further development of personal recording, let's look back at the development of recording studio technology. Return to the behemoth monaural eight-input vacuum tube consoles of the early Sixties to get a glimpse of the overall progress that's been made. All of the circuitry contained in one of those Collins or RCA consoles, weighing several hundred pounds, today would barely fill a single integrated "chip" weighing several ounces.

Generally, tube consoles and other electronics were all built in a "discrete" fashion. Each component, be it a resistor or a capacitor or an inductor, existed as a separate part and was hand wired from tube socket to tie point. Next came the transistor consoles at the end of the Sixties, designed to eliminate vacuum tube's high voltages, transformer isolation and large components, but which remained discrete in wiring and design. The Seventies brought us the era of integrated circuits (IC's), in which the separate transistors and circuit components were created on substrates of semiconductor materials and sealed into indestructible plastic.

CHIPS OFF THE OLD BLOCK

These early IC circuits were the gathering into a single package of very standardized designs that had achieved popularity in discrete usage. There was some savings in cost and an increase in reliability. More and more ICs were developed to replace most discrete circuits. Consoles became first a collection of ICs and then as the technology advanced, the contents of several ICs were further shrunk and placed on a single IC. This process, known as VLSI (very large scale integrated) technology, shrunk the amount of interconnection inside audio products. Costs were greatly reduced and quality improved due to the uniformity of the circuitry.

During the latter part of the Eighties, we began to see ASICs. The application specific integrated circuit (ASIC) allows equipment makers to design their own custom circuitry which is then created in silicon by custom chip makers. Personal audio recording equipment designers can create entirely new products on a chip rather than having standard chip designs dictate a product's function and performance. The revolution we now enjoy in personal and project recording is a direct result of the evolution of electronic "chip" design. In the future, analog and of course digital chip technology will take us to plateaus never before imagined in the recording of music.

BLUE CHIP FUTURES

At the same time, computer "chip" technology has progressed to the point where the original 8-bit pathways under 1 megabyte speed and 64K bytes of random access memory (RAM) of the first personal computers are now found in digital watches. Tomorrow's "personal computers" will become "personal audio recording stations" with 64 and 128 bit pathways, 80 megahertz clock speeds, full gigabyte hard disks, 64 megabytes of RAM and voice input and output. Applications software and plug-in boards will customize the computer for mixing, editing and digital signal processing — simultaneously, if so desired.

Stepping beyond that level, the relentless progress in silicon memory chip development should make motionless solid-state recording a reality sometime towards the end of the Nineties. At this time, about 4 million memory transistors capable of storing one bit of information can be placed onto a single silicon chip. By the end of this century, computer scientists expect to place at least 256 million of the memory units on a single "chip" and at an operating speed 300 times as fast as current units. An array of such chips would provide a viable platform to record and reproduce multitrack digital audio with not so much as a moving part in sight.

2001 — A SOUND AND SPACE ODYSSEY

Our future studio scenario might work like this. A producer and a mixer walk into a control room to prepare for a session. The room is dominated by a large computer with an overhead plasma display. The producer places an optical disk into a drive and loads the specific mixing software for today's session. The software also configures the large memory banks for recording purposes. The mixer "converses" with the computer, modifying the various "console" settings displayed up above on the plasma screen. A laser "pen" is also used to make quick changes directly on the screen. Once the hardware is configured, the session can proceed. All signal processing is done in the digital domain without any noise or distortion. After the session, a different software is loaded into the computer to facilitate editing. After all of the production work on the session is finished, a blank read-only-memory (ROM) storage chip is "cut" with the finished record. Millions more can be duplicated without loss from the "master chip." Consumers will buy these album chips to plug into their car units, their home receivers and their portable walkarounds. A fascinating scenario of "chip" progress, indeed! EQ

Digital Without the THuD

Photo by Peter Monroe

In the digital domain, Total Harmonic Distortion is a whole other beast, one with two heads

BY LEN FELDMAN

You know distortion when you hear it. But do you know what it really means? Understanding the basics of total harmonic distortion, or, as it's frequently called by those who prefer acronyms, THD, is the first step in keeping it under control during the recording process.

As you're probably aware, THD, like other forms of audio distortion, is quoted in percent. But, percent of what? Well, let's start at the beginning. If an electronic amplification circuit (amp, preamp, mixer, you-name-it) were perfect (and none that I know of are), it would deliver at its output terminals an exact replica of the signal or waveform that was applied to it — perhaps greater in amplitude or lower in amplitude, but nevertheless an exact replica. If you feed in a single mid-frequency tone, say, at 440 Hz (that's middle "A" on the piano, in case you're wondering why I chose such an odd frequency), all you should get out is a 440 Hz tone. Not 880 Hz, not 1320 Hz, not 1760 Hz — well, you get the idea. Unfortunately, those other frequencies — all of them multiples or harmonics of the fundamental 440 Hz — do appear, usually in small amounts. And when you add them to the original waveform that waveform changes; it's no longer an exact replica of what you put in at the input.

Getting back to the percentage question, the ratio of the total amount of extraneous or harmonic signal content divided by the amount of the desired, original signal at the output is simply expressed as a percentage. For example, if the output voltage at the speaker terminals of a power amp is 20 volts, but 1 volt of that is the sum of all those unwanted harmonics of the original signal frequency, then 1/20 is the harmonic distortion level or, putting it in percentage terms, the THD of that amp, at that output level and at that frequency, is 5 percent. We'll get to the question of how much THD is audibly disturbing shortly, but for the moment let's stay on the subject of measuring THD. So how do you figure out the "total amount of the unwanted harmonics" when all those signal frequencies are jumbled together? Generally, you use a distortion analyzer. All the distortion analyzer does is to use a filter to null out the desired, fundamental frequency (440 Hz in our example) and what's left, of course, is the total harmonic distortion. Or is it?

DISTORTION ANALYZERS OFTEN LIE!

In fact, common, garden variety distortion analyzers don't normally read only THD. To be perfectly honest, they read THD + Noise! Suppose you have a defective power supply in one of your line amps. It's spewing out hum levels at 60 Hz and 120 Hz (and perhaps even at 180 Hz) that are half as big in amplitude as the audio signal you're trying to recover at the output terminals. The typical distortion analyzer can't tell whether these extraneous signals are harmonics of the desired input signal or not. So it reads out a "distortion" figure of — are you ready for this — 50 percent! In fact, the actual harmonic distortion may be only 1 percent, or 0.1 percent, or even 0.0001 percent. There's no way of telling with an ordinary distortion analyzer.

TELLING THE WHOLE THD TRUTH

Unfortunately, most manufacturers and testers tend to quote the readings they get on a distortion analyzer as THD, when in fact they should call this specification THD + Noise. To separate the real harmonic distortion from the extra noise, it's necessary to use a more sophisticated piece of test equipment known as an audio spectrum analyzer. Generally equipped with a display, such as an oscilloscope, a spectrum analyzer sweeps across a band of frequencies and shows what's there at all audio frequencies. So, in our example, it would show the fundamental 440 Hz as a tall spike of full amplitude and then, further along in the sweep, it might show very small amounts of those harmonic components at 880 Hz, 1320 Hz, 1760 Hz, etc. If there were hum components or other noise at non-related frequencies, those would show up too, at their appropriate frequencies in the display. And if you wanted to really know just the value of THD, you could ignore the noise and the non-harmonically related components when figuring out the value of total harmonic distortion.

But now that we've got the amplitudes of the various harmonic components clearly defined, how do you "sum them up" to arrive at a real THD figure? Anyone who's got a phobia about math computations can skip this paragraph. For the rest of you, you simply take the square root of the sum of the squares of all the harmonic components, (ignoring those components at least 10 dB lower than the largest harmonic component) and the answer you get is the true THD. Don't panic! Let me give you a quick example. Suppose there's a second harmonic component that's 0.01 times as big as the desired component, while another, third harmonic component is 0.005 as big as the desired fundamental. Well, 0.01 squared is 0.0001, while 0.005 squared is 0.000025. Add those two results together and you get 0.000125. Take the square root of that, to get 0.011803, or thereabouts. Drop a couple of leading zeros behind the decimal point to change the number into a percentage and *Voila*!, you come up with a true THD reading, in percent, of 1.11803 percent. There, now that wasn't so bad, was it? Well, if you think it was, you can stick with the simpler distortion analyzer, and hook up an oscilloscope to it so you can look at the THD + Noise waveform itself. That way you can usually tell if the reading consists largely of harmonic content or if there's also noise that's dominating the reading.

LISTENING TO HARMONIC DISTORTION

Back in the good old days (?) of analog audio, you could pretty well count on THD being very low for low signal levels and very high as you drove the piece of equipment into overload or clipping. It was widely believed that harmonic distortion levels of under 1 percent or so, when listening to actual music signals, were pretty much inaudible. Of course, using single test tones, most of us can identify THD levels lower than that, especially if we switch back and forth between the "clean" signal and the one containing small amounts of THD. (Some golden-eared recording engineers claimed to be able to hear even 0.1 percent THD when listening to reproduced music, but I've always believed they were faking it and had advanced knowledge of actual THD levels.) So, all we had to worry about was keeping all signals at levels that would not cause clipping in preamps, mixers, power amps and the like. But then, along came digital — and everything changed!

LOWER THE LEVEL THE HIGHER THE THD

Because of the way digital signals are stored or "quantized," with digital audio, things actually work in reverse when it comes to total harmonic distortion. Look at any "spec" sheet for a DAT recorder or any digital audio product and you'll find minuscule THD "specs," usually no more than around 0.01 percent or even 0.005 percent or lower. What the manufacturer doesn't tell you is that those THD values are taken relative to a maximum recorded signal level — the loudest signal that can be handled by the particular digital system or component. Back off 20 dB from that maximum level and the relative THD for that lower level signal is ten times as great! Record a really low-level musical passage, say, at -60 dB, and all things being equal, the THD for the unit that boasted a spec of 0.01 percent THD will be 10 percent! And 10 percent harmonic distortion is audible to even the most uncritical listener. So, clearly, when working in the digital domain, you want to keep your recording levels right up there and not let the softer passages of music get so low in level that they are swamped by noise and harmonic distortion.

BUT BEWARE THE DIGITAL BRICK WALL!

Would that a recording engineer's life were that simple! The fact is that, when dealing with digital audio, there's an upper boundary you have to watch out for. In the analog world, if you went above a +4 dBm reading for peaks, you got into a bit of distortion, perhaps, but the way in which distortion increased was sort of gradual, so in most cases, a certain amount of over-recording was tolerable. Not so with digital! Remember, in digital audio you're dealing with numbers — zeros and ones — nothing else. And if you're working in a system where digital samples are 16-bits long, the "loudest" signal you can handle is one whose sample is represented by 16 "ones." Push the system still harder, and, digitally speaking, there's just no headroom available. You'll run into the most horrendous looking square wave clipping you ever saw (or heard) and that kind of clipping means equally horrendous and audible distortion that will drive any listener up the wall. So, when it comes to digital audio, harmonic distortion lurks at both ends of the dynamic range scale, and you'd better beware! EQ

Playback It Again, Sam

How to store and restore your magnetic media without flaking out

BY RICHIE MOORE, PH.D.

You put your master tapes away years ago after the record stopped selling. They have been sitting someplace in storage just in case you ever need them. Finally, the record company wants to re-release the album on CD. You go and find the tapes and get ready to re-master them. You put the first tape up and find to your horror that it won't even play because it is sticking to the heads and guides. The next tape doesn't sound as bright as you remember. The tape after that has dropouts in it; the list goes on. At this point you are severely bummed — but not alone.

Magnetic media is a very volatile storage device. Magnetic media includes audio tape (both analog and digital), videotape, cassettes and computer floppy and hard drives. Personally, I rarely see people take care of this media as well as they should. This is also another one of those topics there's not much information on, though there should be.

TAPE ME, I'M YOURS

Magnetic tape manufacturers are taking more and more care to make sure that tape arrives at the end user in great shape. Ampex came up with the plastic tape collar some years back to protect the end of the tape and to add support to the reel flanges. Scotch (3M) now has the tape available in tough plastic boxes to protect the tape and reel. The tape usually comes in a sealed plastic bag to protect it from dust, dirt and humidity while in transit. These precautions all help the tape to reach the consumer in better shape than ever before.

But the buck (and the tape) doesn't stop there. You should store magnetic media under the right environmental conditions. This stuff is made out of plastic and magnetic iron particles. It's very susceptible to extremes of heat, cold and moisture. Media needs to be stored between 55 degrees and 80 degrees Fahrenheit. The relative humidity should also be 50 percent. Extremes can damage the polymer compounds that make the backing of the media and, as we all know, Rust Never Sleeps.

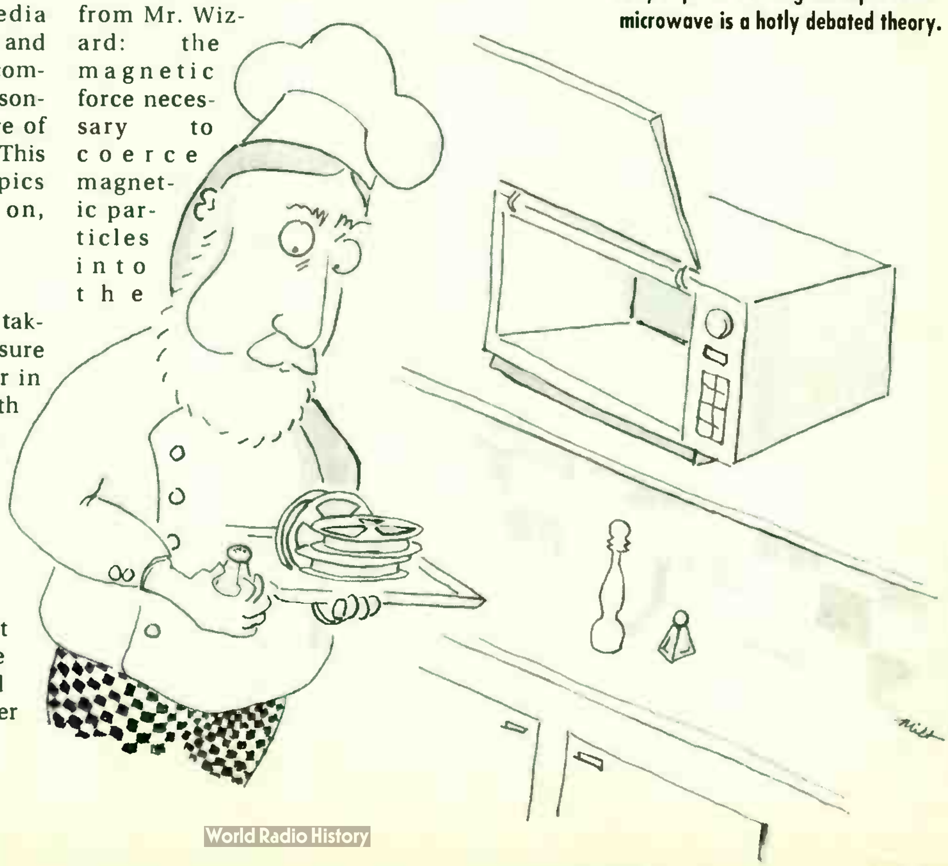
The world is wrapped in magnets other than the North and South Pole. Most of them are not apparent in day-to-day life, however. Power lines create an EMF (electro-magnetic field) that radiates enough energy to be a modern physiological concern. Anything that creates a magnetic field is a concern to magnetic media, and you really have to be on your toes about this situation. Speakers, servo motors, appliances, television sets, clocks, automobile drive trains and household tools are only a few items that can have a magnetic field.

Consider this fact from Mr. Wizard: the magnetic force necessary to coerce magnetic particles into the proper place on a piece of magnetic tape and remain that way is less than 300 Oersteds. Translated from rocket science, this amounts to a very, very small field. One you shouldn't play on.

KEEPING MEDIA THE MESSAGE

At this point in the discussion media will mean any type of magnetic material for data storage, but tape will be tape. You should always inspect the cartons and the individual boxes for dings and scuff marks. Also look for water damage and excessive amounts of diesel soot. Don't be afraid to say that a piece of media looks suspect and that you don't want it. You needn't suffer improper shipping and handling. The dealer can return it for credit.

When you get the media to the studio, immediately put it in a safe place. Most facilities have a library for materials. This, as was mentioned earlier, should be a place with proper environmental controls. There should be no electronic devices in the room.



Tasty Tapes? Reviving old tapes in the microwave is a hotly debated theory.

Illustration by Milton Reyman

Shelves should not be located near AC power lines or junction boxes, and this is no place for refrigerator magnets. Media should be placed upright and not laid flat on its side. It's best to leave it in the original shipping carton until needed.

What follows may seem like a lot of hassle, but is well worth the effort. When opening a box of tape, don't throw the collar away if it comes with one. Also save the sticker on the head of the tape that contains the batch run numbers. I usually attach them to the reel that the master tape will live on. This helps track a roll of tape in case you have to complain to a vendor if the tape disintegrates with your master on it. Always feed the tape from the reel it came on originally to another reel of the same type. This will probably involve buying some spare reels of the same type. Feeding the tape from the original direction maintains the desired position of the magnetic particles that need to be aligned and assures better tape path stability.

At the end of a session, take all the tapes and store them tails out. This will make any print-through to be post-signal rather than pre-signal. It's still there, but not as noticeable. If the machine you have has spooling mode (half the speed of rewind or fast forward), use this speed to pack the tape evenly from top to tail. The slower pack keeps the edges of the tape from skewing and causing possible damage to the edges.

At the end of a project, pack all the tapes at 30 ips. This may be a real pain, but not as big a pain as losing your masters. Do not use all types of masking and leader tape to mark the tape or the reels. The stuff can ooze sticky stuff after a time. Re-check all the splices to make sure they don't ooze into the tape when tightly packed. Pack all the tapes in a plastic zipper bag to seal in freshness, enclosing a packet of silica gel to keep the moisture level low. Put the tapes in the tape locker so they'll be safe. If you must store magnetic materials off-site, check the location well to make sure it's as safe as your own vault.

DAT tapes and videotapes of all formats should be stored the same way. Keep them away from magnets and direct sunlight. Store them upright, wound to the tail for storage. You may not think that video or digital has the same problems of print-through as audio tape, but it does.

BEFORE PLAYING TAPS, TRY THIS

For whatever reason a tape deteriorates, you should take action on it immediately. Time can only make it worse. Some batches of brand tape, which will remain nameless, have had a shedding problem with the oxide. I almost lost a whole Huey Lewis and the News album because the shedding was so bad you couldn't play the tape more than 35 seconds without losing fidelity. Other makers of tape have a big problem with stiction after a time. Stiction is a problem where the lubricant and the oxide particles break down and cause the tape to bond with the heads and guides, rendering them unplayable. Continued attempts to try to play a tape usually ends up destroying it. (If you'd like the brands and the model numbers of these problem tapes, write me care of *EQ*.)

Many engineers have had success saving tapes by baking their tapes for an unspecified amount of time in a microwave oven. It sounds like a last ditch (dish?) effort to me. In 1987, AGFA introduced a process to recover magnetic tapes: the AGFA XT process. IDT (Innovative Development Technology) of West Palm Beach, Florida (800-447-3083) is a tape dealer who has a facility to do the XT process exclusively. They have audio engineers trained in the proper preparation of tapes for the XT process and have the knowledge and experience to deal with the most extensive tape deterioration. Don't take the law (of physics) into your own hands. As the folks at IDT explain, you can make sure that your magnetic recordings live on and don't become "history."

Whatever you do, take care of your magnetic media. You can store and restore media properly if you take the time. There's no substitute for regular maintenance. And remember: "MAGNETS WANT OUR MASTERS. ALIENS WANT OUR MATES." EQ

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ANALOG ENTRY

continued from page 106

a drum sound module containing about 500 excellent, high quality drum sounds. Ross Garfield, the "Drum Doctor," based in L.A., provided most of the drums that were sampled for the unit. Walter and I rent his drums for our recording and sampling sessions. The D-4 has the ability to trigger from MIDI or audio, and will even put out a MIDI note when triggered by the audio source. For mere mortals, the D-4 is fine. I have used it on Rodney Crowell's new album, mostly as an adjunct to the snare that's already on tape.

When triggering off of tape, there's a 6 millisecond delay from the beginning of the drum on tape to the start of the drum produced by the D-4. If you trigger from pads or an audio source that doesn't need as much processing by the D-4, the delay is around 3 milliseconds. The D-4 takes 2 milliseconds to respond to a MIDI note in. My initial response, after measuring the delays, 6 milliseconds seems like a long time (a near eternity for Donald), but when it's used to add to the sound of an existing snare drum, you can't really tell there's any delay. The drums sound so good they blend readily with just about any trigger source.

I'm working on an article that covers every possible way to replace drum sounds and compensate for the timing slop inherent in some of the systems. These methods were passed on to me by an utterance from a smoldering pineapple bush near the 17 mile marker on the Hana Highway on Maui. I have some translation yet to unravel, but the results will appear in a future *EQ*.

FLIP SIDE:

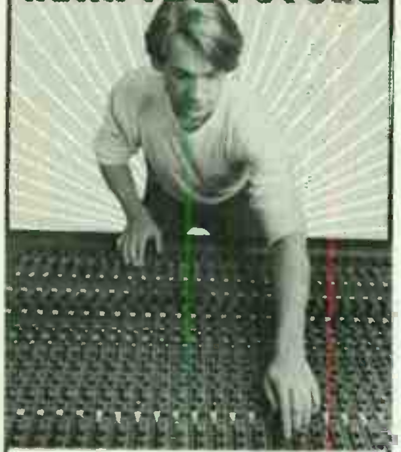
It's nice to ramble on about the unlimited budget projects that pop up now and again, but most folks have to deal with "reality." (Hang on a second, I have to look that up to make sure I spelled it right. Yup, okay.)

"How can I record an album of my own and make it sound good?" you ask Well, I'm gonna have the answer for you real soon now. I'm going to complete three entire record company projects using semi-pro gear. The first test will be using the Fostex G24-S one-inch 24-track machine with Dolby-S noise reduction and built-in synchronizer for lock-up and auto-punch. I share production credits on the Fostex project with Jerry Carrigan, a Muscle Shoals/Nashville drummer, and a singer/songwriter named Conrad Reeder. The Fostex "testees" (I guess I shouldn't use that term, seeing how the Fostex project is analog) will be a country singer named Linda McKenzie, and a pop/alternative band named "Fugitive Blonde." The third project will be with the Alesis ADAT system, also 24 tracks worth. The results will appear right here.

The plan is to use the quietest and most efficient signal paths on both recording and playback. This also tends to be the least expensive route. Instead of a big console, we are using a Roland M-24E mixer for playback of the multitrack and monitor on HD-1s. The recording is done with Rane mic pre-amps, for vocals or acoustic instruments, and Brooke-Siren direct boxes for MIDI driven synths and samplers. The outboard rack is the rack from hell, and I'll get into that more in the up-coming article.

Keep your fingers crossed for me. This has to work, otherwise I'll have to figure out where to get the money for a Sony 48-track. Maybe the benevolent patrons at *EQ*? Naaaah. EQ

HEAR THE FUTURE



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IN REVIEW

Roland SBX-1000

EQ LAB REPORT

MANUFACTURER: Roland Corporation US, 7200 Dominion Circle, Los Angeles, CA 90040-3696; 213-685-5141.

APPLICATION: The SBX-1000 is three devices in one: (1) a cue list section that allows you to create a series of MIDI events triggered at particular SMPTE times, in much the same fashion as a video editor creates an Edit Decision List (EDL); (2) a 16-track MIDI sequencer with a full complement of editing features; (3) a tempo controller that allows you to build a tempo map that controls playback of the internal sequencer and/or connected external MIDI sequencers.

SUMMARY: The SBX-80's long-awaited successor, the SBX-1000, performs as flawlessly as its predecessor.

STRENGTH: The tempo controller section can create a SMPTE-synchronized MIDI tempo map in one of five ways.

WEAKNESS: Doesn't support MIDI Time Code.

PRICE: $3495 **EQ FREE LIT. #:** 126

BACK IN THE MID-EIGHTIES, ROLAND created quite a stir in studio circles with the release of the ubiquitous SBX-80. Despite having an owners manual that may well qualify for the "Worst Documentation of All Time" award, the unit became enormously popular because of its functionality and precision. The SBX-80 was pretty single-minded; it was a SMPTE timecode reader and generator that also allowed SMPTE timecode and MIDI timing commands (both clock and song position pointer) to be synchronized, enabling a MIDI sequencer to lock to prerecorded acoustic instruments.

Roland recently unveiled the SBX-80's long-awaited successor, the SBX-1000 MIDI Cueing Box. The first question has to be: Does the SBX-1000 perform as flawlessly as its predecessor? The answer is hands-down "yes." The second question (at least for SBX-80 owners with long memories) is: Is the owners manual in English? Here, the answer is "sort of." This time, Roland provides not one but three manuals. While they're considerably better-written than previous efforts, they're still far from perfect — but the good news is that the unit itself far surpasses its documentation.

GOOD TRI

Essentially, the SBX-1000 is three devices in one. First, there's a cue list section that allows you to create a series of MIDI events (as well as General Purpose Interface [GPI] events for machine control), triggered at particular SMPTE times, in much the same fashion as a video editor creates an Edit Decision List (EDL). Second, there's a 16-track MIDI sequencer with a full complement of editing features. Third, there's a tempo controller that allows you to build a tempo map — synced to SMPTE — that controls playback of the internal sequencer and/or connected external MIDI sequencers. The SBX-1000 does not support MIDI Time Code (MTC).

On the SBX-1000 back panel are two sets of MIDI in/out ports as well as SMPTE input and output, four GPI output, and audio input jacks. On its front panel, you'll find a full complement of cursor arrows and a weighted data entry wheel. Tape-recorder-like controls allow you to easily play, stop, record, rewind and fast-forward. Roland has also thoughtfully provided six locator points for playback. These SMPTE values can be entered manually from the numeric keypad or can be set on the fly. An onboard 3 1/2-inch disk drive enables you to save the work created in any or all of these sections, and also allows you to save your setups (though not locator points!) and a list of most-used MIDI events, as well as facilitating system updates periodically provided by Roland.

THIS IS YOUR CUE

Let's look at each of the three SBX-1000 sections individually, starting with the cue list. There are three ways to enter events into this list. You can place them manually, much as you create a sequence in step time, only here events are placed at specific SMPTE time points (within a resolution of 1/4 frame) instead of bar/beat points. Alternatively, you can enter cues via MIDI input, a technique which can have particular value in sound effects work. A third means of data entry allows you to record predetermined MIDI events (from a user-created event list) either by hitting the front panel Tap key or in response to percussive audio input.

MIDI controller messages, program change messages, pitch bend, after touch and even system-exclusive messages can be recorded in this way and dutifully played back by the SBX-1000 at the correct SMPTE times. The cue list also allows you to embed sequencer start commands (to start the internal or any externally connected MIDI sequencers) as well as GPI commands, which can start or stop connected CD players or tape decks (pretty much any that have footswitch inputs). All cue list events can then be freely edited; for example, you can move them forward or backward in time or change any of their parameters (such as note number, velocity or gate time). Up to sixteen revisions of a single cue list can be created and stored.

The sequencer section pretty much follows the Roland convention; if you've used any of their MC or MV series hardware sequencers, much of the layout will be familiar to you. Up to twenty songs can be stored, with a maximum capacity of 110,000 notes (the sequencer memory is shared by the other two sections). Maximum clock resolution is 1/96 of a quarter note; there are sixteen tracks, each of which can store data from multiple MIDI channels, and there's full provi-

sion for overdubbing and punch-in recording, as well as track copying and selective extracting and/or merging of data. A "microscope" function allows single event editing. Sequences created in other Roland instruments can be directly loaded by the SBX-1000 disk drive. You can also load standard MIDI files (as well as export in that format) from an MS-DOS formatted disk, although the review unit I used had an annoying tendency to lock up whenever I tried loading MIDI files created in a PC. I'm told by Roland that this problem will be corrected in a future software update.

UP TEMPO

The tempo controller section is where the SBX-1000 really shines. Here, you can create a SMPTE-synchronized MIDI tempo map in one of five ways. First, you can convert the tempo map used by the internal sequencer. Second, you can use the numeric keypad to manually enter in tempos, beat by beat. Third, you can play in the desired tempo by hitting the front panel Tap key. Fourth, you can input a prerecorded percussive audio signal (such as a click on tape). Finally, you can input MIDI clocks from an external sequencer — this allows you to import a tempo map created in another hardware or computer-based sequencer.

Once you've recorded a tempo map, you can edit individual beats and/or smooth tempo changes. You can also use the Correct command to rescale any section of the map (or even the entire map), either by a ratio value (for example, have it play back 2 percent faster) or by having it fit a certain time area (for example, have bars 3 through 6 play back in exactly 4 seconds; the time area you select, by the way, can be resolved to as little as 1/80 of a frame). Meter changes can also be entered, although this can be done only after the tempo map is recorded. Up to 32 different tempo maps can be created and stored in the SBX-1000's internal memory.

ALL FOR ONE

There can also be a good deal of interaction between the three sections of the SBX-1000. For example, you can start the internal sequencer from within the cue list and/or control its playback tempo from the tempo controller. From within the tempo controller, a really hip Shift command lets you automatically move the cue list sequencer start time so that a particular bar/beat occurs at a specific SMPTE point. Thus, you can view a striped video and have, say, the downbeat of bar seven occur at a precise point in the action.

Because the SBX-1000 uses a software-based operating system (like most Roland products that have disk drives), there's plenty of reason to think that it will evolve with future developments in synchronization. One such area may well be MIDI Machine Control, and we can probably expect to see the SBX-1000 eventually act as a master MIDI machine controller as well as a synchronizer.

There's no question that the SBX-1000 can be an invaluable tool in the post-production studio. Although its price (list: $3495) is rather steep, you'd need a computer, a SMPTE/MIDI interface and more than a thousand dollars' worth of software to emulate all of its functions. If you've already got such a computer system, you may not need all the power this unit provides — but if you're building a system from scratch or if you're not already using a computer for synchronization, the SBX-1000 is worth a serious look.

— Howard Massey

Mackie CR-1604

EQ LAB REPORT

MANUFACTURER: Mackie Designs, Inc., 16130 Woodinville-Redmond Rd., NE, No. 2, Woodinville, WA 98072; 1-800-258-6883.

APPLICATION: Primarily a 16 x 2 mixer, easily expandable to 32 or 48 inputs, with limited but useful EQ, and more effects sends than you've ever dreamed possible.

SUMMARY: Compact but ergonomically designed; it's priced in the "keyboard mixer" range but its specs belong to a much more expensive category. Neither too complex nor too simple, it offers a unique combination of features arranged like a good computer program: a straightforward user interface with a great deal of sophistication under the surface.

STRENGTHS: Seven auxilliary sends and four stereo returns. Very clean sound. Gain structure easy to optimize. Inexpensive.

WEAKNESSES: Small faders. Limited EQ. Return levels don't go low enough.

PRICE: $1099 **EQ FREE LIT. #:** 127

A TOUR OF THE RECENT NAMM convention proved one thing — numerous manufacturers have been watching the success of the Mackie CR-1604 mixing console and decided to get a share of the action with Mackie-clones of their own.

The success of the CR-1604 is based on the fact that it's the right product at the right time. That's because, until now, mixing consoles geared towards electronic music studios tended to fall into one of two categories: (1) complex boards with long-throw faders, comprehensive EQ, and multiple output buses, which are often cut-down versions of boards designed for multitrack tape studios or live mixing; and (2) rack-mountable "keyboard mixers," with tiny faders, limited or no EQ, a couple of effects sends, so-so specs, and one or two pairs of outputs. For the studio that works mostly with MIDI tracks, not multitrack tape, neither of these designs is ideal.

The Mackie Designs CR-1604 places itself squarely between these two extremes. It's primarily a 16 x 2 mixer, easily expandable to 32 or 48 inputs, with limited but useful EQ, and more effects sends than you've ever dreamed possible. Perfect for the MIDI studio, it can also fit neatly into an 8-track facility, and even has enough flexibility for broadcast or live sound. It's compact, but ergonomically designed, and although it's priced in the "keyboard mixer" range, its specs belong to a much more expensive category.

The "Mixer Mixer"

ROTATE THE POD, HAL

The CR-1604 is 19 inches wide (it's rack-mountable) and from 12 to 16 inches high, depending on how you set it up. The input and output jacks are on a repositionable "pod." For use on a table-top, the pod is mounted so the jacks point out the top of the mixer. It elevates the rear of the unit, sloping the main controls at a comfortable angle. For use in a rack, the pod can easily be rotated so that the jacks point out the back, which means no extra vertical clearance is necessary in the rack. You can even flip the pod so the jacks point forward, with an optional "Roto Pod" attachment. All of the 16 input channels have high-impedance, unbalanced line inputs, and six also have XLR balanced low-impedance mic inputs with phantom powering. An optional add-on 10-channel mic preamp is available. The main outputs are balanced, but are on 1/4-inch (tip-ring-sleeve) jacks so they can easily be used with unbalanced systems. Each channel has a fader, a pan knob, Solo and Mute switches, and three-band fixed-frequency EQ.

ON THE BUS

Each channel also has four send controls handling seven auxiliary buses. How does four get you seven? Well, the first send on each channel can be switched between an aux bus and a monitor bus, each with its own output. The third and fourth sends on each channel can be switched to access either aux buses 3 and 4 or 5 and 6. Since it's unlikely you'd ever want to use more than four different effects at a time on a particular input signal, this is a clever way of maximizing flexibility without using a zillion tiny knobs. There are four stereo aux returns, each with its own balance control and mono switch, and these returns, as a group, can be soloed. Finally, there are master left and right level faders, a solo/headphone fader, and a pair of 10-LED level meters displaying the range from -20 to +22.

GAIN, NO GAIN

Another unique feature of this mixer is that it's built around a "unity gain" structure. All the faders, sends, and returns have a "click" position at zero gain: the signal going out is at the same level as it was coming in. With a normal mixer, if you set all the faders to "0," you'll overload the bus amplifiers, but this mixer is designed with enough headroom that "0" can be the nominal operating setting. While most mixers make available only a few dB above "0" at each stage, the CR-1604

The CR-1604

gives you lots of room: the aux sends have 15 dB of extra gain available, the aux returns 20 dB, and the faders a healthy 22 dB. This design means the mixer is incredibly easy to set up, and there is no guesswork in arriving at the optimum control settings in terms of gain structure. A simple procedure for setting up the channels is detailed several times in the manual. When it's done correctly, you almost deliberately have to push the levels too hard to make this mixer clip.

UNDER THE SURFACE

There are some very nice touches in the mixer's design, which you discover as you spend some time with it. The Solo buttons are "in-place," which means they maintain the stereo placement of the signal. A very rude flashing light prevents you from forgetting you're in Solo mode. The Solo bus can be assigned either to the headphone jack, for live mixing, or to the main outputs. A Mute button turns off the main outputs but leaves the headphones on so you can hear count-ins before you start recording. Eight of the input channels have direct in/out jacks. If you insert a plug to the first "click," the channel signal is routed out of the jack without interrupting its path through the board, thus acting as a "mult." If you push the plug all the way in, the signal path is broken, so that a limiter, MIDI-controlled fader, or other outboard processor can be inserted. An eight-track tape deck, with its own tape/source monitor switching, can also fit here just fine. The main output buses have access jacks as well, for master limiting or EQ. Although it's mainly set up for stereo, the board actually has two pairs of outputs. The channel Mute buttons are in reality DPDT switches which send the channel's signal to a completely separate stereo bus, called "Alt." The Alt bus has its own outputs, and it can also be switched into the headphones with an "Alt Preview" button. The only thing preventing this from being considered a true four-channel board is that the Aux sends and returns cannot be used on the Alt bus.

THE MIXDOWN

The sonic performance of the CR-1604 is beyond reproach, and easily the best I've ever heard from any mixer anywhere near its price range. All that headroom is not, thankfully, at the expense of a raised noise floor: I never heard any noise at all from this mixer in all the projects I used it on, which included mixing a short film (dialogue and sequenced music), writing a group of library tracks, and editing some hard-disk audio. The sound is completely transparent, and in just the first day I could even smell my mixes becoming cleaner. The EQ points are a little different from what you might be used to: 80 Hz, 2.5 kHz, and 12 kHz. The bandwidths are wide, which means they're designed not for eliminating noise or accenting individual frequencies but for shaping the musical tone of a sound. In that regard, they make a lot of sense and sound quite smooth. The faders are small and very responsive, but they are solid, quiet, and stable feeling. There isn't much room for a "scribble strip" next to them for marking settings, and I wouldn't want to do a complex live TV mix with this board. However, for electronic music, in which the number of fader moves in a piece is usually small, they're perfectly acceptable. Each fader has an overload LED next to it — unfortunately, this is the only signal indicator on the channel, and I would have preferred an additional "-10" light or something like it. The aux sends work extremely well, with no crosstalk, and having so many of them means you can keep all those reverbs, delays, and limiters you've been collecting on line all the time, and forget about patch bays and submixers. There are no master send controls, which can be a little confusing, but the unity gain concept helps. The aux returns have a slight problem: maximum attentuation is about 60 dB, which means that they never quite shut all the way off. The input jacks are very "grabby," and it's hard to disconnect cables once they're in. However, it also means the chances of a cable falling out accidentally are very slim, so I consider this a good idea. Zeroing the board requires a special technique, which is unfortunately not described in the manual. The aux sends can all be turned off by sweeping your finger across each row, but they are so close together that turning off one row turns "on" the row above it. If you work from the bottom row upwards, however, you can get all of them zeroed in short order.

MORE CHANNELS!

You can gang two or three CR-1604s together, using an optional device called the "Mixer Mixer." This is essentially a jack box full of those same grabby jacks, with no controls of its own, that ties together the mixers' auxiliary and output buses. Mackie also sells a cord pack containing enough 36-inch patch cords to tie three CR-1604s together. With three mixers, you get 48 inputs, the same seven aux sends, and 12(!) stereo aux returns. The only problems are that you now have three sets of main output faders and three solo buses. The first can be solved with an optional "Remote Fader," a very smooth long-throw fader that acts as a single fader for all of the mixers simultaneously. It clamps onto the side of one of the mixers, and connects to the Mixer Mixer. The solo problem, however, is a little more complex. If you solo a channel on one mixer, the rest of the channels on that mixer will shut off, but those on the other mixer(s) will be unaffected. The company has a mod to solve this, but it requires soldering a shielded cable to the motherboard of each mixer (voiding the warranty). It works just fine, and hopefully they'll figure out a way to make it easier.

FINAL NOTES

The manual is brief, but good. Telephone technical support is excellent,

> For use in a rack, the pod can easily be rotated so that the jacks point out the back, which means no extra vertical clearance is necessary in the rack.

and there's an 800 number. The company seems to be actively interested in improving the product and is very open to suggestions from customers. The packing material and descriptive literature is very funny: clearly the product of people who like, and believe in, what they do. The Mackie CR-1604 is a remarkably well-thought-out and well-executed product, designed to fill an important niche in the mixer market. Neither too complex nor too simple, it offers a unique combination of features arranged like a good computer program: a straightforward user interface with a great deal of sophistication under the surface. If you have more processing gear than you know what to do with, and don't use a lot of corrective equalization, it's perfect. It also sounds great, and the price can't be beat. I recommend it highly.

—*Paul D. Lehrman*

Opcode Track Chart

EQ LAB REPORT

MANUFACTURER: Opcode, 3641 Haven, Suite A, Menlo Park, CA 94025; 415-369-8131.

APPLICATION: Provides graphics tools for designing and filling out track charts; prints out labels for cassettes, tape boxes and console faders; offers a real-time scrolling display.

SUMMARY: Streamlines the entry of session information, and then organizes and distributes it in ways that engineers are likely to find useful.

STRENGTHS: Track Chart is a cutting-edge product; expect to see its functions built into sequencing and hard-disk recording programs in the future and to see Track Chart itself grow into a comprehensive session documentation system.

WEAKNESSES: Doesn't provide for the inevitable patching of tape tracks to channel inputs whose numbers don't match; leaves large areas of session documentation unaddressed; and cassette J-cards are absent from the label-printing options.

PRICE: $179

EQ FREE LIT. #: 128

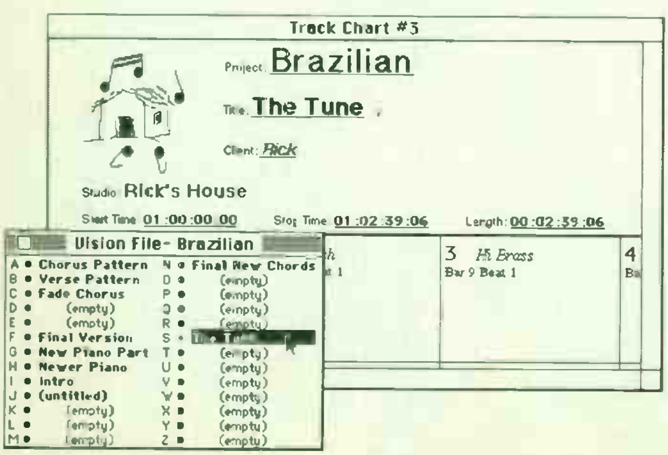
IN RECENT YEARS, THE COMPUTER has usurped the functions of just about every piece of gear in the recording studio. First the tape recorder was challenged by the sequencer. Then hard-disk recording threatened tape itself. Today effects, automation systems, and even the mixer may reside inside a CRT-adorned box filled with computer chips. What job is left for the computer to take over?

The assistant engineer's, of course.

Track Chart (Mac Plus or greater), a new Macintosh application from Opcode, does much of what an assistant is supposed to do (short of setting up microphones). It provides graphics tools for designing and filling out track charts. It prints out labels for cassettes, tape boxes and console faders. And it offers a real-time scrolling display, called a time line, that provides customized mixdown reminders (as well as crude MIDI sequencing capabilities). In short, Track Chart streamlines the entry of session information, and then organizes and distributes it in ways that engineers are likely to find useful.

TRACK-ITTY-YAK

Track Chart lets you design track chart documents to your own specifications. There are provisions for virtually any information you might expect to see on a track chart, including a source designation, which is an icon that indicates whether the track resides on tape, on hard disk, in a MIDI sequence, or is to be recorded live during the mixdown.



Adjacent tracks can be grouped by putting a bold line around them and assigning them a group name. If you enter information about tracks that haven't yet been recorded, it appears on the computer screen but not on printouts. And, nearly all of your page-design specs and general session information can be transported easily to track charts for other songs in the same project.

The handiest aspect of this part of the program is the ability to import track information from a sequence saved as a standard MIDI file (or, more directly, a sequence created using Opcode's Vision sequencer). Any sequence track number can be mapped to any tape track number, making it easy to match the order in which you dump sequenced tracks to tape, or simply to document virtual tracks.

TIME LINES

Track Chart's time line display provides a graphic overview of the contents of a multitrack reel. It depicts each track as a shaded line, much like a sequencer's track display. The purpose of the time line is to keep you apprised of what's coming up as you mix.

A time line can be inserted with tracks lined up either vertically (in columns) or horizontally (in rows). The vertical time line format is suitable for printing a hard copy. If you've set the track-width parameter properly, the resulting document will line up with the faders on your console.

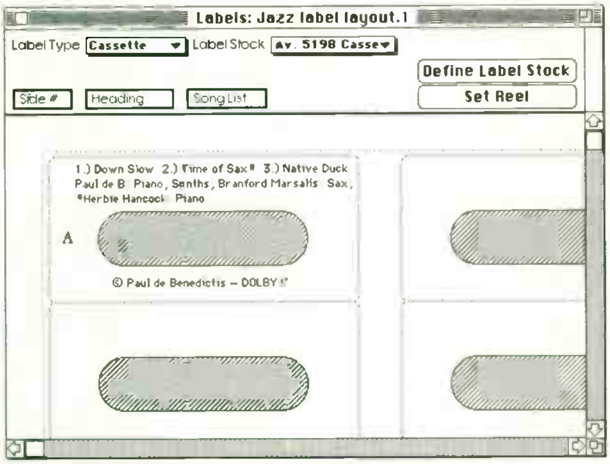
If your computer's monitor is free during the mixdown and you can position it to be visible, the horizontal format is the handier solution. The display scrolls as the music plays (synced either to MTC or MIDI clocks, or running freely), showing instrument entrances and exits as they occur.

The time line also offers animated streamers and punches, text messages, song lyrics, rehearsal marks and other sorts of marker information — virtually anything that might be used to cue mixdown moves. It also sends out MIDI events that can be sent

IN REVIEW

directly to synths or to MIDI-controlled automation devices such as JLCooper's MIDI Mute or the Niche ACM.

There are several ways to enter information about your tracks into a time line, the simplest being to import MIDI tracks from a MIDI file sequence. In this case, the sequence also provides tempo and time signature data, which are necessary for real-time scrolling. The lines representing tracks on tape may be either drawn by hand or "grabbed" on the fly by clicking the mouse as the tape plays to indicate entrances and exits. Owners of Opcode's Studio 3 or Studio 5 interfaces have a special option: You can route the recording, track by track, to the interface's audio input. As a track plays, the interface generates MIDI events, letting Track Chart know that the track is active. If you take the time to play through every track, the program will assemble automatically a time line corresponding to your multitrack.



Regardless of how information gets into the time line, it's all fully editable once it's there. In fact, any changes you make in the time line will appear in the corresponding track chart, and vice versa.

RUNNING WITH IT

Track Chart requires either Apple's MIDI Manager (an official Mac system extension) or Opcode's own OMS (Opcode MIDI System), or both, in order to receive or transmit MIDI data. They control the transmission and reception of MIDI to compatible applications within the computer. When you add them to your system you may run into compatibility problems with other MIDI programs. (My favorite sequencer, Master Tracks Pro 4 doesn't work with the Opcode MIDI driver.) Expect a few hitches getting up and running with them, but

they're nothing you can't work around. If it doesn't seem worth the trouble, keep in mind that you can use the lion's share of Track Chart without accessing its real-time MIDI functions.

Project studio owners may be happy to be able to derive tidy track charts from their MIDI sequences. Few of them, however, will be faced with such elaborate mixdowns that they truly need the program's time line functions, and fewer still are likely to want to tie up their Mac's RAM, processor and screen with Track Chart. On the other hand, larger studios lacking automation that can devote a computer to Track Chart during mixdowns, may well find the program invaluable.

Track Chart is in some ways a cutting-edge product and, as a result, it has its share of omissions and rough spots. For instance, it leaves large areas of session documentation unaddressed. Cassette J-cards are absent from the label-printing options (though the ability to enter a custom label size offers a viable workaround). Patchbay connections and console fader, send and EQ levels aren't dealt with in a direct, graphic manner, although they may be entered verbally as comments. Studio time logging and billing also come to mind. Opcode assures me I can expect to see some of these capabilities in the future.

You can use the lion's share of Track Chart without accessing its real-time MIDI functions.

We were most disappointed to note that Track Chart doesn't provide for the inevitable patching of tape tracks to channel inputs whose numbers don't match. An option to juggle tracks on the time line without affecting their locations on the corresponding track sheet would be helpful here. Similarly, the program only allows the remapping of sequence track numbers to tape track numbers before the sequence has been imported. A dynamic sequence-to-tape map would allow for the kind of flexibility that real-world sessions require.

We'll chalk up this sort of thing to the fact that Opcode and programmer Rick Johnston have done something new here. The fact that it's something useful means that refinements are inevitable. Expect to see Track Chart's functions built into sequencing and hard-disk recording programs in the future, and to see Track Chart grow into a comprehensive session documentation system.

—Ted Greenwald

Analog Entry, Stardate...


Photo by Ed Colver

Random notes from the Fagen Frontier, plus Audio Cross-Dressing, and Hawaiian War Drums

BY ROGER NICHOLS

Well, another couple of months have gone by and another vocal is done on Donald Fagen's new album. I went to my local stationery store to get one of those calendar planners for 1992 so I could start scheduling my time to finish the album. My choices were "Day at a Glance," "Week at a Glance" or "Month at a Glance." They do not yet make a version called "The Rest of Your Life at a Glance," which I think I may need before we get done.

At the level of acceptance that has been set for this project, we find that microseconds are tossed around instead of milliseconds, tempos have decimal points in them, tuning values are expressed in micro-cents and comments on the groove feeling are something like "It felt pretty good on that beat right there." My life outside the studio has started to imitate the reality inside the studio. I now only order egg salad sandwiches if the eggs came from Frizzle Chickens (a mutant chicken whose feathers are on backwards) and only wear clothes made from seedless cotton.

THE MAGIC FLUTE TRACK

In the middle of the Donald Fagen month last October in New York, Walter Becker and I took a one-week break from the Donald's project to record a couple more jazz albums. One of the artists is a piano player named Dave Kikowski. The other artist is a flute player named Jeremy Steig. There was only one fly in the ointment. Jeremy is an amazing flute player. He is also, as it turned out, an amazing negotiator. It says in his recording contract with the label that his flute will be recorded analog. Walter and I could do whatever we liked to the rest of the musicians (recording-wise), but we must keep the flute analog.

Our first notion was just to record the whole band on 24-track analog with Dolby SR, but just the mere mention of actually recording analog in mixed company caused three people to leave of embarrassment. Besides that, the record company had already printed the little "DDD" on the back of the CD booklet. So, we locked up a 24-track analog machine and a Sony 3324 digital machine. We printed all of the other instruments to the digital machine, while recording only the flute on the analog machine.

At the last minute, Walter convinced Jeremy that the sound of the flute was just fine on the digital machine. Jeremy conceded and we went on with the recording. Jeremy's request was a valid one, I guess. But I would definitely put that request in the same category as a Scotsman who preferred to wear kilts.

Walter explained it very well when he called this switching back and forth between formats "audio cross-dressing." (Analog being the female analogy and digital being male.) If you're mostly an analog person, then nobody cares much if every once in a while you dabble in a little of the digital domain. But if you're a "digital-kind-of-guy," people point at you and laugh if they catch you messing with those frilly analog things. I am just glad that Jeremy didn't like the sound of wire recordings or the optical tracks on film. Imagine using a soldering iron to do edits or waiting for the film to come back from Fotomat between playbacks.

BACK TO PARADISE . . .

So now it's the beginning of February, 1992. We're in Hawaii (Donald, Walter and myself) at Walter's studio working our little fingers to the bone, slaving over a hot 48 track. We have all but one of the tunes on tape. The last one is going to be printed tomorrow. Next week we're going to have a sacrificial ceremony and Donald is going to offer a couple of his digital delays to the Hawaiian God of Increments by throwing them off a cliff into the ocean. Walter is going to videotape the ceremony for posterity.

Leroy Clouden is coming from New York to play real drums to replace the sequenced drums on tape. The vocals are going just fine and we should be mixing by September. Hang on a second, I think my tick is coming back, back, back.

As I have mentioned in previous columns, we do have the somewhat dubious distinction of being able to throw technology at any problems we come up with during these Fagen recordings. The never ending quest for drum replacement techniques is one of them. I keep looking for a machine that will replace Wendel, but so far, nothing. The problem with Wendel is that it's old technology. It's based on a CompuPro S-100 computer that you can no longer get parts for, and the operating system is CPM-86. Try calling tech support on that one. When I built it in 1981, one megabyte of memory filled the whole computer and it cost $12,000. (I know I've mentioned this fact before, but I'll keep reminding you every so often.) A couple of memory chips have died, and replacements can't be found.

Alesis just came out with the D-4,

continued on page 94

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