EXCLUSIVE REVIEWS: ALLEN & HEATH GS3V CONSOLE • AKAI CD3000 CD-ROM PLAYER

EQ LIVE: DESIGN-YOUR-OWN LOUDSPEAKERS

EQ

THE PROJECT RECORDING & SOUND MAGAZINE

APRIL 1994

MULTIMEDIA AUDIO REPORT

RAMONES ON TOUR (What Else is new?)

THE AGE OF THE HYBRID STUDIO

PC SOUND CARD BUYING TIPS

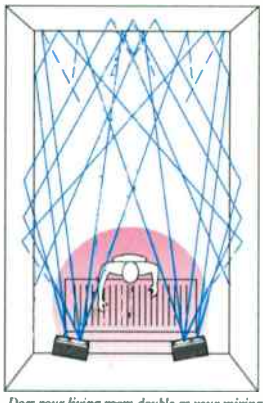
RAY ALLEN: HOW TO MAKE THE MULTMEDIA MOVE

The Truth From

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

Room For Improvement

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



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

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

The Truth From Top To Bottom

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

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

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

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

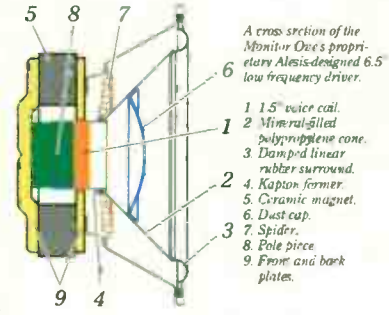
Left To Right

Power To The People

High power handling is usually reserved for the big boys. While most near field monitors average around 60 watt capability, the Monitor One handles 120 watts of continuous program and 200 watt peaks...over twice the power. Also, its 4 ohm load impedance allows most reference amplifiers (like the Alesis RA-100™) to deliver more power to the Monitor One than they can to 8 ohm speakers. That means the Monitor One provides higher output, more power handling capability, and sounds cleaner at high sound pressure levels.
If you like to mix loud, you can.

The Engine

Our proprietary 6.5" low frequency driver has a special mineral-filled polypropylene cone for stability and a 1.5" voice coil wound on a high-temperature Kapton former, ensuring your woofer's longevity.
Our highly durable 1" diameter high frequency driver is ferrofluid cooled (costly, but it's the best way to cool a tweeter), to prevent heat expansion of the voice coil which inevitably leads to loss of amplitude and high frequency response. Combined, these two specially formulated drivers deliver an incredibly accurate, unhyped frequency response from 45 Hz to 18 kHz, ±3 dB. The five-way binding posts provide solid connection, both electronic and mechanical. We even coated the Monitor One with a non slip rubber textured laminate so when your studio starts rockin', the speakers stay put. Plus, it's fun to touch.



A cross section of the Monitor One's proprietary Alesis-designed 6.5" low frequency driver.

1. 1.5" voice coil.
2. Mineral-filled polypropylene cone.
3. Damped linear rubber surround.
4. Kapton former.
5. Ceramic magnet.
6. Dust cap.
7. Spider.
8. Pole piece.
9. Front and back plates.

The Monitor One's five-way binding posts accept even extra-large monster wire, banana plugs and spade lugs. Hookup is fast, easy and reliable

The New Alesis Monitor One™

You don't design good speakers by trying hard. It takes years and years of experience and special talents that only a few possess. Our acoustic engineers are the best in the business. With over forty years of combined experience, they've been responsible for some of the biggest breakthroughs in loudspeaker and system design. The Monitor One could be their crowning achievement. They're the only speakers we recommend to sit on top of the Alesis Dream Studio™.

See your Authorized Alesis Dealer and pick up a pair of Monitor Ones. Left to right, top to bottom, they're the only speakers you want in *your* field.

The Monitor One is the speaker for the Alesis Dream Studio™. Need more information about the Alesis Monitoring System? Call 1-800-5-ALESIS. See your Authorized Alesis Dealer.
Monitor One, SuperPort, RA-100 and the Alesis Dream Studio are trademarks of Alesis Corporation.
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ALESIS
STUDIO ELECTRONICS

CIRCLE 02 ON FREE INFO CARD

EQ

PROJECT RECORDING
& SOUND TECHNIQUES
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FEATURES

EQ LIVE

TECHNIQUES / WORKSHOPS

COLUMNS / DEPARTMENTS

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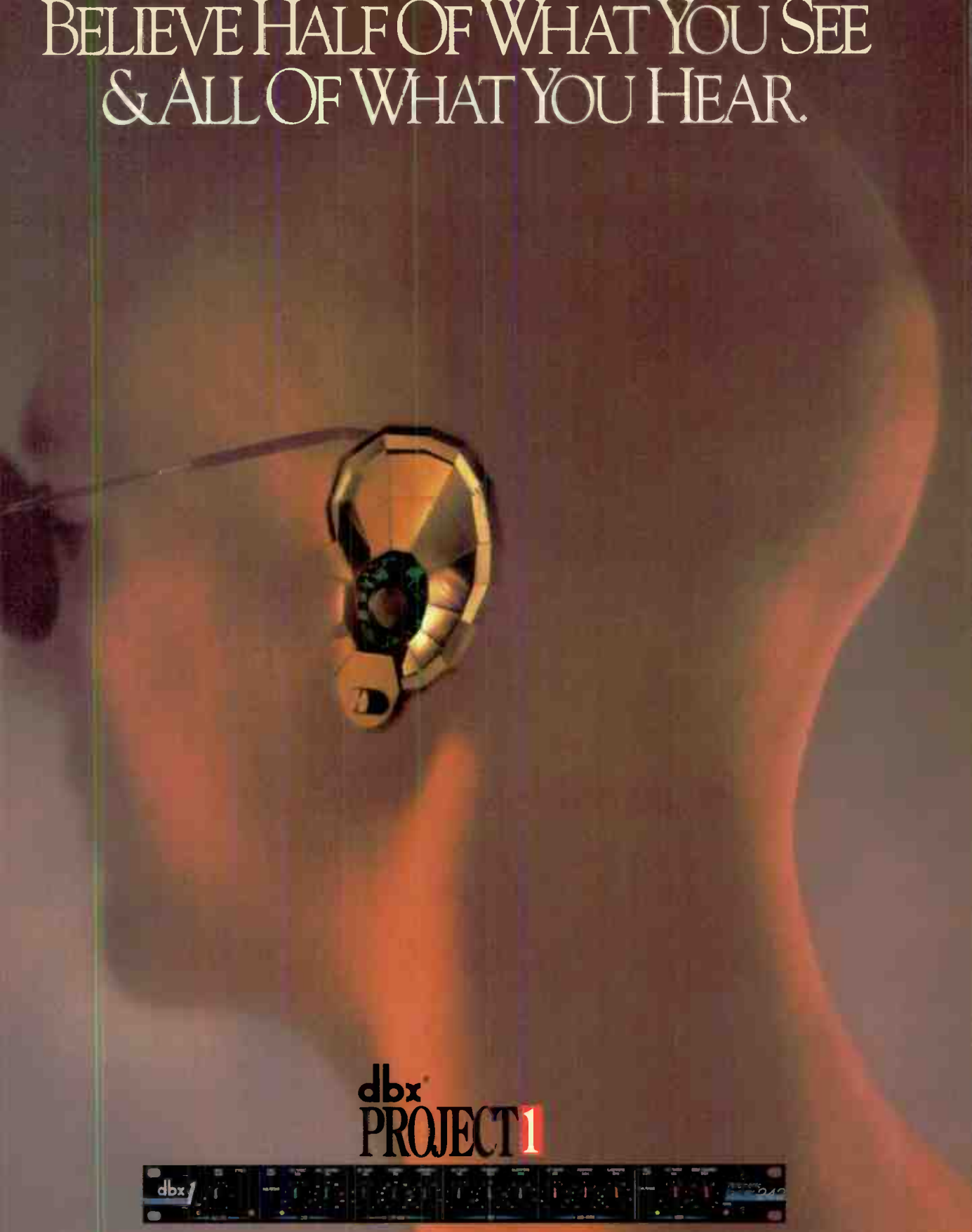
Cover: Murray Allen by Steve Jennings.

BELIEVE HALF OF WHAT YOU SEE & ALL OF WHAT YOU HEAR.



dbx®
PROJECT 1

You hear sound curve as the notes take shape. You've heard our reputation in the business. We build great compressors. Chances are, you've got one in your rig now. We're introducing a Parametric Equalizer built and priced to keep our rep and your music intact. Keep an ear out for it.

THE 242 NEW PARAMETRIC EQ. SHAPING YOUR SOUND. CIRCLE 15 ON FREE INFO CARD

EQ

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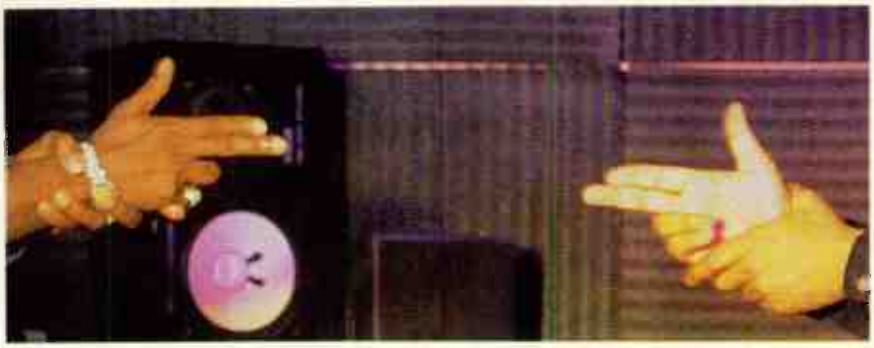
LETTERS TO EQ



WHAT'S THE POINT?: The fingers in question.

WAVE RIDER

As a user of the Waves Q10 software, I found the review by Howard Massey in your Feb. '94 issue to be a very good general overview of this product. I would only add the following footnotes:

While it is true that the parametric equalizer in Sound Designer 2 offers control over only one band (stereo), it should be noted that its graphic equalizer actually has five bands (ten in mono) that have parametric capability, all parameters being continuously variable, including bandwidth. The drawback is that only gain variations can be previewed in real time.

For me, perhaps the best features of Q10 are its stereo gain controls and metering; here it offers something Sound Designer 2 has always needed. Its faders can be used to alter stereo balance with or without [having the] equalization activated. And it will preview a whole file with EQ applied and show available headroom (in dB), as well as clipping, if any.

On the subject of the "bug" that was mentioned, I have found system hangups can be avoided only if you remember to turn off the Preview mode before making any changes whatsoever to any file.

Finally, I have one colleague who worries about using any DSP, such as Q10, that is not dithered.

Doug Pomeroy
Pomeroy Audio
Brooklyn, NY

HOW SWEET IT ISN'T

What are Daddy-O and Richie Hererra communicating by posing with mock guns aimed at each other in *EQ*'s February "Room with a VU"? Are they:

a) Two grown men showing their fun-loving nature by playing "Cowboys and Indians"?; b) Striking a cool pose confirming their status as members in good standing of the "hood" where irresponsible gun use has become a substitute for manhood?; c) Just a couple of talented but idiotic musicians making light of the violence that has produced an epidemic of death and fear among their young fans?

Ask any one of the thousands of parents who has lost a child to a drive-by shooting how funny, hip, or entertaining *they* think the photo you selected for the article is, and you won't get any chuckles or hear reactions like, "Wow, aren't they cool!"

The photo is insensitive at best, and grossly irresponsible in the context of what's happening on America's streets right now. Violence that kills innocent kids, adults, and yes, people who own project studios is not funny, hip, or entertaining — it's disgusting. As are those who make money glamorizing it.

I for one hope the photographer, Daddy-O, Hererra, and whoever at *EQ* chose to run the photo don't themselves lose a loved one to the senseless and appalling violence so casually made light of in the photo, and that they'll use better judgment in the future.

D.K. Sweet
San Jose, California

REP-PREHENSIBLE

I am a manufacturers representative for both consumer and professional electronics in the audio industry. I

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.

Accept no substitute (there isn't one)

There is no other console like BIG in the world. For a moderate price, we deliver a comprehensive audio system with full-specification AMEK automation and Recall.

BIG's audio architecture has complete facilities for all types of creative audio production.

☞ Dual-path modules in the 28 or 44 position chassis provide 56 or 88 inputs.

☞ The standard console includes 4 stereo line inputs and 4 stereo FX inputs.

☞ 12 busses selected to 24 outputs, Direct outs and 8 Aux sends provide massive routing flexibility.

☞ EQ is similar to the famous and much-loved AMEK ANGELA 4-band device.

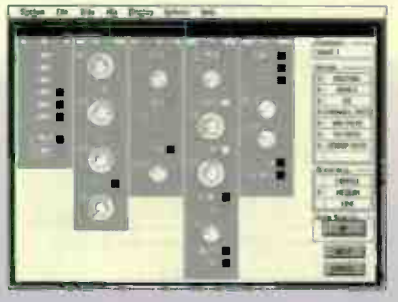
AMEK SUPERTRUE automation is a powerful part of BIG's equipment and is one of the world's leading systems with a user base of over 600 installations.

☞ SUPERTRUE gives complete SMPTE-locked control over faders and mutes; up to 10 VCA groups can be configured.

☞ Extensive automated solo modes allow additive and subtractive mixing using mutes and solos in various combinations.

☞ SUPERTRUE has a range of on- and off-line editing systems including the forthcoming Mix Processor which allows Mix Data to be Spliced, Merged, Shifted, Erased or Extracted and Channel Data to be Swapped, Copied or Trimmed.

☞ The Cue List allows a range of Events both inside the console and externally (via MIDI) to be triggered from time code.



Unlike any competing product, BIG has a Recall system which allows you to store the positions of module controls. This means you can reset the console surface at some later time, allowing you to recreate and modify mixes as required using the graphic screen display or the unique Voice Prompt, which talks you through the console.

Finally, AMEK VIRTUAL DYNAMICS is an option. This proprietary software-based envelope shaping system allows you to select a digitally-controlled Dynamics device, such as a Compressor, Gate, Limiter, Expander or Autopanner to each VCA fader from a menu.

BIG by Langley

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AMEK

CIRCLE 06 ON FREE INFO CARD

couldn't begin to figure out how many consumer and trade publications I've read in the past 18 years — some excellent and some not so excellent!

Tonight I decided to look through your magazine. In the February issue I came across "Room with a VU: Make Room for Daddy-O." I felt pure disgust — almost the same feeling I had while driving in my neighborhood of Northridge, CA! I think the photo taken by Julian Jaime is without a doubt the worst thing I've seen in any publication!

Are these the kinds of "signals" you want to send out to your readers? What are you trying to say? Janet Reno will not be happy with this!

You screwed up. Very poor taste. Herbie Hancock would have probably liked to be associated with another issue!

Mark Mayer
Northridge, CA

SO SHOOT US

Violence, violence, violence. Big Deal. Yawn. Violence is America's unhealthy way of life these days. Drive-by shootings, murders — you name it, it happens every day. Yeah, I've seen it all. *So what.* I've become "comfortably numb" by it all. Or so it seemed.

But then I found myself staring at your photo of "Daddy-O" (you know the one with two men pointing fake "guns" at each other) and it flat out pissed me off. I've got to say that "Daddy-O" blows and so does your editorial staff for printing that photo of those stupid idiots.

Wake up and smell the blood *EQ* Magazine. People who use guns to terrorize our society are the lowest form of life on the planet and not to be celebrated and idolized. Maybe they don't care and maybe you don't care, but *somebody should care!*

P.S. I'd have been more impressed if those two had been playing with real instruments rather than "air" guns.

Daniel Evans Clawson
ClawsOn Productions
Louisville, CO

[With all the media coverage about the portrayal of violence, one can easily lose sight of the real problem — violence. Real violence is beyond the purview of EQ, *our topic is the use of technology to help create musical art. Artists will reflect reality, and unfortunately we live in a violent society. So let's hope that the next time our critics fire their criticisms, they look more carefully before they shoot. Take aim at the real problem and not the chroniclers of the problem.*

Mark Alhadeff
EQ Picture Picker]

TAPE TIP

Here's a tip for all Hi8 tape users. Before inserting a tape make, sure the cassette hood is slightly loose. If it's tight and won't move, do not insert it into the machine. We did, and it took a lot of time and pain to remove the cassette. The hidden code was 5-EAR-10, which does not exist in the manual, and TASCAM never knew about it. The hood can be unstuck with a little

IT'S A SMALL PRICE TO PAY FOR ENVIRONMENTAL PROTECTION.

The good news is you're booked for Saturday night. The bad news is you couldn't pick a worse place to perform.

Low ceilings. Hard surfaces. And more dead spots than you'd care to name.

Fortunately you're in better shape than you think. Assuming you bring along this small environmental wonder. Yamaha's latest stereo EQ. the affordable GQ2015.

Featuring a compact rack-mount stereo configuration. with 15 EQ bands at 2/3-octave intervals from 25 Hz to 16 kHz, the GQ2015 gives you just about all the control you'll ever need.

Plus, with switchable EQ gain – up to ±12dB of cut or boost – it gives you more control to contour the sound the way you like it.

You'll also find THD of less than 0.05% and a noise level below -94 dB – in other words – professional performance at an affordable price.

And to counter the low-end boominess you might find in some of those nightmare spaces, the GQ2015 features a switchable high pass filter that can tighten up your bass at the touch of a button.

It also accepts both TRS and XLR jacks and has switchable levels to balance your I/O with all kinds of equipment.

So whether you're setting the stage for a one night stand or for considerably longer, the GQ2015 can help you save the environment. Not to mention reduce stage rumble, control feedback, sibilence and microphone popping.

Stop by your nearest Yamaha dealer for a demonstration today.

Or call *1-800-937-7171 Ext. 330* for more information.

CIRCLE 83 ON FREE INFO CARD

YAMAHA

force. If all else fails, try a drop on the floor.

Ed Lopata
Moss Hollow Recording
Marlboro, VT

WIRE WRANGLER

Re: EQ&A, February, 1994 issue of your magazine, and the answer of Mr. Tim S. O'Meara (director of engineering, Switchcraft) to Jorge Chinique's questions about the proper method of wiring patchbays:

Mr. O'Meara wrongly implies that there is only one way to wire up Send/Return 1/4-inch TRS jacks. Sadly — especially now that after only a dozen years or so, our industry has settled down on the Pin 2 Hot issue — we find ourselves tangled in yet another wiring controversy.

Send/Return loops done using only a single 1/4-inch TRS jack are popularly wired in two exactly opposite manners: There is the Tip=Send/Ring=Return school (the 3Rs: ring, right, return group) and there is the Tip=Return/Ring=Send school (the other group). Technically speaking, equally logical arguments exist for both designs. Objectively it could be done either way. Unfortunately, it is done both ways.

Your readers need to be warned that every time they use a single 1/4-inch Send/Receive effects loop, they must check which wiring method is used. Failure to do so guarantees a very long night.

I do not have a list of just who does what (in fact, maybe that would be a very good project for one of your future columns), but a quick perusal of our files produced the following list of manufacturers using the Tip=Send/Ring=Return convention: Alesis, Soundcraft, Peavey, Tascam, Mackie, Yamaha, and Rane.

Also see page 294 of Gary Davis & Ralph Jones' book *Sound Reinforcement Handbook, 2d ed.* (Yamaha, 1989)

Dennis Bohn
V.P. R&D
Rane Corporation

[Tim J. O'Meara responds: In regards to the two popular methods of wiring a send/return TRS 1/4-inch jack, I fully agree. Although our EQ&A response made mention of the fact that "...a variety of connection schemes are used by various manufacturers...," we failed to be explicit in this area of potential confusion. I support Mr. Bohn's urging that one "must check which wiring method is used."] EQ

TRIBUTE TO A FRIEND

There are several things everyone should know about Len Feldman who died of cancer in February. Len knew audio. Len lived the audio business. Len was always one step ahead of the cutting edge. Len seemed to be everywhere, know everyone, and have time for everything.

Most important to me and dozens of other younger editorial types, Len was always there to advise, support, and set a standard. That's why I was so honored when he agreed to become the Basics columnist for our magazine. And that's why we extend the love of the entire *EQ* and PSN Publications staff to Len's family and his wife Rayma whose behind the scene copyediting skills have contributed to the quality of the magazine you're now reading.

There are numerous eulogies to Len now being published. (Look for David Lander's article in an upcoming *Audio* Magazine). But none of them have mentioned the significance of Len Feldman's regular audio columns in the 1970s issues of *Rolling Stone* and how they influenced a young generation of stereo buyers. To some of us these articles were more important than the cover stories (other than those about the Grateful Dead.) To others, they were the first introduction to the reality that great sound was possible, affordable and — in many ways — one of the great sensory experiences of life. Thank you Len. —*Martin Porter*

1500 SERIES

WHATEVER YOU DO — DON'T BUY THE WRONG MIXER.

Especially if you're doing multitrack recording — whether digital or analog. Fact is, a mixer that's not specifically configured with the features essential for multitrack recording just isn't a recording mixer. Bottom line is, general purpose mixers make multitrack recording a nightmare.

You see, mixers that aren't designed and engineered for multitrack recording will torture you with the endless hassle of patching and repatching — every time you track, overdub or mixdown. It's frustrating, wastes valuable time and leaves you tangled in cable.

So before you choose a mixer for your studio — be sure it has the features of a dedicated recording mixer.

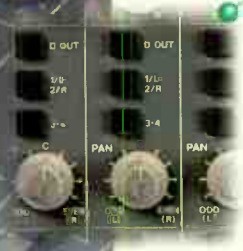
IT'S NOT A RECORDING MIXER IF IT DOESN'T HAVE THESE FEATURES.

MULTITRACK DECK CONFIGURATION

If you don't have dedicated inputs and outputs for your 8-track deck, where do you plug it in? Without this basic recording configuration you'll be repatching day and night and you won't be able to record on 8 tracks at once. With these inputs, tape monitoring is as simple as pressing a switch. Also, because the TASCAM M1500 is a true 4-buss mixer, you can mix any combination of your input signals to any of the 4 output busses directly to tape.

SWEEPABLE MIDRANGE EQ

Ask for it. Because when it comes time to tailor your sound, you need the flexibility where the action is — in the midrange. The M1500's sweepable midrange lets you isolate specific mid frequencies allowing you to make the subtle tonal corrections you want.

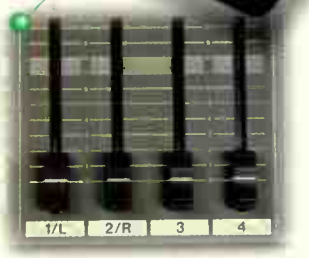


DIRECT OUT AND GROUP OUT ASSIGNMENT SWITCHES

You gotta have these. Because without them you can't directly send a single input to tape, or record several inputs to one track. But with them, assign your inputs anywhere by pressing a few switches. Best part is, you'll never have to refer to any complex patch diagrams.

ELABORATE MONITORING

In a recording environment you need to hear what's going through your board at all times. With the M1500's comprehensive monitoring matrix you are able to hear any sound source at any time — inputs, tape, AUX sends, anything — it's your choice, just press a switch.



IN-LINE MONITORING

A sure sign of a recording mixer. This lets you monitor your tape tracks at any time without sacrificing an input channel. Just press a switch. With the M1500's dual section not only can you monitor tape tracks, it can be used for additional effects sends, or to double your inputs for virtual tracking at mixdown. And do any of this by flipping a switch.

TRUE TRANSPARENCY AND LOW NOISE

In recording, your signal goes through the mixer several times. And each time it goes through, it is important not to lose or gain anything. Especially an identifiable "mixer sound." Test any mixer for its transparency. Take any signal and bounce it 3 or 4 times on your favorite digital recorder. With the truly transparent M1500, you'd be hard pressed to differentiate between the bounced tracks and the original signal.

At TASCAM, we've been making multitrack recording equipment for more than 20 years. We pack that experience into every mixer we make — and we make more recording mixers than any other company in the world.

For our M1500 Series of recording mixers, the result is an affordable mixing console configured for 8-track recording. A truly transparent mixer that makes tracking, overdubbing, and mixdowns easy. An extraordinarily flexible console loaded with the features and specs you'd expect on consoles costing thousands more.

But the M1500 Series of recording mixers are priced less than many general purpose mixers on the market. They're available in a 16-channel/32-input tabletop version (M1516) and a compact rack mountable 8-channel/16-input version (M1508). So if you're involved in digital or analog 8-track recording, you've just found the best recording console value in the industry.

Get your hands on a true recording mixer today: the TASCAM M1500 Series. There's one waiting for you at your authorized TASCAM dealer. Go ahead — test it and play with it. It's your next recording mixer.

TASCAM.
Take advantage of our experience

CIRCLE 39 ON FREE INFO CARD

GOING THROUGH CHANNELS

Q *Regarding the August '93 EQ Workshop, "Building a Balanced Line Driver," I am planning to build a multichannel version of the device. Do I need a power supply for each channel, or will one power supply suffice? How many channels can be supplied by a single power supply? Is there a schematic for building such a power supply?*

Vince Relihan
Schenectady, NY

A You do not need a power supply for each preamp. You do, however, need a power supply with sufficient current capacity to handle the total number of line drivers. According to the spec sheet, the worst case current draw for each SSM-2142 IC is 7 mA with no loading. Under absolute worst case operating conditions — delivering full voltage into a 600 ohm line — the current drain is approximately 50 mA. Therefore, for eight line drivers, you would need 400 mA of total current capacity.

In the real world, it is highly unlikely that this much current would be drawn at one time. Therefore, a power supply that can deliver 500 mA (half an amp) would be more than adequate to provide a large measure of reserve power. A 1-amp supply could comfortably power 16 or more line drivers.

Note that most power supplies have overload protection, so if you decided to be a little less conservative and run more than eight line drivers from a 500-mA supply, no physical damage would occur. However, if the power supply shuts down often, this indicates that you need a heftier supply.

Suitable power supply kits are available from PAIA Electronics, 3200 Teakwood Lane, Edmund, OK 73013 (Tel: 405-340-6300). The BPS-15 ($29.95) provides ±15 volts at 500 mA, and the rack-mountable 7700/K power supply kit ($59.95; front panel $23.95) supplies from ±9 volts to ±15 volts at 1 amp. Contact PAIA for ordering and shipping information. Note that these power supplies are also suitable for the various projects I've published in *EQ*, such as the Tone Tweaker.

Craig Anderton
West Coast Editor
EQ

A REAL PLAYER

Q *I need to get a cassette deck for making copies of tapes, and wondered if you could recommend the types of features that are most important for studio use.*

David Solinski
Des Plaines, IL

A Although there are professional cassette decks designed for studio use, a good consumer-oriented model will often do the job just fine for less dollars. Not all decks are equal, however, and I suggest you look for the following:

Dolby HX Pro: This circuit increases the apparent high-frequency headroom and reduces tape saturation at high frequencies. It does not require decoding, so it's compatible with any tape deck for playback.

Variable front-panel bias: This is a hard feature to find, but well worth it. Being able to optimize bias gives the best compromise between noise, distortion, and high-frequency response for whatever tape you use. Decks with fixed bias require that you find a particular type of tape that works well with the deck and keep using it.

Monitor from tape: A few desks include the functional equivalent of a third head so you can monitor from the tape, not just the source. This simplifies level-setting and lets you know whether the signal is being properly recorded or not.

Non-auto reverse: Since you want something reliable in the studio, avoid auto-reverse decks — they're mechanically more complex than the standard kind and, in my experience, more prone to breakdowns.

Noise reduction: Although it's nice to have Dolby B, C, and S capability, remember to always make copies without using any noise reduction. Unless the playback deck noise-reduction circuitry is properly calibrated, you could get mistracking errors.

Time counter: Check whether the counter can display elapsed time as well as arbitrary numbers. The elapsed time feature generally will not be linked to the tape (i.e., if you rewind, the counter won't follow), but as long as you're in play mode, it should keep relatively accurate track of the time.

Personally, I use a Teac V-670 because it meets all the above criteria (except Dolby S noise reduction), and it has worked very well for me. Granted the spotlight is on DAT, DCC, and MiniDisc, but a properly biased cassette deck with quality tape can do a very good job of recording and playing back music.

Craig Anderton
West Coast Editor
EQ

HUM BUGGED

Q *I record classical music concerts live using a DAT machine and an outboard mic preamp (it uses an NE-5534 op amp and Shure A95 input transformer). My battery-operated electret condenser DAK mics have hard-wired unbalanced cables (16-foot with 1/4-inch phone jacks). Although I try to record at the end of the 16-foot mic cables because the resultant recordings are quiet, I usually have to run 45 extra feet of unbalanced cable to the preamp, which often causes a hum on the recording. I'd like to build a pair of in-line balancing transformers between the mic cable and the extension (the preamp is looking for a balanced line with XLR connectors). How do I do this?*

Danny Ferguson
Springfield, NJ

A A 60-foot run of unbalanced microphone cable is almost certain to cause the kind of hum pickup you experience. The unbalanced cable should be converted to a balanced one as near to the microphone as is conveniently possible, perhaps at the bottom of the mic stand. The circuitry for this converter (see the schematic on page 14) can easily be built in a small metal box.

According to the information we

Gain

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Laser-age manufacturing techniques keep the cost of ProForce mics low. But their sound and ruggedness are uncompromised Sennheiser. Grab a ProForce mic... and gain complete control of your performance.

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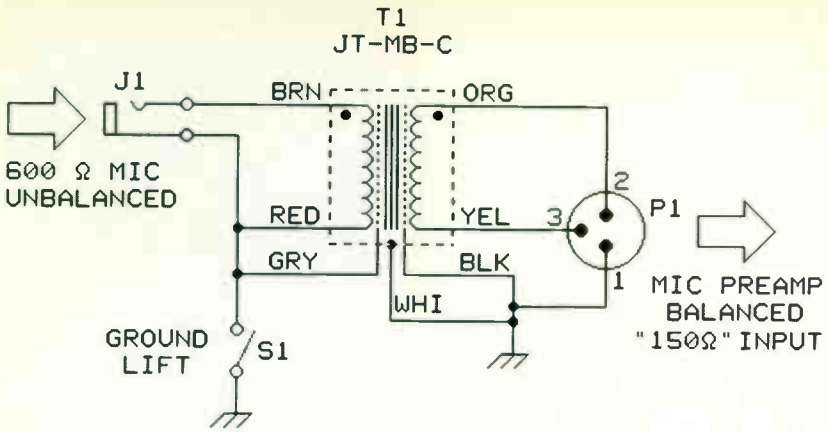
CIRCLE 50 ON FREE INFO CARD



J1 SLEEVE MUST BE INSULATED FROM THE CHASSIS IF GROUND LIFT SWITCH IS TO OPERATE.

obtained from the supplier of your microphones, their output impedance is 600 ohms ±20 percent. The Jensen JT-MB-C transformer, operating from this source impedance and feeding a 150-ohm mic preamp input (whose *actual* input impedance is about 1 kohm), has a frequency response that is flat ±0.2 dB from 30 Hz–20 kHz. The transformer's distortion, under 0.05 percent at 30 Hz, is almost entirely 3rd harmonic, and roughly halves every octave above 30 Hz until it becomes unmeasurable.

Most relevant to your problem, however, is that this transformer has double Faraday (electrostatic) shields for very high common-mode rejection. It prevents the hum picked up by the microphone body and unbalanced cable (common-mode) from being converted to signal (normal-mode) on the balanced output cable. Of course, the output cable should be a good-quality shielded twisted-pair type.

The "ground lift" switch should normally be closed, but under certain conditions, noise can be reduced with it opened. For example, the mic preamp's chassis (and input shield) may be "floating" at up to half the AC line voltage with respect to the local "earth ground." This causes a small AG current — which increases if someone touches the microphone — to flow in the shield of the input cable; this adds hum or buzz because this shield also carries the signal. Opening the ground lift switch interrupts this AC path.

Bill Whitlock
President
Jensen Transformers, Inc.

This is where your questions get answered. Send your query with your name and address to:
EQ Editorial Offices,
939 Port Washington Blvd.,
Port Washington, NY 11050
Fax: 516-767-1745
AOL: MPANDA

THE CLOSER YOU LOOK, THE BETTER WE SOUND.

When you blow away the hype surrounding today's compact mixer market, it still comes down to this. The board that delivers the most flexibility along with the best sound *wins*.

We started with 22 inputs because that's what you need in today's input-hungry world. Then we added 4 Buss capability plus 4 Stereo Returns, 6 Aux Sends, PFL and *true* In Place Solo for unparalleled flexibility in all kinds of mixing situations.

MPL2242. EVERYTHING YOU EVER NEEDED IN A COMPACT MIXER.

MPL 2242 with optional side panels

The exceptional sound of the 2242 is based on its musical 4-Band EQ, wide frequency response, transparent audio path and 5dB more overall gain than anything in its class.

Don't take our word for it. Take a close look at the MPL 2242 and you'll see why it's fast becoming the mixer of choice for discriminating recordists and live engineers who need *more* than the accepted standard.

For additional information about the MPL 2242 rackmount mixer and the full line of Samson Audio products, please write to Samson Technologies Corp., P.O. Box 9068, Hicksville, NY 11802-9068 or call toll free (1-800-328-2882).

SAMSON AUDIO

CIRCLE 31 ON FREE INFO CARD

EQ PRODUCT VIEWS

WINNER TUBE

Peavey's new PVM™ T9000 is a tube microphone that features a self-polarized condenser capsule coupled with a vacuum tube preamplifier. It offers smooth extended-range frequency response and a uniform cardioid pattern. The PVM T9000 easily handles SPLs of up to 137 dB and includes a –10 dB attenuator and 80-Hz low-cut filter switches. A unique shock-suspension system incorporates a finned heat sink to help dissipate tube filament heat. The PVM T9000 comes complete with shock suspension, AC power supply, 25-foot special cable, and case. For more information, contact Peavey, 711 A Street, Meridian, MS 39301. Tel: 601-483-5365. Circle EQ free lit. #101.

TRACK IN A BOX

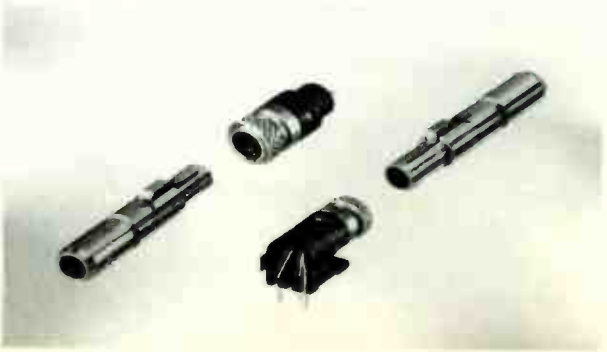
Vestax has introduced the HDR-6 and HDR-4 6-track and 4-track hard-disk digital multitrack recorders. The HDR series come standard with substantial storage capacity. Another regular IDE interface for an additional hard drive is provided for increased capacity. The HDR has a digital mixer with 3-band EQ and 4 AUX sends (pre- and post-fader configurable) and returns built in. The tracks can be merged and tracked down digitally without any external mixer. The HDR also features auto punch in and out. The HDR-4 and HDR-6 retail for $1880 and $2300, respectively. For more information, contact Vestax, 2860 Cordella Road, Suite 120, Fairfield, CA 94585. Tel: 707-427-1920. Circle EQ free lit. #102.

PEAK PERFORMANCE

The new PL-2 from Rocksonics is a dual/stereo peak limiter. Frequency response is 20 Hz–20 kHz ±0.5 dB with 0.05% THD. This compact unit is easy to operate via its front-panel controls. The hard-wired Bypass switch completely removes the unit from the signal path when it's not needed. The Threshold control sets the absolute peak output level. A Link switch engages a special stereo crossmatrix circuit to link the A and B channels for stereo processing. The PL-2 accepts and sends nominal –10 dBv in/out levels. For further information, contact Rocksonics, P.O. Box 442, Los Alamitos, CA 90720. Tel: 714-229-0840. Circle EQ free lit. #103.

EASY CONNECTION

Neutrik has released its "XY" Series IDC XLR connector. No soldering or tools are required to assemble this connector with a balanced cable. The IDC contacts (for contacts 2 and 3) are constructed as a double U-contact element, providing four gastight and reliable contact points. These contact points accept a range of wire cross sections of AWG-24–AWG-26. Shield contact is not IDC. Housings, inserts, chucks, and boots are compatible with Neutrik's popular X Series. For more info, contact Neutrik, 195 Lehigh Avenue, Lakewood, NJ 08701. Tel: 908-901-9488. Circle EQ free lit. #104.



ON THE MARK

Tannoy has finally released the *EQ* Blue Ribbon-winning Mark II Series of PBM compact reference monitors. The PBM 5.5 II, PBM 6.5 II, and PBM 8 II are the smallest and most affordable studio monitors the company makes. The Mark II Series uses injection-molded core materials rather than vacuum-molding. These polyolefin low-frequency cones are mica impregnated for high resolution and enhanced endurance. In addition, the Mark II Series features high-grade minimalist crossover topology and the speaker elements are suspended with a single roll of nitril rubber. The PBM 5.5 II retails for $350/pair, the PBM 6.5 II is $450/pair, and the PBM 8 II is $795/pair. For more information, contact Tannoy, 300 Gage Ave., Kitchener, Ontario, Canada N2M 2C8. Tel: 519-745-1158. Circle EQ free lit. #105.

ONE FOR ALL

Celestion's new Studio 1 is the latest addition to its Pro Monitor Loudspeaker Series. All of the monitors in the Pro Series are compact, 8-ohm units finished in black ash veneer. They also all feature Celestion's proprietary 1-inch titanium dome tweeter as well as felted fiber cone drivers that are now ported for a smoother extended-bass response. An improved crossover design also provides smoother midrange performance with increased sensitivity. The Studio 1 features a 4-inch bass/midrange driver and can handle 50 watts. Frequency response is 78 Hz –20 kHz and sensitivity is rated at 89 dB SPL. Suggested retail is $210 per pair. For detailed information on the Pro line, contact Celestion, 89 Doug Brown Way, Holliston, MA 01746. Tel: 508-429-6706. Circle EQ free lit. #106.

FIXED RATE

New from Z-Systems is the Z-1SRC sample rate converter. It provides a simple and cost-effective way to make all your digital audio signals compatible. Built around the Analog Devices AD1890 chip, the Z-1SRC features 20-bit input, 24-bit output, arbitrary sample rate conversion, format conversion, varispeed to a single rate, and jitter reduction. It's useful for digital transfers, digital copying, layback to video, and miscellaneous interfacing. It features AES/EBU; S/PDIF; coax, optical, and XLR inputs and outputs; and can be externally synchronized. For more information, contact Z-Systems, 4641-F N.W. 6th Street, Gainesville, FL 32609. Tel: 904-371-0900. Circle EQ free lit. #107.

EQ PRODUCT VIEWS

HIGH DEFINITION; LOW PROFILE

Eastern Acoustic Works has introduced two new speaker systems developed for the contract division of Siemens A.V. EAW's JF50 High Definition Compact Full Range System is designed for nearfield applications where optimum sound quality and dynamic range are required with a minimum of visual obtrusiveness. Just about 16 inches high, the JF50 will put out up to 119 dB SPL. The complex crossover includes computer-optimized asymmetrical filters and driver protection. Even smaller than the JF50 is the UB12 Ultra-Compact Full Coverage/Surround System. Its maximum output is 113 dB SPL. It sports a vented enclosure that extends low-frequency response, and a 1-inch soft dome tweeter that produces high clarity and definition. For more information, contact EAW, One Main Street, Whitinsville, MA 01588. Tel: 508-234-6158. Circle EQ free lit. #108.

RACK 'N' ROLL

MIXRAK, a division of Bullfrog, manufactures a complete line of professional studio rack furniture. MIXRAK features modular design, allowing a customized system to be quickly and easily assembled. Available in a variety of heights and sizes, the furniture's surface is protected with melaface and high-pressure laminate. An inch and a half of solid oak edges the parameter of the desk, while durable, high-impact T-molding protects all other edges. MIXRAK is manufactured with a computer controlled, state-of-the-art woodcutting system. For more information, contact Bullfrog, 1503 Prairie Ave., South Bend, IN 46613. Tel: 219-233-4151. Circle EQ free lit. #109.

GREATER HARMONY

DigiTech recently introduced Version 2.0 for the DHP-55 digital harmony processor. The total effects configuration list has been increased to over 175, with more than 70 featuring a new reverb algorithm. A new Volume Modulator adds tremolo and a stereo out-of-phase tremolo. Other advances include an Auto Pitchcorrect mode, intelligent overflow messaging, faster custom harmony editing by instrument input, and parameter editing by MIDI sysex and continuous controllers. The Auto Pitchcorrect mode shifts the input signal directly to the nearest correct scale tone. Intelligent overflow messaging shows exactly which effect is being overdriven. Custom harmony editing is simplified directly from the instrument because the DHP-55 recognizes and stores the pitch as the selected note. All parameters are now editable using MIDI continuous controllers and sysex messages. For complete details, contact DigiTech, 8760 South Sandy Parkway, Sandy, UT 84070. Tel: 801-566-8800. Circle EQ free lit. #110.

HAVE YOU HEARD...

Kurzweil is making a new line of accessories available. The line includes everything from extensive sample-disk libraries for the K2000, to wearables, pedals, footswitches, SCSI cables, and road cases. The products are available through dealers or by dialing Kurzweil direct at 1-800-400-6658...**Symetrix** has added a 44.1 kHz sample rate oscillator to its 601 digital voice processor. The previous version of the 601 had the ability to adapt to either 48 kHz or 44.1 kHz, but could only originate 48 kHz...A line of new accessories was introduced by **DOD** recently. Products in the line include the DOD 210 Cable Tester and the 275 Active Direct Box, which features battery/phantom power switch, a ground lift switch, and an input level selector...**Hi-Mu Amplifiers** is offering the Model 5.5, which is a guitar amplifier designed specifically for project studio applications where loud noise may pose mic leakage or problems with neighbors. It's an all-vacuum tube, handwired, compact two-piece unit that offers the sounds you would expect from a quality tube amplifier, but wouldn't be able to get from a 50- or 100-watt gigging amp with its volume set low...**Peavey**'s new SP™+ is an enhanced version of the DPM SP Sample Playback modules. It features all the capabilities of the original SP, but increases polyphony from 16 to 32 voices in a single rack space. RAM capacity has also been increased to support up to 64 MB of standard SIMM memory...The AM700 Audio Measuring Set from **Tektronix** is a fully integrated test instrument. Its onboard digital capability and its processing power, programmability, and convenient interfacing make it a very versatile test instrument in its class. It offers analog and digital analyzers, analog and digital generators, internal CPUs, a monochrome VGA display, disk drive, and memory.

8•BUS SERIES

DUE NEW GREG

6 AUX SENDS with Solo and Solo LED.

6 STEREO AUX RETURNS. All have 20dB gain, Solo and can be used in stereo & mono. 1 & 2 are pannable & bussable.

MIX B/MONITOR section can be used as an independent stereo out for PA monitor mix, 2-track recording, video/broadcast feed or assigned to L/R mix.

TWO SEPARATE HEADPHONE SECTIONS can be used totally independently of each other. Each features source selection between Control Room & any combination of AUX 3/4, AUX 5/6, Mix-B or External source. Solo allows control room to hear what musicians are hearing in their headphones.

TALKBACK assigns to all submasters, main mix, AUX 1, AUX 2 or Phones 1&2.

SOLO level adjust and ultra-rude LED.

MONITOR section with separate Control Room & Studio levels. Source selection between L/R mix, Mix-B, Tape & External. Can be switched to Mono.

–40 to +10 bar graph LED DISPLAYS for each sub-master & Solo/Main (with main L/R +28dB CLIP LEDs).

EXPANSION CONSOLES let you add channels in banks of 24 to either the 24•8 or 32•8. Expanders have their own internal mix amps so the main board only "sees" one extra channel per expansion console.

Built-in talkback MIC.

Trick BUS SOLO switches send odd-numbered buses to the left speaker and even-numbered buses to the right speaker — unless you've pressed the respective MONO L&R button. When a bus has been mono-ed, SOLO sends the bus to both speakers.

L MIX/R MIX & MONO L & R buttons assign buses to main L/R stereo bus.

All channels have Mackie's renowned discrete, wide-bandwidth MIC PREAMP circuit for ultrahigh headroom & low noise. All mic inputs have RFI choking, ferrite beads and +48V phantom power (switchable in banks of 8 channels).

Optional tilt-up METER BRIDGES are globally switchable to see tape return preamps or channel output and include VU meters for main L/R output. MB•24 meter bridge for 24•8 console is $799*. MB•32 for 32•8 is $899*. Our soon-to-be-released 16•8 16x8x2 console's meter bridge will retail for $699*.

Rugged, non-flexing STEEL CHASSIS.

4-BAND EQ with "Expensive British Console Sound." Includes TRUE PARAMETRIC HI-MID, swept LO MID, shelving HI & LO plus 18dB/oct HI PASS (lo cut) filter at 75Hz. Users are raving about the sound quality.

Optional stand ($295*)

You'll like the increase in both sound quality and versatility.

We paid the same kind of attention to fader quality. Instead of less-accurate D-taper faders, we commissioned a totally new custom 100mm fader with the logarithmic taper found in mega-expensive consoles.

YE OLDE ENGLISH SOUNDE. Greg started out by asking "What is it that makes the finest British mixing boards perform the way they do?" For example, "classic," older English consoles have much wider-band midrange EQ than lower-priced consoles — it really has an effect on overall sound quality. So we incorporated the same capabilities on our new consoles. This also enabled us to add the flexibility of a variable bandwidth control for true parametric HI-MID EQ. It wasn't easy to engineer in the expensive circuitry necessary and still keep our consoles affordable, but we did it.

EVEN THE FEATURES HAVE FEATURES. Naturally each channel has In-line monitoring with split EQ. But our MIX-B Monitor section also has a SOURCE switch to tape off the channel (pre-fader) to create independent mixes for taping, broadcast feeds or headphone mixes. Dual independent headphone sections offer the ability to switch between Control Room and any combination of AUX 3/4, AUX 5/6, MIX-B or External sources. Tape inputs and outputs feature internal

*Suggested Retail Price. Your actual

+4dBU balanced TAPE RETURNS, switchable to –10dBV unbalanced in banks of 8 returns.

Balanced MIC, bal./unbal. LINE IN, MIC/LINE switch, DIRECT OUT & CH. INSERT on every channel.

Three TAPE OUTPUT jacks per bus (total of 24). +4dBU balanced, switchable in banks of 8 to –10dBV unbalanced.

TO THE UNEXPECTEDLY HIGH DEMAND FOR OUR 8•BUS CONSOLES, WE WON'T EVEN LET MACKIE HAVE ONE YET.

HE WORKED. HE SLAVED. He created the 8-bus console HE always wanted to own. In fact Greg kept adding features until we threatened to whack him upside the head with a rancid salmon. Now the first notices are in from 8•Bus owners: spontaneous raves from recording studios, PA companies and videopost houses. Quotes like "It's so quiet I had to check to see what it was on" and "Blows away my old board that cost $20,000." In other words, Greg really DID succeed at creating the first truly affordable high-headroom, low noise, feature-laden 8-bus consoles. Unfortunately, we can't build them fast enough to meet demand. Unlike our competitors, Mackie can't just order up consoles by the container-load. Instead, we build each 24•8 and 32•8 at our factory in Woodinville, Washington. Even though we're working day and night (and shipping more and more each week), there's still a waiting list at Mackie dealers. Even Greg hasn't gotten one yet! Serves him right for designing so much performance into consoles that retail for $3995* and $4995*. Read on for the deliciously explicit details.

220-watt, Class A POWER SUPPLY with enough juice to also power a meter bridge.

+4dBU to −10dBV level conversion so you can use semi-pro tape decks without the inherent noise penalty found in mixers that operate at −10dBV internal levels.

MACKIE'S SIGNATURE MIC PREAMPS. At the urging of legions of satisfied CR-1604 and MS1202 users, we didn't mess with a good thing. Our 8•Bus consoles' mic preamps deliver −129.6dBm E.I.N. at 0.005% THD with a 300K bandwidth, yet can handle +14dBU inputs without a pad. The consoles' working S/N is 90dBu with 116dB internal headroom. For any application where noise is especially noticeable (such as hard disk or multi-track digital recording), you've found your board — for as little as $3995*!

SOME OF THE FEATURES GREG ADDED SINCE WE FIRST ANNOUNCED OUR 8•BUS CONSOLE LINE:

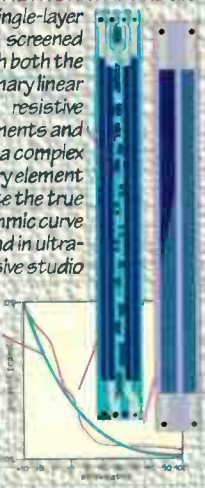
- External 220 watt, Triple-Regulated, Low-Ripple Power Supply
- Mix/Line switch on every ch.
- "Triple-bussed" tape outputs
- +4/−10 tape inputs & outputs (switchable in banks of 8)
- "Before we threatened to whack him upside the head with a rancid salmon if he didn't stop."

EXPANDABLE AND AUTOMATABLE. Need 24, 48 or even 72 extra channels? Add one or more 24-channel expander consoles (complete with inputs, tape returns and their own power supply) at any time in the future. Just by connecting one cable between the expander and your 24•8 or 32•8 board. External fader and muting MIDI automation will also be available soon.

OPTIMIZED FOR PA AND RECORDING. Along with elaborate monitor capabilities, balanced XLR main outputs and 18dB/octave hi pass filters, you get non-flexing steel construction, fiberglass thru-hole plated, horizontal circuit boards that minimize impact damage, gold-plated internal interconnects, sealed rotary pots and a rugged 220-watt, super-regulated power supply. You won't find more roadable, compact PA boards anywhere.

READ ALL ABOUT IT. Call us toll-free and we'll ship you a comprehensive brochure including application hookups. We think you'll be impressed enough to be willing to wait a little while before you get your 24•8 or 32•8. After all, Greg is still waiting for his.

Conventional faders have a second layer of resistive material that attempts to approximate logarithmic a taper. Our PRECISION NETWORK FADERS are single-layer screened with both the primary linear resistive elements and also a complex auxiliary element to create the true logarithmic curve found in ultra-expensive studio console faders (blue line on graph).



In-line FLIP reverses tape and mic/line inputs between channel strip and Mix-B/Monitor section.

AUX SENDS 1-2. PRE button selects pre-fader/post EQ or post-fader/post EQ.

AUX 3-4/5-6. SHIFT changes 3-4 to 5-6.

SOURCE selects signal source of AUX 3-4/5-6 from channel strip to channel's Mix B/Monitor send so you can build an effects mix (pre or post-MIX-B level) to assign to phones during tracking.

True parametric, 3-control HI MID EQ that has seasoned engineers swooning (quotes and raves on file... we're not kidding). Ultra-wide 500-18k frequency sweep range; bandwidth can be adjusted from a very wide 3-octave width to a very narrow 1/12-octave width. 15dB boost/cut.

LO MID EQ with ultra-wide 45Hz-3K sweep, 15dB boost/cut.

±15dB shelving HI (12kHZ) & LO (80Hz) EQ.

Multipurpose 18dB/oct. LO CUT filter @75Hz. Cleans up "mix mud," cuts PA rumble, creates a "neo-peaking" bass control when used with LO shelving boost.

Independent MIX-B (Monitor) section with pan, level & source. During mixdown, use as extra pre-fader stereo AUX send or double your inputs.

Mix-B SPLIT EQ assigns HI & LO EQ to Mix-B.

MIX-B SOURCE can route the monitor section to an extra stereo output for 2-track taping or broadcast feed during live mixing.

Constant power, buffered PAN pot for rock-solid panning.

Overload LED and Hyperactive −20dB Signal Present LED.

Selectable SOLO with CHANNEL METERING allows soloing in full stereo perspective; displays soloed channel operating level on master L/R meters so input trims can be adjusted for optimum levels.

*price, like your mileage, may vary. Prices are slightly higher in Canada.

MACKIE.

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CIRCLE 33 ON FREE INFO CARD

The Original Kramer

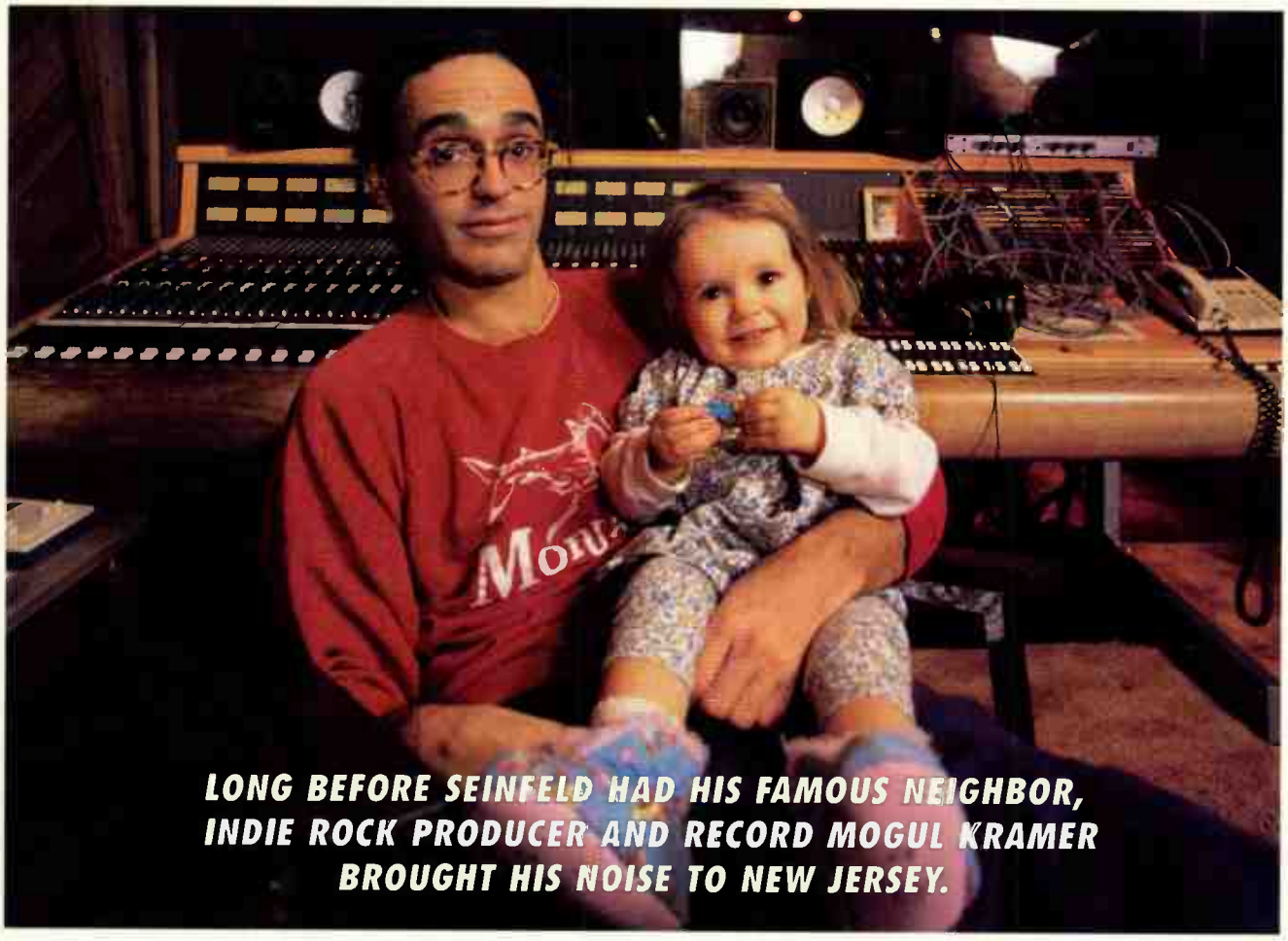
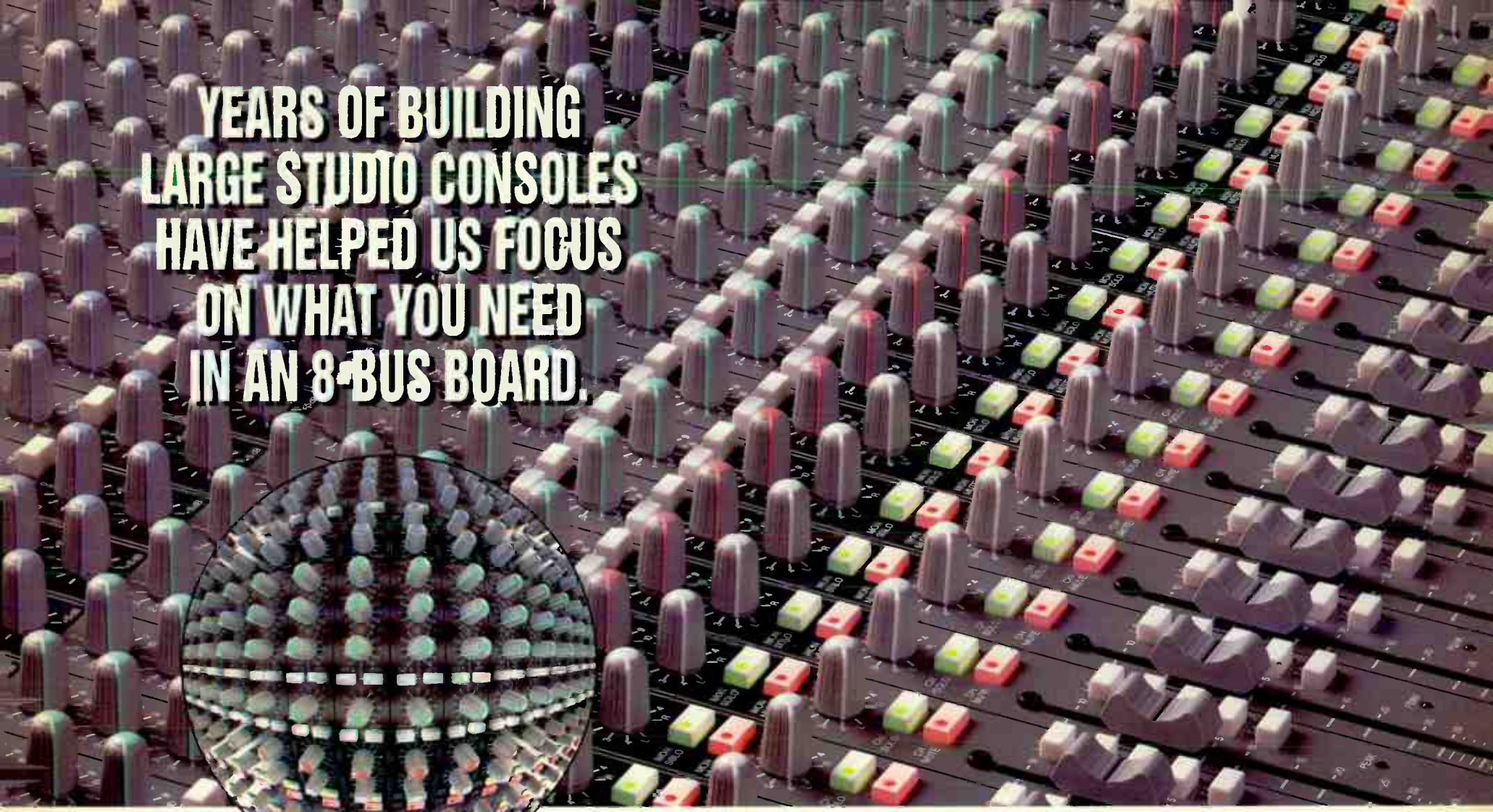


Photo by Julian Jaime

LONG BEFORE SEINFELD HAD HIS FAMOUS NEIGHBOR, INDIE ROCK PRODUCER AND RECORD MOGUL KRAMER BROUGHT HIS NOISE TO NEW JERSEY.

STUDIO NAME: Noise New Jersey
LOCATION: Demarest, NJ
KEY PEOPLE: Kramer (owner); Tess (daughter)
BANDS RECORDED: Devils Wielding Scimitars; Butthole Surfers; Nova Mob; Hugh Hopper; Galaxie 500; GWAR; Unrest; Fred Frith; and Drizzle
CONSOLE: Trident 80B (54 x 24 x 2), modified
RECORDERS: 24-track Sony JH 24-2 (2-inch); 2-track MCI JH110B (1/2-inch); 2-track Technics RS1520 (1/4-inch); DAT Panasonic SV-3500
SPEAKERS: Urei 813B, Yamaha NS10, Auratones
MICS: AKG The Tube, C414, D320B, D12E, and C451, Sony C-37, Electro-Voice PL20 and PL5; Beyer M160N and M500N; Neumann U87, U89, and KM84; Shure SM57 and SM81; Sennheiser 421
KEYBOARDS: Yamaha Baby Grand and DX7; Kawai Baby Grand; Hammond B3; Mellotron; E-mu Emulator II; Farfisa; Ensoniq Mirage; Korg Digital Piano DP2000C; Vox Baroque; Korg M-1; Polaris Synthesizer; Casio MT65
COMPUTERS & SOFTWARE: JLCooper 32-channel Automation; Macintosh IIx; E-mu Procussion
OUTBOARD GEAR: Lexicon Alex, 224XL, and Prime Time; EMT-140 Reverb Plate; Ecoplate III Reverb; Yamaha REV 7 and D1500; Roland SRV-2000; Aphex Aural Exciter Type B; Korg SDD2000; MXR 1500; ART Multi-Verb; Eventide Clockworks H910 Harmonizer; Delta Lab DL-2
COMPRESSION: Urei 1176LN limiter and LA-4 compressor; dbx 160; Dyna-Mite Gates; Drawmer Dual Gates; Audio Logic Quad Noise Gates
MISCELLANEOUS: Meditation pyramid; Ping Pong table
EQUIPMENT NOTES: Kramer states: My most prized possession in my studio is my Mellotron. It's a classic piece of equipment with the sound of the Moody Blues and King Crimson from the '60s, which are both bands I grew up with. It strikes a chord in my heart that is not reproduced by any modern equipment.
STUDIO NOTES: Kramer continues: My previous studio in Manhattan [Noise New York], which I sold, had 16 tracks. This studio is state-of-the-art with 24-tracks and it's *entirely* different. The early Shimmy Disc releases I recorded at the other studio have almost nothing in common qualitywise with what I'm doing here. I have a Mac, but I never use it — it's a drag. To balance my own music with recording bands, I take off December and January every year for working on my more private projects. Overall, I'm not as interested in technology as I am in music. Machines are just the tools to get the job at hand done. EQ

YEARS OF BUILDING LARGE STUDIO CONSOLES HAVE HELPED US FOCUS ON WHAT YOU NEED IN AN 8-BUS BOARD.



TOPAZ. Our reputation for innovation and excellence in high-end multitrack consoles affords us a singular perspective on the art of recording and mixdown.

Insights gleaned over the years have led to Topaz 24, a 48-input in-line console designed with the sonic integrity and smooth, responsive operation of our most prestigious recording consoles.

MORE EQ. Others may claim to offer "British EQ," but we deliver the real thing, and more of it. Not only Soundtracs' world-class 4-band EQ with dual swept mids, but also dedicated EQ on all tape monitors without compromising your primary equalization.

MORE CONTROL. In addition to a logical, fully implemented control surface, Topaz includes SOLO and MUTE functions on all tape monitors, a critical feature in cutting through the mix to isolate problems, something our competitors may have overlooked.

MORE FLEXIBILITY. Our "Floating Bus" design enables you to route Topaz's 8 group outputs to all 24 inputs of your tape machine(s) without repatching. A comprehensive meter bridge is also available as an option for both the 24- and 32-channel Topaz.

MORE AUTOMATION. When it's time to automate, we give you the professional option of 12-bit, high-resolution VCA/Mute automation with 4,096 increments on each fader to eliminate "zipper noise."

Topaz from Soundtracs. Our track record with big boards allowed us to design the first 8-bus console with everything you need. For more information, call (516) 932-3810 or fax to (516) 932-3815.

TOPAZ BY SOUNDTRACS PLC
Surbiton, Surrey, England

Suggested list price for Topaz 24-channel: $3,995; Topaz 32-channel: $4,995. Soundtracs is exclusively distributed in the U.S. by Samson Technologies Corp., P.O. Box 9068, Hicksville, NY 11802-9068. ©1994 SAMSON

CIRCLE 48 ON FREE INFO CARD

The Neumann CMV 563

Get a taste of these long-lasting lollipops

MICROPHONE NAME: Neumann (East German) CMV 563. Microphone used with M7, M8, M9 plug-in capsules and M55K screw-in capsule.
TYPE OF MIC: Condenser
FROM THE COLLECTION OF: Mic-Heaven, Hoboken, NJ
SERIAL NUMBER: CMV 563 — 13601, 13602; M7 — 13078, 13079; M8 — 1620, 1621; M9 — 262, 263; M55K — 1146, 1147
CURRENT VALUE: $10,000
POLAR PATTERN: M7 — cardioid; M8 — figure 8; M9 — omni; M55K — omni pressure
FREQUENCY RESPONSE: CMV 563 — 30 Hz–20 kHz; M7 — 40 Hz–16 kHz; M8 — 40 Hz–8 kHz; M9 — 30 Hz–20 kHz; M55K — 30 Hz–20 kHz
OUTPUT IMPEDANCES: 200 ohms; balanced
HISTORICAL NOTES: This is the last version (1956) of the famous Neumann "Bottle Mic" that first appeared in 1932 as the CMV3A, and uses the same plug-in capsules. The M7 capsule was also used on the West German Neumann M49, M249, U47, and U48 mics. The M55K pressure capsule is a modern (1950) version of the original CM3 capsule used on the original CMV3 Neumann mic from 1927.
SONIC QUALITIES: Each capsule type is optimized for its pattern and is unique.
USER TIPS: M55K pressure omni capsule is unsurpassed for recording acoustic guitar. It has an airy, open sound that brings the acoustic guitar to life, say users Roger Johansen of Mic-Heaven and Jerry Graham of G Prime Ltd. in New York.

Another application is the use of three CMV 563's in a three-point configuration (left, center, right) to record classical orchestras. This technique has been used on many German recordings and is still employed as a favored technique in such applications.
FUN FACT: This mic was replaced by the East German Neumann UM57 three-pattern mic using the M7 capsule in 1957. The Microtech Gefell UM92S is the modern version of the UM57. EQ

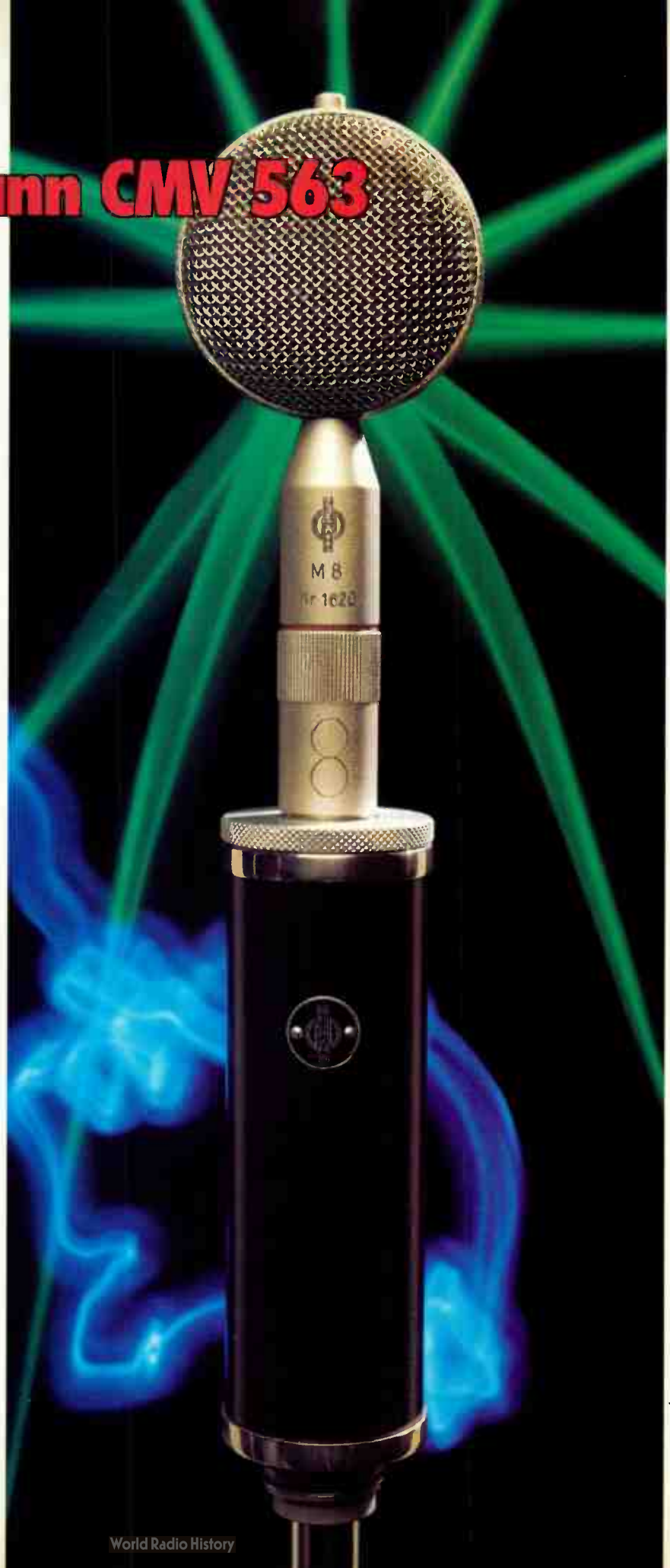


Photo by Ed Colver

The new Kurzweil MicroPiano.™ $499 of pure inspiration.

The new MicroPiano – The half-rack module features 32 Presets including Kurzweil's new Grand Piano samples, Strings, Hammond® Organ, Electric/Electronic Pianos as well as 16 superb digital multi-effects.

Musicians have always envied those who have had the legendary sounds of a ***Kurzweil*** at their command – especially our grand pianos, electric pianos, strings and organs. With the new $499 ***MicroPiano*** sound module, *you* can now add all these great sounds (and others) to your keyboard setup.

The *MicroPiano* features 32 of the most sought-after keyboard sounds with full 32-note polyphony (*64-note with two MicroPianos in the exclusive Link Mode*). In addition to keyboard sounds, Kurzweil's lush string section, played solo or layered with another sound, creates a gorgeous orchestral ambiance. Some sounds are based on the proprietary samples from the award-winning ***K2000***, but many are brand new, available only in the *MicroPiano*. If you don't have a nine-foot concert grand and a great recording engineer, you need the *MicroPiano*.

Besides the acclaimed Kurzweil samples, the compact, half-rack module offers the kind of playability a keyboard player expects, with 16 superb, crystal-clear digital multi-effects, useful MIDI control capabilities and fully-functional soft, sostenuto and sustain pedal response. The user interface is straightforward, easy-to-use and includes Tuning and Transposition as well as Stereo Outputs.

At just $499 suggested retail, we've made it a lot easier for you to play a true Kurzweil. Whether you're a novice or a pro, audition one today at your authorized Kurzweil dealer.

The new ***MicroPiano***.
It's Pure Kurzweil.
And ...

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KURZWEIL

Kurzweil is a product of Young Chang America, Inc.,
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Pure Inspiration.

Testing in the Digital Studio

Using your hard-disk system to make sure your other gear is performing up to par

BY CRAIG ANDERTON

Hard-disk recording and editing systems just happen to make great pieces of audio test equipment: you can record signals, "freeze" them on-screen, and measure their characteristics using the time and amplitude grids of hard-disk editing software.

Let's look at how we go about testing noise gate response, calibrating delay lines, determining sequencer jitter, checking frequency response, and taking gain measurements. (Note: Many of these tests can be completed more quickly with the Russian Dragon, a clever piece of time-calibration test equipment by Jeanius Electronics [see *EQ*, December 1993]: If you often make the types of measurements discussed below, consider augmenting your hard-disk system with this box.)

IS YOUR NOISE GATE HALF-FAST?

One of the characteristics that separates a superior noise gate from a so-so model is how fast it opens up in response to an input signal. To test this, split a percussive audio signal to one channel of your hard-disk system and patch the other split to the noise gate input. Feed the noise gate output to the second hard-disk channel (fig. 1).

Set the noise gate's attack time to minimum, and record the percussive signal in stereo. Measure the time difference between the start of the two events; anything less than a few hundred microseconds is pretty good.

WHEN SEQUENCERS GET THE SHAKES

Just how stable is your sequencer? Are there things you can do to make it more stable? Hard-disk systems can point the way to the best sequencer performance. Although the following is a very basic check — we're not testing for multinote/multicontroller performance over several channels — it can nonetheless point out differences between sequencers and the platforms on which they run.

First, you need a device that responds rapidly and predictably to MIDI note-on messages. Of all my gear, the Alesis HR-16 tested out best, at approximately 1.6 ms of MIDI delay. The important characteristic is consistency. Since we are interested in relative timing shifts more than in absolute ones, the actual delay time isn't too important, providing it's constant.

Second, you'll need to record a test sequence into the sequencer. A series of 16th notes in channel 1 will do for now. Either quantize these or enter them in step time.

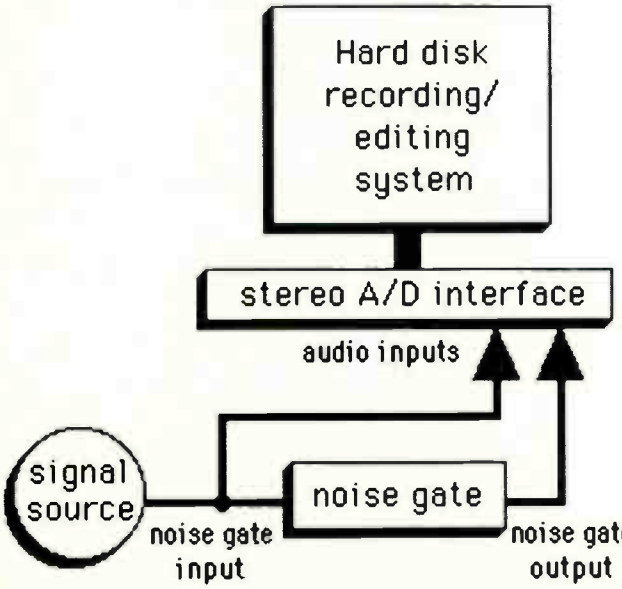


Figure 1

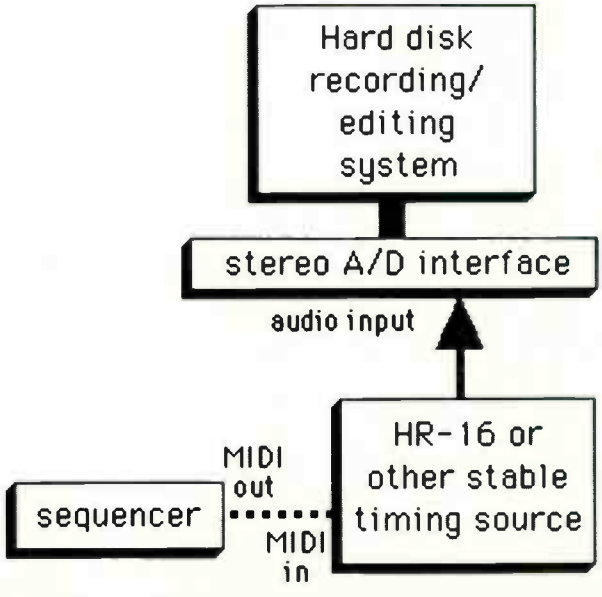


Figure 2

Suggested retail list $279.

Introducing the new DUALFEX II from Behringer.

Have you noticed a difference between your recordings and the sound of CDs and tapes from your favorite artists? You can't fix it with EQ. And adding more parts to the mix just makes everything sound muddier.

With the new DUALFEX II, all your parts stand out bright and clear, up front in the mix. Once you hear how it makes your music jump, you won't be able to live without it.

You've heard about exciters and enhancers. The DUALFEX is both — and more. Its unique Variable Sound Processor lets you manually adjust the processing from "enhance" to "excite," and get every sound in between.

And, unlike other units, the Dualfex's tunable high pass filter allows you to tailor the high end while a separate frequency switchable Bass Processor lets you fatten up the bottom.

Best of all, you get the musical, satisfying level of sonic performance and noise-free sound enhancement you've come to expect from Behringer.

DUALFEX II. The best way to go if you want to take your music to a better place.

THE PERFECT EAR. BEHRINGER

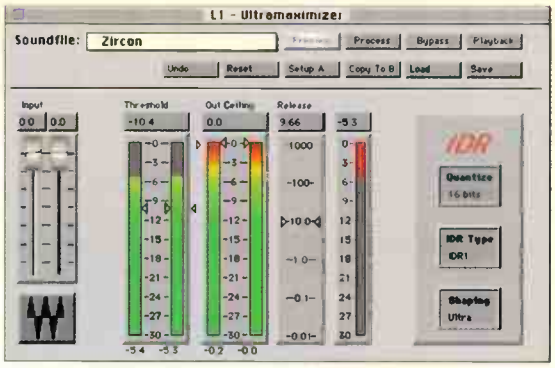
For more information about Behringer Signal Processing, please call 1-516-932-3810 or write to Samson Technologies Corp., P.O. Box 9068, Hicksville, NY 11802-9068.

Behringer is exclusively distributed in the U.S. by Samson Technologies Corp.

CIRCLE 09 ON FREE INFO CARD

EQ STUDIO WARE

TAKE IT TO THE LIMIT



Waves, Ltd. continues its digital audio product introductions with the release of the L1 Ultramaximizer, a plug-in for Digidesign's Sound Designer II and TDM systems. The L1 incorporates a transparent brick-wall limiter and full implementation of Increased Digital Resolution (IDR) dithering technology options. Proprietary technology is used to control peaks with minimum audible effect in the digital domain. For more info, contact Rockwell Digital, 4501 Glencoe Avenue, Marina Del Rey, CA 90292. Tel: 310-577-0480. Circle EQ free lit. #111.

MULTIMEDIA MONITORS

Audio-Technica is now offering the MS337 and MS557 multimedia speaker systems. Designed primarily for computer systems, these speakers can also be used in other applications. One speaker of each system houses a 2-channel amplifier with controls for volume, bass, and treble in both channels, and an LED power indicator. The enclosures are magnetically shielded and feature an internal 120-volt AC power supply and line cord. The MS557 features dual bass-reflex enclosures with 4-inch woofers and 3/4-inch tweeters. The built-in amp provides up to 20 watts total output. Frequency response is 80 Hz–20 kHz, sensitivity is 89 dB. The MS337 utilizes 3-inch full-range speakers, and the amp puts out 7.5 watts total. For more information, contact Audio-Technica, 1221 Commerce Drive, Stow, OH. Tel: 216-686-2600. Circle EQ free lit. #112.

IF I HAD A HAMMER

FWB has introduced HammerDisk•PE 250. It's a SyQuest-based subsystem and is a SCSI-2 removable media drive featuring a 3-1/2–inch drive that allows up to 256 MB of data to be stored on each data cartridge. The drive also offers compatibility for existing 105 MB data cartridges. Suggested retail is $829. For more information, contact FWB, 2040 Polk Street, Suite 215, San Francisco, CA 94109. Tel: 415-475-8055. Circle EQ free lit. #113.

SYNC A SONG

Mark of the Unicorn (MOTU) has introduced the first desktop digital audio synchronizer: the Digital Time Piece. It provides conversion of many digital audio synchronization formats as well as MIDI Machine Control and SMPTE timecode. The Digital Time Piece works with Pro Tools, Sound Tools II, AudioMedia II, ADAT, Performer, Digital Performer, Tascam DA-88, and virtually all professional MIDI sequencers on any computer platform. For more information, contact MOTU, 1280 Massachusetts Avenue, Cambridge, MA 02138. Tel: 617-576-2760. Circle EQ free lit. #114.

HAVE YOU HEARD...

Two new nontraditional CD-ROM libraries from **OSC** have just been introduced: *A Poke in the Ear With a Sharp Stick, Volume III*, which provides over 18,000 sounds, effects, loops and clip-tunes from five different sound designers; and *Textural Environments*, which contains long, evolving atmospheres, soundtracks and beds...**Ensoniq** has added General MIDI compatibility to its TS Series of Performance/Composition synthesizers (TS-10 and TS-12). The O.S. Version 2.5 GM upgrade provides 16-channel multitimbral MIDI reception with all 128 GM sounds, including all eight GM and GS drum kits, permanently stored in ROM...**Kurzweil** has debuted Version 3 software for its K2000 Series samplers. Version 3 offers a 32-track sequencer with 16 song tracks and 16 arrangement tracks. The sequencer supports pattern or linear sequencing, auto punch, multi MIDI-channel record, event list editing, track solo/mute, mixdown functions, and live performance arranger features. The disk file system has been enhanced as well...**Crystal River Engineering** has begun participating in the Digidesign Third-Party Developer Plug-Ins program. CRE will work with Digidesign to market its Spatial Effects plug-in for Pro Tools...Speaking of **Digidesign**, the company has unveiled SampleCell II PC for Windows. It's a stereo, 16-bit, 32-voice, 32 MB sample playback card for the Windows/PC platform.

Mike Pinder likes to keep his Mellotron mellow.

"Part of my bliss is having The CardD at my fingertips. The CardD has become an integral part of my studio, being permanently connected to the output of my mixing console. I am very impressed with this product. It's a pleasure to use, and I am equally impressed with the people who stand behind it."

Songwriter, vocalist, and keyboardist Michael Pinder, formerly with the Moody Blues, is famous for his pioneering use of the Mellotron. The hauntingly beautiful sounds of this instrument became the trademark of the group's early works.

Mike is back in the studio, and has just released a new CD called *"Off the Shelf."*[1] Mike's new CD *"has a jazzier, more sophisticated flavor than his music with the Moodies, while retaining...'that heavenly atmosphere.'"*[2]

When it came time to digitize his recordings for final mastering, Mike trusted only one system. *"...at the end of the chain I mix directly to an IBM clone computer running The CardD™, by Digital Audio Labs... It's fabulous, for a thousand dollars, and it has incredible editing and the A-to-D and D-to-A converters are the best I ever heard. I do all of my mastering there."*[2]

Cards

The CardD Plus™ — $795
Professional-quality soundcard for the IBM.

The CardD™ — $295
Companion to The CardD™, for direct digital transfer to and from your DAT.

The Digital Only CardD™ — $495
Stand-alone card for direct digital transfer to and from your DAT.

Software

FastEddie™ — $199
The *FAST* editor for Windows™ soundfiles.

The EDitor Plus™ — $349
The Professional editor for Windows™ soundfiles.

[1] Mike's CD is available exclusively through Higher & Higher, P.O. Box 829, Geneva, FL 32732. Send SASE for information.

[2] From Higher & Higher, an independent fan magazine focusing on the Moody Blues, Winter/Spring 1994

Digital AUDIO LABS
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Plymouth, Minnesota 55447
phone (612) 473-7626 fax (612) 473-7915

CIRCLE 56 ON FREE INFO CARD

Mixed Media

Getting a lock on analog/digital hybrid studios (with some video thrown in for good measure)

BY WES DOOLEY

It takes a real mix "master" to get a lock on hybrid studios.

Multitrack recording originally meant magnetic tape on a single machine. Then some of us found we needed more tracks or wanted to be locked up to video for sweetening or scoring sessions. The industry is now discovering the joys and pitfalls of digital audio production. Meanwhile, we have to use both older and newly emerging media together. Lots of choices. Lots of chances to be creative. Lots of opportunity to discover some real problems. What follows is a guide to the why and how of mixed media cooperation.

TWO MEN'S EXPERIENCES

Many music projects start with basic tracks on an analog 2-inch machine, then do a cue mix onto a digital machine and start overdubbing on the digital. For mixdown they lock the analog and digital back together. Why? For Cal Harris it's always been a matter of sound. His work on the last three Lionel Richie albums illustrates digital's evolution. On the first album he used his Sony JH24 for the basic tracks because it sounded better than his Sony 3324 digital machines. After upgrading the 3324's with Apogee filters, however, he began to use them for basics too. But he still bounces some material off the 3324 onto the locked up JH24 and then back onto the 3324 to get that analog tape sound.

Cal also runs 24 tracks of ADAT with a BRC and also has a Roland DM80. He uses a house sync video generator to keep all his equipment referenced to the same time base. A 360 Systems audio switcher handles the SMPTE timecode (TC) routing. JH24 synchronization chores are handled by a TimeLine Lynx. This is a serious personal facility, so everything is referenced to house sync and the exact same type of SMPTE timecode is used everywhere. (See fig. 1 for an example of house sync routing.)

Ken Scott is the engineer/producer on Duran Duran's current album in production. They have already done some homework on an Akai Adam digital 12-track. They used a TimeLine Lynx pair to sync the Adam up to an Ampex ATR 124 and transferred those tracks across to the analog machine. The rest of their work will be done on a pair of ATR 124's at Wyn Davis's facility, Total Access (Redondo Beach, CA). Mixdown will be to a 30 ips 1/2-inch machine with a DAT used as a safety backup. This 30 ips mixdown will be used to generate the 1630 digital CD master.

WHY ARE WE TALKING ABOUT ANALOG?

In a word, performance! Did you know that the majority of pop music CDs are mastered to digital from analog? That 1/2-inch 30 ips is the dominant mixdown format in Los Angeles? A producer like Keith Olsen, who is known for using cutting-edge technology, locks his pair of 3324's together at Goodnight L.A., mixes down to 1/2-inch 30 ips, and then immediately takes it over to be mastered. Analog is a mature format and we know how to use it very well. This doesn't mean that digital is not the future. It gets better all the time. But the serious players are still using analog for its flavor and for its performance. If you check out your favorite producers, you will find that their preferences range from all digital to all analog, but most of them still work in both mediums.

HOW LITTLE CAN I GET BY WITH?

If you already have a synchronizer on your analog multitrack, you're home free. Locking an analog slave to a digital master for a music project is fairly easy. The minimalist approach is to

Don't Mess Surround.

There's a lot of confusion about surround sound recording these days, and as a project studio owner you don't want to mess around with the wrong format. ∞ Cinema surround is fine for the movies, but what about your music? Now you can record with the world's finest music surround sound system, Circle Surround™ from RSP Technologies. ∞ Our patent pending intelligent process will enable you to position instruments, vocals, sound effects and so on, anywhere in the circle in conjunction with a four, or five, speaker surround system. Complete smooth panning of the entire 360 degree sound field is possible. ∞ Circle Surround™ uses no artificial ambience effects, and no schemes to mess up your original source material. Just the best surround sound process available for the most dramatic and realistic music and audio/video surround productions. ∞ Compatible with existing surround systems, Circle Surround™ will even dramatically improve performance of those typical cinema surround systems. ∞ So put your music, your soundtrack, your audio/video production, in good hands with Circle Surround™ and leave the popcorn at the movies. Give us a call, or visit your RSP Technologies dealer and quit messing around when it comes to surround.

CIRCLE SURROUND

RSP Technologies

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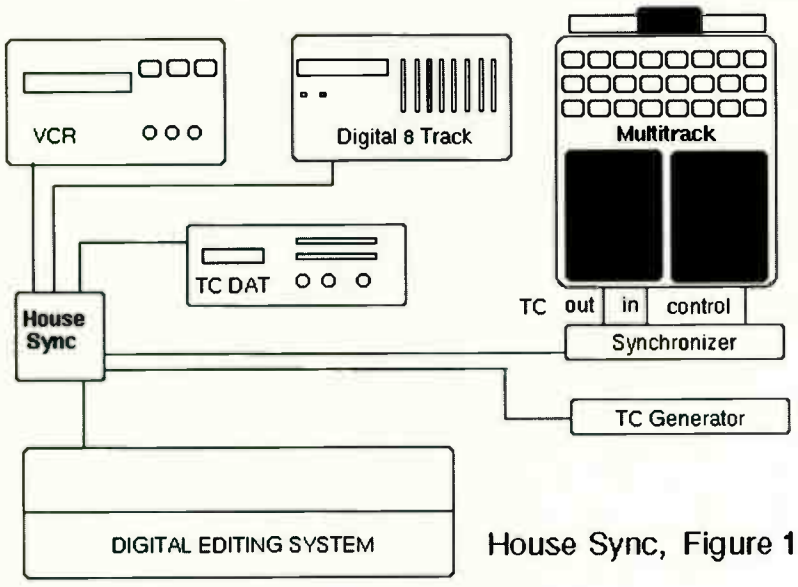


House Sync, Figure 1

Once you combine video and digital in a system, proper use of house sync becomes a necessity. Timecode recorded on a digital machine's audio track just won't be in sync with both the internal word clock and external video.

synchronize the multitrack to timecode from the digital recorder. You'll want to have the exact same timecode format and addresses on both the digital and analog machines. Just lock the multitrack to TC from the digital machine. This can be a bit awkward if the digital machine does not have a dedicated timecode track, but you can use the autolocators on both machines to shuttle to the top of a take to speed things up some.

A REAL WORLD STORY

Once you combine video and digital in a system, proper use of house sync becomes a necessity. Timecode recorded on a digital machine's audio track just won't be in sync with both the internal word clock and external video. Bruce Botnick at Pacific Ocean Post (POP), located in Santa Monica, CA, ran into this problem when he received the audio tracks for a Ravi Shankar concert shot in England. It was recorded on a locked pair of Tascam DA-88's. The timecode format was 25 frame (PAL video) code. It was supplied from the shoot's master TC generator and recorded onto an audio track. That's where the trouble started, as the timecode should have been recorded using an SY-88 chase synchronizer/timecode card. That would have both saved an audio track and kept the timecode properly referenced to the DA-88's digital word clock.

Bruce tried to do a straight digital transfer to his Sony 3348 locked to the concert video, but the timecode kept drifting. Luckily the DA-88's at POP had an SY-88 installed; they were therefore able to manually enter offsets and slave the DA-88's to timecode from the 3348 and do an audio transfer. Another option would have been to transfer to an analog multitrack and then synchronize that back to video.

The conclusions are clear:

1. The same house sync reference and format must be used everywhere while recording.

2. The right house sync/timecode/synchronization interface is critical in digital recording.

HOW MUCH TO DO IT RIGHT?

The real question is, what does it cost to do it wrong? You already know the answer to that, so we'll look at the price of a house sync generator. They start at $289 for a Horita BSG 50. It has six outputs, so you could feed a video machine, a timecode generator, a hard-disk recorder, a timecode DAT, an 8-track digital recorder, and a multitrack synchronizer. Enough for most projects.

The Tascam SY-88 Chase Synchronizer/Timecode card for the DA-88 is $799. The Alesis/TimeLine AI-2 Chase Synchronizer is $1099. These are options that you have to budget for if you need to synchronize to anything external. The good news is that you can synchronize an entire stack of their digital 8-track machines with the one unit.

SO WHAT'S NEXT?

It's your choice. You know what your resources and interests are. I'm just reminding you that people routinely use 30- and 40-year-old equipment when it contributes to their sound. So, develop your own sound and don't be afraid to be retro. But, if you are going to synchronize multiple media, buy a house sync generator and always use it — and use the same timecode format everywhere in your project. You'll never know how much pain and suffering you missed...unless you forget one day.

Wes Dooley works at Audio Engineering Associates in Pasadena, CA. He is well known as an advocate of the MS stereo microphone technique. He has also synchronized analog and digital machines for Barbra Streisand and lived to tell the tale. He can be reached at 818-798-9128 or faxed at 818-798-2378.

Hard Disk Recording Doesn't Have To Be Hard On Your Wallet.

"...in a price/performance comparison, the DR4d would be hard to beat. Thumbs up on this one." George Petersen, MIX Magazine

"...great sound, useful features, and friendly operation... technology that is sure to set a new standard in affordable recording" David Frangioni, EQ Magazine

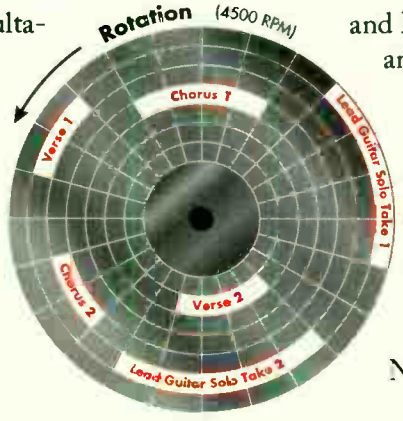
Ah, decisions, decisions. You want to buy a new multitrack recorder, and you want to go digital so that you'll get the best possible sound quality. And you'd like to buy a hard disk recorder, rather than tape, so you can get random access editing power. And finally, it's got to be something you can really afford. But there's a problem.... don't all hard disk systems require expensive add-in hardware and software, to already expensive computers? Not anymore!

The DR4d is the solution for those looking for an alternative to expensive, complex computer-based systems, or the limitations and mechanical uncertainty of tape recorders. It offers a perfect combination of hard disk recording benefits with an easy-to-use interface.

The DR4d can record up to four tracks simultaneously to standard SCSI hard disks, either internal or external drives. An optional 213MB internal disk offers 40 track minutes of recording (44.1kHz) right out of the box. To expand your recording time, simply connect external drives to the DR4d's supplied SCSI port.

With standard tape machine-style controls the DR4d is by far the easiest hard disk recorder to operate, which means that you can get to work immediately creating music rather than setting up and operating a computer system. Punch ins/outs can be performed manually or automatically from the front panel, or by footswitch, just like you'd expect.

Now you can start to take advantage of the power of random access editing. You can cut, copy, and paste sections of audio with ease. Our Jog/Shuttle wheel lets you scrub through the audio at various speeds, forwards or backwards. Try out different arrangements. Create perfect tracks by combining the best sections from multiple takes. Whatever. And you can edit with confidence, because if you change your mind you can instantly Undo your last edit - even after the power is turned off and on again! Imagine it. Do it. It's that simple.



On a spinning hard disk, the various sections of music can be accessed almost instantaneously by the moving heads of the drive mechanism. This allows you to seamlessly output different parts in any order, with no time spent rewinding. Music can be moved and rearranged in ways not possible with tape!

Another DR4d advantage is not having to wait for tape to shuttle back and forth. You can instantly move to 108 memorized locations at the touch of a button, and these locate points may be entered manually or on-the-fly. It's also simple to set up seamlessly looping repeat sections, so it's easy to jam over tracks. No more wasting time on rewinding tape!

Of course, how the DR4d sounds is as important as how it works. Advanced 18-bit oversampling A/D and D/A converters insure crystal clear sound, and with a full 96dB dynamic range, the DR4d offers no-compromise specs. The four balanced 1/4" input and output jacks are switchable between -10 and +4 operation, and 2-channel digital I/O is included standard (AES-EBU and SPDIF) with two additional digital ports optional.

Need more than four tracks? Four DR4d's can be linked to create a 16-track system. And for synchronization to other gear, just add the optional MIDI or SMPTE interfaces.

And best of all, the DR4d is an **affordable** reality: suggested list is only $2495.00 (or $1995.00 w/o hard disk)! Multitrack disk recording *is* within your reach, so see your Akai dealer today for a complete demo!

Intro Verse 1 Chorus 1 Verse 2 Lead

On tape, the sections of music are physically located far from each other, separated by many feet of the tape itself. Since you have to move all that tape past the heads to get where you want to go, it's impossible to jump instantly from one section to another. It wastes time, and limits creativity!

DR4d AKAI DIGITAL

P.O. Box 2344
Ft. Worth, TX 76102
(817) 336-5114
Fax 870-1271

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According to all the hype, multimedia will be the te
substance beyond the hype — multimedia is real.
position to join in on the reality (virtual and otherw
just a natural evolution for the project studio mover
exclusive three project studio vets probe...

AUDIO FO
MULTI

chnology that defines our decade. There is
As a project studio owner you are in a unique
ise). Some even say that multimedia is really
ment. Are you ready to evolve? In this *EQ*

R

MULTIMEDIA COMPOSING BY MURRAY ALLEN

MACINTOSH AUDIO BY MICHAEL TAPES

DOS/WINDOWS AUDIO BY HOWARD MASSEY

MEDIA

Multimedia Composing

Multimedia is more than just a buzzword — it's business

BY MURRAY ALLEN

As the information highway begins to germinate, new opportunities for a career path in music and sound will become available. The new "IHM" (Information Highway Musician) will have to be knowledgeable about music, sound effects, and dialog, and about how these sounds dramatically interweave with pictures, both motion and still. The Hollywood model of composer, sound effects spotter, sound editor, Foley artist, ADR mixer, and dialog, sfx and music mixer is a starting point; the IHM will become jack of all these trades. The IHM must also specialize and become a master of one of these same trades.

Multimedia means many different things to different people. This column, and all that follow, will discuss it as it pertains to games and CD-ROM technology. The reason is obvious, as the use of multimedia technology manifests itself mostly in the form of interactive video games and movies.

There are several "systems" available for the game-playing consumer. There is the Nintendo Entertainment System, Sega Genesis, Sega CD, 3DO, PC and PC-CD-ROM, Macintosh, Jaguar, Game Boy, and more. Every hardware manufacturer is boasting about its next CD-ROM device. And none of these devices are compatible.

On the PC platforms there are at least a dozen different types of sound cards that literally speak different languages. So if you are an IHM, how do you wade through this maze of interactive "Towers of Babel" and come out with product? The answer lies in having an intimate knowledge of the medium.

The sound source in a typical game deck or personal computer consists of one or more integrated circuits. Just as any other device in the system must be programmed, these

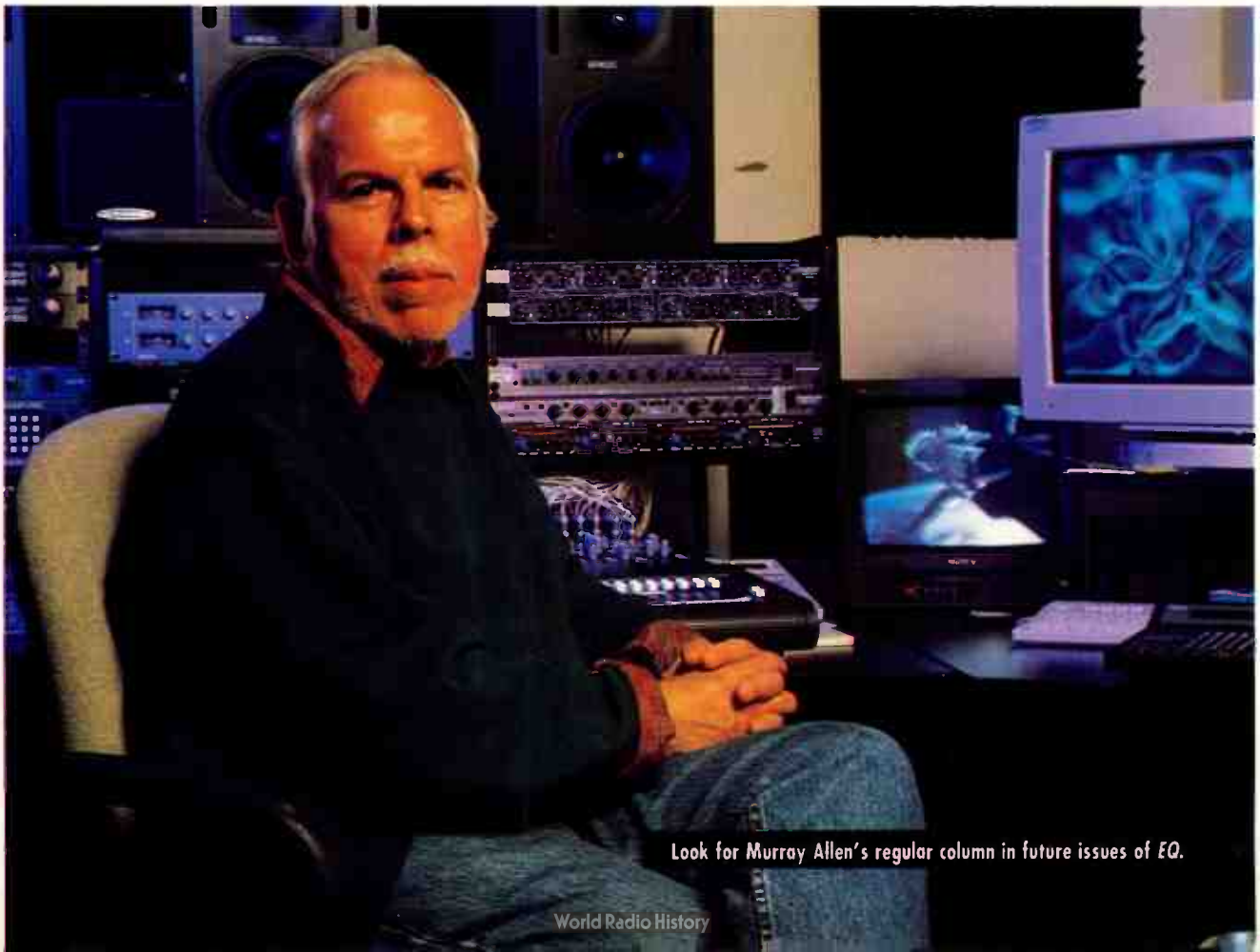

Photo by Steve Jennings

Look for Murray Allen's regular column in future issues of *EQ*.

The SRV-330 Dimensional Space Reverb represents the ultimate marriage—one between a studio reverb with unrivalled sound and Roland's proprietary 3-D technology. Thus, it can replicate virtually any acoustic environment. And the new algorithms and extensive editing parameters give you tremendous creative flexibility. With added features like ducking and fully-controlled gated reverb, the SRV-330 is a must. (Call or write for an audio demo CD of the SRV and SDE-330's, with 79 musical examples featuring various reverb and delay effects. Include $5.00 for shipping and handling.)

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AUDIO FOR MULTIMEDIA

integrated circuits have to be programmed by the microprocessor. In the early days of game development, occasionally the game programmer was responsible for creating the music and sfx. In addition, this same game programmer was responsible for programming the sound system.

THE SOUND DRIVER

As a rule, the game programmer did not want to deal with the sound, and preferring to depend upon a music programmer who knew music and sfx, and also game programming. As games became more sophisticated and tasks more specialized, there was a need for a more highly specialized programmer familiar with the various sound systems on the numerous game decks and PCs and with hardware, and who was also knowledgeable about music and sfx. This programmer wrote a special program called a sound driver.

The sound driver programmer will usually prepare a set of utility programs that the composer and sfx creator can use to create the audio

From Sammy to Grammy

In the '50s, Murray Allen emitted some fine sounds as a woodwind player for bands headed by the legendary swingers — Sammy Davis Jr., Tony Bennett, and Bill Evans. Today, after making great strides in studio sound and audio-for-video recording, Allen has set his sights on the latest frontier, multimedia. As director of audio for Electronic Arts, Allen serves as the point man for what promises to be a multimedia explosion in audio due to his involvement with 3DO, Sega CD, Nintendo, and more. Presently, the patented 3DO technology is at the forefront of interactive CD-gaming technology, combining a live-action 32-bit video system with sonic sound. For a man who used to dream of the day when tape recorders were no longer necessary (a day which has since arrived), 3DO's introduction opens up whole new areas for the audio explorer. Schooled in a variety of studio techniques and musical genres, Allen is enthusiastic about exploring present pathways while laying down new ones.

Defined by a life full of cutting-edge challenges, Allen has remained in sync with changing times by focusing on the opportunities at hand. Whether it meant recording on analog tape in the '60s or dealing with digital devices in the '90s, Allen has been there, innovating. While serving as a studio musician for bands such as The Platters and The Crewcuts, Allen grew dissatisfied with his own sound, leading him to experiment in his home. After buying an Ampex tape recorder and some Altec condenser mics, Allen began recording in-house, a revolutionary move for its time. By the end of the '60s, he had turned himself into a top-notch mixer and was asked to provide his services for a host of hit-makers, Curtis Mayfield and Buddy Miles included. Shortly thereafter he opened his own commercial studio, Audio Finishers.

Things really started rolling, however, after Allen's acquisition of Universal Recording, a Chicago-based recording facility he converted into a superstudio. Over the years — from 1973 to 1985 — Allen's studio was nominated for three TEC Awards as it simultaneously branched out into film soundtrack production (*Top Gun*, *Backdraft*), pushing the limits of what a modern sound facility could do. The turning point for Allen came when he sold off his monster production house in the late '80s and shifted his interests to the burgeoning multimedia market.

"When multimedia came to my attention, I felt that I was seeing a whole new industry unfolding," says Allen. "It was like the recording studio in the sixties — everybody was seeing how far they could push this thing. With multimedia, that's what they're doing now. But this time, instead of magnetic tape and analog recorders, we're dealing with compact discs, data, RAM, and the expanding world of computers."

Working with Electronic Arts, Allen plans to expand the boundaries of interactive sound in video games by introducing live bands, greater interactivity, and new educational touches. Presently, he is also the man the Grammy Awards Show looks to for an impressive sound production.

—Jon Varman

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data for the game program. This finished audio data is given to the game programmer along with the sound driver program. When a musical piece or sfx is required, the game programmer will call upon the sound driver to produce the audio. The sound driver will interpret the data created by the composer or sfx creator and will program the sound chips to recreate the sounds. There may be several different sound drivers for any given game platform, but only one for any given game.

One of the goals of the utility program is to enable the composer to create his or her music with the ability to hear what it will actually sound like on the target platform. A device that we will call an emulator is used for this purpose. Since this device is hardware, it will require the composer to download the sound driver program. It will then interpret the data and produce the same sounds as the game platform.

Another goal of the utility program is to clearly define the limitations of the target platform. The window of acceptability relative to the target platform as well as the emulator gives the composer powerful tools to enable the most creative effort within the limitations of the platform.

Unfortunately, even though many new platforms are using CDs to provide the content, it is presently impossible in most cases to actually play back audio at true CD (44.1 k-16-bit) quality. The graphics are always fighting the audio for data storage space. It is, however, possible in many instances to reproduce 22.05k-16-bit audio. We will discuss this in future articles.

In today's world of sound drivers and utilities the goal is to develop tools that can be MIDI driven. This opens the door to composers and sound designers less familiar with the ins and outs of game programming. This technique will also free up the experienced game composer to spend more time in the true area of creativity. Despite this, many experienced game programmers still prefer to compose their own music and sfx using compiling language.

When a composer is hired to compose music and sfx for a game, the developer will have already selected a sound driver. Unless you are a composer who already has his or her own tried and tested sound driver, you will be expected to use this driver and its utilities and emulator.

Next time we'll build an interactive video sound track. EQ

Studio Notebook

Alex Haas has two new friends!

Audio-Technica AT4033 and AT4051 studio microphones

When we asked Alex Haas, mixer for Eric Clapton's 1993 Grammy song of the year *Tears in Heaven*, about Audio-Technica studio microphones he replied, "The 4033 is very versatile. On my latest sessions I have used it on guitars and as a room mike. It captured the warmth and clarity of Eric's vintage acoustic guitars, and also worked well on his electric guitars."

But that's not the only use Alex has found for these unique microphones. "The AT4033 is an excellent vocal mike. From the crisp top end to the tight bottom end I find that I hardly have to EQ when I use the 4033. I have also been recording a variety of acoustic guitars with the 4051, which sounds clean and clear. On the Fuzzy Logic project with Jeff Bova, I have used both the 4033 and 4051 to sample an enormous range of musical, ambient and environmental sounds."

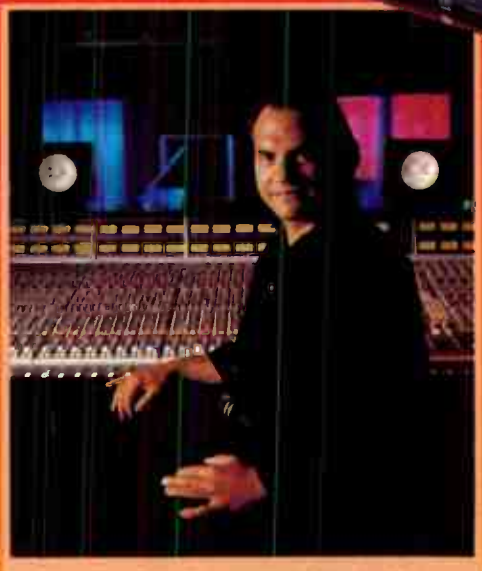

Photo: Skyline Studios, NYC

Alex Haas has mixed for Eric Clapton, the Pat Metheny Group, the Kronos Quartet, the Blues Brothers, Chaka Khan and Michael McDonald, Garland Jeffreys, and many others.

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The Great Macintosh Audio-For-Video Quest

Looking for high-quality Mac-compatible sound cards that will bring better sonics

BY MICHAEL TAPES

I have been working with audio for about 30 years. Audio, sound, and music have given me great pleasure over the years and allowed me to make a good living doing something that I thoroughly enjoy. So I can honestly say that audio has been very, very good to me.

Despite this wonderful relationship I have had with audio, it only took a SMPTE frame or two (at 29.97 fps) for me to turn my back and jump feet first into the desktop video revolution. While I dutifully did my audio thing during "work hours," I found myself wrapped up completely in the world of Apple Macintosh QuickTime (QT) video during my so-called personal time.

I had followed QuickTime from its inception. If I was having a good contact lens day, I could actually be fooled into thinking that those postage stamp-sized QT 1.0 images were playing back at 15 frames per second, even though I knew they were only at 10 fps. In those early days of less-than-perfect QuickTime I turned this video hobby to good use by producing a training video created totally on the desktop for the then-new GUIDE software release for Otari's ProDisk Hard Disk Recorder/Editor. From a technical standpoint, the training video was weak, but Otari's customers were enthusiastically pleased to have a complete run-through of their new ProDisk software right there on their VCRs.

While I believed that this QuickTime technology was going to be very useful, I also believed that it would be a while before the video hardware and software would reach a point where a true industrial-quality video could be produced on the desktop. I never gave a second thought to the audio part since we had been recording "CD-quality" audio on the desktop for years. We just had to wait for the video to catch up.

The Apple Macintosh AV computers were made for multimedia.

Fast forward about a year and a half; the software had arrived. With software such as QuickTime 1.6 and Adobe Premiere, and hardware such as the Radius VideoVision Studio system, full-screen, full-motion (60 field) video was a reality. And Apple's Sound Manager 3.0 gave QuickTime the 16-bit, 44.1 kHz specs that had been promised. My newest training video, for Otari's Concept One audio console, would not only be useful for Otari's customers, but would have good quality as well. I tweaked my Quadra 950 with six GB of disk storage and started the project. I digitized my Hi8 footage and made my rough-cut edits. The project was heading to a great conclusion.

About halfway in, it became time to deal with the audio. No problem. I dusted off my Digidesign Sound Tools system and proceeded to digitize the audio. Not so fast. I needed to update the Digidesign drivers to be compatible with Apple Sound Manager 3.0. Through my industry contacts I obtained a beta release of the new Digidesign driver. Sound Tools now recorded perfectly, but the playback was grossly weird. A few phone calls later, I was informed that

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AUDIO FOR MULTIMEDIA

Digidesign was not going to support Sound Manager 3.0 with the older generation Sound Accelerator card that was the heart of my Sound Tools system. While my simple audio plan was thwarted, it seemed reasonable to me (having designed digitally controlled audio products for much of my life) that Digidesign had to draw the line somewhere. I put the Sound Tools back in the garage and started my Great Mac NuBus Audio Card Search.

My research led me to three products. The first was the newer Digidesign Audiomedia II (AMII) card. It cost about $1100 on the street. It seemed like a safe way to go, but I wanted more for less. This led me to the Pro Audio Spectrum 16 (PAS) from Media Vision. These guys are huge in the PC market, and this board offered the specs I needed for under $400. The only catch was that the PAS didn't have any digital I/O. I would have to go to analog in and out. It seemed fine to me for training videos. I called the MacConnection 800 number around midnight and had the PAS board on my doorstep at 10:00 A.M. the next morning.

I installed the PAS card and software. It was clever, especially at the price. A very nice software panel controlled the four-stereo-input mixer, as well as the playback level and 2-band stereo equalizer. The connector breakout box was small and slick: a bunch of RCA connectors and one 1/8-inch stereo jack, which duplicated the main stereo outputs. At the price, this was quite a package. I recorded a bit through the Macintosh Sound panel. It sounded good. I had found the bargain of the century. I popped open Adobe Premiere and went to set up all the sound input parameters. The problem was that there was no setting for a 44.1 kHz sampling rate. I could set it to various settings of 11 kHz and 22 kHz, but no 44.1 kHz. The only sampling rate that was even close was 44.5 kHz! Thinking that this was a typo in the sampling rate list, I recorded some sound and played it back. It was no typo. Premiere acknowledged that the audio was recorded at 44.5 and when I played it back, it played back at 44.1, which made it a bit slow and down in pitch. The same thing occurred in VideoFusion (a Mac video special effects program I use) and any other Sound Manager 3.0-controlled application that I tried. This would not do. I called Media Vision and of course found out that no one in the entire world had ever seen this problem, and that perhaps I did not know what I was doing. After faxing them some screen shots of a 44.5 kHz pop down menu list, they said, "Oh yeah, Apple

AND THE WINNER: Digidesign's Audiomedia II

screwed up and we'll get back to you." Well they never did. The search continued.

Needing to get on with the project, I borrowed a Digidesign Audiomedia II board from a friend and tested it out. It was perfect. All worked as expected, plus it had S/PDIF. Finally I was able to capture 60-field full-screen video with 16-bit, 44.1 kHz audio. I was about to make the call to purchase the AMII when I checked my research notes and decided to check out the NuMedia board from Spectral Innovations. This uses an AT&T 3210 DSP chip just like the new Macintosh models. In fact, with the NuMedia installed, my Quadra 950 would have Mac AV capabilities. The folks at Spectral Innovations convinced me that the NuMedia was a much better purchase than the Digidesign AMII because I could take advantage of

The Media Vision Pro Audio Spectrum (PAS) 16 is a cost-effective sound-card option.

NuMedia from Spectral Innovations can give an average Mac AV Mac capabilities.

other AV capabilities, such as speeding up Adobe Photoshop. And in a month or so I would get a free upgrade to Dolby DSP compression. This seemed great, especially since I use Photoshop a lot in my video projects. For about $1000, I would get my audio board, a Photoshop accelerator, Dolby compression, and Mac AV capability. I knew that VideoFusion uses the AV to accelerate its processing, so this seemed like the way to go. Another late night 800 call to Mac credit card central, and my latest NuBus sound card was on my doorstep.

The NuMedia card came packaged with Passport Producer and a stock music CD. We were looking good. The card was nice, but no slick breakout box. It had a bunch of 1/8-inch stereo connectors for Mic In, Line In, and Line Out, and the digital I/O was optical. All this was not as slick as the PAS or as convenient as the AMII (a row of RCAs for Line In, Line Out, and S/PDIF), but it was workable.

After a trip to Radio Shack for the adapters, I was in business. I opened Premiere, set the audio parameters, and recorded audio. It was perfect. Now to do some real work and capture audio with video. The audio was still perfect, but now the video dropped frames on a continuous basis (this causes jerky video). I confirmed that the NuMedia card was the cause and phoned one of the engineers. He was aware of the problem and was working on a fix, which would take a week or two. I figured that I could live with that; after all, my Gaussian Blur and Rotate in Photoshop were now over twice as fast!

The weeks passed, no fixes arrived (or seemed to be coming), and additional problems showed up with the board. The input sensitivity was just a little too low, causing me to have to go through a mixer instead of being able to just patch directly. There was no software-controlled input level control (as there was on other boards). The output drive was low. In order to record mono you had to apply signal to both left and right channels or the recorded signal would limit at a low level. Not all AV acceleration worked (the NuMedia is not a bus master, so certain programs such as VideoFusion cannot utilize its DSP capabilities). A discussion with the NuMedia engineer at MacWorld Expo in San Francisco convinced me that the fixes wouldn't be ready any time soon. Next!

Also at MacWorld, the Media Vision folks told me that the 44.5 kHz problem on their PAS board had been fixed. Okay, I was back to the bargain of the PAS card. I got on the phone and got yet another PAS card. The audio worked great, but, as with the NuMedia card, the video frames were dropping like flies. This bargain was much too expensive.

I am now the happy owner of a Digidesign Audiomedia II NuBus audio card. The great Mac NuBus Audio Card Search is over. The only real player (in my research) is the Digidesign Audiomedia II. The product just works and you get Sound Designer II editing software. Because Premiere can export and import its audio files as AIFF, Sound Designer II turns out to be the ultimate QuickTime stereo audio editor because it too can import and export its files as AIFF (or Sound Designer II format). With my great search over, I was finally able to get back to the real work at hand: finishing the training video.

In fairness to the two losing products, by the time you read this, they may have corrected some or all of their problems. Assuming that all three cards work properly, which is a leap of faith as I write this today, I would rate them as follows:

DIGIDESIGN AUDIOMEDIA II

Pros: Does everything it is supposed to do; comes with Sound Designer II software.

Cons: Nothing significant.

For information, contact: Digidesign, 1360 Willow Rd., Suite 101, Menlo Park, CA 94025. Circle EQ free lit. #115.

MEDIA VISION PAS 16

Pros: Very low price; four-stereo-input mixer.

Cons: No digital I/O. (Currently does not allow the capture of high quality video without dropping frames.)

For more information, contact: Media Vision, 3185 Laurelview Court, Fremont, CA 94538. Tel: 800-348-7116. Circle EQ free lit. #116.

SPECTRAL INNOVATIONS NUMEDIA

Pros: Allows some AV Mac capabilities; Dolby compression software is promised.

Cons: Awkward minijack interface; not compatible with all AV Mac applications; some hardware limitations. (Currently does not allow the capture of high-quality video without dropping frames.)

For more information, contact: Spectral Innovations, 1885 Lundy Ave., Suite 208, San Jose, CA 95131. Tel: 408-955-0366. Circle EQ free lit. #117. EQ

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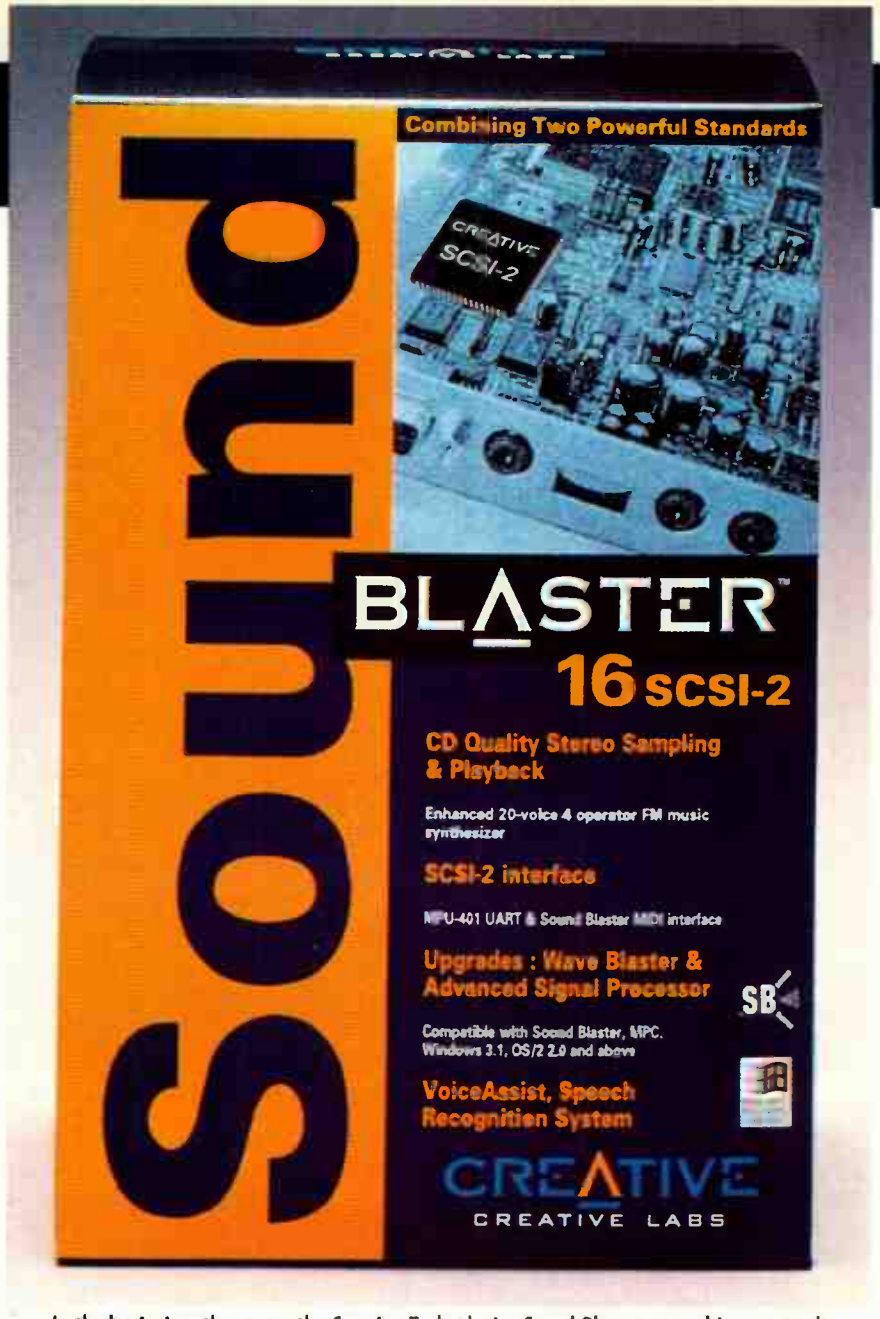
gremlins. If you're planning on using your sound card's digital audio capabilities for casual use, this probably won't be an issue, but if you plan on using a PC card for high-end work, it's a good idea to dig beneath the specs and check for potential audio limitations before you buy.

SOUND-CARD SYNTHESIZERS

Until just a few years ago, the synthesizers on sound cards were so embarrassingly bad they were barely deserving of the name. Of course, for veteran PC owners who were used to hearing just distorted square wave bleeps and bloops out of the internal speaker, these synthesizers were nothing short of miraculous. I guess it just goes to show, as Paul Simon once said, that one man's ceiling is another man's floor.

Early sound-card synthesizers used what was then the only chipset commercially available — Yamaha's OPL2, which provided just two operators of FM synthesis (in contrast to FM keyboard synths, which provided four or six operators). This chipset has 11-note polyphony with a single monophonic output and very limited editing control. As a result, the sounds you can coax out of it are uninteresting, to say the least. You can expect that the brass and string sounds coming from these cards, for example, will be strikingly similar.



In the beginning, there was the Creative Technologies Sound Blaster — and it was good.

More recently, Yamaha released the OPL3 chipset. This provides a four-operator FM system with maximum 20-note polyphony, a stereo output, and more advanced editing features. Though a significant improvement over the original chipset, there are many who would argue that FM is FM, and that FM is old hat. For those who would agree with that statement, many of the newest generation of sound cards instead utilize wavetable synthesis chipsets from manufacturers such as E-mu, Ensoniq, Kurzweil, Korg, and Roland. These chipsets provide ROM collections of sampled waveforms and are roughly equivalent to today's popular sample playback keyboard and rack-mount synths. Some OPL3 cards have provision for optional wavetable synthesizer daughterboards to be connected (such as Turtle Beach's Maui). In the near future, we can look forward to even more advanced sound card synthesizers. For example, industry giant MediaVision has already announced that it will be releasing a "WaveGuide" chipset shortly. This utilizes some of the same physical modeling technology embodied in the brand-new Yamaha VL1 synth.

But have no illusions about the fidelity of sound-card synthesizers — they rarely sound as good as standalone MIDI keyboards or racks. Part of the reason has to do with the crosstalk and clock-noise leakage problems described previously, and part has to do with the relatively limited ROM memory provided by these sound cards, which almost always cost much less than standalone MIDI instruments (street prices for sound cards generally range from about $150 to $500 or so). As a result, the ROM sample data is often compressed and, in those cards that offer wavetable synthesizers, there are rarely multisamples provided. A given voice will therefore only sound realistic over a limited range of notes.

The synthesizer section in a sound card is accessed by software drivers that route standard MIDI files (.MID files) from within applications such as MIDI sequencers or multimedia programs. The Windows Media Player utility can also access a sound-

Avoiding The Installation From Hell

You know the weird guy who hangs out on the corner muttering to himself? Well, this man was probably the chairman of a major corporation right up until the day he attempted to install a sound card in his PC.

Actually, things aren't that bad. The main problem seems to be a proliferation of owners' manuals from hell; while there's nothing we can do about that, we can offer a number of tips that should make your sound-card installation fairly straightforward if not a complete piece of cake.

First off, your sound card will come bundled with a bunch of software that needs to be installed before you can use the card. No big deal, right? Well, pretty much any time you install a new piece of software in a PC a whole array of files — not just one or two — will be copied onto your hard disk, often in different locations. What's more, a few of your most important system files — usually your boot-level AUTOEXEC.BAT and CONFIG.SYS files, as well as the SYSTEM.INI file in your Windows directory — will be altered somewhat. This can be a problem if you ever need to pull your sound card for any reason: you'll need to remove lots of files and you'll also have to restore your system files to their previous state, otherwise, your computer will freak out when it can't find the card it expects to see.

A few manufacturers have actually taken pains to document all these alterations and movements so that, if necessary, you can get your computer back to the state it was in before the sound card ever left the wrapper — but most don't. So here's the plan: Before installing any new hardware or software, get out a floppy disk and create a directory named "CLEAN" or something like that. In that directory, copy the current AUTOEXEC.BAT and CONFIG.SYS files from your boot disk's root directory. Then go to the SYSTEM subdirectory inside your boot disk's WINDOWS directory (normally C:\WINDOWS\SYSTEM) and copy the SYSTEM.INI file. Don't change anything at all — just copy these three files. If you then ever need to deinstall the sound card, all you'll need to do is copy these files back to the places they came from.

Now let's define a few terms. First, all PC peripheral devices — including sound cards — need to establish a unique address that allows the host computer to find then. This address is usually expressed as hexidecimal numbers, with the letter "H" preceding a three-digit number. Second, each peripheral must establish a unique interrupt level (usually called an IRQ); put simply, this is an electrical signal sent by the peripheral to the host computer in order to get its attention. Standard PCs utilize sixteen IRQ values numbered from 0 to 15, and some IRQ values are reserved for specific devices. Other IRQs are unassigned and can be used by any attached peripheral. Finally, sound cards that provide digital audio recording/playback capability must also utilize a Direct Memory Access (DMA) bus for the high-speed transfer of such data to and from the CPU. There are numerous DMA channels (the exact number varies in different computer models), and your sound card must use a unique DMA channel that is unused by any onboard functions or by any other peripheral in your system.

To summarize, for a sound card to peacefully coexist with other peripheral devices in your PC system, it must be set to a unique address, IRQ, and DMA channel. If there are any conflicts with other devices, the likelihood is that your computer will hang up and that you will begin developing that ulcer you have so carefully been avoiding for the past ten years or so. So—preparation is the key. Before installing anything—before even opening the cover to your PC—sit down with a good old analog pencil and paper and make a list. Write down the names of all the peripheral devices in your computer and, next to each name, write down the address, IRQ, and (if used) DMA channel assigned to that device. This information can be found in owners' manuals or by opening the Drivers applet in the Windows control panel and selecting the driver, then clicking on the Setup button. Now look in the owner's manual for the sound card you're about to install and write down the default address, IRQ, and DMA channel used by that card. Check your list. If any other device in your system uses the same address, IRQ, or DMA channel, you'll need to change either the old device or the new sound card (make sure to write down any changed values).

It will probably be easier to make any such changes in the new sound card, since it's still in its wrapper and not already seated in an expansion slot (you did heed my advice to not open the cover yet, didn't you?). The changes can sometimes be made during the software installation, but more often than not require that you move a jumper (a small piece of plastic that connects pins) on the card itself. Don't panic—this doesn't require a techie or a visit to your local authorized service center. All it requires is a pair of needlenose pliers and a gentle touch. The owner's manual for your sound card will include a diagram that shows where each jumper is located and describes its function. Use the pliers to move any jumpers necessary. Then, with the PC turned off (and preferably disconnected from AC), go ahead and remove the PC's cover. Locate an open expansion slot, then ground yourself (either by using an antistatic grounding strap or by touching a piece of bare metal) before picking up the card. Gently seat it in the slot connectors and be sure to replace the slot screw — this is important for electrical grounding as well as for ensuring that the card is seated properly. Finally, replace the cover and turn on the power.

Now it's time to install the software. Be sure to precisely follow the installation procedure described in the manual and to read any README-type files that may be included; these usually contain late-breaking information about the card and/or addendums to the printed owner's manual. After installation is complete, you'll usually need to run some first-time configuration utilities, often from DOS as well as from Windows.

Installation of Windows drivers can sometimes be a little problematic: you'll probably need to open the Drivers applet from the Windows Control Panel and click on the Add... button, then on the "Unlisted or Updated Driver" option in the resulting dialog box. Then place the installation floppy in the drive and click on the OK button; a list of one or more drivers should appear. Select the one you need (the sound card owner's manual will describe each to you) and click on OK. You'll need to restart your computer for the driver to be activated, so click on the resulting Restart Now button. When Windows returns to you, you should be in business. If you've followed all the tips here and things still don't, call your sound-card manufacturer's tech support line. —*Howard Massey*

card synthesizer via another Windows utility called the MIDI Mapper. There are also software utilities that allow you to play synth voices from the computer keyboard, or, if the sound card provides a MIDI interface, via an external MIDI controller. Most current PC sound cards organize their preset voices according to the General MIDI spec; a few provide alternative Roland MT-32/LAPC or Sound Blaster compatibility.

SOUND CARD MIDI INTERFACES

Sometimes you just have to be in the right place at the right time, and sometimes you just have to be smart enough to recognize a good thing when you see it. Back in the early '80s, the folks at Roland were both, manufacturing an excellent PC MIDI interface card (the MPU-401, later called the MPU-IPC) that added one MIDI in, two MIDI outs, and an analog (FSK) sync input/output. As a result, the MPU-401 has been the standard in PC MIDI interfaces for almost as long as there has been a MIDI. Despite attempts by other manufacturers to introduce competing products, the MPU-401 was supported by virtually all DOS MIDI programs, and its popularity continues even into the Windows era.

You'd think that sound-card manufacturers would be aware of this, wouldn't you? Nah. Up until just a year or so ago, most PC sound cards that provided MIDI in/out (and not all did) stubbornly used their own proprietary interface and software driver. More recently, however, they seem to have tired of swimming against the tide and the result is that many of today's sound cards offer MPU-401-compatible MIDI interfaces. There is a downside to this generally positive trend, however: if you already have an MPU-401 or MPU-IPC card in your PC, you'll almost certainly have to pull it when installing one of these newer sound cards in order to avoid potential conflicts (older sound cards that use proprietary MIDI interfaces can usually coexist comfortably with MPU cards).

OTHER SOUND CARD FEATURES

A fair number of PC sound cards provide a CD-ROM interface in order to help satisfy another MPC criteria — the presence of a CD-ROM drive. Most, however, utilize a proprietary connector that often works with only a few specified drives. A few provide a CD-ROM interface disguised as what is claimed to be a "generic" SCSI connector; however, since SCSI in the PC world isn't nearly as standardized as it

AUDIO FOR MULTIMEDIA

is in the Mac world, you may find that this, too, only works with a few specified drives.

Many PC sound cards come bundled with a microphone and/or computer speakers. The microphone is usually of the low-fidelity dynamic type and is best avoided if you plan on recording music as opposed to speech. Computer speakers are distinguished only by the fact that they are compact and magnetically shielded in order to reject interference when placed close to a CPU and monitor, but they generally range in audio quality from poor to fair, so don't even consider using them for serious monitoring purposes.

Some newer sound cards have onboard DSP chips for the addition of effects such as reverb and chorusing. Naturally, this will help sweeten the audio, smooth nasties in the sounds, and reduce the need for using outboard devices. Don't expect anything as fancy as the multieffects processors in today's keyboard and rack-mount synths, though — the cost limitations imposed by the sound-card market pretty much guarantees a no-frills approach to DSP.

Sound-card inputs and outputs, as mentioned earlier, usually are of the 1/8-inch Walkman variety, though a few offer RCA connectors instead. Some cards provide separate mic- and line-level inputs; more commonly, a single Mic/Line input is used with the input level strength switchable via software. Many cards provide an additional line-level CD/Aux input; an on-screen software mixer usually enables balancing of the various input and output signals.

Speaking of software, there are almost always a number of DOS and Windows utilities tucked away in the sound-card packaging. These range from the ridiculous (Creative Labs' Talking Parrot, for example) to the sublime (Turtle Beach's Wave SE editor or Voyetra's various multimedia utilities, including the exceptional Audiostation), but usually can be counted on to provide various ways of routing data to and from the sound card's digital audio and synthesizer

sections as well as any connected CD-ROM drives. A recent trend in sound-card software goodies is the inclusion of voice-recognition utilities. A few even include multimedia authoring programs.

THE UPSHOT

Do you need a PC sound card? Well, if you own a PC-compatible computer and use it for anything other than pure business, the answer is almost certainly a resounding yes. If you're a PC owner who's also a casual musician, the answer is an even more resounding *yes*. But if you're a pro or semipro musician who already owns some MIDI gear, the truth of the matter is that the equipment you're using now is probably already better than what you'll find in most sound cards. Bear in mind that you won't need a sound card to enable the use of a Windows or DOS MIDI sequencer with your external equipment — in that case, all you'll need is a PC MIDI interface, probably of the MPU-401 or 401-compatible variety. Even if you do use a sound card to record .WAV soundfiles, be aware that Windows imposes some timing problems during playback and that reliably syncing soundfiles to MIDI sequences is currently virtually impossible. If you're hoping to use your PC for serious digital audio work, you'll probably get better results from an external device that provides A/D and D/A hardware under PC control.

But — and this is a big but — no matter who you are, if you want to use your PC to have some fun, a sound card is definitely the way to go. Plus it's another way to tick off those guys in suits.

Howard Massey is a MIDI consultant and technical writer who has authored numerous books on music technology, including The Complete Sound Blaster. *In his spare time, he is attempting to develop the world's first MIDI-controlled German Shepherd.*

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EQ LIVE

LIVE DRUM MIKING TIPS — PAGE 64

LIVE SOUND FOR THE GIGGING MUSICIAN

ON THE ROAD TO RUIN AGAIN

THE WORLD'S PREMIER PUNK BAND HAS SEEN AND DONE IT ALL — AND THEY'RE NOT DONE YET

BY JOEY RAMONE

■*ON FEBRUARY 9, 1994, Tokyo, Japan, the Ramones played their 2000th show. To celebrate this event, as well as the band's 20th anniversary and the success of their most recent release,* Acid Eaters, EQ *correspondent Joey Ramone talked with his soundman of some 15 years, John Markovich, about life on the road with the Ramones.*

Joey: So how did you hook up with us?

John: Originally I owned my own sound company and I was doing shows in the Ohio area. I used to do a lot of work with Kenny Slater of Blue Sky Records, who took care of Rick Derringer and Johnny Winter, and he asked me to do a favor for a friend of his. At the time I took the job I really didn't know who it was till I got to New York and met everybody. Although, we had done a Ramones show a few months before that...

Joey: Was that at the Tomorrow Theater?

John: Yeah, in Youngstown. The way that came about was that Kenny Slater said, "I have a friend who needs a soundman for a group that's going to be opening for Iggy Pop." When that tour ended you guys went on to record the live album [1978's highly acclaimed *It's Alive*, so far unreleased in America, which was recorded at Lon-

AND THE BEAT GOES ON

TIPS FROM REAL LIVE DRUMMERS (AND AN ENGINEER) ON HOW TO GET BETTER SOUND THROUGH MIKING

BY TERI SACCONE

■WITH THE PENDULUM swinging back to acoustic drums in recent years, the importance of live drum miking techniques has increased significantly. Considering the regularity of technological audio advancements, professional drum-miking techniques follow rather simplistic guidelines, yet concertgoers can still find themselves at a gig where improper miking can debilitate the sound of the drumkit. And, as anyone can attest, when live drums go south, the rest of the sound usually follows fast.

With just a few basics kept in mind, the live drummer can easily capture the sound he or she desires and one that the audience has come to expect (usually the sound most similar to that of the album).

The optimum drum sound today is one that is resonating, natural, and — above all — live. The ideal is to obtain that sound with minimal modification or tampering when miking the drums. Noise gates, reverb units, and other assorted effects can be added later if necessary.

With regard to size, drum miking is one area where smaller is definitely better. With the exception of bass drums, the rest of the drumkit will benefit from smaller mics. The more compact the mic, the less obstruction the player has to contend with when hitting the drums. Also, smaller mics can result in better isolation of each individual drum sound. Some manufacturers offer individual clip-on cymbal miking systems that further help to inhibit sound washover, and can — in some live situations — eliminate the use of overheads, resulting in a cleaner look.

COUGAR SKINS: John Mellencamp's drummer Kenny Aronoff.

Mic positioning in a live context is similar to studio miking. Placing a single mic from three-quarters of an inch to one inch from the top head is recommended for both rack and floor toms and snare drums, going in 1/10th of the diameter of the drum. Where bass drums are concerned, the usual manner of miking is through a hole (from 6 to 8 inches in diameter) cut out of the front head. The mic is normally placed either close to the batter head (within 2 to 4 inches) or sometimes clipped on to the exterior of the hole. There are no hard and fast rules here because some drummers don't use front heads on their bass drums to begin with, while others use only outside mics (John Bonham was a proponent of this). It's best to work with different techniques to obtain the most individualized sound.

Acoustics are another important consideration in drum miking, as venue size varies, and soundchecking an empty venue in lieu of a filled one requires compensation through the PA and the mixing board, as well as through the miking system. Again, experimentation helps here, as does good communication between drummer and engineer.

Finally, it's important to bear in mind that any kind of dampening will affect miking. For that open, ringing resonance, it's advisable to keep the drums free of taping and dampening (although bass drums are usually dampened to some degree via a pillow or blanket). If a wetter sound is desired for the top kit, any enhancements can be added later through gating.

The "experts" who spend the majority of their time on the road have their own sets of priorities with regard to miking. Drummer Kenny Aronoff (John Mellencamp, Iggy Pop, Meatloaf, Lyle Lovett, Bob Dylan) maintains that one of the most important factors in the miking equation is the venue. "The ambience and sound of the room is the main priority," says Aronoff. "Similar to studio miking, in a live performance the idea is to get the actual sound of the drum captured up close. The difference is basically that in the studio, you're working in a smaller room, and when you play live you use the PA to project your drums to fill up the room."

Aronoff is renowned for his distinctive snare drum sound. To reproduce his crackling, deep snare attack live, he opts for the single mic method. "I don't put a mic underneath the snare, only one from the top," he explains. "I usually use the Shure SM57, and it's about 1/2-inch into the drums, coming in from the rims. Usually, the closer in you come to the drum and the higher you come down into the drum, the fatter, warmer, and more full-bodied the sound. The farther out you go from the drum, the more crack and ring you'll get. I like the combination of the two: the fat and the crack, so it's just not tinny.

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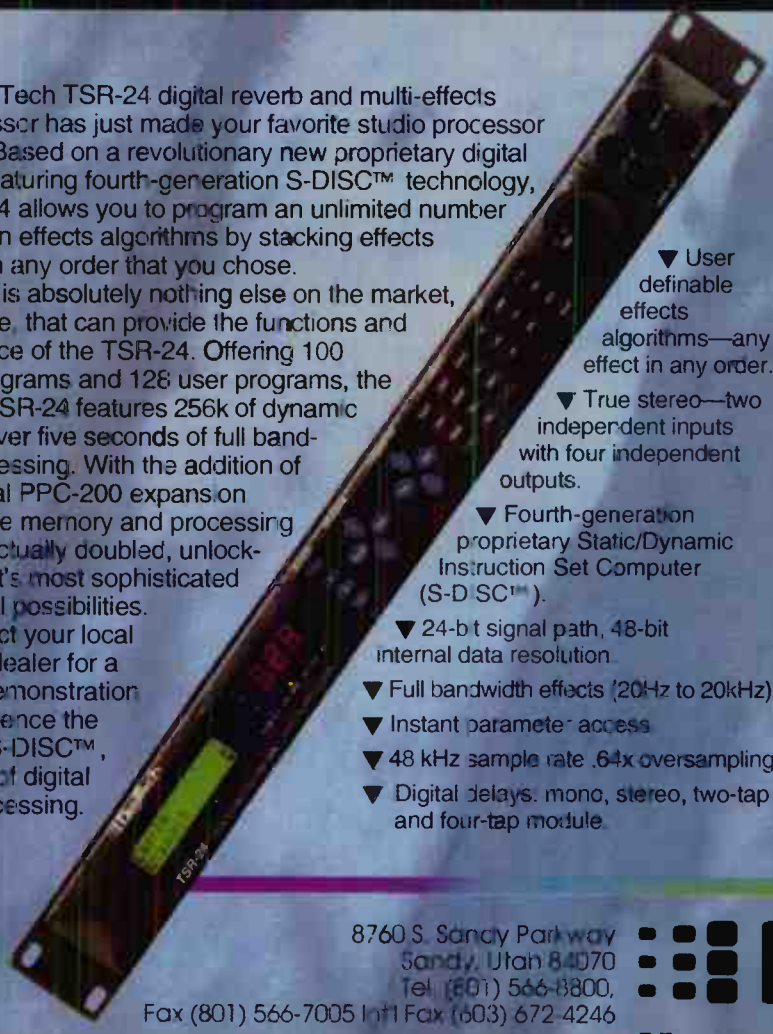

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"But a lot of how you mic has to do with how you hit and how you tune the drums," Aronoff continues. "I play a lot of rim shots, and I always play with a lot of power. In fact, I tend to overpower a lot of mics, so I stick with the AKG D-112 'football mics' (named for their oblong shape) on my bass drums. On the toms, I use the teeny clip-on mics — AKG's C-408's — that attach onto the rim system. You can't see them, so visually they're fantastic. And since they're so out of the way, you don't have to worry about hitting one of them and obliterating it."

For cymbals, Aronoff opts for the individual cymbal mics. "I use the Zildjian Z-MC's," he says. "They're great for catching each cymbal and there's a lot of attack. The fewer mics you use on stage, whether overheads or clip-ons, the fewer problems you have with sounds bleeding into other sounds, so I try to keep it all at a minimum. With an individual cymbal mic stuck under a cymbal, engineers can turn it up or down as they want. That's the beauty of it."

Drummer Danny Gottlieb (Pat Metheny, Elements, Mike Stern, The Blues Brothers) has spent most of the last 17 years touring. Because he often plays in a jazz context, his miking needs are different from those of Aronoff. "Since I play in a lot of European clubs," comments Gottlieb, "I've experienced everything from elaborate miking of the drumset, to one overhead, to no mics at all. And when it comes to miking that's suitable for the drumkit, it really comes down to the acoustics of the room. So I don't have any set rule for miking that will work for every type of room. My advice is to be aware of the full range of the drumset and to try to get as good a sound on it yourself. Hopefully, you'll have someone with good ears who can reproduce that sound in the context of the club.

"What makes my playing a little bit different from that of a lot of drummers is that I'm very cymbal-heavy," he adds. "The time that I play really comes from the ride cymbal and I use those flat Paiste 602's, which have a very brilliant, crystallike

sound. Plus the musical context is often very sensitive, with a lot of dynamics from very soft to very loud — except when I'm playing with the Blues Brothers in front of

continued on page 82

Sampled Drums

Being properly miked shouldn't involve the likes of a quest for the Holy Grail. With an acoustic kit, it's a pretty straightforward affair. Live miking plays a vaguer role when sampled drum sounds enter the picture. Although acoustic drums are often preferred, there is still a large contingent of players who rely on sampling, speculates Chris Tso, professional audio manager at Manny's Music in New York.

"Drummers are sampling more often than not because the audience expects the artists to sound similar to the way they sound on the recorded product," Tso says. "So a certain sound that maybe the producer wanted on the record has to be reproduced live. The way to do that on a big level is to use either pads or, ideally, whatever drumset the drummer feels most comfortable playing, and trigger off those heads.

"There's a big swell back to unplugged now," Tso continues. "Now, a lot of drummers are augmenting acoustic drumkits with electronic samples so that they can have their drums the way they normally sound, and they can also get custom sounds that may not be derived from their acoustic kit."

With sampling being prevalent with a number of major acts, how, if at all, does that affect the function of miking? "If the electronics are only augmenting an acoustic kit, it's not going to effect the live miking," responds Tso. "[The signal] might go through a drum mixer onstage and then out to the house already premixed, but it depends on the custom set up. To get a good sound, there's no getting away from it: a good drum miking technique is important, even in these electronically augmented situations."

Photo by Julian Jaime

Unsanity

BAND NAME: Unsane
MEMBERS: Chris Spencer (guitar/vocals), Peter Shore (bass/vocals), Vinnie Signorelli (drums)
LATEST RELEASE: *Total Destruction* (Matador/Atlantic)
WHERE THEY'VE BEEN: To hell and back — twice. Actually, the States, Europe, and Canada
WHERE THEY'RE GOING: More U.S. and European dates
HOW THEY GET AROUND: 1988 Chevy Ten van (130,000 miles)
GUITARS: '72 Telecaster; '79 Telecaster with Seymour Duncan 54's; '76 Rickenbacker 4001; '80 Rickenbacker single-input; '72 Rickenbacker 4001 Special Edition
AMPS: Custom Fender Twin 300-watt head with 2 x 12 Celestion speakers; 2 x 12 Vox cabinet; GK 400 200-watt head; Ampeg 1 x 15, 4 x 10 cabinet; Crate 150-watt 1 x 15
DRUMS: Tama Artstar II, Zildjian cymbals
EFFECTS: First, Second, and Third Edition Rat distortions
MICROPHONES: [2] Shure SM57's
BEST LIVE EXPERIENCE: Peter Shore states: Emo's in Austin, TX. Great show. Great crowd. Great bartenders. Great employees. Much fun was had by all. It was extremely loud and we sounded really good. The packed crowd was really into it.
WORST LIVE EXPERIENCE: Shore continues: We played in a mall in Melbourne, FL to about ten people, which included the bartender, the promoter, and the promoter's friends. Not only was the place empty, but we sounded terrible. The next day, we had to get a check from the promoter's mom and had to go to her bank to cash it before we could get the hell out of that state. We hate Florida. Sometimes. EQ

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Circuits Maximum Oracle in-ear monitoring system

31 FLAVORS

The new Q™ 231 graphic equalizer from Peavey is a dual 31-band graphic equalizer. While designed to be cost-effective, the Q 231 still includes a list of features normally found on more expensive units,. It allows stereo EQing of the main PA or dual-channel monitor equalization. When a mono PA is used, the Q 231 allows one channel to be used for the main PA, with the other channel used to EQ the monitors. The Q 231 features an internal power supply and is housed in an all-metal chassis with controls that are easy to read and adjust, a solidly built circuit board, and easily serviceable components. For complete information, contact Peavey, 711 A Street, Meridian, MS 39301. Tel: 601-483-5365. Circle EQ free lit. #120.

Peavey Q 231

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The FBX-1802 feedback exterminator is the latest addition to Sabine's line of automatic feedback controllers. It's a single-rack 2-channel upgrade of the FBX-900, with additional features. Like the 900, the FBX-1802 automatically senses feedback, determines its frequency, and places a digital narrow-notch filter to cancel the feedback with virtually no

effect on the rest of the frequency spectrum. It has nine filters per channel and provides more quickness, accuracy, and gain before feedback than its predecessor. You can choose to lock the 1802's filters to prevent the filters from going deeper. You can also select the total number of filters to be activated, as well as the width of the filters. In addition, peak output is increased to 23 dBV. For further information, contact Sabine, 4637 Northwest 6th Street, Gainesville, FL 32609. Tel: 904-371-3829. Circle EQ free lit. #121.

Sabine FBX-1802 feedback exterminator

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MAS/West Test 1-2-3

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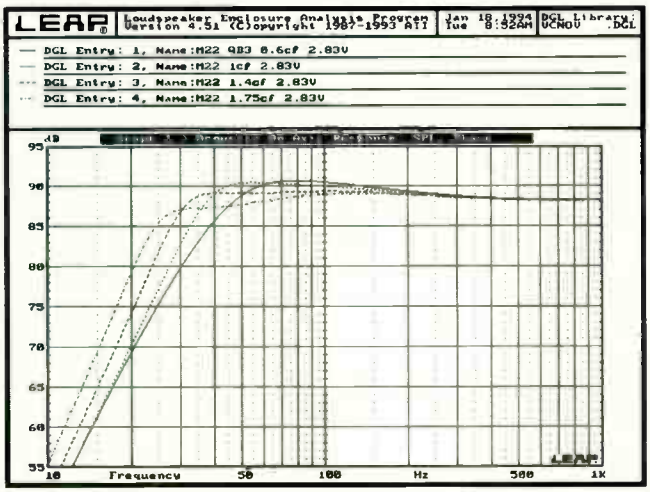
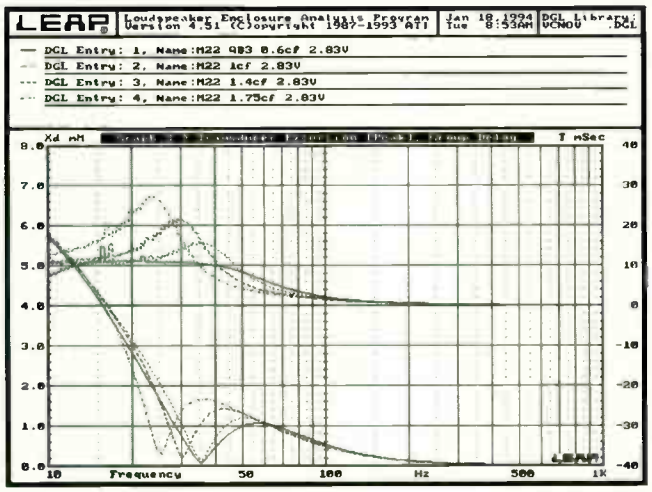
BY VANCE DICKASON

■DESIGNING AND BUILDING your own vented loudspeaker cabinets, such as designing a subwoofer to augment a commercial studio monitor, has become a fairly trivial task. Thanks to the published work of engineers like Olson, Beranek, Novak, Thiele, Small, and others, the mathematics and design procedures for accurately generating the proper volume and vent-tuning frequencies for this type of enclosure are readily available and fairly easily implemented by any competent audio technician or engineer.

Coming up with the appropriate volume and vent dimensions can be accomplished by either solving the pertinent design equations using a hand-held calculator, or by using one of the numerous computer programs available for this purpose. Using a scientific calculator and the Thiele 4th-order model equations is a fairly painless process. Step-by-step details of this procedure are widely available in print and can be found in various reference books such as the anthology *AES Loudspeakers, Volume 1*, the collection of Bob Bullock's Speaker Builder box articles titled *Bullock on Boxes*, or in my book, *Loudspeaker Design Cookbook* (all available from Old Colony Sound Lab; tel: 603-924-6371).

Given that driver parameters have been appropriately measured, the Thiele/Small signal model is sufficiently accurate for the constructed enclosure to measure within a few Hz of the predicted low-frequency roll-off. While the calculator method is reasonably fast and not very difficult to use, most of us would just as soon avoid solving equations whenever possible and instead rely upon the computer.





Figures 1 (top) and 2

GET WITH THE PROGRAM

As editor of *Voice Coil*, the newsletter for loudspeaker manufacturers, I have reviewed more than 20 different loudspeaker computer programs over the last three years. These programs have ranged in price from $50 to over $1000, and all provide a variety of features for streamlining the box-designing task. The vast majority are Thiele/Small calculator programs that incorporate the equations presented in the original AES articles along with the added ability to graph the predicted response, impedance, and power handling curves. Their cost is moderately low, between $50 and $200, and versions are available for both the PC and Macintosh computers. This type of software makes the process a simple matter of entering the usual set of speaker parameters (Re, Sd, Qes, Qms, Qts, Vas, and Xmax), selecting the type of enclosure desired, and letting the computer do the rest of the work. The advantages are convenience and speed, with the results being approximately the same as those obtained using the calculator method.

Bob Bullock's BoxModel and Joe D'Appolito's Top Box are two good examples of the software that's out there. Both have a variety of useful features, such as the ability to compare several designs simultaneously as well as to design speakers with augmenting electronic filters. Among professionals, however, the industry seems to have widely embraced the LEAP (Loudspeaker Enclosure Analysis Program) software from LinearX, Inc. as the weapon of choice.

LEAP is easily the most advanced loudspeaker-simulation software currently available, performing simulations not only of low-frequency enclosures, but also for active- and passive-network designs. Regarding box-design features, however, LEAP goes significantly beyond the original 4th-order model. It offers a number of serious enhancements such as the incorporation of frequency-dependent loss variables; nonlinear modeling of Bl (motor strength), compliance, and port functioning; and the ability to analyze an enclosure at both small and large signal levels. To get a better perspective on the power of

DESIGN YOUR OWN SPEAKERZ

How many times have you thought that you could do a better job of design ng a speaker than the speaker company? Twenty-five years ago I bought some speaker components, crossovers, and speaker box plans (about the size of UREI 833's) from JBL. I cut the wood and glued the boxes together. Then I started experimenting to see if I could make them sound better. I whipped out all of the audio-design books and my K&E slide rule and started calculating. I made the boxes out of thicker wood, changed the port size, changed the crossover components, changed the damping material, changed my mind. It was a mess. I still have the speaker system. I have been carrying it around thinking that I would get to it again real soon now. I have even had the woofers reconed twice because smog ate the surrounds.

Bingo! A Physicist named John Murphy wrote a program called MacSpeakerz that will solve all of your speaker-system design problems. I stumbled upon a demo copy of the program on America Online. I downloaded it and started messing around. I was hooked. I called them up and bought the real version.

MacSpeakerz is a CAD program that allows you to design a very-high-performance speaker system with optimized frequency-response characteristics. The program contains the mathematical models for both vented and closed speakers of either rectangular or trapezoidal shape, crossover components, and a database of over 670 loudspeaker drivers.

I am not the only person who thinks this is a great design tool. MacSpeakerz is used by Apple Computer, Yamaha, Kenwood, Acoustic Research, Fostex, Nakamichi, 20th Century Fox, and Maryland Sound — and the list goes on.

By the time you read this there will be a PC version called WinSpeakerz. I think this program is so good that I am going to do a full review of it in a future issue.

—*Roger Nichols*

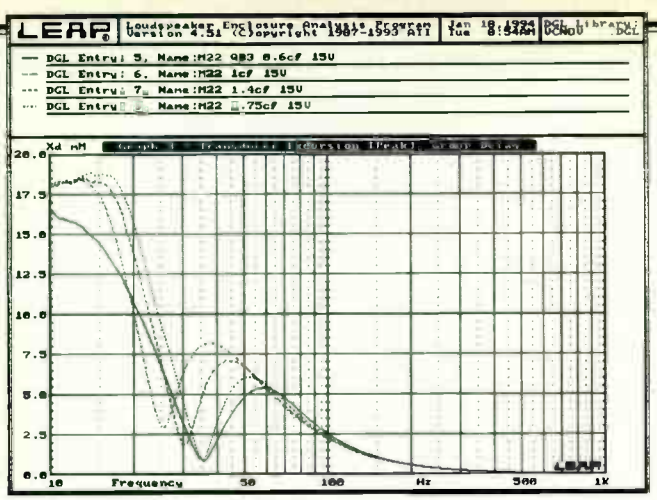
this type of software modeling, I performed an analysis for a typical 8-inch woofer. The speaker's parameters were derived using the LinearX LMS analyzer and the LEAP program as follows:

Revc= 4.2 Ohms; Qms=1.96; Sd= .0206 M2; Qes=0.30; Vas= 82.3 liters; Qts=0.24; Fs= 24.6 Hz; and Xmax=6.5 mm

Using the LEAP Quick Cabinet utility, which provides starting values for cabinet volumes and port tuning frequencies, and suggested box volumes and tuning frequencies from the manufacturer, I programmed LEAP to simulate the response of this woofer in 0.6-, 1-, 1.4- and 1.75-cubic-foot enclosures. The 0.6 box is a QB3 (Quasi 3rd-Order Butterworth response) alignment that provides the best possible transient performance from a woofer in a vented box (woofer alignments are part of the terminology used in the original Thiele work to describe a particular response category). The port area for each of the enclosures was made as large as possible given the depth of the enclosure for the given volume (for a specified port tuning frequency, as the area of a vent is made larger, the vent becomes longer).

IT FIGURES

Figure 1 illustrates the response of the four simulations, each analyzed at a nominal 2.83 V input level.



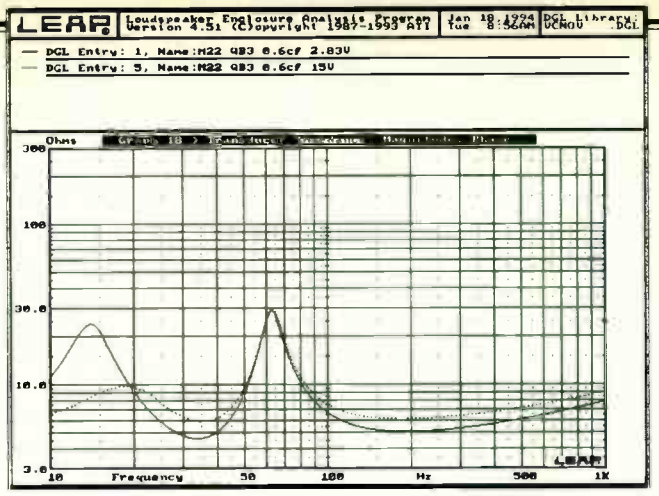


Figure 3 (left) and 4

The –3 dB low-frequency roll-off varies from about 46 Hz for the QB3 alignment to about 23 Hz for the largest box, which is exactly one octave difference. Low-frequency performance in any loudspeaker, however, is always a trade-off between damping and excursion. The larger the box, the less control the motor has of the cone.

Looking at the cone excursion curves and group delay curves in fig. 2, it is easy to see that as the box size gets larger, the excursion increases and the damping, as interpreted from the group delay curves, decreases. The shape of the group delay curve for the QB3 enclosure is very shallow and close to the shape of the group delay curve for a Qtc=0.7 sealed box. As the transient performance of the box/woofer combination decreases, the magnitude of the group delay curve increases and the shape becomes more of a "knee."

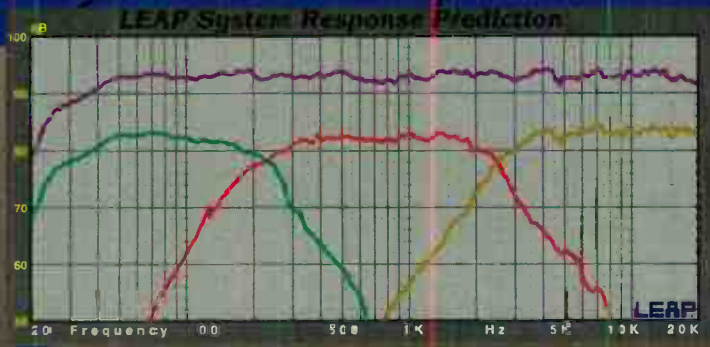
While it is obvious that the larger box has greater group delay magnitude, the consequences of the excursion curves are difficult to interpret at a nominal 1 watt level. Figure 3 shows the 15 V input excursion curves for the same set of enclosures. The output of the woofer has increased to over 102 dB (not shown), but the large enclosure exhibits an excursion above the port tuning frequency (indicated by the dip in the excursion curve) of about 8 mm at 35 to 40 Hz. The maximum linear excursion for this driver, or Xmax (the voice coil length minus the gap height divided by 2), is only 6.5 mm. Using the 15-percent criteria, the woofer

continued on page 83

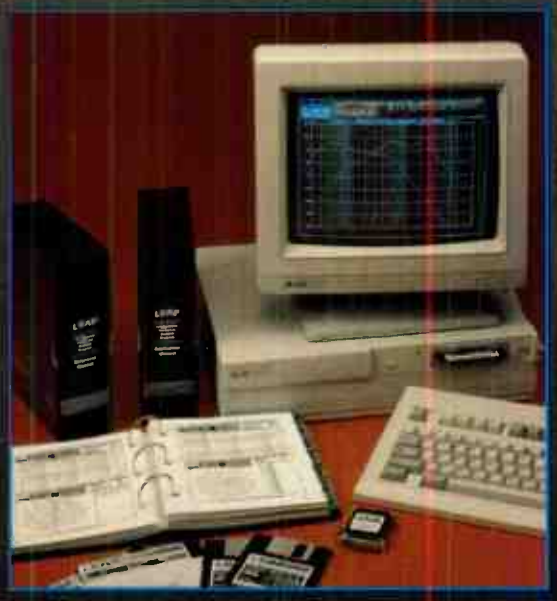
New!
LEAP
Version 4.5

LEAP

Precision Development Tools for Precision Loudspeaker Designs



The Art and Science... of loudspeaker system development today has become more complex than ever before. Competition is tough, and to compete each design must perform to the best of its ability, and make the most out of every dollar's worth of transducer cost. The simple approach of choosing a combination of seemingly appropriate transducers coupled with ordinary networks and filters, has given way to a painstaking process of meticulously blending selected transducers in combination with carefully devised and matched crossover designs.



LEAP (Loudspeaker Enclosure Analysis Program) is a complete full range analysis package which provides virtually all of the tools necessary to develop precision loudspeaker systems for today's demanding audio markets. Whether your applications are consumer audio, car stereo, professional audio, or custom esoteric markets, LEAP provides the power, flexibility, and accuracy to investigate every possible design permutation. The open architecture and broad spectrum of features provided will dramatically reduce your development time, while improving the quality of the final results, and demonstrates why LEAP has become the #1 choice of professional loudspeaker designers worldwide!

Advanced System Analysis Features

- ✔ Use simulation or imported actual measured SPL/Z data.
- ✔ 5-Way crossover system modeling, and more.
- ✔ Time offset between transducers.
- ✔ Active or Passive based crossovers.
- ✔ Hilbert-Bode transform for deriving phase.
- ✔ 22 Passive components per xover section.
- ✔ 16 Active filter blocks per xover section.
- ✔ Passive Network Optimizer for single system response.
- ✔ Active Filter Optimizer for single system response.
- ✔ Frequency ranges from 1Hz to 100kHz.

Printer/Output Formats

When you wish to produce a hardcopy output of your finished designs and graphical data, LEAP supports a large number of printer standards, and even supports numerous desktop publishing graphic formats in both black & white and color! Portrait/Landscape orientations in any custom size and aspect ratio are user controllable.

- ✔ IBM/Epson 8 Pin Dot Matrix
- ✔ Epson 24 Pin Dot Matrix
- ✔ HP LaserJet Series Printers
- ✔ HPGL-Compatible Plotters
- ✔ PostScript EPS/TIF B&W,Color
- ✔ AI-Adobe Illustrator B&W,Color
- ✔ NEC 24 Dot Matrix
- ✔ TOSHIBA 24 Dot Mat
- ✔ HP DeskJet 500C
- ✔ PostScript Printers
- ✔ DXF AutoCAD
- ✔ TIFF, BMP, PCX Plots

Advanced Transducer/Enclosure Simulations

- ✔ Sealed,Vented, Bandpass, PR Simulations with multiple speaker/ port capability.
- ✔ Large signal analysis of TempVC, and Non-Linear BL/ Ports/ Compliance.
- ✔ Acoustic Parallel or Acoustic Series (isobaric) Driver Mounting.
- ✔ Port Standing Wave resonance modeling.
- ✔ Frequency Dependent Rvc and Lvc modeling.
- ✔ Library storage of 99 transducer parameters, and over 24 enclosure parameters.
- ✔ Complex transducer modeling of electrostatic, ribbon and piezo devices.

Extensive Documentation

The two volume manual set comprises almost 1,000 pages of documentation which thoroughly covers the operation of the program- and provides numerous examples of how to maximize your use and understanding of the program's many features. The Reference Manual describes all graphs, menus, commands, and their operation. This manual explains the unique and special non-linear speaker and port models, as well as proper use of the optimizers, importing data, and the many other utilities. The Application Manual provides many exciting examples showing how to use the powerful features of the system in a combined manner to perform both simple and complex design tasks. Both novice and experienced users alike will find this information invaluable for exploiting the full power of the system. Additional information is also provided on loudspeaker measurements, design tips, filter calculations, and complete crossover system development for both passive and active based systems.

✔ 502 Page Reference Manual ✔ 436 Page Application Manual

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Allen & Heath GS3V Console

EQ LAB REPORT

MANUFACTURER: Allen & Heath, 8760 S. Sandy Parkway, Sandy UT 84047. Tel: 801-566-8800.

APPLICATIONS: Mixing and control for recording, MIDI performers and automated audio systems.

SUMMARY: Very good audio performance combined with automation in a multitrack mixer that is ideal for project studios.

STRENGTHS: Flexible layout and compact size combine with a basic automation system to enhance your mixing abilities.

WEAKNESSES: Concentric knobs are too close together; supplied automation software may not be compatible with all Atari computers.

PRICE: $6495

EQ FREE LIT. #: 123

DO YOU OFTEN find that you must remix a project to make small but significant changes? Do your mixes grow in complexity to the point where you begin to sweat at the thought of repeating them? If so, then you are in the market for an automated mixing console. Once these consoles were only available with six-figure price tags and filled large control rooms, but no more. Now a number of mixing consoles are available for under $20,000 and have all of the facilities required in the typical project studio. The GS3V from Allen & Heath not only costs a mere four figures, but also uses up less of your precious control-room space than most similarly featured consoles with VCA (Voltage Controlled Amplifier) fader automation.

Accurate repetition of the mutes and fader moves during the mix and easily making changes as you build a complex mix are important features of an automation system. The GS3V can store all mute settings and fader moves without the need for anything but an external sync-source (MTC or optional SMPTE). Using MIDI proto-

col for transmitting mix data allows the GS3V to store and retrieve mix information with either a MIDI sequencer or the Allen & Heath V_EDIT program. Allen & Heath offers the console in 16- and 24-input versions, with an optional 8-input expander that can be retrofitted to create up to a 32x8x32x2 configuration.

The GS3V is a very compact tabletop format mixer, measuring approximately 28x31x4 inches (719x783x106 mm) for the 16-input version reviewed. This places all of the controls within easy reach, yet there are 64 mono and 4 stereo inputs, all with EQ, available for the final mix. All of the input and output connectors are at the rear of the single-piece steel top panel.

This console makes extensive use of concentric controls on the FX sends, monitor level/pan, and EQ frequency controls. Unfortunately, the lower of the two controls cannot be rotated completely without stopping to take a second grasp of the control because the knobs are so close together. This trade-off has left room for the smooth and accurate 100 mm faders on each input. Of course, on the GS3V these faders actually control DC voltage sent to a VCA (and transmit MIDI information) and are not in the audio signal path. Automation includes control over mutes on the monitor, stereo FX return, FX send and input channels, and VCA control over all input channel levels and the Left/Right Master.

The on-board GS3V automation stores snapshots of the fader and mute positions or fader moves related to an external sync source. By connecting a MIDI sequencer to the MIDI ports on the console, storage and off-line editing of mix data is possible. Automation MIDI data can take two forms: Basic, which uses standard MIDI controllers for fader information and note numbers for mute control; and Enhanced, which increases the resolution of the stored fader information from 128 steps to over 4000 steps by using the double precision of two MIDI continuous controllers (course and fine) per fader. The Enhanced mode also changes the mute information to continuous controllers. The Enhanced mode is currently supported by Allen & Heath's V_EDIT program (currently version 5.04 is supplied with the GS3V) for the Atari 1040ST and compatible computers and a PC version that is in development.

Using the Basic mode of automation a MIDI sequencer can be used to store, edit, and recall automated mixes. Although the lack of fader resolution (in Basic mode) may be a problem in some situations, the excellent design of the VCA control minimizes zipper noise. The advantage of this approach is the tremendous power available to the user in editing the automated mix information with sequencing software. I experimented with these features using Cakewalk Professional for Windows (Twelve Tone Systems, Inc. Watertown, MA. Tel: 617-926-2480) achieving excellent results. Cakewalk, and a number of the other top sequencing programs, allow you to create controller information by drawing on-screen or creating algorithms to define changes in controller values over time. The Fader View mode of Cakewalk allowed me to see the VCA levels changing during the mix and make changes on the GS3V or on-screen.

The Atari ST series of computers remain the most popular computer for music production in Europe and have the distinct advantage in the U.S. of costing only a few hundred dollars when purchased used. Allen & Heath suggests using an Atari (running V_EDIT) exclusively for the automation, freeing your PC or Mac to carry on with the sequencing duties without clogging the MIDI stream with automation control. I experienced difficulty with my old Atari 520STfm, which would hang the GS3V during the recall of stored mixes and occasionally crash when the Null button was selected on-screen.

The software does not allow off-line editing of mix information in an event list, but does offer the user a way to see the current position of all input faders, VCAs, and mutes [see fig. 1]. An additional screen displayed the Master Fader and VCA position and the mute status of FX sends and stereo FX returns. With the addition of a SMPTE-to-MIDI converter, V_EDIT can be used

to synchronize dynamic automation with a multitrack tape machine. Without V_EDIT or a sequencer, the Event and Null indicators at each fader (plus the MIDI RX/TX and SYNC LEDs) are all the information the user has regarding the current automation status.

The Null indicators are two (up and down) triangular LEDs that typically display the VCA setting relative to the mechanical position of the fader. V_EDIT (or a sequencer with MIDI faders) is very useful in keeping track of the current VCA position during the mix. Some of the more expensive automation systems offer moving faders because of the frustration that can result from having the fader in a different place than the automated VCA.

The GS3V does offer the user a range of features for editing the mix without leaving the console. All the automation controls are right on the console, including grouping VCAs or mutes, relative mode for nudging the level of a channel (or group of channels), a punch-in mode so that events can be added without overwriting the entire mix, and the ability to take specific channels off-line (Isolate) for manual operation. There are also seven function keys that are preprogrammed to MIDI Machine Control messages for compatible multitrack recorders to be remotely controlled. The function keys can also be programmed to recall specific VCA/mute settings or send any MIDI information up to 14 bytes (using the V_EDIT program).

The audio performance of the GS3V is very good, with low levels (over 80 dB below nominal) of noise, distortion, and crosstalk. Three-band input channel equalization (±14 dB) includes a fixed shelf at 12 kHz and sweepable mid-frequency (300 to 12 kHz) and low-frequency (20 to 600 Hz) peaking type filters. The equalization lacks high-pass filters, but is effective and musical in operation and can be switched out of the input signal path. The monitor and stereo FX returns have two fixed-frequency controls for ±14 dB at 10 kHz and 100 Hz.

Mic inputs are XLR and all other connections are 1/4-inch phone jacks. The signal levels are nominally –10 dBV, except the channel, group and master inserts, FX sends, control room monitor, and cue sends, which are 0 dBv. All outputs have 21 dB of headroom above their nominal level before clipping occurs.

Four postfader FX sends are available to each input, two of which switch to the monitor signal path. Separate prefade cue sends from the monitor and channel inputs can be connected independently to musicians headsets or combined on the Stereo Cue output that can also include the

continued on page 90

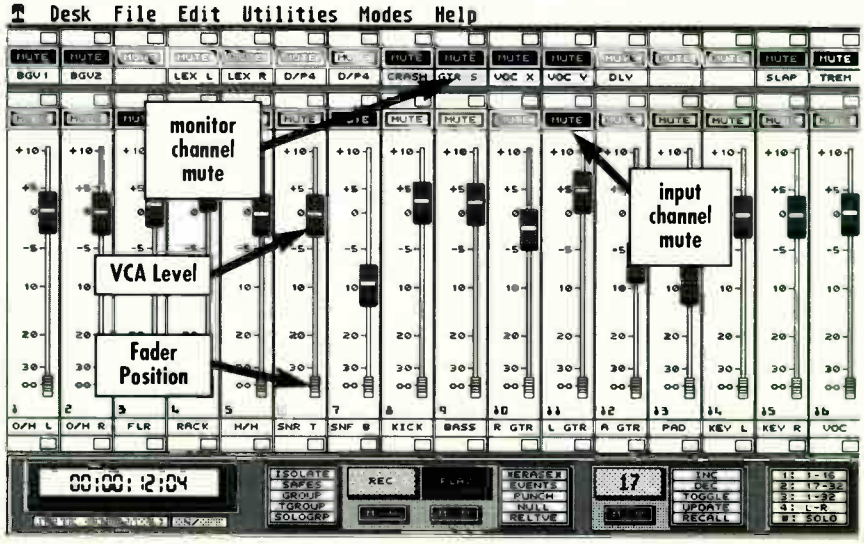
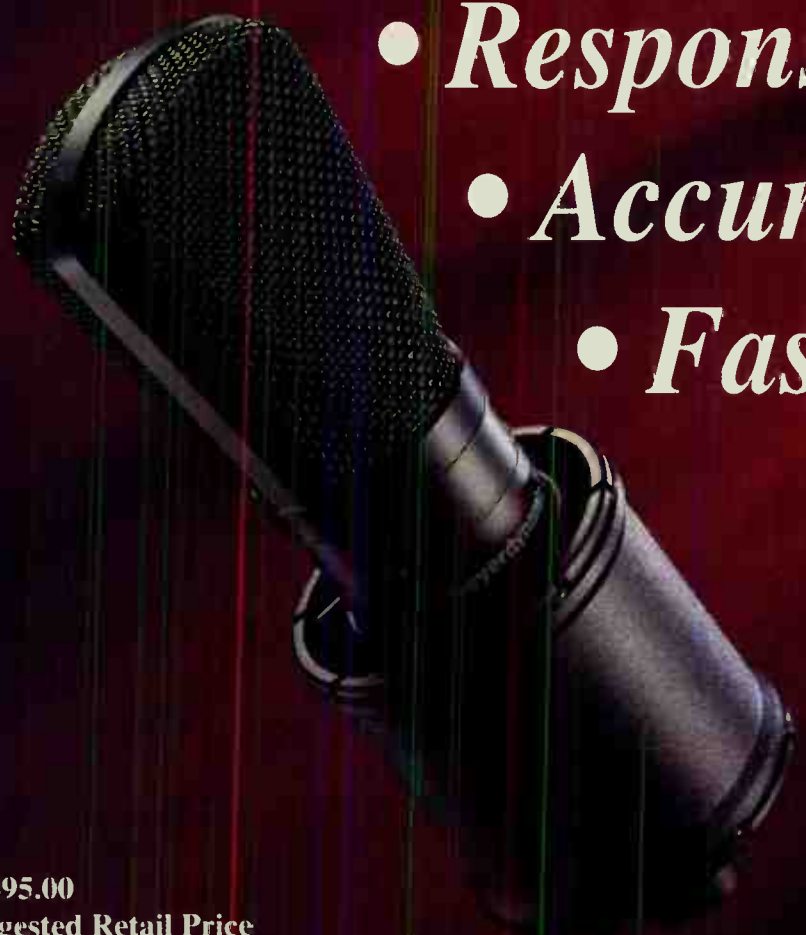


Figure 1: Allen & Heath GS3V automation V_EDIT program - fader view.

The MC 834 —



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Akai CD3000 CD-ROM Player

EQ LAB REPORT

MANUFACTURER: Akai Professional, 1316 E. Lancaster Ave., Fort Worth, TX 76102. Tel: 817-336-5114.

APPLICATIONS: Good for in studio or live sample playing.

SUMMARY: Good-sounding sample player that is easy to operate.

STRENGTHS: Easy loading; can continue to play while loading; high-quality sound; includes five CDs.

WEAKNESSES: You can only sample from the internal CD player; no digital in.

PRICE: $3995 **EQ FREE LIT. #:** 124

The Akai CD3000 CD-ROM sample player is a 3-space, rack-mountable sample player that has the capability of random-access downloading, recording (via CD ONLY), and playback of CD-quality audio data. This includes prerecorded material from CD-ROM libraries or Akai S1000, S1100, and S3200 samplers. If you already own an S1000 or S1100, you can update your disks to the new format for playback and free up your samplers. You can even transmit or receive bulk dumps in the Akai format or Standard MIDI Sample Dump formats between two machines via MIDI or SCSI.

The unit is 32-voice multitimbral and ships with 8 MB RAM, giving you 46.79 seconds mono at 44.1 kHz, or 93.59 seconds stereo at 44.1 kHz. You may also purchase a 10 MB memory upgrade to expand your sampling time to 1.41 minutes stereo at 44.1 kHz. The internal processing is 28 bits with D/A conversion of 20 bits on the stereo L/R outputs and 18 bits on the individual outs; all have 8X over sampling and are 1/4-inch unbalanced. Standard sampling rate of 44.1 kHz is provided with an option for 22.050 kHz recording, resampling, and playback.

SAMPLE LIBRARIES

The CD3000 comes packaged with an assortment of five CD-ROMs from the Hollywood Edge, East West, InVision, and Akai Professional. Upon inserting a CD-ROM you have the option of loading an entire volume (which includes samples, programs, edits, and effects) or just a program and its associated samples. Loading time seemed a bit long (the average was a minute and one half for 1 MB), but keep in mind, that you are loading multisamples from a CD ROM. These drives have always been on the slow side. (Note: although they are getting faster as the months go on; NEC has just released a triple-spin drive.)

One sort of anomaly that kept popping up was extraneous notes being played during a load, or while the disc compartment was being opened or closed. If you tend to load up all of your samples and data before a session or performance using the Setup function (more on this later), this shouldn't bother you. But if you like to keep your music running while loading, some pretty weird things can happen. The plus side is that the unit has the ability to play while loading.

The CD3000 allows for a variety of mixing applications. You may route a program to the L/R outputs, any one of eight individual outputs, and apply the

Imagine Getting Slammed In The Chest With A Sledgehammer.

Now that you know what kind of low end these amps have, let's talk about why. Crown amplifiers are engineered with a damping factor in excess of 1000 (10 Hz to 200 Hz) while most amps are lucky to manage 50 to 100 over the same frequency range. The result of this high damping factor is incredible speaker control for some of the tightest, bone-rattling bass you've ever felt.

Low end isn't the only reason to love these amps. Listen to the other end of the spectrum and you'll hear highs that are crystal clear, revealing every nuance of your music. In other words, sonic purity, with no coloration—as heard only through a Crown.

No other amplifier is as faithful to live and recorded sound as Crown. That's why we're found in the racks of the largest tours and the hottest recording studios worldwide. But don't take our word for it. Compare the sound of a Crown amp head to head with the competition and hear it for yourself.

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send to the effects section. There is no available digital I/O, therefore external sampling or playback must be done digitally off the compact disc.

Assigning a particular program to any output is as simple as turning the data control or entering from the keypad. Basically, with the new cursor buttons and data entry options, entering numbers and naming items can be quite fast; especially when in the edit modes where some functions require several hundred turns of the data control. There is also a "MIDI mixer" that allows the user to assign channels, a key range, transposition, and assign a polyphony value (1-32) to programs. When multiple MIDI channels are sharing the 32 voices, you can set a priority level for programs; those with a lower priority will be silenced when voices run out. Of course, the CD3000 is 16 MIDI channel multitimbral.

THE EFFECTS

The single effects section provides you with 50 preset stereo effects created from four effect types: echo (3-tap delay line), chorus, pitch shift, and a single delay line. These are saved as an effects file that can be moved around and saved with programs and samples when you perform a "volume" save. A simple assignment of program numbers to effect numbers allows you to group programs to share effects or individualize a particular program.

The CD3000 is a RAM-based (memory loss at power-off) sampler, so if you need to save edits of previously recorded material or new sample data, you either have to save to the internal floppy drive or some form of external hard drive. And, although loading programs is quite easy, Akai has developed a disk management system that allows you to create a Setup. At a single keystroke or via MIDI program change, a Setup will load frequently used sounds that you mark from one or many CD-ROM(s)/floppy(s).

DIRECT FROM DISC RECORDING

Because it is, in essence, a CD player, the CD3000 will allow you to play back and also record directly from an audio CD. There are two recording pages; one is the main parameter page. This is where you select bandwidth, duration, original, mono/stereo sampling, and start type. You set the record start by selecting audio threshold, MIDI note, or footswitch start. This screen is a little redundant, because another recording page allows altering of these above settings, as well as a tape-transportlike control for cueing and arming CD audio.

SAMPLE EDITING

You may Cut, Trim, Loop, Join, and Splice together samples with an adjustable, destructive crossfade (for a more seamless splice). Destructive means that the crossfade parameters become part of the sample with no way to recover the original data. The CD3000 will also let you splice two samples on top of each other, effectively creating layers without eating up polyphony. The display is sample-accurate and allows for zooming in and out of the waveform.

From the EDIT 2 page, you have access to the timestretch and resample functions. There are parameters for the amount and area of the time stretch. Two modes are available for this edit: Cyclic, in which a fixed interpolation rate is used for the duration; and Intelligent, which varies the rate according to sample content. The resample edit allows you to change sample rates anywhere from 22.050 kHz to 65 kHz based on a 44.1 kHz original rate. In all of the edit parameters, you are viewing units of samples, but by holding the current page's F-button, the values change to milliseconds. This can be handy with the timestretch functions. When interpolating the sample up or down in pitch, your frequency changes, and along with it, your sample time — the actual number of samples never changes, just the playback speed. This is why it's more useful to use values of samples. As a side note, the EDIT 1 and EDIT 2 pages' parameters are identical to those of the S1000 and S1100 samplers.

Some new editing functions not previously found on the Akai S1000 and S1100 samplers are featured on the EDIT 3 page: Sectional Editing, Normalization, and Digital Fades. Great stuff! Sectional Editing allows you to remove audio data from either side of the start and end points. This becomes very handy if you want to dissect and remove a single snare drum from within a drum groove. You can even leave in the space where it used to live, overwrite the existing sample, or create a new sample leaving the original sample untouched. Level scaling allows you to change the level of the existing sample, and Normalization will find the loudest portion and optimize the level of the entire sample accordingly. This dramatically reduces your noise floor and can be performed on stereo samples while maintaining original phase relationships. The Digital Fade gives you the option of setting a fade that becomes a permanent part of the sample in or out.

ANALOG EDITING

Once you have completed all of your waveform editing, you then assemble your samples in the Edit Program mode. The CD3000 has an extensive implementation of analog style editing called Assignable Program Modulation (APM). With APM, you can assign any modulation source to act on any module (ADSR Envelope, LFO 1 or 2, filter, etc.), just like the old analog synths with modules and patch cords. Complete Envelope, LFO, Filter (with resonance — something you don't see on a lot of today's synths), Key group, and Pitch editing — to name a few — are arranged with a very familiar feel to anyone who has ever done synth programming. The on-screen envelope editing shows values of 00-99 down the side of the screen and a graphic display where you can view one envelope while editing the other.

But don't worry, Akai has provided the option of using some predefined templates of envelope generators that mimic instruments such as organs, pianos, drums, and strings. In fact, you have to create a new program out of an existing one in order to begin any editing, memory permitting. You may even choose the Test Program

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Pure.

AKAI CD3000

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that boots on power-up containing all of the same modulation assignments that are fixed in the S1000 and S1100.

NEW FOR 3000

Akai has developed an ingenious way of entering data that is new to the 3000 series units. Cursor keys have now replaced the cursor knob and a set of "+" and "-" keys can be used in conjunction with the keypad and DATA knob. The Mark and Jump keys allows you to mark a parameter that you most often refer to, and the jump key will take you there quickly from any screen. A new display knob allows you to turn off the high-contrast screen lamp by pressing on the knob itself, thus saving lamp life. Also, as a utility function, Akai is still providing software for programming its ME35T trigger-to-MIDI converter. As with their other samplers, the MIDI pages include various types of MIDI monitoring and generation of note data.

BANG FOR THE BUCK?

With the vast amount of storage (up to 600 MB) on a CD-ROM, and the increasing availability of libraries in this format, I think the CD3000 proves to be of excellent sonic quality and flexibility whether on stage, or in the studio. Editing is very familiar, and with faster data entry, looping and extracting can be seamless in minutes. Even though I would have liked to have seen a digital in, you have to be impressed with not only the sound quality, but the sounds themselves. It's no small offering on Akai's part to have included five great-sounding CD-ROMs with this unit. The usability of the sounds is tremendous and of very high quality. If you already own a sampler (especially an Akai) or are planning to use sounds from a CD or CD-ROM, check this unit out. I believe that sound designers and dance music programmers especially could greatly benefit from this machine due to the fact that it conveniently samples from CD and CD-ROM. In fact, the CD3000 deserves the attention of anyone seriously looking for a sampler/sample playback machine.

Special thanks to David Fournier for his help during this review.

—David Frangioni

LIVE DRUM MIKING

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30,000 people. A lot of sound mixers are used to cranking all the mics up because they have a rock 'n' roll background, so I'll often hear my drums bouncing off the back of the club. For some tours, it's a different rented set of drums every night: different brands, different heads and hardware, different conditions. I just try to get as even a drum sound as I can and then let the soundperson do his or her thing."

ENGINEERING TIPS

Speaking of sound engineers (you remember those folks), what are some of their primary concerns on the topic? Peter Moshay, a live and studio sound engineer (Mariah Carey, Hall and Oates, Journey, The Cars, Barry Manilow), as well as a drummer himself, contends that it all boils down to tuning. "Making sure that the drums sound really good and in tune is my top priority when it comes to miking," he comments. "I always approach everything from the instrument because you can put the best mic in the world in junky, badly tuned drums and it's just not gonna make them sound great. For tours, I spend a lot of time listening to the drums, and I'm usually the one to tune them. Most drummers trust me to do that. If I'm working with a drummer for the first time for a particular tour, I'll work well in advance with them on the sound of their drums, the tuning, and then the miking. The most important thing is to learn how to tune the drums properly."

Moshay does advise experimentation. "Each drummer is an individual," says Moshay, "and you've got to do your homework learning what that drummer wants and learning the music itself backwards and forwards. Once you thoroughly know the music, you can better reproduce the sound via miking. Bands want the live sound to be a close facsimile of that on the record. That's also what the audience wants, and it's your job to give it to them. If you go to a Rush show, for instance, the fans know every fill the drummer's doing, so it had all better be there and be heard. It annoys me when an engineer doesn't pay attention to that either through miking or the board. A drummer can be up on stage, playing fills that are important to the song, and the audience can't hear them." EQ

THE RAMONES LIVE

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boardwalk. The toilet of the world. I would say my favorite venue, however, is the old Ritz in New York City.

Joey: That place was great.

John: I always enjoyed doing a show there. It always had a nice sound.

Joey: Remember that time we were playing in Japan and there was that big earthquake?

John: That was in Tokyo. At first I thought the people were shaking, then the speakers were moving and everybody was just grabbing onto everything. That was exciting.

Joey: It was weird. We went into the dressing room and there were giant cracks in the wall.

John: We were on the 9th floor.

Joey: That was weird, too. It was a department store, wasn't it?

John: It was a department store all through the whole building except for that room on the 9th floor, which was like a small amphitheater. That was a good one. I guess that was back in 1980.

Joey: How many years has it been now?

John: Since 1978. We've spent a lot of time doing a lot of things together. It's become like a family thing.

Joey: I'll say. You ready to do it again?

John: Ready whenever you are. EQ

DIY LOUDSPEAKERS

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could undergo excursion to nearly 7.5 mm and still be producing only about 3-percent harmonic distortion. In this simulation, the 1.75-cubic-foot enclosure would be on the verge of significant increases in distortion at this voltage drive level. The 0.6 -cubic-foot box, however, would require nearly 24 V to produce the same excursion and distortion levels and would be producing 108 dB SPL — 5 dB more than the larger enclosure. Figure 4 illustrates the nonlinear nature of the port output at higher input levels as reflected in the decreasing magnitude of the lower impedance peak for the 15 V impedance curve.

Using a sophisticated program like LEAP, or using the design charts in a book and a hand-held calculator, the ability to design vented enclosures or any of the other direct radiator enclosures (sealed, passive radiator, bandpass) can be easily and accurately accomplished without the need to call in a high-paid industry consultant. EQ

Doing Windows, Part 2

More MIDI mapping in Windows 3.1 for all you Intel-PC audio processors

BY JOHN WORAM

This two-part feature is excerpted from a 22,000-word chapter in the author's *Windows Configuration Handbook* (Random House, October 1993). In that chapter, the Media Vision PAS-16 sound card was used as a typical component in a multimedia system.

Welcome back. Last time we covered the MIDI file format, the MIDI mapper, MIDI mapper configurations, and MIDI mapper dialog boxes. If you haven't read Part 1 yet, do so now. We'll wait. Ready? Here we go...

Patch Maps. Figure 1 shows a custom patch map, whose components are described here.

0/1-based patches Button. The 128 names in the General MIDI Patch Name list may be numbered as follows:

Patches	Numbered as:	Patch Map button reads:
0-based	0–127	1 based patches
1-based	1–128	0 based patches

Use the button to toggle the patch numbers from one base to the other, solely for the purpose of matching the screen display with a synthesizer that uses either base system.

Note that the button legend identifies the base that is currently *not* displayed. If you click on it, that base is displayed and the legend switches to the other base. It's initially confusing — just remember the legend is always the opposite of the current status.

Source Patch. This column lists the 128 General MIDI patch numbers (0–127 or 1–128) associated with the adjacent Source Patch Name column. For each melodic channel recorded in a MIDI file, an embedded Source Patch number associates that channel with a Source Patch Name. If you click once on this or any other MIDI Patch Map row, arrow buttons appear in columns 3–5 for that row, as shown by the example in Row 3.

Source Patch Name. This column lists the names of the 128 melodic channels, as defined by the General MIDI specification. The list is read from the MIDI Mapper driver, as shown in the MIDI Mapper Record in SYSTEM.INI section below.

Dest(ination) Patch. Change the entry in this column to reassign any recorded channel to some other Source Patch Name. For example, in fig. 1, Row 3 shows that any channel associated with the Electric Grand Piano (Source Patch 3) will be associated instead with the Marimba (Source Patch 13).

Volume %. By comparison, this one is easy. Adjust the value up or down to change the output level of the selected instrument.

Key Map Name. Click on the downarrow to open the drop-down list of available Key Maps, such as that shown by the inset in fig. 1. Note that the Bright Acoustic Piano (Row 2) was previously assigned to the +2 octaves Key Map. Refer to the Key Maps section that follows for further details.

Key Maps. By default, any key on a synthesizer keyboard plays the musical note associated with that key, and by convention, middle C (below A-440 = 261.63 Hz) is identified as key 60. If the synthesizer keyboard is set up to play percussion, then each key is instead associated with a specific percussive sound. For example, key 48 is either the melodic note C below middle C, or the High-Mid Tom percussion sound.

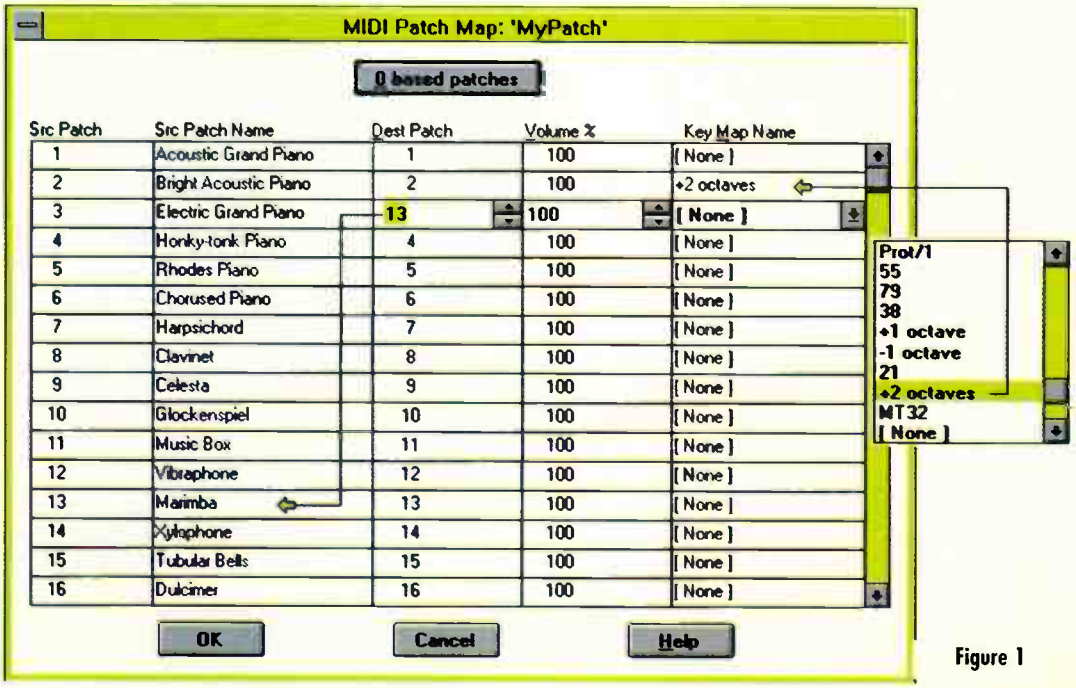


Figure 1

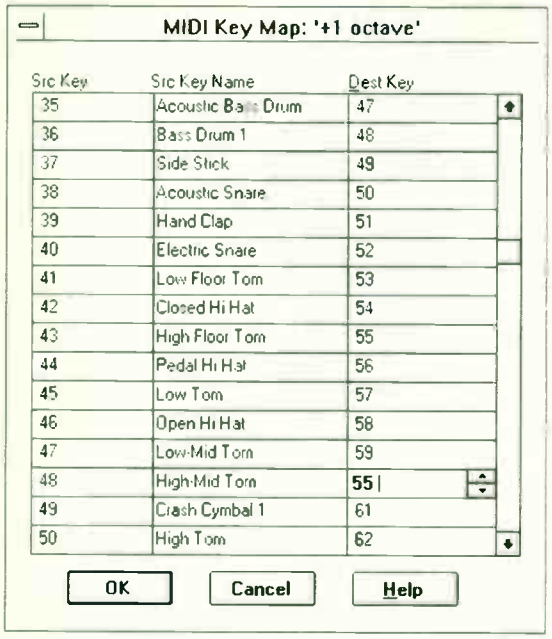


Figure 2

The Key Map can be used to remap one or more melodic or percussion keys. For example, fig. 2 shows a section of the +1 octave key map.

Source Key (Src). This column lists the 128 MIDI keys (0–127) on a synthesizer keyboard or specified within a MIDI file.

Source Key Name (for percussion). The names in this column identify the percussion instruments associated with synthesizer keys 35–81. It, too, is read from the MIDI Mapper driver (see MIDI Mapper Record in SYSTEM.INI, below). The column can be ignored if using the Key Map for melodic instrument mapping.

Dest(ination) Key. If any Source Key is pressed (or specified in a MIDI file), the corresponding Destination Key is heard. If no key mapping is specified, then, of course, source key 35 plays destination key 35, and so on. If the +1 octave key map shown in fig. 2 is used, however, then every source key is remapped 12 semitones, or one octave, higher. Thus, source key 35 now plays destination key 47, and so on.

In fig. 2, source key 48 is remapped to destination key 55, thus making it a perfect fifth (7 semitones) higher, instead of one octave. To create a complete "Fifths" Key Map, edit a new Key Map and add 7 to each Destination Key.

REVISING THE MIDI MAPPER

It's possible to add entries to the available ports list by installing an extra MIDI driver. For example, the Media Vision PAS-16 (Pro Audio Spectrum 16) setup procedure installs two MIDI ports: Voyetra OPL-3 FM Synth and Pro Audio/CDPC MIDI Output. The former is an onboard synthesizer and the latter provides output to an accessory MIDI device, but only if you have one installed. Otherwise it doesn't do a thing.

Install New MIDI Port. The Media Vision PAS-16 is Ad Lib compatible, and there is an Ad Lib MIDI driver included with Windows. To install it, open the Drivers applet, click on the Add... button, and then select "Ad Lib" in the List of Drivers. Click on the OK button to continue the installation. When the installation concludes, the following lines will appear in the indicated INI files:

SYSTEM.INI	CONTROL.INI
[drivers]	[driver.desc]
MIDI2=msadlib.drv	msadlib.drv=Ad Lib

Add New MIDI Setup. With the new driver in place, you can add additional MIDI setups as described here for the just-cited Ad Lib driver. To begin, open the MIDI Mapper applet and click on the New... button. Type the information in the Base configuration column below.

Data entry box:	Base config	Gen config
Name:	Ad Lib	Ad Lib general
Description:	Base-level setup	Gen MIDI setup

Click on the OK button and when the MIDI Setup: 'Ad Lib' dialog box appears, open box 13 in the Port Name column and select the Ad Lib option on the drop-down list. Repeat the procedure for boxes 14–16. This concludes the conventional base-level setup for the Ad Lib driver. If you want to set up a general MIDI configuration as well, repeat the just-described procedure using the information in the General configuration column above. But this time open boxes 1 through 16 and select Ad Lib for each one.

You can make whatever configuration modifications you like, but the procedure described above sets up the two conventional Ad Lib setups usually encountered when this driver is installed.

Delete MIDI Mapper Setup. In order to delete a MIDI Mapper setup, you must first make sure the setup is not in use. That is, when you open the MIDI Mapper applet, the setup to be deleted must not be seen in the Name box. If it is, select some other setup and then close and reopen the applet. Now, select the setup you want to delete and click on the Delete key.

MIDI MAPPER CONFIGURATION TEST

This section may be of use to the MIDI beginner who is still trying to figure out what all this really means.

The CANYON.MID File. Windows is supplied with a grand total of one MIDI file, named CANYON.MID. The file makes a convenient configuration tool, since its header contains a text section in which the various channel/patch assignments are listed. Although some Microsoft documentation states that the file contains information on all channels, it doesn't. Refer to table 3 for CANYON.MID file channel assignments. The file will be used here to illustrate various aspects of MIDI Mapper configuration.

CANYON.MID's base-level melody is on channel 15. Open the MIDI Setup dialog box and clear the Active box for all channels except 15, so that only this melody line is be heard when the file is played. To get an idea of what it sounds like, close the MIDI Mapper and play a bit of it. Then stop the playback and reopen the MIDI Mapper.

Patch Map Test. Edit a new "MyPatch" patch map, and patch the Electric Grand Piano source over to the Marimba destination, as was shown in fig. 1. This particular source was selected because the CANYON.MID file header data shows "melody C15-P3" (that is, channel 15 to patch 3: electric grand piano).

Next, open the Setup dialog box and in the channel-15 Patch Map Name box, specify your new "MyPatch" patch map. Close the MIDI Mapper and replay CANYON.MID. The melody should now bear about as much resemblance to a marimba as it previously did to a piano.

Melodic Key Map Test. Since the CANYON.MID header also reveals that channel 14 is for bright acoustic piano, open your "MyPatch" Patch Map and specify the +2 octaves Key Map for that instrument. Next, open the Setup dialog box and reenable the channel-14 Active check box. The next time you play CANYON.MID, you should hear that marimba accompanied by a high-pitched piano.

Percussion Key Map Test. The percussion channels (base-level 16, extended-level 10) are treated somewhat differently. Listen to channel 16 only, then assign your "MyPatch" patch map to that channel. Edit this patch map and in the Key Map Name column, open the box in the first row and assign any Key Map to it. Note that the Src Patch Name column (Acoustic Grand Piano in this case) has no significance when the patch map is assigned to a percussion channel (16 or 10), but the effect of the assigned key map will be heard on the percussion the next time CANYON.MID is played. If one of the numbered key maps (21, 38, 55, 79) is selected, then all percussion sounds are routed to that key. (Select any such map and note the explanation on the Description line.)

MIDI MAPPER RECORD IN SYSTEM.INI

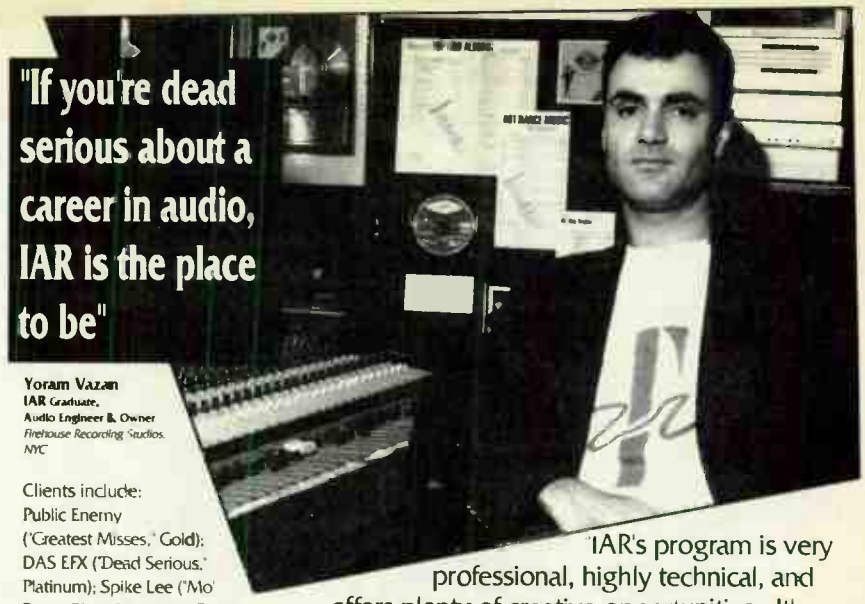
As previously noted, the installed MIDI drivers are listed in the [drivers] section of SYSTEM.INI, with accompanying virtual drivers in the [386Enh] section, as shown by the first four entries below.

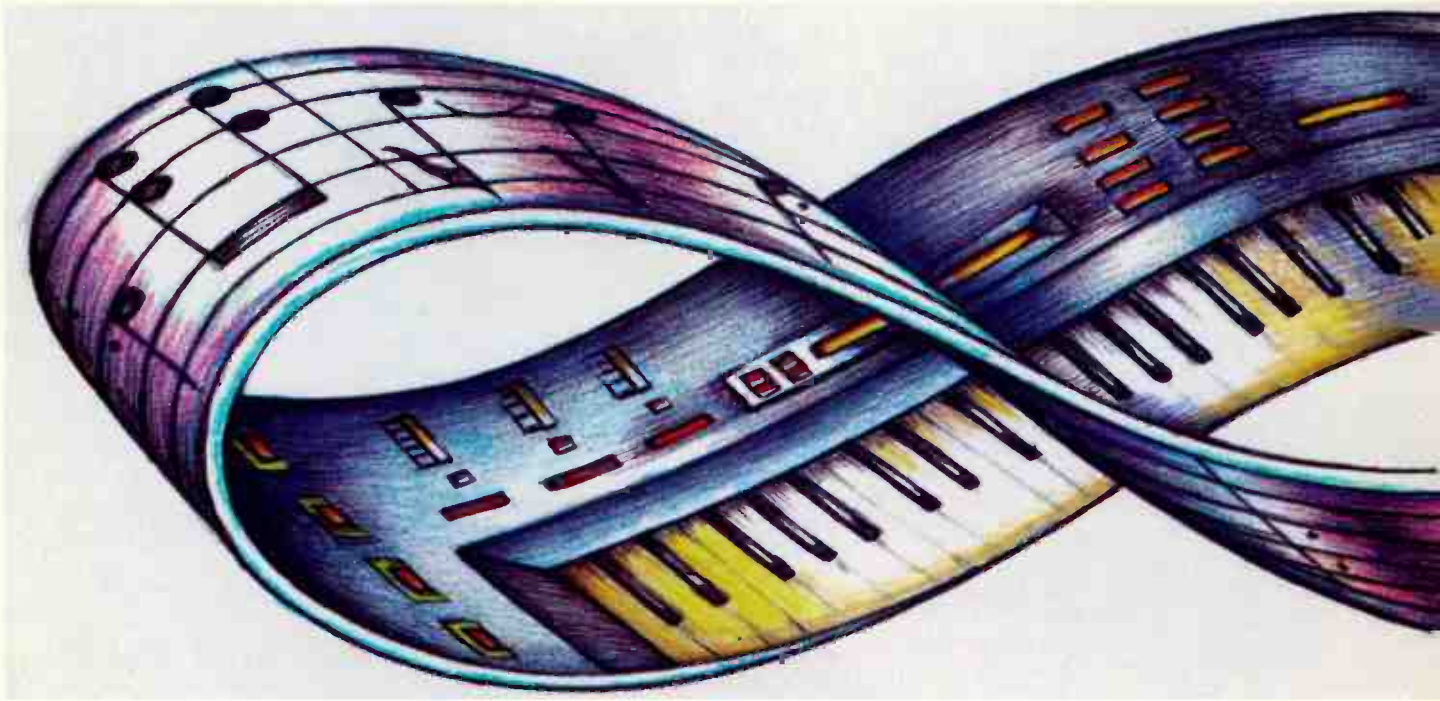
[drivers]	[386Enh]	
timer=timer.drv	device=vtdapi.386	timer
MIDI=msadlib.drv	device=vadlibd.386	Ad Lib
MIDI=mvproaud.drv	device=vpasd.386	Media Vision
MIDI=sndblst2.drv	device=vsbd.386	SoundBlaster
MidiMapper=midimap.drv		MIDI Mapper

The MIDI Mapper driver itself is also listed in the [drivers] section, as shown above. In addition to handling its other chores, the driver contains the following information:

MIDI Patch Map:	Source Patch Names
MIDI Key Map:	Source Key Names

John Woram is the author of the bestselling PC Configuration Handbook, *and the recent Windows Configuration Handbook, both published by Random House.*





Mobius Strip Illustration by Normal "The Math Man" Rockhard • Prints Available

The Revolution Will Be Digitized

Eternally evolving technology and the arrival of artificial intelligence is blurring the distinction between man and machine in the creative process. Should we be celebrating?

BY J.D. SHARP

Musical intelligence" has become a catch-all phrase that loosely encompasses several forms of music-generation software. On the one hand are algorithmic generators, which use existing patch or note information to generate new synth patches or note patterns. The other expanding area of so-called musical intelligence has to do with composition: a variety of software and hardware implementations now seek to assist the intrepid composer in the creation of backing, rhythm, and accompaniment tracks. The question in all this is: Are any of these tools anything other than crutches for the musically impaired? Is there any reason for a serious musician to track the growth of this entire area of software and hardware development?

To properly respond to this question we need to take a few steps back and examine what the current marketplace for musical compositions looks like (don't worry — we'll relate this back to musical intelligence posthaste.) The radio waves are, for the most part, controlled by corporations and investment groups instead of by woolly radio pioneers. And what turns a program director on more than anything is predictability, which in musical terms takes the form of identifiable musical styles that can be attached to certain demographic groups. Once the musical tastes of a certain demographic group can be clearly outlined it's easy to go to potential advertisers and promise to deliver a targeted market. If you're selling muscle cars, for instance, you would most likely want to target males in their late teens and twenties; their identifiable music style, at least in the cognitive map of corporate America, is hard rock or heavy metal, so it's no accident that pictures of hot cars and light beers are accompanied by wailing guitars. The same sort of identification applies to each age and demographic group.

THE FACTS OF LIFE

There's another trend to consider when discussing artificial intelligence for musicians: it's darned hard to become accomplished on a musical instrument. In this era of video games, virtual reality, and instant gratification, the concept of sitting and torturing one's fingers to acquire a unique form of physical dexterity seems almost antiquated. As the world fills up with "vidiots," the accomplished virtuoso musician could well become a rare bird; musicians who can actually think for them-

selves and play their ideas in final form will almost constitute a priesthood. Meanwhile, there will likely be legions of musicians who, given some boost by machinery and software, can produce workable — if slightly banal — compositions that will do just fine for elevators, multimedia presentations, home entertainment, and (shudder) perhaps even radio play!

The connection to musical intelligence and algorithmic composers is this: If musical styles are identifiable and if markets demand conformity to these styles, it's not farfetched to conceive of software that produces the rhythm, drum, and bass patterns that are characteristic of particular musical styles. If you throw in the ability to edit and rewrite all or part of these patterns, a new dimension is added.

For instance, if a music bed for an advertisement demands a swinging country style, it's not totally out of hand to consider borrowing a "stock" country arrangement programmed into a "smart synth," and then revamp the chord progression and bass line. In fact, devices like Korg's Interactive Music Workstation (I2 and I3) extrapolate chords and patterns from the notes you play; you just point them in the general style direction you want to go, and direct the creation of a progression and accompaniment with the notes you play.

Add to this the difficulty in learning everything from scratch and the endemic impatience that's so much a part of the American scene, and you can see why software and hardware manufacturers are falling over each other to produce easy-to-play-yet-powerful instruments and song generators. Here's a quote from a recent ad for the Jammer (song generation software): "The Jammer combines music theory, artificial intelligence, and randomness to create new quality musical parts for you each time you recompose...We took the time-consuming work out of creating music, but when it comes to control...we saved it all for you." Or

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DIGITAL REVOLT

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check out Korg's spiel for the I3: "The I3 is capable of producing musical 'ideas' of its own — called Styles — that can be modified, looped, and combined to block out songs in minutes."

THE CREATIVE TOUCH

What's missing from this picture? In a word, creativity. No matter how smart these instruments and programs get, there is one thing you can be fairly certain of: they are not going to create the next hot musical style. In fact, their very existence is based on the fact that popular musical tastes fall into identifiable categories, and each of these styles of music has rhythmic and chordal patterns that are the essence of its characteristics. As long as audiences and program directors are content to endlessly recycle and reconsume the same musical styles, artificial music intelligence will have its place and will be in demand. If the pendulum swings back toward '60s- and '70s-style creativity, these tools will be less useful because the amount of energy it will take to reprogram them to current styles will be better spent in simply writing great new songs.

There's also the matter of lyrics; if your ambition is to write the great American pop song, you'll have to come up with some decent words, and nobody to date has come up with a lyric generator. (There is quite a bit of repetition and cliche in this area, though. Let's see, just feed the computer "baby," "got to have it," "hurts so bad," "love," and about ten more key words, and away you go.)

The whole issue of music intelligence comes down to this: What kind of music do you seek to fashion? One of the most eye-opening experiences of my career came when I visited the project studio of a popular rock musician. He was hard at work on the follow-up album to his monster hit, and was toying with the idea of having me play some keyboards or contribute a song. He gave me a work tape to listen to. I didn't end up with the keyboard gig or place a composition, but I did receive a lesson in pop-song creation.

There was a slow song on the album with a complete set of lyrics, and a fast track with no vocal track. When the album came out he had taken the lyrics of the slow song and grafted them on to the fast track! I had slaved away for years thinking that my musical compositions demanded certain lyrics, and that once they were united there was no way to move them around like a stack of building blocks. I was wrong. He ended up with *another* monster hit, and I went back to selling musical instruments. In a world where pop songs are made out of so many interchangeable modules, artificial musical intelligence has a bright future, whether you like it or not. EQ

ALLEN & HEATH

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main L/R mix and even follow the control room monitor selection. Conveniences such as a built in oscillator (1 kHz and 10 kHz) and talkback mic (to subgroups or cue sends), make the GS3V a very complete console for project studio applications. Outputs to the multitrack and 2-track mastering machines are switchable to reduce repatching for common applications and tape and line inputs can be swapped to the channel fader.

Although the VCA and mute grouping is very powerful in the studio, the lack of any way to quickly assess the current grouping status of the console would make these features less desirable in most live-sound applications. However, the GS3V could be a powerful addition to a performer's MIDI rig where the mute and level automation could be integrated into the MIDI system with ease and, with MIDI mapping, the function keys, faders, and mutes could even control other MIDI devices.

With the high sound quality and accurate automation of the GS3V console, project studio owners can exceed the expectations of even their most demanding clients. Reliable automation software (which I'm not convinced V_EDIT is yet) is paramount to quickly making changes to a project when the studio clock is ticking. Fortunately, the GS3V automation can function very well without V_EDIT. Allen & Heath has built a good-sounding console that should fit into any project studio where there is time to build complex mixes but not to relearn the mix when last minute changes are required. You could even have the luxury of letting the band sit at the back of the room listening to the mix instead of helping you to move faders and mute channels.

—Wade McGregor

PowerPC to the People

Hold on to your mouse! The PowerPC chip is poised to revolutionize the computer industry.

BY MARTIN POLON

Nineteen ninety-four may well be the most important year in the history of digital audio since the inception of digitization in the late 1970s. The arrival of the PowerPC chip from the consortium of Apple, IBM, and Motorola heralds a change in the power of personal computers on a magnitude that may well approach ten- to twentyfold for some applications.

What the PowerPC does is utilize a standard designed by the triumvirate of Apple Computer, IBM, and Motorola. The new chip's architecture is based on reduced-instruction-set computing (RISC), as opposed to the complex-instruction-set computing (CISC) of the X86 chips from Intel that are the basis of the PC family or the 680XX processors that are the current processing engine for the Macintosh family. RISC chips move instructions at a much faster rate than existing processors, due in large part to the reduction of instruction complexity.

The PowerPC will first appear as the 601 chip, made by Motorola, in IBM and Apple computers which are scheduled to be sold as this issue goes to press. This chip will ostensibly clock at 66 MHz, but advanced samples have indicated 80 MHz options are available. The PowerPC is based in some part on the IBM Power architecture used in the RS/6000 workstations, and IBM will coproduce the 603 PowerPC chip with Motorola.

This 603 chip, which could be released as early as the fall of 1994, will post similar performance ratings to the 601 but with a much reduced power draw, allowing use of the PowerPC in portable computers and other transportable equipment. These chips will run twice as fast as the Intel 66 MHz 486 and can benchmark twice as fast as the Intel Pentium. Manufacturers have expressed interest in the PowerPC family due to the considerably lessened power draw and heat dissipation — plus the fact that the PowerPC costs equipment makers considerably less than the Pentium.

More advanced members of the PowerPC family will utilize 64-bit data paths. This will take clock speed well out to and beyond the 100 MHz barrier. The 604 processor will be focused at high-end processing systems while its stable-mate, the 620, will power workstations and small computerlike PowerPC mainframes. Both the 604 and the 620 could appear before the end of 1994.

One of the astonishing features of the PowerPC family is the diversity and range of operating systems that the PowerPC chip family accommodates currently or will in the near future. The list includes (but is not limited to) Apple UNIX (A/UX), Apple Macintosh System 7.5 Power, IBM UNIX AIX 3.x, IBM Workplace OS Mach 3, Microsoft Windows NT, NextStep, Novell Netware, Sunsoft Solaris, and Taligent Custom.

The initial offerings of PowerPC 601-equipped machines will include a substantial range from Groupe Bull, IBM, and Apple. Initial offerings will be in the $2000 range. Later implementations of PowerPC systems are expected to include the "Fire Wire" Apple/Texas Instruments replacement for the SCSI interface as well as for the serial and ADB (Apple desktop bus) ports on Macintosh systems. PCI

(peripheral connect interface) bus slots are also expected to replace Nubus options. Another expected option is the use of the Sony MiniDisc format for floppy-disklike optical storage of approximately 130 MB per disk.

How does all this translate for the personal and project studio marketplace and all other audio users as well? One way to measure the gain in speed and power is to use benchmark studies showing Beta-version PowerPCs running 3D modeling software that utilizes the native PowerPC version of the Mac toolbox. These machines operate at three to four times the speed of the current top-of-the-line Macintosh Quadra systems. Modeling software is strikingly similar to audio software in being very space- and speed-needy. The promise of PowerPC, even in its initial implementation, is to bring the "studio in a box" to fruition in a way that will further empower the user. Mixing, postproduction, editing, sweetening, CD prep, commercial production, film work, scoring, etc., will all benefit from the increase in processing speed and data throughput. The ability of the PowerPC to move at speeds approaching and eventually eclipsing 100 MHz obviates the need for onboard DSP chips, with audio and video functions possible via software implementation. All in all, a real case of providing power to the people!

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

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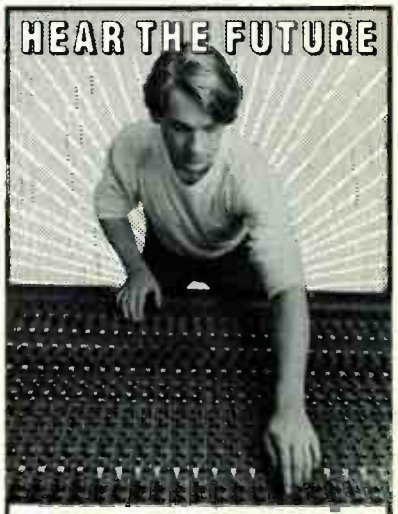

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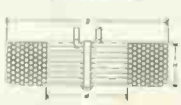

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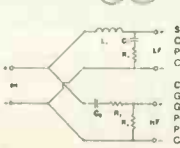

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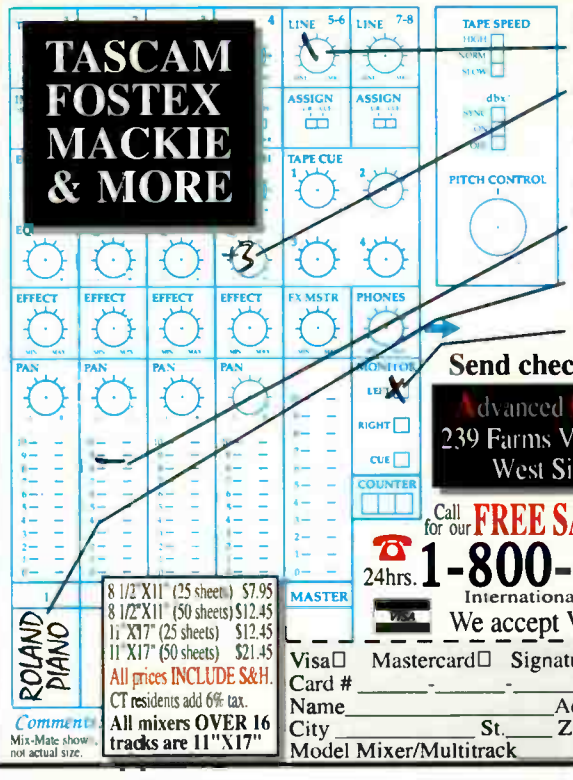

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More On the Floor

Shoot for the stars in your project studio by first looking at your feet

BY JOHN STORYK

I'm certain that the most commonly asked question in studio design and construction is, "What will it cost?" Another question near the top of that list is, "What should the floor of my project studio be constructed of?"

Project studios come in virtually every shape and size — enclosed surfaces can change in an almost infinite combination of designs. The studio floor is, of course, somewhat special in that (until we make a studio in zero-gravity space) we have to walk, sit, and function with our weight on this surface. And with very little exception, this surface will be perfectly horizontal, and will have to be durable enough to withstand hours upon hours of use.

Studio floors are divided into floor structure and floor surfacing. The design criteria for each are sometimes similar, but more often have very little to do with each other.

FLOOR STRUCTURE

Ideal Conditions. There is little question that an ideal project studio control-room floor structure would be a concrete slab on grade with disconnected perimeter walls (similar to that shown in fig. 1). Unless you are constructing a studio from the "ground up," this is not likely to be the case. A variation of this type of structure might be a typical residential basement, where you would have "slab on grade" construction, but no disconnectivity at the perimeter (i.e., as in fig. 2). This will cause acoustic flanking via the wall boundaries, thus diminishing the amount of acoustic isolation between the studio space and the rest of the building spaces. Both of these systems, however, will satisfy the most important characteristics of a studio floor structure: that it be rigid and stiff. Typical nonconcrete structure slabs (i.e., steel or wood joists with plywood underlayment) are designed to have a flex

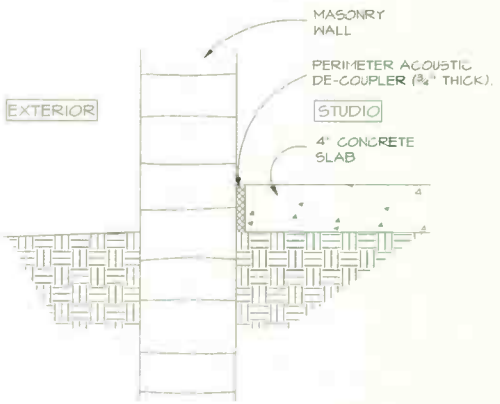
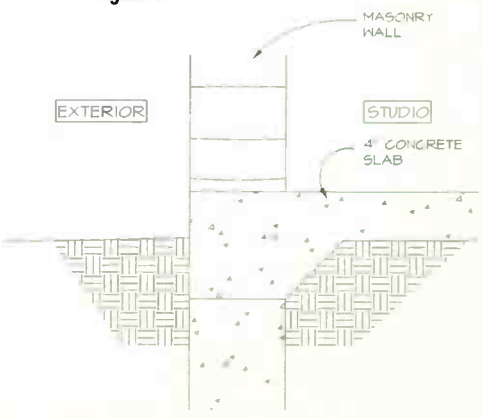


Figure 1



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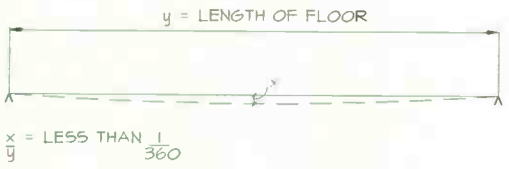

Figure 3

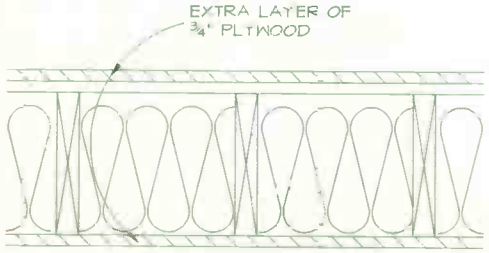

Figure 4

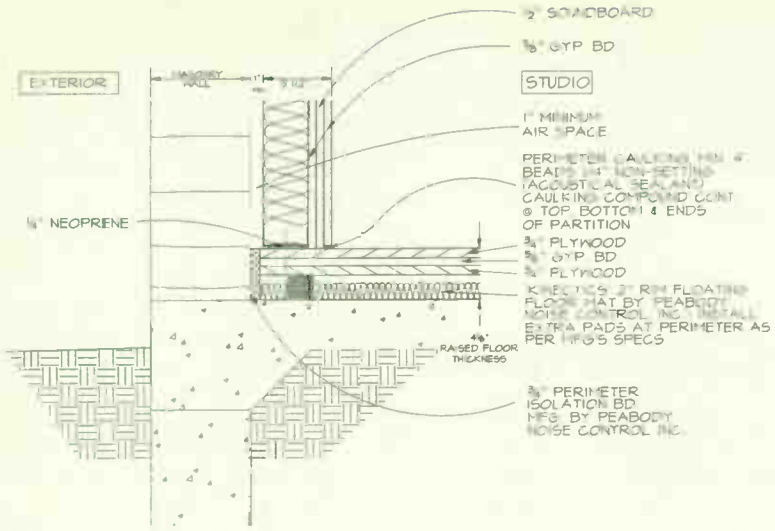

Figure 5

characteristic of less than 1/360 (see fig. 3). This seems very small, but over a span of 30 feet (360 inches) flexing 1 inch is okay for load-bearing purposes, but quite unsatisfactory for acoustic purposes.

Stiffness. A studio floor must be stiff! Prior to any "floating floor" design — prior to any "room-within-room" design for that matter — the studio floor structure should be made extra stiff or determined to be stiff enough. One can quickly see why concrete slabs are virtually always preferable. If a typical wood joist and plywood underlayment structure is the existing condition, stiffening this slab with one or two extra layers of plywood (cross seamed) is recommended — top and bottom (fig. 4). If the structural members cannot take an extra layer of plywood on the underside of the joists below, install additional joist bracing.

Floating Floors. A floating floor will only be necessary if isolation to and from the studio environment is required. Floating floors are expensive; they chew up valuable space (it is usually the studio height that you don't have enough of); they take too much time to build; and are not always necessary! It starts with required NC value (level of quietness) required for the studio environment (see the January '94 column). Project studios that are essentially postproduction environments do not need to be as quiet as rooms that will have open microphones. Project studios with separate iso/vocal booths can get away with an NC 30, while the recording rooms can approach NC 22. In that case the vocal booth could typically be "floated" while the studio would not. Alternatively, if there are external noise levels that will intrude on the studio, then floating the entire studio is a must.

The typical floating floor works by being simultaneously flexible and stiff. The decoupling device is flexible and the "floated slab" is stiff. Figure 5 depicts one of several commonly used floating floors. Note the extra mass given to the "floor" part of this system by simply adding 5/8-inch gypsum board. If more weight is required (a more complex calculation and evaluation than space permits), then simply adding layers of gypsum board is an easy and economical solution. Always cross seam each layer. Never attach the floated structure rigidly to the underlayment.

The typical basement project studio, located in a residential environment, will require a floating floor (and, of course, wall-ceiling) construction if you expect to completely isolate the studio from the house. My advice would be to first try and to control noise to and from the studio with scheduling or location. In most 8-foot-high basements you will quickly run out of ceiling height. Note: complete room-within-room construction has been installed in Whitney Houston's home project studio, but the basement slab had been deepened about 3 feet

prior to this work to obtain good studio ceiling height!

Never confuse the required floor structure (floating or nonfloating) with its final surface treatment. Treatments are a function of:

- internal room acoustic requirements
- wear and tear
- personal tastes (we won't be discussing that here!)

THE FLOOR AND ROOM ACOUSTICS

Most small project-studio rooms (control rooms or studios) can be described by three sets of parallel or near-parallel walls — three dimensions. In most cases, project studio rooms are, in fact, orthogonal (rectangular) in nature. There is nothing wrong with this as long as mid- and high-frequency ray trace analysis is performed and appropriate treatments are then added to certain surfaces. Low-frequency analysis (modal in nature) is very much a function of simply these three numbers — i.e., of x, y, and z values. Even a more complex room can be converted to x, y, z values. The z dimension is typically the vertical one (height) and is usually the lowest. Since floors are almost always perfectly horizontal, the smaller vertical dimension gives a studio designer less opportunity to introduce angles, slopes, and so on. This is the reason why studio designers always want more height in control rooms.

A simple ray trace in the vertical plane will show why a more acute ceiling angle is typically preferable in control room design. Since this would be difficult or even impossible in smaller rooms, one is left with little choice but to introduce mid/high-frequency absorption on these surfaces — i.e., carpet. In smaller rooms, carpet is usually the safe way to go. In larger rooms, where there would more of a chance of ceiling angling, hard control room surfaces could be installed. There is nothing wrong with carpet, but remember:

1. A carpeted floor surface is simply one more absorptive surface, reducing natural reverb time (RT60) and room ambience.

2. Carpet will wear out much sooner than a more durable resilient flooring surface — especially at the console listening position.

FLOOR SURFACES

If carpet is required, try and use the most sound absorbent type you can. Carpet specifications will give you the weight in oz/sq. yard (a typical number is 35-45 oz). Use commercial-grade carpet with a weight as close to these as possible. Add to that the weight of the padding — preferably 40 oz. — and you will get a total carpet and pad weight close to 80 oz. The backing and latex will bring the total unit weight up to 120 oz.

Antistatic carpeting is a must! Static information is typically given in a KVA static number. Use carpets with no greater than 2–3 KVA. Certain fibers are better than others for this. Antron III fiber, for example, is one of the best. Look for a product with extra static control built into it for extra protection.

I rarely recommend direct glue down. Use pad with tack-down installation. The carpet will have to be replaced approximately every two or three years — this will be difficult with glue down.

Resilient floors come in many varieties. The most common in the studio world is, of course, wood flooring. It's tough to beat and when a resilient floor is wanted, it's usually the best. Wood, however, wears out and often needs re-finishing. People often describe wood as having an acoustic naturalness (I suspect trying to describe some sort of absorption at certain frequencies). A finished floor has very little of this, but is still a great flooring surface. Again, if it's wood, install the flooring cross seamed to the structural underlayment. Another excellent flooring surface in these instances is vinyl. It is a very affordable treatment and, in the past few years, this industry has introduced a great variety of colors and finishes.

A FINAL WORD

Unless there's a reason for a step or platform, I would avoid them. They reduce room flexibility; they become something for people to trip on; and, of course, they increase costs. If you need a platform, make it at least 6 inches high so that you have a "real" step and so that you can build it with relatively stiff construction. If you determine that you need a "floating floor," do not penetrate it with empty audio conduit or troughing. This virtually defeats the purpose of the installation.

Good luck building! EQ

Basement Taping

Barry Eastmond goes underground (mostly) to create the project studio of his dreams

BY RICHARD ALDERSON

When producer/composer Barry Eastmond (Freddy Jackson, Anita Baker, Tom Jones, Regina Belle, and Jonathan Butler) moved from Brooklyn to his Hudson River town 30 minutes north of New York City, he knew two things for certain: he wanted to build an in-house recording studio and he wanted it to look exactly like Ground Control, the recording studio of HEA, what was at that time the city's hottest jingle production house.

Eastmond came to me to build his basement project studio because I had designed Ground Control. I faced three key problems in designing this room: a very limited ceiling height, a desire to keep windows in the recording studio, and the need to isolate the studio acoustically from the rest of the house. But the basic problem was fitting a professional studio into a small, awkward space.

There were other considerations as well. Eastmond plans to someday build another studio outside his house, so he did not want to spend a lot of money tearing down walls and changing the basic structure of his space. For that reason, a set of plumbing pipes that jutted out into the proposed studio couldn't be moved. I had to design around them. Today, Eastmond's Amek Angela 36-input deck surrounds the pipes.

The space did offer some pluses. Since it is literally the ground floor, there was no need to float the floor. I

The low ceiling in Barry Eastmond's Amek room was just one of the obstacles that had to be overcome.

Eastmond, who generally works alone, had to have all his gear easily accessible.

was able to soundproof both the studio and the adjacent voice-over room by isolating the walls and ceiling from the upper structure.

The windows were not really a problem. Just because people work with little black boxes doesn't mean they have to work in them. There is very little ambient sound penetrating the windows in Eastmond's studio and, when the need arises, they can be blocked off with soundproofing drapes.

The look and feel of the finished room is warm and comfortable, not the typical "high tech" coldness of many studios. The walls are covered in a rich blonde wood and Alderson installed a stone wall behind the Amek deck. Eastmond's home is built on a hill so that the back half of the studio is not below ground, but is actually at ground level. As a result the windows offer a pleasing view of the trees outside and also bathe his work area in natural light.

When there is a recording session in progress Eastmond works with an engineer and other people, but most of the time he uses the space by himself. He has to reach everything without moving. Consequently, the equipment has been installed in a circle that comfortably surrounds Eastmond when he works. The equipment includes the Amek deck, an Otari MTR-90 24-track recorder, a Roland DM80 8-track hard disk recorder, a Neve Prism, and a full rack of Focusrites.

Today, Eastmond can do a full-fledged 24-track recording in his basement studio and only has to go outside for final mixing. He is currently at work producing Anita Baker's latest album. EQ

IN YOUR INTERFACE

EQ&A

Q I am presently using a Tascam TSR-8 tape machine, Macintosh computer, and several MIDI modules in my project studio. (Now I stripe track 8 in record and then in playback I slave the synths, Mac, and drum modules to tape via a Tascam MTS-30 synchronizer.) I would like to upgrade by purchasing an 8-track, digital tape machine (DA-88, ADAT, or RD-8). Is there a way to sync my equipment to 15 tracks (minus track 8 on the TSR-8)? Could my Mac, MTS-30, or other MIDI gear via Mark of the Unicorn's MIDI Express (MIDI interface) tell the digital machine to start/stop and sync up to the analog/Mac system?

Ron Chassner
Miami, FL

A Using the Tascam DA-88 with the optional SY-88 sync card installed represents the simplest way to accomplish your goal. After striping SMPTE timecode onto the TSR-8 and the DA-88 (the subcode area of the DA-88 is used for timecode so that you do not sacrifice an audio track), it is then just a matter of sending the timecode from track 8 of the TSR-8 to the Master Timecode input of the SY-88 card. The DA-88 will then "chase" and lock to that incoming timecode.

There is also a timecode output available on the SY-88 card that can be used with any SMPTE-to-MIDI converter for "slaving" your MIDI devices. If the DA-88 has to be the "master" in your set-up, then a chase-lock synchronizer like the Tascam ATS-500 would have to be used. The ATS-500 also has a timecode output that can be used in conjunction with a SMPTE-to-MIDI converter for your MIDI devices.

If you are using a computer-driven sequencing package that supports MMC (MIDI Machine Control), then the computer can take the role as "master," using the MIDI ports also available on the SY-88 card.

Neal Faison
Supervisor, Customer Service/Technical Support
Tascam

ACROSS THE BOARD

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changed digitally in the Akai. The output of the SRC-2 was sent to the input of a Fostex D-10 DAT machine. All sample rates were 44.1 kHz.

The only manual moves were to fade up the surf at the beginning and to fade it out at the end. Everything else was automatic. After a couple of run-throughs, the music-to-surf levels were adjusted for each section and the final pass was printed to DAT.

After living with the DAT for a month or so, we decided to change some of the music levels. The changes were programmed into the Akai and the Master DAT reprinted. We listened back to the new DAT a few times and decided that we were finally finished. I logged on a piece of paper exactly where I wanted the various CD tracks to appear on the final product, and made a CD on the write-once Marantz 610 CD recorder. We then listened to make sure that the start IDs were in the right place and then sent the CD to the record company. They sent it on to the CD plant, and the write-once CD was used as the master for pressing the CDs. The start IDs were duplicated exactly as I had placed them on the master. The only expenses were for blank CDs, DATs, optical discs, ADAT tapes, and a couple of airline tickets to Hawaii.

There were no outside charges for mastering. There were no charges for studio time, since I did most of it in my basement. An album project like this won't have to sell very many copies before the royalty checks start rolling in. So crank up your project studio and start working on a project. A lot of record labels are much more interested in a potential project that is completely finished or only needs some final mixing. The chances of closing that "killer" deal are much better.

As for my next project, I'm trying to talk the record company into upping my budget so I can go to The Great Barrier Reef off of Australia and do some underwater recordings of the Great White Barking Sharks. I told them it might take a lot of trips to get a good recording. EQ

WE INTERRUPT THIS COLUMN...

The Roger Nichol's World Tour is on the road and doing well, thanks to all of you who continue to support and participate in it. Roger enjoys meeting friends and subscribers from around the country. A special thanks for a great event and seminar in February. Here is Roger's schedule:

April:

- Steely Dan tour of Japan

May:

- Wrap up of Walter Becker Project
- The first annual South American Pro Audio Expo, Buenos Aires, Argentina May 25–27, 1994
- Dates are still open for summer bookings

Fall:

- We are taking reservations for a grueling, intensive two-day master's engineering class taught by Roger in Nashville, Tennessee.

For more information on these events or to book your own event with Roger, call or fax Gail Carson at 615-646-1309; Fax 615-662-1951.

Un-Natural Disasters

Fighting a cold war of a different type

BY ROGER NICHOLS

I'm late with my column this month. On February 10th, Nashville and much of the mid-South suffered an ice storm. It was the worst one since 1951. The ice collected on the branches of all the trees. The build-up of ice became too heavy for the trees to support, so the branches at the top of the trees started breaking off. In some cases the trees became so top heavy that the entire tree fell over. In the process, the trees took the power lines with them. Most of the Nashville area was without electricity for about three days. Out where I live, it was ten days before I got my power restored. To top it all off, the average temperature for that ten days was around 20 degrees. We are located in an all-electric community, so there was no heat or hot water either.

How do I play with my studio toys without electricity? I checked with Tascam, Fostex, Panasonic, Sony, Roland, Akai, and Digidesign. None of them make propane-powered digital audio gear. Maybe since I was dealing with project studio gear I should have been asking for semi-pro-pain powered equipment.

I fired up a generator to get some lights and water (electric booster pumps). This worked pretty good most of the time for the studio, but the roar of the generator precluded any microphone-based recording. I also learned to save my computer work regularly after the first time I forgot to refill the generator gas tank in time.

If you have equipment that is sensitive to power surges (sequencers that lock up when your refrigerator kicks in), then you should have either a U.P.S. (uninterruptable power supply) or an Auto Transformer (a transformer that keeps the output voltage stable even though the input voltage varies) supplying your gear.

Battery-powered computers such as Apple Powerbooks or laptop PCs are pretty immune to these fluctuations because the internal battery smoothes out all of the voltage swings.

BACK TO SCHOOL

There are recording schools all around the country. I have done seminars at many of them. I was asked by a friend of mine at The Musician's Institute in Hollywood, CA if I would be interested in being involved with a recording school that they were starting.

I said that I would only be interested if I could have some control over the course study. I would like to help set up the class schedules and help determine how much emphasis was going to be placed on which areas of study. They said fine. So that's it. I am now affiliated with the Musicians Institute's new Audio Engineering Program, which will be a totally accredited degree program. This project will be a lot of fun.

I am also planning a couple of Master Workshops. They will be geared for people who have finished a course at one of the engineering schools, or those who have had plenty of project studio experience. It is really hard to make a great-sounding record when you have not had the chance to work with the best musicians in the best studio with the best equipment.

BEST NEW ARTIST

Probably not, but I just finished my first project as an artist. The Roger Nichols Project *Harmonic Ocean* on Triloka Records will hit the streets in April. Hopefully it won't hit the streets as recycled asphalt. Don't listen for it at your local dance club, but you might hear it next time you get a massage. It is an environmental surf tape with a music bed. I have purchased many of these tapes in the past, but could never find one that I liked, so I recorded my own.

The criteria here was to get the most quality for the least outlay of cash. I wanted to complete the whole project without having to use a commercial studio or mastering facility. It took two years of multiple attempts to get a good surf bed. The surf had to be uninterrupted for 70 minutes. I would not allow loops or edits of any kind. Hearing edited surf or overdubbed birds on "nature" tapes always made me nervous while trying to listen to them.

After selecting the right surf bed, my wife, who is a "real" artist, arranged and recorded the music using Opcode's Vision and her bank of samplers. A copy of the surf was put onto ADAT for a guide. One section of the music score needed voices, and they were recorded on the ADAT. All of the music (without surf) was mixed live from the sequenced modules and ADAT, through Apogee A/Ds, to DAT. Sometimes there would be more than one version of a mix. They were all stored on DAT tapes with SMPTE timecode. We locked up each version of the mix with the surf and decided which one worked the best. The winners were transferred digitally to the Akai DD-1000 Optical Disk Recorder.

The original surf master was played back on the Fostex D-20B, which spit out SMPTE for the Akai to chase. The digital output of the Fostex was plugged in to the A input of a Roland SRC-2 2-channel sample rate converter/digital domain mixer. The digital output of the Akai was plugged in to the B input of the SRC-2. The clock reference for the output of the SRC-2 was selected to follow the A input. The overall level of the surf did not have to be changed, and the music levels were

continued on page 105

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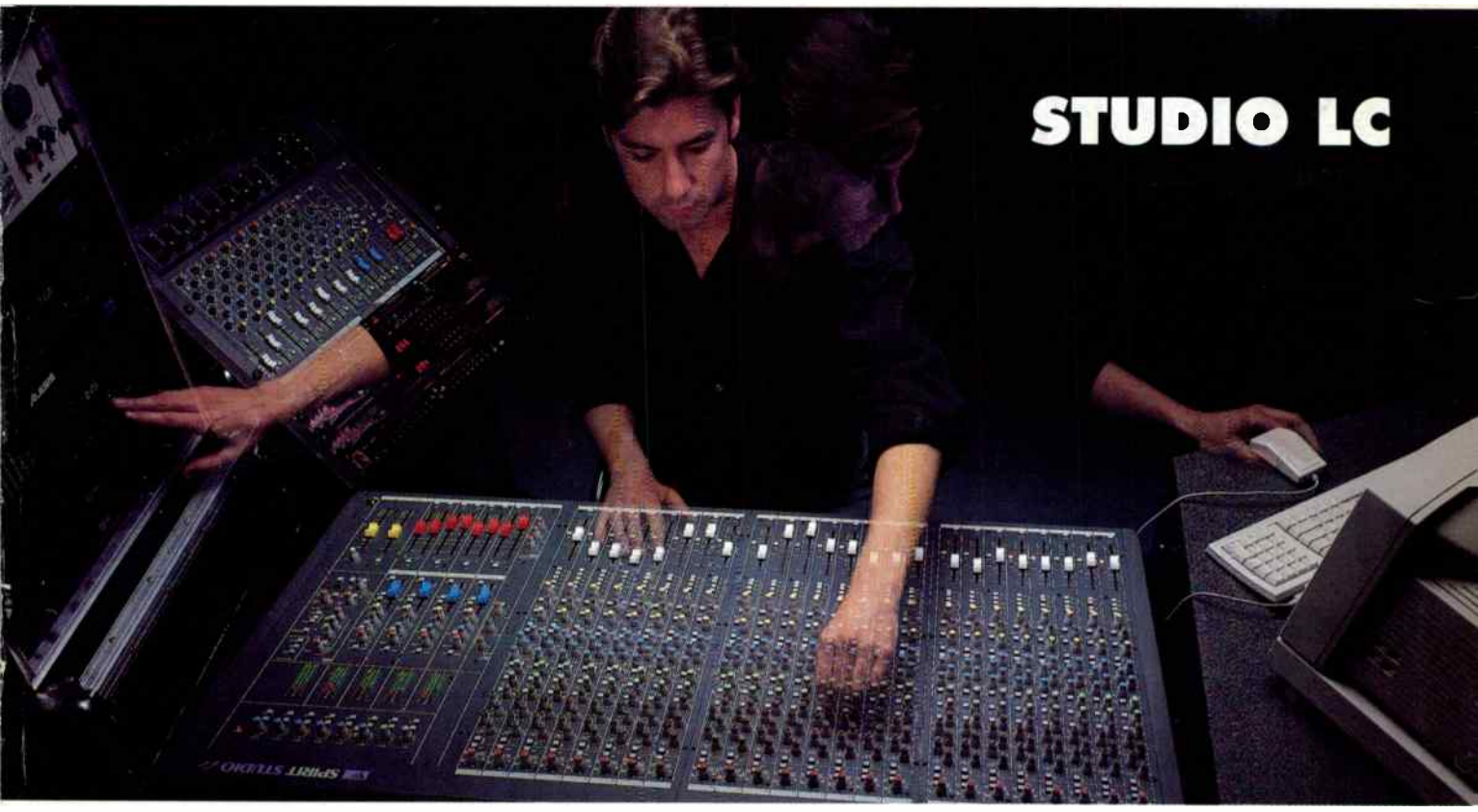

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