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BUILD YOUR OWN DIGITAL 8-TRACK STUDIO

EQ

THE PROJECT RECORDING & SOUND MAGAZINE

MAY 1994

SPECIAL ISSUE: WOMEN IN RECORDING

LES PAUL'S SOUND REINFORCEMENT TIPS

ROGER NICHOLS' SAMPLE RATE CONVERTER REVIEW

STEVE MILLER'S LONG DISTANCE SESSION NOTES

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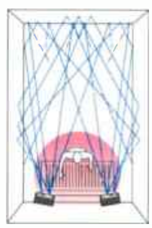
GAIL SKY KING: ROCKING THE DOUBLE STANDARD

The Truth From Left To Right

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, with 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. But in the near field, reverberant sound waves have little impact, as shown in the illustration. The Monitor One takes advantage of this fact and is built from the ground up specifically for near field reference monitoring.



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.

The Truth From Top To Bottom

The Monitor One's proprietary soft-dome pure silk tweeter design delivers natural, incredibly accurate frequency response while avoiding high frequency stridency and listener fatigue–typical of metal-dome tweeter designs. The Monitor One overcomes wimpy, inaccurate bass response–the sad truth about most small speakers–with our exclusive SuperPort™ speaker venting technology. The 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. The Monitor One's top-to-bottom design philosophy is a true breakthrough for the serious recording engineer.

Power To The People

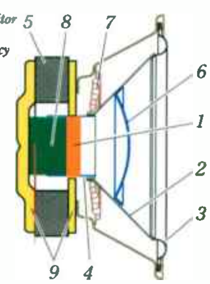
While most near field monitors average around 60 watt capability, the Monitor One handles 120 watts of continuous program and 200 watt peaks...over twice the power. The Monitor One provides higher output, more power handling capability, and sounds cleaner at high sound pressure levels. If you like to 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. Combined, these two specially formulated drivers deliver an 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 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 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.

For more information about the Monitar Ones and the Alesis Monitoring System, see your Authorized Alesis Dealer or call 1-800-5-ALESIS.

Alesis Corporation 3630 Holdrege Avenue Los Angeles CA 90016

ALESIS
STUDIO ELECTRONICS

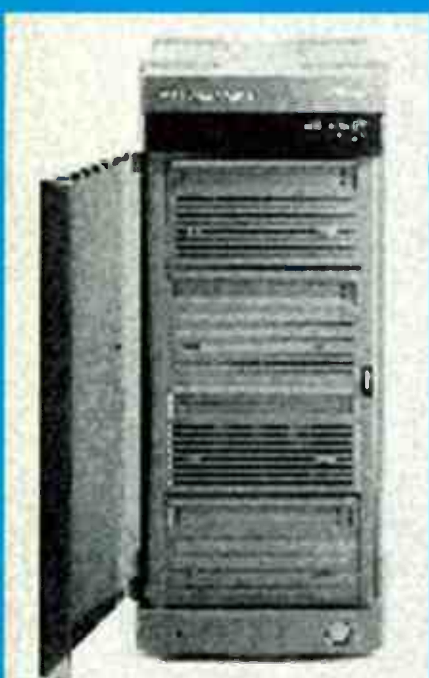
CIRCLE 02 ON FREE INFO CARD

EQ

PROJECT RECORDING & SOUND TECHNIQUES

VOLUME 5, ISSUE 5

MAY 1994



FEATURES

EQ LIVE

TECHNIQUES/WORKSHOPS

COLUMNS/DEPARTMENTS

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Cover: Gail "Sky" King by Julian Jaime.

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Everywhere you find the most demanding musicians and recording engineers, you'll find AKG mics capturing the powerful dynamics and emotion of their music. Now, that clear, clean AKG studio quality sound is available in a rugged dual-pattern microphone—the C1000—for **stage** or studio use.

The C1000 can be converted from a feedback-fighting hypercardioid vocal pattern to a broader cardioid instrument pattern—and back—in less than a minute. When your mixer doesn't have phantom power, the C1000's performance-balanced design hides a self-adjusting battery compartment to give you over 200 hours of full-headroom, self-powered operation from a standard 9V battery.

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CIRCLE 01 ON FREE INFO CARD

EQ

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LETTERS TO EQ

THE MIC IS RIGHT

I was flattered by the many phone calls I received in response to my letter published in the February issue of *EQ*. I am pleased to find that there are so many fans of old RCA ribbon mics, and that they read *EQ*.

Unfortunately, I do not do ribbon mic repairs as my time is fully taken up by my work for *Audio* and as an acoustical consultant. If you do need your ribbon mics repaired, I recommend you send your mics to the repair persons listed below. I am not sure their ribbons are all tuned to the numbers I think are right, but many customers are satisfied with the results.

Microphone Labs: Bill Hayes — Tel: 714-774-0342

Enac; Clarence Kane — Tel: 609-589-6186

Chas Grant — Tel: 310-867-1078

Jon R. Sank
Cross Country Consultants
Haddonfield, NJ

KORRECTION

The article in the February issue of *EQ*, by Francis Daniels, entitled "Special K," was a great little piece on our studio. There was, however, one (rather large) factual error. I have already received several calls wanting to know how we built our studio so cheap. The $15,000 number is quite misleading.

The article states that Studio K was built for under $15,000. This is not exactly correct. The cost is as follows:

Basic Materials (sheet rock, insulation, etc.) and labor for basic construction **$15,000**
Air Conditioning, electrical, acoustical treatment, glass doors, and windows **$20,000**
Wiring (snakes, patchbays, etc.) finishing, miscellaneous **$15,000**
Total Cost **$50,000**

This is a far cry from $15G, but still not bad.

Edward J. Kinslow Jr
Studio K
New York, NY

BYTE OFF MORE

As a subscriber I find it very frustrating that you have not gone out of your way to adequately explore the emerging world of computer audio. I've yet to see any articles devoted to actually comparing the sound and features of the various boards or the new multitrack software packages that are now being offered. Instead, I am having to look over the scattershot world of computing magazines (which do not specialize in audio) in order to get my questions answered. Some comparison tests, a look at what is to come, and info on the drawback or advantages of various chips would certainly be appreciated.

Emmack
Chicago, IL

[Editor's response: EQ*'s "Studio Software Report" (now incorporated into the magazine's regular body) has for 18 months provided information on the use and effectiveness of sequencing, editing, and sampling software, as well as details on MMC (MIDI Machine Control). Also check out Craig Anderton's MI Insider column, where Craig regularly discusses music computer hardware and software. Murray Allen has joined our staff as our multimedia columnist. In addition to one-time articles,* EQ *is committed to covering the ways that computers are revolutionizing the project studio. So stay tuned, we'll keep you covered.]*

CORRECTION

In last issue's Product Views (April, 1994) we printed the wrong phone number for Z-Systems. The correct phone number is 904-371-0990. For more information on the the Z-Systems Z-1 sample rate converter, see Roger Nichols's review on page 58. EQ

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.

DIAMONDS ARE FOREVER

...and Peavey invites you to experience the most innovative application of diamond strength, precision, and eternal durability ever. Diamonds are judged by the "Four C's" -- Cut, Color, Clarity, Carat. Peavey takes it one step further with the 5th and most important "C" of all -- Customer satisfaction. The new Peavey "Diamond" Microphones share these rare descriptions and more...

Cut: The new PVM™835Di and the PVM™880Di microphones "cut" through stage instrumentation and room ambiance like no other mics on the market.

Color: The less color in a diamond, the greater its value. Likewise, the less color in a microphone, the more accurate the audio signal.

Clarity: Microphone clarity is what it's all about...the PVM 835Di and PVM 880Di offer the epitome in clear signal reproduction.

Carat: The optimum level of "diamond" application has been determined through extensive transducer research and development.

CIRCLE 62 ON FREE INFO CARD

Stop by a Peavey dealer to learn more about the "5 C's" of the new Peavey diamond microphones...the PVM 835Di and the PVM 880Di.

PEAVEY

is a publication filled with the latest information on Peavey microphone technology. For a free copy write: ... Meridian, MS 39301 ©1994

EQ&A

SYNCHRONICITY

Q *Is there a way to synchronize my Tascam 488 cassette recorder with my Korg M1 sequencer?*

Conrad Gallmeyer
Bloomfield Hills, MI

A The synchronization of the Tascam 488 to the Korg M-1 can be accomplished in one of two ways. The most direct method would require the sequencer to have a Tape Sync feature. You would then use the following procedure:

1. Using your sequencer, create your song. (Be sure of the tempo you have chosen, because once you have committed this to tape it cannot be changed!)

2. Connect the Tape Sync output to the 488's Sync In jack. (The Sync switch at the rear of the 488 should be in the On position.)

3. Place track 8 into Record (Record + Play) and allow the tape to run for about 3 seconds (pre-roll).

4. Gentlemen, start your sequencer! (At this time a sync tone generated by the sequencer is recorded by the 488.)

5. When your song has ended, stop your sequencer. Allow the tape to continue recording for another 3 seconds (post-roll).

6. Take track 8 out of record and rewind your tape. Connect the 488's Sync Out to the sequencer's Tape Sync input. (You must inform the sequencer that it should start to play based on information coming from this input. Consult your operations manual for exact procedure.)

7. When the 488 is played, the sync tone recorded on track 8 will start the sequencer. You can now record audio on the seven available tracks of the 488 that will play in "perfect sync" with the audio from your sequencer.

If a Tape Sync feature is unavailable, synchronization can be accomplished by using the Tascam MTS-30 MIDI tape synchronizer. This device is known as a MIDI-to-"smart"-FSK converter. ("Smart" FSK includes location data or Song Position Pointer information that normal FSK does not. This allows you to start the tape at any point in your song and the sequencer will be able to follow.)

The procedure would remain the same as in the "Tape Sync" example. The differences are as follows:

1. Connect the MIDI Out of your sequencer to the MIDI In of the MTS-30. Connect the Tape Out of the MTS-30 to the Sync In of the 488. (The Sync switch at the rear of the 488 should be in the On position.)

2. Press the MTS-30's Tape key until "SAVE" appears in the display.

4. Take track 8 out of Record and rewind your tape. Connect the MIDI In of your sequencer to the MIDI Out of the MTS-30. Connect the Sync Out of the 488 to the Tape In of the MTS-30.

5. Press the MTS-30's Tape key until "LOAD" appears in the display. Following the procedure in your sequencer's operations manual and set up the sequencer to accept "External MIDI Clock."

6. When the 488 is in Play mode, the sequencer will once again start and play in sync with the tracks you will record.

This process is explained with much greater detail in a publication available from Tascam called "MIDI Basics (Synchronization and Control)." If you would like to receive a copy free of charge, call 213-726-0303, ext. 463.

Neal Faison
Customer Service
Operational Support Supervisor
Teac America

GET A CUE

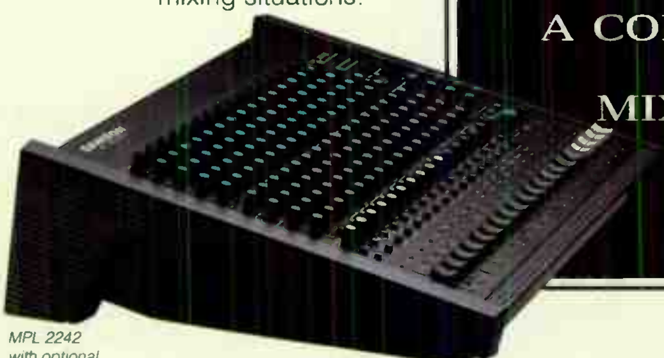
Q *My goal is to lock up a Fostex D-10 DAT recorder to a Tascam DA-88 8-track digital machine. I do production work, and when mixing*

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 rack-mount 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

to DAT I want to go back over segments and punch in until I get the mix I desire. I heard JLCooper's CuePoint might lock the machines together, but since the D-10 has no sync card, that seems unlikely. Is it true JLCooper is coming out with a serial card this May that'll allow this lock up? Also, will I need the optional DA-88 sync card?

Gordon Nicol
Dallas, TX

A Thanks for considering the new JLCooper Electronics CuePoint Universal Autolocator. Although CuePoint is designed to control various devices including the DA-88, it is an autolocator, not a synchronizer. CuePoint provides low-cost, versatile transport control of multiple machines. It features track arming, automated transport functions, a built-in SMPTE reader/generator with SMPTE/MTC conversion. In its basic form, it will control any MIDI Machine Control-compatible computer software or hardware unit. It is fully compatible with the Alesis ADAT, Fostex RD-8, and Tascam DA-88.

CuePoint also incorporates an additional expansion slot that can be fitted with a variety of optional control modules. These allow CuePoint to be customized to fit individual requirements. Modules presently shipping include "dataCARD" for syncing to ADAT without using an audio track. Future modules include an RS-232 card for controlling serial tape transports, RS-422, and Apple Desktop Bus.

According to Fostex, the D-10 is not designed to lock to external sync of any kind. Other Fostex "timecode DAT" models like their D-20, D-20B, or D-30 models have both stereo audio tracks and a subcode sync track.

The Tascam DA-88 can be master or, optionally, a slave master, so it would work in a similar scenario using a timecode DAT in place of the D-10.

Eli Slawson
Product Specialist
JLCooper Electronics

HOW SWEDIEN IT IS

Q *In an EQ article [June 1991] about producer/engineer Bruce Swedien's project studio, I noticed he had an MCI 16-track, 2-inch analog tape machine. I also own such a machine — bought in 1974 — and wonder what tape speed, tape type and noise reduction Mr. Swedien uses.*

Burt Norton
Tallahassee, FL

A I tracked Bruce Swedien to a major studio where he was knee deep in music charts, sequences, and tape working on another mega project. Always the gentleman, with one hand on the pause button and the other on the telephone, he said that, although he loved that MCI machine, he has since sold it. Nonetheless, while he owned it, he did not use noise reduction; always operated at 30 ips; and initially used 3M 250 tape and then 3M 996 (996 @ 46 operating levels).

Hector G. La Torre
Executive Director
EQ Magazine

EQ TIPS

EDITED BY STEVE LA CERRA

COVERING ALL BASSES

In many project studios the bass guitar is recorded direct for convenience and lack of an isolation booth in which to put the bass amplifier. Some basses, however, lack clarity and depth when recorded direct.

The next time you are mixing, try this idea to give the bass sound some depth and bite: Run the bass track directly from the multitrack tape machine into a small amp or a power amp/speaker cabinet combination. A bass amp is great, but a guitar amp can work just as well as long as you are careful with the volume level. Mic the bass amp and return the mic into the mix. While you are mixing, the bass amp will be playing live into the microphone and you will be adjusting the level of the mic into the mix. Since it is likely that the studio will be vacant at mixdown time, you needn't worry about isolating the bass amp from other instruments. You can try using compression on the microphone signal or you could patch a compressor between the tape track and the amplifier.

For even more versatility, patch the tape track into a guitar or bass preamp and then from the preamp out to the amplifier. These techniques can help avoid that typical "flat" or "lifeless" tone that a bass can have when recorded direct. You can also run the signal into the amp from a send on the console, thus giving you the ability to mix both the original direct track and the "new" miked track.

SIMPLY SMPTE

When locking your computer to a multitrack tape machine via SMPTE timecode there are several things to be aware of. When you are recording SMPTE code onto a tape track, run a cable from the SMPTE output of the interface directly to the tape machine input. Also run a cable directly from the output of the tape track back into the SMPTE interface, bypassing any patchbays. SMPTE code has a nasty habit of leaking through channels and can find its way into your music through marginal grounds in a patchbay. By running the cables directly between the tape machine and SMPTE interface, this problem can be avoided

If you are recording on an analog tape machine, record SMPTE on an edge track (usually the one with the highest number) at a level between -5 and -10 VU. If you print SMPTE too hot it will leak into the next track. Leakage can be reduced by using an edge track and also leaving the track next to the SMPTE track empty. If you must record on a track next to the SMPTE code do not print a hot signal or it may modulate the SMPTE code.

With some computers you may need to turn a screen saver program off when locking to tape via a SMPTE-to-MIDI interface. The graphics of a screen saver can require a substantial chunk of memory and processing power, and for some slower computers (like the Macintosh Classic) performing this function while locking to SMPTE code is too taxing for the microprocessor. The result will be timing errors in the MIDI information being played from the sequencing program especially during songs with fast tempos. EQ

Have any special techniques that help you get the job done? Share with your fellow audio professionals and send your tips to:
EQ Editorial Offices,
939 Port Washington Blvd.,
Port Washington, NY 11050
America Online: MPANDA

The Seriou

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

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

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

Goes Faster, Lasts Longer And Takes A Beating

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

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

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

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

You Already Know How To Operate It

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

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

s Machine

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

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

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

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

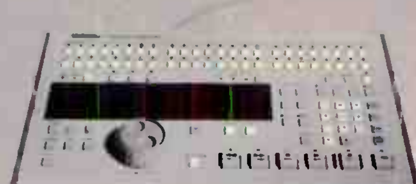
That's not all, you also can set your pitch (± 6%), sample rates (44.1 or 48K), as well as crossfade and track delay times. All from the front of the DA-88.

Complete Synchronicity

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

A Digital Recording System That Grows With You

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



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

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

Listen To The Rest

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

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

TASCAM

Take advantage of our experience.

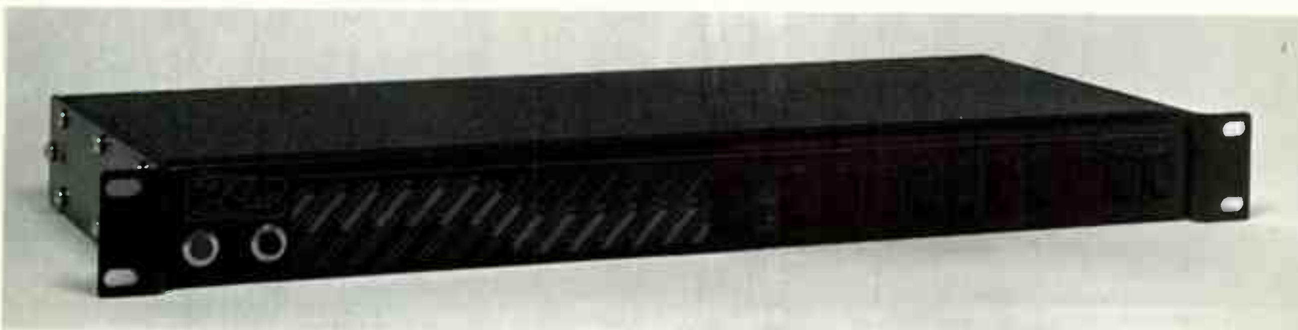
7733 Telegraph Road, Montebello, California 90640
(213) 726-0303

CIRCLE 55 ON FREE INFO CARD

EQ PRODUCT VIEWS

HARMONIZER PLUS

Eventide's new H3000-D/SE Dynamic Ultra-Harmonizer features a new Mod Factory algorithm that adds dynamics and 100 programs. In addition, 100 new presets from several top-name artists, including Bob Clearmountain, plus new 3D speaker-based spatial imaging programs from Empirical Labs are at your disposal. The H3000-D/SE's spatial effects actually create desired effects in space, instead of just adding a third dimension to already-processed sounds. The 3D presets manipulate phase, frequency response and interchannel delays to achieve psychic space. For further information, contact Eventide, One Alsan Way, Little Ferry, NJ 07643. Tel: 201-641-1200. Circle EQ free lit. #101.



SOLID GROUND

New to Whitenton Industries' Juice Goose line is the Rackpower 300 isolated ground distribution center. If a ground loop exists because of an insufficient primary ground system, a ground select switch routes the grounds for all powered equipment to a "STAR Ground Terminal" on the back of the unit. A cable attached to this terminal can be connected to a true earth ground. If the loop is caused by grounding problems within the equipment rack, 10 separate ground lift switches on the front of the Rackpower lift the ground for any combination of the 10 outputs. For more information, contact Whitenton Industries, 7320, #104, Houston, TX 77081. Tel: 713-772-1404. Circle EQ free lit. #102.

MULTIPLE PERSONALITY

Ramsa's new WZ-DM30 is a one-input/four-output digital multiprocessor. In one unit, the WZ-DM30 offers compressors/limiters, a graphic equalizer, a 4-way crossover network, and 4-band parametric equalizers. All controllable parameter settings can be stored in 16 event memories in each mode and can be retrieved instantaneously. The unit can be connected to external MIDI equipment such as a synthesizer and an effector. It features 20-bit digital floating A/D and MASH D/A converters to provide a typical dynamic range of 110 dB. Retail price is $3600. For more information, contact Ramsa/Panasonic Pro Audio, One Panasonic Way, 2A-2, Secaucus, NJ 07094. Tel: 201-348-7846. Circle EQ free lit. #103.

NEW AND IMPROVED

Klark-Teknik has introduced the DN3600A programmable, digitally controlled, graphic equalizer with Version 2.0 software. The DN3600A features a brighter screen display than the previous DN3600, which was introduced last year. Owners of the original DN3600 can have their units upgraded for a nominal charge. With Version 2.0, additional information is displayed at the top of the screen, including frequency and level display of the selected fader, name of the last recalled memory, Q mode ,and channel selected. For more information, contact Klark-Teknik, Mark IV Pro Audio Group, 448 Post Road, Buchanan, MI 49107. Tel: 800-695-1010. Circle EQ free lit. #104.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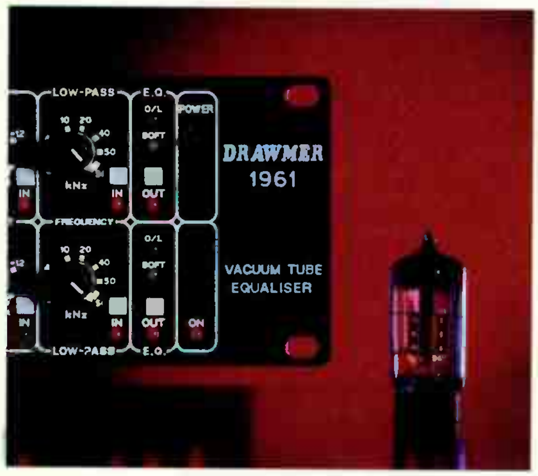
EQ PRODUCT VIEWS

SUPER TAPE

3M has introduced the Audio S-VHS Digital (ASD) format mastering cassette for ADAT-type recorders. The 3M ASD 40+ will provide recording engineers with about 44 minutes of record time. The cassette has been optimized by a refined tape process to produce very low head wear. Other features include an exclusive binder system and a stabilized polyester backing that work together to ensure very low errors and an increased durability in the editing process. In addition, it has a durable shell made from warp resistant, temperature stabilized polymerized plastic that protects the tape from mistracking and edge damage. The 3M ADS has antistatic leader and trailers, comes packaged in both sleeve and album, and includes labeling designed for digital audio mastering. For complete details, contact 3M Pro audio, 3M Center, St. Paul, MN 55144-1000. Tel: 612-733-3477. Circle EQ free lit. #105.



TUBE TALK

Drawmer recently began offering its 1961 vacuum tube equalizer to the U.S. market. It combines wide spectral control with the warmth and sweetness associated with tube circuitry. Each channel has an input level control with a level meter that allows you to optimize the signal and drive the tubes hard or gently. Each of the four main equalizer bands uses a separate tube section for increased harmonic clarity, providing soft clipping that is easily controllable via the input-level adjustment and LED indicators. The four main equalizer sections have six switchable overlapping frequencies, and bandwidth variable from .3 octave to 3 octaves with ±18 dB boost or cut. The fully balanced system includes a variable high-pass section allowing a 12 dB/octave roll-off from 20 Hz–500 Hz. Retail price is around $2700. For complete details, contact QMI, 25 South St., Hopkinton, MA 01748. Tel: 508-435-4243. Circle EQ free lit. #106.

DIGITAL MONITORING

Genelec's 1030A active studio monitor is a fully powered compact monitor system designed for project studios using ADATs, DA-88's, or hard-disk recording systems, and for smaller post suites. It uses Genelec's DCW technology in conjunction with high-end driver technology. The woofer is a high-efficiency, 6-1/2-inch polymer composite cone. The metal dome tweeter measures 3/4-inch. The enclosure is constructed of a vinyl-sprayed MDF and utilizes an unusually thick front baffle that results in a rigid, inert cabinet, minimizing tonal coloration in the midband. Retail price is about $2000 a pair. For more information, contact QMI, 25 South St., Hopkinton, MA 01748. Tel: 508-435-4243. Circle EQ free lit. #107.

TIME IS ON YOUR SIDE

Brainstorm Electronics has introduced the SR-15+ distripalyzer, a timecode analyzer, timecode distributor, and pilot tone stripper, all in one single-rack unit. The timecode analyzer identifies 24, 25, or 30 fps code, drop frame or nondrop frame, and reference to video rate 29.97 or 30 fps. The timecode distributor/reshaper offers five buffered and balanced outputs with individual level controls. The pilot tone stripper extracts pilot tone (50 or 60 Hz) from timecode, video, or AC and is synchronous with the source. The SR-15+ detects and displays on the front panel all common timecode problems. Retail price is $1490. For further information, contact Brainstorm, 1155 N. La Brea Ave., West Hollywood, CA 90038. Tel: 213-845-1155. Circle EQ free lit. #108.

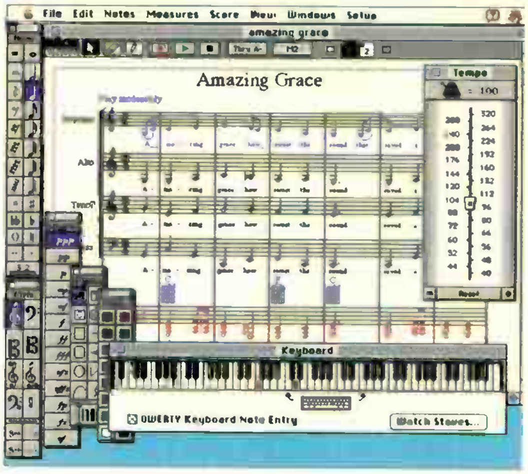
NEED A LIFT?

Vision Audio is manufacturing a clean solution to the placement of aural and visual monitors for the project studio. Typically, the small studio has the nearfield, computer, and TV monitors placed on the meter bridge of the console or in a similar position, creating clutter and putting these devices in a tenuous position. The company's Clearview Monitor Lift stands neatly behind the console and safely supports the monitors in an aesthetic fashion. Previously only available in large-frame configurations, Vision audio is now offering the Clearview Monitor Lift Series E for smaller-frame consoles. For more information, contact Vision audio, 611 Anchor Dr., Joppa, MD 21085. Tel: 410-679-1605. Circle EQ free lit. #109.

EQ STUDIO WARE

I WRITE THE SONGS...

MusicTime 2.0 from Passport Designs allows a standard QWERTY computer keyboard, as well as any MIDI instrument, to trigger your desktop music making. It will record your performance live and immediately notate it in standard music notation. You can compose sheet music, charts, and lead sheets and edit your compositions any way you wish. Compositions can be scored with up to eight staves. For more information, contact Passport Designs, 100 Stone Pine Rd., Half Moon Bay, CA 94019. Tel: 415-726-0280. Circle EQ free lit. #110.



KEEP THE POWER

Controlled Power Company is offering the MD Series of uninterruptible power system. Computers run on power provided by switch-mode power supplies, which use power from standard outlets to provide the pulses of power the computer power supply requires. The quality of that power must meet the machine's requirements, or glitches and crashes occur. The MD series eliminates sags, surges, spikes, noise, brownouts, and even total blackouts. It provides protection for mini and mid-range computer systems, LAN/WAN systems, and other critical applications. Battery run-times are available from 10 minutes to hours. SIzes range from 3.1 kva to 7.1 kva. For more information, contact: Controlled Power Company, 1955 Stephenson Hwy., Troy, MI 48083. Tel: 810-528-3700. Circle EQ free lit. #114.

PERFORMER POWER

A major upgrade in Mark of the Unicorn's Performer Macintosh sequencing software, namely Version 5.0, has been announced. Performer Version 5 has a realistic, three-dimensional look in its support for color monitors. A new MIDI machine control window allows Performer 5 to work with any MMC-compatible device. Performer has a graphic display of online MMC devices and can recognize

a variety of MMC products to automatically configure your setups. Performer Version 5 also offers 50 "DNA" Grooves created by WC Music Research. For complete details, contact MOTU, 1280 Massachusetts Ave., Cambridge, MA 02138. Tel: 617-576-2760. Circle EQ free lit. #112.

ACTION PACKED

Macromedia's Action! CD-ROM bundle for Macintosh and Windows combines the Action! presentation program with professional sound-editing software and royalty-free ClipMedia elements. The Mac version of the bundle includes Action!, SoundEdit Professional, and ClipMedia 3 elements. The Windows version combines Action!, Turtle Beach's Wave Tools and MIDI Tune-Up sound editing software, and ClipMedia 2 elements. Retail price is $399. For more information, contact Macromedia, 600 Townsend St., San Francisco, CA 94103. Tel: 415-252-2000. Circle EQ free lit. #113.

UP, UP AND ORRAY

Pinnacle Micro has premiered its Orray technology, which is the first 5.2 GB optical disc drive product to combine the multihead, multiplatter approach of magnetic disk drives with the removability and reliability of magneto-optical into one complete storage system. The Orray is ideal for digital audio and digital video; and it is plug-and-play ready for Macintosh, PC, Sun, HP, Silicon Graphics, and other computers. The Orray offers up to an 8 MB/second data transfer rate and writes simultaneously to one set of optical media, equaling 5.2 GB of storage capacity. Retail price is $14,995, and each 5.2 GB media set sells for $799. For complete details, contact Pinnacle Micro, 19 Technology, Irvine, CA 92718. Tel: 800-553-7070. Circle EQ free lit. #111.

chassis ground bus. This grounding scheme is common in consoles in this price range. In this case, because of the unbalanced connectors, I would have liked a permanent ground-lug attachment for studio installation. All told, the Mackie represents a great value for the money, but it requires some savvy to wire it up to a balanced patchbay.

Now we get to the tricky part. We wanted the unit to be easy to use but flexible and sophisticated. The way we achieved this was through a custom patchbay that puts everything where you want it for tracking operation but enables you to put anything into any place to be creative — within reason (let's not put outputs into outputs). We went with a custom manual patchbay designed by Sapphyre Productions and built by Audio Accessories.

For processing reverb and delay, I am a firm believer in products that are quiet. We also needed tools that would allow a producer or production engineer to repair tracking problems in areas such as vocal, drum, and piano tracks, either in mixdown or during premastering sessions. This is an area where a parametric equalizer is invaluable. We specified an Orban 642 dual 4-band or mono 8-band unit for these chores. I also recommend Cadac's 6833. In addition, there are some utility 1/3-octave equalizers for tone coloration and gentler repair work. In this case we used dbx 153IX.

To save money without compromising quality, we chose two Symetrix 564E quad gates that we have had very good luck with. They are used for percussion and special effects.

We also like to recommend a fairly good compressor section for premastering work and for vocal tracking (in digital facilities I tend to think in terms of limiting rather than compression), and used JBL/UREI LA-12's.

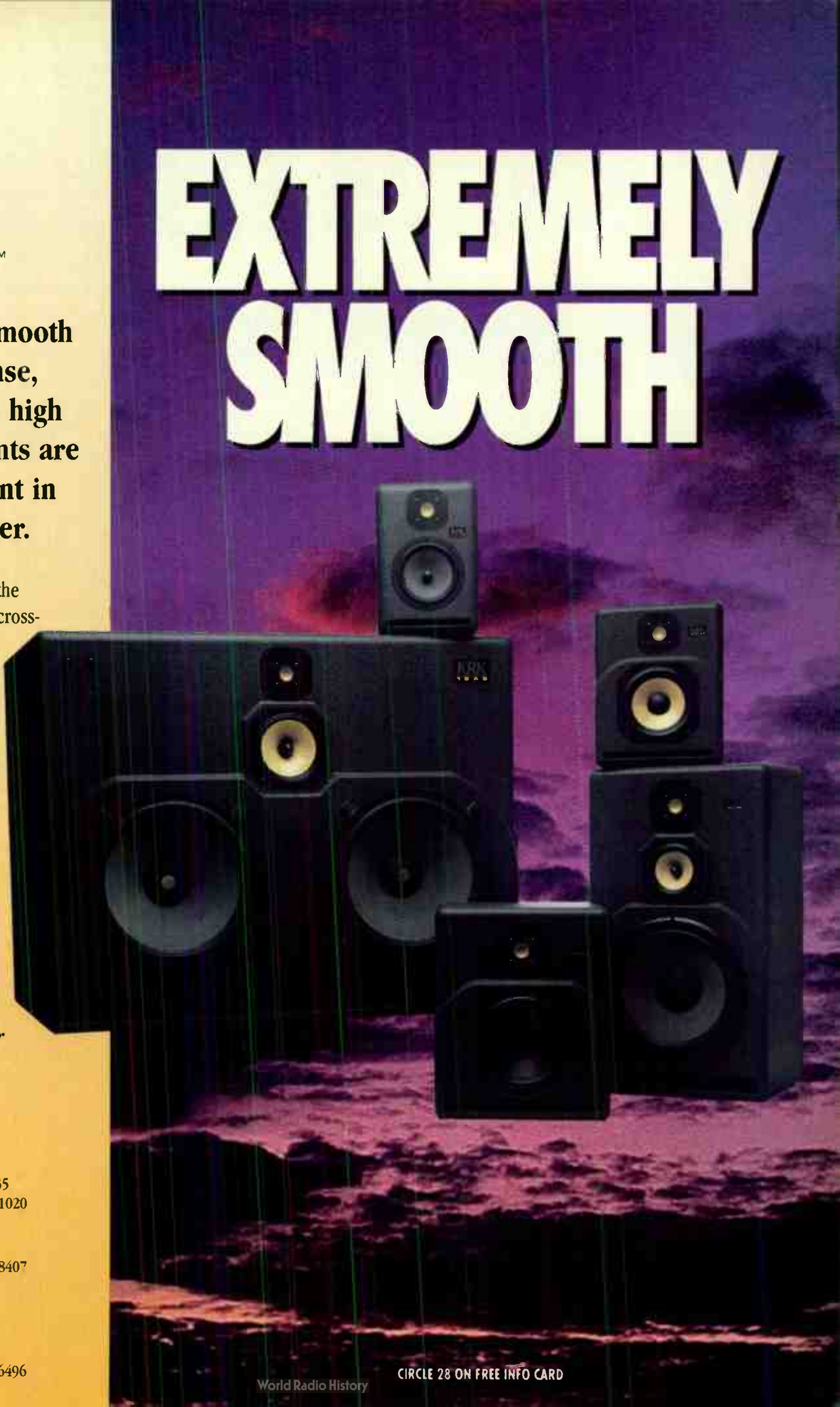
For effects, I prefer MIDI central control. We chose good workhorses in the Lexicon LXP-1 and LXP-5, with an MRC controller — the LXP-1 for its solid reverbs and the LXP-5 for its good VCO section (nice pitch shifting, special effects, chorus, and flanging).

For the delay and chorus, I wanted a high-quality unit for mixing because this is one of the areas that has not really been of quality in older units. We ended up choosing a Sony DPS-D7 that had some very good presets on which to base any audio experiments one might wish to make.

THE MIC STUFF

All this stuff is great, but to get any material from people or things making that good old studio air resonate you need good old microphones, and I wanted to provide something flexible that wouldn't clean out the budget for the rest of the facility. I had used the mics in the AKG Blue Line Condenser Series when they first came out and liked them as an alternative to the AKG C451 and C460 for budget reasons. Three preamp SE-300B sections were specified with the CK-94 figure-8 capsule, CK-92 omni capsule, and three CK91 cardioid capsules. Also included was a Shure SM-81 condensor, which has a slightly different response and tone from the AKG SE 300B/CK91 cardioid.

For lead vocal and acoustic instrument recording, an AKG CK-414B/ULS condensor was included. This can have its patterns changed to figure 8, omni, cardioid, and super cardioid.



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For high-volume instruments and percussion-type recording where a condensor can be overloaded, a few of the industry standards came into play. A couple of Sennheiser MD 421's (a staple of the pro audio-reinforcement industry) and Sennheiser MD 409's, that look like a gold and black spatula but work really well for drums and guitar were chosen.

To monitor all this we needed good, compact monitors and high-quality headphones. As control-room monitors, Tannoy NFM System 8 VER II's were selected for their coaxial design and flat response. For headphones, a set of AKG 240DF Calibrated was used because of its good response.

Probably the most difficult aspect of the whole design process is that there is no standard input, output, connector, pin out, grounding, and shielding scheme by manufacturers. This means that a good portion of the patchbay has to be wired in by hand. A good example of this is that the Mackie 8•Bus uses its chassis ground for a common signal carrier on its channel inserts.

The majority of pro equipment has balanced inputs and outputs using connectors that are either XLR cannon type; or 1/4-inch TRS (tip, ring, sleeve) type; or barrier strips, which I prefer because there is usually a provision for separating the chassis ground from the signal ground. Unfortunately, semi-pro gear tends to have either 1/4-inch TRS or 1/4-inch TS (tip, sleeve) unbalanced connectors.

What this means is that in a smaller system with only 16 channels of record/playback and a full patchbay, you will have to hand wire (unless your middle name is Midas) most of your processing gear and some of your console equipment to the patchbay if it uses "professional" jacks such as 1/4-inch-long frame type, which is a balanced-type connector with a TRS configuration, or if you use "MINI," which is also a miniature TRS-type connector, to save space.

I would like to thank Jerry Putnam (Airwaves Studio), Mike Biladeau (Strings 'n Things), Timothy Symonds (Audio Accessories), and Bill Scranton (Boynton Studio) for all their help on this project.

Jonathan Lang is the owner of the New Hampshire-based Sapphyre Productions, a full-service engineering, sales, and installation company.

The Electro-Voice 950

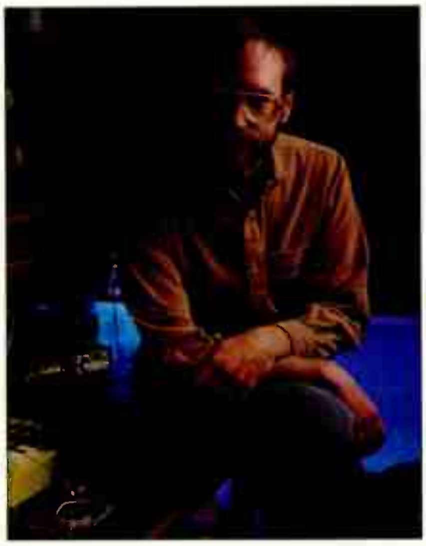
Crystal-clear sound from E-V's vintage 1940's classic

MICROPHONE NAME: Electro-Voice 950 Cardax
TYPE OF MIC: Crystal
FROM THE COLLECTION OF: Bob Chadderon, Electro-Voice
SERIAL NUMBER: 9724
YEAR RELEASED: 1943
VALUE WHEN RELEASED: $42.50 (1952 price)
VALUE NOW: Acoustic: Unknown; Aesthetic: Priceless
POLAR PATTERN: Unidirectional Cardioid
CASE: High-pressure diecast zinc
FINISH: Satin chrome
FREQUENCY RESPONSE: 70–9000 cycles per second (c.p.s.) switchable ring response
MEASURED RESPONSE: 100–5000 c.p.s.
IMPEDANCE: Hi-Z
OUTPUT Z: 100 kΩ
APPLICATIONS: A real workhorse that has been used for everything from PA to music.
OWNER COMMENTS: This mic is a real example of how far we have come in 50 years. It looks a lot better than it ever sounded. The response is optimized for what the spec sheet called "a clean and brilliant" sound with "increased articulation and sibilance." Mics for these applications today have the emphasis on "quality sound" rather than on maximum intelligibility and styling.
ORIGINAL SPEC SHEET BOASTS: The Cardax is designed for all types of voice and music reproduction, indoors and outdoors. It performs faithfully under adverse acoustical conditions because of its directivity and dual response. The wide-angle front response and increased pick-up range are ideal for group work. The cardioid pattern gives the user more freedom of movement. Recommended for all types of better-quality public address, recording, paging systems, remote broadcast, dispatching, and radio communications. EQ

CIRCLE SURROUND CONTROLLER

CIRCLE SURROUND DECODER

Studio-in-a-Box: Virtually Yours



Imagine the day when the bulk of your project studio setup consists of a computer, keyboard, mouse, and monitor

BY CRAIG ANDERTON

Computers have redefined the music scene in several ways, but one of the most significant ways is that they have changed the physical to the abstract. MIDI sequencers have replaced the actual tape recorder with a model of a tape recorder — gears, belts, and heads are now lines of code. And that's not all. Vibrating strings are modeled with software, as are the characteristics of signal processors and even the unique qualities of tube gear.

But why stop there: why not model an entire studio in a computer? Already, there are several signs that the "studio-in-a-box" is virtually here:

• The Macintosh AV series with built-in DSP. Deck II already runs on these machines and provides sophisticated multitrack digital recording.

• Atari's Falcon030. Team this with Cubase Audio, and you have not only digital audio, mixing, and sequencing, but some signal processing as well.

• Peavey's MediaMatrix, which was announced at the last AES and that puts all the modules needed for live or studio use into an IBM PC — just add I/O cards.

These build on the precedents set by the Fairlight, Synclavier, WaveFrame, and similar workstations (or for those with an analog mindset, the Portastudio concept), but offer greater flexibility due to increases in the state of the art of both digital audio and computers. The downside, of course, is that all-in-one systems come with limitations — or do they? Consider this:

• "Plug-ins" for Digidesign's Pro Tools, Sound Tools, and OSC's Deck allow third-party developers to create pieces of software that customize digital-audio programs to do specific tasks. Need sophisticated compression or limiting? Plug in Jupiter Systems' MDT. How about guitar-style processing? Plug in Hyperprism.

• Software upgrades have become a fact of life. Older workstations were limited by their hardware; thanks to general-purpose DSP, a virtually infinite number of variations on a theme are possible. This is also a "greener" approach since it gives hardware a longer shelf life.

NOT READY FOR PRIME TIME?

The advantages to the studio-in-a-box are obvious: interfacing ceases to become an issue since all elements are integrated via software; automation is a given (not an add-on) that can capture all parameters; and a single control surface and interface works with every aspect of a system. (And we don't even have to wait to enjoy these advantages: protocols such as MIDI Machine Control can already link disparate units into a more unified system.)

But technology is always a double-edged sword, and we have some serious issues that need to be addressed before the studio-in-a-box becomes a better way of creating music. How about:

• **Physical controllers.** Ever try to mix 8 channels with a mouse? For your sake, I hope not. Unless we can quickly access the power offered by the studio-in-a-box, that power is for naught. It's no accident that physical controller boxes such as the JLCooper FaderMaster and Peavey PC-1600 have been extremely successful, since they let you get a lot more mileage out of the silicon they control.

• **D/A and A/D "effects loops."** There will always be that vintage tube compressor or analog goodie that needs to flow into the digital data stream. Unless provisions are made for these devices, it will be much harder for the diehard tweakers to come up with truly unique combinations of the old and the new.

• **Crashproof software.** Imagine this: you have a recording studio in a box, or a live-sound setup, and the software freezes. The results could be horrifying — do you lose everything you've recorded? Does a live setup go into uncontrolled feedback? It's bad enough when a synth or sequencer goes down, but imagine what would happen if all your signal processors, recording devices, and sound generators flip out at once. It will not be a pretty sight. Redundancy is not a possible answer; software has to be bulletproof, which is not an easy task. Expect to pay for that privilege.

• **Interdevice communication standards.** MIDI was a wonderful, and perhaps unique, fluke — just try to move multitrack digital audio between the various workstations and you'll probably experience an exercise in frustration. If each studio in a box is an island unto itself, we will all be in trouble. Perhaps the best option is to devise a common backup standard with a standardized way of dealing with digital audio and an optional set


ALESIS
QUADRASYNTH
ALESIS
QUADRASYNTH

> Advantages to the studio in a box are obvious: interfacing ceases to become an issue since all elements are integrated via software; automation is a given; and a single control surface works with every aspect of a system.

of parameters (EQ, effects sends, etc.) that can be implemented to a greater or lesser degree. The model here is MIDI: all keyboards deal with notes the same way, but not all keyboard implement the same types of controllers or pressure options.

• Standardized interface. Walk into a 24-track analog studio anywhere in the world and you'll be able to figure it out — faders control levels, knobs control EQ, and so on. I believe one of the reasons why the Mac was able to hold its own against the PC clone glut was the standardized interface: commands to save, quit, create files, and so on did not change from one program to the next. In fact, Mac users would look down on programs that didn't follow the "Mac interface." Let's not revisit the world of synthesizers, where combinations of programs are called multis, combis, performances, etc. Not only do we need some kind of "interface rulebook," we desperately need a standardized vocabulary.

THE WAY WE'LL WORK

As someone who has assembled a studio out of bits and pieces over the years, I've come to appreciate anything that can create a more unified working environment. Even something as apparently simple as being able to control a tape recorder from a sequencer can make a huge difference in ease of use and efficiency. Having MIDI sequencers follow along with digital audio recorders provides a similar benefit. Multiport MIDI interfaces such as Opcode's Studio 5 and Mark of the Unicorn's MIDI Time Piece II have saved countless hours that would have otherwise been spent repatching. Overall, the studio-in-a-box concept is a real time saver, and now that the third wave of personal computers is coming on-line (as exemplified by the PowerMac), there should be enough processing power to do what we really need to do.

But in the process of being seduced by a new technology, we have to be careful. We must never lose sight of the fact that technology must be a servant to the music-making process. Unless the studio-in-a-box is fast, interactive, intuitive, and flexible enough to allow for "happy accidents," we'll be taking an artistic step backward even as we take a technological leap forward.

Craig Anderton is co-author (with Bob Moses and Greg Bartlett) of the ultra-cool new book, **Digital Projects for Musicians**. *There is a subliminal message hidden in this bio that will cause you to run out immediately and buy it.*

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"...so quiet I had to doublecheck to make sure it was on." M.C., Los Angeles, CA
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"The mic input circuitry is the remarkable low noise design that first brought Mackie into the spotlight." H&SR (UK) 2/94
32•8 shown (instead of 24•8) because we had a cooler picture of it.
"I'm happy to report that the desk maintains Mackie's reputation for clean, quiet circuit design. Some of my tests, involving CDs, showed up the noise on the original recording quite clearly. Even without the EQ switched in, the desk displayed a very open, transparent quality." Sound on Sound (UK/Europe) 12/93
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MACKIE DESIGNS 8-BUS MIXER
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NEW! 24•E 24-ch. expander
NEW! 11-rack space "Sidecar"
"The board's price may put its primary market in the personal studio and small project studio, but it's crammed with truly professional features. Home recordists can stay with the Mackie 8•Bus as they upgrade from semi-professional to professional gear, thanks to the board's ability to run either +4 dBu or –10 dBV operating levels. Everyone (and I mean everyone) who saw the 8•Bus wanted one, and the desire was intensified if they stuck around to hear it." Electronic Musician 2/94
"Amazing. Beautiful. Sexy. I've been waiting for six years for someone to come out with a mixer like this." J.C., Charlotte, NC
"With excellent sonic quality, frequency response, harmonic distortion and crosstalk specs, number of inputs, plenty of headroom, good-quality mic preamps, and the upcoming automation package, the price of the Mackie 24x8 seems insignificant." MIX magazine 2/94
"When I read about your 'quiet' fan in your power supply manual, I almost fell over. When I didn't hear it, I fell to my knees. When I brought up fader after fader and still heard nothing, I almost blacked out! Who in the world EVER realizes that audio gear must be quiet? I love you people." D.S., Palmdale, CA
"Replaced a $20,000 console with the 24•8. Your console kicks butt over my old one. I love the EQ, the headroom and even the pans." D.C., Burbank, CA
"Used a competitor's console while waiting for your 8•Bus and will never use the other board again. Yours is quieter, has better mic pre's, better EQ, more logically laid out, much cleaner sound and better quality construction." P.P., Salt Lake City, UT
"The back of the board has 24 submaster/tape outputs incorporating a triple bus system normalling your submaster to tape ins on the multitrack. When you send a signal to submaster 1 output, for example, it appears at submaster outputs 1, 9 and 17, which simplifies operations with 8-, 16- or 24-track recorders." MIX magazine 2/94
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SHORT TAKES

VITAL EQUIPMENT

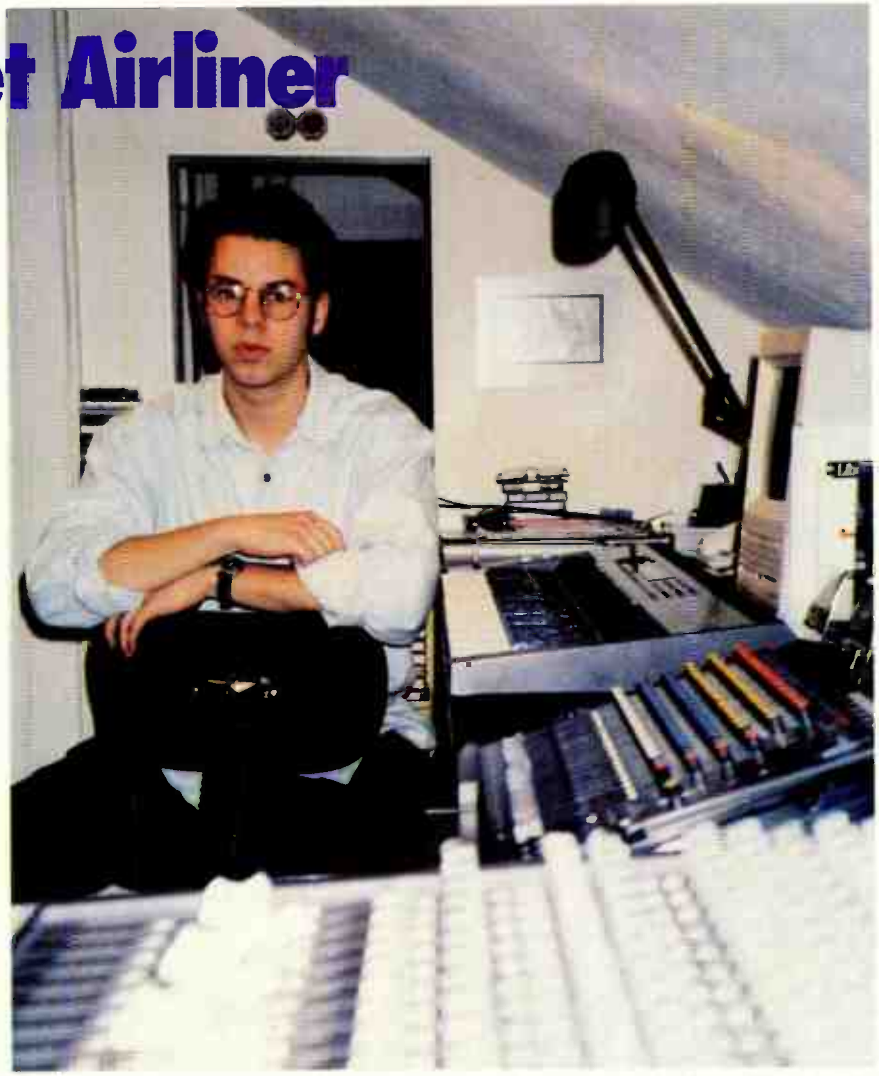
beyerdynamic has just begun offering the complete line of Vitalizer psychoacoustic equalizers in the U.S. All Vitalizers use a unique feedback loop-filtering system to enhance sound across the entire spectrum without the harshness

associated with equalizers that work by adding controlled distortion. While the Vitalizer circuitry is designed to bring highs and mids into sharp focus, at the low end, the Vitalizers employ "dynamic equalization" to accentuate the sub-bass without incurring boominess. Bass material can be processed either tight and punchy or warm. For more information, contact beyerdynamic, 56 Central Ave., Farmingdale, NY 11735. Tel: 516-293-3200. Circle EQ free lit. #115.

DIRECT CONNECT

Sprocket Digital is offering the PrismSound MR-2024T 20/24-bit interface adapter to the U.S. market. The interface adapter connects directly to a Tascam DA-88. The unit provides a choice of 16-bit 8-track, 20-bit 6-track or 24-bit 4-track recording and production formats. A simple single cable connects the MR-2024T to the DA-88 recorders, which retain full functionality. The MR-2024T accommodates both AES/EBU and SP/DIF formats. Rear panel connections include four AES/EBU format 2-channel ins and outs, a pair of SP/DIF I/Os that can be assigned to any of the data channels, four individual word sync outputs for workstations, connection to A/D converters and similar devices, and two DB-25 sockets for direct digital I/O connection to main and back-up DA-88's. For more information, contact Sprocket Digital, 211 North Victory Blvd., Burbank, CA 91502. Tel: 818-566-7700. Circle EQ free lit. #116.

No More Jet Airliner



Despite a lull in frequent flyer miles, Leo Sidran's long-distance relationship keeps right on track with his Alesis ADAT recorder and Mackie console.

How my son and I used our project studio to help long-distance collaborator Steve Miller finish his upcoming album

BY BEN SIDRAN

While others may debate the vicissitudes of multi-ADAT lockup or engage in the "I have more tracks than you" envy-game, my teenage son Leo and I have found a way to keep it simple. And powerful.

Our new project studio — based on a Mackie 8•bus console, Opcode's Vision on the Mac, and a single ADAT — has greatly simplified our collaboration with Steve Miller in his remote mountain hideout/personal studio. Writing with Steve has usually involved a lot of travel. Back in the sixties, I had to fly from England to San Francisco to work on the legendary "Space Cowboy" sessions. And even up to last year we had to get on a plane to Utah to collaborate with The Joker on his recording projects. At the time, Steve was primarily working with Leo — ultimately, Leo wound up writing and arranging four of the songs on the *Wide River* album — so I was more or less along for the ride.

And while the ride can be fun, when you're sitting by the edge of a mountain watching the river flow it can also be a drag...literally. In order to re-create the preproduction programming that Leo had done in our Mac/sound module virtual studio within Miller's tape-based facility, we had to fly in all the hardware to replicate our home setup (the keyboards, the synth modules, even the computer architecture).

This time out, we simply FedEx each tune to Steve, recorded on one to three ADAT VHS cassettes containing from 8 to 24 tracks of music. Rather than mixing virtual tracks to tape, we supply Steve with unmixed instrument tracks of independently miked or sampled drums. A second cassette takes a second pass of the sequencer, this time recording rhythm-section instruments. A third cassette might contain several takes of solos or horn and string parts. We don't even have to sacrifice one track per 8-track cassette to SMPTE because the ADATs lock every time via their own internal system.

Because Steve prefers to track his parts using two Sony 3324's (rather than six ADATs) for his 48-track digital productions, his multiple ADATs are most often used simply to transfer the tracks of his ADAT-using collaborators (namely us and the rest of his band) to the 3324's. Although recently, Steve tells me he has started locking some of his own ADATs to his two Sonys, giving him potentially up to 96 digital tracks if he wants them (a scary thought).

AFFORDABLE QUALITY

The sonic integrity afforded by my Mackie/ADAT system blurs the distinctions between "demo," "preproduction," and "tracking" studios. The submissions we send Steve certainly demo the tunes, but in some cases they can also supply tracks that will end up in the final mix. Because they supply each instrument on separate tracks, Steve has the option of leaving out instruments or parts he doesn't want, replacing parts with different instruments or singers, or keeping tracks he feels shouldn't be messed

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Ben and Leo Sidran send mixes to Steve Miller via S-VHS videocassettes and Federal Express.

with. Track modularity also affords us the ability to cut several different takes of multitrack parts like strings or vocals for Steve to compare — a track-eating operation that in the past would have forced Steve to use various multitrack "slaves" of the master to accommodate it.

Though it was the economics of rap records that originally opened the eyes of record company accounting departments to the positive aspects of project studios, there's no doubt that their A&R staffs are now aware of a more important benefit brought about by affordable digital project studios: increased creativity.

And because digital 24-and-larger


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multitracks can be afforded by only a limited number of superstars and large recording studios, affordable, high-quality gear like the Mackie consoles and ADATs has brought the capacity for convenient collaborations to, if not the masses, at least a large number of professional musicians who loathe the expense and hassle of long- or short-distance commuting.

HOMEWORK

Leo and I also completed production of the soundtrack for *Hoop Dreams*, a feature-length film that was the surprise winner at this year's Sundance Festival. Several of the film's final tracks were created, recorded, and mixed at our project studio. My basic feeling toward my room is quickly becoming, "Why not try and see if we can get it done here at home before we go to the big city and pay big bucks?"

It's also giving Leo a really advanced education in the process of writing, arranging, and producing music. He's seen firsthand that every little thing you do, especially those arising from spur-of-the-moment ideas, can count toward the creation of a great piece of music. And now, with the ADAT, it can count all the way to the bank. But seriously, the more we can keep our work playful and our play serious, the better the music is going to feel.

And thinking back on earlier, less-convenient times, such as one of our midwinter excursions to Steve's mountain retreat, we are downright excited about the benefits brought about by our new project studio. Chief among them? Not watching racks of gear sit on the Minneapolis Airport tarmac while hearing that my flight to Salt Lake City has just been canceled because of yet another surprise snowstorm. Now when the fat snowflakes pour from the sky above our home and project studio, Leo and I can look up from our 24-channel mixing console and offer silent words of thanks to the folks who keep us warm and dry: Greg Mackie, Alesis, and the good men and women who are freezing their butts driving around in Federal Express trucks.

Ben Sidran is a piano player, singer, producer, and broadcast journalist. Recent productions include Diana Ross, Mose Allison, and the Steve Miller Band's Born To Be Blue *album.*

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]

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[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

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Studio Engineering Etiquette

Mr. Manners's guide to what should and shouldn't be done by those who sit behind the console

BY JULIAN MCBROWNE

Despite the perception that working in a studio is a "cool" job, most recording sessions are hotbeds of stress. Being a first-hand observer and recorder of a musical event calls for more than just technical skill. In fact, most sessions require you to be an engineer/producer/waiter/secretary/psychologist, and more. Studio etiquette is the art of being able to handle the technical and emotional dynamics of the session. How you handle yourself can make the difference between a one-shot session and a continuing client.

Studio etiquette isn't such a hard concept if we take a good look at what's really happening at a recording session. Here are some studio etiquette tips and reminders:

WELCOME TO THE FISH BOWL

Like it or not, recording is a communal art form. Unlike writers or painters, recording artists have to create in public spaces — in front of a mic, in a glass booth — all the while being observed by a group of onlookers whose job it is to record and scrutinize their performances (a creative situation no writer or painter would tolerate). Ultimately, the tone of the session is set by the behavior of the observers.

CHECK YOURSELF

Do you wince at bad notes? Yawn between takes? Take phone calls during overdub sessions? Remember, performing artists thrive on reaction, and you are being watched. There's no escaping the power of the person who sits in the engineer's chair. Even when you're "just doing your job," everyone looks to you for a reaction to their performance, sound, or production decision. Unless you're coproducing the session, try to adopt an attitude of supportive neutrality; it'll make it easier for you to deal with the inevitable "How was that?"

BE DISCREET

Everyone's heard those "underground tapes" — they've been making the rounds for years. One features a ripping argument between the members of a late '60s British rock group, and a more recent one claims to be the monitor mix of a superstar spouse. Everyone has heard them, but nobody knows where they came from. They're amusing, but they're definitely not cool. Your clients should be able to trust that you're not peddling their rough mixes and outtakes to the local radio station.

Let your clients know that you respect their privacy. Exercise one of your ten personal digital noise gates, and turn off the mic when the producer goes into the booth for that heart-to-heart with the artist. Turning your back to the booth also signals the other control room observers to turn their attention elsewhere.

TALKBACK CONTROL

Communication between the control room and the studio can be a nightmare if the engineer has to control the talkback. If you have a remote talkback button, pass it immediately to whoever is in charge of the session and show them how to use it. This lets you out of the role of the taxi dispatcher having to anticipate when the producer will finish talking and when the artist wants to speak. On the other hand, sometimes you should take it

Illustration by Amy Hill

upon yourself to speak directly to the artist. Some artists will keep playing or singing every time they hear their cue. They're great — wind 'em up and watch 'em go. But when the producer says, "Let's listen to that," hit the talkback and say, "Listening!" When a long control room conversation results in a long silence for the artist, let the artist know that the producers are "still talking." Artists appreciate some relief from the "control room silent movie."

There's no escaping the power of the person who sits in the engineer's chair. Even when you're "just doing your job," everyone looks to you for a reaction to their performance, sound, or production.

TAKING SUGGESTIONS

Nowadays everyone's a recording engineer. The guitarist has a mic suggestion for his amp, the sax player would prefer "a large diaphragm condenser," and the vocalist feels she sounds best on a Neumann U87 (with a big foam windscreen, if you don't mind). Most of these folks can be handled with a request to "listen to it this way, and if you're not happy we can change it." More difficult are the purists who insist that you should be able to achieve a crisp, separated recording of their ensemble with three (maybe four) mics. Usually these guys show up when the house drum kit is bristling with mic stands. In this situation don't give the impression that you can only do things one way, and be prepared to change. A thorough conversation with the client well before the session should let you know how many mics to set up.

A FEW SHORT TIPS

• Don't Indulge: Even if the client has second thoughts about his performance "under the influence," he shouldn't have second thoughts about the engineering.

• Take Responsibility: Missed that punch? Miscalculated how much tape you had left? "I'm sorry, that one's on me. Can we try it again?"

• Eat Something, Drink Something: A hungry engineer is a dangerous person. Lack of food can seriously alter your mood and make it harder for you to do a good job. Hungry? Tired? Fried ears? Be honest with yourself about your physical state and let your client know when you need to take a break. EQ

Peter Grueneisen shows what *not* to sit on.

both in the back and to the sides. Armrests are nice, and most people prefer having them on their chairs. More exotic features, like flexible headrests, are also available on some models.

Nobody sits still for hours at a time while working in a recording studio. Leaning to the side or forward to adjust a fader, sitting upright, or sliding back to relax and listen to a track are all common positions. A suitable chair should allow and absorb these movements with natural flexibility. But one of the most important features of any chair is its adjustability. Audio engineers, like all people, come in different sizes and shapes, and a chair that is worth its price should be able to accommodate one and all.

Different elements that should be adjustable are the back support (both vertically and laterally), the height of the seat, and the pitch of the surface. Better chairs feature gas-filled cylinders, which can be operated very easily with one hand on a lever. Easy adjustability is very important, because nobody likes to get out a toolkit to change the height of their seat. Chairs typically can swivel around a full 360 degrees, but the ease of turning depends on the quality of the materials and design.

Regulations require at least a five-star base for office chairs, and for a good reason: Chairs with less than that tend to fall over when used aggressively, and accidents have happened with lesser models.

We have seen recording studios with those Scandinavian kneeling devices that look like accessories for a torture chamber, but they are, in fact, one of the healthiest choices for prolonged sitting, and force the user to sit in a healthy position.

For most people, a normal chair is the choice, and there is a broad range of styles and price tags available. As it happens so often in life, the more you pay, the more you get. The major manufacturers of office chairs, like Vitra, Knoll, Herman Miller, and so on, can offer the widest choice, although at a price.

As architects and studio designers, we believe that besides the all-important acoustical performance, the visual aesthetics in a studio make a difference to the well-being of its inhabitants. So we like to recommend chairs that not only meet all the requirements mentioned, but look good as well. But tastes differ and you should certainly shop around before you pick a model that suits you best. Chair finishes also come in different materials, from vinyl to different fabrics, and ultimately leather. It helps if the finish is stain resistant and can hold up to a few years of good use. Coffee spills have been known to occur in studios and the color of the fabric should be chosen to downplay the effect if it occurs.

Even before a suitable chair can be placed at the infamous sweet spot, the floor underneath it has to be made of the right material. While a carpeted surface is usually considered beneficial for a control room environment, the area behind the mix board has to fulfill different requirements than just acoustical ones. It is in essence the highway for the hot wheels attached to the chairs we are concerned with.

This is not only true behind those 18-foot long, 96-input consoles, where travel speed becomes important, but applies for any smaller setup as well. Hardwood floors (no V-grooves please) have been the standard for most studios since the '70s, but a good linoleum or vinyl floor, installed properly, will do as well. For a more industrial feel, finished concrete surfaces work fine, and for discriminating tastes there is always granite or marble. Of course, you can run your carpet right over the work area and later fix the problem with one of those plastic roll-around mats — the floor covering equivalent of pocket protectors or elbow patches.

The wheels or casters on the chair come in different materials as well, from soft to hard plastics and even metal. It is worth finding out which sort works best with your floor, and in the case of some floors, which ones cause the least damage over time.

All in all, don't ignore the sitting needs of your body. Pick the right chair and your work will be more relaxed and concentrated, not to mention a lot healthier, too. Who knows, we might even be able to hear the difference on your next record.

Award-winning A.I.A. architect Peter Grueneisen is partners with George Newburn and Peter Maurer in the international firm studio bau:ton. They have designed for major facilities such as Record Plant/LA and Bad Animals/Seattle, as well as project studios for artists such as Baby Face, L.A. Reid, Bill Botrell, and Ice T.

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WOMEN

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IN VARIOUS FACETS OF THE RECORDING INDUSTRY ABOUT THE
WOMEN WHO WAN

BY VANES

IN AUDIO

RDER IF YOU'RE A WOMAN. WE ASKED FOUR PROMINENT WOMEN CAREER EXPERIENCES AND WHAT ADVICE THEY WOULD GIVE TO O MAKE IT IN AUDIO.

Photo by Julian Jaime

GAIL "SKY" KING

Born and raised in Queens, New York, Gail "Sky" King got into production and writing through playing in bands and, believe it or not, rollerskating. Last year she got a TV gig working for PBS's Children's Television Workshop on its show Ghostwriter, *which led to her working on* Sesame Street *beginning with the fall '93 season. She continues to produce and write for several artists.*

I got started on guitar when I was twelve; it was at an experimental camp that brought 50 black and Latino New York City kids together with 50 white rural Vermonters. Sandy Paton, still my friend to this day, taught music there, recognized my enthusiasm, and stuck a guitar in my hands.

I worked my way through Emerson College [Boston, MA] playing in funk and fusion bands while earning my degree in communications. I DJ'd on the college FM radio station and, as an African-American woman guitarist in some pretty progressive groups, I managed to swing interviews with a lot of artists for my radio show. I even jammed with quite a few, including Kool and the Gang, Carlos Santana, and Tower of Power.

I eventually got tired of the bar-band scene, so after I graduated, I returned to New York and discovered rollerskating. That led to a whole new career as a DJ. I played in all the skating rinks and also performed with a troupe called The Wizards. I put our music together and for five years we skated professionally in the U.S., Europe, South America, and Canada.

DJ'ing got me interested in the studio. Around 1983, a woman coworker told me that producer Arthur Baker was looking for some people just to run errands or whatever in his new studio, and I jumped at the chance. Arthur was willing to hire women at a time when few studios would, and particularly not someone as inexperienced as I was, though I had a lot of drive and enthusiasm. After a couple of years I became an engineering assistant, but I realized I'd rather produce than become a full-fledged engineer.

At that point, I started editing, which in turn led to remixing. I eventually began doing production using what I'd learned from doing radical remixes where I'd keep only the original vocals and create all new tracks underneath.

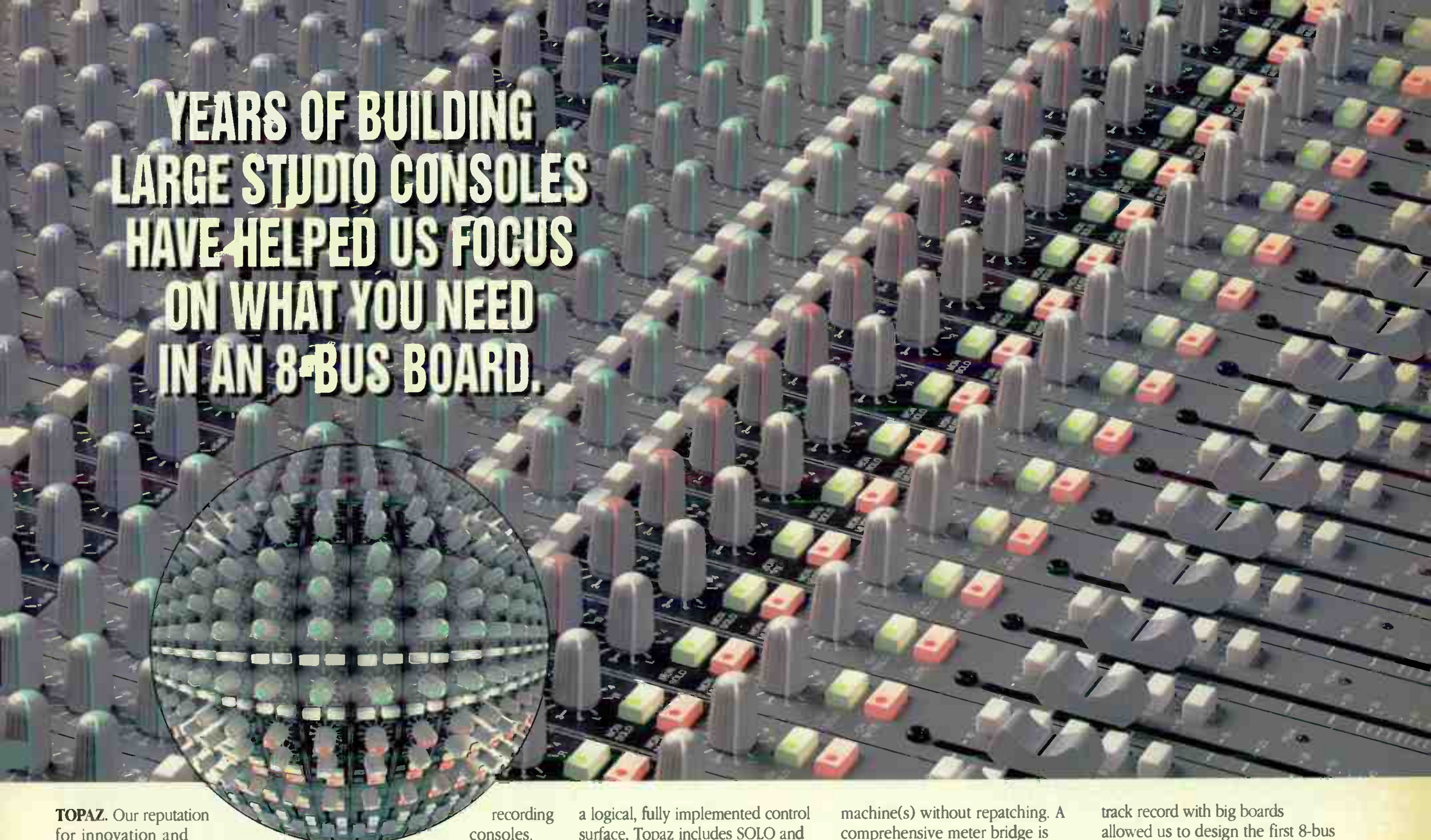
Around that time I got married and had a daughter, who's now four. Working while pregnant was often amusing. I remember how the busses on the SSL got progressively farther away. Everybody was afraid I'd have the baby in the studio. I turned in my last project the day before I went into labor, took three months off, and then brought my daughter to the studio with me. My schedule was crazy: I was in California for six weeks, in London for a couple of weeks, and she came along 'cause I wasn't about to leave her home! I'd fly somebody out who could watch her while I worked. It was tough juggling being a producer mom.

So, to accommodate having a kid, I built a home studio. Now I can work and be home more often. Having a personal studio has allowed me to widen my focus to include composing for TV, as well as doing records. Though TV work is seasonal, it's generally long-term and enables me to explore a wider variety of music. Doing records is great, too, but you've always got to hustle up the next gig. Diversity is the key, so I try to wear as many hats as possible.

Being a woman has both hindered and helped my career. I've had some negative experiences. Once I discovered that a male DJ was paid more than I was even though I spun the more successful weekends. When I confronted the owner he said simply that "a man needs more money." He couldn't see the inequity in this and I ultimately split. I also had an engineer tell me that women have no place in the studio. He said he would do everything in his power to make my life miserable — that nothing would make him happier than to see me cry, then quit. So, of course, I did neither. I tried to avoid his sessions, but I was the newcomer and the senior assistants weren't particularly fond of him, either. He finally stepped way over the line and we had a blowout, but I earned his respect. We actually kind of became friends, years later.

As a guitarist, being a woman helped because I was a novelty, an attraction. If you can battle your way into a position where you're doing what you want, being a rarity can work to your advantage. Getting there is the hard part.

When I became a mom, I ran into a fair amount of flak, as if motherhood were somehow unprofessional. At Electric Lady Studios, however, they babyproofed to accommodate my daughter: put up gates, buried the wires, hooked up the lounge. It was great. When I had to breast feed her, my engineer and assistants (all men) weren't quite comfortable, so I sat on the other side of the console where they could hear but not see me, and I'd keep right on directing the session. I think some guys had difficulty seeing me play the roles of both mother and producer. There *are* obstacles, so women have to excel to gain equal standing with men. There's also a certain amount of racism in the industry,



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though in my experience it's usually less of an issue than sexism.

The paucity of role models also impedes our progress. It's a boy's club. And some men just don't feel at ease with women in the studio. Women have to learn to deal with that, to still get the respect they deserve and do the job well without getting stepped on. Certain personalities are going to fare better than others, and if you're easily intimidated you might get pushed out.

Standing up for yourself can be a problem, too, because there's a double standard: when it's a man, he's a go-getter; when it's a woman she's a bitch. Keeping your balance can be tricky. Sometimes you gotta put up and shut up and try to make the situation work for you. Sometimes you're up against people who just aren't gonna budge, in which case, maybe the place you're at is not the place to be, though opportunities are so few and far between that you really can't afford to chuck 'em. So if you get a chance to do this, there can't be any half-stepping. Don't go out there and be all fluff and no substance.

I'm not here to be a proponent for women in audio, but I know many extremely competent professional women who are everything from studio managers and engineers to producers and musicians, yet the stereotypes still reign. I can rock the computer on an SSL as well as any man. People need to get over it and get on with it. I'm not a man, but I can do the job and be a mom, too — and a good one!

Leslie Ann Jones

Leslie Ann Jones has engineered for 15 years, working with Bobby McFerrin, Herbie Hancock, BeBe and CeCe Winans, Kim Carnes, Holly Near, and Joan Baez, as well as doing film and TV scoring. She has mixed for the Grammy Awards for three years and is currently on staff at Capitol Records in Hollywood.

About 15 years ago I began doing live sound for bands to make extra money, and found that I had a talent for mixing. This led to starting a PA company while I worked for ABC Records in publicity. I never really intended to be an engineer; I wanted to produce and thought it best to learn some engineering. I took some classes from Bill Lazarus, then asked to be transferred to ABC studios.

Studio head Phil Kaye was very honest and said he didn't know how the clients would react. ABC had a woman mastering engineer, but unlike a recording engineer, a mastering engineer doesn't become a part of your life for the next six weeks. Phil said we'd try it. I recall only one negative incident where someone's wife had me taken off a project because she didn't want any other women in the room with her husband.

I never made a big deal about being the only woman in the room; besides, I have a good sense of humor and that makes people more comfortable. When I left L.A. for San Francisco, however, the training process started all over with a whole new set of clients and engineers who had to be convinced I actually knew what I was doing. When I started at Capitol they had already had experience with a woman staff engineer which made that transition easier.

Some people want to work with me *because* I am a woman, and I don't mean just female clients. There's a certain way I hear things, and they like that. The bottom line is that we're not the same as guys.

Perhaps because of ego and competition, it's difficult for an all-male staff to suddenly work with a woman who's not in a traditionally female role. Men are used to women acting as caretakers, notetakers, studio managers, assistants, etc. — not as the ones making the technical decisions. That perception may be why women have such trouble making the leap from second to first engineer. When you're the engineer, nobody cares how detailed your track sheet is; they want to know if you can make the snare sound the way it should.

One of my main mentors at ABC was Roy Halee, and another was David Rubinson, who owned the Automatt. On a fluke, I sent a resume there because Fred Catero was the chief engineer and I really admired his work. Three days later they called me, and I was there for six years. David was one of those people who was forthcoming about anything having to do with the business; I learned an immense amount from him.

My advice to aspiring women in audio would be not to have a hot date on a Friday night. Kidding aside, you should know that engineering will take over your whole life. Long hours don't go away just because you attain a certain level of success, and I think that's difficult. Particularly for women. My career has always come first. Interestingly, the guys I work with are married, yet most of the women engineers I know are not. Also, don't be afraid to ask questions. If you have the attitude that you're always supposed to know everything, you're never going to learn. People will help you if you're interested in finding out how things work.

Women should do more outreach at recording schools, do more guest lectures, and write articles. Letting people know there are successful women in a certain field furthers the choices women have. I would have felt much less alone if a guest lecturer had talked at school about her experiences, like the women in audio seminars at past AES shows. I think it's also helpful to combine women and men in this work because the experiences aren't the same. That keeps it from straying to "how the guys have done us wrong"; I don't think that's really the point anymore.

SUSAN ROGERS

Susan Rogers started in 1978 as a receptionist at the University of Sound Arts recording school in Hollywood. She then became a field service technician for MCI, followed by staff maintenance tech for Crosby, Stills, and Nash's studio. In 1983 she was hired sight unseen as a maintenance tech/engineer for Prince (or whatever he's calling himself these days), and worked with him for five years before moving back to L.A. where she has worked with Wendy and Lisa; Michael Penn; Tevin Campbell; Edie Brickell and the New Bohemians; Public Image; and others.

I started during a recession, and heard if you get into maintenance, you'll always have a job. Besides, being a woman in this business I wanted to be technically adept. I called my local U.S. Army recruiting station, told them I wanted to join when I got out of high school, and for $1.75 in postage they sent me the complete army electronics training, from DC principles to microwave technology. So I started studying electronics.

Prince had asked for a technician, but he also needed an engineer. He assumed if you could do one, you could do the other. I wasn't about to tell him differently, since I wanted to engineer. I had never really done any engineering; I was just thrown into it, but Prince had his own methods of working, anyway. He was my engineering mentor, because I didn't come up through the traditional studio system.

I definitely had role models. When I first wanted to engineer I knew of only two women doing it: Leslie Ann Jones and Peggy McCreary. I idolized them. It's important for women to have female role models, because I believe you have to have a competitive drive and it's easier to compete with people who are your peers. In retrospect, I did not compare myself to the great male engineers. I admired their technique, but I didn't have that ambition to outdo them as an engineer.

I think being a woman initially helped me get jobs, most importantly with Prince because he liked working with women. As far as hindering things, I wish I knew. I think it's safe to say the industry somewhat discourages women in audio, not just as engineers and producers, but also as A&R reps, managers, and record company executives. I now have a 15-year body of work so it's no longer as much of an issue for me.

It's vitally important to have a mentor. Very often the way engineers move up is through the buddy system: an older engineer takes you on as an assistant, and helps you learn the ropes. For women, this can be a problem. Women are not moving up in the numbers you would expect, possibly from a reluctance to hand the chair over to a woman. This needs to be examined by men and women.

If I have any advice for women it is to think seriously about what you're gaining and what you're giving up to pursue this career. For a man, it's difficult to spend a lot of time away from his children while in the studio constantly. For a woman, it's a near impossibility. Also, success is when preparation meets opportunity; you must have a wealth of knowledge behind you. Read the manual for every piece of gear you use. Educate yourself so others will have confidence in you, and you will have confidence in yourself. For example, when I was 22 I was sent to repair a tape machine, and the chief technician called my boss to say that I was not acceptable. My boss was very supportive and said, "When the work is completed, give me a call if you have any complaints." I fixed it in about an hour and the tech was satisfied. It really impressed upon me that one's work will be judged by its quality, and that you can change ingrained attitudes if you do your job well.

If a woman's work is great, she'll be helping herself and the women who come after her. If it's not great, it will be more difficult for the next woman (which is not the same as for men, who are more apt to be judged as individuals).

Aside from being technically prepared, women should maintain a zero tolerance level for sexual harassment no matter where it comes from. You must say something when the first incident happens. If it persists, you must either make it very clear to the perpetrator that it is illegal and that you will have to see a lawyer, or change your job. I don't think harassment is an epidemic, but it does happen. What a man can do is help a woman move up through the buddy system — be a mentor and provide her with the criticism she needs to improve her skills and move up. And when he's ready to move on, let *her* sit in that chair. I think getting women into the industry is up to both sexes.

LAURIE SPIEGEL

Composer Laurie Spiegel wrote interactive music and image software at Bell Labs from 1973 to 1979 on the alphaSyntauri, McLeyvier, and other "vintage synths." In 1985 she wrote the popular program Music Mouse (available for Mac, Atari, and Amiga computers). She has consulted for IBM, Eventide, and others, and is currently a freelance composer, consultant, and programmer in New York.

My father taught me to solder when I was about nine, so engineering felt natural. By age seven I knew I didn't want to become just a housewife like my mother, who was very bright but had no identity back then besides family. I wasn't inspired by a role model, but by avoiding one. Growing up, I felt neuter, outside gender roles. If I had role models, they were men like Bach, DaVinci, and Shakespeare. But I was always more motivated by enjoying the process of doing than by images of others.

I got into writing computer software accidentally as a tangent to being a composer. My B.A. degree was in the social sciences, but I loved music best and decided I'd forever regret not giving myself a year to really try it. By the end of that year I was composing, performing, teaching electronic music, doing soundtracks, and earning a living at electronic music. So all those people who told me I couldn't do it because I was a girl and starting late were wrong — though if one less person had encouraged me and one more *dis*couraged me, I might have given up.

By 1973 I realized I needed computer control to overcome my frustrations with analog gear. I apprenticed myself to Emmanuel Ghent and Max Mathews at Bell Labs, and started by helping to debug FORTRAN programs. I never formally studied engineering or programming; I just learned by doing. While I was at Bell Labs I still taught, composed soundtracks, concertized, coded, and did video. I worked hard.

Being a woman has both helped and hindered my career. When I was younger I had many discouraging experiences, such as sexual advances from film directors. I wouldn't tolerate that now, but when you're younger you're at a disadvantage. You're willing to do almost anything to complete a job because it's so hard to build any track record.

A lot of men assumed that if you were working for them they could get as intimate as they wanted. I'm amazed that men find it hard to believe how common and pervasive sexual harassment is. On one of my very first jobs, my boss assumed that sex with him was part of it. I ultimately quit and wound up not getting paid. It's really hard to say no when you're a subordinate, so I've stayed mostly freelance — a client won't make the same assumptions about your dependency that a boss will. I'm a strong believer in multisourcing my support system, too.

On the tech side, being a woman helped because I was an oddity, which attracted interest. Overall, though, I think if I were a man and had done the same work, by now I'd be better known, have more money, and be considered a much more important figure in my field's evolution. There are lots of subtle "techie-macho" things that undermine your confidence. A lot of men engineers try to show me how to do things that I already know how to do, treating me as though they have to watch over me to make sure I do them right. Or there's the covert insult of being praised for things men would be expected to do as a matter of course, implying that I wouldn't be expected to be able to do them.

Today there's a lot less sexism, but many men still fear women having power and control. By now, women are in a lot of the lower, non-decision-making jobs in the industry, but that's like there being a million actresses yet very few women directors or producers. And we have to remember that women are socialized differently. I've been called aggressive for participating on the same level as men in a team situation, because a woman isn't supposed to be as forward. "Aggressive" is a positive term when applied to men and negative when applied to women.

Ultimately your individuality is the most important thing. People will usually prefer working with somebody who knows what they're doing and is an enjoyable person over someone who doesn't know what they're doing and is to relate to — regardless of gender. So it's most important to get known, as an individual, for your strengths. EQ

ON THE TECH SIDE, BEING A WOMAN HELPED BECAUSE I WAS AN ODDITY, WHICH ATTRACTED INTEREST

WOMEN IN AUDIO: THE TECHNET SURVEY

Last spring, Technet and *EQ* magazine distributed two surveys to research the evolving roles of women in audio. The first survey targeted companies for statistics on the levels of participation and success achieved by women in the audio industry. The second targeted women working in the audio industry for information on individual perceptions.

The first survey was distributed primarily through SPARS. Eighty percent of the respondents came from an equal number of studio businesses and manufacturing and design firms. Seventy-five percent of these companies had fewer than 25 employees.

Employers reported that women work predominantly in administrative and support positions. In fact, over 33 percent of the employers stated that over 80 percent of these low-paying jobs are "manned" by women. For most firms, less than 20 percent of all technical positions are held by women; with less than 2 percent being "First" or "Lead" positions. Approximately 50 percent of the firms state that women are employed in less than 20 percent of all management and marketing positions. Overall, the larger companies reported slightly higher percentages of women employed in managerial, marketing, and technical capacities.

The employers were asked to evaluate a list of suggested changes that might lead to increasing the participation and success of women in audio. Positive changes in college recruiting practices, hiring practices, music-marketing techniques, and secondary education biases were overwhelmingly rated by all respondents as being "very important."

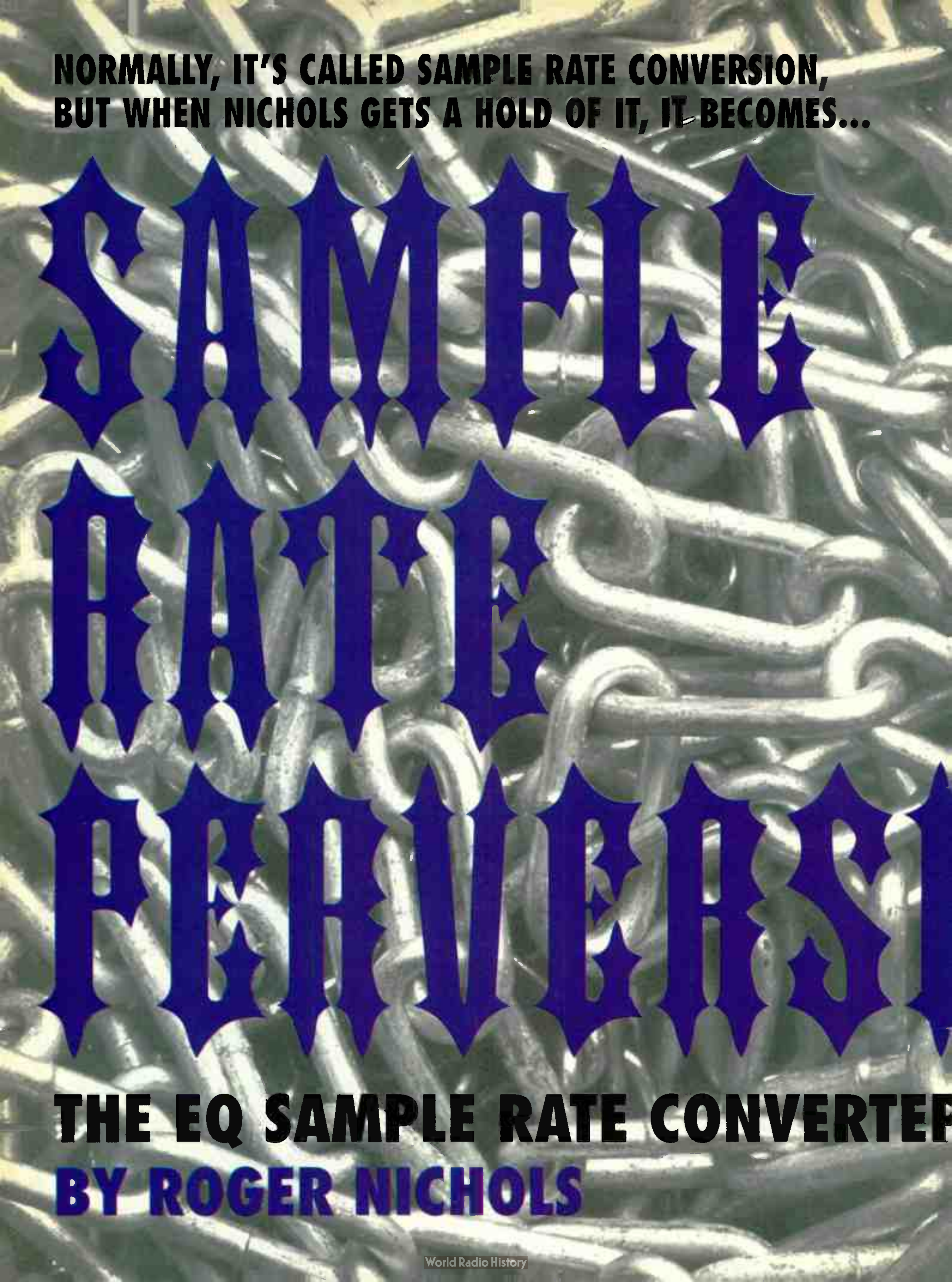
The second survey was distributed primarily through Technet. Almost half the women who responded are currently working in studio production. Marketing/publicity/journalism professionals accounted for 20 percent of the respondents, as did wholesale/retail sales professionals. Approximately 50 percent are self-employed.

Individuals in their 20s reported little exposure to gender discrimination and sexual harassment. Almost 33 percent of respondents in their early 30s had experienced sexual harassment and most knew of others within their company who also had. Approximately 66 percent of those over the age of 45 reported that their salaries were at least 10 percent lower than those of their male counterparts. Overall, 75 percent of the women claimed that gender discrimination has had a negative impact on their individual ability to enter the audio industry and succeed in it. Two percent reported a positive impact.

Asked to evaluate a list of several factors that contribute to the scarcity of successful women in audio, responses from women in the under-30 group emphasized the lack of prominent role models and the lack of encouragement by primary and secondary educators. Responses from the over-30 group emphasized the existence of a male-dominated music industry. All individuals regarded widespread gender discrimination as being a lesser, or secondary, force determining the overall number of successful women in audio.

Fifty percent of all respondents, regardless of age, believed that their careers would be jeopardized if they reported sexual harassment to their superiors. Meanwhile, 54 percent of those responding to the first survey stated that their firm had no written policies for dealing with gender-related issues such as job discrimination, salary discrimination and sexual harassment.

—Sally Dorgan Potts



NORMALLY, IT'S CALLED SAMPLE RATE CONVERSION, BUT WHEN NICHOLS GETS A HOLD OF IT, IT BECOMES...

SAMPLE RATE PERVERSI

THE EQ SAMPLE RATE CONVERTER

BY ROGER NICHOLS

Roger Nichols's Den of Iniquity. Photo by Roger Nichols

ON

ROUNDUP

WELL I thought that was what they wanted me to write about, but after *EQ* got the invoice for 750 feet of chains, I received a call admonishing me for not cleaning the wax out of my ears.

It has no doubt come up that something you recorded on DAT was at the wrong sample rate. If you mixed your record to a consumer DAT machine through the analog inputs, then it is probably at 48 kHz, but it has to end up at 44.1 kHz to produce CDs. If you mixed to a professional DAT machine that offers you a choice and you chose 44.1 kHz, things may look fine — but what if you want to transfer the final mix to ADAT and add another part or two? You can't win.

Sooner or later, you are going to have to use a sample rate converter. Now the question is, which one? I gathered up a few to review, including the Alesis AI-1, NVision NV4448, Roland SRC-2, the Z Systems Z-1, and the now six-year-old Sony DFX-2400. As a comparison, I used a set of Apogee converters to convert sample rates by transferring *analog*.

The type of sample rate converter that I will discuss in this article are fixed-rate converters. A fixed-rate converter is one that can only convert between exactly 44.1 kHz, 48 kHz, and sometimes 32 kHz. The relationship between the sample rates remains fixed. You cannot VSO the input to the converter, and you may not be able to convert to 44.056 kHz or 47.952 kHz.

The first sample rate converter made by Studer was such a fixed-rate converter. Also the converter originally used by Harmonia Mundi was also fixed (their latest version uses the AD1890, like just about everybody else). During the early days of digital, Mitsubishi X-80 digital 2-track machines used a 50.4 kHz sample rate and 3M used 50 kHz. The Studer also allowed these sample rates, but at a price in the neighborhood of $50,000. I don't know anyone who moved into that neighborhood, but I do know a few people who visited there regularly.

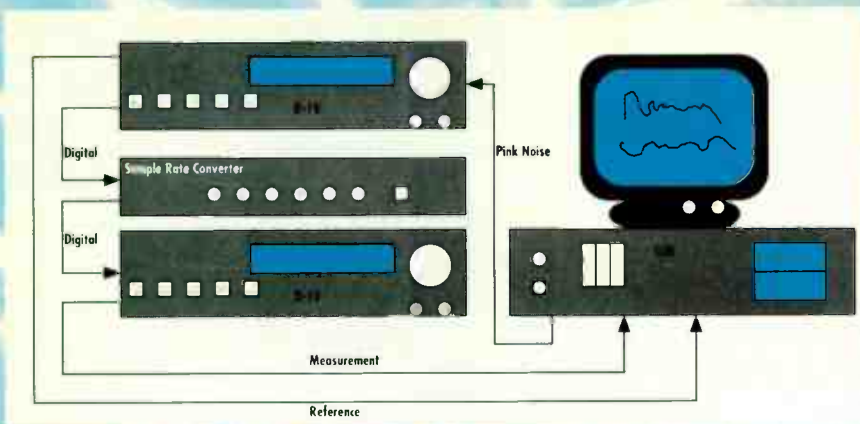


Figure 1: Test Setup
Measurement and Reference are analog from the D/A section of each D-10.
Pink Noise is the analog signal from SIM to the A/D converter of the reference D-10.

SAMPLE RATE CONVERSION

Digital sample rate conversion is a complex mathematical process. The process is one of interpolation and decimation. If these mathematical processes are not done well enough, it could be called "decimation of character."

If we wanted to double the sample rate, let's say from 25 kHz up to 50 kHz, we would need twice as many samples in the output. We would have to manufacture a new sample to go halfway between each of the samples we already have. Let's say that our sample resolution was 100 (as opposed to 65,536 levels in 16-bit sampling). Now let's assume that we have one sample that is at 100 and that the next sample is at 50. The newly manufactured sample would be at 75, halfway between the two values. If we did this for every space between samples, we would have a new signal at twice the sample rate of the old one. For sample rate changes that aren't as easy to calculate, such as 44.056 kHz up to 48.084 kHz, the math gets a little bit harder, but now you get the idea.

Analog Devices is a company that makes D/A and A/D converters. They have produced a sample rate converter on a chip called the AD1890. This chip does everything necessary to convert from any sample rate to any other sample rate. A lot of companies are using the chip as the heart of their converter systems. What makes for differences between converters is what additional features are implemented along with the chip. Some higher priced converter boxes include additional DSP processors to allow dithering and noise shaping after sample rate conversion. No longer will sample rate converters cost the same as cars.

The first sample rate converters I ever used were the fixed-rate Harmonia Mundi and the Sony DFX-2400. It was nice being able to make transfers to foreign formats without having to worry about the analog chain of events. Sample rate conversion has come a long way, and what follows are some samples of what you can get your hands on.

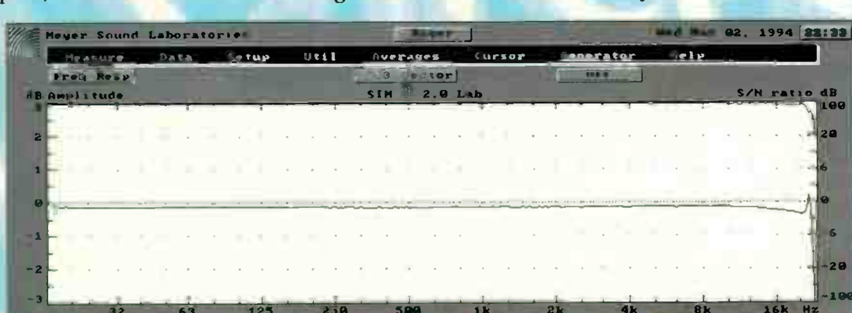
THE TEST SETUP

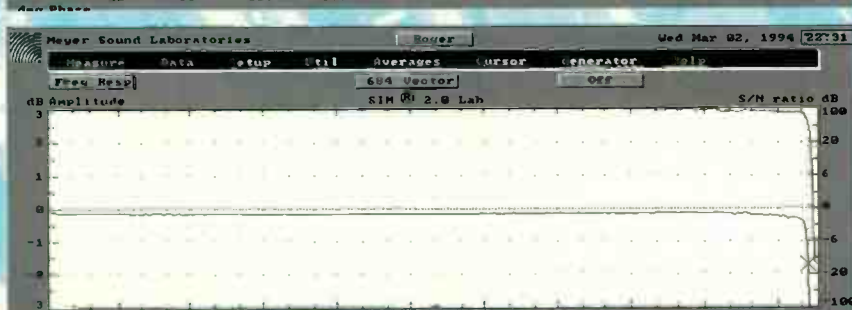
The digital copies were made between two Fostex D-10 DAT machines. The test machine was the Meyer "Source Independent Measurement" system, or SIM machine. The SIM machine performs real-time FFT analysis of two signals and displays frequency response, level, phase, and signal-to-noise ratio. The SIM machine can use music as well as pink noise as a test source. We used both to make sure there were no differences.

To test the converters, we needed to compare the analog signal before and after the conversion. We did not want the A/D and D/A converters in the DAT machines to influence the test. To nullify any conversion anomalies, we compared the output of the D/A section of the source D-10 with the output of the D/A section of the destination D-10. You can see in the graph for the flat D-10 transfer that the control setup was flat as a pancake (unless you eat at IHOP, in which case it is as straight as a ruler).

The sample rate converters were inserted between the two D-10's. The source sample rate was selected by the source machine. In the single case of VSOing the source, a Fostex D-20B DAT machine fed the signal digitally to the first D-10. The digital input of the D-10 followed the VSO output of the D-20B. This way we could vary the input sample rate while still comparing the outputs of the two D-10's.

The standard tests performed were: 44.1k to 44.1k; 44.1k to 48k; 48k to 48k; and 48k to 44.1k. Some units did not have a hard bypass and were always in the circuit. Some performed as if they were bypassed when the input and output sample rates were locked together. Units that allowed





Left, top to bottom: Apogee 44 to 48; Apogee 48 to 44. Above: The Apogee UV22 super CD encoder.

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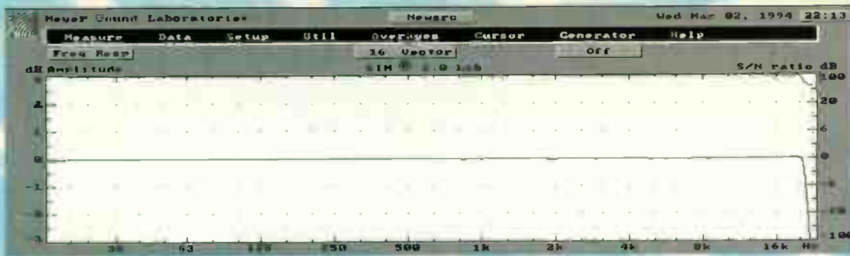

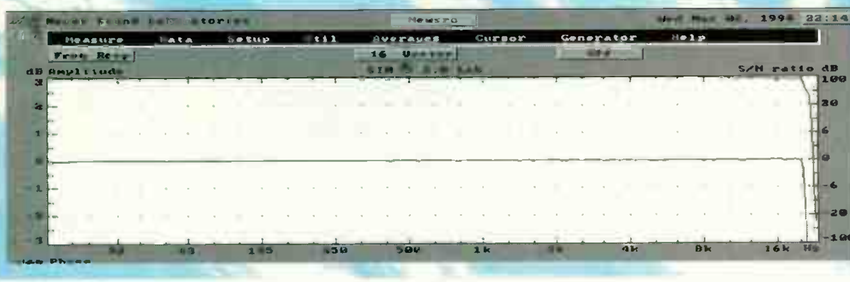

Left, top to bottom: Z-Systems 44 to 48; Z-Systems 48 to 44. Above: Z-Sys Z-1.

VSO input or external synced output were given additional tests. Because of the number of graphs required to show all the results, I have only included one graph where results were identical.

Harmonic distortion tests were run on all units. When audio levels were up around zero, THD was not degraded at all by any of the units tested unless otherwise noted for an individual machine.

SONY DFX 2400

I wanted to start here because I have used this converter on and off since 1988. At about $12,000, it was the cheapest sample rate converter around at the time. The input and output could be just about any frequency combination you could throw at it. The format choices were AES/EBU or SDIF-2. The output frequency was determined by feeding an external reference to the unit. The clock could be word clock or another AES signal. There was no internal reference, so the external clock was mandatory.

As you can see by the graphs, the Sony is about par with the group, except for a decrease in S/N ratio below 33 Hz coupled with a slight level rolloff and phase shift starting at the same frequency. This may have been an attempt at DC removal, but the rolloff is not quite enough. The high-frequency cutoff and S/N ratio stayed pretty much the same, no matter what the input/output frequency combinations were.

If the winner of the converter contest were to be picked on the basis of size and weight, the Sony would win hands down. It was the only unit that kept the big spring on my back door from pulling itself shut.

Digital sample rate conversion is a complex mathematical process. If these mathematical processes are not done well enough, it could be called "decimation of character."

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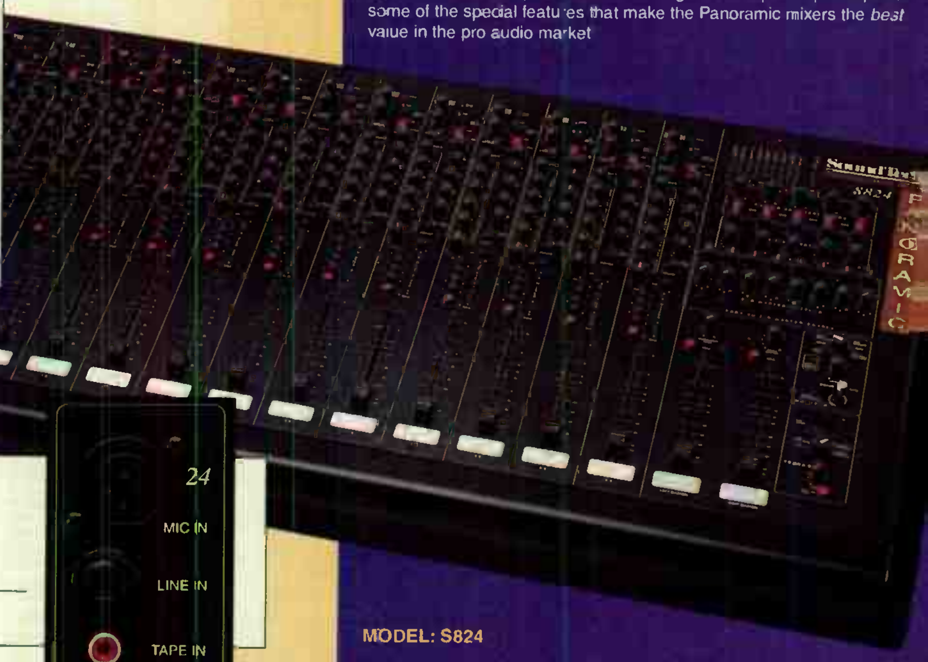

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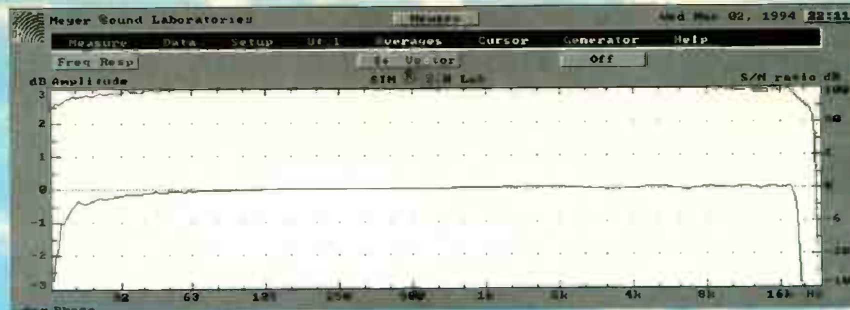
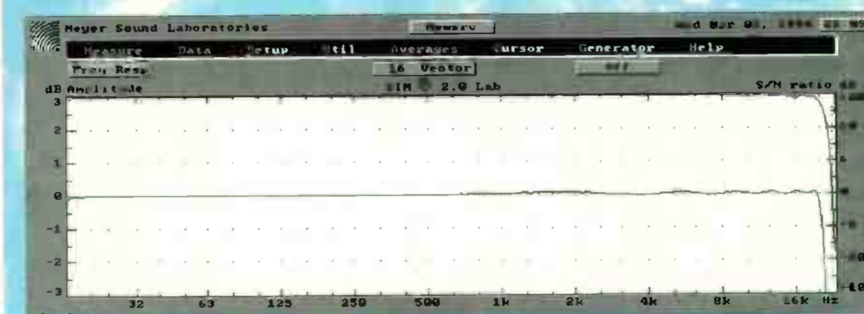




Above, top to bottom: Roland SRC 44 to 44 DC CUT; SRC 44 to 48; SRC 48 to 44. Right: Roland SRC-2.

Pros: Sony was there when needed, with an inexpensive (comparatively, at the time) and reasonably good converter; I would not be ashamed of running my mixes through it if I needed a good conversion; has a nice built-in briefcase handle.

Cons: Weighs more than my dog.; expensive by today's converter pricing; no internal clock reference.

ALESIS AI-1

The Alesis AI-1 sample rate converter was designed as an interface between the ADAT system and the external digital world. Since ADATs run at 48 kHz (without VSO), the AI-1 included sample rate conversion so that you could move data digitally between the ADAT and DAT or hard-disk editors, where you wanted to be at 44.1 kHz.

Examples of applications:

• You mixed your project to two open ADAT tracks and want to transfer it to 44.1 DAT for mastering.

• Your entire sample library is at 44.1, but you just recorded some killer samples on an ADAT at a friend's studio.

• You copied a sample loop from CD onto your hard disk, edited it into a song, and now want to put it on ADAT to start overdubbing.

Without the BRC in the system, the AI-1 will let you select one of four track pairs as the ADAT source or destination. That means you could choose tracks 1&2, 3&4, 5&6, or 7&8 to be transferred. With the BRC doing the routing, any two of the ADAT tracks from any of the transports in the system could be transferred.

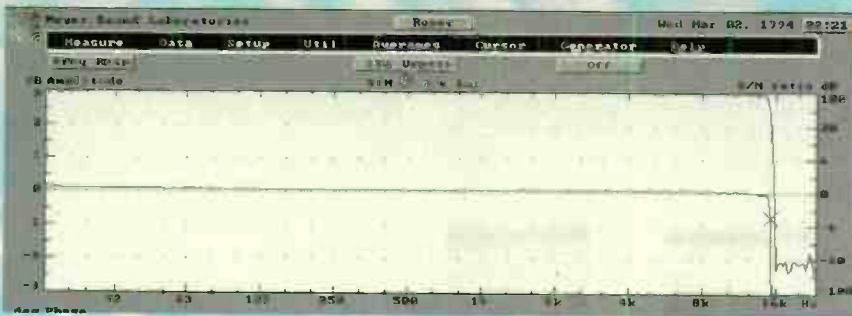
The AI-1 will also work independently of the ADAT system. The source and destination can be DAT machines, or other pro or consumer digital devices. I used it to transfer some old 48 kHz DATs to 44.1 kHz.

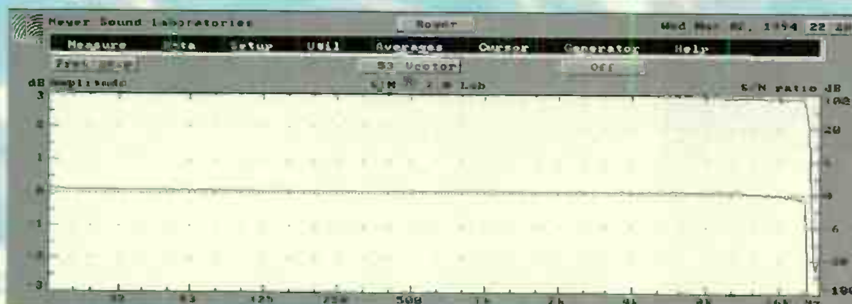
Pros: Handles variable input rates well; is the only one to support ADAT; has sync output (48 kHz only); sounds good.

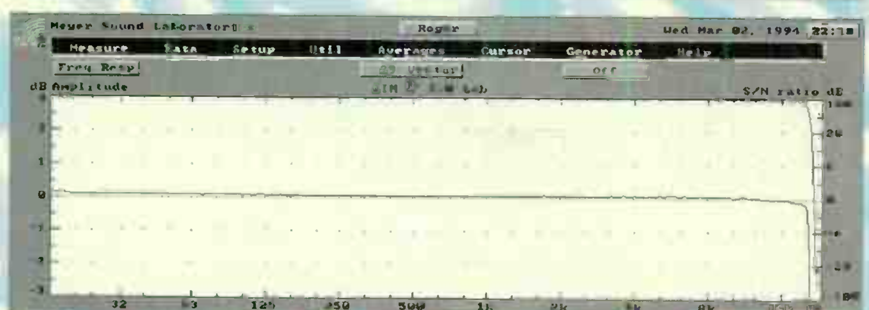
Cons: Can only choose ADAT tracks in pairs next to each other without the BRC; no hard bypass.

NVISION NV4448

The NVision stand-alone converter is used at many mastering facilities as







Above, top to bottom:
NVision 44 to 32; NVision 44 to 48;
NVision 48 to 44. Left: NVision NV4448.

their "meat and potatoes" conversion box. I borrowed the one I tested from Masterfonics in Nashville. The unit performed flawlessly and measured well. Except for the Sony, the NVision is the only unit to support SDIF-2. Under normal circumstances you won't have much need for SDIF-2 unless you are going to transfer your DAT to a Sony 1630 or Sony reel-to-reel 3402 digital 2- track machine.

Pros: SDIF-2; external sync; plenty of sample rate choices; sounds good; just plug it in and go.

Cons: No hard bypass.

ROLAND SRC-2

The Roland is a unique box. It is not only a sample rate converter, but a dual-stereo input mixer, digital level adjuster, a DC remover, a flag-setting box, and an emphasis remover.

The SRC-2 has two digital inputs that can be different sample rates and different formats (pro or consumer). The level of each input can be adjusted independently. If either input contains emphasis, it is digitally removed — not just flag removal, but deemphasis EQ. Any necessary sample rate conversion is performed and the two signals are added together. The output level can be adjusted, as well as the balance between the two inputs. Finally, you can set copy protection flags, select either pro or consumer format output, and choose whether or not emphasis should be put back.

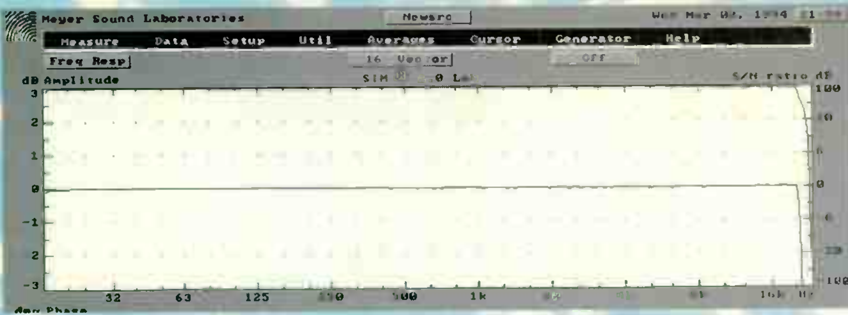
The SRC-2 has level indication for input A, input B, and output. It also has indicators for emphasis and the sample rate of each input signal. The output sample rate can be internal 32k, 44.1k, 44.056k, 48k, 48.048k, 47.952k, referenced to input A, external word clock, or video sync.

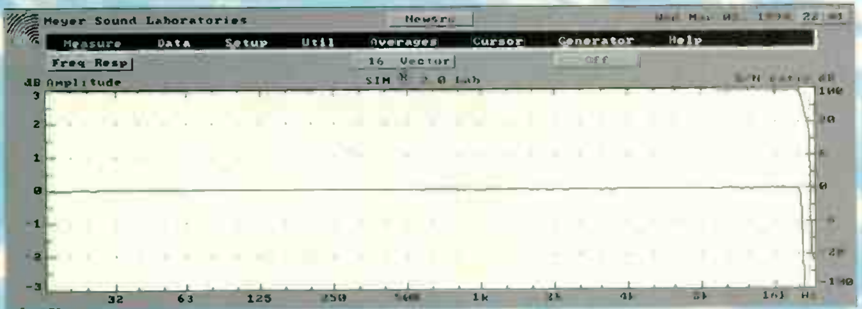
Pros: Level control; mixes two digital inputs; emphasis removal; DC removal; video sync; hard bypass.

Cons: Cutoff frequencies are the lowest of the bunch; I would have to buy 23 more of these to have a 48-input digital console.

Z-SYSTEMS Z-1

The Z-Systems unit is a nice box. The output reference can be either internal 44.1k, 48k, 32k, or external.





Left, top to bottom: Aleses AI-1 44 to 48; AI 48 to 44. Above: Alesis AI-1.

If you remember from a couple of issues back [March '94], I solved an impossible task with the Z-1. The input to the converter was VSO'd one way and the sync driving the output sample rate was VSO'd the other. The Z-1 didn't care, it just did the job. I also really liked the idea of being able to switch among the digital inputs.

The Z-1 has AES, Coax, and optical inputs and outputs; it can synchronize with either another AES signal or word clock. The input capture range is from 25 kHz to 50 kHz. The capture window can be set to "narrow" if the input signal is not changing frequencies. This will improve clock jitter specifications and improve harmonic distortion and noise figures by about 1 dB.

Z-Systems is also coming out with a lower priced unit (under $1000) that is a fixed-rate converter for sample rates of 32 kHz, 44.1 kHz, and 48 kHz. It will not track variable speed inputs and will have an output resolution of 16 bits.

Pros: An excellent machine; switchable inputs; external sync; plenty of sample rate choices; sounds good; hard bypass.

Cons: None.

APOGEE A/D-D/A

Digital sample rate conversion is not the only way to get from one sample rate to the other. Before there were sample rate converters, engineers would connect audio cables from the analog output of the source machine to the analog input of the destination machine. The play machine would play at the source sample rate and the record machine would be set to record at the destination sample rate. As if by magic, the conversion was performed.

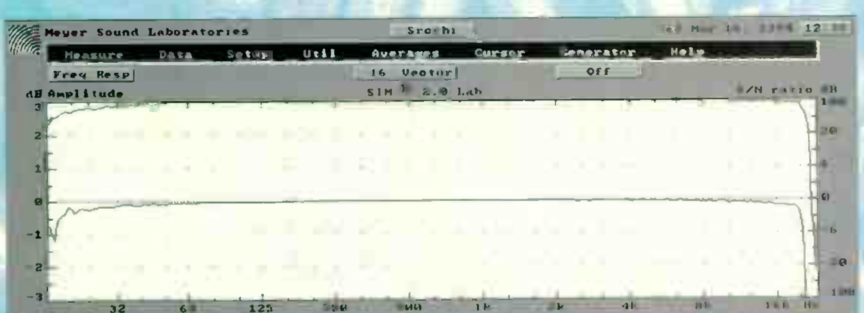
In the early days of digital, the analog-to-digital-and-back-again conversion process left a lot to be desired. The additional distortion and digital noise added to the signal during analog transfer bordered on the unacceptable. But it was sometimes the only game in town. If you mixed your record to a Sony F-1 at 44.056 kHz, for instance, you were up @$%&'s Creek.

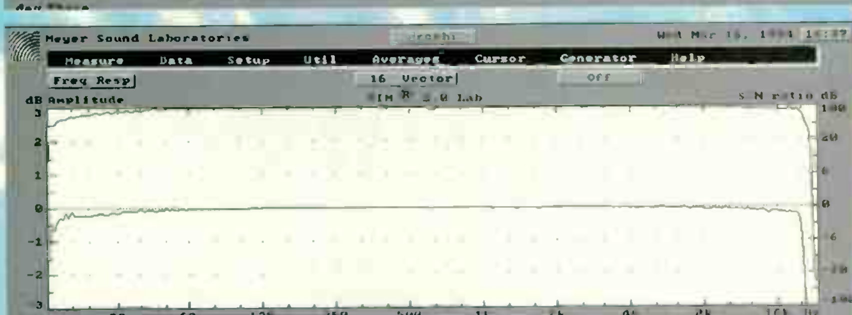
The analog conversion process has improved to the point where the differences between converting digitally and converting in the analog domain are slim, especially if you use a good set of converters like the Apogees.

For our test, we just plugged the Apogee D/A into the digital out of the source machine and connected the Apogee A/D to the digital input of the destination machine. The sample rate of the destination was selected by the sample rate switch on the Apogee A/D. The analog connection was made with three-foot lengths of Mogami with Neutrik gold plated XLRs on each end.

Pros: If you already have a set of Apogees, you may not need a sample rate converter. You can also mix through them and improve the sound of your DATs.

Cons: None.





Top to bottom: Sony 44 to 48; Sony 48 to 44.

SUMMARY, BUT IT STILL FEELS LIKE SPRING

All the sample rate converters I tested had their strong points. None of the units did everything, but each did its job rather well. If I had one wish for a sample-rate converter genie, it would be for a box like the Roland SRC-2, but with the converter quality of Z-Systems, an interface to ADAT and DA-88, an integral 6 x 6 matrix for routing digital audio around to all my machines, and a built-in Apogee

On the Mic Amp:

In keeping with the pedigree, I use a switch which accurately sets and resets the gain. This enables optimum circuit configuration for best performance at any position. There is massive headroom at all settings and at 0 gain this "Microphone" input will accept + 25dBu without overload. This is achieved by a "T.L.A." circuit which gives the advantages of a transformer without the penalties. A common mode inductive filter protects against RF interference.

The double-balanced microphone amplifier output is available on an XLR Male connector at the rear. This output is balanced but not ground-free.

On the Filters:

The lowest frequency on the High Pass Filter is 20Hz. Below this there is virtually nothing musical but the filter does let you remove any very low frequency building rumbles or other unwanted low frequencies while at the high end it is possible to remove a great deal of the "spill" from adjacent low frequency instruments in multi-track work.

The Low Pass filter extends as high 30kHz in order to remove inaudib distortion and processing products whic can, nevertheless, affect the way we he music within the conventional audio ban

On the Mid Range Equalization:

The overlapping mid range sections the SYSTEM 9098 EQ are very powerf and much more flexible than found earlier generations. This enables t frequencies specific to a musical instrume to be raised or lowered; or, on the oth hand, a broad bandwidth may be select to adjust the tonal balance.

On Low Frequency Equalization:

The Low Frequency section is one of t most important tools in the equalizati toolchest. In the more extreme setting

System 9098 EQ by Rupert Neve the designer

SYSTEM 9098 EQ FUN

MICROPHONE SENSITIVITY switch

The Mic Amp has 0 to 66dB gain switched in 6dB steps; the Mic/Line Level control provides an additional 6dB of Gain if required.

PHANTOM switch

Switches in a 48 volt balanced phantom feed for powering capacitor microphones.

MIC/LINE switch

Selects between the output of the microphone amplifier and an XLR Female connector at the rear. Selected to LINE, the rear XLR connectors can be used as an insertion.

The Line Input is a high level T.L.A. circuit which behaves like a transformer but can accommodate unbalanced sources.

MIC/LINE LEVEL potentiometer

When the MIC/LINE switch is selected to MIC, + or -6dB gain adjustment is provided, enabling a continuous bridging of the switch positions.

When selected to LINE, +12 or -6dB adjustment accommodates low level (eg,domestic hi fi) line sources.

At the centre (detent) position the gain is unity. This centre zero control enables a smooth transition between microphone switch positions combining the advantages of a continuous control with the precision of the switch.

PHASE switch

Inverts the signal phase at the Line Input.

FILTERS switch

Steep cutting (18dB/octave) continuously swept filters. The LPF range extends well outside the conventional audio band.

FILTERS IN/OUT switch

Enables / disables filter circuitry independent of the EQ circuitry.

ALL EQ IN/OUT switch

EQ is either by-passed or in circuit depending on position of MID and LF/HF switches.

LF EQ - FREQUENCY potentiometer

Controls the frequencies at which the LF EQ section is effective.

LF EQ - LEVEL potentiometer

Provides up to + or - 18dB variation of level at the selected frequencies.

GAIN +/- 9 switch

Limits the level variation to 9dB, providing greatly enhanced resolution. I have found this "zooming" feature very helpful in achieving fine control and ease of reset.

SHELF/PEAK switch

Provides that frequencies below the peak are either maintained or die away.

GLOW switch

My traditional Equalizer curves are steep-sided, providing very powerful tools. GLOW changes the curve shape to provide greater or less "warmth", altering the overall sound without changing its character.

LMF and HMF EQ

LMF and HMF sections are identical apart from frequency range.

lramatic effects can be imparted to the :ick drum, for example, and many famous ·ecordings bear testimony to its ffectiveness. At the other extreme it is ›ossible to provide the most subtle varming of the signal without otherwise ıffecting its naturalness. The circuitry has ›ery low basic phase shift imparting a very 'solid" quality to the sound making a ·elatively small amount of equalization ›ery effective.*

)n Bandwidth:

Professional audio engineers have :nown for many years that the sonic uality of audio equipment does not relate ully to the technical specifications. Human ›earing demonstrably "cuts off" somewhere ›elow 20kHz for most of us and it seemed o make sense that our equipment should ·ut off there too.

More recently research has shown that the experienced listener is right. It has now been positively established that equipment which can handle signals well above 20kHz faithfully gives a greater sense of enjoyment and fulness. These are qualities that we would have associated with mid and lower frequencies. It seems that the presence of these super high frequencies - provided that they actually occur in the original acoustic sound - can influence the way we hear sound within the traditional hearing band. Even if the out-of-band frequencies are severely attenuated, the effect is noted.

The HIGH FREQUENCY control is designed to peak above 20kHz, and can give scintillating quality to the sound.

Needless to say, out-of-band response must be free of self-generated noise and distortion. If the original signal contains unwanted noise, typically produced by digital processing, the LOW PASS filter is likely to be the preferred tool!

On the Shelf/Peak Curves:

If the PEAK function is selected, the fairly steep slopes of the equalization curve can alter the nature of an instrument by changing the relationship of harmonics to the fundamental.

The SHELF function raises or lowers the whole range of frequencies beyond the turnover frequency and can allow subtle modification of the tonal balance without altering higher harmonics.

IONAL DESCRIPTION

***FREQUENCY** potentiometer*
Controls the frequencies at which the MF EQ section is effective.

***MF EQ - LEVEL** potentiometer*
Provides up to + or - 18dB variation of level at the selected frequencies.

***GAIN +/- 9** switch*
Limits the level variation to 9dB, providing greatly enhanced resolution.

***"Q"** potentiometer*
Defines the bandwidth over which the EQ action is effective. Allows smooth transition from gentle enhancement/depletion, to a hard resonant sound.

***NOTCH** switch*
Provides a very sharp depletion filter, enabling discrete frequencies to be notched out with minimal interference to the desired sound. Works with the Frequency, "Q" and Level controls to "home in" on a given frequency. Notch attenuation can exceed 30dB.

***MID IN/OUT** switch*
Enables or by-passes the two Mid EQ sections.

***HF EQ - FREQUENCY** potentiometer*
Controls the frequencies at which the HF EQ section is effective. This frequency range has been extended to be effective well above the conventional audio band.

***HF EQ - LEVEL** potentiometer*
Provides up to + or - 18dB variation of level at the selected frequencies.

***GAIN +/- 9** switch*
Limits the level variation to 9dB, providing greatly enhanced resolution.

***SHELF/PEAK** switch*
Provides that frequencies above the peak are either maintained or die away.

***SHEEN** switch*
The mirror image of GLOW. A very gentle curve which operates over a wide range of frequencies, SHEEN changes the curve shape to provide an alternative balance which can alter the overall sound without changing its character. Gives that "old-fashioned" character to the sound.

OVERLOAD
This LED measures the signal level at the equalizer output and indicates that the (internally re-definable) level has been reached.

***POWER** switch*
This switch enables an AC signal fed to the IEC connector at the rear, to provide power for all the magical controls.

N.B. if this switch is not operated, the LEDs will not light and certain other tonal deficiencies may be noted, but there will be lovely silence.

Supply Voltages between 90 and 125 or between 180 and 250 volts RMS, range selected by internal switch, from 40 to 400 Hz may be used.

AMEK

" The SYSTEM 9098 EQ is a high performance Equalizer and Preamplifier designed to originate microphone signals of the highest quality and to process signals generally in terms of frequency response. The circuitry is based on the research I put into the 9098 console and the approach bears many similarities to that used in the 9098. Paramount importance has been given to the sonic quality of the audio path, taking great care to retain the highly-prized musical character of the famous old designs of this pedigree.

What is special about my approach to EQ ?

Purists might say they would like to set everything flat in the hope that this might provide a "natural" or unprocessed sound. In nature, however, nothing is flat. Every sound we hear has been influenced by acoustic or mechanical resonances, be they from rooms, instruments, loudspeakers, microphones or other sources. The finest musical instruments have achieved their qualities through years - sometimes centuries - of listening and adjusting resonances and by inference, the generic frequency response and attendant shifts in phase. Therefore let us never be reluctant to employ a good equalizer to get musically desirable results !

Early work on the EQ "tool", as used in professional sound, showed that gentle raising or lowering of the level of a band of frequencies might compensate for a deficient piece of equipment but could not be effective as a creative tool. In fact the term "EQUALIZER" was applied to equipment which equalized poor landlines and other equipment.

In professional audio production we are concerned with the use of the equalizer as a creative tool and with a microphone amplifier which will amplify and deliver the best possible signal to the equalizer.

Back in the '60s, one of my earliest requests was to "lift" the guitar from an already mixed tape - bearing in mind that this was before the days of multi-track and could not be re-mixed as we would today. The guitar frequencies were lifted with a high "Q" mid-band boost circuit. If the band was too narrow, the guitar lost its harmonics; if it was too broad, unwanted frequencies were lifted. Eventually a curve shape was found which did not change the nature of the guitar but changed its perspective relative to other instruments.

This led to the development of a range of equalization curves which provided maximum usefulness without harshness or unnecessary distortion of individual instruments. These have been used and acclaimed by generations of professional music people around the world.

The SYSTEM 9098 EQ embodies the original curve shapes now enhanced by improved circuitry which provides swept frequency bands in place of the discrete switched steps of the past. Thus the EQ has become even more powerful yet remains a subtle and creative tool, using the same basic circuit configurations which have been successful over many years. However, new amplifying devices and better quality components have resulted in lower noise, lower distortion and the ability to handle higher frequencies.

In the light of this contrast between the old and the new, I am often asked about how I configure my circuits and my views on Class A as opposed to alternative circuitry.

All my original equalizers used "Class A" circuits. These use the linear portion of the amplifier characteristic which is reasonably flat and results in second and third harmonic distortion which does not sound harsh. Unfortunately a Class A amplifier is inefficient, runs at high current and produces heat.

"Class B" or "Class AB" circuits consist of symmetrical amplifier pairs so arranged as to cancel out much of the inherent non-linearity, producing higher output with less heat. Their efficiency makes them very desirable where many amplifiers must be housed within a small volume. Most of the well known Integrated Circuits fall into one of these categories.

The two semi-linear curves of the symmetrical "Class B" amplifier do not fit exactly together resulting in a kink or discontinuity. This produces a "spike" and high order distortion of an audio signal. Negative feedback is often not very effective in reducing this type of distortion which is particularly unpleasant to the ear. The reason for this lies in the fact that many of the harmonics introduced by the "crossover" distortion are caused by the "spike" or switching "click" and are not related to the music content of the signal.

The SYSTEM 9098 EQ makes use, where possible, of Integrated Circuits which are so designed as not to produce crossover distortion; and where the requirement is for more power, I have used a biasing circuit which provides much of the efficiency with a "Quasi Class A" performance. At a number of points "discrete" transistors have been used to further extend the I.C. performance. The result is an equalizer which has the solidity and sound of Class A without the cost, heat and weight penalties and thus provides the 'best of both worlds'. We have also left behind cumbersome and expensive hand cabling, noisy connectors, heavy separate power supplies and outdated assembly techniques which contribute nothing but nostalgia. Apart from the robustness, repeatability and reliability, we have now made one of my designs more affordable than ever before."

Rupert Neve

SYSTEM 9098 EQ
is a product of

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Technology Group Plc

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TABLE ONE

BOX	MODE	-1 DB	S/N<90	S/N @16K	S/N KNEE	SYNC	SAMPLE RATES OUT
AI-1	44>48	>22266	20313	90	85	output	1,2
	48>44	20703	17969	90	74		
DFX 2400	44>48	19922	16406	86	73	input	8
	48>44	19922	16406	86	73		
SRC-2	44>48	18750	14453	95	80	input	1,2,3,4,5,6,8
	48>44	20313	8984	95	89		
ZSys	44>48	21094	17188	95	85	input	1,2,3,4,5,6,8
	48>44	20703	15234	90	85		
Nvision	44>48	20703	20313	90	85	input	1,2,3,4,5,7,8
	48>44	20703	20313	90	85		
Apogee	44>48	22266	9375	86.4	78	input	1,2,3,4,8
	48>44	21094	20313	93.8	85		

1=44.1k, 2=48k, 3=32k, 4=44.056k, 5=47.952k, 6=48.048k, 7=50k, 8=external word clock

Mode is the direction and sample rates. *−1 dB* lists the frequency at which the response was 1 dB down from the reference level. *S/N<90* is the frequency at which the signal-to-noise ratio became less than 90 dB. *S/N &16k* is the signal-to-noise ratio at 16 kHz. *S/N Knee* is the signal-to-noise ratio at the knee in the frequency response curve, usually up around 20 kHz. *Sync* is whether the unit has external sync capabilities. *Sample Rates Out* is a list of the selectable output sample rates.

UV-22 processor. Oh, yes, and a cost of under $1000.

Making the final decision on which sample rate converter to get first (sooner or later you will have more than one) won't be easy. As you can see by the graphs, the highest frequency cutoff was obtained using the Apogee analog converters to change sample rates. Who'd have guessed? The lowest cutoff was the Roland with all the cool features.

There are other sample rate converters on the market that I didn't have time to include in these tests, but as they become available I will test them and let you know what I find.

Since most of the new converter boxes coming out now use the Analog Devices AD1890 to do the sample rate conversion, they are all going to sound pretty much the same. What is going to distinguish one manufacturer from another is what additional features are included in each box — input switching, the mixing of multiple inputs as in the Roland SRC-2, level indicators, or even automated level changes triggered by DAT start IDs. I can even see the next generation of DAT machines with sample rate converters built right into the digital I/O. You could VSO the machine without changing the output sample rate.

Next time I will report on non-real-time sample rate conversion like that available from Digidesign, Sonic Solutions, and Sadie. If you want, I'll tell you how to do your own sample rate conversion with a calculator and a lot of free time on your hands. I will include converters like the NVision NV1000 (1890 based), Digital Domain's VSP, and a couple of others that didn't make it in time. I can't wait.

Oh, remember the 750 feet of chains? Well, if anyone knows of a magazine that would be interested in the perversion article, or if you have a lot of tires that need preparation for winter driving, drop me a note. EQ

TABLE TWO

BOX	LIST PRICE	DSP	I/O FORMAT	DC CUT	EMPH. REMOVAL	WORD CLOCK	AES SYNC IN	OUT WORD LENGTH
AI-1	$895	Mot 56001	1,3	no	no	out	—	20-bit
DFX2400	$12,000	Sony	4,5	no	no	in/out	XLR	??
SRC-2	$2595	Roland	2,3,4	yes	yes	in +video	—	20-bit
ZSys	$1750	AD1890	2,3,4	no	no	in	XLR	20-bit
NVision	$3950	Mot 56001	2,4,5	no	no	SDIF	XLR	20-bit
Apogee A/D/A	$5770	None	2,3,4	yes	yes	in +video	XLR,opt, BNC	16-bit

1=ADAT, 2=SPDIF (RCA), 3=Optical (Toslink), 4=AES/EBU (XLR), 5=SDIF-2 (BNC)

List Price is just what it says. The Apogee price is for total package of A/D, D/A, rack-mount power supply, cables, and rack-mount frame. *DSP* is the type of digital signal processor used to perform the sample rate conversion. *I/O Format* is the digital I/O connections available. All converters will accept any format signal on any of the input connectors. All output connectors will have the same output format, consumer or pro — whichever is selected. *DC Cut:* By going analog with the Apogees, all DC is removed. *Emphasis Removal* is performed digitally by the SRC-2. Emphasis is removed or imposed as necessary by the Apogee converters. *Word Clock* is for synchronization with other digital equipment. *AES SYNC In* is for locking the output sample rate to the sample rate of a third AES stream. *Out Word Length* shows the resolution of the converter.

EQ LIVE

LIVE SOUND FOR THE GIGGING MUSICIAN

LES PAUL: A LIVING LEGEND LIVE

■IF EVER THE term "living legend" could be used in its truest sense, it would be to describe Les Paul. For those with a fuzzy memory, Paul built the first solid-body electric guitar (which the Gibson guitar company referred to as a "broomstick") and the first 8-track tape recorder, and pioneered overdubbing, close miking, and tape delay. It would be accurate to say that without Les Paul, contemporary music would be very different.

For the past ten years, the Les Paul Trio has been performing weekly in New

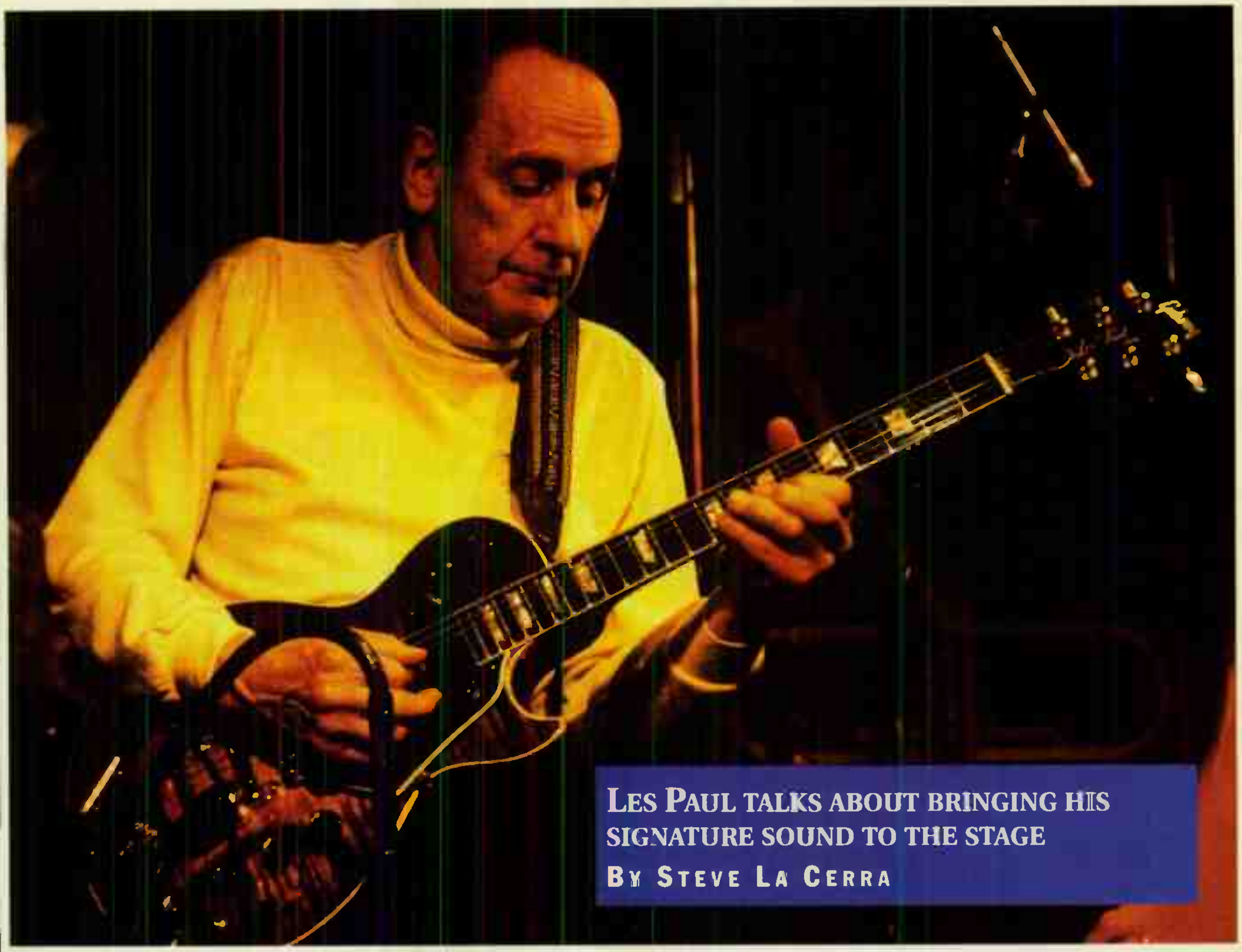

Photo by John Bellissimo/LGI

LES PAUL TALKS ABOUT BRINGING HIS SIGNATURE SOUND TO THE STAGE

BY STEVE LA CERRA

If At First...Try Tri Again

■MANY CLUB bands that carry their own PA systems are using what could be referred to as "monoamped" systems. In this type of system the mixing board feeds a single amplifier that is used to drive all the speaker components (i.e., the woofers, midranges, and tweeters). While there is nothing wrong with this, improvements can be made to the system by switching to a biamped system, in which separate amplifiers are used for the woofers and tweeters, or to a triamped system, where an additional amplifier powers the midrange drivers.

Using multiamplification in a PA system can afford several advantages. In a monoamped system, high- and low-frequency signals can modulate each other, causing intermodulation distortion ("IMD"). In a multiamped system the amount of IMD is reduced because each amplifier now carries a smaller workload of varying frequencies. Reducing IMD improves the clarity of the PA system. Second, headroom will be increased. A monoamped system "spends" most of its power on the low frequencies, leaving little available to reproduce the high frequencies without clipping. Multiamped systems avoid this problem by giving the high frequencies their own power amp. This amp can have a lower power rating (and reduced cost) since high-frequency reproduction requires less energy than low-frequency reproduction. And, as we will soon see, an active multiamped system can be more reliable than a monoamped system.

THE CROSSOVER

To fully understand the operation and connection of a multiamped system we must first recall some information regarding PA speakers. A full-range speaker cabinet has within it several component speakers or "drivers" that are used to reproduce various frequency bands: a woofer for reproducing low frequencies, a tweeter for reproducing high frequencies, and sometimes a midrange driver for the midrange frequencies. (Incidentally, the amount of component drivers in a speaker enclosure is not always indicative of its quality.)

Before an audio signal arrives at one of these drivers, it must "know" where to go. This is accomplished with a crossover — a series of filters that direct the various frequency ranges to the appropriate drivers (see fig. 1). Most full-range speaker cabinets have at least some kind of simple crossover network. A crossover that is built into a speaker box is generally referred to as a "passive" crossover because it does not require any power of its own to operate.

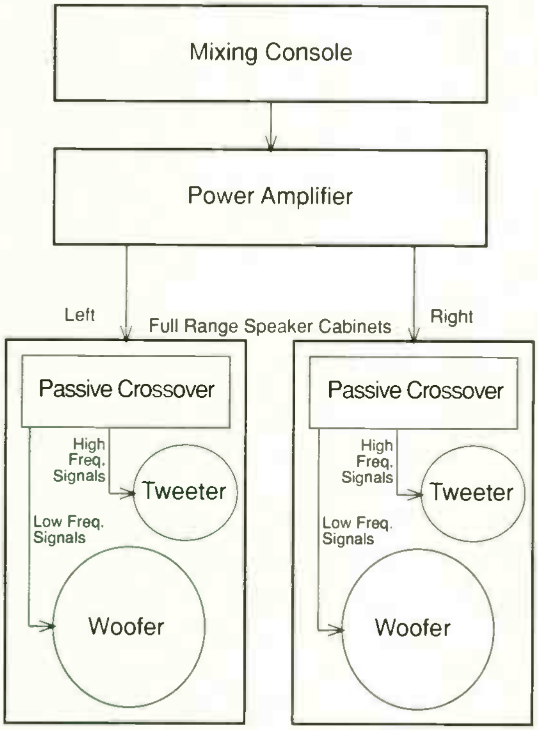

Figure 1

USING A BIAMPED AND TRIAMPED SOUND-REINFORCEMENT SYSTEM TO BRING BETTER SOUND TO YOUR GIGS

BY STEVE LA CERRA

Passive crossovers are convenient because the user does not have to mess around with the thousands of variables that can affect their operation (the R&D department of the manufacturer has already figured that out).

Passive crossovers, however, can sometimes have difficulty handling high power levels. After all, the full brute force of the power amplifier is being directed into the crossover before the signal reaches the speakers. In severe cases this can produce enough heat to melt the crossover components. As a result, many speaker manufacturers are now placing extra inputs on their cabinets that allow the user to access the high-frequency and low-frequency drivers directly for biamping purposes. Since these inputs bypass the internal passive crossover, the user will now need an active crossover.

This type performs the same function as a passive crossover but at a different location within the signal flow of the system. Figure 2 is a block diagram of a biamped system where the output of the mixing board is connected to the input of an active crossover, a rack-mounted outboard device (author's note: for clarity, the PA systems in figs. 2 and 3 are monoaural). The active crossover is doing its job at line level (roughly 2 volts) so it is not subject to the abuse of high power levels as is the passive crossover, making it much less likely to fail. Furthermore, the active crossover is user-adjustable for parameters such as crossover point and output level for each of the high, low, and

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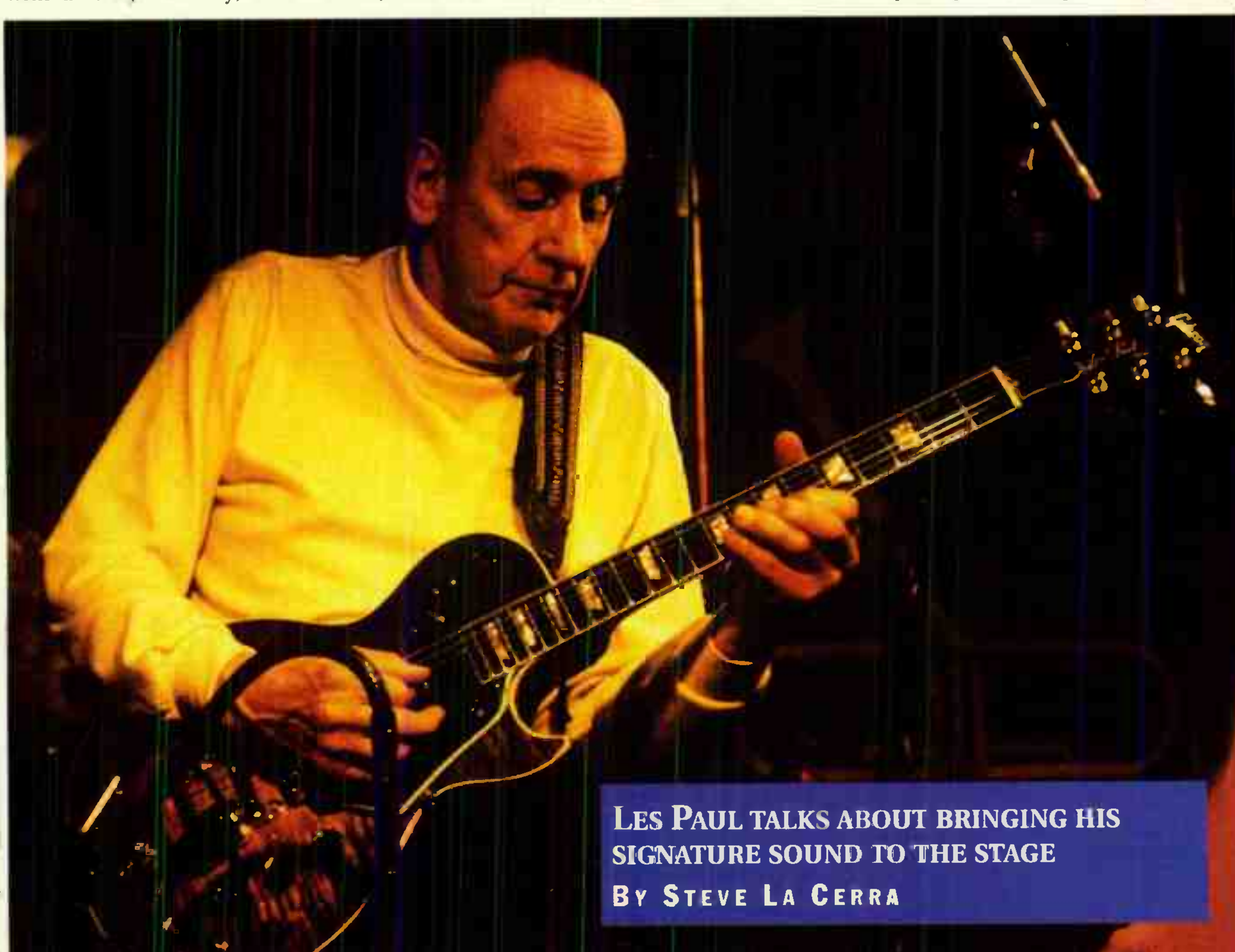


LES PAUL TALKS ABOUT BRINGING HIS SIGNATURE SOUND TO THE STAGE

BY STEVE LA CERRA

Photo by John Bellissimo/LGI

EQ LIVE

York City at Fat Tuesday's. *EQ* recently had the honor of speaking with Les Paul to discuss this and other topics.

Q: Ten years is a long time to do the same gig. How do you keep it fresh?

A: The audience is fresh — it's a transient audience. The audience is coming from all areas of the world and different types of musicians come to Fat Tuesday's. We have had people from India, China, and Alaska, and when a Russian group came in we got them up on stage.

It's a kick every week because there are always some celebrities in the audience. When Tony Bennett comes to sit in, we do the songs that he is known for. Steve Miller was in last week and he does his thing. Or it might be Al Di Meola or Bob Dylan or Jimmy Page — we play music that these guys are known for. They all have their own style, so all we have to do is give them a background. Although the trio stays the same, a guest can walk right in and away we go. Larry Carlton, Danny Gatton...they keep your mind in step with what's happening. It's a great, great place, and that is what has kept us here for ten years.

Q: Who are the other two players in the trio?

A: Gary Mazzeroppi plays stand-up bass and Lou Pallo plays rhythm guitar and sings. These guys do an excellent job.

Q: What equipment are you using these days? How are you getting your sound?

A: The guitar I'm using now is the Gibson Les Paul I have always used and it's modified to give the sound I'm known for.

Q: Did you modify the pickups?

A: Oh yes. I make my own and I work with a lot of guys in the business who make pickups, such as Seymour Duncan. [*Editor's note: When Duncan was a teenager he met Les Paul backstage at a show where Paul explained to the youngster what a guitar pickup is and how it works. Today Duncan's pickup manufacturing company is one of the most successful in the business.*] By moving the pickups you can alter the sound. A pickup near the neck will give a thick, dark tone for a jazz player, but moving the pickup toward the bridge of the guitar gives a piercing sound used by many rock players.

Q: Are the pickups on your guitar movable?

A: They were movable until I determined where I wanted them to be. To this day most of the guitars sold have the pickups in the place I have them, but they are probably not there for the same reason. The people who positioned those pickups put them there because I put them there, but I did that for the tone. I angle the pickup for a reason — and if I angle it, so will every one else.

Q: What type of guitar amp are you using?

A: At the moment I'm using an old Fender Twin Reverb amplifier.

Q: Are you using any effects units with that?

A: I've got a whole tray of them but I won't use them unless, say, Jimmy Page sits in and wants to get a Zeppelin-type sound. Then we will go for an effect. My rhythm guitar player has some effects available that he doesn't use either, but they are there if someone wants to sit in and use them.

Q: So you are plugging straight into the amp...

A: There are two outputs on the side of my guitar: one high impedance and one low impedance. The low impedance pickup is 250 ohms and gets connected directly into the board. When it goes into the console I don't need to touch any EQ or anything. All I do is set a level and the board is very happy with the signal from my guitar. The high-impedance pickup goes straight to the input of the Fender Twin Reverb.

Q: So the sound of your instrument is really coming from the instrument, not from effects devices or amplifiers...

A: Exactly. The key to the whole thing is the pickups.

Q: And the guitar amp is not miked into the PA system?

A: No.

Q: Is there a particular person who handles the sound system for you?

A: I have a fellow named T.W. Doyle who handles sound for me and is also a fine luthier. He has a great ear and knows what I want to hear.

Q: Do you use any hearing protection?

A: If I can hear the system humming I know I'm going to steal someone else's plugs. When I was on Letterman they asked me if I needed plugs and I said, "No I don't need them." But then Paul Shaffer's band hit that first chord and I said, "Oh God give me those plugs." So I use them if it's a loud situation.

Q: What is the most exciting piece of technology you have seen lately?

A: It's really something I saw in 1928, but it's here now: digital technology. In 1928 we had a player piano and we could speed up or slow down the piano without its changing the pitch. But if you put your hand on a phonograph record to slow it down, the pitch would drop. At the time I had a question: What is the difference between the phonograph and the player piano? The player piano was a form of digital storage but the phonograph was analog. I was already aware of "digital" through the player piano back then.

Analog has got to step aside for digital, and it's already happening in broadcast with digital switchers, and so forth. The road [to technology] no longer has a sign that says, "Dead End." That doesn't exist. Instead it says, "Go Ahead." We have now broken the barrier and it's almost like the question: How far can you go into outer space? You can go as far as you can imagine. I'm not sure it will stop here, but digital is a tremendous step forward. Music doesn't change that much but the technology does — and it's terribly difficult to keep up.

Q: Do you have any plans to record in the near future?

A: Oh sure. I just talked to Phil Ramone the other day and we're going to roll up our sleeves and get into it. We plan to walk it up to a record company and there shouldn't be a problem getting a company to release it.

Q: What would you say to a person who told you that he or she wanted to be a successful performer?

A: You really have to pick your vocation carefully. No matter how much you love music that does not mean that you have the materials — the God-given talent — to make it happen. It's what you do with what you've got. There is a behind for every seat, but you have to find the right seat. Keep your antenna up or you won't receive the broadcast! EQ

IF AT FIRST...TRY TRI AGAIN

USING A BIAMPED AND TRIAMPED SOUND-REINFORCEMENT SYSTEM TO BRING BETTER SOUND TO YOUR GIGS

BY STEVE LA CERRA

■MANY CLUB bands that carry their own PA systems are using what could be referred to as "monoamped" systems. In this type of system the mixing board feeds a single amplifier that is used to drive all the speaker components (i.e., the woofers, midranges, and tweeters). While there is nothing wrong with this, improvements can be made to the system by switching to a biamped system, in which separate amplifiers are used for the woofers and tweeters, or to a triamped system, where an additional amplifier powers the midrange drivers.

Using multiamplification in a PA system can afford several advantages. In a monoamped system, high- and low-frequency signals can modulate each other, causing intermodulation distortion ("IMD"). In a multiamped system the amount of IMD is reduced because each amplifier now carries a smaller workload of varying frequencies. Reducing IMD improves the clarity of the PA system. Second, headroom will be increased. A monoamped system "spends" most of its power on the low frequencies, leaving little available to reproduce the high frequencies without clipping. Multiamped systems avoid this problem by giving the high frequencies their own power amp. This amp can have a lower power rating (and reduced cost) since high-frequency reproduction requires less energy than low-frequency reproduction. And, as we will soon see, an active multiamped system can be more reliable than a monoamped system.

THE CROSSOVER

To fully understand the operation and connection of a multiamped system we must first recall some information regarding PA speakers. A full-range speaker cabinet has within it several component speakers or "drivers" that are used to reproduce various frequency bands: a woofer for reproducing low frequencies, a tweeter for reproducing high frequencies, and sometimes a midrange driver for the midrange frequencies. (Incidentally, the amount of component drivers in a speaker enclosure is not always indicative of its quality.)

Before an audio signal arrives at one of these drivers, it must "know" where to go. This is accomplished with a crossover — a series of filters that direct the various frequency ranges to the appropriate drivers (see fig. 1). Most full-range speaker cabinets have at least some kind of simple crossover network. A crossover that is built into a speaker box is generally referred to as a "passive" crossover because it does not require any power of its own to operate.

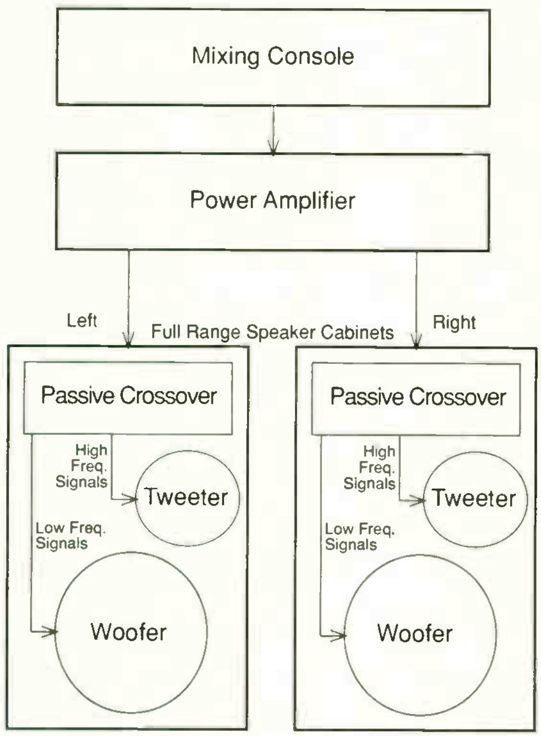


Figure 1

Passive crossovers are convenient because the user does not have to mess around with the thousands of variables that can affect their operation (the R&D department of the manufacturer has already figured that out).

Passive crossovers, however, can sometimes have difficulty handling high power levels. After all, the full brute force of the power amplifier is being directed into the crossover before the signal reaches the speakers. In severe cases this can produce enough heat to melt the crossover components. As a result, many speaker manufacturers are now placing extra inputs on their cabinets that allow the user to access the high-frequency and low-frequency drivers directly for biamping purposes. Since these inputs bypass the internal passive crossover, the user will now need an active crossover.

This type performs the same function as a passive crossover but at a different location within the signal flow of the system. Figure 2 is a block diagram of a biamped system where the output of the mixing board is connected to the input of an active crossover, a rack-mounted outboard device (author's note: for clarity, the PA systems in figs. 2 and 3 are monoaural). The active crossover is doing its job at line level (roughly 2 volts) so it is not subject to the abuse of high power levels as is the passive crossover, making it much less likely to fail. Furthermore, the active crossover is user-adjustable for parameters such as crossover point and output level for each of the high, low, and

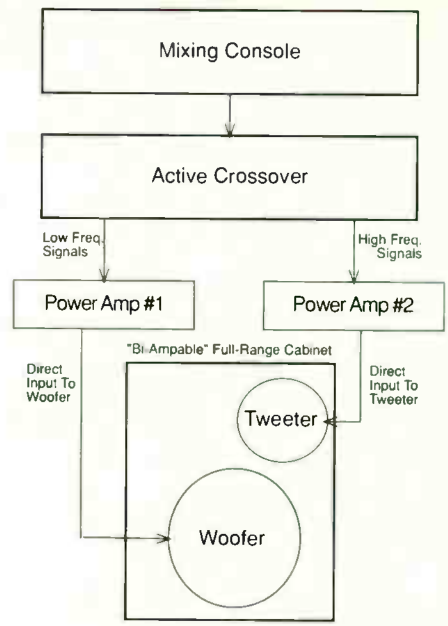


Figure 2

mid (if any) frequency outputs. The user can compensate for varying environments by making modifications to the crossover balance of the system even before involving any equalization.

COMPRESSION

Any club PA system should include compression as a means of protecting the power amps and speakers as well as for maintaining a smooth audio level in the environment. When biamping and triamping, compression can be patched into several different locations. Using a single compressor patched

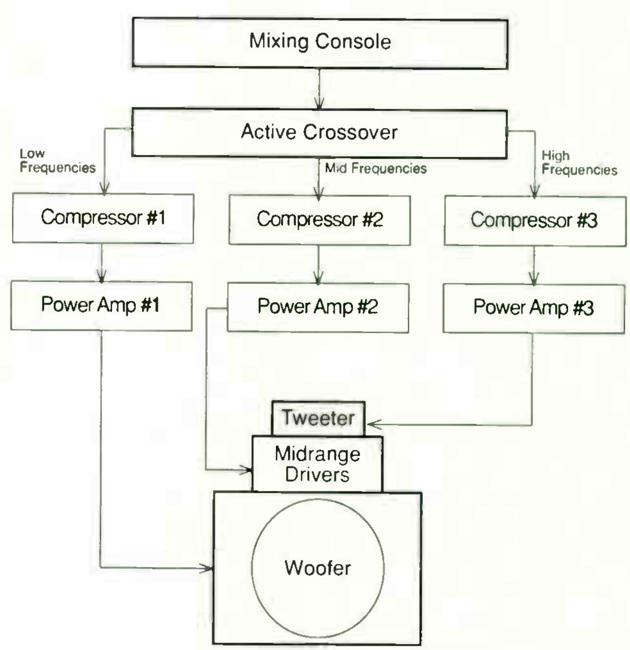


Figure 3

between the mixing board and the crossover will be effective and economical. But sometimes compressing the whole frequency range with one unit can result in "pumping" of the high frequencies when there is a lot of energy in the low-frequency range. A slick (albeit expensive) solution to this problem is to patch compression after the crossover. Figure 3 shows a stacked-cabinet, triamped system in which the high-, mid-, and low-frequency outputs of the crossover have been patched into their own compressors. Each frequency band can now have its own compression ratio, threshold, attack, and release times. The good news is that this keeps the compression action of the different bands from interacting with one another. The bad news is that we now need two or three compressors for the system. There are some compressors available that will allow the user to set separate compression ratios for the low and high frequencies, but maximum flexibility is attained by the use of separate compressors for each frequency band.

Anyone using a bi- or triamped system for the first time should seek the manufacturer's recommendations for crossover frequencies and maximum power input level. The entire system should be set up and tested, and all gremlins should be flushed out prior to using the system in a live performance situation. Once the parameters of the system have been adjusted, the crossover, compressor(s), and power amps can be prewired into a rack, leaving only the mixing console and speakers to be connected at the venue. EQ

SHORT TAKES

WIRELESS OPTIONS

Nady Systems has just introduced two revolutionary products that allow any microphone with an XLR-type connector to be operated wireless. The ENG-12 is a multichannel wireless transmitter operating in the UHF band. Features include a compact metal case, up to 160 user-switchable channels, tone squelch, and Nady's patented companding circuitry for a dynamic range of 120 dB. The unit snaps on to the microphone via an XLR connector to provide a one-piece mic/transmitter combination only slightly larger than the mic itself. The ENG-12 is compatible with many Nady UHF receivers.

Also available will be the ENG-11, capable of converting any XLR-type microphone to VHF wireless operation. The ENG-11 is a single channel VHF transmitter that has a dynamic range of 120 dB and is designed for use with the Nady 551 VR portable 2-channel receiver. Minimum range under adverse conditions is stated as 200 feet. Stay tuned to *EQ* for more information as it becomes available. or contact Nady Systems Inc., 6701 Bay Street, Emeryville, CA 94608. Tel: 510-652-2411. Circle EQ free lit. #124.

PIONEERING DJ MACHINE

Pioneer New Media Technologies has recently introduced the CDJ-500G professional CD player. Aimed at DJ's who want the same control over compact discs as they have with vinyl discs, the unit has a palm-sized jog wheel that allows a DJ to perform "scratching" on CD's without the worry of mechanically wearing out the discs. The built-in sampler can hold one second of sound and can repeat that sound continuously with the push of a button. "Quick Start" eliminates waiting time between pushing the Play button and the actual time that the music starts.

The CDJ-500 has a 100 mm fader that allows the user to adjust speed or pitch over a ±10 percent range, but the unit can also be set with a Master Tempo that will lock the tempo of the music without changing the pitch. Other features include an automatic transport lock for shipping, antivibration construction, and a heavy-duty motor.

For more information contact Pioneer New Media Technologies, 2265 E. 220th Street, Long Beach, CA 90810. Tel. 310-952-2111. Circle EQ free lit. #125.

New Gear For Your Next Gig

Lots of Kick

Carlsbro Electronics's new Alpha 94 series of loudspeaker enclosures features Celestion speakers. The enclosures have been designed as portable cabinets and feature a kickproof grille that does not suffer from the noise resonances inherent in some designs. All models operate at 8 ohms and have newly designed corners and an integral stand adapter. The 70-watt Alpha 10 has a 10-inch speaker and a single Piezo horn, while the 100-watt Alpha 12 has a 12-inch speaker and the same horn. Operating at up to 150 watts RMS, the Alpha 15 features a 15-inch speaker and twin piezo horns. The Alpha 158 also operates at 150 watts RMS; it has 15-inch and 8-inch speakers and twin piezo horns. For further information, contact Carlsbro Electronics, Cross Drive, Kirkby in Ashfield, NOTTS NG17 7LD, England. Tel: 0623 753902. Circle EQ free lit. #117.

Carlsbro Electronics Alpha 94

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

What Would Jimi Say?

Sabine is introducing a new version of its Feedback Exterminator, the FBX-901. Like its predecessor, the 901 is a digital signal processor-controlled filtering system that automatically senses feedback, determines its frequency, and places one of nine very narrow digital-notch filters to cancel only the ringing frequency. Utilizing filters that are 10 times narrower than 1/3-octave EQ, the Feedback Exterminator automatically controls feedback faster and with less sound muffling than graphic or parametric equalizers can. With 901's new features, you can choose to lock its filters to prevent them from going deeper, and can also limit the number of filters to be activated. The unit also provides a more efficient design for impressed effectiveness and reliability, and employs a new algorithm that greatly reduces the chance of mistaking music for feedback. For further information, contact Sabine, 4637 N.W. 6th St., Gainesville, FL 32609. Tel: 904-371-3829. Circle EQ free lit. #118.

Pos-i-tively Wireless

Telex's new FMR-450 is a high-performance UHF wireless mic system that operates in the 524 MHz to 746 MHz range (UHF TV channels 23–60). The system is designed to operate up to 50 systems at once, using hand-held or belt-pack transmitters, without compromising operating range or quality. The FMR-450 includes Pos-i-Phase true diversity circuitry for stable RF performance. Pos-i-Squelch II provides a greater degree of integrity in maintaining overall system quieting, and the new compander design results in a signal-to-noise ratio of greater than 110 dB. Other features

Sabine's FBX-901 Feedback Exterminator

include a specially matched 1/2-wave collinear ground-independent antenna system that offers a gain improvement over 1/4-wave designs. The receiver features RF, audio, and diversity LED indicators, and a transformer-isolated balanced mic level output with attenuation control. The micropack transmitter features a new silent On switch and separate mic mute, overload/low battery LED, removable antenna, and gold-plated LEMO mic connector. A hand-held transmitter with a choice of four

Telex FMR-450

mic-head options is available. Systems start at under $1400. For more information, contact: Telex, 9600 Aldrich Avenue South, Minneapolis, MN 55420. Tel: 612-884-4051. Circle EQ free lit. #119.

Big Bottoms

Quantum Sound, which recently purchased Servo-Drive technology from Intersonics, has expanded the use of this technology with the introduction of Bass Tech 7 enclosures. Utilizing a patented servo-motor drive concept instead of the typical voice coil, the Bass Tech 7 increases linear excursion and the overall efficiency of this subwoofer system. The ruggedness of the servomotor additionally enables the device to handle large low-frequency peaks without damage or sonic compromise. At a full rate power of 1600 watts, an array of four Bass Tech 7 systems will generate 142 dB in the 28 Hz –125 Hz range with an essentially flat frequency response and a 15 to 20 percent lower harmonic distortion than traditional designs. For more information, contact Quantum Sound, 3453 Commercial Ave., Northbrook, IL 60062. Tel: 708-272-1807. Circle EQ free lit. #120. EQ


Quantum Sound Bass Tech 7

YAMAHA SPX990 PROCESSOR

■IT'S FUNNY how obsolescence works, especially in the audio world. Some archaic devices, like the 8-track cartridge and quad decoder, are virtually worthless today, while others, like vintage ribbon microphones and Minimoogs, are coveted by audio aficionados. Yamaha's version of Old Faithful — the SPX90 multieffects processor — definitely falls into the latter category. Despite the fact that it hasn't been manufactured for many years now, this workhorse remains a staple of both project and commercial studios, as well as a mainstay of live rigs.

The major appeal of the SPX90 has always been its versatility. When it was first introduced back in 1985, there were few if any signal processors that could produce as wide a variety of effects. Today, however, the situation has changed; there are literally dozens of multieffects processors on the market. Happily, Yamaha's recent release of the successor to the SPX90 — the SPX990 — demonstrates the company's commitment to not only keep up with the times but to blaze some interesting new territories. And what a difference that extra "9" makes!

First and foremost, the SPX990 utilizes state-of-the-art 20-bit A/D and D/A conversion (at a 44.1 kHz sampling rate) with a claimed dynamic range exceeding 100 dB. In contrast, the original SPX90 utilized 16-bit D/A and A/D at a sampling rate of only 31.25 kHz (with a claimed dynamic range of only 75 dB). These extra bits of headroom, in conjunction with the higher sampling rate, make for a sound quality that is stunningly superior to that of the original SPX90. This is most apparent when listening to the 990's various reverb programs: instead of the graininess SPX90 users have been accustomed to, there is now shimmering, clear audio. Other 990 effects programs demonstrate a similar increase in fidelity, putting this unit in a class with many other pro-level effects processors costing several times as much.

Beyond the improvement in sound quality, there are many other features that distinguish the 990. For one thing, it has a stereo input (instead of the single mono input provided by the 90). This input can be used as two separate mono feeds. Both inputs and the stereo output are balanced, with both XLR and 1/4-inch TRS jacks provided. As with the SPX90, there is a rear-panel switch that allows selection of either +4 or –20 dB input strength. Two separate rear-panel footswitch jacks are provided; one can be used for patch selection or as a bypass/active switch and the other is used to trigger certain effects functions such as opening noise gates, setting delay tempos, or initiating sample recording in the "Freeze" program. Finally, a MIDI input and switchable MIDI output/thruput is provided.

The SPX990 front panel is uncluttered and includes a dual-concentric input level knob (labeled to show unity and +10 gain); an eight-segment LED input level meter (which includes an all-important "clipping" segment); and a number of small LEDs that show the memory area currently selected (Preset, User, or Card), the type of input (stereo or mono) currently in use, and the presence of incoming MIDI signal. A large LED display shows the number of the program currently selected and a backlit two-line LCD shows the program's name, as well as indicating the Main effects type used by that program (more about this shortly). Six soft buttons under the LCD take on a variety of functions depending upon the mode selected. For example, in "Memory" mode (really just playback mode), these buttons allow you to jump instantly to any of four preselected programs — a particularly handy feature in live performance when you might need to rapidly call up multiple programs in a single song, for example.

The center of the front panel is dominated by a large notched Data Entry wheel, which is used to dial

ROAD TEST

MANUFACTURER: Yamaha Corporation, PA Division, 6600 Orangethorpe Ave., Buena Park, CA 90620. Tel: 714-522-9011.

APPLICATION: Multieffects processor for live and studio applications.

SUMMARY: Solid processor with an above-average reverb section

STRENGTHS: Excellent reverbs and panning effects; pre- and post-fx processing; extensive real-time MIDI control; balanced stereo inputs and outputs; programming power galore for users who like to "roll their own" effects.

WEAKNESSES: No visual indication that an effect has been edited; poor documentation.

PRICE: $1149

EQ FREE LIT. #: 121

BY HOWARD MASSEY

EQ Magazine We Define The Project Studio

project studio \praj-ekt 'st(y)ud-e-o)\ *n* **1 :** the personal work place of a musician, engineer, songwriter, producer or multimedia artist, designed to maximize musical and sound creativity in an affordable technical environment **2 :** the most active aspect of the professional recording and sound market circa 1993, due to advancements in desktop audio, MIDI, hard disk recording and developments in affordable analog and digital technology **3 :** the market trend first defined by EQ Magazine in 1990, which now serves its readership with practical, hands-on, how-to information

I FOUND THE REAL STRENGTH TO BE ITS REVERB EFFECTS. THERE IS A UNIQUE ECHO ROOM EFFECT THAT CAN BE USED FOR REALISTIC AMBIENCE.

up programs and also to change data when editing. To its right are a series of six dedicated buttons: two serve as Page Up/Page Down keys (used when navigating through edit menus), while the other four allow you to store programs, select different memory banks, enter edit mode, or place the 990 into Bypass mode. Finally, a data card slot enables additional memory storage.

Speaking of memory, there's lots here. The SPX990 comes with 80 ROM ("Preset") programs, and there are another 100 RAM ("User") slots into which user-modified versions of these programs can be stored — all this in addition to the extra memory slots you can gain by adding an optional RAM card. There is also provision for a MIDI bulk (system exclusive) dump so that you can offload your 990 programs to any standard MIDI data filer or patch librarian.

Good as most of the factory programs are, though, there will be times when you'll want to reach in and tweak away in order to create custom effects, and it's at this edit level that things get really interesting. Each SPX990 program actually consists of up to three effects working in tandem. The input signal first passes through a "Pre-effect," then into the "Main effect," and finally into a "Post-effect" (any of the three can be turned off if required). There are four Pre-effects and three Post-effects options: 3-band parametric equalizer; compressor; harmonic driver (the Aural Exciter™ program licensed by Aphex for Yamaha's use); or (for Pre-effect only) a combination of compressor, distortion, and equalizer. Seasoned engineers know the value of equalizing, compressing, or otherwise processing an effects send or return signal — in effect (pun intended), the SPX990 allows you to perform this kind of advanced sonic wizardry without tying up two or three external devices to do so. In fact, it is the judicious use of these Pre- and Post-effects (most usually the parametric equalizer or harmonic driver) that give many of the factory programs their clarity and definition. Of the four Pre-effect types, it is really only the distortion program that is a disappointment: thin and weak, it probably won't do anything for you heavy metal guitarists out there.

Of course, it is the selected Main effect that does the brunt of the work. In all, there are 36 Main effect types, which include several kinds of "generic" reverbs, early reflections, delays (including multitap delays), modulation programs (such as flanging, phasing, chorusing, and rotary speaker), pitch change (harmonizer) programs, panning, "Freeze" (user sampling), and various "complex" (dual) programs. The latter allows two kinds of effects to be generated simultaneously; in many cases, the left and right input signals can be routed independently to each. Within each Main effect type, there are a multitude of user-editable parameters that are accessed by "paging" through nested menus. While I realize this isn't everybody's favorite way to work, things are laid out fairly logically so that the process of editing is pretty straightforward despite an owner's manual that is less than illuminating.

In fact, I really only have two complaints about editing on the SPX990. The first is that there is no provision for choosing a Main effect — instead, you have to begin by editing a ROM program that already contains the Main effect you wish to use. The second is that there is no indication when a program has been edited and no edit buffer that saves your previous editing session if you neglect to save it (your most recent unsaved edits are, however, preserved in memory even when the unit is powered off, but they disappear for good when you recall another program).

I found the real strength of the SPX990 to be its reverb effects. In addition to the standard replications of halls, chambers, and plates, there is a unique Echo Room effect that can be used for realistic ambience. This effect type allows the user to define the height, width, and depth of the room model, as well as set the listening position and other room variables. Most reverb effects pass through a programmable internal gate that can be triggered not only by input signal strength but also by a user-defined MIDI Note On, or even by the press of a footswitch. The maximum reverb time is a monstrous 480 seconds (eight minutes!) while the maximum delay time available is a quite respectable 1480 ms (about one-and-a-half seconds).

Delay effects include mono and stereo echoes, as well as a multitap delay line with six independently adjustable taps. There are also three different "tempo" echoes, where the delay times can be set either by manually tapping in a tempo (using front-panel softkeys or a footswitch), or, most impressively, by inputting MIDI clock signals. These techniques allow you to set up synchronous delays in the SPX990 even if the source material contains tempo fluctuations. This isn't the only MIDI control offered by the 990, either — the majority of parameters within each Pre-, Post-, or Main-effect type can be altered in real time with any MIDI controller from 1 to 95; in fact, within each program, you can assign two controllers to two different effect parameters. There's also a "Range" parameter that allows you to scale the minimum and maximum effect of any assigned controller. This can be used, for example, to preset the precise "slow" and "fast" speeds when using a MIDI controller to change the rate of a rotary speaker effect.

In addition, the SPX990 "Freeze" effect (which allows short samples of up to 1.35 seconds to be recorded and played back) can use MIDI Note On messages as a trigger source. Both the "Freeze" effect and the various pitch change effects also allow incoming MIDI Note On messages to be used for real-time transposition. One pitch change feature of particular interest is called "Intelligent Pitch." This allows you to choose from among 12 different musical scales (ten preset ROM scales and two user-defined scales) as the basis for pitch change. In a live performance, this would enable the sound engineer to add musically relevant harmonies (not just block harmonies) to backing vocals or horn lines.

EQ LIVE

The SPX990 can also act as an autopanner, providing three different pan effect types, including one complex one that allows for independent panning of left-right input signals. In addition to the standard kinds of left-right/right-left pan directions, there are two "dimensional panning" directions called L-TURN and R-TURN. These simulate the action of rotating a signal in a near or far circle between the two speakers.

There is also a series of modulation effects that enable an incoming signal to be flanged, phased, or chorused. Of the three, the chorusing is the richest (the "Symphonic" effect is quite good, as is the rotary speaker), but in general I found these kinds of effects to be the 990's weakest. Finally, a handful of "complex" effects combine two effects types, with independent routings for incoming stereo signals. The trade-off here is that since one processor is doing twice the work, there are fewer editable parameters and a slight but noticeable degradation in sound quality. My advice would be to skip these altogether unless you have a serious shortage of effects processors in your setup.

All in all, I give the SPX990 a hearty thumbs-up. It produces extraordinarily beautiful reverbs and provides more than enough flexibility, editing power, and MIDI control to satisfy even the most demanding applications. The bottom line is, if you liked the "obsolete" SPX90, you'll love the "thoroughly modern" SPX990!

Howard Massey heads up On The Right Wavelength, a MIDI consulting company, as well as Workaday World Productions, a full-service music production studio.

Digidesign's DINR

EQ LAB REPORT

MANUFACTURER: Digidesign, 1360 Willow Road, Ste. 101, Menlo Park, CA 94025. Tel: 415-688-0600.

APPLICATION: For Sound Tools users who want to increase capabilities and lower noise.

SUMMARY: Sound Tools Plug-In with two noise-reduction modes.

STRENGTHS: Flexible; easy to use.

WEAKNESSES: Noise can't be louder than the signal you need to improve.

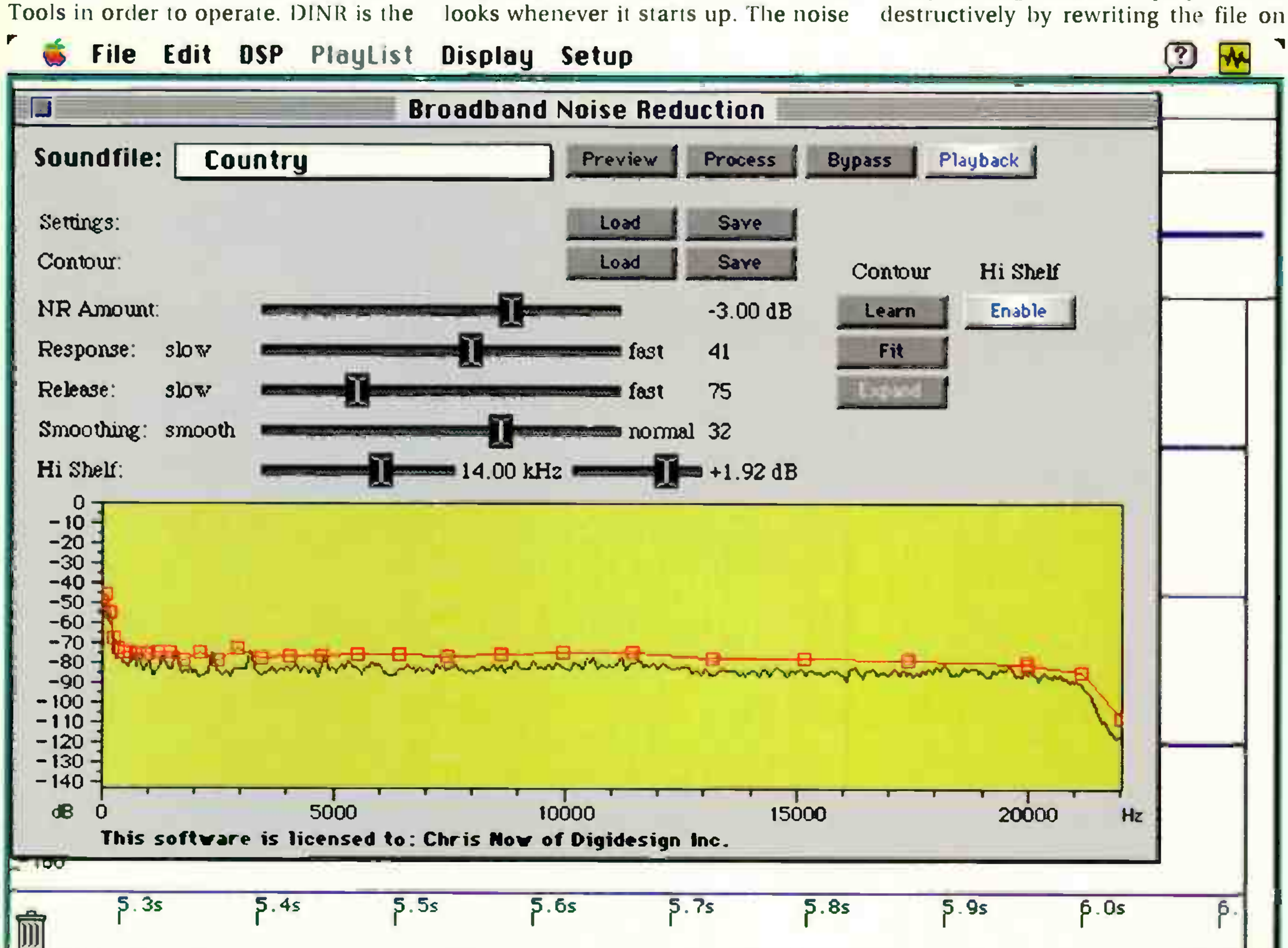
PRICE: $995

EQ FREE LIT. #: 122

FIRST OFF, you cannot review DINR (Digital Intelligent Noise Reduction) without talking about Digidesign SoundTools. DINR requires Sound Tools in order to operate. DINR is the first in a series of "plug-ins" that expand the features available to Sound Tools users. The plug-ins are placed in a special folder where Sound Tools looks whenever it starts up. The noise reduction choices then show up in the DSP menu.

DINR has two noise-reduction modes: One for wideband noise, such as tape hiss and preamp noise, and the other for pitched- or hum-based noise. In the case of broadband noise, DINR is best suited to removing noise whose overall character does not change very much during the course of the program material. This includes tape hiss, air conditioning rumble, mic preamp noise, and other similar noises. If there are different kinds of noise in the same piece of material, each type can be treated separately.

DINR can operate nondestructively during real-time playback or destructively by rewriting the file on



disk. It can also operate on mono or stereo files from 8 to 24 bits and sample rates up to 48 kHz. All of the controls in DINR's control panel operate in real time, allowing you to instantly hear the effects of any changes you make.

However, DINR does have some limitations. The noise must not be louder than the signal you are trying to clean up (if this is why you want DINR, then you can't read my columns any more), and the noise must be louder than –96 dB. If your noise is down there, then you aren't going to be needing DINR anyway.

CONSIDER THIS

You must take into consideration three things when jumping into the noise removal arena: the amount of noise removed from the signal, the amount of signal removed from the program material, and the amount of new artifacts added to the signal. In any noise-reduction system, you will have control over those three parameters and you must decide on the balance among them.

Noise reduction systems allow you to trade off between adding unwanted changes to the sound (at more extreme settings), and the amount of noise reduction. In most cases, the artifacts that are created with DINR are of a different type and less objectionable than the noise that you are trying to remove. This is not a side effect of just DINR. All noise removal systems work basically the same way. Even the $100,000 Cedar system I used to clean up the original Steely Dan recordings has these side effects to deal with.

READY, SET, GO!

You start DINR by selecting the type of noise reduction you need under the DSP Tools menu of Sound Designer II. The region you wish to deal with must already be highlighted in the main waveform display window. For best results you need to select only a short (100–300 ms) area that is "noise only" for DINR to analyze. This could be a quiet spot before the program material starts. (If you don't have any small area in your sound file that is just noise, DINR can still work on your file using a modifiable generic "noise fingerprint").

Next you click on the "Learn" button on the DINR window. DINR will build an editable contour line that most closely matches the frequency spectrum of the noise. (If you need more editable points for your contour line, you can hold down the option key when you click on "Learn"). You then choose the amount of noise reduction, or modify other parameters, until you get what you like. DINR gives you default choices for everything other than the amount (in dB) of noise reduction. This method seems to work the best most of the time. If there is some part of the frequency range that you do not want included in the removal process, then you can change the contour to fit your specific needs, or edit the other parameters for controlling other aspects of DINR's processing. Remember, since you can change parameters in real time while DINR is playing back, you can experiment until you get something that you like.

Until you get used to what DINR does, let DINR learn the noise and decide how much noise reduction to use. Like any new piece of audio processing gear, you have a tendency to overdo it until the novelty wears off.

I used DINR to get rid of hum on a Fender Rhodes performance on Donald Fagen's *Kamakiriad*, and it worked great. I transferred the file digitally from the 48-track Sony to Sound Tools. I used the automatic feature in DINR that analyzes the noise and decides the best way to remove it. This is best done on a segment of the sound file that contains only noise and no sound from the track you are trying to clean up. You highlight a segment of the file between notes or before the instrument starts to play. Select Analyze from the DINR menu and let DINR decide what to do. You can then listen to a segment of the sound file while turning DINR on and off to preview the results. If you like what you hear, then tell DINR to process the whole file. After the file is completely processed, put it back on the tape and you are done.

I have also used DINR to remove the air conditioning rumble that was present on the vocal track recorded in my wife's demo studio. The price of the DINR package was a lot less than what it would cost to soundproof the basement.

DEEP THOUGHTS

If you think about it, a lot of studio time is spent trying to avoid hums and buzzes. How many times have you had to tell the guitar player to turn sideways to get rid of the hum picked up by his single-coil pickup? It never goes away completely, and it usually shows up the most when you are trying to record some nice quiet music.

What about the sample that your partner brings over to your studio on a cassette? It is the perfect sample for the song, but the background hiss and noise are too distracting. One quick run through the DINR and you're back in business.

How about the case of a demo that you recorded a year ago on a 4-track cassette and you want to include it along with your new ADAT demos? One pass through DINR and the A&R guy will never know the difference. Okay, he probably wouldn't have known the difference anyway, but you will!

SURVIVAL TOOL

DINR is one of those packages that after you use it, you can't figure out how you ever got along without it. I am currently transferring old 2-track analog tapes to CD for archival purposes. I am using DINR to clean them up. These are tapes of old albums that I recorded that will never be released on CD, so I am going to make my own. I had the album covers reduced on a color copy machine so they will fit in the CD jewel case. Now if I ever want to listen to them again, I won't have to unpack my 2-track Revox as I can just throw on the CD.

As a departing note, a third party company, ksWaves Ltd., is marketing a ten-band equalizer plug-in for Sound Designer software. [*See the February issue for a review. —Ed.*] This looks like a great after market outlet for all of you DSP hackers out there. Congratulations to Digidesign for opening the Sound Designer platform for third-party contributors. This type of approach is beneficial to everybody involved, especially the project studio owner who ultimately benefits from the flow of technology.

Well, make sure you let me go for a ride in the new car you buy with the $99,000 you saved when you bought DINR instead of Cedar. I'll even free up a space in my driveway if you want to leave it at my house.

—Roger Nichols

CIRCLE 42 ON FREE INFO CARD

E-mu Emulator EIIIxs

EQ LAB REPORT

MANUFACTURER: E-mu Systems, Inc., 1600 Green Hills Rd., Scotts Valley, CA 95067-0015. Tel: 408-438-1921

APPLICATION: Comprehensive sampler for studio use.

SUMMARY: Good all-around sampler that comes packed with features that make it applicable in many situations.

STRENGTHS: *Lots* of features; good sound quality; fast.

WEAKNESSES: Does not support direct-to-disk recording; can work on only one page at a time.

PRICE: $4495

EQ FREE LIT. #: 123

THE NEW-GENERATION Emulator III is upon us, and believe me, it was worth the wait. Several years ago, E-mu released the EIII claiming it was an affordably priced workstation. At a base price of over $10,000, it was clearly positioned to compete with the likes of very high-end machines. In fact, I recall a press release that compared it, feature for feature, with the Fairlight Series III. Although the EIII sounded great and offered a lot of sampling power, it had some serious shortcomings. These included monophonic outputs, antiquated SCSI implementation, and only 8 MB of RAM. In response to the piles of user feedback, E-mu has released the EIIIxs.

The E-mu Emulator EIIIxs is a 32-voice (16-voice stereo) sampler that ships standard with 8 MB and an option for maximum expansion to 32 MB. There are different configurations available: the EIIIxp, the EIIIxs, or either model in a turbo version. The EIIIxs features stereo 64x oversampling sigma/delta A/D converters, the EIIIxp is digital-only sampling, and the turbo versions feature the 32 MB RAM/120 MB internal hard drive (although, as of May, it will be a 270 MB hard drive). Default sample rate is 44.1 kHz with 16-bit resolution, yielding a 20 Hz–21 kHz frequency response. The analog output incorporates 18-bit resolution through eight individually balanced 1/4-inch jacks, and the main outputs have both 1/4-inch and XLR balanced connectors; the stereo inputs, however, are balanced 1/4-inch only. Digital I/O is provided via XLR AES/EBU or S/PDIF, and there are two SCSI ports on the rear panel. The manufacturer states

that phase coherency is maintained at ±1/1 kHz, and signal-to-quiescent-noise is spec'd at better than 100 dB (which means that in the best case, the unit has a better than 100 dB noise floor).

THE INTERFACE

As an owner of the original EIII, I found it quite easy to get up and running with the "x." The unit I tested was a Turbo, chock full of factory sounds stored on the internal hard drive (kudos to E-mu). Those of you familiar with E-mu samplers will have no problem with this unit. In fact, I find this to be both a strength and a weakness. The downside is that the current architecture is a little dated. For example, I find it cumbersome to work on only one page at a time, constantly being thrown back to step one every time a menu choice is entered. The upside, however, is that the EIII is familiar to many professionals and works well in its current state. I guess E-mu figured that if it ain't broke, don't fix it. The front-panel layout is exactly the same as that of any other unit in E-mu's Emulator series. There are buttons that access the different modules; these form six general groups that in turn contain several submodules to accomplish recording and editing tasks. Within these submodules lies another layer of pages with even more options. All these options offer a quick and simple way to harness the tremendous capability of the EIIIxs. To tell you the truth, there are so many features that they would probably take several months of in-depth reviews to cover them all.

When a module button is pressed, you enter the Module Identifier and choose a submodule. Now you have the option of viewing each of the modules, or, when you begin to memorize the different submodules, you can simply press the corresponding keypad number and quickly enter that submodule. Many of these submodules have several pages and rows of parameters that are accessible via the four cursor buttons, Increment/Decrement buttons, keypad, or data

slider. After selecting a submodule and performing a function such as editing, the EIIIxs always reverts back to the Module Identifier.

The Master/Globals section is the hub of the unit. Here you'll find utilities for backing up and formatting hard and optical drives, extensive MIDI mixing, and various SCSI options.

THE PRESETS

The EIIIxs comes packed with a wide variety of high-quality samples on the internal hard drive. You simply call up a particular bank and download all of its presets into RAM. E-mu has developed a unique way of loading banks using MIDI program change. With two program changes you can load an entire bank of presets; the first program number is called the "Magic Preset," which you define, and then you send the bank number. You can also hook-up a CD-ROM drive to the rear and access the different presets from the demo CD-ROM shipped with the unit.

Bank loading takes several seconds, depending upon the size of the bank, and there is the capability for creating up to 99 banks and 1000 presets. You can always check the size of the presets by scrolling through and instantly viewing the percentage and byte size. The Stack Mode links presets, enabling you to play several presets with one keystroke.

You can also create new presets from the existing sample data, and because most acoustical sounds are multisamples, you not only load the sample data but also the zone from which it was created. This allows for a more realistic playback. On the front panel there is a button labeled Multimode. Pressing it turns an already powerful machine into an even more powerful multitimbral device with separate pan, volume, output, and preset for each of the 16 MIDI channels.

SAMPLE MANAGEMENT

The idea here is to set up sampling so that when recording is complete, you can immediately play the sample. In fact, after sampling is completed, the EIIIxs instantly creates a preset for you, allowing you to continue sampling. This can be especially handy in the studio when your machines really need to fly. You can even audition the sample directly from the front panel using the Audition key. The hard-drive space is quite large and the RAM (when equipped with 32 MB) is sufficient to handle a large preset allocation and to perform recording and editing tasks without slowing you down.

Within the Setup submodule, you can select either the digital or analog inputs, and you have a choice of 44.1 kHz, 48 kHz, or 32 kHz (44.1 kHz or 29.4 kHz analog) sampling rates. Here you can also select the recording length and a few "Auto" features. The first is auto truncate, which will automatically remove empty space from the start and/or the end as soon as the sample is taken. The next is auto normalize, which will search for the highest peak and maximize the gain upon sampling. The last is an auto placement feature that will instantly place the sample on a key range of a single key to the whole keyboard, or, by selecting "white keys," will place each new sample on the next white key. This can be handy if you are compiling a lot of drum or foley sounds.

Once you set the threshold and gain, you have to select a particular range on the keyboard where your sample is going to initially live. You can then choose to "arm" your sample (breaking threshold starts sampling) or "force" sample start by pressing button #8, the Escape button, to terminate sampling. The sample is instantly ready for playback. This method makes the EIIIxs extremely fast and easy to use.

The SCSI Sample Dump Standard and EIIIxs formats are supported for transmission between two samplers or to a personal computer for waveform editing (for example, with SDII or Alchemy).

UNDO-REDO

Just as E-mu has provided a module

The idea here is to set up sampling so that when recording is complete, you can immediately play the sample. In fact, after sampling is completed, the EIIIxs instantly creates a preset for you, allowing you to continue sampling.

for sampling, it has also provided a separate module for accomplishing editing tasks in the digital domain. Within the Digital Processing module, you can loop, truncate, taper, and gain change a sample. You can also cut, copy, and paste portions of samples to the same or another sample. And don't forget, this sampler has a submodule called "Undo" that retrieves the original state of the sample after an edit. You can then swap the original with the copy stored in the "clipboard," edit it, swap back, edit again, and continue this swapping until you are happy with the one you would like to save.

Data entry is provided concurrently with reference to units of samples and of time (in milliseconds). You can use either the data entry slider, Inc/Dec buttons, or the pitch bend wheel on your master controller. Bending the pitch up will scrub the start or end points (depending upon where the cursor is located) from slow to fast as you bend less or more. The scrubbing quality is fairly analog and can make looping or cutting extremely fast once you get a feel for it. I prefer to use a wheel for editing, but if you have some sort of definable fader system, like the Lexicon MRC, you can assign a fader to pitch bend for real-time remote editing; you can't actually scrub any audio unless you send it pitch bend information.

Truncating will trim the head and tail of your sample, while taper allows you to create an artificial decay. Keep in mind that unlike other samplers, when using digital processing (as oppossed to dynamic processing) these edits are completely undoable and are destructive (irreversible) when you choose to save; you are simply dealing with two edit buffers instead of one. Gain change is flexible enough to offer ±96 dB of deviance and includes several types of head-tail taper slopes. When it comes time to loop, you can auto correlate (for amplitude looping), compress (for smoothing the amplitude of the loop portion), and/or crossfade the loop (for fading the start and end points together to achieve a more seamless loop).

DYNAMIC PROCESSING

Within the Dynamic Processing module, you assign the usual analog-style envelope, filter, LFO, etc., to the different zones on the keyboard. The concept behind zones is that you should be able to process identical or different samples among different areas of the keyboard. In this way, you can have completely different LFO or VCA settings for one or several keys. There's even an auxiliary envelope that can be placed at any of several points in the processing path. Velocity from your keyboard may be assigned to affect pitch, VCA level, VCA attack, VCF cutoff, bandwidth and attack, Pan, or sample start.

It just keeps getting better...

The unit I tested came with Version 2 software. This new software adds a plethora of new features, including sample auditioning directly from hard disk and "monitor thru" to the main outputs while sampling (assignable). Also, Akai S1000/S1100 and Emax II support is now offered. Samples and presets from either sampler can be loaded.

Another unique feature has to do with defragmenting memory. When you perform edits such as cut and truncate, the memory is left fragmented into different sections before actually saving to a bank. Memory defragmenting moves the samples into the lowest available memory address space, which results in more apparent sample time. This software update is available through your E-mu dealer.

THE BOTTOM LINE

In comparing the EIIIxs with other professional samplers, you will find some basic differences. The EIIIxs, for example, doesn't provide enough synthesis to call it a synthesizer/sampler (à la the K2000). It also doesn't support direct-to-disk recording, digital effects such as reverb, delay, echo, chorus, etc., or sequencing (á la the ASR-10 and S3200). I find some of these features very useful, but it's up to you to decide what your needs are in a sampler. The sound quality and features of the EIIxs are outstanding, however, and it is very fast to use, So E-mu clearly has a winner. In the course of the months that I used this sampler, I did complete dance remixes, mix-to-picture, and extensive sound design. (The EIIIx's current sound libraries are second to none, especially in the field of Sound Effects.) In many cases it wasn't necessary to use anything besides the EIIIxs Turbo. Even with my above-mentioned wish list, I recommend checking out the EIIIxs.

Special thanks to David Fournier of DFE for invaluable assistance.

—David Frangioni

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AD INDEX

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New Computers: Friend Or Foe?

Getting that hot new computer may initially cause more problems than you expect

BY MARTIN POLON

It is a curious phenomenon of the 1990s and certainly a given of life in a studio with computers that upgrading to gain speed or power may become an annual exercise as more powerful processors evolve at a rapid pace. Add to that the availability of ever-faster accelerator boards or plug-in upgrade processors and the project studio user finds an almost overwhelming array of options.

The question that must be asked, however, is whether an upgrade is really needed and if so, when. Only in this way can the studio user avoid ending up behind the proverbial eight ball trying to solve compatibility and software problems that did not exist on an older platform.

One project studio operator described his recent experiences thusly: "I had been lusting after a faster Macintosh computer for my studio. After several years of making do with a Mac IIci, I stepped up in class to a Macintosh Quadra 650. The machine is a real powerhouse, running four to six times as fast for some applications. I used all the same software and SCSI (Small Computer Systems Interface) devices that I had been using on the older machine. I thought I would just 'plug and play,' but in fact I lost two months in one way or another as surely as if I had entered a black hole."

Stories of difficulties in dealing with new computers are approaching the level of legend, and not just for the Mac world. Similar stories are told by some of those going to Intel Pentium (P5) platforms from older Windows applications that had run on 386 and 486 systems. In fact, new UNIX-based systems have had difficulties that literally trivialize those reported by Macintosh users. The tips that follow can help prevent a problem in your studio.

1. New Chips — A move to a faster processor will yield increased operating speed and processing power — sometimes! Yes, the Motorola 68040 will run anywhere from 2.5 to 10 times faster than the 68030 chips used in older Macintosh computers and the PowerPC chips will run 2 to 6 times faster than that, but if your speed problems are caused by slow hard drives, insufficient system or application RAM, incompatible applications, old system software, bad system settings, or combinations of the above, then more speed will not help.

2. New Versions Of The Operating Systems — The newest chips must have the very latest operating system, be it Windows NT, IBM OS Mach 3, or System 7.5 Power. (Some of these versions are not yet available.) Each system is designed for its respective chip and anything else is a waste of speed and money — period.

3. Higher Speeds — It is possible to buy a processor whose blazingly fast speed will incompatible render the applications you are now using. While it may be possible to upgrade the application, the change in the program code may not reach the marketplace at the same time as the new chip. Similarly, with new families of chips such as the PowerPC, Pentium, and Alpha chip, software utilizing each chip's special "tool box" may not follow until the application developer is convinced that a large enough audience exists for the revised product.

4. Upward Compatibility — The issue of using existing software with a new computer is not as simple as it might seem. Compatibility on new-generation computers with new microprocessor technology, especially with new system architectures and operating systems, is not guaranteed for existing software. There is an old saying usually used for those driving fast cars and using fast drugs, but sometimes applicable here: "Speed Kills." On computers running eight to ten times faster than previously owned systems, the compatibility of software that has not been optimized for higher speeds and altered architectures will vary from application to application.

5. Third-Party Software Vendors Caught Unawares — One of the reasons that software is frequently not compatible with newly released hardware is that a surprising number of software developers do not use the latest systemology to develop their programs. Although nearly everybody eventually achieves compatibility, it can involve an endless stream of software patches from bulletin boards and upgrade disks. And a compatibility problem may not show up on the software responsible for the problem but rather elsewhere, such as on a digital audio editing software program, for example. Fax modem software and screen savers are amongst the culprits that have conflicted with other, more important software on new systems.

If there is a bottom line here, it is that there is an old saying, "Be careful what you wish for; you might just get it!" A new computer will undoubtedly improve studio operations, but do not assume that these improvements will come without a price!

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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The Secret to Composing by Code

Getting started writing music for multimedia applications

BY MURRAY ALLEN

Last time I mentioned that some experienced game composers wrote their music in Compiling or Assembly Language. This form of electronic composing closely resembles the classical method of scoring and orchestrating using traditional music-score paper.

Both these methods of composing have simple advantages and disadvantages. On the plus side, both use many short cuts that do not require writing every note of every phrase. In the world of music-paper composition, this saves time and eliminates the boredom of writing phrases over and over again. In the computer world, it saves data that is hard to come by in games.

On the negative side, both methods do not allow you to hear what you have composed until it is performed. Live musicians are required on the traditional side, while a computer is required on the electronic side.

Figure 1 is a four-measure composition we will use as an example to show the similarities between traditional composition and Compiling Language Composition (CLC). Our target platform will be the Super Nintendo Entertainment System. Our target video system will be PAL (European).

The first issue the composer must address is the tempo. The composition calls for 120 quarter notes per minute. In a traditional composition one would just set the metronome at 120. In CLC the number 42 is entered to define the byte (db). This number is derived using the formula:

256 [maximum value a byte can hold]/(HxS/(BxN), where H=Hz value (50 Hz in Europe, 60 Hz in the U.S.), S= 60 (seconds in a minute), B=120 (number of beats per minute), and N=4 (notes per beat; for this exercise we will choose the sixteenth note as our smallest or fastest note — i.e., four sixteenth notes equal one quarter note). So our formula becomes: 256/(50 Hz x 60 seconds/(120 bpm x 4 sixteenth notes)) = 40.96 (or 41). When computers perform integer math they always round down and discard the decimal position. Therefore, 6.25 becomes 6, resulting in a new answer of 42.6667 that rounds down to 42.

The experienced composer does not go through this calculation. Just as a traditional composer knows a typical march tempo is 120 bpm, a CLC composer knows the number 42 (which, coincidentally, is the answer to life, the Universe and everything according to author Douglas Adams).

We have three parts (Bass, Flute, and Trumpet). They will be defined (dw) respectively as tuneA1 (Bass), tuneA2 (Flute), and tuneA3 (Trumpet). The five remaining Nintendo channels will be at rest. The actual instrument sounds will be samples called from the instrument folder or list. Using CLC the composition begins as follows:

```
db    42
dw    tuneA1, tuneA2,tuneA3,
      restt,restt,restt,restt,restt
```

The traditional composer will list the instruments to the left of their respective staffs on the score paper (fig. 1).

The next step in CLC will be to define the length of tune. This is accomplished for each instrument as follows:

```
tuneA1  dw   A1pl (=tuneA, part #1, phrase #1)
        dw   cutt (stop tune)

tune A2 dw   A2pl (=tuneA, part #2, etc...)
        dw   cutt

tune A3 dw   A3pl
        dw   cutt
```

The traditional composer will draw in the clef signs and double lines to signify the end of the composition. If the composition to be repeated three times he or she will add the appropriate dots and additional double lines at the beginning and notate to repeat 2X. Using the CLC the following method is used:

```
tuneA1  dw   A1p1, A1p1, A1p1 (play phrase
             A1p1 3 times)
        dw   cutt

tuneA2  dw   A1p1,A2p1,A2p1
        dw   cutt
```

Figure 1

```
tune A3 dw   A3,p1,A3p1,A3p1
        dw   cutt
```

Volume can be notated using: Vppp=1 (softest sound) through Vfff=8 (loudest sound). Whatever volume is designated for part A1 will carry through for all the parts (instruments) unless otherwise noted. The pitch of the notes will be designated by their position on a keyboard; i.e., c0 would be the lowest c, while c6 would be six octaves higher. The length of each note (L) will be notated by counting the number of sixteenth notes that make up the value. The traditional composer writes his volume marks and draws his note symbols to determine pitch and duration. In CLC the computer keyboard is used.

We are now ready to type the CLC equivalent of fig. 1.

```
A1p1   db   Inst, Bass

       db   mvol, Vmf
       db   L8,c2,L4,g2,c2

       db   mvol, Vmf
       db   L8,e2,L4,g2,c2

       db   mvol, Vpp
       db   L8,f2,L4,g2,g1

       db   mvol, Vf
       db   L16, c2
       db   x

A2p1   db   Inst, Flute

       db   L2,e3,g3,e3,g3,e3,g3,e3,g3,

       db   e3,g3,e3,g3,e3,g3,e3,g3,

       db   f3,c4,f3,c4,d3,f3,d3,f3

       db   e3,g3,e3,g3,L8,e3
       db   x

A3p1   db   Inst, Trum

       db   L10,c4,L2, g3,c4,b3

       db   L10, as3,L2,g3,as3,c4

       db   L6,a3,L2,gs3,L4,g3,b3

       db   L16,c4
       db   x
```

There are also symbols to denote: detune slide, vibrato, stereo L & R, loop, pause, tied note, and transpose. An experienced CLC composer can create a composition in about the same amount of time it would take a traditional composer to write his or her masterpiece.

Murray Allen is Director of Audio at Electronic Arts.

ACROSS THE BOARD

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this case you would need some type of format converter box such as the Digital Domain FCN-1 to convert from consumer to pro bit stream. The other consideration is levels and impedances. Consumer format is .5 volts and the impedance is 75 Ohms. Pro levels are 5 volts with an impedance of 110 Ohms. Canare (the cable company) makes transformers to match these two signals. The FCN-1 box also matches signal levels and impedances. The transformers do not change bit stream formats, just levels.

Just like getting away with driving 90 miles per hour on Mulholland Drive without getting killed starts to set a standard, audio cables may work sometimes, or even most of the time, but Murphy says the one time they "must" work you will be up "Bit Creek" without a paddle. Apogee and some other companies make excellent digital audio cables for transferring AES/EBU data streams. Get a couple of good cables for the job. You will be glad you did.

DIGITAL CONVERTERS

There was some question about the quality of the converters in the ADAT and DA-88 machines. These converters are much better than anything that was available in professional digital machines just a few years ago. Because of the millions of consumer products that use digital converters, the development of high-performance converters has progressed at light speed, while in the professional field, there haven't been enough machines sold to warrant such upgrades. The converters in the early Sony 3324 and the Mitsubishi X-800 and X-850 are quite inferior to the converters in the ADAT and DA-88 machines.

As of this writing, Mitsubishi is no longer making digital multitrack machines. They dumped the remaining X-880 32-track machines for about $35,000 each. Otari will only build a machine after it has been ordered. Studer has come out with a new 48-track DASH machine and discontinued its old version.

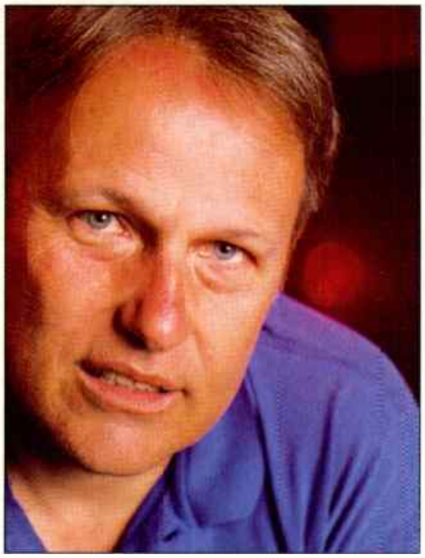
I heard rumors that Sony, Otari, and Studer have signed on to produce versions of the videotape based digital 8-tracks. I am not sure who has gone with DA-88 and who has gone with ADAT, but it should make for an interesting few years ahead.

A company called Prism, in Cambridge, England, has a box that will let you record 6 tracks of 20-bit or 4 tracks of 24-bit on a Tascam DA-88. A U.S. company is building a similar device. (see page 32.)

One additional observation. In multitrack recording there is a masking effect that takes place when you play multiple tracks of digital information. There is an effective resolution higher than that of any individual converter. If you played back 24 tracks of 20-bit data and compared it with the same 24 tracks of 16-bit data, you would not hear any difference. This is not true with just two tracks of information, where the complex waveforms need the higher resolution. So if you need to improve your converter arsenal anywhere, it should be on the mix machine. Also keep in mind that a good quality 16-bit converter like an Apogee is as good or better than a poor 20-bit converter.

Bottom line, the converters in the ADAT and DA-88 machines are right up there. You might see a little improvement if you were bouncing 6 tracks to 2 through a set of external Apogees (the ones you bought to mix to), but other than that you should be pretty much set. EQ

Network News



Avoiding road kill, speeding tickets, and potholes on the information highway

BY ROGER NICHOLS

I have been surfing the computer networks lately to find out what questions people have regarding digital audio and recording. I have found quite a bit of interesting information that I would like to share with you.

DATA TAPES

First, I want to touch on the use of data-grade DAT tapes for use in audio equipment. I covered this in a previous column, but at that time data-grade tapes were rather hard to find. Now, however, they are available at computer stores as well as tape suppliers like Project One in Los Angeles. I have been using data-grade DAT tapes for three years now, and never had a problem. In critical applications, like mixing, I will use nothing else.

Data tapes are measured by the physical length of the tape, not the playing time. A D-90 audio tape will last for 90 minutes. A D-90 in data tapes means that it is 90 meters long and will play for 3 hours — double the length of D-90 audio tapes. The specifications for audio tapes states that they all contain the same thickness of tape, just different lengths. When you get into the D-90 data tapes, the tape is actually thinner and there is an additional hole in the shell that conveys the thin tape information to the transport. When the transport detects the hole, the tape tensions and length information in the transport is adjusted to handle the thinner tape. Detection for the extra hole is not provided on audio machines.

On Internet, someone mentioned that they tried a 90-meter tape in a Panasonic SV-3900, and after checking the error rate, determined that the data tape was not as good as TDK audio-grade tapes. This was because of the tension not being set properly. If you use data-grade tapes, you will have superior performance, but you must stick to D-60 (60-meter) tapes or shorter.

DDS (Digital Data Storage) tapes are designed to withstand more stop/backup/start in computer backup operations. This is necessary when the data block is read incorrectly and must be re-read. This motion does not normally happen with audio decks. Audio deck operation usually compares to the data deck's "streaming" mode, just playing in the forward direction. The DDS standard for data-grade DAT tapes was developed for these start/stop operations. This stipulates more stringent mechanical, environmental, reliability, and durability specifications than the DAT standard.

One last point. Never try to use audio-grade DAT tapes for computer backup. I guarantee that you will never see your data again.

CABLES FOR DIGITAL AUDIO

Problems when making digital transfers were scattered around the Internet like rock salt wounds in the arse of a chicken poacher. (I'm learning a lot about the American language living here in Nashville.)

SP/DIF (consumer format) and AES/EBU (pro) digital data streams operate at a frequency of about 5 MHz. I don't care what any of you say, not many people can hear up in this range, so it is considered way out of the audio spectrum. It seems intuitive then, that audio cables will not solve the interconnection problem. Yes, you can sometimes squeak by using audio cables to substitute for digital cables, but I would include this method in the same category as substituting a speed wrench (chisel and hammer) for a nice socket set.

There are provisions in the digital data stream for CRC error correction, but it is not implemented in any DAT machine that I know of. Partially this is because if there was a block transfer error, you couldn't tell the playing machine to resend the bad block. I would like to see it implemented just to show you if there are any transfer problems.

So what to do? In the case of consumer format machines with RCA plugs, use video dub cables. You can get them at any Radio Shack or Circuit City and most hardware stores. The video cables are designed to operate much closer to the digital frequency than audio cables. If your decks have optical outputs and inputs, then use them. That is the best possible solution. Besides making a great digital transfer, the machines are isolated from any ground loops.

If the transfer is AES/EBU, don't be thrown by the use of XLR connectors. This could be the most stupid thing that was allowed to creep into the AES/EBU specification. I have seen studios run 50 feet of audio cables across the room between digital machines and wonder why the signal wasn't making it. Long audio runs with multiple connections will almost guarantee problems.

Another thing to watch out for is transferring between formats. If your play machine is consumer only and the record machine is pro, there are two things to consider. One is the data format. Some machines will automatically detect the input format and receive whatever is there. Other machines, like the Panasonic SV-3700 series, require the correct format to be presented to the input connector. In

continued on page 104

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