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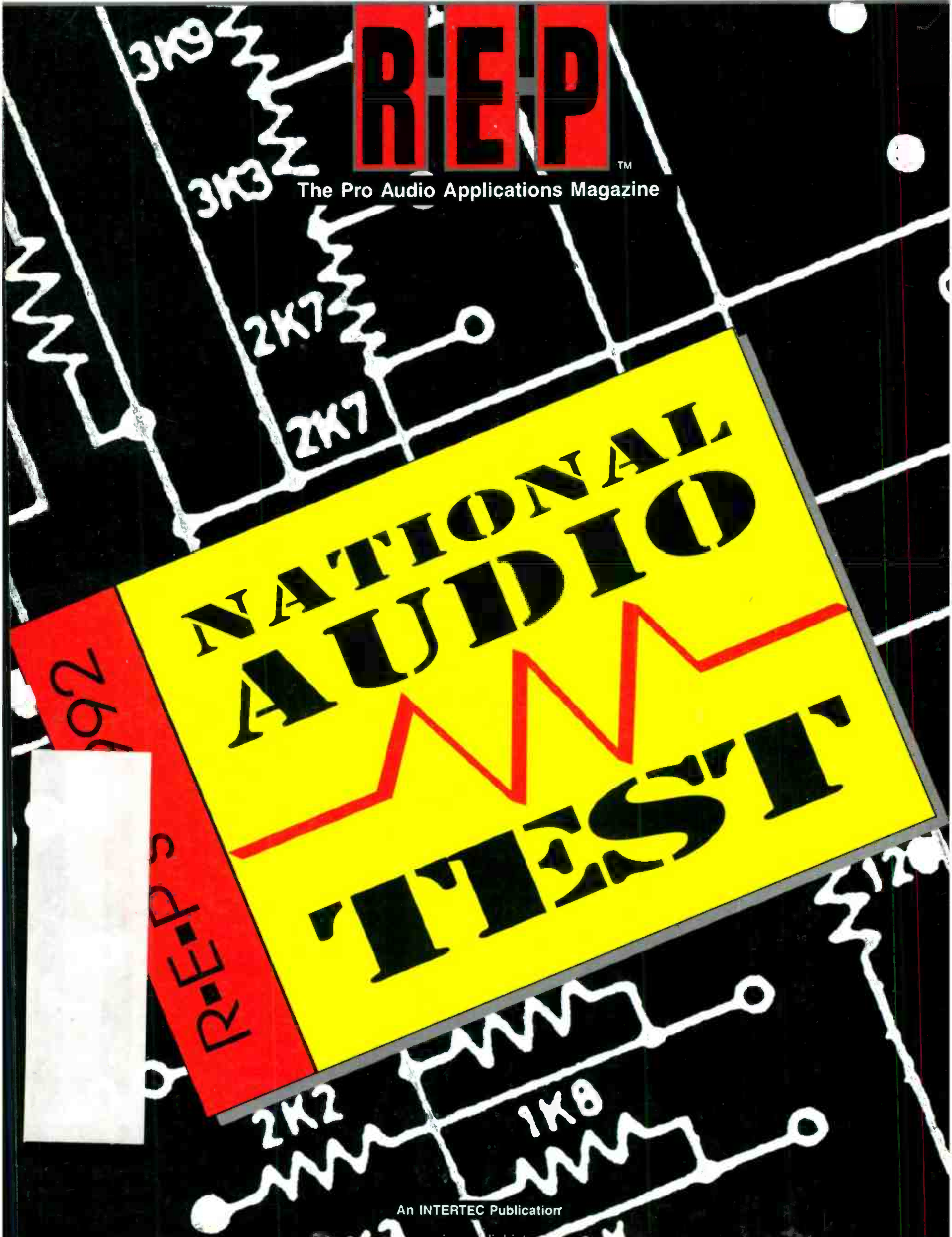
R·E·P™

The Pro Audio Applications Magazine



An INTERTEC Publication

The Most Widely Used Studio Microphones.

Take a look around the next time you're in the studio. Ask the most demanding musicians and recording engineers what microphone they use to produce the brightest and clearest sound. You'll find that most of them prefer AKG 414's for recording their music's incredible dynamic range, subtle nuance and emotion.

It's no wonder that the 1990 and 1991 *Billboard* surveys found that AKG microphones are the "#1 Most Widely Used Studio Microphones."

For years the 414 series and its AKG signature sound have been a part of recording the best music in the business. That's why the 414 is a studio classic.

AKG

Circle (1) on Rapid Facts Card

WHEN IT CAME TO BEHRINGER WE LET THE EXPERTS MAKE THE NOISE.

Leading experts in recording, film post-production and live sound are discovering the many advantages of the Behringer 2-channel and 8-channel DeNoisers. They know Behringer takes the noise out of the dirtiest signal path without altering the audio quality. Their reactions show why Behringer is now the most talked about name in professional audio circles.

"Simply lovely. Smiles all around. Room agreement was unanimous: We want this thing on all our tracks." **Mike Joseph—Editor REP, March 1992**

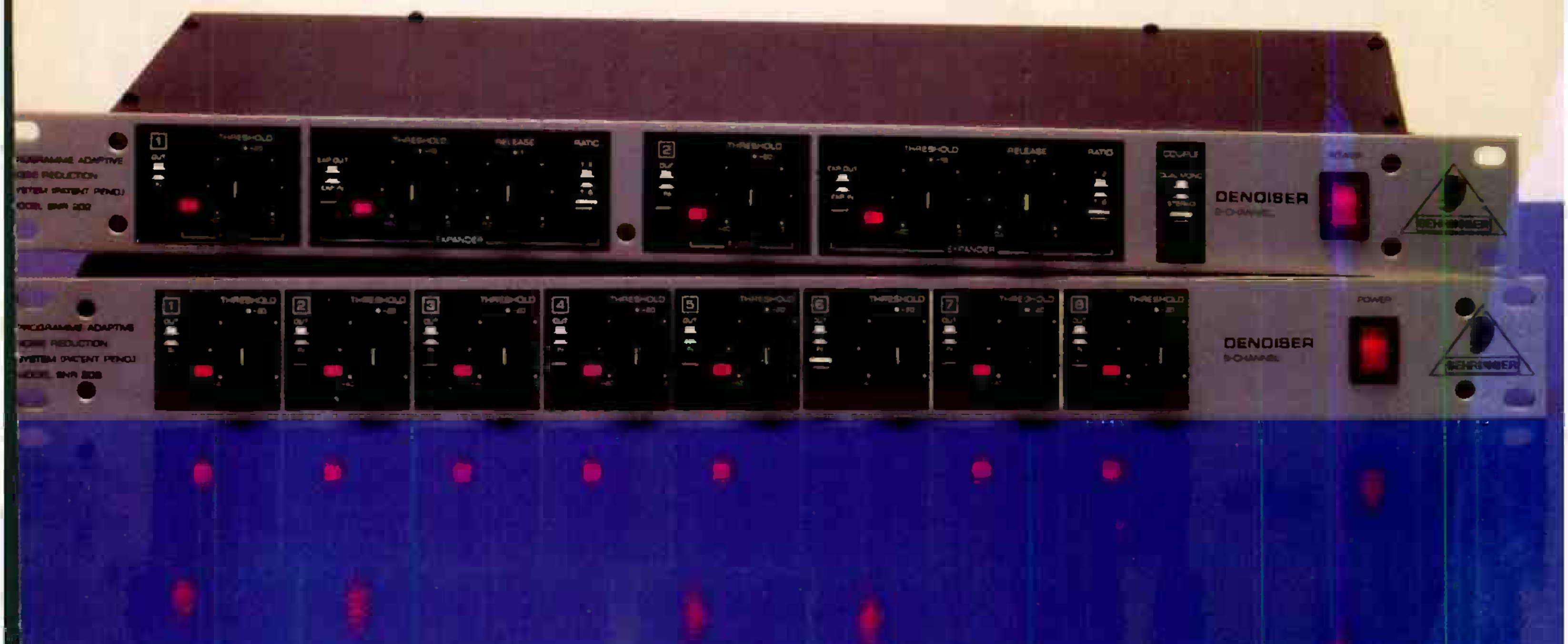
"I have used similar 'single-ended' devices on the mixes of 'Ghost' and 'Godfather III' and found the Behringer Mark III to be superior in every category—from ease of operation to final result.

"Consequently, I am—without hesitation—recommending to LucasArts/Skywalker Sound that they buy at least four channels of Behringer Mark III DeNoising for each mixing console here and in Los Angeles; a total of twelve mixing rooms."
Walter Murch—Film Editor and Music Mixer, LucasArts/Skywalker Sound

"If the phrase *noise floor* is in your vocabulary and you would prefer that it was not, get a Behringer single ended noise reduction unit to the top of your *got to have one* list." **Robert Scovill—Sound Engineer/Mixer, Rush/Def Leppard**

The experts know why Behringer DeNoisers let them take the noise out and leave the audio quality in. Isn't it time you discovered all the good things Behringer can do for your audio?

THE PERFECT EAR. BEHRINGER



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Circle (3) on Rapid Facts Card

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913-967-1711
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913-341-1300

R•E•P is an applications-based publication targeted at professional individuals and companies active in the commercial business of studio and field recording, audio for video, live sound production and related fields. Editorial content includes descriptions and demonstrations of audio production techniques, new products, equipment application, maintenance and audio environment design.

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From the Top

Space Versus Speed

It seems that every time a new technology comes along, there's a fresh protocol one has to learn to achieve functional operability. Take computer workstations, MIDI devices, or digital processors in general. Digital control of circuitry has gotten us away from analog's "one-knob, one-function" control architecture into multi-page nudge button/scroll wheel topologies and mousing about on screens.

Is this a good or bad thing? Well, if you want a lot of variability in a very small piece of real estate, you can't beat it. All the functions that you don't need to access very often can be hidden away, multiple layers down. How many times do you play with diffusion density distribution or pre-delay frequency response on a complex reverb program? Probably not often. If all your digital verb parameters were laid out like an analog parametric EQ, you'd have a panel 2-feet wide with 120 knobs on it, of which only eight would be used with any repetition.

The downside of screens, LCD windows and nudge buttons is that sometimes you *do* want to address items lurking down there on page four. Then it's menu scroll, scroll, scroll. For that reason, mixers-on-a-screen haven't gained any headway over anolog control-surface consoles. Some of the digital EQs and compressors we've seen, both realized and on the drawing board, suffer from this slow digital operation interface. It's often nice to reach out and touch some knob, tweak-in your change and move on. Elapsed time: two seconds. Ergonomically, it's all a trade-off. Space versus speed.

The important point to remember here, though, is that technological capabilities will allow further improvements in the balance between space and speed. The current breed of digital recording consoles provide reasonably sized analog-type control surfaces, with active, flexible knob "function assignments" that can remap those deeper or alternate pages onto the console-top when needed.

Those of you who know your way around video editing systems or computer workstations are already several steps ahead. Much of the technology that trickles down to audio control comes from computers and video.

But how about the rest of you? Have you kept up with all these changes? Ask yourself: have you gone to a major show — an AES, NAB, NAMM or even a SPARS conference and played with one of those systems? Even if the company checkbook says "Sit Tight," are you being fair to your future and keeping yourself educated as to the new developments in audio?

Here are some ideas:

•Get some time on someone else's system, Rent it or the room. Use a project (and/or its budget) to make it happen.
•Buy the book.
•Get involved in our CompuServe R•E•P: On-Line Forum, and ask *a lot of questions!* There are others out there who use the devices, know techniques you may want, or have access to the answers. Also, many of the manufacturers have a presence on CompuServe audio and MIDI Forums themselves.
•Join SPARS. The members are a wealth of information.
•Learn parallel technologies, i.e.: computer video/graphics design. The rate at which the technologies are crossing over is astonishing. And, it's fun.
•Buy a simple (or starter) version of the system, if it's a workstation, and grow into it, building as you go.
•Take a short course from one of the many quality schools or universities who cover the subject. If it's in a nice vacation area, bring the family!
•Hang out at pro-audio stores that allow you to demo. Make an effort to understand. Ask questions and make the salesman dig for the answers. You'll both learn. Demo the unit at home, and offer to teach the staff at the store how it works after you figure it out. Better yet, let them learn through a seminar at your studio. You can assist, plus get potential clients in your door.
• Get a collection of local audio people together and have someone from the factory come out and do a presentation, either at your studio or at a sales outlet. They'll usually be happy to. That's how they sell things!

Do your part. Be good to yourself and your future. Keep up with the new technology. Grow. Learn. If you don't get the current technology and techniques under your belt today, you'll be completely lost when the *totally outside* stuff comes at ya next year. And that's a threat! ■

Mike Joseph
Editor

Letters

Apples and Oranges

From: John Hardy, The John Hardy Company, Evanston, IL.

I must again point out problems with statements made by Jim Williams of Audio Upgrades, this time in his letter to the editor in the March '92 issue of R•E•P (which was a reply to my letter from the November '91 issue, which was written in response to the original Williams/Schwartz article in the R•E•P June '91 issue).

First, regarding the 990 discrete op-amp, he states, "As to the basic 990 design, large improvements can also be made." Part of his basis for this statement is a comparison of the National Semiconductor LM394 super-matched transistor pair that is used in the 990, and the rival Precision Monolithics MAT-02 transistor pair. According to Mr. Williams, "The input transistor, the LM394, has a noise spec of 1.8nv/Hz squared. The PMI MAT-02 is a direct replacement and offers an improved noise spec of .85nv/Hz, or about half the noise." Here is where he goes wrong:

A. Careful examination of the data sheets of both devices reveals that those noise specs were derived under *different operating conditions*. The collector current of the LM394 was 100 microamps, while the MAT-02 was 1 milliamp. If the test currents (and *all* other conditions) had been the *same*, the noise specs would have been identical. Mr. Williams is therefore comparing apples and oranges.

B. Mr. Williams is apparently implying that, by using the MAT-02 in place of the LM394, the 990 would be a quieter op-amp ("... half the noise"). Did Mr. Williams actually test both devices in the 990 circuit and find that the MAT-02 provided half the noise? Mr. Williams seems to continue to make erroneous or misleading statements. I contacted PMI to get their views regarding the MAT-02 versus the LM394 issue. After reviewing the specifications of both devices, the PMI engineer concluded that they were extremely close in both ac and dc performance.

Regarding the other alleged advantage of the MAT-02, Mr. Williams states, "It also sounds better, at least to the many people who have installed it in Sony/MCI tape machines and consoles." I cannot debate that, since I haven't heard a comparison. However, I would like to know the complete circumstances under which the comparisons were made. Were there other changes made to the circuits at the same time, changes that might have been the cause of the audible difference? Perhaps the circuits might not have been properly optimized for the LM394 in the first place (some of those designs aren't exactly brilliant), and the MAT-02 was coincidentally better. Did anyone do a thorough analysis of the circuits, before and after the modifications, to see what was really going on? Or was it more like, "Hey... I plugged it in, and it didn't blow up (yet), and I think it sounds better too! Pass the Oreos!"

C. Another aspect of the "large improvements" that Mr. Williams says can be made to the 990 is moving from 100ppm resistors to 50ppm resisitors. I am gradually incorporating 50ppm resistors into the production of the 990, but more importantly, I have special trimming procedures to adjust critical resistances to well beyond the 1% specified accuracy. Great attention was also paid to proper layout of the p.c. board to assure the tightest possible tracking of critical resistances and other parameters by minimizing thermal gradients. Meanwhile, there are other resistance values that are very *non*-critical, so it doesn't matter whether they are 1% or 5%, or 100ppm, or 50ppm, or 200ppm. *Again*, you must analyze the circuit to determine what is really important.

Perhaps in certain areas of life, "It's not the size that counts, it's how you use it!" as Mr. Williams says. However, as I mentioned in my previous letter, there are issues of power dissipation and maximum voltage levels where there is no substitution for larger size. Note that the MAT-02 Mr. Williams seems to prefer over the LM394 uses a silicon chip that is about 1/16" square, just as the LM394 does. These chips are roughly the size of an entire monolithic op-amp chip. I am certain that National Semiconductor and Precision Monolithics would have used smaller chips if they could have, but to make the best possible matched transistor pair, a "large" chip was required.

Mr. Williams made erroneous assumptions about how the MAT-02 would perform in the 990 application, bashed op-amps in general, and the 990 in particular. We deserve much higher accuracy in reporting.

Responding to Noise

From: Marvin Caesar, Aphex Systems, Sun Valley, CA.

Your review of the Behringer De-Noiser was most unfortunate in that it lends credibility to a company whose stated business plan is to copy successful products developed and introduced by other companies, including Aphex. I am not talking about about using a feature or a part of the design, but rather the entire product, including PC layout and componentry. They have even admitted in writing that they have copied the owner's manual for our 612 Gate.

Where possible, legal action against Behringer has already been instituted or will soon be initiated. Legal costs are extremely high, and, when viewed in relationship to our small marketplace, may be prohibitive.

However, it is not only manufacturers such as dbx, Rane, Symetrix and Aphex that are hurt, but also your readers – the ultimate consumers.

Products may never be developed for fear of theft. Those which are developed may not include all available innovations or may contain performance compromises to protect circuitry — the consumer loses.

Schematics and technical information become limited and service restricted — the consumer loses.

Owners of copies of products do not receive the benefit of updates and technical support from the original manufacturer — the consumer loses.

When patent rights are upheld, the purchasers of the copy may be subject to legal action (a patent being the exclusive right to manufacture, sell and use) — again the consumer loses.

If the above arguments are insufficient for some, I would ask that they imagine that one of their songs — the result of their creativity and hard work — is blatantly stolen, not just a sound or a line, but an entire song with cover art! That thought will be their guide when confronted with a choice between a rip-off and the real thing. That thought should also guide them out of a store which deals in goods of questionable origin. ■

Send letters to R•E•P, Box 12901, Overland Park, KS 66282-2901; or fax 913-967-1905. Letters must be signed and may be edited for length and clarity.

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STUDER

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Random Access

"Hello this is Paul Nado. Recently, Bay Bank of Boston took p[...] New England Digitals' assets. If you are calling with any questions rela[...] England Digital, you should try to reach Tracey Burllock at Bay Bank i[...] The number is (617) 556-6511. Tha[...]

Phone calls to NED's 603-448-5870 main numb[...] the week of July 6, 1992) led to the comptrolle[...] and the voice-mail message listed[...]

NED POWERS DOWN ...BANK PULLS PLUG

Brad Naples, who stayed on at the NED facility in Lebanon, NH on a volunteer basis, confirmed these facts: NED ceased operations on June 22, 1992. NED was neither in chapter 7 nor 11. The Bay Bank of Boston was simply seeking to sell NED assets to a "bulk buyer." Among those expressing interest, in no particular order were: **Fostex, Harmon International** and various investor groups.

"Part of the reason we (NED) got in this position was the continued recession, the New England banking environment and competition in the lower end (of the market). I feel optimistic that some form of the company, if not the whole company, will live on." – Brad Naples

BIG APPLE USERS RESPOND!

Approximately 50 users met in Manhattan – some even flew out from Chicago. A similar meeting was held in Los Angeles.

"Synclavier is the one system in the world where you can press one button and set an entire studio in motion."

– Quote from users meeting

The big questions on most users' minds at these meetings were:

1.) What is really going on?
2.) Where do we get parts and service?
3.) What can we do?

SYNCLAVIER OWNERS CONSORTIUM

Late-breaking news: R•E•P received this fax from the just-formed SOC representing the focused response of the NED user universe:

"The Synclavier Owners Consortium has been re-formed, based jointly in New York and Los Angeles. The primary aim is to provide ongoing service and support for the current users. An eastern action committee, chaired by Mike Thorne, was approved, the other members being Dave Behuniak of Magno Sound, Valerie Ghent of Ashford + Simpson and David Klein of RMI. A western committee chaired by Bruce Nazarian and Martin Royer of Gnome Productions was also approved. Membership is nearly total among active Synclavier users in the U.S."

Stay Tuned.

PEOPLE

NADY has promoted **Mike Perez** to the position of artist relation manager ... At WaveFrame, **Dan Radford** has assumed duties as chief financial officer and **Dennis Eveland** has been named as field service manager ... Engineer **Ed Cherney,** producer **Don Was,** and artist **Bonnie Raitt** were recently awarded a 3M Visionary Award for their work on Raitt's "Luck of the Draw" album ... Tannoy has selected ACS's **Paul Consalves** as their new sales rep for the regions of Ontario and Quebec. ACS Acoustical Services of Canada) is an independent sales and marketing company ... **John Scott** has joined Musical Infinities creative staff to head the Octopus service department ... Master's Workshop announced the appointment of **Tim Archer** to the position of chief engineer ... Harvestworks assistant director **Brian Karl** has been named co-director, and has also been added to the Editorial Board of *Tellus*, the Audio Arts Magazine on cassette and compact disc ... **Robert Maxwell,** a vice-president at HBO Time/Warner Inc. and Harvestworks' board member has been elected Chairperson of the Board ... **Michelle Andersen** has been named promotion manager for The Welk Music Group in Santa Monica, CA ... **Erica Reitz** has been chosen to head the international advertising and marketing department and **Kent McGuire** has assumed responsibilites for domestic advertising and marketing at Circuit Research Labs (CRL). ■

TREND WATCH

Everything you need to know

The University of Miami's College of Engineering and School of Music have formed a new undergraduate curriculum option for students wishing to study audio engineering. The Department of Electrical and Computer Engineering will administer the degree program providing a Bachelor of Science in Electrical Engineering degree with an emphasis in Audio Engineering — the first degree program of its kind in the U.S. The Audio Engineering option combines traditional electrical engineering studies such as circuit theory and electronics, with audio studies in areas such as acoustics, digital audio, transducer, signal processing, post production and recording. Prerequisite courses in areas such as calculus, differential equations, physics and chemistry are also included in the curriculum. ■

Who you gonna write (complain) to?

A program called Personal Advocate has the names and addresses of 2,500 key people to complain to, along with 40 pre-composed complaint letters.

On the list are every member of Congress, government agencies, Better Business Bureaus, credit bureaus, health and medical associations, insurance regulators, state banking authorities, child welfare groups, consumer and civil liberties advocacy groups plus the heavy hitters at most Fortune 500 corporations, Medicare and Social Security and the Department of Veterans Affairs. Personal Advocate for the IBM is available from Parsons Technology of Hiawatha, Iowa. Now, where's the Mac version for us left-brainers? ■

Just in time for Christmas

Philips Consumer Electronics and PolyGram have launched the manufacturing, marketing and consumer awareness campaign for its new Digital Compact Cassette (DCC), with the delivery date for DCC as Sept. '92. PolyGram has already opened the world's first pre-recorded DCC factory in the Netherlands; Philips has received the first shipment of digital heads; DCC industry groups have been established worldwide; the first DCC decks have been distributed to the industry and a new DCC logo has been created. Selective consumer demonstrations of the new DCC technology will begin in the third quarter of 1992. ■

STAR SEARCHERS

Six months after the British Record Producer Guild launched its hunt for new musical talent, a number of Guild members have discovered promising artists among the more than 1,000 tapes which have been received.

The BRPG's talent hunt is not a competition, it is simply an attempt by Guild members to promote new talent and encourage new and innovative ideas. By using their production skills and contacts with mainstream record companies, members hope they will be able to help talented newcomers find recording deals.

The initial sorting process is entirely random, but members have adopted a policy of referring promising tapes to appropriate colleagues. Some are well produced, but it has been noticed that a large part of the work is badly recorded and/or packaged. Many tapes are arriving without SAE's, photographs, or information about the band. Many have been received with only the tape and a telephone number. So, okay, the business side is weak. But did you hear that guitar solo? ■

Random Access

STUDIO UPDATE

Facility/Location	Details
NORTHEAST	
Bear Tracks/Suffern NY	Finished upgrading its SSL E Series console with a SRI G Series studio computer and software.
Hit Factory/New York City	Purchased a Neve VRP72 with Flying Faders.
SOUTHEAST	
River City Productions/Memphis	Recently installed a new Tascam 3500 series 24-input console.
MIDWEST	
Ron Rose Productions/Southfield, MI	Placed an order for an AMS Logic 1 SPECTRA workstation.
SOUTHWEST	
San Antonio Shoe/Boerne, TX	Purchased a Neve VR60 with Flying Faders Automation.
SOUTHERN CALIFORNIA	
Goodnight L.A./Van Nuys	Added a second studio, L.A. "B," which is a 24-track digital room with MIDI.
Capital Records/ Hollywood	Added a Neve VR72 with Flying Faders.
NORTHERN CALIFORNIA	
Focused Audio/San Francisco	Acquired a 16-output AudioFile Plus digital editing system.
NORTHWEST	
Digital One/Portland, OR	Recently installed an SSL ScreenSound.
CANADA	
Pizazzudio/Ontario	Acquired a Studer A827 multitrack recorder, a D&R Avalon console with Optifile, an Apogee AD500 converter, Demeter tube mic preamp and tube direct box and Genelec 1031A loudspeakers.
CARIBBEAN	
Music Works/Kingston, Jamaica	Began construction on a new facility featuring an ARcoustics architectural design.
GREAT BRITAIN	
Berwick Studios/London	Has added 62 channels of Uptown Automation's 2000 moving fader console automation system to its AMR 24 console.
EUROPE	
Rádiotelevisao/Lisbon	Has purchased one 16-channel and two 8-channel DAR SoundStation Sigmas.
FAR EAST	
Synchrosound/Kuala Lumpar, Malaysia	Has taken delivery of a Neve VRL 60 with Flying Faders.
DESIGNERS	
Walters-Storyk Design Group/New Paltz, NY	Has completed new digital post production facilities for Henniger Digital Audio (Arlington, VA), including a matched pair of Lexicon Opus suites.

NEWS NOTES

According to its founder, Dr. Christopher Jaffe, **Jaffe Acoustics, Inc.** has been reorganized. Dr. Jaffe's two senior colleagues, Mark A. Holden and Paul H. Scarbrough, have been named full partners in the company, which will be called Jaffe Holden Scarbrough Acoustics, Inc.

Musical Infinities, a San Francisco-based recording studio, has opened an integrated recording studio, creative services and equipment rental business called "Octopus." The service allows clients to rent Mac-based workstations for off-line prep- and pre-production tasks before finishing projects on-line at the Musical Infinities facility. Musical Infinities also recently opened a new foreign language and translation department.

Roadworx, headquartered in Greensboro, NC, has opened their fourth U.S. office in the Atlanta suburb of Smyrna, GA. The company which specializes in touring audio, lighting, video and staging, services entertainment and corporate clients.

The **Crown** DC-300, generally accepted as the first high-power solid-state amplifier, is celebrating it's 25th anniversary. The amplifier is still available as the DC-300 A Series II.

The **AES of Great Britain** is sponsoring a 2-day conference from Sept. 14-15 at Kensington Town Hall, London, England, featuring 18 experts in the field of DSP for Audio. Sessions include fundamental issues, filter design and topology, code generation, DSP and psycho-acoustics, acoustic environment correction and control, pro and consumer applications and audio signal restoration. For information call 0628-663725; fax 0628-667002.

PUBLICATIONS

The new book **"Practical Recording Techniques"** provides coverage of digital tape recording, hard-disk recording, keyboard and digital workstations and MIDI. Authors Bruce and Jenny Bartlett cover such topics as the basics of sound and the tools for recording including studio setup, proper acoustics and recording equipment. The is book available from Sams; 800-428-5331. ■

By Dan Levitin

Bronx Style Bob: "Grandma's Ghost"
Label: Sire/Warner Brothers
Produced by: Bronx Style Bob, Carmen Rizzo Jr., John Myles O'Brien, Roy Campanella III, Sam Catalona, Mike Hightower, Jeff Connor, Danny Saber, Kenny Villenuve, Skate Master Tate, Matt Hyde
Engineered by: Carmen Rizzo Jr., Alan Blazek, Danny Saber, Sean Freehill, Kalique Glover, Dean Burt, Sam Catalone, Mike Hightower, Kenny Villenuve, Sean Freehill, Ramsey Embick
Recorded at: Power Trax, Studio 55, S.S.R., California Recording, Kitchen Sync, Sound Castle, Chappell, Beat Street, Tempo, Image

Comments: One of the most interesting and exciting new artists to cross my CD player this year, Bob is something of an anomaly. Blurring the distinctions between rap, R&B and pop, he blends these influences skillfully, sometimes even within one tune. BSB co-produced all the tracks with great consistency, and by the sonics, you'd never know so many different producers were involved. (I consider this a plus.)

Of special interest: The smoothest tunes are those co-produced by Carmen Rizzo, such as "Forbidden Love," "I'll Be There For You" and "Children Play." Perhaps what appeals to me most about Rizzo's tracks are his melodic use of guitars and the instruments he tucks into the corners of the mix. "Children Play" features a gypsy-style violin throughout, and a guitar solo that sounds like something from "Axis: Bold As Love." Acoustic guitar leads in the intro and throughout lend a classical quality to the song.

FOCUS:

CARMEN RIZZO JR., Producer and Engineer, "Grandma's Ghost"

R•E•P: How do you get drum machines to sound live?
CR: The first thing is the drum programming — I try to get a live feel. A very important thing is the panning of the drums — wide panning isn't the answer and mono obviously isn't the answer either. I also try to incorporate live samples — I use the MPC60 and the Forat F16. The guys from LinnDrum, Ben and Bruce, own their own company and make the Forat, a 16-bit drum sampler. It's a very common box that a lot of people use.

Also, I tend to mix dry, I don't like too much reverb or effects usually. The kinds of records I listen to — older- style records — don't have a lot of tricks involved. Too many records today are too compressed and too wet for me — I think drier and simpler mixing is my style.

R•E•P: When you talk about old style-records ...
CR: I mean like Sly Stone, older soul records, old Marvin Gaye — those are the records I tend to listen to more. It's wierd because people sometimes think you have to be a flash mixer to be a good mixer when you really don't. A lot depends on the song and just making sure the song comes across on the record. One of my favorites is Bill Bottrell (see the June 1992 R•E•P Interview for Bottrell). I had a chance to work with him, too.

R•E•P: What did you learn?
CR: To kind of get your chops down as a mixer; by chops I mean balancing, and tending to mix dry. I wasn't a second engineer that long, unfortunately, so I kind of assisted for him and also co-engineered with him. Susan Rogers is another favorite — she did all those Prince records. Between the two of them, listening to their stuff and talking to them, I learned a lot about mixing.

They always tend to do records I like. I think being a mixer is like being an actor or a model — people tend to judge you by the product.

R•E•P: How do you record vocals?
CR: I always use a dbx165A limiter as I'm recording. With EQ I might add some top and take out some low. On Bob I used an AKG "The Tube." When recording I try to get as much of the sound to tape as I can.

R•E•P: How involved were you with the arrangements?
CR: We changed them around a bit. The instrumentation was done from scratch. That's one thing I'm kind of proud of — the instrumentation is very important to me. We used vintage instruments — a real violin in "Children Play," a Wurlitzer, real bass on all the songs. On all the records I do there's pretty much a posse — Allen Kamai on Bass, Chris Bruce on Guitars and John Barns on Keyboards, including synclavier strings.

R•E•P: How did you mix?
CR: I mixed on SSL to analog 2-track and to DAT; 30ips on 996 +6 and I usually go to two DATs at the same time. This album was cut analog but I usually go 24 digital and 24 analog. I like the mixing technique of "push up the faders and it's time to go." ■

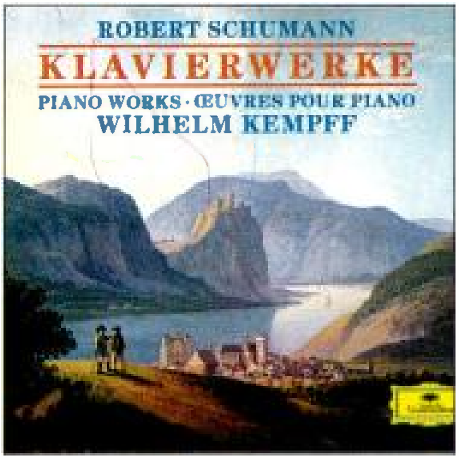
Dan Levitin is R•E•P's music production editor.

Fresh Tracks

Ugly Kid Joe: "As Ugly As They Wanna Be"
Label: Stardog/Polygram
Produced by: Ryan Dorn and Ugly Kid Joe
Engineered by: Ryan Dorn
Mixed by: Ryan Dorn
Recorded at: NRG Studios (Los Angeles)
Mixed at: Summa Studios (Los Angeles)
Mastered by: Eddy Schreyer at Future Disc

Comments: All the high-budget hard rock production tricks are employed on this 5-song EP with a sense of humor and spirit: doubled electric guitars, solo lead guitar doubling lead vocal line for emphasis, full band breakdowns punctuated by blistering lead fills and choog choog driving rhythm guitars. Propelled by the hit "(I Hate) Everything About You," UKJ is getting some well-deserved attention. Crane is a vocalist with good control and delivery, and the rhythm section is tight.

Of special interest: The instrumentation is the standard two guitars, bass and drums. But close attention to the arrangements and some clever planning keeps the songs interesting. The arrangements have obviously been worked over so that not everyone is taking fills at once, and when someone does take a fill, it fits perfectly into the song as a whole. The lyrics avoid the usual hackneyed approaches to cars, drugs and women by putting on a new spin, i.e., "Madman" is about a psychopath in Disneyland who wants to administer LSD to his victims. The CD booklet graphics feature a scruffy version of Nipper urinating on a Victrola. (Notice they're not on RCA). ■



Wilhelm Kempff: "Schumann Piano Works"
Label: Deutsche Grammophon
Produced by: Cord Garben, Manfred Richter, Rudolp Werner
Engineered by: Klaus Heimann, Klaus Scheibe, Hans-Peter

Maurizio Pollini: "Chopin 4 Scherzi, Berceuse, Barcarolle"
Label: Deutsche Grammophon
Produced by: Werner Mayer
Engineered by: Günter Hermanns

Comments: Asked to name the most difficult instrument to record, many engineers single out the grand piano. The fundamental problem is getting a recorded piano to match the sound heard in one's head while sitting in the room with the piano. This turns out to be insidiously tricky. The instrument itself is very large and the sound emanates in every direction, from the soundboard toward the floor, from the sides, front and back, and from the string bed upwards, reflecting off the partially opened lid and back into the soundboard again. In popular music, where the piano is accompanied by other instruments, the sound quality is not so incredibly crucial. The recording of solo piano is truly the test of engineering skill and artistry.

These two recordings represent very different approaches to the recording of solo piano, different production styles and sonic conceptualizations. Though different, almost opposite, they are both superb examples of solo piano recording technique married to extraordinary and sensitive performances.

The sound of Kempff's Schumann is warm and rich, encompassing the darker tonal shades. The recording is at once haunting and delicately surreal in performance and tone. Scheibe's 1973 recording of the Kinderszenen does become slightly distorted in some of the louder passages, but this is only momentary and not a serious detraction. The soft passages are recorded far from the noise floor and sing beautifully.

Hermanns' recording of Pollini captures a much brighter piano sound, with no corresponding lack of warmth. The tone itself feeds the playfullness of Chopin's Scherzi, the ambiences all surely organic. The small amount of noise heard in some of the louder passages (on Scherzo No. 2, for example) may sound like distortion introduced by the recording process, but are far more likely the sounds of the piano itself vibrating and the strings buzzing with the intensity of the performance.

The stereo imagining on the two recordings is quite different as well. Where Kempff's piano is rendered in luxurious ambience washing over the stereo field, Pollini's instrument is presented with perhaps greater spatial accuracy: Chopin's rapid multi-register runs dance through the stereo field with the sort of naturalness and majesty one might expect to hear sitting right on the piano bench. ■

Lyle Lovett: "Joshua Judges Ruth"
Label: Curb/MCA
Producers: George Massenburg, Billy Williams, Lyle Lovett
Engineered by: George Massenburg, Nathaniel Kunkel
Mixed by: George Massenburg
Recorded at: Ocean Way, Conway (Los Angeles)
Mastered by: Doug Sax, Alan Yoshida, Ron Lewter at The Mastering Lab (Los Angeles)

Comments: It is recordings such as this that make being a reviewer worthwhile. "Joshua" has clarity, spaciousness, depth and sponaneity. And it was recorded predominantly live to multitrack! Lovett's vocals are captured with great attention to detail so that every nuance comes through, creating an intimacy not often found in recorded music. The heavy compression that Massenburg says he used is not sonically evident. The songs are wonderfully eclectic, stunningly reproduced by one of the greatest engineers of our time.

Listeners who focus on the tones of individual instruments may argue that the individual tones are not "amazing." Alright, so maybe the piano's a tad bright. The real brilliance here is how everything hangs together to create a package that is better than the sum of individual parts, and that is first and foremost, musical. Where every minor detail adds to the total soundscape. A skillful, artful marriage of songwriting, playing and production values.

FOCUS:

GEORGE MASSENBURG, Producer and Engineer, "Joshua Judges Ruth"

R•E•P: I thought we might talk about the Lyle Lovett record.
GM: I gotta tell you, I really hate going through the "I use this microphone" and blah blah blah. I think it's pointless. What people generally do when they read about what kind of equipment someone uses is — they end up buying something expensive that really doesn't help at all.

R•E•P: You mean because ultimately it's in the ears ...
GM: Right. I would love to talk about the engineering side of the album, but if it runs the danger of being "I use this number of such-and-such microphones" because, hey, it looks like I've got a promotion deal with a manufacturer, then I'd really have to decline. Someone was walking through a Little Feat session I did at Conway once and saw the keyboard talkback mic that I had on Bill, a 56 that had been pushed over into the corner. And they were oohing and ah-ing and just looking at this mic and saying, "That's how he gets that drum sound!" That's certainly wrong.

R•E•P: So, I won't ask you what mics you used. But I'm interested in how you miked the piano.
GM: It's a technique I've used for years. It's MS (mid/side sum and difference) done with a custom-capsule C24. We used the 7-foot Steinway in Ocean Way Studio A — it's a very good piano. That is most important. It's pretty bright, you don't have to reach for it. And the player, Matt Rollings, is great.

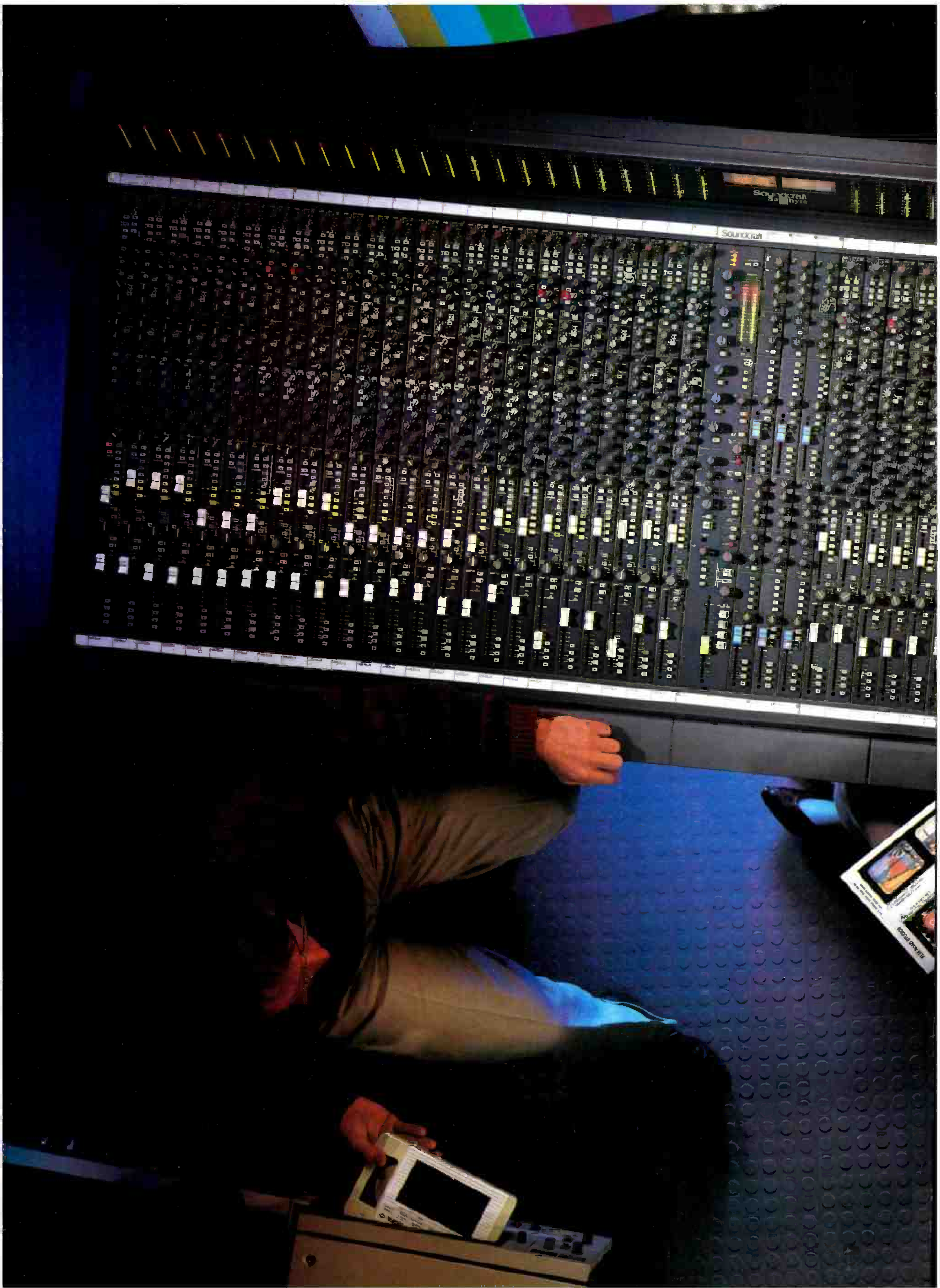
R•E•P: Was the lid on or off?
GM: The lid was not only on, but because we recorded with musicians sitting physically as close as possible to each other, we had a lot of blankets on it and baffles around it. It's amazing that it came off.

R•E•P: On "She's Already Made Up Her Mind," you get a very atypical rhythm guitar sound — it's very dark.
GM: That's a lot of things. Nathanial Kunkel and I kind of meander until we get something that strikes a nerve. Now, Lyle drove this session in the sense that he knew what he didn't want. The sound of that guitar is probably in the way Lyle works the mic against what he was hearing in the phones. Because it was live, he was singing and playing quite close to the mics.

R•E•P: Do you use any kind of baffling between his chin and his guitar?
GM: No, but we've been there. Doesn't work that well. Nobody's been through it more than Peter Asher and James Taylor, who've tried every known combination of plexiglass and Saran Wrap and what-have-you. But no, for me, nothing.

R•E•P: The overall sound is very clean.
GM: I guess that's from Lyle's and my real lust for doing the stuff live. Almost everything is live except for a couple of one word vocal fixes, some solos and a few backgrounds. The contributions of the individual musicians (Russ Kunkel, Leland Sklar, Dean Parks, Ray Herndon, Matt Rollings, et. al.) were selfless and trancendent — the guys were so careful to meld with Lyle and his tunes. Everything was done through external GML mic pre's, EQs and compressors. The bass went direct, right to tape, through either an Alembic or a Demeter direct box.

R•E•P: Do you have your own wire?
GM: Nah, I'm careful but I'm not crazy. ■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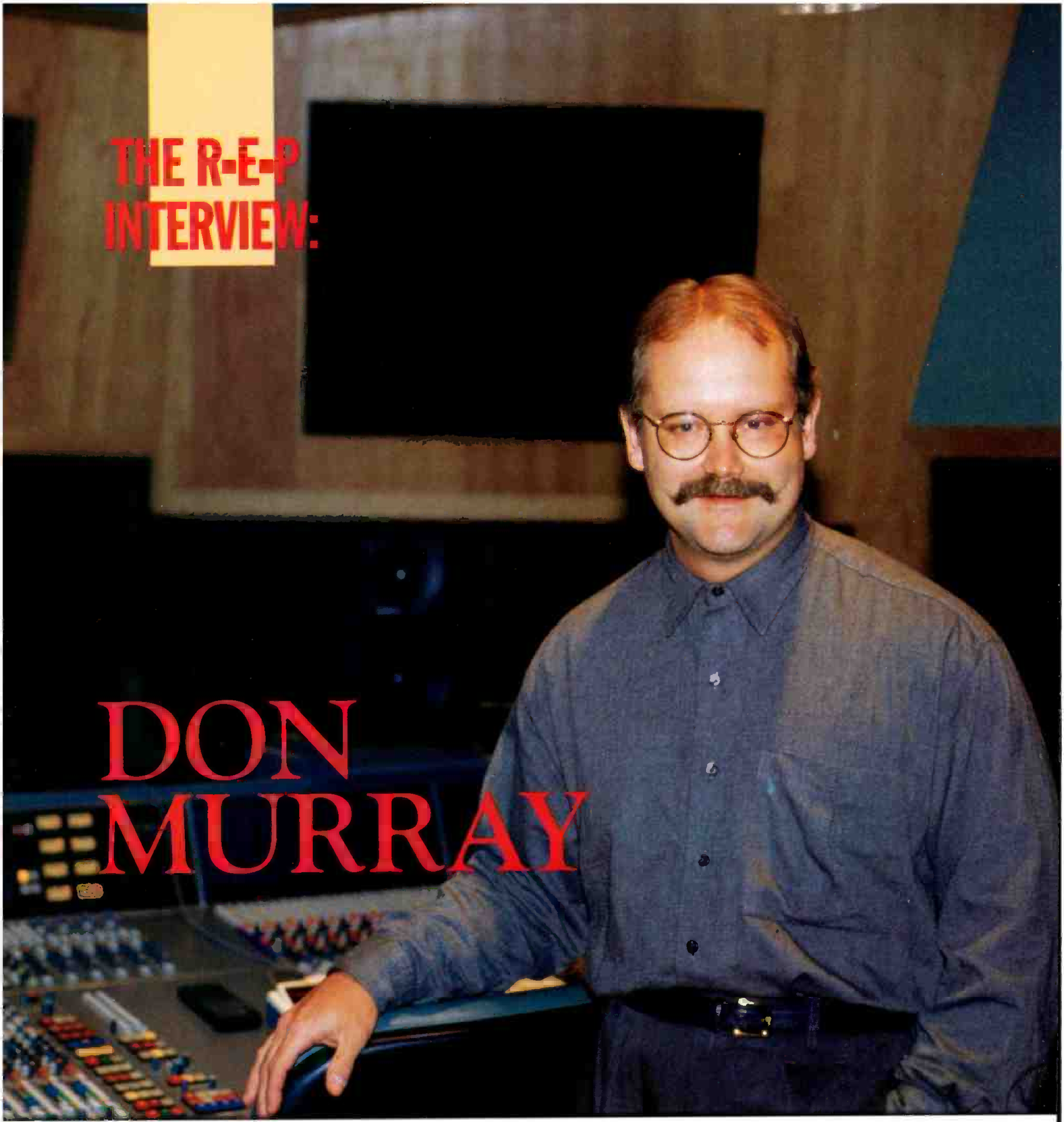
THE R-E-P INTERVIEW:

DON MURRAY



By MAUREEN DRONEY

It's a typical Grammy listening party for the Best Engineered Album Award. CD after CD and, yes, cassette after cassette are given a 10-second listen and a quick yank. (Sad but true, some record companies actually send cassettes to be judged for sound quality!). Thumbs down circle the room. Engineers are a notoriously picky bunch when it comes to analyzing other engineers' work. And the well-known and well-respected engineers who sit on the

Maureen Droney is a production manager and engineer based in Los Angeles.

A technically revealing conversation with one of the industry's most respected recording engineers.

Grammy selection committees are even pickier. So when someone receives seven nominations for Best Engineering, it's time to sit up and take notice.

He may not be as well-known as Bob Clearmountain and Tom Lord-Alge, but he should be. He's repeatedly praised by his peers, the people with the golden ears, and he owns a contented client list a mile long. His name is Don Murray.

Murray got his start in Philadelphia where he worked with legendary R&B producer Thom Bell on records of such greats as the Spinners and the Stylistics, and that influence is apparent in his current work. Although Murray currently specializes in jazz, he's not into the overly reverent jazz of some, recorded exclusively with 30-year-old mics and no EQ. Using modern production techniques and exquisite taste, jazz recorded by Murray grooves as hard as any New Jack Swing track. He's a master of live recording, as evidenced by the GRP Records disc called "Fourplay," currently breaking records with 25 weeks in the No. 1 slot on the Billboard Contemporary Jazz Chart. R•E•P booked Murray for an interview at Andora Studios, a new Los Angeles facility where he's been spending a lot of time.

R•E•P: How did you learn engineering?

Don Murray: I was in music school in Philadelphia, and I had a friend who was a second engineer at Sigma Sound. Things got busy and they needed another second. They asked me, and that was it for music school. The first day I worked a double shift. One of them was a rhythm section for Gamble and Huff, the Delphonics or somebody. It was the early '70s and Philadelphia was just taking off. It became like a factory of producing R&B records. I worked with Joe Tarsia, a great engineer with great ears. My first big client as an engineer was Thom Bell, who was just becoming very hot. The first session I did for him was the Spinners' "One of a Kind Love Affair," which became a No. 1 record. From then on I worked with him on all of his stuff. I was still playing then, so I played some guitar on those records. I'd hear a part, and at night after he'd gone home I'd lay it down. Often he'd like it and keep it. And, if the piano player wasn't available, I'd sit in with the rhythm section and play piano. But after a while I got too busy as an engineeer, and with no time to practice, I stopped playing pretty much.

R•E•P: Although you've had seven Grammy nominations for Best Engineered Album, the Grammy award that you actually won was for production...

DM: It was a live recording done in Japan. Another GRP Records extravaganza! As far as I'm concerned, there wasn't a whole lot of production involved. It was basically "get the best sound you can" on one show, with a very limited sound check. It was great to win an award for that, but it would be nice to win the engineering award eventually. But really, it's a committee of engineers who make those nominations, and it's an honor to be nominated by my peers. Just to have all those nominations means a lot to me.

R•E•P: I think there are a lot of engineers out there who would like to hear real specifics of how you track a record.

DM: I have a process of doing things, and use certain mics that, over the years, I've decided I like and will start off with, but I'm very open to change and to trying things. I'm not one of these people with a notebook full of

DON MURRAY DISCOGRAPHY

THE SPINNERS
- "The Spinners" Atlantic
- "Mighty Love" Atlantic
- "New And Improved" Atlantic
- "Pick of the Litter" Atlantic
- "The Spinners Live" Atlantic
- "Happiness Is Being With The Spinners" Atlantic
- "Best of the Spinners" Atlantic

JOHNNY MATHIS
- "I'm Coming Home" CBS
- "Sweet Love of Mine" CBS

THE STYLISTICS
- "Rock and Roll Baby" AVCO

ELTON JOHN
- "The Thom Bell Sessions" Rocket

THE JACKSONS
- "Destiny" Epic

DENIECE WILLIAMS
- "My Melody" CBS
- "I'm So Proud" CBS

LEE RITENOUR
- "Captain Fingers" Epic
- "The Captain's Journey" Elektra
- "Feel The Night" Elektra
- "Friendship" Elektra
- "Rit" Elektra
- "Rio" JVC/Elektra Musician
- "Rit II" Elektra
- "On the Line," Direct to DiscJVC/Elektra Musician
- "Banded Together" Elektra
- "Earth Run" GRP
- "Portrait" GRP
- "Festival" GRP
- "Color Rit" GRP
- "Stolen Moments," Direct to Disc GRP

FOURPLAY
- "Fourplay" GRP

DAVE GRUSIN & LEE RITENOUR
- "Harlequin" **1985 Grammy Nominee GRP Best Engineered Album

Continued on next page

DAVE GRUSIN
"Cinemagic"**1987
Grammy Nominee GRP
Best Engineered Album
"The Collection" GRP
"Migration" GRP
**1989 Grammy Nominee

"The Fabulous BakerBoys"
soundtrack GRP
"Havana"
soundtrack**1991Grammy
Nominee GRP
BestEngineered Album

"The Gershwin Collection"
GRP

GRP VARIOUS ARTISTS
"GRP Live in Session"
**1986 Grammy Nominee
GRP
BestEngineeredAlbum

"GRP Superlive"
**1988 Grammy Nominee
GRP Producing Best R&B
Instrumental

"Happy Anniversary,
Charlie Brown" GRP
**1989 Grammy Nominee
GRP Best Engineered
Album

"GRP All-Star Big Band"
Album GRP

HIROSHIMA
"Third Generation" CBS
"Go" CBS
"East/West" CBS

DIANNE SHUUR
"Shuur Thing" CBS
"Timeless Album" GRP
"Diane Shuur & the Count
Basie Orchestra" GRP
"Talkin' 'Bout You" GRP

ERNIE WATTS
"Sanctuary" Quest

DON GRUSIN
"Raven" GRP
"Zephyr" GRP

PATTI AUSTIN
"Love Is Gonna Getcha"
**1990 Grammy Nominee
GRP
Best Engineered Album

NANCY WILSON
"With My Lover Beside
Me" Sony ■

exactly the way to EQ every microphone. I just sort of take the moment as it comes. I like being spontaneous, and not thinking about it too much beforehand. I go with what the musicians in the room are doing and what the song is like. I also do a lot of listening in the room before I record something. I have an idea of mics I like on certain instruments, but before I listen to the mics I go out and listen in the room. I listen to the bass drum, the snare and how the drums feel. If it's not sounding right, then we'll change them or try to find another part of the room that feels better, that projects more.

I start with a Sennheiser 421 on the bass drum. It has a nice attack to it, and it doesn't have such a big bottom to it that it comes off too puffy sounding. I usually end up putting bottom on the kick drum and taking low mids out, because the low mids are the part of the sound that will distort a speaker.

R•E•P: Like 300Hz?
DM: Yes, 300, 400 cycles. And then I'll add low end, 50, 100 cycles, and then maybe put a boost at 7 to 10,000 cycles to add even more attack, depending, of course, on what kind of song it is. You have to go by instinct, on how the bass will interact with the bass drum. But generally, I find with the 421 that you have to notch out some low midrange, and then stretch the high end and low end around it.

R•E•P: You get fat, punchy bass drum sounds. Do you usually use compression to get that?
DM: Yeah, I compress bass drums to begin with during recording, and I'll gate them also. I like to use dbx 165 series compressors. They're my favorite for bass drum.

R•E•P: They have attack and release controls.
DM: Yeah, but they have an auto function also, which I usually use. It sounds fatter to me for some reason. It just comes from experimenting, but that's one thing I've found about the 165s, with the auto attack and release in I like the sound better. I've been using Aphex gates lately on the kick. I find that they are very smooth and transparent; you have a lot of room to tweak them and the VCAs sound better than some other gates. I'll either use GML EQ or an API 560 graphic. The API 560 graphics are great for bass, kick drum and snare, and I like to use them when I'm recording.

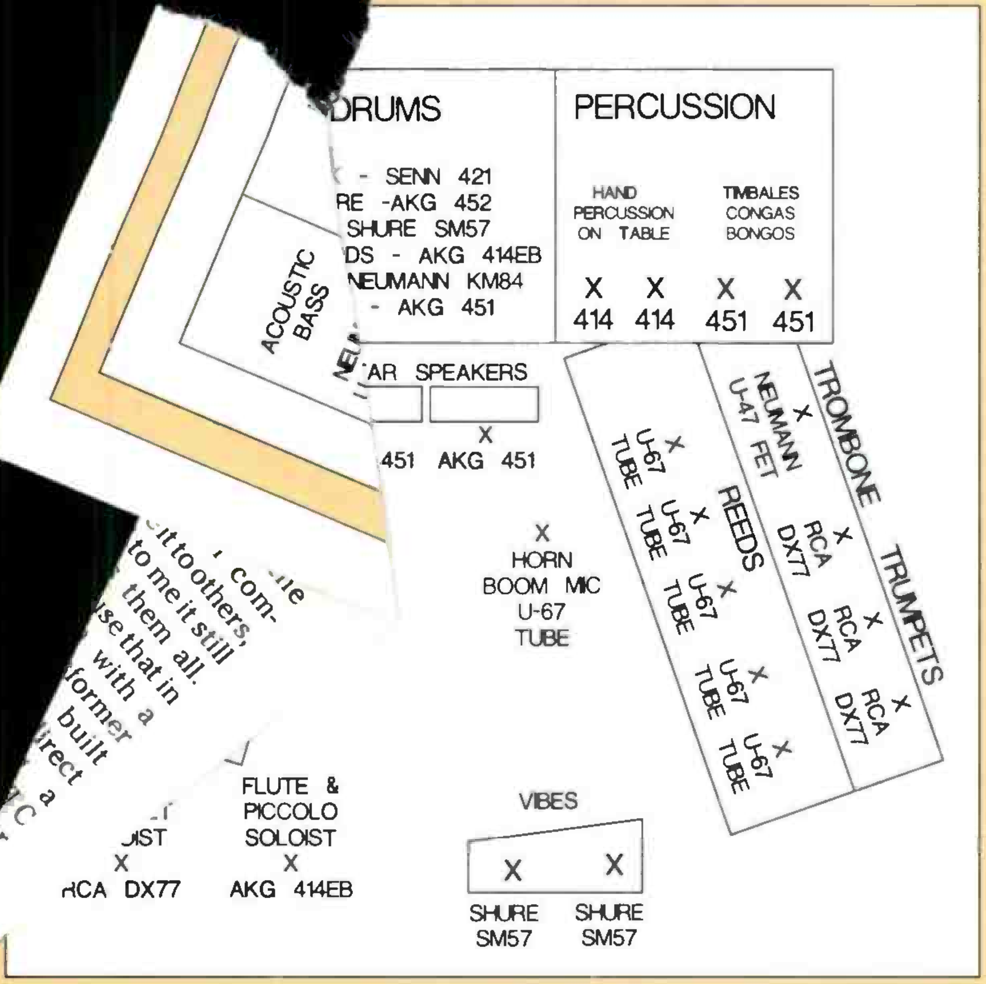
On snare drums I usually use a dynamic mic on the top, like a Shure SM 57, and I'm a fan of AKG 451s or 452s. I'll use one of those on top also, and I'll blend the sound of the two. You get more crack out of the 451, but it doesn't have as much punch as a Shure 57, so I'll blend the two together. I'll usually put a bottom mic on; sometimes I use it and sometimes I don't, and it'll usually be a Shure SM 57. Depending on whether I want to hear more snares or not for that particular song, I'll blend in a little of that. I gate the snare also, probably fairly loosely, because often the drummer will play rolls or cross stick, and you don't want to lose that. I might gate it again in the mix, and fine-tune it then, so that I don't lose anything. Again, I like the API 560 on the snare.

I'll bus all the mics that I'm using to one track, and I'll run the group output throught the API 560. I really like their sound. They are not at all masked-sounding. They have a nice presence, and a nice hype, a very musical high end, and a good sounding low end. They are graphics so you can really tweak things in and see what you are doing. And you can notch out at certain frequencies, so you can really make musical adjustments to the sound of those instruments.

I like to be open to trying new things and realizing that it's a whole new day, a whole new project starting, and maybe there's something new we can get out of it.

On the hi-hat, I usually go for the Neumann KM84. It's not as bright as some of the other condenser mics, but it has a meatier quality to it that I like. I don't like it when the drummer hits the hi-hat cymbal and it screams at you. I like more of the midrange quality of the hi-hat, and then I'll brighten it up if I need to, or I'll take out low end, because sometimes you get a bass buildup around the hi-hat that is a little annoying. I'll use a high pass filter to get rid of some of that.

I'll usually use two overheads that are fairly close to the cymbals, and they'll be AKG 414 EBs without any rolloffs or anything. I typically spread them out about three or four feet and angle them down over the cymbals, and I don't have them



RECORDING THE GRP ALL-STAR BIG BAND ALBUM

For you engineers tired of overdubbing one instrument at a time, here's a recipe guaranteed to alleviate boredom: Take 20 top-flight musicians set up in big band formation, change players, arrangers and instrumentation for each song; have the horn section and percussionist change instruments during each song; add one video crew, miles of extra ac cable and lighting snakes in both the control room and the studio; and mix it all together to get what engineer Don Murray laughingly calls "a recording nightmare."

GRP Records did just that when it decided to celebrate its 10th anniversary by recording a big band record containing performances by as many GRP artists as possible. Compiled from two shows recorded live to video and to 24-track in one day, the result sounds great.

Just in case you find yourself in a situation like this, here's a diagram of how Murray set up the session at Los Angeles' Ocean Way Studios for the GRP All-Star Big Band. ■

Don Murray: "I did a rather historic live performance album for GRP in January, an all-star big band record. It includes a lot of tremendous artists, and we cut it live by recording two shows, which were also being taped for a video. It was recorded at Ocean Way, and the band was set up very tight, in a traditional big band setup with no baffles or anything. Because the cameras had to be able to get everyone in the shots, I had very little room to set up in with 20 some guys.

It was an experiment in leakage. The best players in the world, Dave Grusin, Kenny Kirkland, Russ Ferrante, David Benoit, different players for different tunes. Dave Wechtel on drums, John Patituchi on upright bass, Alex Acuna on percussion, Gary Burton on vibes, Lee Ritenour on guitar, Eddie Daniels on clarinet. The horn section was Tom Scott, Bob Mitzer, Eric Marienthal, Ernie Watts, Arturo Sandoval, Randy Brecker, Sal Marques. Sax players on the front riser, trumpets right behind them. I had to put all the horns on separate tracks,

PHOTO CREDIT: Harrison Funk

because they were changing instruments during almost every song! The reeds played sax, flute and oboe, and there were English horns and muted trumpets. The leakage really worked, but that way I had enough control in the mix. ■

too high off the cymbals — maybe three feet.

R•E•P: You get a lot of stereo spread in your drums.
DM: Yeah, I like it so that when the drummer hits the cymbals on the left, you really hear them on the left. I do that with the toms, too. I like stereo.

R•E•P: Do you use both top and bottom mics on toms?
DM: No, I just use AKG 451s or 452s, which have a nice attack, and I usually wind up putting low end on.

R•E•P: You get a lot of powerful low end on toms, but they never sound muddy. How do you do that?
DM: Well, I basically stretch out the sound of the tom, adding low end and high end. I'll also put up a directional dynamic mic next to the 451 on each tom and I use those mics as trigger mics for gates. I use gates on the toms, and if I used the sound coming through the 451 to trigger the gate, then what would happen, since I'm EQing a lot, is that each time the drummer would hit a cymbal next to the tom, it would open the gates, and I hate the sound of cymbals going through EQ'd tom mics. So I just use the dynamic mic, and I drastically EQ it, rolling off the high and low end and boosting midrange, so I'm just using the attack for a click that will trigger the mic through the external trigger input of the gate. It doesn't open when he hits the snare or the cymbals, and it gives me a lot of control over the sound of the toms. It helps a lot to give me a nice tight sound on the drums. It also helps in being able to use the overheads as your total overhead sound on the drums. Of course, you can't do this if it's a ballad with brushes on the drums!

If I need a room sound I'll usually use Telefunken 251s. I'll go for a nice vintage tube sound, but I don't back up my overheads, because I still want that presence on the cymbals. If I need a room sound I'll just add other mics and record them on separate tracks. Then I have control later. Of course, if they want a gated room sound, I'll create it then, triggering the room mics to open from the kick, toms and snare.

R•E•P: Are most basses that you record taken direct?
DM: Yes, I like the sound of the direct bass, and I have a couple of direct boxes of that I use on the bass that I like the combination of.

R•E•P: Your own direct boxes?

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great. It really has a nice punch to it, and I'll wind up using both the Fat Box and the transformer, two separate inputs into the console. I'll combine them to one track and, depending on what I need, I'll vary the amounts in the combination. On the "Fourplay" album, with Nathan East, we just used the transformer.

R•E•P: It sounds as if you set yourself up to be really flexible.
DM: Well, I have to be. What works in one studio sometimes doesn't work at all in another one.

R•E•P: Do have a favorite bass limiter?
DM: The dbx 165 when I'm recording. The Inovonics 201 is also very good. I don't like Neve compressors as well for bass or kick. They seem to take too much of the attack away. The same thing with tube compressors. I wouldn't normally use them on a bass or a kick. They're just too spongy on those instruments, although I use tube compressors and Neve compressors on a lot of other things.

R•E•P: You've worked with pianist Dave Grusin a lot. What mic setups for piano have you developed?
DM: In general I go for AKG C12

tube mics, two of them, not too close to the strings, and I don't put them together in the old stereo fashion. I like to open up the piano. I like for the piano to be in its own room, because if you have a piano sitting in the middle of a big sound stage, the piano goes out, and it dissipates, and it sounds deader and smaller. If the instrument is placed in an iso booth, or a room where the sound can go out, and then come back at you, the sound builds up in the room and the room resonates, and it sounds louder and more present.

Of course, it has to be a good sounding room. The same thing with acoustic guitar. The sound of it so depends on what room you're playing it in, and you really want the room to resonate with the guitar. It comes alive then. It projects much better.

R•E•P: Do you prefer to record the tracks on albums that you will be mixing?
DM: I do. It's more difficult for me to be happy with the way things turn out if I don't also record the tracks. There are exceptions, but so much of my sound is in how I record something, so if I happen to take a mix session, and someone wants my sound, it's difficult. I don't like being just a mix engineer. People expect something from me that's hard to give them because I wasn't there from the beginning of the process.

R•E•P: Do you have a favorite console?
DM: I prefer mixing on older Neve consoles, specifically the 8078. I like the sound of the 4-band EQ. The Neves have both the warmth and the presence, more than any other console that I've worked on. Sunset Sound Studio One has a custom console that I like a lot. I mixed the "Fourplay" album and the Gershwin album on that. It's basically their own design, with API 550 EQ in that console and GML (automation). It's a simple signal path in that console, and that's why it sounds good. Very clear sounding. I definitely go for the simple consoles. The more processing you get in a console, the more I don't like it.

R•E•P: So you're not a big SSL fan.
DM: I'm not, although if it were the right project, I might be. I like all the control and the compression on every channel. The quad compressor is probably the best part of the console, in my opinion. You can make it sound so punchy. I've done a lot of records on SSLs. I end up bypassing everything I need to when I'm recording. The same thing with the V-Series Neve. I have a way of setting it up where I'm going through as little processing as possible, and it sounds fine to me. Some of the more complex consoles are good for tracking because you have options like more headphones mixes and loads of inputs, so you can stretch out, and gates and compressors on each channel if you want them. It's great if you have a big session. When I'm mixing, though, I like the personality of the old consoles.

The first session I did for (Thom Bell) was the Spinners' "One of a Kind Love Affair," which became a number one record.

R•E•P: Do you usually record to digital?
DM: Well, I used to, but in the last couple of years I've gotten away from it. I mean, you have to wind up there eventually, but I don't record to it. I really like working with Dolby SR, and the new tapes that are out, like Scotch 996, really got me away from digital. I did comparisons, and the high end is much more 3-dimensional with 996 compared to digital. You can hear air and depth around instruments. With digital it becomes flat and grainy, 2-dimensional. Even with low end. The digital has more perfectly reproduced low end, I suppose, but with analog, the tape compression that goes on is more pleasing to the ear. So that combination, much nicer high end, tape compression on the low end, and Dolby SR so you don't have to worry about tape hiss, and I'm sold on analog.

R•E•P: Do you also mix to analog?
DM: I sure do — -inch, Studer 820. Scotch 996 and SR sound great on the Studer. ATRs have a low midrange bump, which a lot of people like, but for me, I like the machine to be more transparent. Then I make a transfer to 44.1KHz and I do my EQing through analog EQ at the same time so that my master only has to be transferred once, with no more format conversions. Then we work with that 44.1 EQ'd master to assemble. It saves steps and transferring.

R•E•P: Is there a piece of equipment that you can't live without?
DM: Well, I'll give my digital workstation a plug. It's built by Hybrid Arts, and it's called an ADAP 2. Hybrid Arts is known for its SMPTE track program. It runs with an Atari computer, and it's very easy to use and extremely reliable. I've used the ADAP ONE for years. It's also a stereo sound sampler. The ADAP 2 has 500Mbyte hard drives and can sample up to half an hour of stereo program at 48kHz. It's in a little rack, and I use it a lot for mixing. I can sample the piano solo from one take and put it into my master take. I can fly things, edit, change pitch and time compress, and it sounds great. It also has MIDI capability, and it slaves to SMPTE. I can do segues in it, and it doesn't compromise the sound. It's a great mixdown tool, and at this point it would be hard to work without it.

R•E•P: You have a lot of long-term clients. What's your secret?
DM: I like to be open to trying new things and realizing that it's a whole new day, a whole new project starting, and maybe there's something new we can get out of it. It makes it more fun to be open to new ideas. And I'm lucky that I work with people who have the attitude that they are there to make music and to have a good time.

R•E•P: How far in the future are you booked? Is it true that if Sting called you'd be too busy to work with him?
DM: Well, I have a lot coming up, but things always change, and I'd love to do a project with Sting! ■

R•E•P's 1992 NATIONAL AUDIO TEST

Compiled and edited by Eric Wenocur

Welcome to the second R•E•P National Audio Test. This collection of questions covers subjects of concern to anyone involved in the technical fundamentals of sound recording and audio production. The following 100 questions are divided into five categories: Electronics/Technology, Acoustics, Recording Techniques, Digital Audio and Sampling/MIDI.

Last year's section on sound reinforcement was dropped to condense the breadth of subject matter and to encourage live sound professionals to check out R•E•P's new sister publication, *Live Sound! and Touring Technology*.

Although we realize that it is not possible to cover every topic in-depth, nor address every level of knowledge found among recording professionals, we'd like you to consider this quiz a guide to the kind of information you may need to know, pending your special areas of activity.

The answers to these questions can be found on Page 47. Space does not permit printing complete explanations for every question, but readers are encouraged to write in about questions that were particularly perplexing. We will respond to these in an upcoming issue. Have fun!

ELECTRONICS/TECHNOLOGY

1. From the audio engineer's perspective, which of the following statements about the lowly decibel is the most accurate?

A. It describes level changes that reflect non-linearities of the human ear.
B. It is a unit of level change based on the amount of difference the average person can detect.
C. It is by itself the basis of a system for describing various magnitude-related relative measurements.
D. It is by itself the basis of a system for describing various magnitude-related absolute measurements.

2. In reference to something called Thermal Noise Floor, what sets this level and how can it be made lower?

A. TNF is the result of cosmic background noise left over from the Big Bang. Only time will lower this noise level.
B. It is the result of current flow through a component and is reduced by increasing the negative feedback of the circuit.
C. It is the result of current flow through a high gain amplifier and can be reduced by properly matching source and load impedances.
D. It is the result of Brownian motion in the atoms of the component being measured. It can be reduced by lowering the temperature.

3. Assuming negligible loss in the speaker cable, what is the amplifier power output to an 8Ω speaker if the measured Vrms is 24.5V?

A. 75W
B. 196W
C. 20 log(24.5)W
D. 245W

4. The predominant characteristic of a long audio cable that limits its high-frequency response is the:

A. Shunt capacitance.
B. Series inductance.
C. Series resistance.
D. Shunt transconductance.

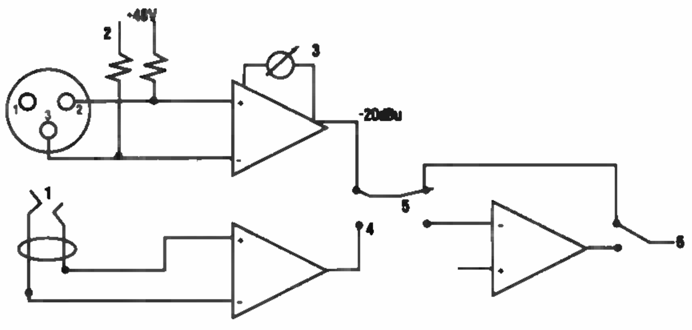
5. What is the dynamic range of an amplifier with a maximum operating level of +30dBu and a signal-to-noise ratio of 70dB, assuming a +4dBu nominal operating level?

A. 120dB
B. 104dBu
C. 96dB
D. 104dB

6. What is the amount of headroom of the amplifier just described?

A. 40dB
B. 26dB
C. 34dBu
D. 34dB

Eric Wenocur is chief engineer at KLM Video, Bethesda, MD, and a recording engineer, producer and musician in the Washington, D.C. area.



7. Figure 1 is a simplified block diagram of a console input section. What does item #1 represent?
A. Input coupling capacitor.
B. 3-conductor (balanced) patch jack.
C. 2-conductor (mono) patch jack.
D. RCA plug.

8. What is the purpose of item #2?
A. Supplies power to the microphone.
B. Adjusts common-mode rejection of balanced input.
C. Not enough info provided.
D. Supplies dc bias for the mic preamplifier.

9. Identify item #3:
A. LED overload indicator.
B. Input metering.
C. CMRR adjust pot.
D. Preamp gain pot.

10. Assuming a mic source impedance of 150Ω, what is the gain of the input amp in Figure 1?
A. 62dB
B. +28dBu
C. -50dB
D. Can't tell.

11. What is the purpose of item #4?
A. Input level attenuator.
B. Input selector switch.
C. Impedance matching switch.
D. Balanced/unbalanced terminator.

12. What is the purpose of item #5 in Figure 1?
A. Additional gain stage.
B. Polarity reverse switch.
C. EQ in/out switch.
D. Summing amp.

13. If this were a recording console, what is likely to follow after #6?
A. EQ
B. Insert point
C. Post aux send
D. A or B

14. A ground-compensating, cross-coupled balanced output stage has what primary advantage over simpler op-amp inverting outputs?
A. Lower output impedance and can drive lower impedance inputs.
B. Unit will not "blow up" if shorted to ground.
C. There is not a 6dB loss in level when run as "single-ended."
D. Maintains absolute polarity.

R-E-P's 1992 NATIONAL AUDIO TEST

15. New analog tape formulations are making it possible to raise record levels from +6 to +9 without overload. What does this really mean?
A. The level for 0VU is 9dB above the old Ampex reference fluxivity of 185nW/m.
B. The level for 0VU is 3dB above the reference fluxivity of 200nW/m.
C. The voltage of the signal is twice as loud.
D. The level for 0VU is 3dB lower, therefore, record levels appear higher.

16. To accurately measure the amplitude of an audio signal in a non-terminating system, the measurement device should have:
A. An input impedance of 1MΩ or more.
B. A terminating impedance input of 1kΩ or less.
C. A low-impedance bridging load.
D. A balanced input.

17. After making some changes to the 24-track's wiring harness, one channel's output is now 6dB lower at the console than the others. This is a tip-off to what?
A. Failure in an output amp of multitrack.
B. One side of the balanced line shorted to ground.
C. Console preamp problem.
D. Open ground connection at output.

18. For a balanced or differential input, the ability to reject noise with the same amplitude and phase on both inputs is known as:
A. Transformer isolation.
B. RFI/EMI rejection.
C. CMR.
D. Noise reduction.

19. In an analog tape recorder, tape is driven by what transport component?
A. Take-up reel.
B. Pinch roller.
C. Capstan.
D. All of the above.

20. Excessive thumps on punch-ins could be caused by:
A. Misadjusted record EQ.
B. Bias frequency set too low.
C. Bias level set too high.
D. Bias ramp time set too fast.

ACOUSTICS

21. A room has one dimension of 20 feet. What is the fundamental frequency that can develop along this dimension?
A. 56.5Hz
B. 1.13kHz
C. 113Hz
D. Cannot be calculated from the information given.

22. The room described above also has one dimension of 10 feet. What are the first 3 axial resonant modes that could become problematic in this space?
A. 113Hz, 226Hz and 283Hz
B. 28Hz, 113Hz and 226Hz
C. 113Hz, 226Hz and 452Hz
D. 56.6Hz, 113Hz and 170Hz

23. Which ratios of room dimensions will produce the most desirable room modes?
A. 1:1.55:3.60
B. 1:1.5:3
C. 1:4:6
D. 1:1.28:1.54

24. Which of the following materials has the highest sound absorption coefficient at low frequencies?
A. Painted or sealed concrete.
B. Medium weight velour drapes.
C. Faux marble.
D. Gypsum board.

25. A room's walls are covered with 1-inch-thick fiberglass insulation and fabric. It seems "dead," yet excessive bass build-up still occurs. Why?
- A. The bass frequencies are louder and require more absorptive material.
- B. More diffusion is needed.
- C. The longer bass wavelengths don't "see" the fiberglass and reflect off the wall behind.
- D. The fiberglass must be denser to "capture" all of the bass energy.

26. The Reflection Free Zone concept seeks to produce:
- A. No sound reflections in the front of the room.
- B. No sound reflections in the rear of the room.
- C. A "sweet spot," which receives mostly room reflections.
- D. A "sweet spot," which receives no early direct reflections.

27. Which wall construction will have the highest average STC rating?
- A. Single stud wall with two layers of 5/8-inch drywall on each side.
- B. Two single stud walls with one layer of 5/8-inch drywall on each side of each wall.
- C. Double stud wall with two layers of 5/8-inch drywall on each outer side and 4-inch spacing between walls.
- D. 4-inch concrete block wall.

28. One truism in studio acoustic design states that the isolation capability of a solid, heavy door is greatly diminished if:
- A. The door does not seal well around all sides.
- B. There is a window within three feet of the door.
- C. The door is made only of wood.
- D. The door is not "sound rated."

29. Why use several thinner sheets of glass laminated together for studio windows?
- A. Laminated construction stops sound passage better.
- B. Laminated construction reduces resonances in the glass.
- C. Laminated glass is easier to handle and mount.
- D. Laminated glass increases the tint factor.

30. Locating speakers in the corners of a room will:
- A. Create a tighter "sweet spot."
- B. Decrease bass output caused by diaphragmatic cancellation.
- C. Increase bass response.
- D. Create a weaker stereo image.

31. When mounting large main control room speakers, it is preferable to:
- A. Attach them rigidly to the front wall to improve bass coupling.
- B. Isolate them from the structure as completely as possible.
- C. Isolate them from the front wall, but attach solidly to side walls.
- D. Attach them rigidly to the floor or ceiling, isolated from the walls.

32. What are the characteristics of reflective and diffusive materials?
- A. Reflective material causes high frequencies to bounce from the surface. Diffusive causes low frequencies to bounce from the surface.
- B. Reflective causes complete reentry of the sound into the air. Diffusive causes some absorption at certain frequencies.
- C. Reflective material causes sound to bounce from the surface at roughly the same angle and amplitude it arrived. Diffusive causes sound to leave at random angles and intensities.
- D. Reflective causes changes in time alignment of high and low frequencies as they bounce off. Diffusive preserves the phase coherency of sound bouncing from the surface.

33. The studio HVAC system has a noisy, vibrating air handler. To reduce sound transmission into the studio, what is the best mounting method for this unit?
- A. Very stiff spring isolators.
- B. Attach rigidly to building structure.
- C. Sit on rubber waffle pads.
- D. Spring isolators with neoprene rubber pads.

34. Reflections from the control surface of a recording console can combine with direct sound from the monitors to produce:
- A. Haas effects.
- B. Comb filtering.
- C. Psychoacoustic effects.
- D. Acoustic ringing.

35. Mean Free Path (MFP) is:
- A. A method of ray tracing.
- B. The average wavelengths of all bandwidths within a space.
- C. Half the distance between the sound source and the first reflection.
- D. The average distance between reflections in a space.

36. Equal Loudness contours refer to:
- A. Acoustic power.
- B. The curves of a Butterworth filter used in a crossover network.
- C. Frequency dependent characteristics of human hearing.
- D. The distance from a loudspeaker where direct and reflected sound are equally loud.

37. The Sabine when referenced to 0 represents:
- A. Absorption equivalent to an open window.
- B. The broadband absorption of a room with an RT60 measured at 1 second.
- C. The dead end of a LEDE environment.
- D. The absorption properties of an anechoic chamber.

38. Which of the following measurements predict speech intelligibility?
- A. RASTI
- B. ANSI
- C. %ALcons
- D. A and C

39. Which of the following dB SPL meter weighting ranges are used to reflect equal loudness contours?
- A. A-weighted SPL below 70dB, B-weighted SPL between 70dB and 80dB and C-weighted SPL over 80dB.
- B. A-weighted SPL below 70dB, B-weighted SPL between 70dB and 90dB and C-weighted SPL over 90dB.
- C. A-weighted SPL below 80dB, B-weighted SPL between 80dB and 90dB and C-weighted SPL over 90dB.
- D. A-weighted SPL below 70dB, B-weighted SPL between 70dB and 100dB and C-weighted SPL over 100dB.

40. Atmospheric absorption of sound is most affected by:
- A. Altitude.
- B. Barometric pressure.
- C. Temperature.
- D. Humidity.

RECORDING TECHNIQUES

41. You are in the fifth hour of a mixing session. Your producer brings in his coffee for the umpteenth time and remarks, "Why is it so damn loud in here? It wasn't this loud this morning!" What is going on?
- A. The compressors used on your mix have brought the sound level "power" up, with lower average to peak ratios.
- B. In the process of adding more tracks, the mix has become louder.
- C. Our hearing has become desensitized.
- D. More people have entered the control room, so the room has become warmer and more humid, changing the room absorbtion amount.

42. How are the threshhold and compression ratio set for dbx-type noise reduction?

A. They are automatic, dependent on the input signal level and dynamics.
B. They are adjustable on the circuit card, but set at the factory.
C. There is no threshhold and the ratio is fixed at 2:1.
D. None of the above.

43. SMPTE time code cannot be used for which of the following?

A. Identifying video frames.
B. Identifying clock rate on digital audio signals.
C. Directly controlling a console's automation system.
D. Synchronizing a MIDI sequencer with a VTR.

44. Drop-frame time code is valid at which of the following rates?

A. 30fps
B. 29.97fps
C. Either 29.97fps or 30fps
D. 25fps

45. When transferring field audio from a Nagra tape with Pilot-tone to a 1/4-inch tape with center-track time code, the primary rule to observe is:

A. Make sure the time code is the same frame rate as the Nagra Pilot-tone.
B. Make sure the time code machine is locked to house video sync.
C. Lock the Nagra to its internal crystal at all times.
D. Make sure both transports are locked to a common time base.

46. When making an audio dub with time code from a 3/4-inch videotape to audiotape, you should:

A. Regenerate the code with a "continuous" jam-sync.
B. Regenerate the code with a "momentary" jam-sync.
C. Dub the code directly with a video-type cable.
D. Reclock the code to house reference.

47. While posting audio for a music video shot on film, you notice the speed of the audio on the videotape is slightly slower than the original song master, but it is in sync with the picture. Why?

A. Someone bungled the audio transfers.
B. Probably just a speed fluctuation in your playback VTR.
C. The time code Nagra used on the shoot was set to the wrong frame rate.
D. The film was slowed down during the transfer to videotape.

48. Which of the following is not a good troubleshooting method?

A. Substitution.
B. Comparison.
C. Bypass.
D. Addition.

49. You have just finished a miserable daytime outdoor remote in 100 degree weather with 90% humidity. What should you do next with your microphones?

A. Put them away in foam-lined boxes for their next use.
B. Take them back to the studio and set them out.
C. Leave them on the stands into the night until the place cools off.
D. Move them immediately to a well air-conditioned area to cool down.

50. You are engineering a session for a rock 'n roll band of older, experienced players. The reason they are asking for more monitor volume in the cans is:

A. Their hearing has begun to weaken because of normal aging.
B. They are so accustomed to loud stage levels that they can't get the "feel" at low volumes.
C. Years of exposure to high volumes has caused their hearing to deteriorate.
D. All of the above.

51. You are attempting to capture a literal ambient soundscape for a field shoot on video for later stereo broadcast. What recording method will create the most realistic and versatile result?

A. Set up multiple cardioid mics along the width of the scene being shot and mix them on the set.
B. Set up an M-S microphone above the camera perspective and record the raw outputs for later sweetening.
C. Set up multiple cardioid mics along the width of the scene and record them to multitrack.
D. Set up a near-coincident pair of mics aimed at the center of the scene and record them in stereo.

52. The Dolby 4-channel matrix system for film sound is used for:

A. Creating left, center, right and surround signals.
B. Reproducing 4-channel quadrophonic audio in a home theater.
C. Condensing four channels of information into two for a standard 2-channel print.
D. Creating stereo-surround channels from mono.

53. The Lucasfilm THX system for film sound is:

A. A modified Dolby-surround matrix.
B. A prefabricated speaker system for theater playback.
C. A specification for equipment, alignment and fidelity of theater playback systems.
D. A 6-channel stereo-surround system compatible with home theater setups.

54. The modern-day equivalent of "looping" in film sound production is:

A. Dubbing.
B. ADR.
C. Conforming.
D. Spotting.

55. Videotape machines and DAT machines have what in common?

A. Both use rotating heads that record a helical scan pattern on tape.
B. Both use specially formulated tape stock.
C. Both may use timecode in some form.
D. All of the above.

56. An advantage of using multiple mics on a single sound source is:

A. One mic will smooth out the frequency response "holes" of the others.
B. The sound will be louder and tighter overall.
C. More level or EQ control is possible.
D. Phase response interactions are minimized.

57. A disadvantage of using multiple mics on a single sound source is:

A. Proximity effect becomes overwhelming.
B. Low-frequency response is less predictable.
C. Distortion rises.
D. Phase anomolies are likely.

58. The proper way to gain-stage a console when recording is:

A. Set the faders and pots at the "0" references, as the manufacturer's markings indicate.
B. Set the channel faders at the indicated "0" and adjust levels with the mic trims.
C. Set the mic trims just below signal clipping, then set channel faders for correct metering.
D. None of the above.

59. When the end of a tape reel is reached, the following should be done:

A. The reel should be taken off carefully and stored.
B. The reel should be rewound at a medium shuttle speed, then removed.
C. The reel should be rewound quickly but smoothly, so as to preserve the "pack."
D. The reel should be shuttled back and forth once at a high speed to repack the tape smoothly.

60. The proper way to set a compressor to preserve transients, yet still control level is:

A. Fast attack and slow release.
B. Slow attack and fast release.
C. High threshold and compression ratio.
D. Low threshold and compression ratio.

DIGITAL AUDIO

61. The magnetic tape used for open reel digital audio recording is:

A. The same tape as used for open reel analog recording.
B. The same oxide coating, but with a thinner backing than analog tape.
C. A different tape with different magnetic and physical properties.
D. The same magnetically as analog tape, but with a smoother oxide coating and thinner backing.

62. Recordings made on 1/4-inch DASH format machines are:

A. Not playable on ProDigi format machines.
B. Not capable of being edited with a razor blade.
C. Not capable of being synchronized with video.
D. All of the above.

63. A-Time can be read as SMPTE time code on a DAT deck:

A. Never.
B. On some machines.
C. On all machines.
D. Only on units with external time code synchronizers.

64. Print-through on digital audiotape occurs:

A. Never.
B. When digital overload occurs.
C. Only on DASH format tapes.
D. When the tape is stored heads out.

65. Which of the following digital filters exhibits linear phase response?

A. IIR.
B. Transversal.
C. FFT.
D. FIR.

66. Current consumer DAT recorders include a Serial Copy Management System (SCMS). This system restricts:

A. Any copying in either digital or analog mode.
B. Second generation digital dubs.
C. Any copying in digital mode only.
D. First-generation digital dubs.

67. The proposed MADI hardware interface is based on which of the following impedances and connectors?

A. 100Ω output impedance, 250Ω input impedance, BNC connectors.
B. 110Ω output impedance, 250Ω input impedance, XLR connectors.
C. 75Ω input and output impedance, XLR connectors.
D. 75Ω input and output impedance, BNC connectors.

68. Sampling at 44.056kHz is used when the digital system is:

A. Running slower than normal.
B. Locked to SMPTE time code.
C. Locked to video sync.
D. Synchronized with an audio recorder.

69. Digital resolution refers to the:
A. Number of words per sample.
B. Number of samples per second.
C. Smallest amplitude levels between samples.
D. B and C.

70. Resolution is defined by:
A. dBu.
B. Number of samples per second.
C. Number of bits available per sample.
D. The databus width.

71. When using a multitrack digital tape recorder, what rules should be followed when placing program material on adjacent tracks?
A. No rules are necessary.
B. Avoid placing high-level, low-frequency material next to low-level material.
C. Keep highly transient material isolated from low-frequency material.
D. Avoid placing high-level, high-frequency material next to low-level material.

72. In a digital system, aliases are:
A. Artifacts caused by an insufficient number of bits in A/D conversion.
B. Artifacts caused by attempts to digitize signals above one-half the sampling frequency.
C. High-frequency spikes left over from D/A conversion.
D. Pseudonyms used by unscrupulous digital equipment salespeople.

73. What is the primary drawback to brick-wall, anti-aliasing filters?
A. They are difficult to design.
B. They cause significant group delay in the signal.
C. They exhibit significant high frequency roll-off.
D. They limit the use of oversampling.

74. Pitch-shifting is a digital effect produced by:
A. Changing the sampling frequency of digitized audio.
B. Delaying the audio through a digital storage device.
C. Expensive DSP processors.
D. Changing the rate at which digitized audio is read out of RAM.

75. When recording live drums to digital tape, it is important to:
A. Not exceed digital zero.
B. Get the low-frequency to high-frequency balance exactly as you want it.
C. Not record the high frequencies too softly.
D. Roll off anything below 20Hz.

76. Which of the following most determines audio frequency response in a digital system?
A. Sample rate conversion method.
B. Number of bits used for each sample.
C. Sample rate.
D. Amount of oversampling employed.

77. Which of the following most affects dynamic range?
A. Sample rate conversion method.
B. Number of bits used for each sample.
C. Sample rate.
D. Amount of oversampling employed.

78. Overdriving a digital signal causes:
A. Square wave distortion of the audio.
B. Loss of high-frequency response.
C. dc offset.
D. Dropouts or "splatter."

79. Recorded digital audio is sometimes described as "sterile" sounding. This may be because of:
A. Brick-wall, anti-aliasing filter designs.
B. A/D and D/A conversion methods employed at the inputs and outputs.
C. Lack of coloration typically found in analog devices and storage mediums.
D. None of the above.

80. 1-bit differs from 16-bit sampling in that:
A. It is slightly lower fidelity.
B. It only registers the difference from the previous sample.
C. It has a lower sample rate.
D. It is only compatible with other 1-bit systems.

SAMPLING/MIDI

81. You want to use a sampler triggered from a MIDI sequencer. The sampler must, at minimum, be able to respond to:
A. MTC (MIDI time code).
B. MIDI note-ons.
C. MIDI sync.
D. MIDI song position pointer.

82. Given equal amounts of internal RAM, the difference in the amount of time two samplers can record is dependent on:
A. Their word length.
B. Their sampling rate.
C. Whether they record mono or stereo samples.
D. All of the above.

83. If you have several MIDI synthesizers "daisy-chained" by connecting one's THRU jack to the next one's IN jack, the resulting total delays down the chain will be:
A. Less than the inherent response delays of each keyboard.
B. 1.5msec per synth.
C. 3msec per synth.
D. Dependent on the number of devices in the chain.

84. On MIDI cables, the connector case should be:
A. Not connected.
B. Grounded at the send end.
C. Grounded at the receive end.
D. Grounded at both ends.

85. The standard for a MIDI output connector is:
A. 5-pin DIN female, pin 1 = shield, pin 2 = +5V, pin 3 = data.
B. 4-pin DIN female, pin 1 = shield, pin 2 = +12V, pin 3 = +5V, pin 4 = data.
C. 5-pin DIN female, pin 3 = sheild, pin 4 = +12V, pin 5 = data.
D. 5-pin DIN female, pin 2 = shield, pin 4 = +5V, pin 5 = data.

86. When using a MIDI sequencer running on a personal computer, increasing the computer's processing speed will:
A. Increase MIDI data speed and reduce timing delays.
B. Increase MIDI data speed and reduce screen redraw time for the sequencer software.
C. Reduce screen redraw and computation time for the sequencer software.
D. Cause sequencer tempos to increase by the same percentage as processor speed increase.

87. Synthesizer patch data can be transmitted between equipment using:
A. MIDI poly mode.
B. MIDI system exclusive commands.
C. Continuous controller data.
D. A MIDI thru-box.

88. When using a MIDI-controlled effects device on the same cable as other MIDI equipment, make sure it is:
A. In poly mode.
B. In omni mode.
C. On its own MIDI channel.
D. A and C.

Continued on page 47

The continuing saga of the evaluation

By MIKE JOSEPH

In last month's issue, we dove deep inside four currently popular, reasonably priced digital processors, evaluating them with percussion, keyboards, guitars and voice for sources. We learned a number of things: You don't always get what you pay for; inexpensive doesn't mean poor quality; and some little boxes sure do a whole lot of things really well.

Evaluated in Part One were the ART DR-X 2100, the Digitech 256XL, the Dynacord DPS 20 and the Ensonic DP/4. This month we cover the Lexicon LXP-15, the Roland RSP 550, Sony's DPS-D7 and R7 and the Yamaha SPX1000.

Let's get to it!

THE PLAYERS

LEXICON LXP-15

DESCRIPTION: This single-rack-space stereo device uses a proprietary Lexicon chip to process audio at 31.25kHz. 16-bit linear PCM converters are fed from two unbalanced inputs, with 128 factory presets and 128 user configurable registers available for program storage. The algorithms which make up the patches include Delay/Reverb, Pitch/Delay, Gate, Plate and Chorus/Delay.

RETAIL PRICE: $1,049

FUNCTIONS AND EFFECTS: Programs include Hall, Room, Plate and Gated Reverbs; Delays; Chorus/Flange and Pitchs. All of these are available as 1- or 2-part combinations.

REVERBS OFFERED: Large, Bright and Dark Halls; Large, Medium, Small and Closet Rooms (more like ambience than reverb); Long, Small, Bright and Regular Plates; Small, Medium, Large, Bright and Slap Gates. Some of these are also available in combination with other programs, such as delays and pitch detune.

EVALUATION: Has anyone out there *not* experienced a Lexicon 224 or 480 reverb program? If you're the rare one who hasn't, you've certainly heard them on the vast majority of American pop, rock, jazz and dance records produced over the last decade. Lexicon has always been known for its excellent sound quality, and the LXP-15 does not disappoint. What can we say about an industry standard's smaller sibling?

The Halls, Rooms and Plates differ from each other, not only in apparent size and decay time, but character. They are all smooth (unless you muck with the diffusion enough to expose the chitter of individual reflections), and deep, without compromising the shimmer or sparkle. Low-end reflections are tight and controlled. All personality flavors are available, from warm and mellow to bright and percussive. The patches sound good right out of the box.

What is slightly surprising is the relatively limited palette of parameter adjustments available (compared to such units as the Dynacord or the Ensoniq). On a typical reverb program, such as Bright Plate, multiple pages allow access to pre-delay (0-262ms), pre-delay feedback, treble (roll-off slope only, from a 320Hz LP knee to full bandwidth in 16 steps), bass multiplier (0.35 to 2.4 times in 32 steps), size (eight meters to 87 meters in 1-meter increments), diffusion (0 to 100%), LFO rate, wet/dry mix, pan and levels.

Arguably, these controls are all you need if the sound is good, and it is. Other programs in the presets, especially combinations with delay or detune, provide a greater number of adjustable parameters. The ambience settings (rooms) are especially tasty, creating

CONTROLS: Single input and output level pots; 5 "soft" buttons (associated with LCD readout labels); page view button; continuous data parameter wheel; 16-position page/function knob; power switch.

INTERFACE: The unit has two each 1/4-inch unbalanced I/O jacks; five foot controller jacks; Dynamic MIDI IN, OUT and THRU jacks. Input levels are fully adjustable with front-panel pot and configuration software, 0dBu being nominal at 50kΩ stereo and 25kΩ impedances. The outputs are stated as being 600Ωsource impedance, with +4dBu nominal and +8dBu max into a 600Ω load (+14 max into >10kΩ).

the artificial acoustic space Lexicon is masterfully known for.

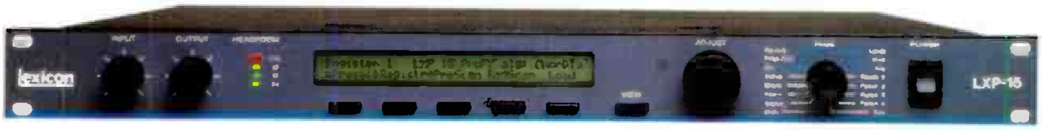
Some small presence of gated hiss and modulated noise is noticeable behind some of the setups. The Long Plate and Vocal Plate were hissy, as were the 1/4 note MAT (MIDI Auto Tempo) and 1/8-1/16 MAT presets. Many programs displayed ever-so-slightly noticeable gating on background hiss. With no signal present there was complete silence. Although better than most of the units evaluated last month, the presence of *any* hiss, modulated and breathing, or static, on a Lexicon unit was unexpected. A price vs. performance issue? Hmmm...

Mike Joseph is editor of R•E•P.

of mid-priced digital signal processors.

DELAYS OFFERED: Slaps, Delays, Cloud (cluster) Delays, combinations with reverb, ambience, chorus, etc.

EVALUATION: As you can imagine, a range of high-quality left and right echoes, delays and slap-backs exist as stand-alone effects, in addition to echoes and tunnel effects incumbent in different combination programs. They are all generally clean and flexible.



MODULATION EFFECTS OFFERED: Chorus, Chorus and Delay, Detune Chorus, Flange, Flange Room, variable envelope Chorus.

EVALUATION: Several subtle but effective single-program chorus presets are offered, but they are nothing to rush out and rave about. The sounds are generally clean and flexible, with three delay settings, feedback, treble rolloff, reverb diffusion, LFO, chorus rate, mix and level parameters available for adjustment. Some slight background hiss is perceivable on chorus and delay combo settings. Although conservative, the stock sounds are good, and quite usable. Wanking the parameters way out can give you tunnel sweep and other special effects. In combination with room ambiences and DDL, multiple alternatives are possible.

DYNAMICS OFFERED: Gating as part of specific programs.

EVALUATION: Several levels of gated reverbs are available, and each is effective. The gated verb is its own algorithm, so the process is more than just a dynamic alteration of an existing program. Also offered is a slap/gate combination that might have some interesting applications. No new ground here, but everything is done right.

EQUALIZATION OFFERED: High and low cut fixed slope, variable frequency rolloff; treble rolloff (identical to high cut filter); bass multiplier (boost/cut), as part of presets.

EVALUATION: These settings are available selectively, internal to specific programs. The treble and bass multipliers are part of the reverb page, and the high and low cut are part of the pitch/EQ page. Their effect is strong and usable, with great variations possible to the applicable programs.

PITCH SHIFT OFFERED: Pitch shift and detune algorithms as part of ambience, reverb and delay programs.

EVALUATION: No real stand-alone pitch shifting is offered. Up to six serial effects can be grouped, of which pitch shifting is one. Shifts to one octave up and two octaves down are possible, and by minimizing the delay or reverb mix percentages, the pitch shifted signal can be made the prominent element of the program. Tweaked as dry as possible, the quality is slightly warble-prone, with some glitching and/or hiss noticeable. However, within the context of the programs, judicious use of the pitch/delay algorithm adds a marvelous quality to the chorus, ambience or reverb programs. And the flexibility to adjust the parameters to taste makes the pitch shifting an effective and useful tool, with high enough quality for the application at hand.

OTHER EFFECTS OFFERED: RevWarpEnv (LFO's tunnel-sweep reverb), Cascade (pitch arpeggiation), LFO Roll (like chorus), Through A Ringer (regenerative tunnel flange), Jumpin Beans (LFO warble), Glub Glub (excellent pitch shifted lower partial with ambience), Wander Room (random panning), Fade In Space and Big Flush (dynamic envelope-altered verbs), AftFlngVerb (ramped envelope buzz verb), Gated Dive (noisey descending echoes), assorted slap reverbs and altered envelope reverb attacks and decays.

EVALUATION: All of these sounds are quite good or interesting, but definitely fall into the realm of special effects. How useful? How much sound design or science-fiction movie special effects do you do?

WRAP-UP: Clearly, the LXP-15's big strength is its capability to emulate its big brother's wonderful reverb sounds. Except for some modulated hiss on several of the programs, the verbs themselves are some of the best sampled for this article — lots of shimmer, depth and sparkle. The modulated effects are good and solid but wouldn't necessarily fit into a lead guitarist's wet dreams. The pitch shift, delay and other algorithms are nice window dressing to the reverb programs, but don't pay the price of admission.

... the verbs themselves are some of the best sampled for this article.

It's worth mentioning that Lexicon has taken a different approach to housekeeping with its choice of control parameter grouping and augmentation. The 16-position Page knob, which allows direct access to factory presets and user registers (all via the Setup position), as well as the ability to spin the knob again for direct access to deeper pages without having to scroll through one to get to the next, is nice. The View button further simplifies control by allowing either multiple parameter names or more detailed values to be addressed directly.

Attesting to the professional leanings of this unit is the fact that it is delivered with all programs balanced to a 100% wet mix. Stick it in the mono or stereo send, stereo receive loop of a quality board, and you won't be disappointed.

ROLAND RSP-550

DESCRIPTION: The single-rack-space, 2-in/2-out stereo processor provides 39 permanent factory presets and 160 user program locations — 199 possible stored effects. 16-bit linear A/D and D/A conversion is applied, with a sampling frequency of 48kHz. The 550 is designed for in-line use with instruments and in mixer high-level send/receive loops.

RETAIL PRICE: $1,295

FUNCTIONS AND EFFECTS: The 39 basic algorithms include items such as Hall (made up of reverb, gate and equalizer elements), Room, Plate, Gate Reverb, Reverse Gate, Ambience (equalizer, edge expander and ambience); various Delays (delay, modulation and equalizer); Penta, Space, 2 Band, 4 Band and Stereo Chorus (chorus and equalizer); Flanging and Phasing; Rotary; Pitch Shift; Vocoder; Enhancer; and Multi 1 through 5 (created from combinations of EQ, chorus, Pitch Shifter, Delay, Reverb, Phaser, Panning, Vibrato and/or Flanger).

CONTROLS: Dual-concentric input level pots; program/page scroll, value scroll, parameter, control assign, name, control, exit, write, system, and bypass/space buttons; power on/off switch.

INTERFACE: Two each unbalanced 1/4-inch input and output jacks, each pair switchable between -20 and +4 levels. Input impedance is stated as 47kΩ, output at 220Ω. Four 1/4-inch jacks provide program shift up and down, control signal and bypass footswitch control. Three MIDI DIN jacks offer IN, OUT and THRU connections.

REVERBS OFFERED: Rooms, Plates, Halls, Ambience, Gated Reverb, Reverse Gate and Modulated Reverb are available stock. All can be altered, renamed and saved to user locations. Various combinations of reverbs and other effects are offered in five groupings called Multis.

EVALUATION: In subjective analysis, it is hard to avoid the realization that the reverb choices offered sound pretty much alike. No matter the factory patch chosen, the general timbre and density of the reverbs seem to be based on the same internal quantifiers. Some settings are marginally brighter, denser or fuller than others (exclusive of parameter alterations), and some slightly noisier. The overall tone is generally metallic and harsh, with some high-frequency chitter noticeable on high-frequency transient inputs (variable by diffu-

PART TWO POWER PROCESSING

sion amount), and midrange artifacts return even when low-frequency stimulus only is applied.

Ever been to a YMCA swimming pool? Even by altering parametric values or preset, you keep being reminded of a basic gymnasium/arena sound. The plate sounds like a brighter large hall, somewhat midrange-y. It's not that the basic verb is bad; it's just that there's not much variation to be had, other than the changes that varying diffusion or EQ contribute.

Reverb parameter adjustments cover the following functions: 3-band EQ with sweepable mid, total reverb level and wet/dry balance, reverb time, pre-delay time (to 450 mils), HF damping frequency and level and diffusion amount. The controls provide a fair range of variation, and decent flexibility is available in the areas of reverb length and pre-delay.

Gated reverbs additionally allow the adjustment of gate threshold level, hold time, attack, release and leftover (a nice and usable touch for setting the amount of "leaked" signal after gating). The gated reverbs are basic, but they work well enough. The reverse gated reverb is more of an effect. All told, these reverbs are best used in-line with keyboards and/or non-critical instruments on stage or in a cut-it-live environment. They're OK, but they probably won't satisfy the critical ear. The exception might be the Ambience patches, which could find use building air and space around too-dry sounds. The latter's Edge Expander element adds an interesting dimension to the sound.

DELAYS OFFERED: Simple delay, stereo delays, tempo delays, and 4-tap and 8-tap delays.

EVALUATION: The delays available are flexible enough to do most of the things you'd want to do, relatively easily and cleanly. The 4- and 8-tap algorithms provide capabilities to place repeat echoes anywhere in the stereo panorama, with adjustable levels and spacing. Basic but usable stuff.

MODULATION EFFECTS OFFERED: Stereo, 2-band, 4-band, Space and Penta Chorus; Multi-Phaser, Dynamic Phaser; Stereo Flanger; Rotary Speaker.

EVALUATION: What can we say? The modulated sweeping effects featured on Roland devices throughout the years have been excellent, and the 550's offerings are no exception. The phasers and flangers are just what you might expect, adjustable from shallow and subtle to deep and intense. But the various and sundry chorus algorithms are what make this unit worthwhile and worth owning. Just as the original Roland Dimension D set a new standard of aural action, so, too, do the chorus settings offered in the six algorithms achieve the lofty goal of excellence.

On all of the chorus settings, your choice of depth in sweep, width and "churning" (to apply a phrase) are available. You won't want for quality and usable motion with these. They're all good.

The rotating Leslie effect is better than average, but not great, with a smooth adjustable transition between slow and fast horn speeds. The ability to overdrive the sound, adding a touch of distortion, is nice. You won't mistake the total effect for the real thing (a hotly biased and slightly battered tube preamp under the pedals of a B- or C-3, pushing a stack of wooden 122 tone cabinets), but in a musical context, the sound works well. As with most digital Leslie emulations, the high and low frequencies accelerate as one. It's not like that in real life! But they do have their own rise and fall times (swell periods). That's better than most of the others.

DYNAMICS OFFERED: Gating.

EVALUATION: The gating is available only as part of specific reverb and reverse reverb programs and, as such, are part and parcel of the effects themselves. In context, they work well enough.

EQUALIZATION OFFERED: High level, mid level and frequency, low level, HPF, LPF.

EVALUATION: As with the gating above, the EQ is integral to the specific reverb programs. As such (because you're EQing the wet or to-be-wet signal only), the sound is colored initially. The range of the mid sweep is good, and enough level range is offered to make a difference.

PITCH SHIFT OFFERED: Stereo, 2-band and Quad Pitch Shifter.

EVALUATION: Not advisable for use as stand-alone effects, the pitch shift capabilities are a little too warbly (with noticeable splices) to be out front. However, as timbral touches behind and under other sounds, such as full-fingered guitar chords, keyboard or group background vocals, the effect works well. The key to success is subtlety.

OTHER EFFECTS OFFERED: Vocoder, Enhancer, space sounds, etc.

EVALUATION: The Vocoder uses an element of hiss in the formula and might have its interesting uses. The enhancer is especially nice with kick drum and other percussive sounds, although it probably wasn't intended for that application. The other effects are strictly effects, if you know what we mean.

WRAP-UP: The strength of the Roland RSP-550's performance lies in its modulation effects. The reverbs are OK, but not great. However, the sweeping, swimming, swizzling sounds of the chorus are as good as any previous Roland, and possibly quieter and cleaner.

A complete 90-page Algorithm Guide is included with the 70-page owner's manual. Both are thorough and educational, even if the Japlish is sometimes noticeable.

All told, the 550 is a good live performance or MIDI studio unit. But even with its +4 inputs and outputs, it falls somewhat short in the areas of versatility, noise and ultimate fidelity as a studio tool. Compared to some members of the competition, the price is high for the features provided.

SONY DPS-D7

DESCRIPTION: The D7 is a seemingly simple single-rack-space stereo digital delay. That's all it does. It features 100 factory programs and 256 user edit locations, all based on one of seven algorithms. Presets include single, double or panoramically positioned, moving echos. An oversampling 18-bit A/D and pulse-type D/A operating at 49.152MHz provide the conversion. A proprietary DSP provides the algorithm processing.

RETAIL PRICE: $1,100

FUNCTIONS AND EFFECTS: Delays, multitap delays, tremelo, auto panned delays, extremely short delays (for basic acoustic environment simulation), calibrated time shift delays.

CONTROLS: Concentric stereo input level pots; dry and effect pots; load, help, edit, save, bypass and enter buttons; parameter adjustment wheel; power switch.

INTERFACE: Two pairs, each 1/4-inch and XLR input and output jacks, -10 and +4, respectively. Max input is +10 and +24, with 50kΩ and 10kΩ loads on the unbalanced and balanced lines. Outputs are also -10 and +4, with max levels being +10 and +24 with 10kΩ and 600Ω source impedances on the unbalanced and balanced outputs. MIDI IN, OUT and THRU 5-pin DINs are offered, with two proprietary D-sub 9-pin remote jacks provided for the system remotes.

DELAY OFFERED: You name it, if it echoes, slaps back, warbles or flies around your head, the D7 does it.

EVALUATION: I never thought I'd say "now I've heard everything" when it comes to delays, but ... anything that can possibly be done with slapbacks and multitap, autopanned DDL is in this box. Despite a little hiss on some of the settings, the unit is exceptionally clean, crisp and clear. The range of effects is much larger than one might think delays and multitaps offer. The D7 allows the ability to get in and alter parameters with simple yet powerful controls, so any sound effect present can be further tweaked to your own idea of perfection.

The autopan function available on all programs is powerful, and sounds as far ranging as psuedo-chorus, vibration, a telephone filter effect and stereo tremolo (reminiscent of a stereo Rhodes) are represented. Even believable reverb-like programs are offered.

Aside from the creative applications, many useful production programs exist: calibrated NTSC (and PAL/SECAM) 1-frame and 1-field stereo time shift, 10-meter stereo time shift, 7mm stereo time shift and ambience are here.

EQUALIZATION OFFERED: A comprehensive and powerful 3-band equalizer is internal to the programs. They all cover ±12dB, with bass sweeping from 16Hz to 6.3kHz, treble covering 400Hz to 20kHz, and a parametric section active from 63Hz to 20kHz with a variable Q of 0.267 to 34.62 (very narrow to very wide).

EVALUATION: This is possibly the cleanest and most powerful equalization offered on a digital

YAMAHA SPX1000

DESCRIPTION: This modern, single-rack-space update of the original SPX 90 features 40 factory presets and 59 additional user RAM locations. Unique to all processors evaluated, the 1000 features digital inputs and outputs (in the Yamaha format) and high/low switchable level analog inputs. Licensed Aural Exciter™ enhancement is offered, and the stereo sampling is 16-bit, with a conversion frequency of 44.1kHz.

RETAIL PRICE: $1,595

FUNCTIONS AND EFFECTS: Halls, Rooms, Plates and Echo Rooms, Gated Reverbs and Reverse Gates; Delays and Stereo Echos; Stereo Flange, Chorus, Phasing, Tremelo, ADR Noise Gate, Pitch Change, Freeze (stereo sampling), Panning, Distortion and Aural Exciter.

CONTROLS: Power switch; concentric stereo input level; parameter select (forward and back), parameter change (up and down), EQ, internal parameter, external controller assign, level, store, program select (up and down), recall, utility, trigger and bypass buttons.

INTERFACE: Two each, unbalanced 1/4-inch input and output jacks, switchable in pairs between -20dB and +4dB level. 8-pin DIN Yamaha format digital input and output jacks. DIN MIDI IN and switchable OUT/THRU jacks. 1/4-inch unbalanced Increase/Decrease and Trigger 1 Switch jack, Analog Trigger 2 jack (switchable between "mic" and "line" level), and a bypass control jack.

REVERBS OFFERED: Hall, Room, Vocal, Plate, Echo Room, Early Reflections, Gate Reverb and Reverse Gate.

EVALUATION: Without mincing words, the reverbs offered by the 1000 will not by themselves entice you to rush out and grab the unit. They are pedestrian at best. The basic sound, identifiable as the working element in all the verb patches, is a big, warm chamber. As evidenced in the Hall patch, the sound is full and dark, with just a touch of high-frequency chitter. Playing with diffusion or damping can bring out the negative side effects, but at no point is the sound smooth or brightly silky. That's not necessarily a bad thing, if you're looking for a rougher snare return or something with a touch of bite.

The Room patch, as, too, the Vocal and Plate programs, can be made to sound virtually identical to the Hall patch by noodling with the HPF and LPF, EQ settings, diffusion, damping and decay. It's more than a 1-sound-box, but the range of play is limited, if you don't want to go too far into chitterland.

Are there algorithmic differences in the verbs? Yes. Some parameters are unique to different programs, but as you've heard us say before, the basic quality of the timbre is consistent throughout, and it's not a Lexicon or a Dynacord. Wanking-in EQ or going to HPF/LPF extremes will definitely change the sound quality, but you're more into effect territory here than basic, usable reverb.

One strange anamoly is worth mentioning. Dialing in *any* HPF introduces a rough, grainy hiss to the program. The noise varies by the frequency selected. Very disconcerting.

The Gated Reverb and Reverse Gate are usable, albeit somewhat "buzzy" and chittery. The Reverse Gate especially evokes the sound of a zipper. Again, a usable effect, certainly applicable to specific things, but not a general purpose verb.

The Early Reflections fall into this category, too. They are less room ambience creations than combos of short reverb and delay meant to *simulate* an ambient environment. The rough grain is clearly in evidence here.

DELAY OFFERED: L/C/R Delay, Stereo Delay.

EVALUATION: Some noise or hiss is noticeable, although the distraction is negligible. The L/C/R Delay is especially nice. Single shots on a tom, for example, can be turned into an entire tom roll, right around the stereo spread. In short, these are two simple but effective programs.

MODULATION EFFECTS OFFERED: Flange, Stereo Phasing, Chorus, Symphonic.

EVALUATION: The Flange program is very good, almost more like a strong chorus than a flange, whereas the Chorus is weak, demonstrating none of the depth that a good chorus patch should have. It does, however, provide a solid sense of panning, churning sweep, almost stereo flange-like, back and forth at a selectable rate in the stereo spread.

Like the Flange preset, the Stereo Phasing and Symphonic programs are also strong. The latter is more like a shallower chorus setting, with a lower midrange emphasis. Symphonic is applicable to keys, guitar and vocal spreads, as well as percussive effects.

Overall, the modulation effects are good, but disappointment lies in the weak chorus. Go figure.

Pitch change 2 deserves special mention -- it is an absolutely wonderful setting for guitars, vocals and keys. It is almost worth owning the device for!

DYNAMICS OFFERED: Compress/Expand, ADR-Gate, Gate as part of the Reverb programs.

EVALUATION: The compression/expansion algorithm provides the prerequisite basic number of controls — attack, release, threshold, ratio — but excludes an in-program drive or input level (other than the analog input pots). The compressor and expander work, after a fashion, but they wouldn't cut it as stand-alone devices. The sound is flat, with limited life, even after playing with the attack and release settings. It is probably OK for use on synths and such (strings, swells, etc.).

The ADR-Gate algorithm fairs somewhat better, because this program is far more usable on its own. Tweaking the settings provides usable noise removal for hissy guitar and keyboard rigs. Some bounce on release is noticeable, but overall the program works well enough.

EQUALIZATION OFFERED: 2-band parametric, HPF, LPF, BPF, PEQ, LFO and Dynamic Filter.

EVALUATION: The EQ algorithm, a separate function available to any program, provides a wide range of action among limited controls. The basic settings include grouped stereo 2-band selectable peak or shelf EQ (high or low Q in peak modes only), variable in frequency from 32Hz to 2.2kHz on the bottom and 500Hz to 16kHz on the top. The boost or cut is 15dB, and the EQ functions effectively.

Also available are Dynamic Filters, controllable by an internal LFO or the input signal's level to the analog trigger. Either of these signals can drive either an LPF, HPF, BPF (bandpass) or a PEQ (parametric) filter. The various effects can create some wonderful sonic signatures, dependent on the nature of the drive signal dynamics and input tonality. Discovery and play is in order here.

The combination programs that exist between factory presets 31 and 35 additionally allow 2-band parametric control of the left and right channels separately, with the same parameters as the 2-band EQ stated above.

The equalization generally is effective and useful, although the Q does not provide much variation between its high and low setting. Additionally, the noise level rises substantially with the adjustment of certain EQ parameter settings, irrespective of the amount of boost or cut added. Ah, the joys of digital filters...

PITCH SHIFT OFFERED: Pitch Change 1 and 2, Stereo Pitch Change.

EVALUATION: The nicest-sounding program of the factory presets is patch 21, Pitch Change 2. It is absolute magic. Nothing more than a fine downward detune on one channel (-8 cents) and a fine upward detune (+8 cents) on the other, the effect adds life to everything we tried it on. Simply wonderful.

The maximum shift capability on any of the settings is up or down two octaves, with fine-tuning available in 1-cent increments. Too much shift introduces the inevitable warble.

The Stereo Pitch Change preset definitely suffers from too much of the latter.

OTHER EFFECTS OFFERED: Distortion, Panning, Triggered Panning, Freeze (sample), Tremelo.

EVALUATION: Forget the distortion program. Two back-to-back diodes across a speaker would sound more realistic. The three Freeze or sample patches work well, with triggering of the playback available by various means. Freeze 1 and 2 allow 5.8 seconds of sound to be captured, with Stereo Freeze allowing 2.9 seconds. All provide the means to trim down the front and back of the sample by indicated milliseconds.

Extremely usable is the triggered delay, configurable from L to R or R to L in pan direction. This program could see a lot of use. The basic Pan program is what you think, but with the addition of a front-to-back parameter adjust, the net effect is a "dipping" of the signal level in the middle on one leg, as if the signal were completing a circle in front of you, receding and then passing closely by.

The tremelo certainly won't replace a Fender amp, but it may find use on some retro-sounding production.

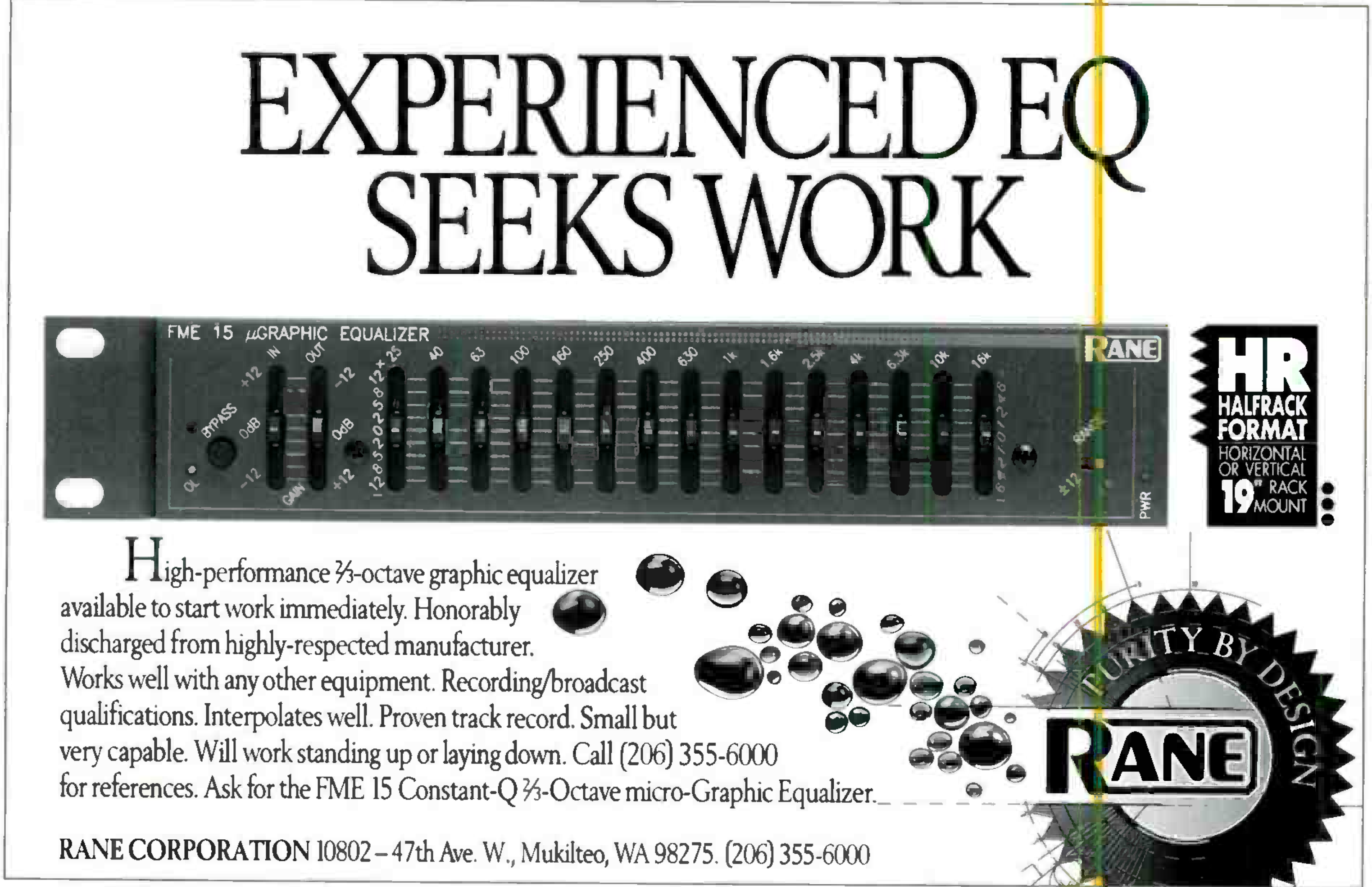
WRAP-UP: Although the reverbs are usable, they are somewhat weak compared to several of the competitive units. The modulated effects, however, specifically the Phasing, Flanging and Symphonic programs, are very good. Pitch Change 2 deserves special mention — it's an absolutely wonderful setting for guitars, vocals and keys. It's almost worth owning the device for! The Triggered Pan and standard Pan patches are usable, but avoid the Distortion setting. It's a real speaker-blower.

An interesting Utility parameter change allows setting up the inputs in reverse stereo (left input feeds right channel, etc.), normal, left mono or right mono signal feeds. The Utility mode further allows switching between an internal digital clock and clock signal embedded in the received digital input signal.

Additionally, the digital I/Os can be used as pre-effect or post-effect looping outputs (inserts). Conceivably, the A-to-Ds could be isolated and used independently, as could the D-to-As. We didn't try it, but someone should and let us know. Remember, though, that they are Yamaha formatted digital protocol.

In short, the Yamaha SPX1000 is a flexible, sophisticated processing device that delivers numerous effective programs, a number of them dynamically driven by the signal.

If any of the products in this compendium have piqued your interest, we highly recommend that you rush right out to the nearest pro audio playground and start nudging buttons. We'll take no responsibility for the fun you'll have. Good luck! ■



STAYING IN SYNC

IN THE REEL WORLD

Practical tips on being sync savvy.

By JOE MOORE

So just what is time code anyway? Let's start with a definition: Time code is an electronic digital signal that can be recorded along the length of audiotape or videotape in the same way that sound is recorded. It is a square wave with a bandwidth of 15kHz. All information carried in the time code word is binary — it's made up of ones and zeros. Its main function is to give each frame of video a unique address that can be easily found. Time code can also contain additional useful information.

Joe Moore is senior audio engineer of Post Edge, Hollywood, FL.

One frame of time code is called a word and each word is divided into 80 parts, called bits. The 80 bits are assigned as follows:

1. 26 are for time information: hours:minutes:seconds:frames.
2. One bit is used to indicate drop-frame mode. We'll be covering this topic later.
3. One bit is used to define the video color frame.
4. One is for phase correction.
5. 16 are to identify the end of the frame and the direction of travel, called the sync word.
6. Eight 4-bit groups, individually called binary groups and collectively referred to as user bits, are available to store such data as the date, scene number and take number.
7. Two bits are used to select one of four ways of interpreting the user bits.
8. One unassigned bit is reserved for future use.

All these bits fall into three categories: time information, user information and sync information.

TIME INFORMATION

Each video frame is identified by an 8-digit address broken down into hours, minutes, seconds and frames. This provides a number of advantages in producing and editing audio and video material.

First, it can represent the actual clock time when something was shot or recorded. Producing a chronological report or making a list or log of takes is easy. Second, the duration of an event, such as an edit or the actual running length of a program or commercial, can be calculated with frame accuracy. Third, individual scenes or takes can be identified by programming a specific time code number to them.

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TIME CODE STANDARDS

Originally, black-and-white TV receivers in this country used the 60Hz power line on which to generate their 30Hz frame sync pulse. Their vertical hold controls had sufficient range to accommodate any variation from the sync frequency of the station. The video frames were advancing at a rate of 30 frames per second (fps). If time code had been around to measure the program, everything would have agreed — the program length, the time code display and the clock on the wall.

Then along came color television and things quickly became complicated. To avoid producing beat frequencies caused by the 60Hz ac power line, as well as various oscillator harmonic interaction, and provide enough room in the spectrum for color information, a reference frequency of 59.94005234Hz was chosen. Thus, the color TV frame rate became 29.97fps, a change that produced a stable picture but some interesting side effects.

Even though the color reference changed, there were still 30 frames in every second of video. So, if you clocked a color program at 30fps, you picked up an extra 0.03 seconds every second because the video was running at a slower frequency. At the end of an hour, you would wind up with 3.6 extra seconds of video.

If you were trying to time a program accurately, this would not work because there would never be an agreement between your time code clock and the clock on the wall. The problem was how to lose 108 video frames every hour and still run the time code at a constant speed. To compensate for this, the drop-frame time code standard was developed.

DROP-FRAME EXPLAINED

Instead of the time code address numbers always advancing by one frame, the counting was altered so that whenever the time code ended a minute, it dropped the first two frame numbers beginning the next minute. As an example, frames 1:34:00:00 and 1:34:00:01 would not exist. Frame 1:33:59:29 would advance to 1:34:00:02.

The chances of the ¼-inch deck at the post house running at the same constant speed as the recording studio's deck are about as good as winning the lottery.

This omission did not affect the video picture. The video frames still progressed in a continuous frequency rate of 29.97fps. The only thing altered was the time code number count. However, this method of dropping two frames every minute lost 120 frames during the course of an hour. The goal was only 108.

To compensate further, time code was allowed to keep the first two frames of every 10th minute. It is important to note that although 29.97fps drop-frame keeps real time on the average, at any given point it can be almost 0.10 seconds fast or slow, depending on when the last frame number was dropped.

Admittedly, the drop-frame standard is very confusing. It is also *very* important to understand that there are 30 frames in every second of video, even though it is running at a frequency rate of 29.97fps.

To visualize this clearly, imagine two identical bicycle chains made out of rubber and containing the same number of links. If you stretched one of them slightly, the links would become longer and it would take longer to follow the length of the chain all the way around to where you started. But the number of links would not change.

The shorter chain would correspond to 30fps full-frame time code, but the stretched chain would be 29.97fps full-frame time code. Now, if you periodically removed links along the stretched chain, eventually the length of the two chains would match fairly close. That is exactly what happens with the drop-frame standard.

To complicate matters, it is possible to have the following: 30fps full-frame, 30fps drop-frame, 29.97fps full-frame and 29.97fps drop-frame time code.

We will discuss "why" shortly, but when dealing with audio and audio for video, you will be using 30fps full or non-drop-frame and 29.97fps full or non-drop-frame time code most of the time. Full frame means that every number is counted and that none is dropped.

SMPTE TIME CODE

There are two types of time code in use today. The first, developed by the Society of Motion Picture and Television Engineers, is SMPTE time code. This type is treated just like an audio signal and is recorded longitudinally — lengthwise on an audio or cue channel of audiotape or videotape. This is referred to as LTC. In the case of a multitrack audio deck, it is usually recorded or striped on an outside channel, such as channel 24 on a 24-track. A buffer channel should be used between it and any production channels. Whenever possible, it should be recorded at a normal operating level of 0dB. However, adjacent-channel leakage on a

STAYING IN SYNC

multitrack tape may require recording levels 6dB to 10dB lower.

SMPTE LTC can be post-striped onto audiotape or videotape and can be read only in the play mode. Some machines and computers use a wide bandwidth reader board to read the code in fast shuttle. Most computers, especially in the audio studio, must rely on counting tack pulses in shuttle. This is because the tape lifter mechanism does not allow the time code channel to come in contact with the heads during rewind or fast-forward. Doing so could damage your speakers.

One disadvantage to this type of code is that it cannot be read in slow shuttle. To circumvent this with videotape, you can have the time code numbers burned into the picture electronically, a process called window code. Because the numbers are part of the picture, if you pause, stop or step through the picture slowly, the numbers are still visible on the screen. There is, of course, no audio equivalent of this except with random access digital audio workstations.

Finally, some professional videocassette machines have a special audio track called the address track. It's used just for recording time code. This frees the regular audio channels for mono or stereo production audio.

VITC TIME CODE

An alternative to SMPTE LTC was introduced in 1978. Called Vertical Interval Time Code or VITC (pronounced vit-see), it contains the same address and user information as SMPTE but is recorded inside the video signal itself, yet outside the visible picture, in an area called the vertical blanking interval.

VITC has a number of advantages over SMPTE code. The most important is its ability to be read in slow motion or "still frame." Remember, video is recorded and played back on a helical scan or spinning head assembly while the audio record and playback heads are stationary. Because VITC is part of the video signal, anytime a picture is visible, the time code is detectable with a special reader. This eliminates the need for burned-in time code, because a VITC reader can detect and display frame accurate code in any mode.

Second, VITC can identify a video field. As we discussed, a time code word is one frame long. But each video frame is really constructed of two individual fields. Field one is produced as the picture tube scans the odd numbered lines of the video; field two consists of all even numbered scan lines. VITC indexes both fields of a frame with an address number that includes field identification. A sophisticated computer editing system can use this information to make extremely accurate edits.

VITC frees the audio channels for production audio and eliminates crosstalk from the code signal. Its biggest disadvantage is that it cannot be post-striped onto the tape but must be recorded simultaneously with picture, called VITC. In most cases, as an audio engineer, you will never come in contact with VITC.

PRACTICAL APPLICATIONS

Now that all of the technical stuff is out of the way, let's talk about the real world. Whether you work in a traditional recording studio, do location audio recording or mixing, operate a MIDI studio or coordinate engineering in an audio sweetening room, chances are you've encountered time code. In fact, most of you probably use it on a daily basis. Some of you cannot do your job without it.

TRADITIONAL RECORDING AND MIDI STUDIOS

Probably the most common use of time code in the recording studio is to lock two machines together. This is done with a synchronizer and either a couple of multitrack tape decks or a multitrack and a video deck. The syn-

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the master and the offset would then be <->4:00:00:00.

er is a computer that will designate one of the machines as a master as a slave. It will then both machines, con- slave and servo lock-up time onds. As the

Offsets are initially a little hard to grasp, but most synchronizers do all the arithmetic for you. It's wise to have at least a general understanding of offsets, however, because the chances of the master and the slave having the same numerical time code is slim.

Incidentally, it is good practice to start your program material (music bed, commercial, etc.) on an even minute mark. This is standard procedure at any professional post-production facility. A sure sign of an amateur is starting your work on some obscure number, such as 6:15:12:22. Not only does using the even minute mark keep things neat and tidy, but it makes it easy to compute the duration of your work. If your music starts at 5:00:00 and runs to 5:26:18, its length is 26 seconds and 18 frames long. How convenient.

A common function of the traditional recording studio or MIDI room is to record music, and an offshoot of that is scoring to picture and sound design. A typical scenario: A copy of the video master, whether it's a commercial or program, is made on a videocassette, usually a ¾-inch U-matic with address track time code and corresponding burned-in window code. The address track is a special additional channel on a professional ¾-inch machine used for recording time code. This keeps the two audio channels available for stereo or split audio signals.

The videocassette is played and its address track time code is read by your synchronizer. If an offset is necessary, it is computed and a multitrack machine or MIDI setup is locked to picture. Music is performed, layered and built onto the multitrack or sequenced on the keyboards until the piece is complete. A final mix is created and copied to ¼-inch audiotape. The tape is sent to the video post house to be laid back to the video master.

Will it stay in sync? No way. Why? First, no two free-running audio decks in the world will run at the same constant speed no matter what brand they are or how stable their crystal reference. In the world of music and record production, this is of little consequence. But in the world of frame-accurate video production, this fact is critical and sometimes deadly. So the chances of the ¼-inch deck at the post house running at the same constant speed as the recording studio's deck are about as good as winning the lottery.

Furthermore, the free-running deck at the recording studio is said to be line-locked, using the power line as a reference. The machines at the post house are all using a common house reference and are said to be sync-locked. The difference is that sync-locked machines are running 0.1% slower that their line-locked brothers. The two will never match. More later.

THE SYNC-LOCK SOLUTION

The answer to this is a threefold process. First, the multitrack slave or MIDI setup *must* be locked to the videocassette master during the mixdown process. Second, the ¼-inch mix *must* be striped with code while the mix is being recorded. This is usually done on a center track time code machine, or, in the case of a mono signal, the code can be recorded on channel two of the 2-track. If possible, it should be the same numerical code as the slave but it absolutely *must* be running at the same rate as the slave. And because longitudinal time code does not copy well, it should be re-generated with a time code generator. In any case, *never*

THE HISTORY OF TIME CODE

Once upon a time, possibly before you were born, television was live. It was broadcast directly from a studio somewhere. Viewers sat down after dinner all across the country and watched the same program at the same time. But a problem quickly arose. When it was 8 p.m. in New York, it was only 5 p.m. in Los Angeles. Some folks were not even home from work yet. The producers wanted a way to delay their broadcast to the West Coast so that everyone could enjoy the show at a convenient hour.

This was solved in the mid-'50s when the Ampex Corporation invented the first videotape recorder. Shows were produced live in New York, recorded on a VTR and fed to the West Coast hours later, days later or whenever they wanted. The term pre-recorded was introduced. This flexibility opened a whole new world of production possibilities. More complex shows were taped and many new methods of keeping track of the recorded material were developed.

One of the first and most basic methods was the VTR's footage counter. Fairly accurate, it could be zeroed at the start of the tape and a log kept to find the location of specific stuff. However, footage counters vary slightly from machine to machine, not to mention reel size, the pack of the tape and stretching. Frame count was hopeless and editing was a distant dream.

The next method to come along called for counting control track pulses. The control track is a series of equally spaced electronic pulses that are recorded along with picture on videotape. When the tape is played, the phase of the control track is compared to the phase of the machine's internal or external reference. This ensures that the tape plays back properly.

In a way, the control track is to videotape what sprocket holes are to film. But unlike sprockets that can be numbered and counted, all control track pulses are identical. If you try to count them and lose count because of noise or distortion, you must start from the beginning. Accurate editing was still over the horizon.

Not to say they didn't try. Editing in the '50s was done by physically splicing the tape together just like audiotape. The results were sloppy and created picture roll and breakup. They were well below the standards we know today.

Then along came electronic editing. This involved playing a videotape on one machine (the source or playback deck) and re-recording it, segment by segment, on a second machine (the destination or master record deck). Although this method eliminated the breakup and roll of physical edits, it was still primitive and not accurate. Many versions of electronic editing came along, all based on counting control track pulses. None was frame-accurate. (Most off-line VHS editing systems still use the control track with which to edit today.)

Then in 1967, a new method was developed. This method used an electronic signal called time code to define and identify each video frame with a unique location. It did so by dividing the video into hours: minutes: seconds: frames. With this time code, editing became efficient and fast. Most important, it was frame-accurate. Also, it allowed an edit to be rehearsed or previewed many times with everything remaining in sync.

In the early '70s, the CMX company furthered the editing process with the introduction of computer assisted electronic editing. With sophisticated computers, edit points could be identified using time code numbers. The edit decision list (EDL) could be stored in memory and automatically performed anytime. As this technology matured, the electronic editor was able to do even more tasks besides just executing an edit. The computer could control a video switcher, allowing the operator to program dissolve rates, fades, cuts, wipes and special effects. It could also send discrete commands to digital video effects devices called GPIs, general purpose interfaces.

Today, all on-line and most off-line electronic audio and video editing are based on the use of SMPTE time code. ■

STAYING IN SYNC

mix rates or drop and non-drop-frame time code.

Finally, the ¼-inch tape mix *must* be locked to the video master at the post house as the mix is laid back. If this chain of locked and resolving protocol is ever broken, the process is doomed and fixing it is quite costly and time-consuming.

In place of a ¼-inch center track deck, an R-DAT is just as good. Like videotape, it contains a type of control track and is self-resolving. Because the R-DAT does not have time code, it cannot be locked to picture and must be transferred to something that can. Then that copy must be locked to the video master during layback.

The term "resolving" is playing back a tape so that the program comes off the tape at exactly the same speed that it went on the tape. Locking a slave to a master resolves the slave. When playing an R-DAT, its control track is referenced to the internal sync of the machine and it, too, is resolved.

Obviously, there is no control track on a piece of ¼-inch audiotape. The audio deck is simply moving the tape across the heads at a speed dictated by its internal crystal referenced to the local power company's 60Hz ac or line-locked. Anything can happen and usually does. Time code takes the place of the control track on the ¼-inch tape, but to be resolved it still must be slaved to something.

Let's examine another scenario. A music track is produced in a traditional recording studio or MIDI room with everything done properly. All slaves and masters are locking together. The ¼-inch mix is striped with code. Then, as with every master that goes out of the studio, the conscientious engineer leaders each cut on the tape. By doing so, he has rendered the tape useless.

Leadering the head of a cut means no pre-roll, no lock-up before the audio starts, and no resolving, because it takes a second or two for the resolving process to begin. *Never* leader time coded tapes! One of the purposes of the code is to give you an exact location to your program content, therefore leadering is not necessary. To lock up and resolve, there must be at least 10, and if you can, 20 seconds of pre-roll. This goes for the end of the program content as well. Ten seconds of post roll is perfect, but *no leader tape*! Simply note on the label the time code start location of each cut. The sweetening engineer will find it easily.

Now let's talk about prepping for a music video shoot. Most of the time, music videos are shot on film. At some point, the film will be transferred to videotape on a special machine called a telecine. Your objective as an engineer is to give the on-location sound man a copy of the song to which the performers can lip-sync, thus assuring them that when they edit their film later, the music will stay in sync with the picture. This is an extremely confusing task and requires a great deal of understanding of video and film speeds and rates.

Except for special effects or unusual circumstances, film is shot at a standard rate of 24fps. Audio recorded or played back on the set of a film shoot is running at a standard speed of 7.5ips. This is most often done on a Nagra tape recorder. For those that don't know, Nagras come in three flavors: mono with 60Hz neo-pilot tone, stereo with FM pilot tone and stereo with SMPTE time code. The pilot tone and time code are the "control track" used to resolve on playback.

When film is played back on a telecine and transferred to video tape, it is slowed down slightly from 24fps to 23.97fps. This difference amounts to 0.1%. The reason is that everything in a video post-production facility, including the telecine, is locked to a central external reference sync generator running at a color frequency rate of 29.97fps. This is normally referred to as house sync, and all of the equipment is said to be sync-locked.

The music that was used during the shoot was played back at 7.5ips. But once the film is slowed down on the telecine, that music will no longer sync to the picture. It will be running 0.1% too fast. In the final edit, the singers will visually drift out of sync with the music. So how do you compensate? If you know that it will be a film shoot, proper planning and prep work will ensure flawless results.

FILM SHOOT PLAYBACK

The first step is to copy the final music mix to videotape *before* the shoot. From the standpoint of quality, a direct digital transfer from R-DAT to D-2 digital videotape is best. Remember, D-2 audio is an AES format running at a sample rate of 48kHz. Don't show up with your DAT recorded at 44.1kHz! The next best thing is a CD copy of the song. If this is not available, then a ¼-inch 15ips studio master is fine.

Once the music is on videotape, that tape now becomes your new "music master for video purposes" and all copies being prepped for the film shoot *must* be made from it. Most post houses have a center track time code machine. Have them make two copies at 7.5ips on 5-inch reels. Two copies always ensures that there is a backup on the set in case one gets trashed. The reason for the speed and size of the reels is because field Nagras cannot hold large reels nor play at 15ips.

Now comes the secret to the slowdown speed dilemma. Make a note on the box that the ¼-inch tapes are striped with code running at a rate of 29.97fps and *must* be cross-reference resolved to 30fps on playback. This way, the field Nagra will cause the music to play slightly faster, 0.1% to be exact. The talent will be lip-syncing to a faster playback and when the film is transferred to videotape, their actions will be slowed down by 0.1% to match the music already on the video master. On the final edited music video, the singers will be in perfect sync and you'll be a hero!

This overview doesn't cover every scenario or every use of time code you will encounter, but these are common areas where many mistakes are made. As you deal with time code, you'll come across others and develop alternate ways of dealing with them. The important thing to remember is that time code is a powerful tool in your audio engineering arsenal. It can make your life easier, your job more efficient and the products more professional. ■

Choosing the Right Power Amplifier

By David Scheirman

"I have a speaker cabinet that says it's rated at 300W. What size power amplifier should I get to hook up to it?" That simple question, posed recently by a part-time musician who had to deal with his group's sound system, made me stop and think for a minute about power. About watts, ohms, and all of those other things named after English and German scientists and inventors a century or two ago.

On the surface, it seems simple: you need to raise the level of an audio signal and connect that more powerful output to a loudspeaker, at which point you will convert electrical energy into acoustical, with a bit of waste heat as a by-product. Hopefully, the power amplifier will handle most of this process ... if the speaker ends up being responsible for too much of the waste heat, then it's goodbye sweet sound, hello burned-out voice coil.

The musician with the 300W speaker cabinet assumed that he should get a 300W amplifier. A watt is a watt, right? That might at first seem like a good rule of thumb, but things can be a bit more complicated than that.

What type of program input was the amplifier measured with? How about the loudspeaker? Are these "AES standard" watts, or "music power watts," or "peak power" watts, or something in between? Do some loudspeaker manufacturers measure their speaker products' power handling ability conservatively (so as to not be embarrassed by too many failures) and do some power amplifier manufacturers go the other direction with their wattage rating claims, so as to appear to be the most powerful for the buck? And, if so, how do you tell what model to get for a certain application?

Specsmanship and marketing claims aside, there are some more basic questions that anyone should ask when power amplifiers are under consideration for sound reinforcement use.

• How much total system electrical headroom is desired for the type of use that the system is most likely to encounter? [Power amplifier headroom will help to prevent audible distortion and will actually place less stress on loudspeaker components than a zero-headroom system. 10-20% is a minimum for everyday normal-use applications, 25-50% for heavy-duty rock 'n' roll usage, and some audio fidelity purists providing sound reinforcement systems for opera and symphonic use like to go with 100% headroom for very accurate amplified transient response.]

• What's more important: a good pricing deal for mass quantities of amps from a single supplier, or optimized amplifier applications based on exact needs? [Lows, mids and highs often work best with different design types or products from different manufacturers, depending on the loudspeaker component characteristics and system use requirements.

• Does the power amplifier need to do anything else beside amplify an input signal? [Many amp manufacturers are now offering plug-in signal processing options, such as electronic crossover modules, high-pass filters and limiters. Some offer computer-control modules that enable the unit to be linked into a network of similar devices, all monitored and controlled by a personal computer in a remote location.]

• What controls and features are required? [Is it crucial to be able to adjust amplifier gain? Should it be on the front or back panel to meet your system needs? Do the gain attenuators need to be step-adjustable for consistent settings? Do you need to see clip indication, or thermal status?]

• What environment will the amplifier operate in? [Are loud, single-stage fans acceptable in a quiet performing area, or must you have thermo-sensing and multi-stage fan operation? How capable must the air filters be ... is foreign airborne matter going to be a hazard?]

•What type of input and output configuration is required? [Connectors vary in different types of situations. Do you need XLR's on inputs and binding posts on outputs? RCA or -inch phone jacks? All of the above? Should the A and B channels be strappable for bridged-mono operation?]

•What input sensitivity best suits your system needs? [An amplifier will develop its full, rated power output in response to a specified voltage level input. This varies from amplifier to amplifier, and is often adjustable internally. In a multi-amp system, calibration of input sensitivities is important for optimum system performance.]

•What is the intended duty cycle of the amplifier? [Will it be run 24 hours per day in a permanent installation? Will it receive a few hours' hard use on a daily basis in a tour situation? Is the amp being purchased for long-term use in a certain application in a specific system, or will it be a building-block module as part of a rental company inventory?]

• How important is serviceability? [Should it have modular plug-in circuit cards for quick road replacement? Will it rarely pass through the shop to sit on a test or repair bench? Where are the fuse caps located? Are the output transistors readily accessible?]

•How will the amplifier be racked? [An amp that sits in an air-conditioned recording studio, built into a wall, will suffer far less abuse than an amp that travels around the country in the back of a truck. Mechanical design and chassis construction is a very real consideration for portable system use.]

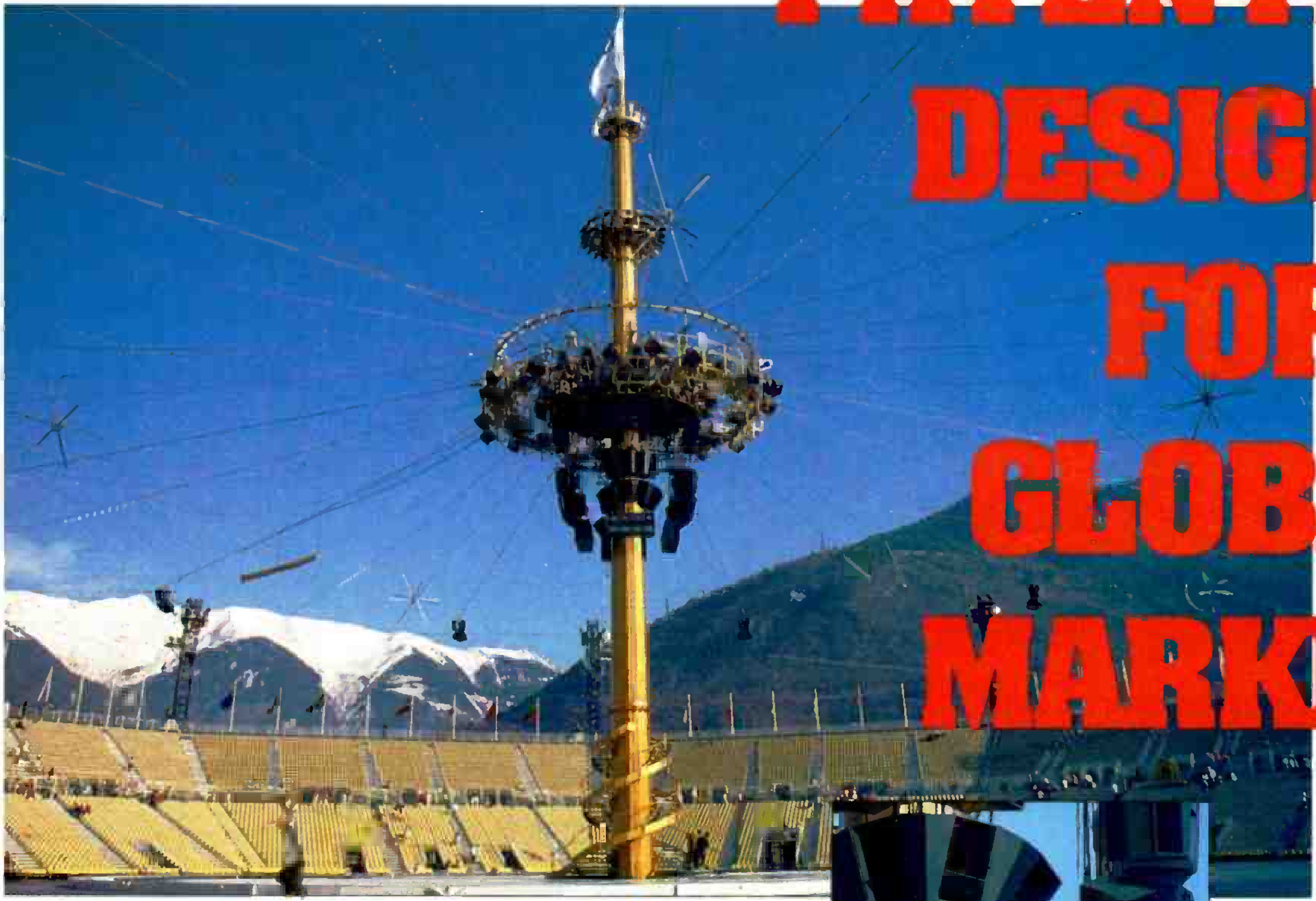
• Will the unit be used for years until it gives out, or will it be sold or traded off as part of a regular system upgrading or expansion plan in the future? [Just like automobiles, some brands and models of power amps have better resale value than others. If future resale is a consideration, this may influence your choice in suppliers].

An amp is not an amp is not an amp ... there is a tremendous variety of designs and models available from many different suppliers. Taking the time to closely examine a power amplifier's performance, features and construction can mean the difference between choosing the ideal unit that won't require much care and feeding, or in choosing a problem device that never quite delivers what was promised or is expected of it.

If you are involved in purchasing, specifying or using professional power amplifiers, it makes sense to be very clear on what you expect of this important part of your sound system. A good power amplifier may be practically invisible, and it may just be a piece of "wire with gain" that "has no sound of its own," but whether or not it is truly good for your needs will be based on the amount of time that you take to determine how well it meets your specific requirements. ■

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

U.S. SOUND: PATENTED DESIGNS FOR A GLOBAL MARKET



The 360° array of U.S. Sound enclosures covering the 34,000-seat outdoor 1992 Winter Olympic Games stadium. Total hung weight is 3.5 tons.

What do Madison Square Garden, Atlanta's Omni and the 1992 Winter Olympics all have in common? Here are the facts.

By David Scheirman

Sound system designers have never had it easy. First, there are the laws of physics. Then there are the size and weight limitations, truck packs for portable systems and rigging considerations for hanging the speakers. And no one really wants to hear about the mechanical engineering considerations to make it all work, because the sound of the system is what it's really all about.

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

Yet the very compromise that makes one part of it work (how it will fit into the truck, for example) is often the thing that hurts the acoustical performance. ("Hey, what happened to all the low end?")

One system designer who's looked at these things from many different perspectives for several decades is Cliff Henricksen. Formerly an engineer with Electro-Voice and Community, where he honed his transducer-developing and horn-designing crafts, Henricksen is currently the design engineer for U.S. Sound, a company that has been quietly working for the past few years to change the way that full-bandwidth audio is presented to large audiences.

Henricksen is no ivory-tower theorist. He owns a home recording studio, is a performing musician, and admits to liking funk, especially James Brown's music. With partner John Lemon, he has patented some innovative technology that is more than just theory. Working in New Jersey, the pair have already manufactured hundreds of lightweight, arrayable enclosures based on these new ideas.

BASIC SYSTEM DESIGN PRINCIPLES

Two patents have been granted to Henricksen and Lemon. One is related to lightweight enclosure construction

("Ultralight loudspeaker enclosures"); the other to a method of combining them with minimum multisource interference ("Method for large-scale multiple source sound reinforcement"). Both patents were granted by the U.S. Patent Office in 1989, and U.S. Sound has been hard at work constructing systems based on these ideas.

Ultralight loudspeaker enclosures. How light are they, one asks? "Nothing in this enclosure ever grew in a tree," responds Lemon, company owner. "We are using a material for construction of the enclosures that is more durable than wood and much less heavy." In fact, the average weight of a single low-frequency box, housing two large-diameter speakers, is said to be only 110 pounds. A mid/high box, handling all frequencies above about 160Hz, weighs even less — a trifling 98 pounds.

For comparison, check that against the "typical" 350- to 400-pound full-range touring sound enclosures. Putting a U.S. Sound low and mid/high box together would tip the scales at only 208 pounds, a 30% to 50% savings in weight over the standard, direct-radiating road box. Is this a fair comparison? Horn-loaded systems have long been known to be more "efficient" but many have argued that direct-radiating systems provide better fidelity, regardless of weight disadvantages or increased power requirements.

> *The patent states that one of the invention's objects is to "provide large-scale multiple source sound reinforcement without interference between any of the multiple sound sources."*

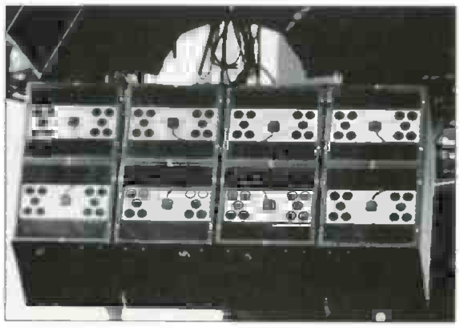
"Direct-radiating style systems require very careful component choice and placement on the baffleboard," said Henricksen. "The wavefront that develops from a flat baffleboard will rarely be phase-coherent, because of the multiple devices that are placed in close proximity to handle output in the same frequency band. I've chosen to work with horn-chamber technology over the years, knowing that the key to a lighter, more efficient and coherent system is going to have to come from the proper design and use of waveguides. The big step that we have taken is to use narrow-pattern devices, to maintain pattern control over longer distances."

Briefly, this new system is fully horn-loaded according to the designer's philosophy based on physics — exceptionally lightweight, because of advanced materials technology, and highly efficient, because of narrow-pattern, "long-throw" characteristics.

WHAT'S IN THE BOXES?

"That's proprietary," said Lemon. "It's a 3-way system. We use large-diameter motors for the low box and compression drivers for the mid and high sections. They are not parts that are available to anyone else. We had to develop special components for this application and we worked hard to find the right combination of materials, physical size and motor construction."

A key to understanding the mechanics of this system might be to note its heat-transfer characteristics. In fact, the company's granted patent goes to great lengths to describe the importance of "good thermal conductivity" and "thermal energy dissipation" in reference to the suggested means of



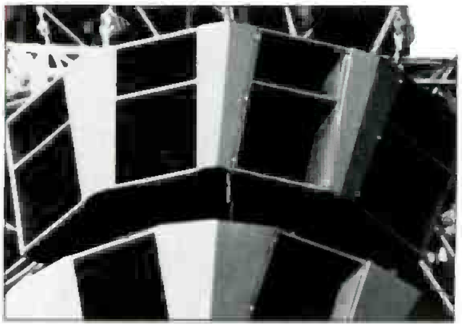
Four low-frequency enclosures, weighing 110 pounds each, use an innovative heat transfer system to avoid speaker thermal damage.

You'll notice that our system doesn't look as if it is built in the same way that others are and that it's not put up in the same manner.

enclosure assembly and construction. The low-frequency box, when viewed at close hand, appears to contain panels made of a heat-conductive material, presumably to allow the speaker motors to operate at high output levels without experiencing thermally induced failure.

The mid/high enclosure appears to rely on two deep-throated horn chambers, one for low/mid frequencies and one for mid/high frequencies. Although it is impossible to determine what is hidden deep within those horn mouths from a visual examination, we can surmise that the system would exhibit relatively good pattern-control characteristics within its intended frequency range, as do most long-throw horn devices.

U.S. Sound's second patent, "Method for large-scale multiple source sound reinforcement," gives us a clue that the system designers think that a minimum number of sound sources is preferable to prevent interference or "overlap" from adjacent sound devices. The



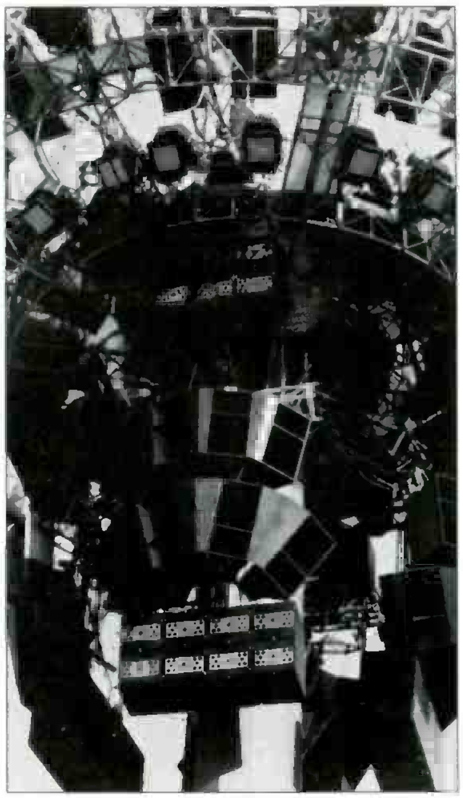
The deep-throat horn (or waveguide), narrow pattern-control mid/high enclosures contain a large-format compression driver and a smaller high-frequency driver each.

patent states that one of the invention's objects is to "provide large-scale multiple source sound reinforcement without interference between any of the multiple sound sources."

In part, this includes the limiting of horizontal dispersion coverage angles for each enclosure to a narrow (20° to 408) pattern. If the manufactured product adheres to the expressed design theory, then it follows that a single enclosure will use the minimum number of drivers, and an array will be constructed in such a manner that no two devices are directed toward the same part of the audience listening area.

Astute analysis of an array that follows this theory, as shown here, would note several unusual things: the method of placing the identical boxes into the array using joining "plates" or dividers, and the curving or "arcing" of the line of enclosures in a downward manner, similar to a rainbow. Does this have anything to do with the system's high articulation and apparently "seamless" sound coverage in the audience area?

"We've felt in the past few years that sound at a lot of public events and concerts was not all that good, and we noticed that there were certain similarities in the way the sound systems were constructed and put up, regardless of what company was providing sound services," noted Lemon as he demonstrated the system array in Albertville, France during the 1992 Winter Olympics.



Closeup of one of the four arrays mounted from the center-stadium production tower.

"There are some really good soundmixers out there and some great recording artists who do live concerts, yet things often sound unclear, harsh, just not quite right," he said. "Why not? What's the common denominator that contributes to large-scale sound reinforcement not being as good as it should be? You'll notice that our system doesn't look as if it is built in the same way that others are and that it's not put up in the same manner. You'll have to decide for yourself if it's a better way of doing things based on what you are hearing."

1992 WINTER OLYMPICS

In February, U.S. Sound was contracted to deliver a stadium-sized speaker system to Bose S.A.R.L. (Bose France), the company that was designated the official professional sound system supplier to the Olympic Committee for the XVI Olympic Winter Games. Telema, the French production company, subcontracted with U.S. Sound to handle sound reinforcement for the Olympic opening and closing ceremonies. The 34,000-seat outdoor venue needed full-bandwidth sound reinforcement, with 360° coverage.

"It's a fairly straightforward design process with regards to this speaker system," said Lemon. "Our pattern-controlled enclosures allow a designer to calculate what areas he wants to cover, and our arraying method ensures the most intelligible sound for that application."

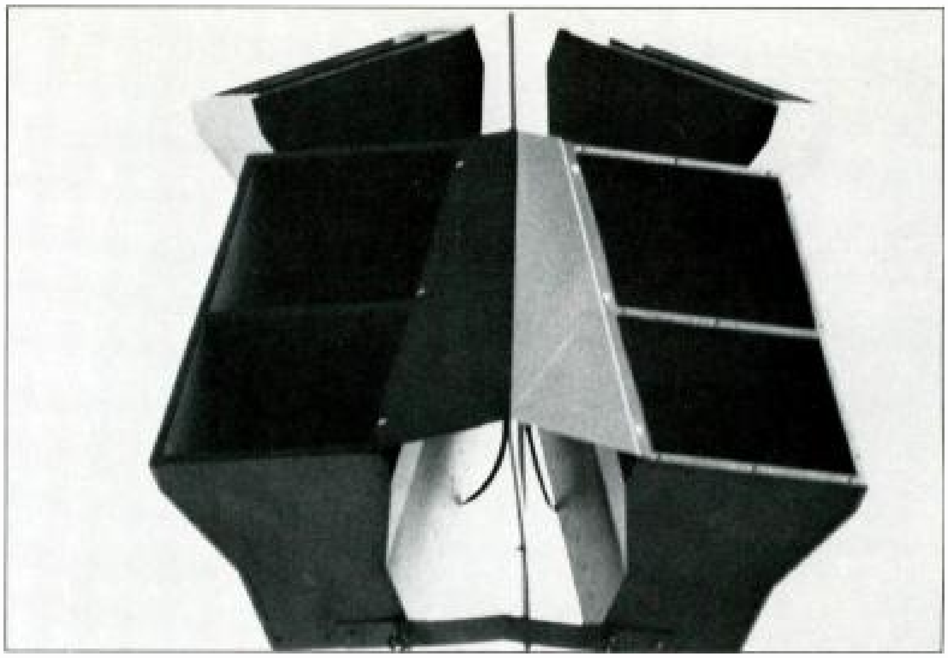
U.S. Sound is carefully considering how to best implement its new Sound Reinforcement technology for the world marketplace.

In Albertville, 64 of the company's boxes (32 low and 32 mid/high units) were suspended from a tower-like production mast located in the center of the stadium area. The total weight of this 4-quadrant array was only about 6,656 pounds, or less than 3.5 tons. "They had a serious weight restriction for this tower," said Lemon. "I cannot think of any other loudspeaker systems that could have accomplished the same thing here without being seriously overweight."

Additionally, smaller versions of the company's mid/high enclosures were suspended over the audience from Kevlar cables in a signal-delayed ring 50 meters from the center of the stadium. Other boxes were mounted atop the stadium's perimeter and were used as a source of special-effects sounds during the opening and closing ceremony programs, which featured a fully digital soundtrack playing the sound of wind, insects, bells, musical instruments and more.

CAREFUL CONSIDERATIONS

With a world-class special event and several prestigious permanent installations under its belt, U.S. Sound is carefully considering how to best implement its new SR technology for the world marketplace.



Smaller mid/high enclosures, with exposed bracing and baffling, were used at the 1992 Winter Olympics for special-effects signal sources around the stadium and suspended over the audience.

"We've sold the enclosures used in the Winter Olympics to the European sound contractor for the event," said Lemon. "And we have several major permanent-venue installation designs in progress. The Madison Square Garden and Omni systems have brought some pretty interesting customers to our door. There are another 200 enclosures in the warehouse waiting for our next project."

Beyond that, our next phase of research and development will make it even easier to ship large quantities of these enclosures in an even smaller cargo space.

Is the technology applicable to the touring concert world? Is such a radical horn-type system suitable for symphonic, jazz, rap, pop, rock and heavy metal music? Can the system be rigged to work as flying arrays in a live-concert environment? Are such lightweight boxes able to withstand the rigors of rock 'n roll trucking and transoceanic airfreight? These are real questions to pose about any new loudspeaker technology.

"We just finished our first prototype of a ruggedized road-version enclosure," said Lemon (as of March 27). "It does not sacrifice the weight advantage that we have now, but should give excellent protection for daily handling, truck stacking and shipping damage. Beyond that, our next phase of research and development will make it even easier to ship large quantities of these enclosures in an even smaller cargo space. And we are already having discussions with major concert sound companies and production managers, people who have real concerns about freight costs and total system weight based on their needs to do global touring."

There you have it, a speaker system intended for the year 2000 and designed to minimize interference between multiple sound sources. Easily transportable. Very intelligible. Although no touring versions exist, they soon will. It will be interesting to see how applicable they are to concert sound as we know it now. ■

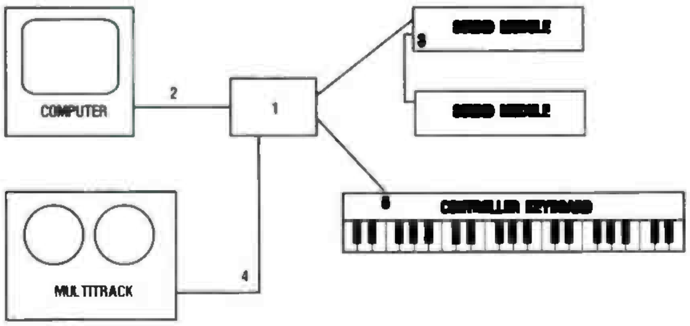
Editor's note: This feature has been written with reader interest and education in mind. The mention of specific brand names of manufactured audio equipment is not to be taken as an endorsement by the author or Intertec Publishing.

Continued from page 27

89. When you purchase sample libraries on CD-ROM, the sample data must:
- A. Have at least 16-bit resolution.
- B. Match the data format of the particular sampler.
- C. Load quickly into RAM.
- D. Be played in a drive that can also play compact discs.

90. When recording your own samples live, better results can be produced by:
- A. Recording many samples throughout the range of the instrument.
- B. Carefully editing the sample for good dynamic characteristics.
- C. Applying velocity-mapped sample blending, where possible.
- D. All of the above.

91. The number and size of samples that can be used simultaneously is limited by what factor?
- A. The amount of RAM in the sampler.
- B. The size of the hard disk.
- C. The number of MIDI channels in use.
- D. The clock speed of the sampler.



92. Figure 2 is a block diagram of a simple system for using a MIDI sequencer synchronized to tape. Identify item #1:
- A. MIDI thru-box.
- B. MIDI patchbay.
- C. MIDI/computer interface.
- D. Time code generator.

93. What kind of cable is item #2?
- A. Serial interface cable (usually RS-232).
- B. 50-pin disk drive cable.
- C. Audio cable.
- D. MIDI interconnect cable.

94. What connector is item #3?
- A. Time code out.
- B. MIDI out.
- C. MIDI thru.
- D. Audio out.

95. What kind of signal is carried on cable #4?
- A. MIDI system exclusive messages.
- B. Longitudinal time code.
- C. VITC time code.
- D. Tape deck control commands.

96. What connector is item #5?
- A. MIDI in.
- B. Time code in.
- C. MIDI thru.
- D. MIDI out.

97. For music sequencing with MIDI, which is more accurate in terms of timing?
- A. Dedicated hardware sequencers.
- B. Dedicated software sequencers.
- C. There is no difference.
- D. It depends on the machine.

98. MIDI note-on/off commands are used to:
- A. Change presets on MIDI effects processors.
- B. Trigger a note from musical instrument.
- C. Execute MIDI channel muting.
- D. B and C

99. Which of the following is not true?
- A. Two devices can operate (respond) on the same MIDI channel at once.
- B. Audio samples can be transferred via MIDI.
- C. Wireless MIDI data transfer exists.
- D. MIDI uses four wires to send and receive data.

100. Sysex software messages are:
- A. Used to open sequencer application programs.
- B. Used to command customized MIDI device presets.
- C. Used to make a MIDI music sequence quantize or conform to prescribed timing commands.
- D. None of the above. ■

I would like to thank the following people for their contributions to this article: Chips Davis, Del Eilers, Paul D. Lehrman, Tim Sadler, Rick Schwartz, Andrea Wetherhead, Anthony McLean and Mike Joseph. Special thanks to Barry Thornton and Bob Gilmartin for going beyond the call of duty.

National Audio Test Answer Key

Electronics/Tech	Acoustics	Recording Techniques	Digital Audio	Sampling/MIDI
1. C	21. A	41. C	61. C	81. B
2. D	22. D	42. C	62. A	82. D
3. A	23. D	43. B	63. B	83. A
4. A	24. B	44. C	64. A	84. A
5. C	25. C	45. D	65. D	85. D
6. B	26. D	46. A	66. B	86. C
7. B	27. C	47. D	67. D	87. B
8. A	28. A	48. D	68. C	88. D
9. D	29. B	49. B	69. C	89. B
10. D	30. C	50. D	70. C	90. D
11. B	31. B	51. B	71. A	91. A
12. B	32. C	52. C	72. B	92. C
13. D	33. D	53. C	73. B	93. A
14. C	34. B	54. B	74. D	94. C
15. A	35. D	55. D	75. A	95. B
16. A	36. C	56. C	76. C	96. D
17. B	37. A	57. D	77. B	97. D
18. C	38. D	58. B	78. D	98. D
19. C	39. B	59. A	79. C	99. D
20. D	40. D	60. B	80. B	100. B

Routine Maintenance

By M. Raymond Jason

The two parts of any ATR preventive maintenance program are: mandatory upkeep procedures, including demagnetizing, cleaning, lubricating and aligning; and exploratory performance testing to reveal progressive failure.

DEMAGNETIZING

Magnetized tape-path components partially erase tape, and magnetized heads degrade an ATR's signal-to-noise ratio. Heads can become magnetized by a dc current from faulty or poorly-designed upstream circuitry. Heads and other components can acquire magnetization from sudden accelerations (such as dropping) or from repeated exposure to rapidly-changing transport solenoid fields.

The cure for these ills is manual demagnetization of the tape path using a degausser, provided some critical cautions are followed. First, an ATR must be turned off before demagnetizing, because the induced signal from the degausser is powerful enough to damage head pre-amp input circuitry.

Second, degaussers with momentary on-off switches are dangerous, because mistakenly turning one off while it's touching a tape-path component can magnetize that component to a level beyond the degausser's demagnetizing capacity. For this reason, any switch on a degausser is a potential hazard, as are degausser power cords that are too short or in poor condition.

Third, effective demagnetization requires deliberate and careful movement of the degausser. Plug in the degausser at least three feet from any tape path. Bring it toward the tape path, moving slower the closer you get. Inside the tape path, never exceed a 1-inch-per-second movement. Do not hold the degausser up to a rotating component and then spin that component; instead, rotate the component slowly once or twice around. When finished, back the degausser off, not faster than 1-inch-per-second, until a foot away, then quicker if you desire until at least three feet away before unplugging it.

M. Raymond Jason is an electronic engineer at National Public Radio in Washington, D.C.

CLEANING

Three cleaning chemicals suffice for any tape machine: isopropyl alcohol, water and a mild alkali-based liquid cleaner (such as Formula 409). Isopropyl alcohol is the very best general cleaner to use: it is non-toxic (unlike methanol, which is highly toxic even through skin); readily available in pure form (unlike ethanol, which is cut with kerosine unless purchased prescription-grade, and unlike "rubbing" alcohol, which usually contains glycerin); its surface tension allows reasonably controlled application (unlike Freon, which is more likely to penetrate and so indirectly damage bearings); and it is available inexpensively, in bulk, from chemical suppliers (unlike special-purpose "head-cleaning" fluids, which by comparison are priced exorbitantly).

Water is useful for those pinch-rollers that degrade with frequent application of alcohol. Many pinch-rollers tolerate alcohol, but some tend to loose resilience. If in doubt, call the manufacturer for advice or simply use water. Alkali-based liquid cleaners work better than alcohol on ceramic capstans, guides and erase-heads. You must, however, finish by cleaning off the alkali-cleaner with alcohol.

Finally, if your facility can tolerate the extra expense of one-piece foam-tipped applicators, using them will provide three benefits: They don't contain glue, which on cotton-tipped applicators can dissolve and accumulate in the alcohol bottle; they are incapable of leaving behind fibers; and they are easier to wring-out after dipping and before use, to prevent alcohol migrating down into bearings.

ALIGNING

ATR service manuals all contain alignment procedures, so I won't use space here to duplicate that information. Here are some ideas, however, on achieving consistency in alignment.

In terms of obtaining and maintaining a "sound" for a facility, it is perhaps less critical precisely how tape machines are aligned than how consistently they are aligned from session to session and from one machine to another. All your ATRs should be aligned consistently to the same reference level tape. To keep azimuth consistent, use the same piece of tape for all azimuth alignments, or, alternatively, purchase a full reel of azimuth sweep or tone and cut it onto smaller reels, splicing it into your other alignment tapes.

Because few ATRs exhibit perfect EQ performance, bias and EQ alignments usually entail compromises. All engineers responsible for aligning your ATRs should agree upon a common alignment technique. This means that all your engineers should employ the same biasing procedure and go for the same biasing criterion, and that they should all set high- and low-frequency repro- and record-EQ in the same way. Ideally, you should not be able to distinguish the sound of one machine from another when each records and plays back pink noise. This test is an excellent check between formats, as well, such as ATR to VTR, etc.

There are some simple tests and observations you can make to find problems before they become big problems. Check power supply voltage and ripple to catch drifting dc before it causes a logic glitch, and to catch dying capacitors before they give up the ghost. Listen for noisy bearings so you can lubricate or replace them before they produce wow and flutter or total transport failure. Measure wow and flutter often. Many sources of it, including bad motor or tachometer brushes, bad bearings and aging pinch-rollers, are progressive in their negative effects and can often be caught before operators notice trouble.

SCHEDULING MAINTENANCE

Operators should be trained to recognize improper transitions between transport modes, tension problems, audio problems, noisy bearings and misadjusted brakes.

They should also know about demagnetizing, electronic and azimuth alignment, checking tensions and brakes, inspecting for head-wear and pinch-roller condition, exercising switches and checking for bearing noise.

The third tier is exploratory testing and should be performed every two to three months. Check all power supply voltages and ripples; perform a wow and flutter measurement, check pinch-roller pressure; EQ circuitry and head condition with a flux-loop measurement; check bias frequency and erasure depth; and lubricate bearings according to the ATR service manual's schedule. ■

Writing on the Wall

By Rick Schwartz

If you attended the recent NAB show in Las Vegas, you probably already know about Scenaria — the new post-production workstation/mixing system from SSL. What sets it apart from the pack is its ability to record video as well as audio. This breaks a long-standing post-production tradition. Since the departure of the optical soundtrack in the '50s, sound and picture have been treated as distant relatives. As a result, tremendous amounts of time are spent matching one to the other. It doesn't matter whether you work with film or videotape, every time picture cuts are made, the audio tracks need to be conformed to match.

Today, we have the technology to bring high-quality multitrack audio and video back together at a reasonable cost. Witness the current generation of products from Avid and others. Multimedia may still be an over-hyped buzzword, but desktop video is starting to gain widespread acceptance. Let us not forget that these non-linear video editing systems have high-quality digital sound and powerful audio editing capabilities. Non-linear video editors could do to audio 'post' rooms what the project studios have done to music recording rooms. We need to ask ourselves what good is random-access audio without random-access video? A conventional VCR takes more than a minute to a locate point that would take less than a second on a hard disk-based system.

GETTING INTO THE ACT

So how does one go about integrating digital audio with picture? The multimedia craze has created off-the-shelf technology that allows us to incorporate digital video at a reasonable cost. There are four things we need to integrate video into our digital audio workstations: live NTSC video-in-a-window, true color display, real-time data-compression and a standard file format for storage. Although it's possible to buy a single card that incorporates them all, we will look at each separately.

COMPUTER TV

The best way to get your feet wet is to buy a card that allows you to display live NTSC video on your computer monitor. A SuperMac Video Spigot will digitize full-motion video for less than $500. The card simply maps over selected pixels on your screen with the converted NTSC image. The picture appears in a resizable window that can be placed anywhere on your screen.

TRUE COLORS

Although most computer monitors display color, they were not designed to display full-motion video. 24-bit video is needed to accurately display the colors of a photographic image. Why 24-bits? It matches, and in some cases, exceeds our limits of color perception. It is the difference between 256 (8-bit video) and 16.8 million colors. Without it, you can see 'steps' in a color transition or blend. 24-bit cards are available for less than $1,000 from Radius, RastorOps and others.

MAKING MOVIES

Now that we can display high-quality full-motion video, we need a way to play it back from disk in sync with our audio tracks. The key to storing video on disk is the use of data compression. One example of such a technology is QuickTime from Apple Computer (although the name is a mystery to me – QuickTime is about as 'quick' as a HyperCard is 'hyper'). QuickTime is more than a way of playing full-motion video on any Mac (with an '020' processor or higher). It is a software architecture that encompasses a wide range of elements.

HARDWARE HURDLES

Normally, QuickTime decompresses data using your computer's CPU and software algorithms. This is why sound and picture are sometimes jerky. Anyone who saw an early demonstration of QuickTime was most likely less-than-amazed. It's hard to get excited about grainy video in a window the size of a matchbook running at 10-12 frames per second with jumpy distorted audio. When you playback a digital video image, QuickTime works to retrieve the stored data at the best rate your hardware can provide. With slow hardware it may mean that QuickTime can only display every eighth frame. The frame rate of playback is limited by the speed of your hardware. Don't expect 30fps playback from a Macintosh LC. Fortunately, the architecture used in QuickTime allows the use of plug-in codes that can be hardware- or software-based. Using a third-party card with hardware data-compression, full-screen 30fps playback should be possible on most Mac computers.

STORAGE CAPACITY

Even with data compression, you still need a fair amount of space on a hard disk for video. A 10fps QuickTime movie with 11kHz mono sound eats up about 6Mbytes per minute. Imagine, then, how much space full motion video would require.

NOT SO QUICK

The QuickTime toolbox should make it easy for software developers to incorporate QuickTime movies into their applications. Most likely the QuickTime movie would be recorded using third-party software and simply imported into your EDL software. Until then, we will have to live with real-time video in a window. Although most QuickTime movies have limited audio capabilites (22kHz 8-bit mono), RasterOps makes a card that digitizes 16-bit audio at 44.1kHz, right along with 24-bit digital video. Keep in mind that most of the time production sound on tape is only used for reference purposes — 48kHz stereo sound is not needed to phase two audio tracks together. What is more important is that the sound from videotape is digitized to create a visual waveform that allows sample-accurate editing.

CROSSING PLATFORMS

QuickTime movies can be converted to formats that will play on IBM PCs and other platforms using MacroMind Director software from Paracomp. Avid recently announced their Open Media Framework (OMF), a common interchange format, that allows the contents and description of edited programs to be shared between different systems. NED plans to support this standard with their Post-Pro SD. Digidesign is also teaming up with Avid on a 4-track audio editor. Hopefully it will not take other manufacturers long to follow their lead. ■

Rick Schwartz is a contributing editor to R•E•P and director of The Post Complex in Studio City, CA (formerly Music Animals). He can be reached via CompuServe at 70672,1377.

R•E•P: On-Line

Harping On-Line

By Tim Sadler

Clearly the issues surrounding the home/project studio vs. the commercial studio are still on everyone's agenda. Here are some excerpts from the R•E•P: On-Line forum that demonstrate the depth of feeling and the validity of arguments on both sides of the fence.

Message: #9729, S/14 REP Magazine
Date: Sat, Jun 6, 1992 6:00:05 p.m.
From: REP 75300,3141
To: Tim Sadler [REP] 75300,3142
Tim–

Re: Home vs. Pro studios ... It has now gotten to the point where L.A. HARP members are reporting the home/project studios to the zoning officials ... Hearing about the financial hoops they must jump through to satisfy L.A. zoning codes, it is easy to empathize with the Pro side of the story. Maybe what is required is a level playing field.
– Anthony McLean, Features Editor of R•E•P Magazine

Message: #9825, S/14 REP Magazine
Date: Mon, Jun 8, 1992 11:19:27 a.m.
From: Tim Sadler [REP] 75300,3142
To: REP 75300,3141
Anthony–

If you define a "level playing field" in terms of everybody obeying the law, no question, I agree, but ... the advantage of many home/project studios is the very fact that the playing field is being redefined every time some smart vendor gives the home market access to technologies or capabilities previously out of reach. The concept that many (by no means all) large commercial facilities fail to recognize is that to stay competitive they must become much more flexible, able to reconfigure their rooms and reposition themselves in the market as the playing field changes. If one looks at how this dynamic is playing out in other industries, the trends are clear. Those who can make rapid adjustments are the winners. Even as we speak, how many new rooms are being designed with the consideration that they may be totally reconfigured within 24-36 months? It is more expensive to build-in this flexibility up front, but it will more than pay off when the update occurs.
– TS

Tim Sadler is president of IntraMedia, a media software consultancy and Sysop of R•E•P: On-Line. His CompuServe address is 75300,3142.

Message: #9975, S/14 REP Magazine
Date: Wed, Jun 10, 1992 2:11:26 a.m.
From: Jeff Lorenzen 70262,2441
To: Tim Sadler [REP] 75300,3142

Interesting, don't you think, the fight going on between HARP and project/home studios. All for the love of money and artistic freedom. I've been unclear as to many of the reasons HARP uses to forcibly shut down home studios (with the exception of zoning laws). It seems that they're just pissed about not having the corner on the market that they used to because of broadening technology. What about the artist, songwriter, or film composer that has a studio in his den in which he or she works quietly on his or her own stuff?

What if you're Springsteen and you just rent a pair of Sony 3348's, a Neve and a load of gear and set it up in your 800-square-foot library? Is *this* a commercial studio? There seems to be a grey area between "office-in-home" and "commercial work-place." Who's making the decisions on what is what? What proof are they obtaining? ... I think that HARP is going to find themselves giving in a few years down the road. The technology is going to become too great and too inexpensive, and the market for home studios will increase 10-fold. I bet IBEW and the record labels fought tooth and nail back when garage studios started to pop up. I'm willing to bet studio time that a few of those "garage studio" owners went on to own Ocean Way, Sunset Sound, Conway, or Record Plant. Maybe they ought to think about where they came from before they start stepping on toes.
– JL

Message: #10006, S/14 REP Magazine
Date: Wed, Jun 10, 1992 5:30:24 p.m.
From: Tim Sadler [REP] 75300,3142
To: Jeff Lorenzen 70262,2441

The technology is going to become too great and too inexpensive that the market for home studios will increase 10-fold. Can't argue with your math!
–TS

Message: #10035, S/14 REP Magazine
Date: Thu, Jun 11, 1992 3:24:14 a.m.
From: Bob Olhsson 72340,17
To: REP 75300,3141

The city of Los Angeles has outrageous parking space per square foot of commercial space rules that make it very expensive for businesses that require a lot of square footage like studios. To match the rates in the home studios, the studios located on commercial property found themselves being literally asked to pay people to use their facility. After seeing a number of major, top-quality, honest studios go under, a group of studio owners got together to demand an even playing field. They would have been just as happy to have the zoning requirements eased, but they couldn't see being forced to compete with one hand tied behind them by the city.
–Bob

Message: #10036, S/14 REP Magazine
Date: Thu, Jun 11, 1992 5:36:03 a.m.
From: Kevin W Smith 76334,3232
To: Jeff Lorenzen 70262,2441

You're absolutely right about all this HARP junk. It's the Spanish Inquisition of the recording business. But I note that most of these "residentially orientated" studios are still going ... This has already been discussed here I'm sure, but what'll kill The big L.A. studios are shrinking recording budgets, not living rooms with SSLs. The meat and potatoes of the recording business, even for so-called high- end studios, is midline and new artists that either don't exist anymore, or just don't have $1,500+ a day to record with.
– Kevin

Message: #10040, S/14 REP Magazine
Date: Thu, Jun 11, 1992 8:19:11 a.m.
From: Tim Sadler [REP] 75300,3142
To: Kevin W. Smith 76334,3232
Kevin–

Shrinking budgets are the *reason* for living rooms with SSL's.
–TS

The changing modes of product support and engineering pay scales were also very active topics last month. We'll cover these subjects soon. See you in the ether. ■

Now You Hear it, Now You Don't

By Murray R. Allen

One of my closest friends is a fellow saxophone picker. He and I agree on almost everything except how a recorded sax should sound. We use the same type of horn, the same mouth piece and even the same reeds. When we play in a club together we actually sound quite similar, except that he has better ideas. But when it comes to recording and mixing, we share no common ground. I see the same differences among string, rhythm and horn players. I have a theory that in every decade the way in which sounds are perceived gradually changes, and ultimately, vast differences in our listening judgement develop.

Back in 1974, while recording the Stan Kenton Band (the "Fire, Fun and Fury" album), Stan hated my drum sound. He kept saying that, "drums don't sound like that." The producer, Bob Curnow, and I won out even though I knew what Stan was hearing. He liked the sound of drums as they were recorded in the Capitol Records studios in Hollywood 20 years earlier.

Stan Kenton worked in big ballrooms and night clubs all over this country. He usually sat about 20 feet or so from the drummer. When the band played, he heard the drummer's sound bouncing off the walls. Some of the snare was absorbed by the drummer and drum kit, and the bass drum filled the room like a sub-woofer. That's what he wanted to hear.

What Bob Curnow and I wanted to hear was a more "modern" drum sound, maybe the way a drummer would perceive his own sound. Peter Erskine's drums sounded great and with close miking we nailed our "sound." Even though the record was nominated for a Grammy, I know Stan would have preferred his perception of drums compared with ours. I believe that much of what we want to hear on a record is influenced by what we hear in real life.

Murray R. Allen is president of Allen & Associates, audio director for the Oprah Winfrey Show, audio designer for the Grammy Awards 10 years running, and a former president of SPARS.

When I started out in this business, music was performed without the use of microphones — except for the singer, and even then, singers with exceptionally big voices would often go acoustical. When Bing Crosby recorded "White Christmas," there were only two microphones used, one on Bing and the other on the band. Unfortunately, the microphone on the band was not working, so the entire record was recorded with only one mic. When people went to ballrooms, that was pretty much what they heard. That sound was accepted and at last count I believe Bing's record had sold close to 30 million copies.

It really pleases me when I hear a younger patron remark, "Gee, that's a new sound."

Throughout the early days on Broadway, all performances were totally acoustical. Then the lavalier microphone was invented and it was suggested that weaker singers could be helped by this deivce. Now, several decades later, every singer and every instrument on Broadway is electrified. When someone goes to a big musical today, they expect to hear a record-like sound. Personally, I hate this use of electronics. When an actor walks across the stage and his voice stays in the same place, I find it to be less than a dramatic treatment for a "live" show.

This use of electronics even leads to lip-syncing, which is far from "live" no matter how you shake it. I also feel that it is a shame that future generations may never hear an unamplified pit orchestra.

In the middle '60s, the Fender Bass began to replace the acoustic bass. Everybody loved it, because it was always difficult to mic a string bass and there was always leakage from the drums floating into the bass player's mic. But the acoustic bass was not to be put out to pasture. The contact mic came along and we could make an acoustic bass leak-proof. With the exception of a few traditional bass players, such as Harry Connick Jr.'s Ben Wolf, the sound of acoustic basses today is usually a combination of electronic devices and acoustic mics. Personally, I like this arrangement because it gives you the best of both worlds. But many old jazz musicians shake their heads and say, "That's not what a bass sounds like."

In recent decades, the boutique studio evolved. With the proliferation of the multitrack recorder, it was possible for all the musicians to play at different times. This really made recording "leak proof." The new found isolation opened the door for all different kinds of sound tinkering. Pianos could be highly compressed. Drums could be gated. Instruments began to sound "synthesized," as opposed to what musicians really heard in live performance.

Boutique studios, with smaller rooms and low ceilings, created an alteration in the recorded sound of strings and horns. The so-called "tight" mic sound on acoustic instruments, minus all leakage, became the norm rather than the exception.

In live performance, it was difficult to recreate these new sounds. Enter MIDI and the sampler. Then you could play an acoustic piano and make it sound like a steamship. So John Q. Public goes to a concert and hears highly amplfied sound created by musical plastic surgeons. (I always loved the comment that "those toms sound just like cannons." We recently scored a film and we used toms to create the sound effect for cannons. "Man, those cannons sound just like toms!")

There are still a few clubs that have un-amplified jazz groups. It really pleases me when I hear a younger patron remark, "Gee, that's a new sound." Personally, I feel that every song demands its own sound. The diversity of music includes the best sounds from history, making possible such hits as "Unforgettable" and "Silent Lucidity." I hope that folks in the music business keep their ears open, and don't get trapped by limiting their live listening experiences.

As for my buddy, the saxophonist, he likes his sound slightly compressed, a little more bottom and top than comes natural, and a slight delay accompanied by some digital stereo reverb. As for me, I like an analog tape delay through an EMT plate. ■

The Society of Professional Audio Recording Services is the industry's best source of business information. For details on activities or membership, contact SPARS at 4300 10th Ave. N., Lake Worth, FL 33461; 407-641-6648; fax 407-642-8263.

HANDS ON:

EVENTIDE H3500

By Barry Gibbons

I'm right in the middle of a session and UPS delivers Eventide's latest device, the H 3500. Coincidentally, the client is sitting with his arms folded, waiting for the work station to time-correct a 1-minute, 3-second voice-over down to a minute. Question: Do I chance pulling the new unit from the box and, perhaps, make myself look like an idiot (Gosh, I don't understand why it won't work ... uh, gimme a second to look through the manual)? Fear has no fear! I had the 3500 sitting on the meter bridge, ready for operation within a few minutes.

With a fair amount of ease I loaded a sampling preset and began dumping the information in. A couple of taps on the parameter button and I was compressing the program down to where I needed it, and *without peeking in the manual!*

Quite honestly, I pushed the time compress value much further than needed, so I could rid myself of the sense that it wasn't working. The result? Very impressive! The 3500 is easy to use and the learning curve is much less than other gear with similar functions and features.

DO-RE-MI

The H 3500, also called the "Dynamic Ultra Harmonizer," has a solid feel. Corners definitely haven't been cut on buttons, keypad or wheel quality. The front panel has a 2-line x 40-character LCD that is clear and crisp from all but very sharp viewing angles. In the function mode, the Harmonizer wheel can be used to modify the display. The back of the box is quite simple and uses Cannon XLR connectors for interface. Of peculiar interest is a transistor heat sink that protrudes from the back left shell. The cool running temperature of the 3500 is evidence that this design works efficiently, requiring no forced air.

I'm not sure why, but I was surprised to find the 3500 looks like the 3000 SE's twin brother (paternal in this case). Those of you already love smitten with the 3000 will feel at home with the 3500. From the "soft keys" to the "Harmonizer wheel," the unit is the same. In fact, the 3500 houses all of the processing found in the 3000 SE. One difference, according to Eventide, is the front plate. The 3000 series are made of a sheet metal and the 3500 uses steel, making it more road-worthy.

The 3500 is a true stereo processing device with differentially balanced inputs and outputs. The inputs are factory set to +4dBm and can be changed to -10dBm by simply removing the top cover and transplanting a couple of jumpers. Eventide recommends that users run input levels as high as possible before clipping to increase S/N (which is quite good to start with). I found this to be helpful as there are some presets that have minor noise artifacts caused by the nature of the processing.

The H3500 dfx provides 11.8 seconds stereo and 27 seconds of mono sampling. For $1,000 more the H3500dfx/e delivers 47.5 seconds of stereo and 95 seconds of mono sampling. With 16-bit converter resolution, these units sample at a 44.1kHz rate, with a stated frequency response of 5Hz to 20KHz, ±1dB. The 3500 uses 32-bit internal architecture and is fitted with Sony A/Ds and Burr-Brown D/As.

Users of the 3500 will find the manual easy to read and informative. It is written in a relaxed style and has a logical flow, keeping the reader from working too hard. A quick reference chart would have been helpful. This would be a welcome addition for those of us in the "play before read" camp, or as a console-side primer.

PIECE OF CAKE

Let me emphasize how easy the 3500 is to operate. For most people the manual will end up in the reference closet rather than the space next to the outboard rack. This is not to say, however, that the box is simplistic by any means. Eventide allows the user to get into very complex patching of filters, delays, LFOs (I think I spotted one late last night), envelopes and so forth. Initially, one of the biggest challenges comes in trying to remember what the hundreds (450) of presets do.

Barry Gibbons is a song writer, producer and engineer. He owns and operates Platinum Productions/Lab Studio in Salt Lake City, Utah.

To facilitate this, Eventide has wisely included an "origin" parameter that allows the user to trace the preset back to its mother algorithm. The operator can then refer to the manual to see what is actually occurring in the signal path and what options may be available.

The 3500 begins with 23 basic algorithms, four in addition to the 19 found in the 3000 SE. Of those four, there are two "Mod Factory" algorithms with extensive patching schemes and two powerful sampling algorithms, one mono and one stereo. These four algorithms were previously available as two diFfferent option packages designed for the 3000 series. Together, the 23 algorithms can be loosely grouped into six categories: pitch shifting, delays, reverbs, patch factory (a do-it-yourself approach to preset building), special effects and sampling.

MULTIPLE EFFECTS

Eventide plays second fiddle to none when it comes to pitch shifting. Arguably, the company invented the game. Its latest unit has a potential range of three octaves, both up and down. It allows "coarse" adjustments in half-step increments and "fine" adjustments in "cents." And just a button push away are 1,400 milliseconds of usable delay time with feedback control.

One helpful feature in the pitch shift algorithm is the "sustained loop." As an example of its use, I came across some flat flutes while pulling a mix recently. The parts were just far enough out of tune to make an otherwise beautiful recording sound awful, no matter what level I ran them at. Using the "sustain loop" feature, I captured a note and tweeked the "fine" adjustment until the pitch settled in tune. I opened up a couple of tracks and dropped the flutes back down. They sounded great! Of course, I could have just processed the original tracks during the mix, but I didn't want to employ the box as a "band-aid."

Many of the delay type programs on the 3500 have some unique qualities that bear mentioning. First, most of the algorithms make available the use of delays, meaning it is possible to pitch shift and create delay effects at the same time. On top of this, many include a high and low pass filter so you can shape the sound to taste.

One of the features integrated into most of the delay type presets is a soft "tap" key for automatically programming in delay times. Three taps at a desired note value and you're locked in. Also, many of the presets allow you to manually enter beats/minute.

Although not an original idea, another welcome feature included in the Mod Factory presets is dynamically responsive effects. Vocals sound much cleaner when the delays are dynamically attenuated until no input signal is present, allowing the delays to spill into the holes. I found the same to be true of reverb effects. The 3500 allows the user to assign chosen amounts of dynamic response to its many parameters. The dynamic presets are useful in keeping a mix clean while using some aggressive effects.

The 3500 is easy to use and the learning curve is much less than other gear with similar functions and features.

I remember seeing Tommy Bolin in concert a number of years ago where he used an Echoplex to record a loop. He would then cut these amazing solos over the top. The 3500 would have been helpful to him. Not only is it many times cleaner than 1/4-inch tape with several passes on it, but it offers a loopable delay time of 32 seconds. The 3500 would also be useful in building ambience tracks and putting Rap grooves together. Looping is a big part of clean pitch shifting, and Eventide excels at it.

VERBS

As a reference for reverberation, I set the 3500 up next to a Lexicon 480L and a 300. To generalize, the Eventide 'verbs were overall brighter and slightly more grainy. I wasn't able to find a preset that was as warm or smooth as either of the two Lexicons. As far as simulating natural acoustical environments, the 3500 also came in second.

On the up side, I could get the 3500 'verbs to cut through a mix where the smoother units wouldn't. Among other things, it sounds nice on drums. Realizing this, Eventide has included a long list of reverbs designed for drums and percussion. At a mix date, I used the "gated kick" preset to liven up an otherwise drab kit and had great success. I also found several other presets that made toms and snares sound rich and fat.

As a closing note on the 3500's reverbs, they are among the most creative that I've ever heard. One gets the impression that many of the presets were meant to be anything but natural sounding. If you're into creating something out of the ordinary, there's no shortage of options here.

The 3500 also has several onboard special effects. From the "Jet Fly By" to the random "Stutter"(Max Headroom)

SPECS AND DESCRIPTION:

Inputs:	Stereo, true differential balanced
Outputs:	Stereo, differential, transformerless
Dynamic Range:	Greater than 92dB "A" weighted
Distortion:	.01% (.007% typical) @ 1kHz, 1dB below clipping in "pitch change" mode, 0 shift, levels all at 0dB
Sampling:	Full 16-bit resolution at 44.1kHz sampling rate
Frequency Response:	5Hz to 20kHz =/-.5dB typical
Delay:	Up to 23.7 seconds or 95 seconds with H3500 - dfx/e
Pitch Variation:	3 octaves up, 3 octaves down
Power:	75W, 110-130V, or 200-240V, ac 50/60Hz
Size:	Inches: 3.5h x 19w x 13.5d Centimeters: 8.9h x 48.3w x 34.3d
Weight:	13 pounds net, 18 pounds shipping weight ■

preset, there are "toys-O-plenty" for production room gymnastics. Many of these effects are especially well-suited for broadcast use.

TRACK 24 GOING ON 25

I'd go without shoes all winter to get my hands on the sampling package of the 3500. Not only is it easy to use, but it sounds incredible. Here's an example of how it works:

Let's say you want to fly some vocals around. Sample the information into the 3500 and then choose the edit parameter. Using the Harmonizer knob, scrub edit the material to find the start and stop points. The display will show you where you are to the nearest millisecond.

Far beyond simple patch changes, each parameter in the 3500 can be automated via MIDI control.

Now let's say the attack is too abrupt. Tapping the parameter button three more times calls up a window that allows us to adjust the attack and release to smooth things out. We are then given three options for flying the vocals back on to tape: triggering by hand, using an input signal, or assigning a MIDI note number. With a little more esprit de corps, we could even pitch shift up a half step, use time correction, or fly in a modulated last chorus.

MY FAVORITE THINGS

The H 3500 is a creative tool and it's tough to decide where to use it. While mixing several sessions, I found myself frustrated that I didn't have a few more of the units available. Owners of the 3000 series beware! You'll find yourselves "puppy dogged" into feeling like you can't live without the extra features on the 3500.

Examples? A trick learned years ago is to take two PCM 42s and modulate them against each other for a great stereo chorus. Preset 763, "Moving Vocal Spread," grabs you by the ankles and body slams you. It really sounds terrific! This preset gives acoustic guitars a transparent shimmer and has just enough body to give a vocal life. It also sounds great on bass and keys. Almost everything that was run through the preset sounded richer. I found myself using this preset all the time. (Operator hint: Pull the function generator modulation down just enough to make the effect a bit more subtle.)

Dialing up "Sweep 8" turns a bare kick and snare groove into a widespread rhythm machine. This is not your father's delay line! The "Stereo Tom Delay" sounds great, too. It follows the toms through the stereo spectrum, or you can reverse the image for contrary motion. If it's placed properly in the mix, it'll raise the hackles.

THE EDELWEISS STRE-E-ETCH

The extensive MIDI implementation developed for the 3500 is quite impressive. When it comes to patch changes, there are many more presets than available MIDI numbers. Eventide has circumvented this by assigning each set of 100 presets to a different group. (For example, group 1=100-199, group 2=200-299 and so forth.) You choose the group you want and the patch change information will do the rest. It is also possible to change group numbers via MIDI.

How seamless is a patch change, you ask? Well, the machine appears to go into the bypass mode (technically a digital zipper) for a few hundred milliseconds while loading a new preset. This is because of the diverse nature of the algorithms. Some are fairly basic and others are incredibly complex. If a processor is loading algorithms too slickly, we can suspect that there are few, if any, differences in the algorithms. The transitions are fairly smooth and, unlike some other boxes, there are few ugly artifacts in the transition. Obviously, changes are smoother and quicker when moving between presets with the same algorithm.

Far beyond simple patch changes, each parameter in the 3500 can be automated via MIDI control. Think about being able to set up beats/minute by tapping a damper peddle. Simply assign any parameter to a source (volume, pitch bend, etc.) and you're ready to roll. It is also possible to assign several parameters to a single source. For example, the Mod Wheel could be used to change both the left and right delay times while increasing the feedback. This could be recorded on any sequencer and automated in a mix. The possibilities are pretty exciting.

One of the most creative tools included in the 3500 is the function generator. You can think of the function generator as being an LFO with 19 different periodic and triggered waveforms from which to choose. These can then be assigned to modulate any of the parameters in the 3500. Say you want to vary the amount of pitch shifting. No problem! Just assign the left and right pitch shift to the function generator. You can then punch or dial up the amount of modulation at the desired rate. I was amazed at what this would do to a static pitch shift. The pitch shift creates an image, but the function generator gives the image life. I rediscovered keyboards I've had for years just by trying "3500" patches that use the function generator.

Dialing up "Sweep 8" turns a bare kick and snare groove into a widespread rhythm machine.

The MIDI monitor function is especially handy, as it allows users to view MIDI notes flowing to the Harmonizer. Such status information should minimize the Rolaids consumption of engineers and keyboard players managing extensive 3500 MIDI implementation.

ON BALANCE

On a more critical note, it would have been nice if Eventide would make a model available with digital AES/EBU ports, allowing us to use the power of the 3500 in the digital domain.

SO LONG, FAREWELL

At $3,495 for the H3500 dfx or $4,495 for the dfe/x, I'll take two to go. This is a substantial processor. Unlike some other companies that can't seem to do anything but "warm up leftovers," Eventide offers something unique. Rarely does any gear come along that inspires such creativity. If that sounds a bit too theatrical, give one a try.

Eventide is to be congratulated on making such a sophisticated processor accessible to users on all levels. The 450 presets cover just about every conceivable need and can be a great starting point for experimentation.

Whether you're involved with music production, post-production or broadcast, the 3500 will be a valuable addition. It does a lot of things very well, some things no other processor will do, and its shortcomings are few. Any current financial squeeze not withstanding, the 3500 is a solid value and a tool that will get used regularly. ■

Circle Number (100) on Rapid Facts Card

First Look

By Laurel Cash-Jones and Fred Jones

CD'S, THEY ARE A'CHANGIN

Pioneer Communications of America is introducing the CAC-V3000, a double CD player that is also a 300 disc autochanger. Up to 32 of these little beauties can be daisy-chained together and controlled by your computer via RS-232C or RS-422A ports, thus giving you control of an astounding 9600 discs at one time. The most interesting feature is that the CAC-V3000 can crossfade between discs. It does this by having two complete laser assemblies.

Variable volume control allows smooth crossfades, and a variable speed playback allows for even more flexibility in programming and production. The unit also sports digital outputs via RCA connectors and analog stereo and mixed mono RCA outputs. I don't know why there are no XLR connectors on this professional unit. Perhaps it is also designed as a Karaoke machine.

Pioneer claims that the CAC-V3000 is primarily designed for on-air use at radio stations, but wouldn't it make an incredible home stereo? Seriously though, you probably don't need this unit in your home. However, if you have any need of a large library of CDs on line at any one time (such as sound effect and music libraries, etc.) this unit is a wonder

Circle Number (101) on Rapid Facts Card

THE BLUE LINE IS NOT JUST IN L.A. ANYMORE

AKG is introducing a new modular microphone system called the Blue Line. This system is comprised of one preamplifier and eight different microphone capsules with a full range of dedicated accessories. The Blue Line also uses a new positive action bayonet coupling system called ModuLock. Each capsule has a distinct slotted sound port for each pattern that provides both visual and tactile confirmation of the type of pattern selected. Think of it. Now you can put the right type of capsule on the preamp in the dark.

This new system uses the latest industrial materials, designs and construction techniques. The electronics have been made more reliable by the application of High Density Surface Mounted Device (HDSMD) technology, thus making for a lighter and more compact system. A new active transducer capsule design called TransAct, the HDSMD technology, and transformerless output stage deliver higher sensitivity, lower self-noise, better phase and frequency response linearity, lower distortion and lower current consumption.

The AKG Blue Line features two cardioid mic capsules, the CK 91 and the miniature cardioid CK 97-C; a hypercardioid capsule, the CK 93; two omnidirectional capsules, the CK 92 and miniature CK 97-0; a figure eight, the CK 94; and a short shotgun, the CK 98. The miniature cardioid is also available on an integrated 15-inch gooseneck as the CK 97-CVR. A complete range of accessories is available.

Circle Number (102) on Rapid Facts Card

GML HAS A BABY!

GML proudly introduces to you its new 12-input module, line level mixer. It can be used as a side-car expander or a stand-alone mixer with 12 inputs and four buses. Plus, it has four auxiliary sends and direct outs. Solo, Mute, and Insert buttons are just above the precision tapered master rotary level control. Metering is 30 segment, Peak and RMS, with either dot or bar mode optionally available.

It's cute too, with a black anodized aluminum chassis and silver lettering. We think that they might have come up with a nicer name than the HRT 9100 for it though. Why not George Jr.? Oh well.

As you might have guessed, it fits into a rack very nicely, so you can take it with you where ever you go without a stroller. Weighing in at 10 pounds, this lightweight little feller measures just 19" x 7" x 10" and uses no electrolytic or tantalum interstage or output coupling capacitors. The external multi-voltage power supply is 19" x 3.5" x 13.5" and weighs 25 pounds.

Using an all-discrete transistor, minimum amplifier technology the HRT 9100 is designed to accept future analog and digital processing modules. Up to six main chassis with 80 input modules may be connected to form a very low noise mixing system.

The specifications are quite good. Noise measured line in to line out at unity gain is -92dB, crosstalk is 104dBv, ref 0dBv at lkHz, and harmonic distortion, at any frequency from 20Hz - 20kHz is less than 0.005%, with 0.002% SMPTE intermodulation distortion. Our sincere congratulations to the new father.

Circle Number (103) on Rapid Facts Card

IS THAT WHY IT'S CALLED PSYCHO-ACOUSTICS?

Many cannot agree on the diverse theories of psycho-acoustics. Is it because sound people are crazy? To further The Cause, Sound Performance Laboratory introduces the new Vitalizer Equalization System. Using a highly complex system of inter-active, variable filters and phase shifters, this new product promises to have the "golden ear types" talking for months on end.

The Vitalizer is said to bring mid-range material into sharp focus and enhance high frequencies, although *not* using controlled distortion, as used in other products of this type. While under the control of a specially designed analog computer that responds to both the frequency content and dynamics of the input signal, this unit was specifically developed to be compatible with the way the human ear perceives sound. A stereo width expander is incorporated within the Vitalizer. It can be switched so that it may be used without the rest of the enhancer system.

Our favorite feature on this unit is the one that controls the low frequency spectrum. The control that brings dynamic equalization of the sub-bass material into play is labeled "soft" or "tight." ■

Circle Number (104) on Rapid Facts Card

Laurel Cash-Jones is an editorial consultant to R•E•P and a free-lance writer. Fred Jones is a free-lance engineer, producer and writer, best known as the engineer/producer for the legendary comedy group, The Firesign Theatre. Among his many credits are an uncountable number of commercials and TV shows. He has won almost every advertising award, and several of his recordings have been nominated for Grammys.

We hope you enjoy this column, as both informative and humorous. To quote Monty Python, "Always look on the bright side of life." — Written in Los Angeles in the middle of the 1992 riots.

Cutting Edge

INTERFACE CONSOLES

Interface consoles, jointly developed by DDA and Dynacord and manufactured by Dynacord in Germany, come in 8-,16-, 32-channel mainframes. Various stereo master, group output and input modules are available.

Four group mixing buses allow the use of up to four group output modules. Six auxiliary buses are also provided, giving six additional mixes with master level controls. Optional input and output transformers are available to isolate the electronically balanced XLR connections.

Interface offers 5-element LED level indicators on each channel, padded mic inputs, aux sends switchable to direct channel output, and a pre/post switch on aux sends 1 and 2 enables use as a monitor or effects send.

The group module is equipped with extensive switching to configure the console for effects return, PA or recording applications.

Circle Number (105) on Rapid Facts Card

CARTOON TRAX

The Hollywood Edge has released the CARTOON TRAX Collection of cartoon and comedy sound effects. Many of the sound effects in this collection were literally under lock and key in film storage vaults for decades before The Hollywood Edge digitally restored these selections. The CARTOON TRAX Collection is available on five compact discs, is fully cross-referenced and indexed.

Circle Number (106) on Rapid Facts Card

RSP INTELLIVERB

The RSP Intelliverb offers 2.5 seconds of memory in a 24-bit, MIDI-controllable package featuring 64X oversampling rate and 254 memory programmable locations. While the Intelliverb is capable of simultaneous effects such as chorus, delays and pitch shifting, single effect operations are the Intelliverb's strength. The virtual room reverb algorithm offers 53 parameters including room width, depth and height that enable the user to create his or her own room. Once in the room, both listener and source may be moved. Other reverbs include plate, hall, dual and stadium, as well as reverb ducking. A gate operation controls the room's absorption by closing down the reverb and prevents the sounds from one envelope being heard decaying off in the next envelope.

Eight-voice chorusing includes individual parameters for delay length (up to 740ms), modulation rate, depth and panning of each voice. Pitch shifting is possible with up to four separate voices, each covering a 3-octave range from +1 to -2 octaves. Digital HUSH (noise reduction) is available in any operating mode at the front of the effects chain.

Circle Number (107) on Rapid Facts Card

AUDIO-TECHNICA CONDENSER SHOTGUN

The new AT835a condenser shotgun mic from Audio Technica is intended for video and distant miking applications. Based on the original AT835 shotgun, the 835a can be powered from either a AA battery or a phantom power source of 9V to 52V including any video cameras providing phantom power.

Circle Number (108) on Rapid Facts Card

TANNOY SUBWOOFER

The new Tannoy CPA 5 SB subwoofer is a passive, bandpass low-frequency device designed for use with small, high-powered monitors such as the Tannoy CPA 5 ITC. Intended for limited space environments such as workstation monitoring applications, the CPA5 SB delivers frequency response from 46Hz to 210Hz.

Using four 5-inch low-frequency drivers, the CPA 5 SB is wired directly to its power source and can be positioned independently from its satellite speakers. Stand alone impedence for the device is 12Ω and the in-use working load (when coupled to mid/high satellite speakers) is rated at 6Ω. Sensitivity for the CPA 5 SB is rated at 93dB 1W at 1M, while the unit's power handling is listed at 150W.

Circle Number (109) on Rapid Facts Card

SHURE ILP-1

The new Shure in-line preamplifier, model ILP-1, was designed for use with the SM93 miniature condenser microphones and SM91 surface-mount condenser mics. ILP-1 control features include a 10dB gain switch and a selectable flat or low cut response. The low cut position provides a 12dB/octave rolloff below 80Hz, and is useful to minimize low-frequency noise.

The ILP-1, has a tube-style design measuring 5 1/2" in length and 13/16" in diameter, which allows the preamp to be plugged directly into a 3-pin XLR mixer or snake input. The ILP-1 can be powered by an 11V to 52V dc phantom supply from sound reinforcement, recording or broadcast equipment.

Circle Number (110) on Rapid Facts Card

UREI DYNAMIC CONTROL

The UREI LA-22 compressor/limiter, one of three new UREI models, is a dual-channel unit with expansion capabilities. In addition to traditional dynamics control, the 22 can selectively compress or expand specific widths of the frequency spectrum between as little as 1/6th of an octave up to three octaves, leaving the remainder of the signal unchanged. In PA applications, configuration of selected bandwidths can be used to remove or limit unwanted signals such as low-end feedback and can help to prevent damage to the system's components.

The LA-22 is also capable of de-essing speech or music in radio and television broadcast applications, preventing distortion because of saturation or over-deviation of the FM transmitter. A "Side Chain Monitor" button on the LA-22 allows zeroing-in of the filter output only, giving audible reference of the unit's effect on the selected signal.

Circle Number (111 on Rapid Facts Card

ORBAN TRANSMISSION LIMITER

Orban has introduced the new Transmission Limiter 4000, designed for broadcast applications including network audio distribution, and protection of digital audio systems from overload and overmodulation. The 4000 features a front panel design that is optimized for error-free setup, with only input and output level controls, and switching for the built-in line-up tone generator that enables quick and accurate level setting.

Circle Number (112) on Rapid Facts Card

Hardware & Software

FRONTERA HARD DISKS

Frontera Electronics recently premiered its i*cove line of hard disk storage systems, designed specifically for direct-to-disk recording applications and available in rack-mount and desktop configurations and compatible with Macintosh, Atari and PC software including SoundTools, ProTools and Digital Master. The most popular Frontera model is the 680MByte (rack-mount). Other configurations that start with the desktop 50MByte, include CD-ROM drives and fixed capacity drives up to 1.2GBytes. All drives are supplied with 40MBytes of free samples created for the Ensoniq EPS and EPS16+.

Circle Number (113) on Rapid Facts Card

MIDI TIME PIECE II

MIDI Time Piece II is a networkable MIDI processor that functions as an 8-cable computer interface, an 8 x 8 stand-alone MIDI merger/router/filter and a SMPTE synchronizer with adjustable freewheeling.

It is programmable from the front panel. A 16 x 2 LCD reports information about the status of the device. Four rotary encoders provide access to all programming parameters and also double as MIDI control knobs. Two foot-pedal inputs can accept switched or continuous foot pedals and can transmit controller data. Up to four MTP II's can be connected to expand a MIDI system to 512 MIDI channels.

MIDI Time Piece II is compatible with the Performer and Digital Performer sequencing programs, the MOSAIC music publishing program and other music software programs for the Mac.

Circle Number (114) on Rapid Facts Card

MEDIATOR MS-124

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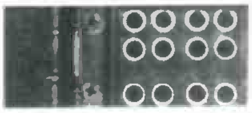

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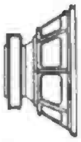

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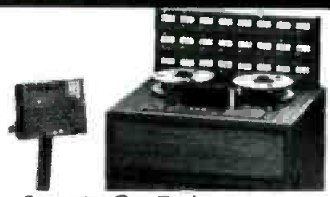

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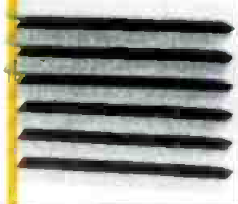

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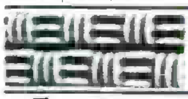

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Anything Audio	58	31	617-426-2875
Bauer Communications	61	39	800-627-7277
Behringer	1	3	516-932-3810
DB Engineering	60	38	617-782-4838
Demeter Amplification	60	35	818-986-7103
Dolby Labs, Inc.	25	16	415-558-0200
Fast Forward Video	43	19	800-755-TIME
Frontera	45	23	619-727-3410
The Hollywood Edge	31	17	800-292-3755
Innovative Audio	60	35	818-986-7103
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Studer Revox America, Inc.	7	6	615-254-5651
Tascam	IBC	2	213-726-0303
T.C. Electronic USA	33	15	805-373-1828
Technical Audio Devices	4	5	213-816-0415
UREI Electronics Products	BAC		818-893-8411

Sales Offices

CLASSIFIEDS
Renée Hambleton
913-967-1732
9800 Metcalf
Overland Park, KS 66212-2215
Telefax: 913-967-1901

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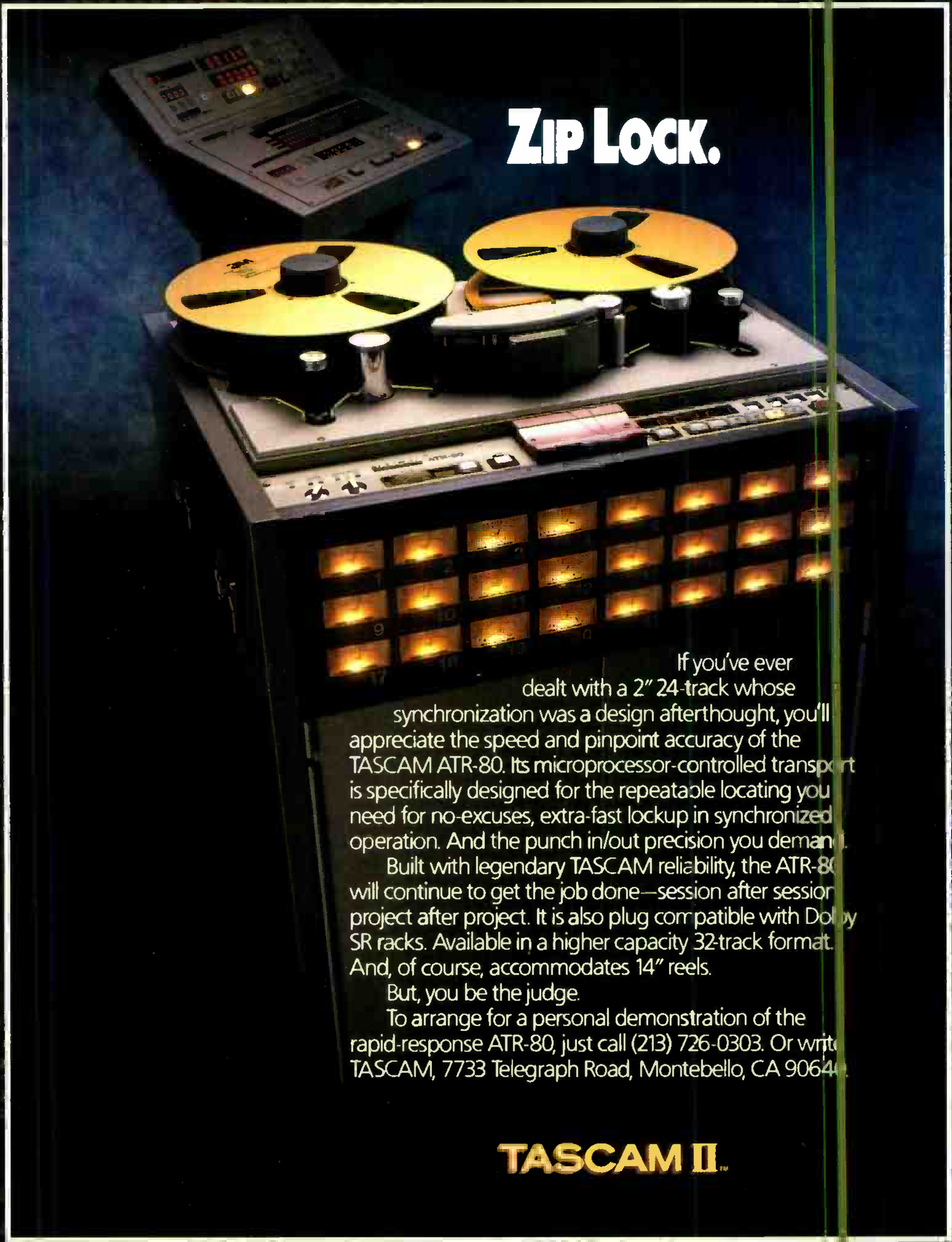
Herbert A. Schiff
310-393-9285
Jason Perlman
310-458-9987
Kelly Daugherty
310-451-8695
501 Santa Monica Blvd., Ste. 401
Santa Monica, CA 90401
Telefax: 310-393-2381

NEW YORK, NY
Ron Lew
212-332-0615
38th Floor
888 Seventh Ave.
New York, NY 10106
Telefax: 212-332-0663

OXFORD, ENGLAND
Richard Woolley
Unit 3, Farm Business Centre
Clifton Road
Deddington, Oxford
OX15 4TP England
Telefax: +44(0) 869-38040
Telephone: +44(0) 869-38794
Telex: 837469 BES G

LIST RENTAL SERVICES REPRESENTATIVE
Lori Christie
9800 Metcalf
Overland Park, KS 66212-2215
Phone: 913-967-1875
Fax: 913-967-1897

TOKYO, JAPAN
Mashy Yoshikawa
Orient Echo, Inc.
1101 Grand Maison
Shimomiyabi-Cho, 2-18
Shinjuku-ku, Tokyo 162, Japan
Tel: (3) 3235-5961
Telex: J-33376 MYORIENT
Fax: (3) 3235-5852


ZIP LOCK.
If you've ever dealt with a 2" 24-track whose synchronization was a design afterthought, you'll appreciate the speed and pinpoint accuracy of the TASCAM ATR-80. Its microprocessor-controlled transport is specifically designed for the repeatable locating you need for no-excuses, extra-fast lockup in synchronized operation. And the punch in/out precision you demand.
Built with legendary TASCAM reliability, the ATR-80 will continue to get the job done—session after session, project after project. It is also plug compatible with Dolby SR racks. Available in a higher capacity 32-track format. And, of course, accommodates 14" reels.
But, you be the judge.
To arrange for a personal demonstration of the rapid-response ATR-80, just call (213) 726-0303. Or write TASCAM, 7733 Telegraph Road, Montebello, CA 90640.
TASCAM

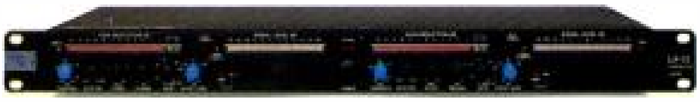
BACKED BY POPULAR DEMAND.

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

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

Full Parametric EQ Section.

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



All three LA Series models are housed in a compact 1U rack space and are designed to deliver years of reliable service, in the studio or on the road. Whether your application is sound reinforcement, recording, broadcast or for permanent install, you'll find LA Series to be a trustworthy and hardworking addition to any system. From the people who set the standard, UREI.

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