March 1990

Recording ENGINEER/PRODUCER

The Applications Magazine for Audio Professionals

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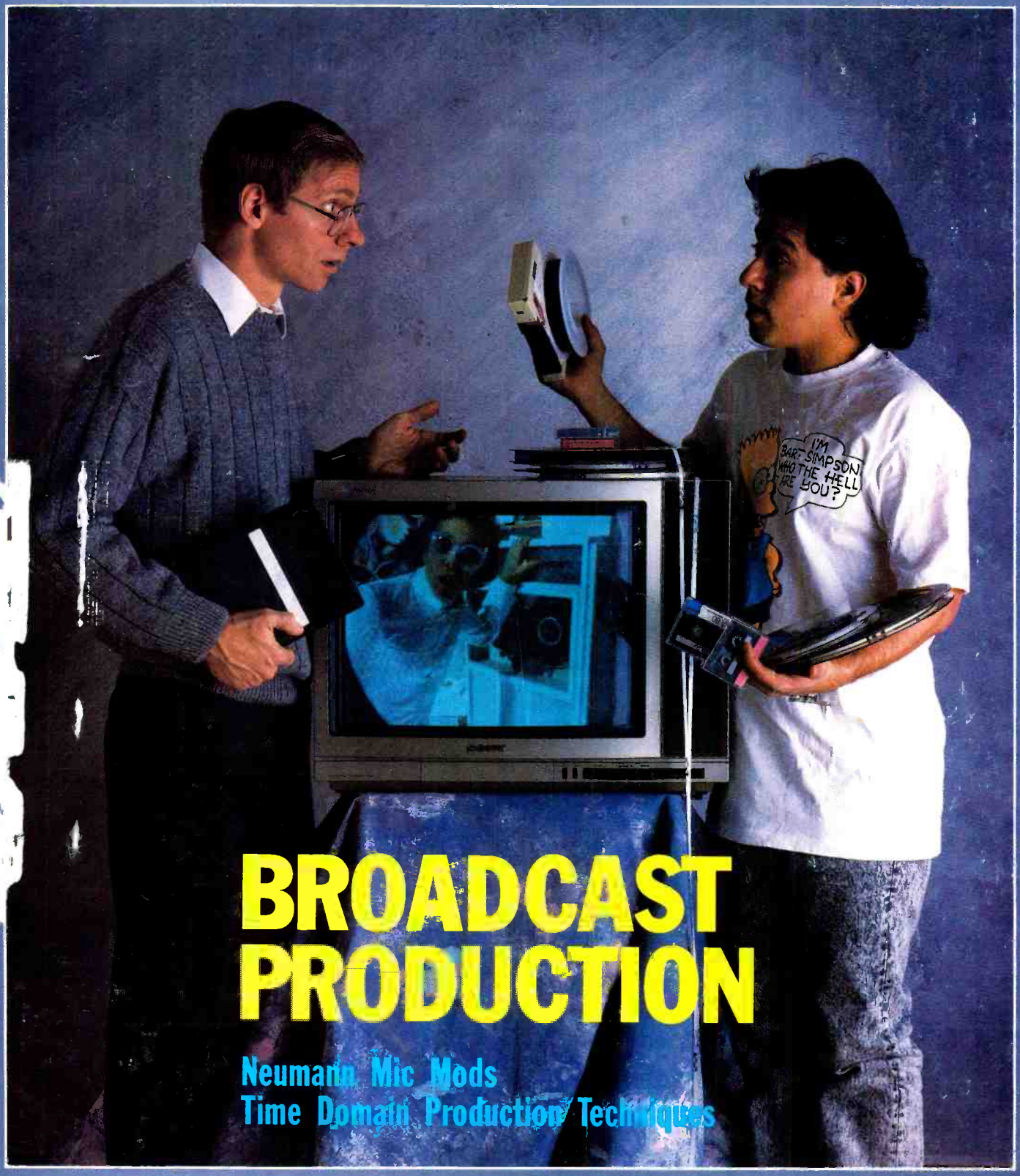

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Different audio formats, different audio approaches: To present the best audio to the public, audio and broadcast engineers must work together to present audio that is aesthetically pleasing while meeting transmission standards. (Photo by Mike Regnier. Technical assistance: City Spark Studios and AudioLine, Kansas City, MO.)

Departments

Volume 21, No. 3

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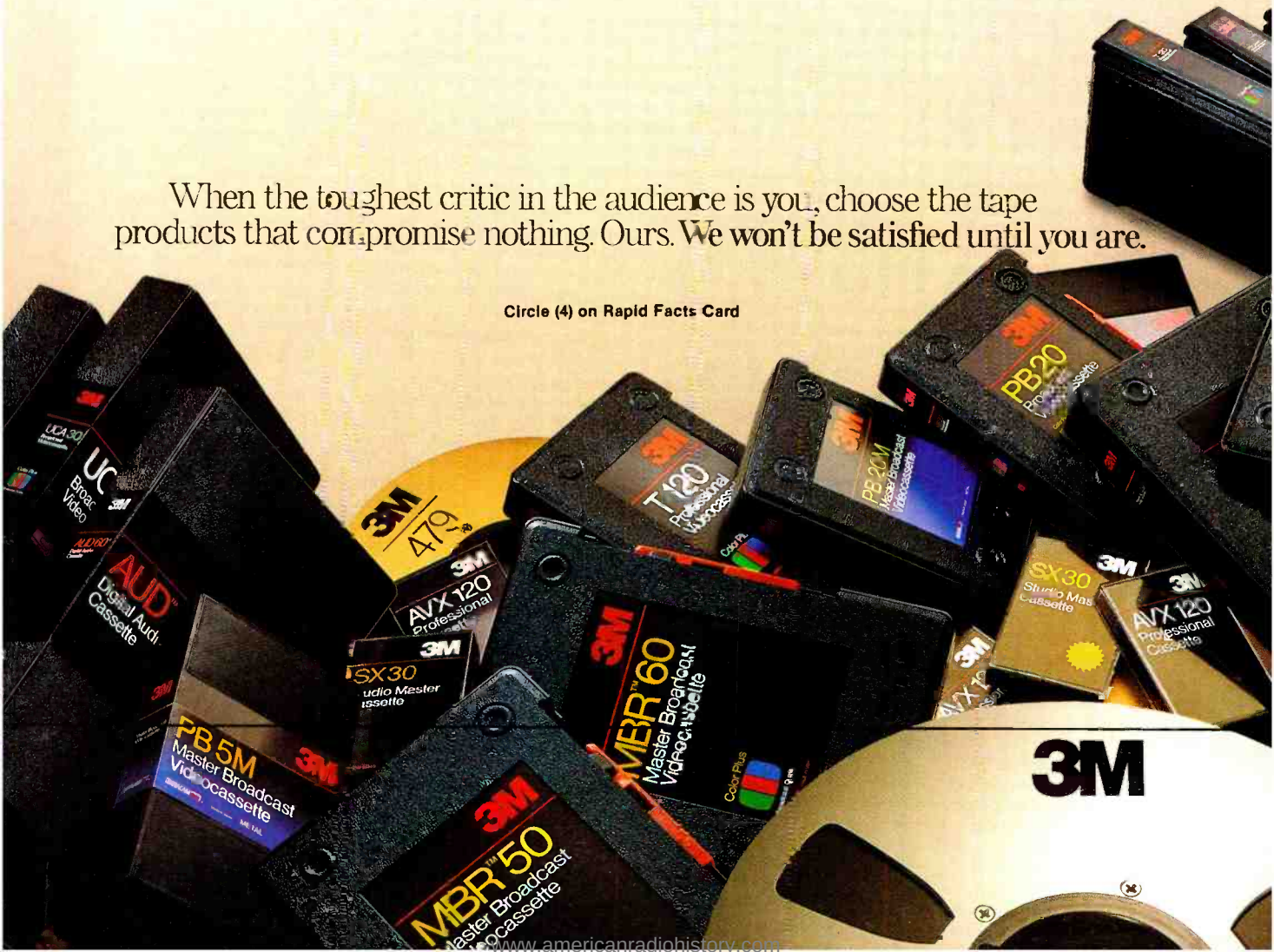

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Recording Engineer/Producer is an applications-based publication targeted at professional individuals and companies active in the commercial business of studio and field recording, audio for video, live sound production and related fields. Editorial 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

Yearning to Learn

On the job training. OJT. Up until 1970 or so, it was the only way any of us learned how to slit tape with a razor blade or tweak out the record cal pots hidden inside the refrigerator-sized transports. If we were lucky, an older studio veteran would take us into the shop and pull out a ratty set of schematics to actually explain what the hell it was we were trying to accomplish.

Usually we'd hang out in the CR, peering over a maintenance guy's shoulder as he repaired a hot-seat damaged console module or find the headroom limits on a new reel of 456. And we'd ask the pertinent questions, lapping up the answers like a puppy laps up fresh milk: What frequency does the head bump move to at 30ips? Should I accelerate to +6 if I'm doing a soft rock 'n' roll date? Which of the noise/distortion/frequency curves are favored at 1dB over peak bias on 3M 250?

In the 1970s and early 1980s, the explosion of music in our mass culture made audio production a hot commodity. Studios were swamped with applicants. Responding, university broadcasting or music departments and many small independent recording studios offered training programs for students.

Granted, initial attempts closely resembled the "let's all learn together through experimentation" school of audio gymnastics, and some graduates found out later that a good course load in child psychology and music composition would have prepared them better than the five ways to mic a classmate's acoustic guitar taught in Music Recording 201. Things have greatly improved with time, and many schools today are highly respected.

All of which is fine and well for students, or for schools where education consists of razor blade editing and tape biasing, mic selection and board operation — things which had remained technologically stable for more than 20 years. To those already in the business, along comes MIDI, digital verbs, computer programmable processors, samplers and disc-based recording, and the whole training thing goes out the window. It seems that if you're not a keyboard player these days, you're lost.

The ability to cut and splice the 2-inch and tweak the record EQ for Agfa 468 will have to sit there inside the brain right next to the bit about MIDI channel routing for the S-1000 triggered sample (called up via the Mac Librarian interface), the sequence on auto-bias calibrate for the new multi, the shortcut on altering the kick feel on the drummer's aging RX-11, and the dreck about session billing on the office IBM clone.

Which brings us to: Where does a busy professional go for a knowledge update? Who do you talk to for a product and technique refresher if you're not a new-on-the-block student?

In the computer hardware industry, IBM has long provided education for its users. In software, freely enterprising independent seminar companies have effectively filled in the blank between new personal computer products and the need to know. In the video world, manufacturers like Sony with its Institute of Applied Video Technology provide both generic production technique and product-specific operation courses targeted at both recent job market entries and self-improving professionals.

To the companies above, it is very clear that more new products sell and get used if the operators are comfortable with operating them. This is motive enough to justify IBM and Sony's efforts behind their education programs.

But who's doing it in audio? For students, some groups like Full Sail Center for the Recording Arts are blending new keyboard and computer technology with traditional recording education. But for those already active in the industry? You tell me. There's no Syn-Aud-Con for the magnetic storage medium crowd, yearning to learn.

Great incentive exists for an audio manufacturer to sponsor training seminars centered around the new technologies. Professional organizations also have much to gain in membership and tuition revenues by providing the managerial and operation support behind intensive education programs. Even recording schools and universities, with their facilities and training staff, can provide an ideal environment to supply the needed educational resources.

At the next industry convention, I would like to see a manufacturer walk up to a recording school representative or an organization sponsor and say "Let's talk. Let's pool our resources and insure our mutual futures. Let's get the industry through this stage of accelerated technological fear and loathing, at least until the interface is so transparent that education and training aren't needed anymore."

Mike Joseph
Technical Editor

GUEST EDITORIAL

By Roger Lagadec

Digital Audio: A Status Report

Hindsight is a great thing. It is a wonderful way of deciding how best to approach a problem — when the problem has long gone by! It makes today's uncertainty tomorrow's fact, and today's mistakes tomorrow's wisdom.

One of the beauties of nature ensures that what we do today affects what kind of a day we will have tomorrow. The sweeping changes in Eastern Europe last year gave the world a new appreciation for the phrase "cause and effect."

The Japanese look at this in a similar way. If you want to be sure of the final product, you have to take part in its design. And if you want to be sure that everything works out, you had better get involved in the plans. So it is with digital audio.

While my favorite definition of a pro audio show is "the discredited showing the unavailable to the insolvent," I actually believe pro audio to be a fairly simple business driven by a few simple forces. We provide the equipment for making TV and movie soundtracks, LPs, CDs, audiocassettes, videotapes and live performances. We also live by renting our expertise, our studios and our equipment.

Change one of the above media, and our industry changes. Compact discs made the label "digital" a selling point, and digital mastering a necessity. So along came digital audio for the studio — surely an easy one to predict. By the same logic, there is almost no digital audio in broadcasting yet, because digital audio is still expensive and the broadcast formats still analog. Film soundtracks are mostly analog because analog technology serves it much more elegantly than today's digital equipment can. And television is also mostly analog with poor sound—no major incentive for all-out digital post-production.

I expect this will change in the 10 years to come. In radio broadcasting, for example, the countries civilized and affluent enough to offer their citizens a nationwide mix of private and public radio programs, with some level of quality, have satellites in orbit, and digital radio broadcasting on their technical agenda. As a natural reaction, terrestrial broadcasting also champions digital formats, with the stunning advances in data reduction recently demonstrated in Europe for broadcasting and transmission formats. Combine this technology (mostly DAT, some disks and DSP), and you will begin to see a new market emerge, while analog broadcast technology will begin its gradual descent into nostalgia.

Television also changes, and will affect our audio world. Television is admittedly analog, but a strong new trend is the emergence of professional digital VTRs. They happen to offer channels of 20-bit digital audio recording, and in the last year 1,000 machines were sold. With high-end professional video going digital, and casually dragging along underused digital audio channels, a very large market is being created.

The third trend I see is linked to the cost of technology. Major changes in our media create new markets, which call for new digital audio technology. The basic technology itself is cheap and available: Error correction and digital signal processing can be adequately taught with textbooks and technical papers of 15 years ago. But making the chips is costly, making the drives and recorders even more, and making the software really work is a costly nightmare. Professional audio of the digital kind is still a very expensive gamble. No wonder that acquisitions and mergers have recently flourished in our industry, and surely major ones are still to come.

Finally, I see the music software industry as a potential source of major change. CDs initiated the first wave of digital audio, which at the time was a technical, not an artistic, innovation. The endless controversies about the sound quality of digital made that clear enough. A few of the large companies that dominate the music recording business are capable of using their technical resources to launch a second wave of professional audio in the recording studios. And few top artists would be indifferent to their creations being saved for posterity in all the glory of, say, High Definition Video with advanced digital audio.

Roger Lagadec is general manager of professional audio at Sony's Atsugi plant in Japan.

RE/P

THE NEVE PRISM SERIES

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With the **Neve Prism Series**, there's a whole new world of sound waiting to be discovered.

The Prism Series offers the unsurpassed EQ and dynamics capabilities of the Neve VR Series console — the console of choice in today's leading recording and post-production facilities — in one extremely powerful and flexible rack-mount package. The two Prism modules include the four-band parametric equalizer and the mic amp/dynamics unit, consisting of compressor/limiter/gate/expander. In addition, the mic amp/dynamics unit also has a mic input and variable high and low pass filters for unmatched definition, clarity, and size of sound.

The competitively priced **Prism Series** is even more affordable with our own Siemens financing. No matter what kind of work you do, the **Neve Prism Series** will enhance it dramatically.

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NEWS

Syn-Aud-Con workshop

Synergetic Audio Concepts' Intelligibility Workshop II is scheduled for May 24-26 in Bloomington, IN. Dr. Larry Humes of Indiana University will supervise.

The workshop will deal with the measurement of speech intelligibility and its uses in planning sound reinforcement systems with acceptable intelligibility characteristics. Special emphasis will be given to the role of the pinna and ear canal in intelligibility; hearing threshold and pinna response measurements will be made on each workshop participant. In-The-Ear recordings will be made in the pressure zone of the eardrum of several participants. DAT recordings will be made of 5%, 10%, 15%, 20% and 25% articulation loss of consonants.

For more information, contact Synergetic Audio Concepts, R.R. #1, Box 267, Norman, IN 47264; 812-995-8212; fax 812-995-2110.

HR format gains

The half-rack (HR) format is gaining acceptance, according to its developers. Jointly designed by ART, Ashly Audio, Crown, dbx, Furman, Lexicon, Rane and Symetrix, the format accommodates compact, flexible, custom-designed installations and enables signal processing modules to be mounted in a standard 19-inch rack horizontally up to two modules across, or vertically with up to 10 modules across. Each HR module is self-contained, can be mounted flush or recessed, and will interface directly with other professional equipment via terminal strip, 3-pin (XLR) and/or ¼-inch connectors.

At present, the HR format is an informal standard. Because of the number of cooperating manufacturers that helped with the HR design, several product redesigns are taking place to accommodate the format. Larry Winter of Rane, chairperson of the HR technical committee, says that formal acceptance by ANSI could take more than two years.

An official HR Mechanical Standard Document, published by the HR technical committee, is available from Rane Corporation, 10802 47th Ave. W., Everett, WA 98204; 206-355-6000.

Music and video business program

The Art Institute of Philadelphia has added a 2-year music and video business (MVB) program to its curriculum. The program, designed to train students for career placement in the music and video business industry, is divided into four segments: a foundation program, studio and technical skills, marketing/communications and business management.

Students will receive in-depth instruction from working professionals; state-of-the-art equipment will be available in studios at the Institute and at nearby professional studios.

For more information, contact Greg Walker, Art Institute of Philadelphia, 1622 Chestnut St., Philadelphia, PA 19103-5198; 215-567-7080.

News notes

Neve is offering engineers and producers two free hours of studio demo time on its Power Package, which consists of a VR console with Flying Faders automation and a Mitsubishi 32-track X-880 or X-850 digital tape recorder, at any of five participating studios. Participating studios include The Enterprise (Burbank, CA), Cherokee Recording (Los Angeles), Sigma Sound (New York), Digital Recorders (Nashville) and The Chicago Recording Company (Chicago).

Concertech is a newly established studio designed for cost-effective audio engineering and equipment rental services for smaller or medium budget acts in the New York metro area. Carl Foster, co-founder of N.Y. Musicworks, is the president. The studio is located at 108 W. 17th St., Suite 7, New York, NY 10011; 212-255-5986.

John M. Storyk Inc. has designed two new facilities: Context Music, a 6-room rehearsal/recording facility located on the Lower East Side of New York; and Margarita Mix, a 5-studio post-production facility in Hollywood owned by L.A. Studios' partners Sunny Blueskyes and Jim "Bunz" Bredouw.

Americ Disc has relocated to 1290 Avenue of the Americas, Suite 2540, New York, NY 10104; 212-767-0750, 800-767-0752; fax 212-767-0753.

The Minneapolis-based **Southern Thunder Sound Company** includes in its inventory the following Electro-Voice speakers: more than 100 EVX-150 woofers; more than 100 EVM-12Ls; more than 50 subwoofers with EVX-180 and EVM-18B speakers; and monitors using the EVM-15L speakers.

Clair Brothers Audio Enterprises recently formed **Clair Brothers Audio Systems,** a company targeted at the growing installation market. It will provide sales of portable and permanent installations for sound reinforcement venues. Offices are scheduled to open in London, Tokyo, Los Angeles and New York; the main office and warehouse are shared by both companies in Lititz, PA.

Hybrid Arts has relocated to 8522 National Blvd., Culver City, CA 90232; 213-841-0340; fax 213-841-0348.

Los Bukis, the international Grammy Award nominees from Mexico City, are the owners of the largest **QSC**-powered, privately owned tour sound system in the world. The system includes 40 model 3800 amps; 64 model 3500 amps; and 72 JBL model 4852 Concert Series loudspeakers.

Audio Intervisual Design has been named the U.S. agent for W. Albrecht GmbH Studiogerate, the West German manufacturer of sprocketless magnetic film recorder/reproducers.

People

Neve North America has named **Charles Conte** public relations administrator for the North American market.

Joe Naccarato has been appointed general manager of Rupert Neve Canada.

Neve Electronics International has appointed **Hazel Simpson** director of sales.

Rick Bos has been named marketing manager at DOD Electronics.

Samson Technologies has appointed **Joel Singer** product sales manager for the complete line of Soundtracs mixing consoles.

Steve Lawson, president of Lawson Productions, Seattle, was elected to the SPARS board of directors during the annual meeting at the AES convention in New York. Lawson is the first SPARS director from the Northwest.

James A. Dismore has been named president/CEO of Ultimate Support Systems. RE/P

WE CARE

AGFA

Left to right: Barry Bongiovi, Studio Manager; Bob Walters, President; Tony Bongiovi, Vice President, Studio Designer

Power Station leads the way. It's inspired numerous studio designs. Countless hit records. And a generation of producer/engineers.

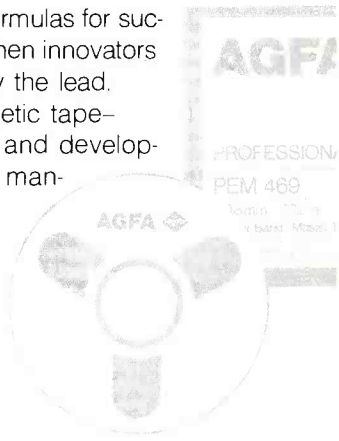
No wonder the New York scene is already buzzing about their new Studio D, a hybrid room that can mix music, audio-for-video, and audio-for-film. It's an innovative look at tomorrow's state-of-the-art.

Innovation. It led Power Station to Agfa bias-compatible PEM 469. Long before it became the year's most exciting new mastering tape. Long before anyone else. "The high-end holds together," says V.P./Studio Designer Tony Bongiovi. "It's transparent. It's musical. It's leading-edge."

It's a new vision for the 90's:

A decade of professionalism and excellence. A time when studios must try new formulas for success. An era when innovators will always play the lead.

AGFA magnetic tape—from research and development, through manufacturing, to delivery and service—we care!



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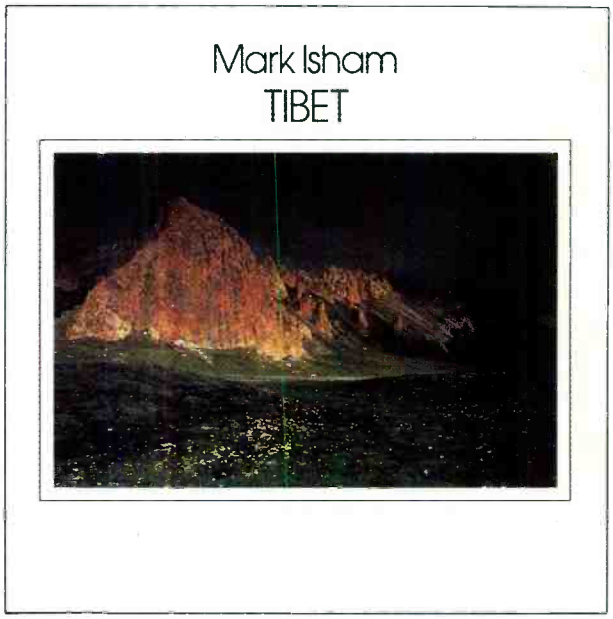
FRESH TRACKS

By the RE/P staff

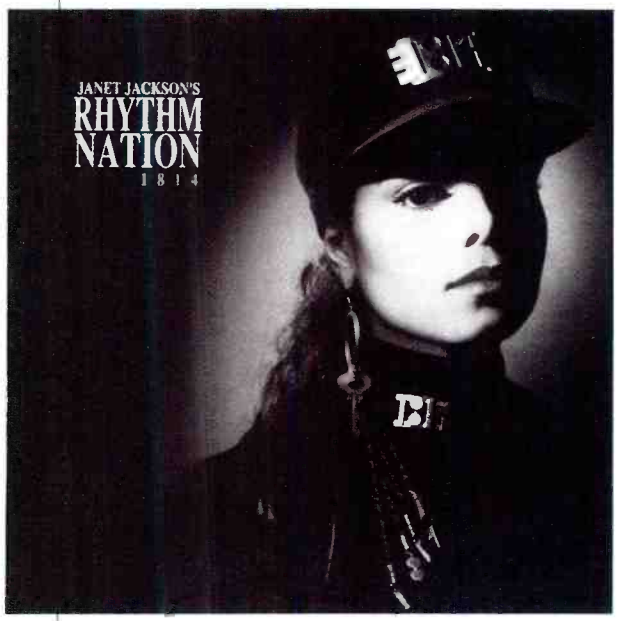
Mark Isham: "Tibet"

Label: Windham Hill Records
Producer: Mark Isham
Recorded by: Ben Rodgers, Stephen Krause
Mixed by: Mark Isham, Ben Rodgers
Studios: Earle-Tones Studio; Producer's Studio; Mad Hatter Studios
SPARS Code: ADD



Comments: Mark Isham, Bay Area horn player/synthesist and member of the seminal Group 87, presents here a beautiful collection of sonic tone poem-like flavorings, some barely melodic, all rich and deep with "sound." Although begging to be called New Age, this material is more than just string swells and large plate reverb F/X, as the stellar lineup of talented jazz and contemporary musicians proves. Isham is a master of trumpet and flugelhorn, flavored downstream with the various color palettes and assorted electronic. His material is instantly recognizable, and there is much to absorb here concerning balances, blends and timbral shading.

Of special interest: The tracks, labelled Part I through Part V, all have their own sonic quality and theme. Note the ambient and spatial effects around the different acoustic and electronic instruments, all creating more of an environment of sound than an actual song. The bass trombone, electric bass, French horn, flugelhorn percussion, trumpet and electronics blend in a way closer to colors on an impressionistic painting than instruments being mixed in an arrangement. This is good stuff, not Yuppie elevator music.



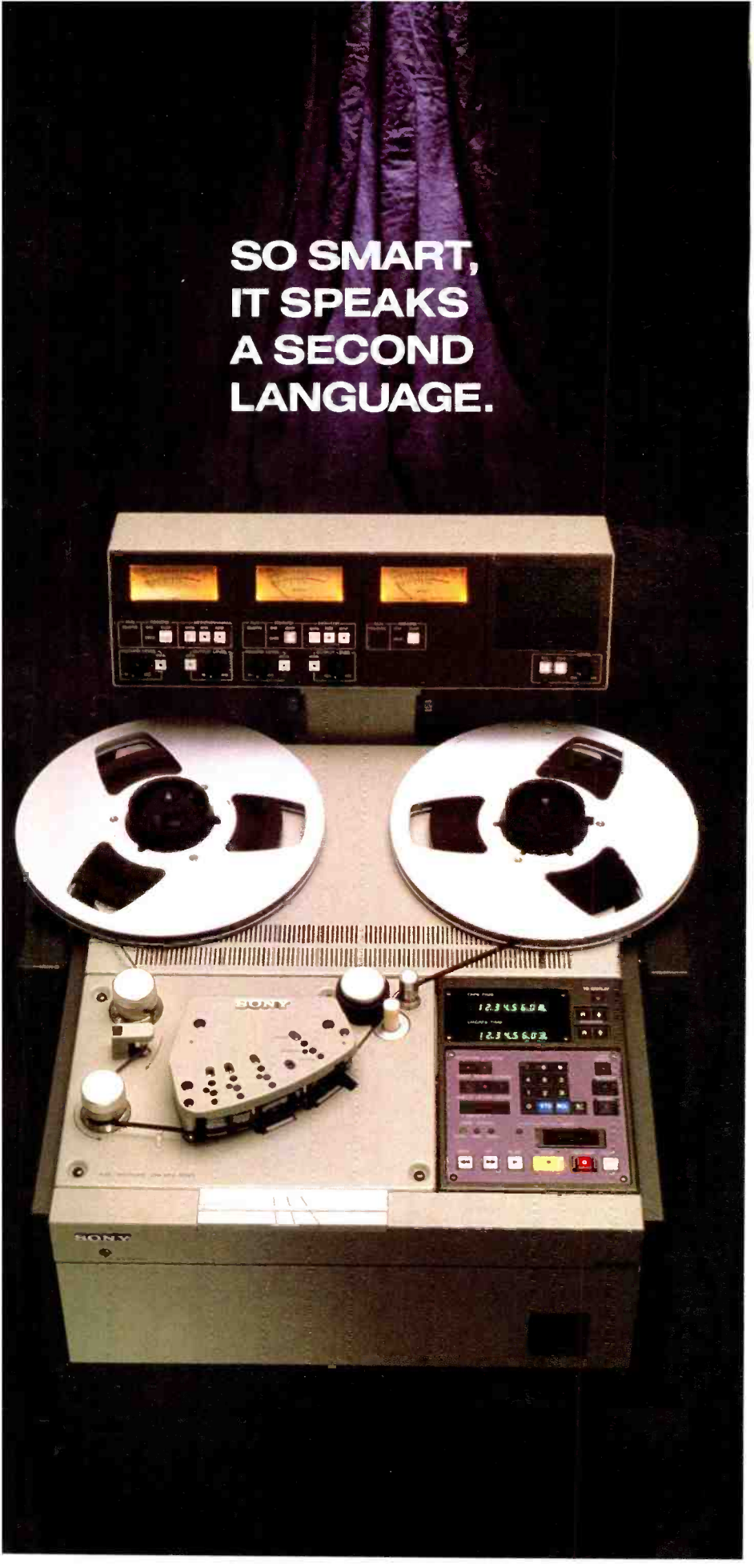
Janet Jackson: "Janet Jackson's Rhythm Nation 1814"

Producers: Jim Jam & Terry Lewis for Flyte Tyme Productions
Co-Producer: Janet Jackson
Executive Producer: John McClain
Recorded and mixed by: Steve Hodge
Studios: Flyte Tyme Production Studios A, B and D, Edina, MN; NPR Studio M, St. Paul, MN
SPARS Code: AAD

Comments: When you divorce yourself from all of the commercial effort surrounding the promotion of this release and *listen* to this work, you'll immediately recognize the talent of the artisans who produced one of 1989's finest recordings. Of particular merit are the variety and creativity of sound sources — from a sampling and orchestrated nature — and the tasteful and unusual combination of outboard signal processing. Processing on background vocals and auxiliary rhythmic tracks are both refreshing and timely.

Of special interest: Perhaps the most noticeable sonic pleasures stem from the basic mix of levels, amplitude modulations, panning and filtering, courtesy of Flyte Tyme's GLW SeriesTen console. The accuracy and clarity of the rapid-fire panning, muting and auxiliary automation is nothing short of astonishing.

SO SMART, IT SPEAKS A SECOND LANGUAGE.



Now, there's an audio recorder that talks video. The APR-5003V.

You can actually control the APR-5003V with a video editor. Because its 9-pin serial port communicates with Sony's BVE-9000 or BVE-900 editors. And ease of control is just one of many advantages for video post production.

The APR-5003V also offers a wide range of synchronization features, including resolve on play. Chase synchronization. Timecode generation with video reference. And offset with bit resolution.

Of course, the first language of any audio recorder is still audio. And you get the superb sonic quality you'd expect from Sony.

9-pin serial port for control by Sony video editors BVE-9000 or BVE-900.

If you're in post production, and you're in the market for an analog recorder, remember whom to speak to. Contact your Sony Professional Audio Representative. East (201) 368-5185; West (818) 841-8711; Central (312) 773-6001; South (615) 883-8140.

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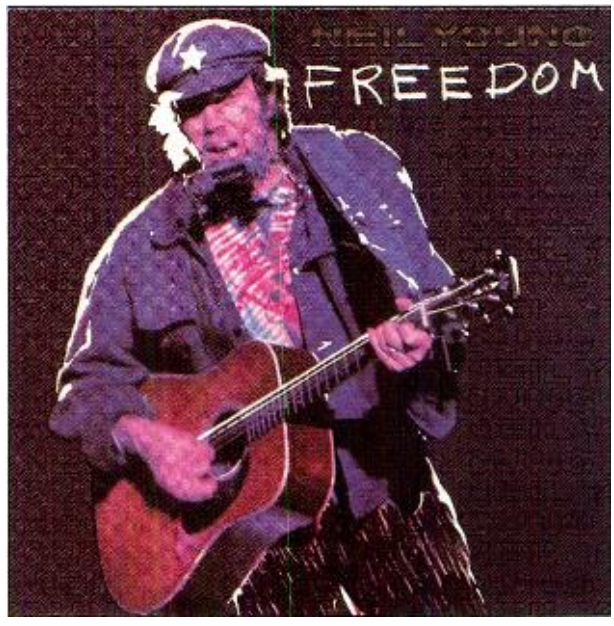
PROFESSIONAL AUDIO

FRESH TRACKS

Neil Young: "Freedom"



Label: Reprise
Producers: Neil Young, Niko Bolas
Recorded by: Dave Hewitt, Gary Long, Tim McColm, Brently Walton, Dary Sulich, Tim Mulligan, Alan Abrahamson, Bob Vogt
Mixed by: Dave Hewitt
Studios: Hit Factory, New York; The Record Plant Remote Truck; The Barn/Redwood Digital
SPARS Code: DDD

Comments: Musically, "Freedom" runs the gamut from the sweetest Ronstadt/Young duet, to Dire Straits-like journeys, to classic wanked-out Neil Young rock 'n' roll roughness, basic as always. The production, excluding the sometimes questionable foreground-background balances of solo instruments and vocals, is digital in clarity and very open. Relying heavily on acoustic instruments, the recorded quality of the guitars, harmonica and electric band is raw but good. The weakest link is in the positioning (left/right, front/back) and imaging of the mixes.

Of special interest: Interesting, original sounds throughout, such as overly wet or overly dry verb treatments; the occasional, intentional(?) board or digital tape clipping on more than several tracks; the unusual rendition of "On Broadway" recorded at the Hit Factory, seemingly with two distant mics and a medium room patch on the verb; and the unique reverb and guitar sounds on "Don't Cry." All said, it is hard to judge whether many of these sounds were recorded poorly (intentionally or otherwise), or if this CD is an incredibly creative artistic statement. You be the judge.

Stevie Ray Vaughn and Double Trouble: "In Step"

Producers: Jim Gaines, Stevie Ray Vaughn and Double Trouble
Recorded by: Jim Gaines, Richard Mullins, Richard McKernen
Mixed by: Dave McNair, Jim Gaines
Studios: Kiva Studios, Memphis, TN; Sound Castle, Los Angeles; Summa Studios, Los Angeles
SPARS Codes: DDD

Comments: Can rock 'n' roll survive in the all-digital domain and still have that analog crunch? Yes. There are no real surprises here — just a solid collection of tightly crafted tunes each with a personality to please, and with enough musicality and emotion to spare. Solid tracks from solid players and engineers.

Of special interest: "Riviera Paradise" reveals a side to Vaughn and band not readily known to their listeners in which their performance and musicality stretches outside the boundaries of their trademark sound.

AMR 24

Track Record

BUSY

Today, some of the world's busiest studios are built around the AMR 24 console. Designed for a higher standard of performance, the AMR 24 offers more inputs, more flexibility and the highest levels of audio quality for day-to-day recording and mixdown.

TRACKING

As a tracking console, the AMR 24 consistently achieves superior results the *first* time — no re-recording required. With exceptionally low crosstalk and noise, and a total dynamic range of 100 dB, it is fully compatible with digital audio.

MIXDOWN

Configured for mixdown, the AMR 24 offers unrivaled flexibility. It handles all kinds of inputs — up to 84 of them in a standard format. Just push *one button* and the 24 Track Select switch changes the monitor returns to full-function line returns normalled to a second 24 track or to synchronized MIDI "virtual tracks." Simple switching creates an additional 24 inputs through the cue faders for effects returns.

Up to 64 channels can be automated with a wide range of factory fitted systems available.

Of course, these DDA design innovations are complemented by the highest quality components and designs that *feel right.*

INVESTMENT

Studio owners know why the AMR 24's track record represents an impressive return on their creative and financial investment. If that news sounds interesting, write us on your letterhead for full information.

DDA

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Circle (9) on Rapid Facts Card

FRESH TRACKS

The Grateful Dead: "Built to Last"

Label: Arista
Producers: Jerry Garcia, John Cutler
Co-producer: Bob Bralove
Recorded by: John Cutler, Tom Flye, Jeff Sterling, David Roberts, Peter Miller, Justin Kreutzman, Chris Wiskes
Mixed by: John Cutler
Studios: Club Front, San Rafael, CA; Skywalker Ranch, San Rafael; Studio X, Petaluma, CA
SPARS Codes: ADD

Comments: The Dead's first album in two years, "Built to Last" is a technologically current effort worth listening to for its openness, acoustic detail and attentiveness to reality, both in sampling and synthesizer programming. (See "Artist Focus" in this section for more details.) The ambiance, stereo spatial image placement and tasteful, creative use of processing/FX on tracks like "Victim or the Crime" and "Picasso Moon" clearly show attention to detail and long hours thinking clean thoughts.

Of special interest: Pay attention to the final cut on the album: "I Will Take You Home." The music box (real, sampled or synthesized?), backed by piano and an as-good-as-it-gets string sample is not the Dead you've heard before. Think what you will of the Grateful Dead's past musical flavorings, this album is sonically modern without losing the essence of what makes them unique.

Artist Focus: The Grateful Dead

By Bob Bralove

My two years working with the Grateful Dead have been one continuing, exciting experiment in music. The current album, "Built to Last," was no exception. The band was eager to try new and different techniques of recording and arranging. It was a privilege to be the associate producer with the likes of Jerry Garcia and John Cutler.

Generally, the arrangements for each tune were worked out at Club Front, the Dead's studio in San Rafael, CA, where we recorded on a Studer A880 using a Neve V Series console. Each writer would bring in his tune and teach it to the band. When everyone had a general idea of what the tune was about, Jerry, Bob Weir and Brent Mydland worked out the specifics of the arrangement. A drum machine provided an elegant metronome for their work.

The complexity of the drum programming depended on the tune. For some tunes, all we needed was a 4-bar groove that could sit in the background as they played and experimented with different parts. Other tunes required specific percussion accents or syncopations that the band wanted to hear over certain sections.

When the overall form of the tune had been sketched out, the tune's singer put a guide instrumental and vocal track down to the drum machine. Once these guide tracks were in place, the rest of the band worked out their parts. The arrangement and parts defined, the song was placed in the hands of the drummers, Bill Kreutzmann and Mickey Hart. They added their parts while listening to the other instruments and an occasional click track.

Once the drummers had established the "human groove," everyone readdressed their parts. The drum machine was no longer listened to and the integration of the entire tune was considered. Slave reels were generated for band members to work on their parts in their home studios or at Club Front. When everyone was happy with a part, it was done, whether the part was recorded at the initial ensemble stage or alone in a private studio. The final stages were the background and lead vocals, and the mixing.

Bob Bralove is a San Francisco-based synthesist/programmer and producer, previously with Stevie Wonder.

Although this may sound like we had a formal and systematic way of working, there weren't any hard-and-fast rules. Any player could ask to record with any other player, or record alone, or try any kind of changes he wanted to.

Below are technical considerations for some of the tunes.

"Foolish Heart": This is a great example of Mickey Hart's use of percussion in a more traditional way. For this album, he dug into his enormous collection of percussion instruments from all over the world. There were many hours spent testing shakers, panderas, tambourines or wood blocks to get exactly the right colors for each of the tunes.

In some cases, the sounds were sampled using an Akai S1000 and played back through MIDI triggers. Other times, they were played live, and there were still other times when Mickey wanted an exact doubling of something Bill Kreutzmann was playing. The sounds were layered in, using Bill's drums as triggers, and remapped to allow for processing delays in the trigger-to-MIDI conversion.

"Just a Little Light": Brent Mydland's original keyboard parts were retained from the ensemble recording, but the sounds were changed using the MIDI recording. The horn stabs were added as an overdub, but the electric piano string-like sound was a reorchestration of the original part.

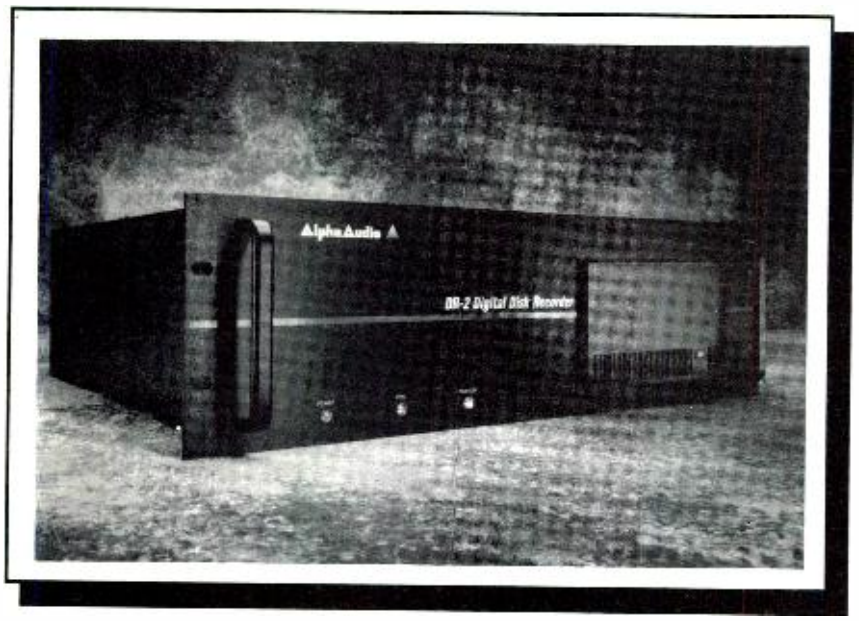
"Built to Last": People familiar with our recent shows know that Jerry Garcia has been experimenting with a MIDI guitar. Here, he played a piccolo trumpet solo.

"Blow Away": This tune has quite a few synth and organ overdubs. There's a high part doubling the piano on the choruses that was an interesting MIDI exercise. I had recorded the full piano parts in the Mac and had to extract just the upper sections of the right-hand parts, transpose them up an octave, sync them back to tape and lay in the new part.

"I Will Take You Home": Brent initially played this on his MIDIed Yamaha grand piano. I tracked the MIDI on the Macintosh and created a sampled music box. (The original notes are taken from a music version of Beethoven's "Moonlight Sonata.") I added strings from the same piano performance and Phil Lesh added bass. Jerry plays that French horn sound through a MIDIed guitar, and the soft mallet rolls on the cymbals were panned across the stereo field.

RE/P

The DR-2™ Digital Hard Disk Recorder



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Circle (10) on Reader Facts Card

MANAGING MIDI

By Paul D. Lehrman

Doctor, Doctor

I got a call the other day from a downtown studio. Actually, I should say the other *night*—I was already in bed. The owner was frantic. The studio had invested many thousands of dollars into an elaborate computerized SMPTE-compatible MIDI setup, and here it was several months down the road and they still couldn't get any of it to work.

The dealer who had sold them the stuff couldn't figure out what was wrong. The studio engineers had read all the manuals, but this only confused them. Could I please help?

I told him to calm down, and had him tell me what equipment he was using and how he had it wired. It sounded like he had tried to match two pieces of gear that I knew were essentially incompatible, so I told him to disconnect one of them and substitute a different piece he had on hand that could do the same job. He should call me the next day if this didn't work, and I'd be happy to come in and check it out.

When I hung up, I wondered how his situation had gotten this bad. I also wondered about the state of MIDI support in general. Every engineer in the recording business knows something about audio electronics, and what you don't know can easily be found elsewhere. Either your studio has a technical support staff, the dealer you buy from has a crackerjack service person on board, or the manufacturer of the piece you're having trouble with has a helpful tech-support department.

Also, in many cities there are independent technicians who have the benefit of training courses given by the major hardware manufacturers. They will come in and take care of problems on a contract or "as-needed" basis.

When it comes to MIDI, however, the situation is very different. MIDI items like synthesizers, modules, interfaces and software programs are small-ticket items, and they change very rapidly. It's hard for manufacturers to justify training anyone outside the company, knowing they'll have to do it all over again in six months.

Likewise, dealers aren't going to commit many man-hours to learning every new product that comes through the door; the small profit margins simply don't justify it. Audio equipment conforms to certain easily definable standards. If a device's input is labelled "+4 balanced, pin 3 hot," you know exactly what kind of signal to send to it, and how it will respond to that signal.

It's hard for MIDI manufacturers to justify training anyone outside the company, knowing they'll have to do it all over again in six months.

If, on the other hand, it says "MIDI In," all you know is that it will do *something* when it receives MIDI data. But *what* it will do—and whether that data should be notes, controllers, program changes, time code and/or system-exclusive—is an open question. To make matters worse, the vast majority of problems facing MIDI users don't involve getting a particular piece of gear to work. They involve getting it to work *along with everything else* in the room.

When the computer in your console goes down, the manufacturer will have procedures for fixing the problem and will be happy to walk you through them. But when your MIDI system starts spewing out random data and erasing all your sequences, there's often no easy way to know what is at fault, or even where to start looking.

Certainly IBM or Atari or Apple aren't going to help. The MIDI standard notwithstanding, it's true that some MIDI gear has trouble interacting with other MIDI gear. Manufacturers are surprisingly myopic about that fact. Rarely will a manual accompanying a piece of MIDI gear tell you much about how that item is supposed to work with other items.

If you call a manufacturer's technical-support department, you will invariably be told that their equipment is not to blame, and something else in your system must be screwed up. The reaction of most studios to this situation is to try to become as MIDI-self-sufficient as possible. But that's not easy. You can make your engineering staff read all the hardware and software manuals, although they probably don't have any more time for that than the salesman in the music store. There are plenty of "Intro to MIDI" books on the market, but given that MIDI development moves much faster than the wheels of the publishing industry, most of these books are obsolete as soon as they hit the stands.

But just as few studios would care to learn absolutely everything about their computerized mixdown and digital editing systems, trying to learn everything about MIDI can be an exercise in futility. Although MIDI equipment is relatively cheap, that doesn't mean it's not every bit as high-tech as everything else in the studio.

Therefore, you might think, there should be a growing market for a new type of technical consultant, a MIDI Support Specialist, or "MIDI Doctor." But except for a few folks who do this sort of work as a sideline (including yours truly), this job category doesn't seem to exist: There are no classified ads in the back of RE/P promoting the services of "East Coast MIDI Support" or something like that.

Why not? The supply is there; there are plenty of MIDI-studio owners (many of whom work independently out of their homes) who have gone through their own trials by fire, and who would be happy to help others avoid the same. In addition, schools and colleges all over the country are turning out MIDI-literate graduates by the hundreds, at least a few of whom have enough of a knack for the technology to be successful at this kind of work.

When your MIDI system starts spewing out random data, there's no easy way to know what is at fault, or even where to start looking.

But the demand apparently is not there, and the reason for that apparently has more to do with most studios' image of themselves, and of MIDI, than with the real world. MIDI is supposed to be a democratizing force—anyone with a couple of thousand bucks and a desire to make music can set up a MIDI studio. It's

Paul Lehrman is *RE/P*'s electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

cheap, it's fun, and a 5-year-old can do it. A studio that prides itself on its professionalism might well feel that it doesn't need to hire a MIDI consultant any more than it needs to find a specialist to change the light bulbs.

A studio might feel that it doesn't need to hire a MIDI consultant any more than it needs to find a specialist to change the light bulbs.

But as Groucho Marx once said, "Get me a 5-year-old—I can't make head or tail of this!" Interfacing MIDI equipment with professional audio gear in complex ways for complex production tasks is *not* easy. Engineers accustomed to plugging in new tape machines, microphones and signal processors without any hassles are surprised when they find out how many things can go wrong when they try to throw together synthesizers, samplers, hard-disk editors, automated mixing consoles, processors, SMPTE readers, and video and audio decks.

While at the most basic levels MIDI has established itself pretty firmly, the standard is still very much in flux out at the cutting edge. A good MIDI specialist will know what's going on at the fringes—more than you will, and even more than the manufacturers will.

There's also the issue of money. Studios that have shelled out thousands of dollars for MIDI equipment understandably feel a bit cheated when confronted with the fact that *more* money has to be spent to hire someone to teach the staff how to use it. With conventional audio gear, this initial setup and instruction is usually handled by the dealer, and the studio only pays if something goes wrong after the warranty expires.

With MIDI stuff, what goes wrong usually happens right at the start. Invariably, dealers are powerless to do anything about it (especially if, as is commonplace in the highly price-conscious musical-instrument business, the dealer is several hundred miles away). With so many studios using MIDI systems, it's surprising that the MIDI Doctor has not become a common phenomenon in the recording industry. Maybe it's time it did. MIDI really isn't all that simple, and to make it work at a professional level requires professional-level knowledge and experience.

Studios shouldn't be afraid to seek out individuals with that knowledge, and benefit from it. The idea may not fit in with their image of MIDI as the "poor sister" of the pro-audio world, but it could save them a lot of time and money.

As for my friend downtown, he never called back. I guess I solved his problem for him. Now I wonder—do I send him a bill?

RE/P

SPARS ON-LINE

By Steve Lawson

Workstations for Broadcast Production

Changing technology is the only constant in our industry, and those with educated foresight are those who will prosper from the ongoing technical advances. In the relatively short history of audio recording, big changes have come quickly — from cylinder to disc, to wire, to tape.

From full-track mono, we moved to 3-track and 4-track formats with the obvious production and mixing benefits. More tracks were what we wanted and needed; along came eight and 16, quickly followed by 24. Plates and springs replaced too many echo chambers. Noise reduction did away with tape hiss. Then we welcomed computers.

Computers, in fact, took over the studio. They paved the way for synchronization, which allowed us to lock our various machines together. They controlled our consoles through automation. Not only could we fine-tune our work; we could easily recreate a sound mix two weeks later, making the inevitable production revisions and variations easy to accomplish.

Computer technology brought us great digital reverb units and delays, and we could simulate the ambiance of a large hall or a small room with the twist of a knob. We kissed our small spring reverbs goodbye and forgot about the hassles of tape delay. We were thrilled with these new toys, and we thought we had found recording heaven.

Then came the biggest change of all — digital recording. As I recall, we didn't exactly rush in with our checkbooks, did we? Remember that last New York Waldorf AES Convention? We didn't quite grasp the concept. "You record sound on ¾-inch videotape? Why? How? Bits of data? Zeros and ones? What do you mean I can't edit it?"

I doubt that many could have foreseen the remarkable changes that would sweep across the entire industry, but recording onto ¾-inch videotape became the norm for CD mastering. Many studios bought the Sony F-1 digital processor, which was inexpensive and small. We adjusted to the need for a VCR for data storage, but there was frustration because you still couldn't edit with the degree of accuracy necessary in sophisticated audio post.

The digital audio workstation has changed forever the way we approach production. These marvels of computer science sound great, but so does analog. Simply choosing to use a digital workstation for better fidelity is mighty costly at best. Although digital recording is wonderful, it's the *workstation* aspect that is truly revolutionary.

No one would have switched from a typewriter to a word processor if all it did was print a page. But just as a word processor allows the editing, repositioning and effortless manipulation of the written word, the digital workstation allows you to cut, paste, move change and merge sound. The advertising agency producer is in a big hurry and wants to hear take 1? Just click on it. He wants to hear how the beginning of take 42 sounds with the end of take 27? Click on that.

It's as fast as the point-and-click of a mouse, or the twist of a knob and the push of a button. No rewind, no chasing — and no waiting, which is the name of the game.

The digital workstation is a dream come true for the production engineer. Recording, editing and multitrack layup are finally interrelated. Sound is converted to digital data and is stored on hard disks. Once stored, it can be edited to your heart's delight without destroying your original recording. There's no more searching through the waste basket for that elusive half-inch of tape you thought you would never need again. The engineer has a massive library of music, dialogue and effects at his fingertips, and instantaneous manipulation is made easy.

Sound effects can be placed in real time as on a multitrack recorder, or by time reference. If your initial placement isn't right, there is no need to re-lay. Just move it a few frames. Wait: We're starting to talk like video editors. We're even starting to think like video editors: hours, minutes, seconds, frames and sub-frames.

You're ready to buy several, aren't you. Slow down. Now is the time to do your homework. For the production studio, a workstation is likely to be the biggest purchase you'll ever make. Digest all the literature, and get a personal demonstration on the units that interest you most. Make a note of what you like and dislike about each. Talk to owners. Ask how well they work, how often they crash and how responsive the manufacturer is when a machine does crash. Ask if the machine is making them any money.

Then, choose the device that works best for you, your engineers and your clients. And plan to pay it off quickly, as you never know when the newer, better and cheaper will be thrust upon us. Yours will still work great. But that studio down the street will certainly buy the new one, if only to make your life miserable.

In September of 1989, SPARS members from around the country met in Chicago to see the latest from our advisory members, the manufacturers who build these machines. We saw presentations from AMS, DAR, Lexicon, New England Digital, SSL and WaveFrame. We spoke to the manufacturers between presentations and had informal conversations between breakfast, lunch and dinner. And they spoke to each other, discussing their common problems and opportunities. We spoke of such concerns as our need for standards that would allow one company's machine to talk with another's.

It is through this type of dialogue that we, the users, can participate in the evolution of new technologies. We all left Chicago feeling that we had gained valuable insight into an exciting future.

As technology changes, the recording business inevitably becomes more complex. Membership in SPARS gives you a crucial edge by exposing you to new information and new ideas while they are still in the formative stages. SPARS is a communication network, an open forum where you can share and learn.

Believe me, digital workstations are here to stay, and they do sound great. But that's only the icing on the cake. The digital workstation is changing not only the way our work sounds, but how it gets through the production labyrinth to the final master. If you aren't yet a member of SPARS, join us to help make the changes that will work best for you in the 1990s.

REP

Steve Lawson is president of Steve Lawson Productions, Seattle, and a SPARS director.

SPARS stands for the Society of Professional Audio Recording Services. For information on activities or membership, contact SPARS at 4300 10th Ave. N., Lake Worth, FL 33461; 407-461-6648; fax 407-642-8263.

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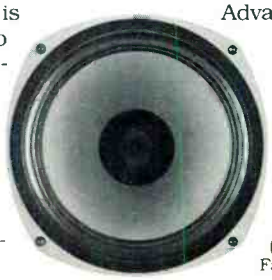

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Circle (13) on Rapid Facts Card

CARING FOR VACUUM TUBE NEUMANNS

△By Dan Alexander and James Gangwer

Straightforward advice for improving and repairing your tube-type Neumann microphones.

Tube-type Neumanns are often among the most prized possessior dio can own. As with all equipment, they occasionally need mainte For some reason, many technicians capable of repairing complicated circuits in tape machines or consoles shy away from even the simplest microphone repair. Considering the simplicity of the circuits, this reticence seems unfounded.

Many times, we have fixed tube-type mics and power supplies by replacing their decades-old resistors and capacitors. Broken wires and dirty capsules (diaphragms) are also common problems.

We do not recommend that you start cleaning your own diaphragms, but many other repairs should present no real problem. With a few mics, gaining access to the interior requires some familiarity with their construction. But most come apart with the simple application of a quality jeweler's screwdriver.

A word of caution: Try not to puncture the diaphragm with your screwdriver, and don't strip the screws that hold the mic together. Many of the microphones are constructed in such a way that the threads screw into the mic body, instead of being unscrewed from it. We use jeweler's screwdrivers made by Starett, and these seem quite acceptable. Don't use cheap tools on your expensive mics! And don't lose those little parts. They can be very hard to find.

Connections

When rebuilding mic systems, one common problem is the connectors into and out of the power supplies. Most often, we replace the old Tuchel and custom connectors with 5- or 6-pin XLR connectors and put a matching XLR on the cable coming from the mic.

Dan Alexander and James Gangwer are with Dan Alexander Audio, Berkeley, CA.

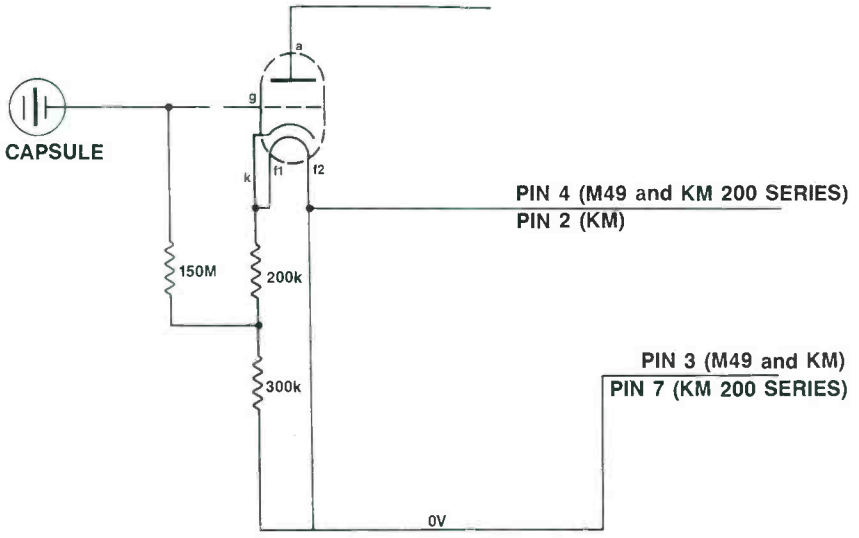


Figure 1. Partial schematic of stock Neumann M49, M50 and KM Series microphones.

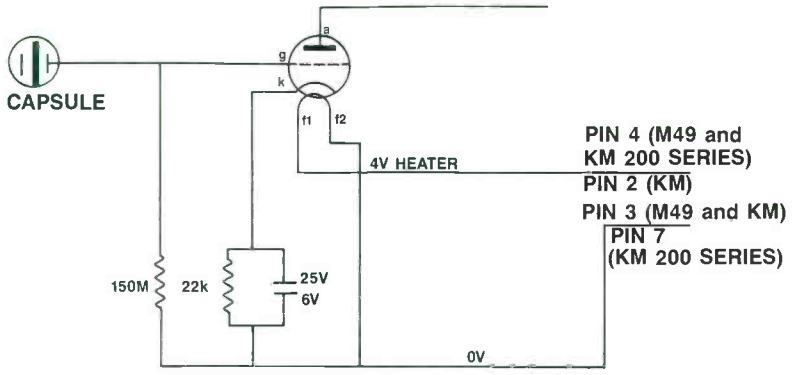


Figure 2. Partial schematic of M49, M50 and KM Series mics after modification of Telefunken AC701K vacuum tube circuitry.

It's nice to replace the old cable with Mogami or some other high-quality, low-resistance wire, but that requires rewiring the connector that goes into the microphone. That's sometimes a chore, as you have to figure out how to get it apart to begin with. Most of the old Tuchel connectors for the Neumann mics can still be obtained, but some of the custom-made connectors for the old AKG's are not so easily found.

Once you have taken the connectors apart, it's relatively easy to build a new cable from the old parts. It helps to have a schematic, but careful notetaking will suffice in a pinch. The cables are mostly straight-through connections, but it can be confusing when you are changing both the connectors.

We always change the audio output of the power supply to a male 3-pin XLR, if the mic wasn't supplied with one originally. This sometimes requires construction of a small metal plate to hold the XLR, as the original connectors were often larger than the XLR. It's not too difficult, but it helps to have the proper hole punch and a drill press. We often replace the ac connectors with a captive ac cord. Be sure to use sufficient strain relief when doing so.

Leaky filter capacitors in power supplies is a common problem, and finding direct replacements can be difficult. It is not the value as much as the mounting and size that presents difficulty. We sometimes have to secure replacement caps with a gob of silicon glue or some epoxy cement. Remember, the filtering of the power supply current is very important, so use at least as much capacitance as the original supply provided.

Most manufacturers had the foresight to use variable ac voltage-tapped transformers in their supplies, but there are some (for example, the Neumann N52A) that were strictly 220V. Here you can use a commonly available 110/220 step-up transformer between the ac and the supply. You should not use the very small autoformers intended for traveling electric razors! Get a 40W to 80W (or larger) transformer from your local electronic supply house. It should cost from $15 to $30.

You can run more than one 220V power supply from a single step-up transformer by using a power strip. Label the strip well — do not plug any 110V gear into the 220Vac! However, it won't hurt anything if you plug 220V gear into 110V, although it will make your mics sound awfully noisy.

When you change the voltage on a power supply from 220V to 110V, you will have to double the value of the internal fuse. Although they are more expensive than a U.S.-style fuse, the smaller European-style fuses are available from a decent electronics supply house. Or you can try an imported auto repair shop.

A mod for less noise

With the intent of improving the noise characteristics of these valuable microphones, here is a bias modification that Neumann developed for its M49, M50 and KM series of tube mics. We do this particular mod to all of these models that we sell. In the manufacturer's terminology, we are modifying an M49 or M49B to an M49C, or a KM54A to a KM54AC. The end result will be a noticeable reduction to the noise of the microphone.

The M49, M50 and KM types use the Telefunken AC 701K vacuum tube. Bias modification of the AC701 microphones requires that you separate the cathode (K) from the heater lead (F1) and connect it to a resistor in parallel with a cathode bypass capacitor to ground. The resistor value for KM or M49/M50 mics is 2.2k, and the capacitor is about 25μF at 6V. The desired effect is to have 1.6V from the cathode to ground, as this was found to produce the lowest noise level.

For those readers who have not seen one, the AC701 is a tube with leads instead of pins. The leads are fairly fragile. There is one lead coming from the top of the tube and four from the bottom. The four

leads from the bottom of the tube are arranged in a square pattern about the circular bottom surface of the tube. There is a spot of red paint next to one of the leads that identifies it as the plate (anode) lead. The leads are arranged, clockwise, in the following order:

A: anode (red spot).
F2: heater ground.
K: cathode lead.
F1: heater lead.

The biggest challenge is separating the leads of the AC701 without breaking them off. In the original assembly, the leads are soldered together and curled into loops. You must be careful when uncurling and desoldering them. We suggest placing a medium-sized alligator clip on the leads as close to the body of the tube as possible, in order to isolate the base of the leads from bending while you are straightening them out.

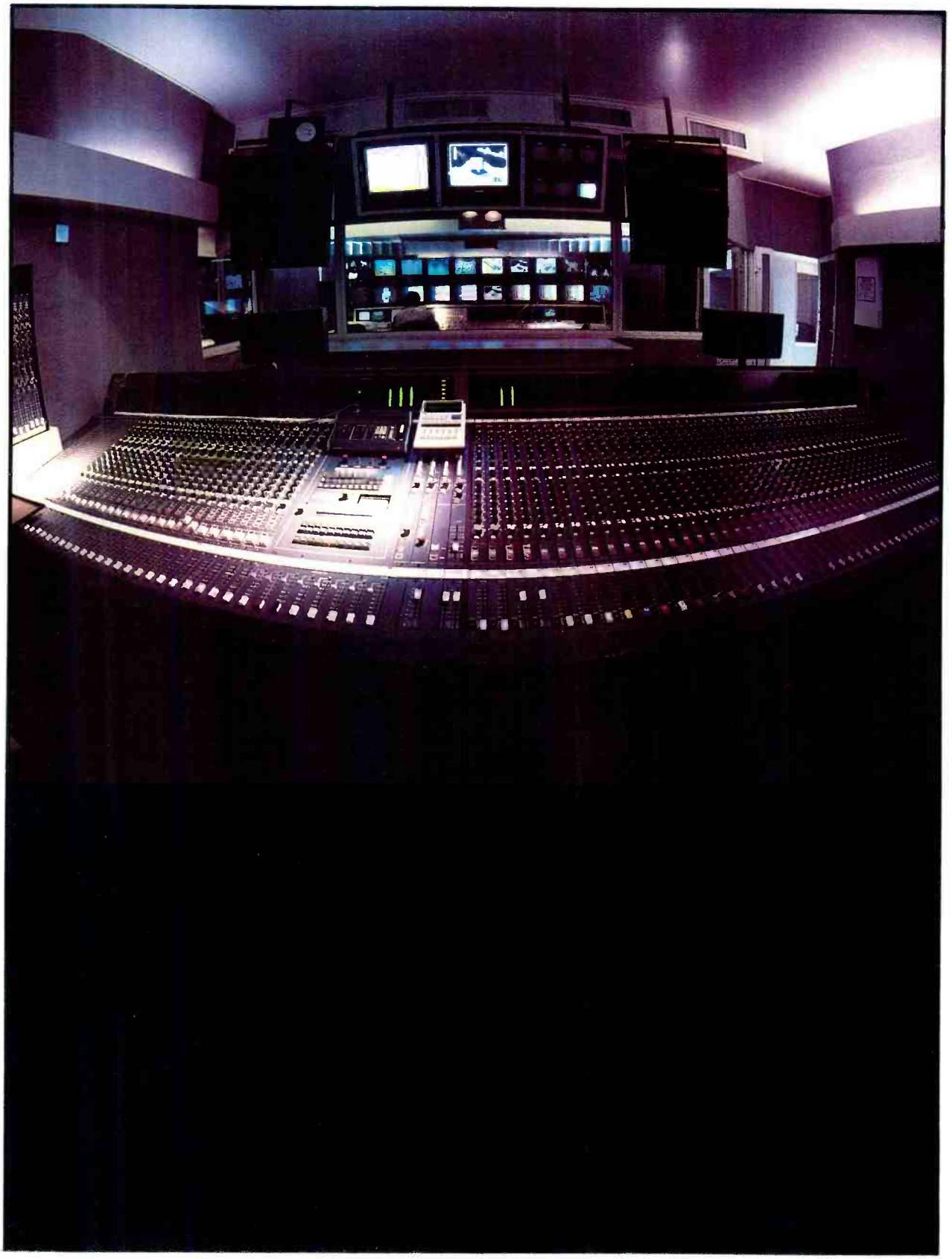
In addition to separating the leads from each other and adding the necessary parts, there are some parts to be removed. Where the cathode and heater leads were connected, a 200k resistor is wired in series with a 300k resistor to ground and a 0.01μF capacitor to ground. From the grid lead (single wire from the top of the tube) there is a 150meg resistor that connects between the 200k and 300k resistors mentioned previously. The tandem 200/300k resistors should also be removed. The 150meg resistor should run to ground.

In the M49/M50 series, the 200/300k tandem resistors are located on a pair of terminals to the left of the bottom of the AC701. You should use these terminals to mount the new parts. Before you run the newly straightened cathode lead over to them, put a piece of shrink tubing or sleeving on it to insulate it from the other leads because it will have to go under the other leads to get to the terminal.

In the KM-type mics, the 200/300k tandem resistors are located on the little circuit board that fits over the transformer. This is also a convenient place to put the new parts, if you have ones that are small enough to fit. Be careful to use the pads that the original resistors use, as you don't want to connect them to anything live.

Using a little care, you should be able to improve the noise spec of your AC701 mics without affecting the beautiful sound in any way.

One final word: To keep your mics founding lovely, always use a wind screen when recording vocals. RE/P



MAKING THE FRENCH CONNECTION

By Peter Jostins

Sandwiched between the U.S. Embassy and the Champs Elysee in Paris is one of the most amiable television studios in the world, not by chance but by design.

This is a story of success on a grand scale. The success of one man, Michel Drucker, who has topped the TV ratings in France for 12 years with his show "The Champs Elysee," a Saturday night spectacular combining talk, music and comedy. Every week, the show is seen by 60 million people in 35 countries and is renowned for the celebrities that it attracts: Clint Eastwood, Elton John and Sylvester Stallone, to name three.

Not content with just presenting the show, Drucker has become increasingly involved with the operational and business affairs of the studio. He now owns the studio, its equipment and the rights to the show itself. This monopoly gives Drucker complete control over the show. Consequently, he can offer his guests not only a trip to Paris on the Concorde and a day's shopping on the Rue du Faubourg Saint-Honore, but also provide the perfect atmosphere for uncandid conversation and spontaneous musical performance.

Before Drucker took control of the show, the studio had a poor reputation for its in-house sound. The first step in Drucker's master plan to provide the ultimate studio atmosphere was to employ the best sound engineers and install the best house and stage monitoring system. In this way, every performer would have the same monitoring quality and volume that they were used to on stage. This was a difficult task, considering the diversity of performers that the show takes on board.

The second stage was to discard the OB vans normally used for transmission and build a complete recording and post-production facility within the studio complex, so that the show could be prerecorded and then transmitted after editing and sweetening.

Although the design brief was fairly straightforward, the variety of musical styles to be accommodated made the choice of suitable equipment quite difficult, particularly as the engineers had suffered from poor equipment in the past. It was obvious that expert advice

Peter Jostins is a London-based studio owner, producer, design consultant and free-lance writer.

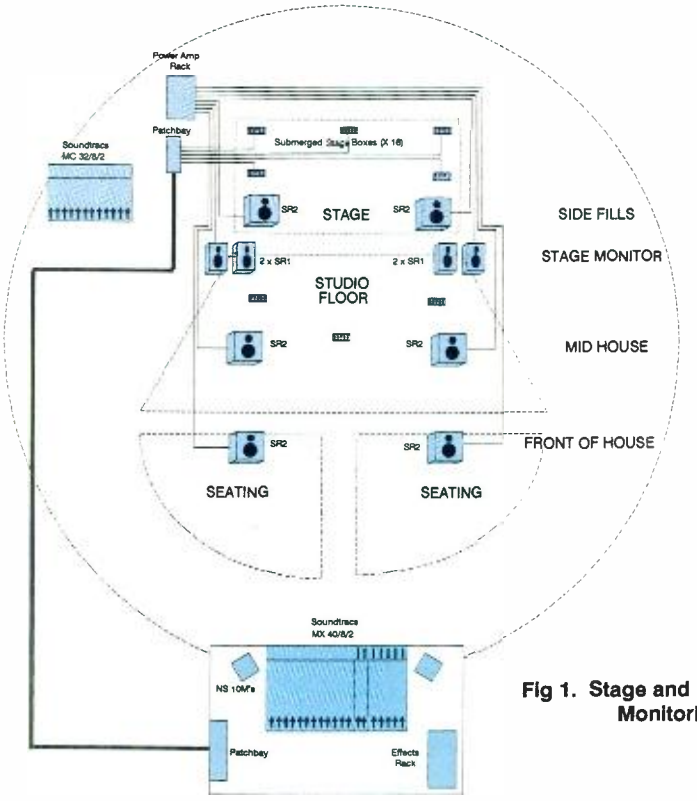


Fig 1. Stage and House Monitoring

Figure 1. *Stage and house monitoring.*

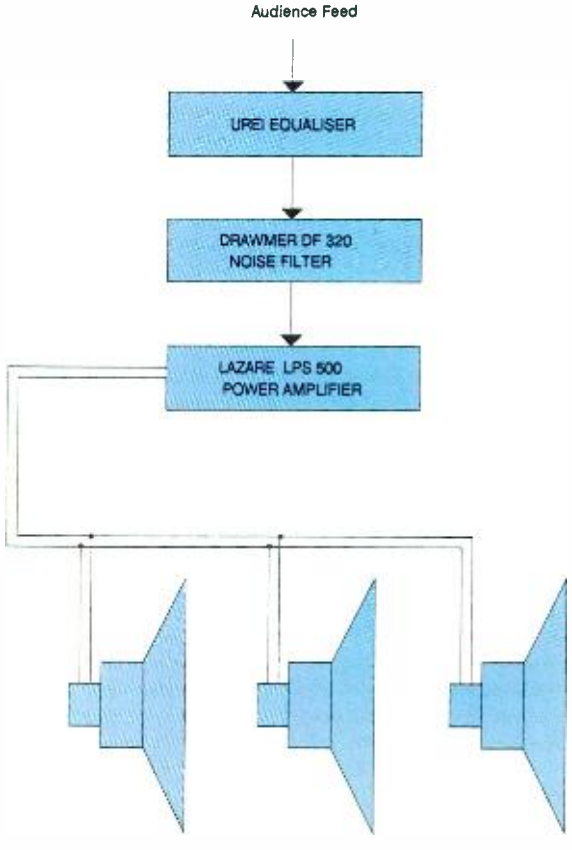


Figure 2. *Seat monitoring system used with lavalier mics.*

was needed. Drucker called upon Lazare Electronic to specify and install the necessary equipment. Lazare Electronic has years of experience in installation and also manufactures PA and recording equipment, so it was the logical choice for the installation.

After much deliberation and on-site tests of equipment, the design spec was completed. In 1988, during the summer shutdown of just 10 weeks, the studio was transformed from a simple show stage and lighting rig to a complete recording and post-production facility.

The building work

The first stage in redevelopment was to remove the partition walls to the front of the studio, which was originally the kitchen. Once the space had been cleared, a new concrete floor was laid; partition walls enclosing the audio and video control rooms were erected. To maintain isolation, these partition walls were also made of concrete. However, to ensure clear visual communications between the video and audio control rooms, the two were built back to back, separated by a glass partition.

The walls were then plastered and acoustic tiles were mounted on 3-inch battens with rockwool in between. Even the corridors were treated in this way to minimize the noise made by footsteps. The false floor, consisting of 15-inch square flooring slabs, was then fitted with each slab mounted on aluminum pods. Each pod was then electrically earthed with copper braid to ensure against ground effects and static. Also mounted beneath the floor were the audio video cable ducts, with access hatches at regular intervals and junction boxes at the entrance of each room.

Finally, the suspended ceiling, made of acoustic ceiling tiles, was fitted along with flush-mounted lighting and brackets for the video and loudspeaker monitors.

In-house monitoring

During a typical show, the studio will play host to a rock band one minute and an aging French cabaret star the next, so the monitoring equipment must be flexible enough to accommodate all styles of performance. Moreover, it must be able to deliver large amounts of power to excite the audience and performers, yet not be so large that it dominates the stage set. As a result, the monitoring specified is a blend of rock 'n' roll PA and TV amplification, carefully chosen for its power/size ratio and its flexibility.

Gilles Hugo, the front-of-house engineer, says, "Before I started here, the relationship between the sound engineers and the performers was very poor, particularly

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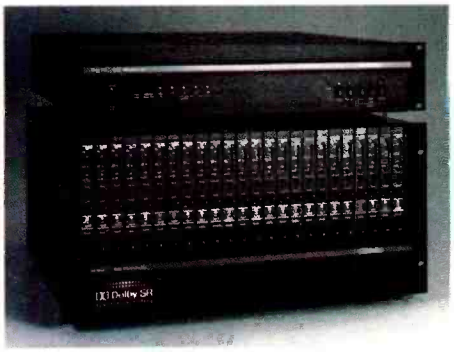

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where rock bands were concerned. The engineers would provide the artists with a quiet and distant monitor and then ask them to keep it down! The first thing we did was to request better stage and house monitors and more comprehensive consoles.

"This didn't automatically bring in the right equipment. In fact, we had to try out numerous speakers and amplifiers, and about 10 consoles, before we decided on the most suitable equipment. Eventually we chose the Renkus Heinz SR series monitors powered by Lazare Electronic LPS 500 amplifiers and Renkus Heinz X22 processors."

The monitors are flown in at either side of the studio flush with the lighting gantry and arranged as two SR2s for front-of-house, two SR2s for mid-house, four SR1s for stage and a pair of SR2s as portable side fills on the stage area. (See Figure 1.) Each of the monitors is bi-amped and fed in pairs via a Renkus X22 processor. The processors and amplifiers are mounted in a vented amp rack above the stage.

In addition to the flown monitors, 20 Renkus/APG standard wedge monitors and six APG micro wedge monitors were supplied to cover all eventualities of on-stage monitoring. With such a large amount of wedge monitors, and with so many scene changes within the show, it was necessary to provide speaker outlets on the stage so that monitors could be moved and re-patched quickly. Consequently, Lazare installed permanent speaker wiring from the 30 power amps mounted above the stage to the submerged stage boxes below. An XLR patchbay is located at the side of the stage for patching to the relevant amplifier. This use of permanent amplifier wiring also reduced the number of cables on stage and kept wiring out of camera shot.

Hugo says, "Of course, having chosen the equipment, we had to convince the filming crew that in some cases it is necessary to place a couple of wedges in camera shot so that the bands could hear themselves properly."

Given Drucker's determination to improve the studio in the eyes of the performers, the filming crew agreed that the wedges were necessary and now the studio has an enviable reputation for its in-house sound.

For normal programming (chat show, comedy, etc.), Sennheiser SK 2012 lavalier microphones were supplied, but as with most lavalier microphones, the problem of feedback occasionally reared its head. The engineers at Lazare Electronic developed a unique system for audience monitoring. They attached 50 small loudspeakers, normally found in automobiles, to the rear of the audience seats. Each of these speakers was fitted with a 100Ω line transformer; all were then fed in parallel from a UREI equalizer, Drawmer noise filter and Lazare Electronic LPS 500 amplifier. (See Figure 2.) The result is an apparently omniphonic sound field with excellent speech intelligibility and no disconcerting monitor speakers pointing at the audience or in camera shot. In addition, the seating acts as a baffle for the speaker, amplifying and directing the sound at the audience rather than the stage, and increasing the sound pressure level before feedback.

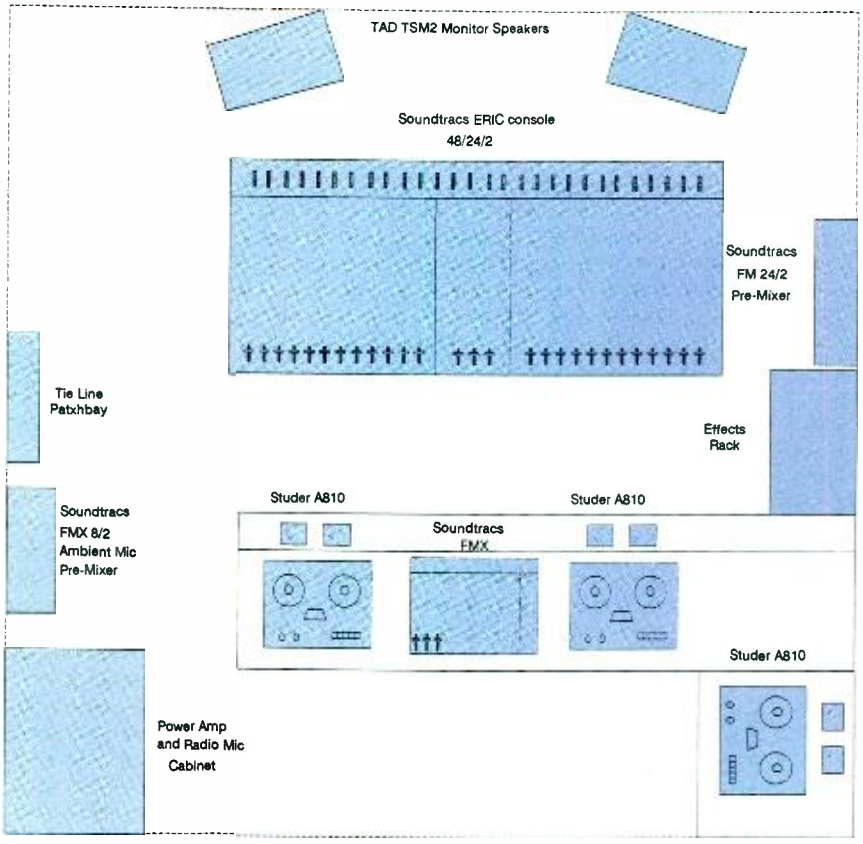


Figure 3. *Control room and Second Wave studio.*

The remaining house equipment was mounted in a open-windowed booth at the rear of the audience seating. The main mix is controlled from within the booth via a Soundtracs MX console with 40 inputs, along with a vertically mounted Soundtracs FM 8/4/2 console for submixing strings and providing foldback to the dressing room and foyer. Monitoring within the booth is via Yamaha NS 10M monitors; a comprehensive rack of effects provides extra coloration of the in-house sound for the audiences benefit.

As far as microphones are concerned, the studio was already equipped with almost every microphone ever manufactured. Radio microphones, however, had been a problem in the past, and the ones used were either ugly or bulky in appearance or they were only useful for speech or for singing, not both. Fortunately, the Sennheiser range of hand-held radio mics satisfied all criteria, being both small and attractive, and responsive to both speech and singing. Ten SK4031 mics and EM 1036 receivers were installed, operating between 670kHz and 702kHz. The signal from the EM 1036 receivers, which are housed in a 19-inch rack in the control room, are then distributed via tie lines to the control room, front of house booth and stage.

Finally, communications between stage, control room and the front of house booth is via a network with full duplex transmit and receive on each station. Through the network any station can call any of the other 12 stations on the network. In addition, a Clearcom system is used for priority communication between Drucker, the producer and Hugo so that any problems on stage can be relayed quickly without being mixed up with the normal, technical communications lines.

Control room monitoring

Situated behind the video control room is the sound control room where the main sound sources are mixed and sent to the video recorders, or to the transmitter,

when the show is live. At the heart of the control room is a Soundtracs ERIC console with 48 inputs. The ERIC console is digitally routed with digital control of mutes on all channels, auxiliary sends and monitors, and also digital control of input switching. Digitally routed consoles are becoming increasingly popular in live applications due to their capability to be programmed during rehearsals, but they are not commonly used for television. Hugo says the ERIC was chosen for its versatility.

"The console obviously has all of the necessary facilities in terms of EQ, aux sends, etc. Using the digital routing we can program the console during rehearsal and then reconfigure it during the show by pushing a single button," he says. "But we also wanted a console that could be used in post-production for automated mixing so that we could mix the sound after the recording.

"The engineers here all felt that the quality of sound for television should be improved upon and, therefore, they wanted a console which would double as a live board for on-air and a recording console. Before we finally chose the ERIC, however, we asked Soundtracs for input metering on all channels and VCA subgroups. Soundtracs kindly obliged."

On the effects front, the studio wanted to offer the artists signal processing comparable to that of a top recording studio, so Lazare supplied a Lexicon 480 XL, an AMS RMX 16 and an EMT 246 reverb, with a couple of Yamaha REV 5s and an SPX 90 thrown in for good measure. Gates and compressors are all Drawmer. For sound effects, a Roland D550 sampler was added. To prevent hum and noise from third party equipment degrading the system, three BSS MSR 604 splitters were supplied.

Second wave studio

Behind the ERIC console is the second wave studio which is used for premixing and editing all stereo material. (See Figure 3.) Equipment includes three Studer A810 master machines flush-mounted within a customized console for ease of editing. The meter bridge for each machine was then built into the angled bridge panel of the console and on racks to the side are all the other stereo sources, such as tuner, CD, cassette machines and turntable. All of the stereo sources in the control room are mixed together on a Soundtracs FM 12/2 console with stereo inputs.

Submixer

Although the ERIC console is capable of handling most of the sound sources, submixers were added to provide more flexibility. In the event of three or four large bands appearing on one show, the work can be distributed to satellite consoles, each with its own mix engineer. To this end, a Soundtracs FM console with 24 inputs was vertically mounted to the right of the ERIC console as a premixer for strings or drums. And to the left of the ERIC console is yet another vertically mounted FM console for premixing the three ambience mics in the studio.

The results of this refurbishment and augmentation of the studio have been on trial for the last year, and everyone involved with the show agrees that the programming quality has improved dramatically. The return of Sting for the third time and an impromptu 25-minute live performance from Elton John stand as testaments to Drucker's philosophy: The performers come first.

RE/P

FROM LACQUER TO LAYBACK

One facility's transition from music to broadcast recording and mixing

By James Porteous and Robert Predovich

A recent MIT study revealed that the television audience would tolerate poor-quality picture reception as long as there was good sound, but not the reverse. Does this mean that television really is just radio with pictures? Not quite, but the important role of good broadcast sound is finally emerging after decades of playing a poor cousin to visuals.

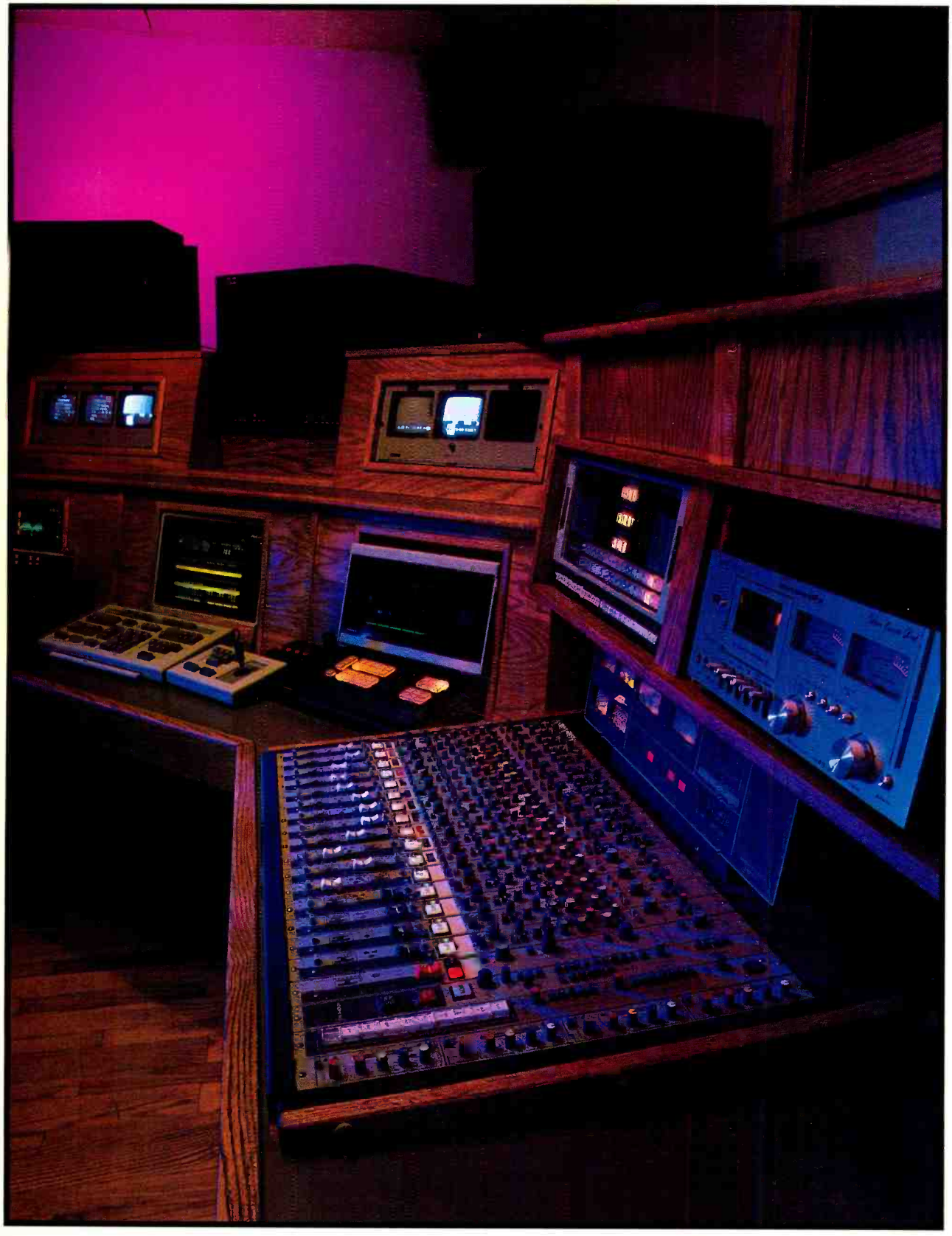
As the demand for quality increases, so does the need for the expertise and talent to deliver it. Music recording and mix engineers have long honed a work ethic that stresses attention to detail. This commitment to excellence will continue to be required if we are to attain the full potential of broadcast sound in the 1990s. Although there will still be opportunities to direct one's attention toward strictly being a music engineer, the merging of audio and video media has already channeled a growing number of engineers into the realm of mixing to picture.

Master's Workshop was strictly a music studio for the first 10 years of its 17-year existence. Today, a blend of music, broadcast and film engineers produce the entire soundtrack for television programs such as "Friday the 13th," "War of the Worlds" and Farrah Fawcett's ABC mini-series "Small Sacrifices." Additionally, Master's produces audio for feature films and IMAX 6-channel presentations.

If you're used to the precision and flexibility available when cutting tracks for a CD release, you might be shocked at the limitations lurking in the broadcast sound chain. Yet, with a road map of the potholes along the way, you should be able to achieve excellent results.

One such difference between music mixing and mix-

James Porteous is the director of re-recording, and Robert Predovich the vice-president and general manager, of Master's Workshop in Toronto, the audio post-production division of Maclean Hunter Ltd.



Northwest Teleproductions/Kansas City, MO. Photo: Don Yaworski.

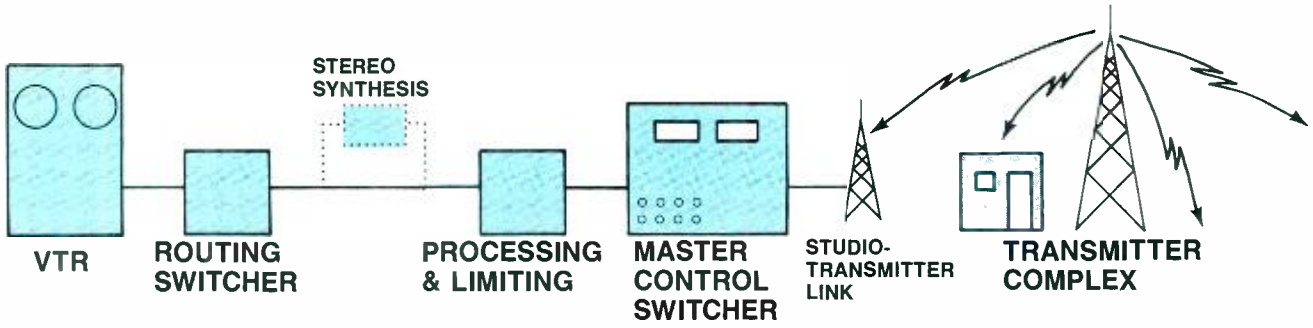


Figure 1. Simplified flow chart of the broadcast chain.

ing for broadcast is the usable dynamic range. Although digital and Dolby SR recording offer signal-to-noise ratios exceeding 90dB, practical considerations in television limit typical program level deviations to a much narrower range. Most network standards suggest that the highest VU peak be no greater than +1VU, and the lowest non-ambient information not less than −10VU. This 11dB window is where the dialogue, music and specific sound effects live. Elements of the track mixed lower than these specifications risk being lost while the kids play soccer in the living room. Deviations above risk the wrath of numerous, unforgiving compressor/limiters.

Ubiquitous throughout the broadcast chain, to ensure compliance with regulatory edicts and prevent overmodulation and distortion from some overzealous "red hot" mixing engineer, these compressor/limiters are generally set to instantaneously pull back peaks. Unfortunately, the recovery time could be set for as long as five to 10 seconds. The first machine gun shot might make it through, but the rest of the round will be blanks!

The damaging action of these devices cannot be taken lightly. Broadcasters, assuming the lowest common denominator, sometimes set these units to adequately control excessive, and seriously improper, level deviations only to the horror and detriment of those engineers trying to stay within the guidelines.

Clearly, you need to be just as familiar with various processes outside your direct control as those within it. If you are going to be serious about mixing for broadcast, and are confident that you are doing your part of the process correctly, be prepared to spend numerous hours solving other people's problems.

Tracing the audio chain

A few years ago, the first episode of a series Master's had mixed went to air locally sounding atrocious. It was dull, lifeless and lacked dynamic range. Yet all agreed that the show sounded tremendous in our mixing theater. Welcome to the most common underlying problem in mixing for broadcast: What happened to the mix after it left our building?

Having previously mixed hundreds of programs in our theater and heard them translate well on a home receiver, we embarked on a lengthy troubleshooting process to determine where things were going wrong. One of our first clues came when a representative of the broadcast organization reported that they had found a gunshot in our mix that peaked exactly at 0VU. They were used to seeing gunshots pin the VU meter's needle, and were therefore convinced that we were overcompressing!

Once we explained that we deliberately controlled our levels with sophisticated compressors so that peaks wouldn't hit their "brick wall" units, and then spent some additional time building their confidence in us, the conversation shifted to the transmission side of the equation. To their credit, they invited us to help them trace their audio chain, right from their originating videotape machine through network control, the local station's master control and out to the transmitter site.

We found misaligned distribution amplifiers and videotape machines that were affecting program levels and frequency response. As well, one of the limiters in the chain was actually set to a threshold of −2VU. Our 0VU gun shot peak was being trounced!

We learned that the limiter had been set that way to adequately deal with the majority of soundtracks that the network received. Most were excessively hot and had to be controlled. We recommended increasing the threshold to 0VU, and asked if the limiter could actually be taken out of the chain for the transmission of our program, as we had already carefully controlled our levels.

Thanks to a supportive team at the broadcaster, the next episode went to air with a properly balanced and aligned chain, and the limiter was removed from the circuit. A VHS Hi-Fi off-air copy sounded remarkably like our master mix, to the extent that many people were surprised that a broadcast signal could deliver such quality. More and more broadcasters are now devoting resources to better audio delivery, thanks to the advent of stereo transmission.

Proper levels aren't the only key to navigating through the tolerances of the transmission chain. You also need an understanding of the interdependence of specific frequencies. The broadcast spectrum is fairly generous, allowing a range from about 80Hz up to about 15KHz. This is roughly similar to the FM broadcast bandwidth.

However, exact tolerances must be followed by FCC and DOC law in order to stop crosstalk or bleeding onto adjacent frequencies. In satellite transmission, excessive energy at around 5kHz has the ability to deviate the carrier frequency. This is important to note as 5kHz is a warm frequency when equalizing dialogue. Too much and it's hello Jupiter!

Needed: audio standards

One of the hardest things to come to grips with in the television industry is the lack of meaningful standards for sound. Often, the standards that do exist are disregarded. Although recording engineers are used to laying tones such as 1kHz, 10kHz and 100Hz at the head of a tape and aligning an out-of-house tape for playback using the supplied sweep, the broadcast industry has been very slack in this area. The standard is simply to record program level tone, typically 1kHz, while a complete set of color bars is recorded for picture reference.

Not only are possible deficiencies in the high- and low-frequency range not discovered during the record stage, but an operator has no idea on playback if he is correctly reproducing the proper spectrum of sound. Laying just a program level tone for sound is like recording color bars with-

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out the chroma (color) information. If this procedure were followed for video alignment, peak white information (maximum video level) could be adjusted, but no data would be available to set proper flesh tones or color rendering.

The video industry would never consider this acceptable, yet this is exactly what the standard has been for many years for the sound component of a videotape. One can just imagine the problems that this can cause. If the lows or highs are boosted on playback due to improper alignment, those specific sound effects that were ever so carefully mixed to stay within +1VU now are elevated to levels that hit the transmitter's compressors with a vengeance. If the high frequencies on playback are deficient, your mix is going to sound pretty dull on air.

Take this one more step, and it really gets frightening. You have just finished your music special and sent a ¼-inch stereo master with center-track show time code to a video facility to be "married," or "laid back," to the video master. If the videotape recorder is a 1-inch C-format unit, the most common today, the chances are that the audio section has been aligned once in the last six months. Maybe. If you are lucky, the tones on your ¼-inch tape are aligned for playback, but the odds are that bars and tone have already been recorded at the head of the videotape. That videotape machine might be down 3dB at 10kHz, but the operator will never know as the program content will simply be dubbed across, not your tones.

Figure 2. *Master's Workshop attaches this label to all tape boxes to ensure that the tape will be properly aligned machines.*

Terry Gordica (left) and James Porteous during a mix at Master's Workshop Digital Theatre 1.

A layback solution

We spent half our lives tracing these types of problems until we purchased our own videotape layback machines five years ago. In an effort to bring a standard into the audio component of "bars and tone," Master's Workshop has been laying 1kHz, 10kHz, and 100Hz, as well as Dolby tone when noise reduction is requested, along with the color bars. We created a label to be applied to the tape box and reel that states that the three frequencies are present and explains what to do with them.

Don't think that this is overkill. Many tape operators line up videotape players with the speaker *off,* having had to listen to enough tone over the years. If that videotape player is misaligned in the high or low end, and he adjusts the 100Hz tone, which is +2VU referenced to 1kHz, as a program level set, guess where your peaks are going to end up?

One-inch videotape layback machines are professional audio quality units that have head stacks that access the two program audio channels and the time code channel on the tape, without reproducing any picture. Their electronics and heads have been optimized to deliver superior audio response than most videotape machines in record and reproduce modes. Because they possess a full-response reproduce head, which is not part of a videotape machine's configuration, bias and record alignments can be accomplished in real time while watching the reproduced result.

Without this capability on a videotape machine, this simple bias and equalization procedure becomes quite involved. Every tweak results in the recording of a section to check its effect, then rewinding and playback. As well, the alignment controls on a typical machine are placed in a position that assumes they won't be used very often. This helps to explain the frequency with which these adjustments are attempted in most facilities.

We take advantage of the ease with which these tasks can be accomplished on a layback machine. Playback equalization is checked before each session, and the unit is biased and equalized for recording each individual tape. As well, recordings

can be monitored off the playback head to catch anomalies such as dropouts.

Some of the alignment problems associated with 1-inch videotape machines are giving way to the excellent linear response and freedom of setup that result from digital audio. Digital videotape machines, in the form of the D1 and primarily D2 formats, are now being used in the broadcast environment. This will remove a major pothole on the road to good broadcast sound, but it will be many years before the industry in general can afford to switch to this new technology. In the meantime, either keep control of the layback to videotape if you can, or have the conviction to attend the restripe at an external facility and ask to see a sweep on the videotape machine.

Mixing styles: stereo and beyond

When it comes to styles of mixing for broadcast, remember that, just as monitoring environments available for a hit single range from a car radio to an audiophile's state-of-the-art speakers, there are several devices that will be used to listen to broadcast sound. For this reason, certain windows must always be maintained in many facets of a mix, ranging from the presence that dialogue can have to mono compatibility of a stereo program. Various speaker configurations that can reliably simulate the ultimate listening devices must be at hand so that a mixer can quickly do cross-checks.

In the last few years, stereo has quickly become a standard for television series. Though many experimented with other permutations at the beginning, most mixers now blend mono dialogue and stereo music and effects. Many of the stereo effects are ambiences, with a splattering of specific effects positioned to follow screen action. Phase error induced may sound great in a stereo mix, but disappear completely in mono. You cannot forget the least common denominator.

Matrixed surround sound processes, where four channels of Left, Center, Right and Surround are encoded into a 2-channel transmission compatible pair, will become much more prevalent in the 1990s. Manufacturers of consumer receiving equipment can now install for a few dollars a large scale integrated circuit containing the active decoder in their products. The consumer demand for encoded television programming will force the mixing community to become more and more familiar with the process and its idiosyncrasies.

Additionally, discrete forms of multichannel program presentations may be around the corner. The HDTV digital videotape machines contain eight channels of digital sound. Having produced the digital soundtrack for the world's first HDTV series, we at Master's are involved in discussions toward a discrete multichannel sound delivery standard for HDTV that would parallel the emerging digital optical film technology. This would allow for five discrete channels of full bandwidth plus a sixth sub-bass channel. Thus, as a music engineer ponders a future mixing to picture, considerations must be given to expanding his skills beyond the limitations of a stereo pair.

A final note: The revenue generated by all this new activity in the broadcast sound arena is significant. Not only has it provided much needed employment in the recording industry, but it has spurred manufacturers to new heights. All this trouble just to produce sound that could come out of a 2-inch speaker! As for the last assistant who honestly believed that phrase, we'll be promoting him out of our shipping department around 2000.

RE/P

FIVE QUESTIONS:

BROADCAST PRODUCTION

By the RE/P staff

Answers to questions on limiting, usable bandwidth, stereo panning, videotape formats and tape prep.

Q: *Because of broadcasting's limited dynamic range, is it better to limit the entire mix or the individual tracks?*

A: To properly answer that, you need to look at why the broadcast industry uses limiting. Way back, when there was only AM radio, stations used to maximize coverage by pinning the meters. Because the AM signal is amplitude modulated, the louder the signal, the farther it would go. Because the FCC would (and is still supposed to, in theory) fine stations for exceeding 100% modulation, the chief engineer would install a limiter right after the console. This allowed the modulation, but no fines. Additional limiters were set up at the studio-to-transmitter link (STL) and at the transmitter.

Because FM (and TV) is frequency modulated instead of amplitude modulated, it is not necessary to use limiters to get coverage. The reason is to keep the signal sources more even.

Generally, when you're mixing, it's best to use a soft-knee curve limiter on each track. You'll get the dynamic range you want without the peaks, thus avoiding the broadcast limiters.

Q: *How much fidelity on my finished tape will end up in the public's living room?*

A: More than you might think. TV bandwidth specs are 50Hz to 15kHz, the same as FM radio. Compare this to AM's 500Hz to 8kHz. Remember the 11dB window suggested in "From Laquer to Layback," where the dialogue, music and effects should be placed. The key is to recognize these parameters and make them work for you, rather than against you. Properly recorded and mixed audio for broadcast should sound great.

Q: *How wide can you pan in stereo without getting into trouble?*

A: Generally, as wide as the individual situation would call for. Remember, you aren't mixing for records. Panning requirements probably wouldn't be as wide. (The obvious exception is music-related programs.)

However, be aware that you can get in trouble if a station is running its limiters "un-strapped" for stereo, or if two separate limiters are set independently. If this were the case, then extremely wide panning could sound pretty strange. This is not a common occurrence, but keep it in mind as you are mixing.

It's also best to check the panning, and the entire stereo mix, as it relates to mono. While prevalent, stereo reception is not available in all areas. Check for stereo/mono compatibility by pushing the mono button while you mix.

Q: *What is the most common videotape format that I'll deal with?*

A: For most TV stations, the most common format is 1-inch Type C, but ¾-inch U-matic is still common. Digital, of course, exists in the form of D2, but is still limited.

Q: *Is there a universal way to prep a tape? If not, what will be accepted by most broadcast outlets?*

A: Unfortunately, there is not. If you are going to err, err on the side of giving too much information. For radio stations, provide a 2-track mono or stereo 7½ips reel tape with audio slates and complete instructions on the label.

For television stations using Type C, there should be a visual slate with all pertinent information, such as the agency, the client length, spot number and air dates. Audio should be on tracks 1 and 2, with time code on track 3. Do the same for U-matic, except for the time code, which is on the address track. RE/P

Research assistance: Fred Jones Recording Services, Hollywood.

TIME DOMAIN

PRODUCTION TECHNIQUES

Making TEF measurements in the field can ease post-production by giving you the information needed to re-create the location audio

△By Barry McKinnon

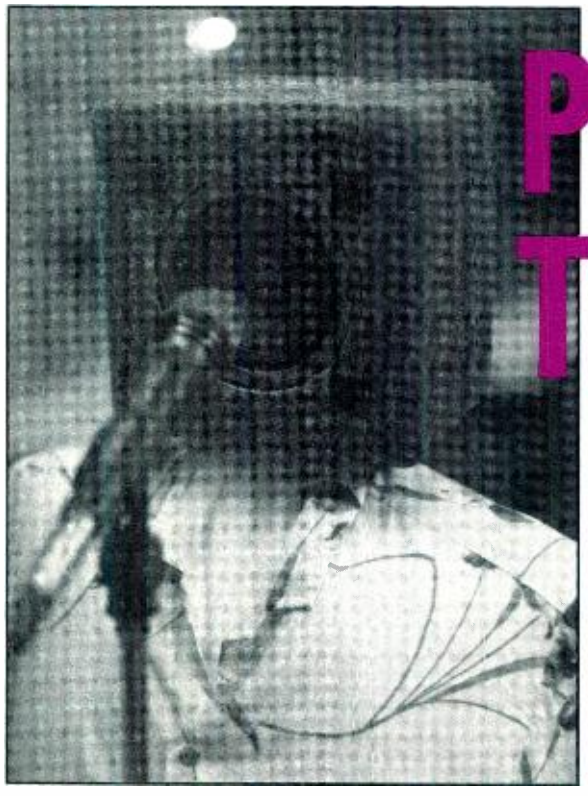
You're about to enter a dimension without frequency, a dimension of sound and perception, a dimension of critical aural impressions. You're about to enter the Time Domain.

Actually, the time domain is just another way of looking at the world of audio that we deal with every day. Thinking in the time domain is not as difficult as it sounds. Much like the calculus you use when you walk toward a curb and have to decide when to lift your foot, you are already working in the time domain when you decide on a microphone because of its dry pickup character. Clarifying and quantifying the time domain by presenting some measurement methods will help you speed up and simplify such difficult tasks as ADR.

The first step is to look at what the time domain shows us, and how to interpret it. Figure 1 shows a talker and a listener in a room. The shortest, most direct path between the two is the direct sound, labeled A. Reflections from nearby surfaces, such as the table top and objects on the table, add together to form early arrivals, labeled B. Reflections from major surfaces, such as walls, floors and ceilings, have longer paths to travel but eventually will end up at the listener as well. These are labeled C. Reflections that bounce off a number of surfaces before arriving at the listener's position are labeled D.

These sounds completely characterize the space in which the listener and the talker are found. If the listener were replaced by a microphone, all of these sounds arriving at the microphone contribute to the character of the pickup. If the room is large or small, live

Barry McKinnon is sales manager at BSE Production Equipment and Services Ltd., Calgary, Alberta.



"Mr. Speakerhead," a test dummy with a speaker attached to the trunk, is used to make time domain measurements.

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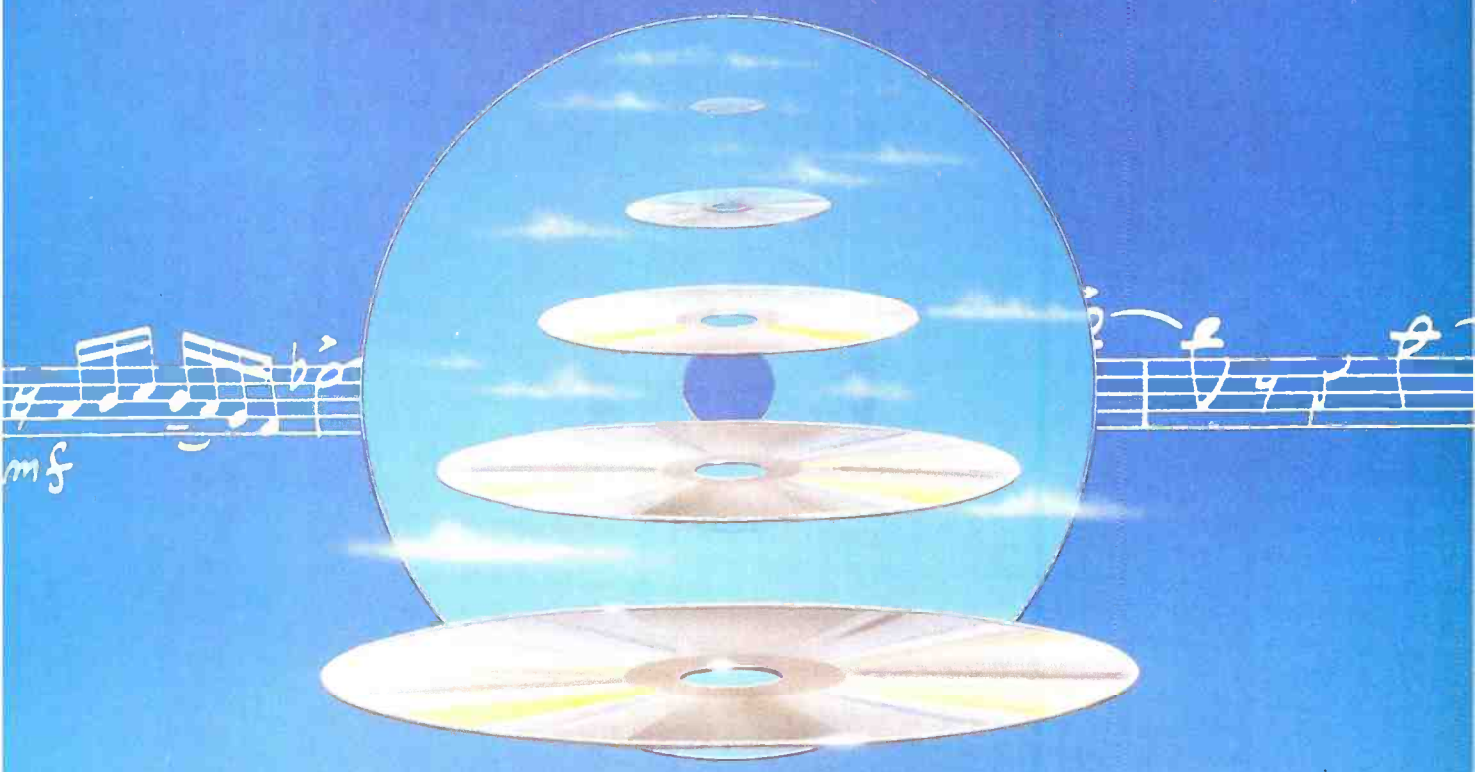

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or dead, and if the talker and listener are close together or far apart, the character of the pickup is changed.

If we were to graph these sounds in the time domain, as opposed to a frequency domain graph we are used to associating with microphones, we would see a graph like Figure 2. In this graph the horizontal axis shows increasing time; the vertical axis shows the energy of the arrival. Note that there is no indication of frequency content, which is a facet of the signal we become blind to when we rotate ourselves into the time domain. In Figure 2, we see the same sound arrivals we had in Figure 1. Again, A though D represent direct sound and early, late and diffuse arrivals, respectively.

Figure 2 is known as an Energy Time Curve (ETC) and can be generated with measurement devices currently on the market, such as the Techron TEF12 Plus and the DRA Laboratories' Melissa system. The ETC is a valuable tool for sound system work and studio control room tuning. It can also be a handy and informative method for comparing microphone techniques and investigating their time domain response. These can then be duplicated using electronic reverbs and time delays for ADR and radio production work.

A TEF analyzer was used to demonstrate the effects of various commonly used mic techniques. The TEF sweeps were sent through a 5-inch speaker mounted on a test dummy, and picked up using several different types of microphones.

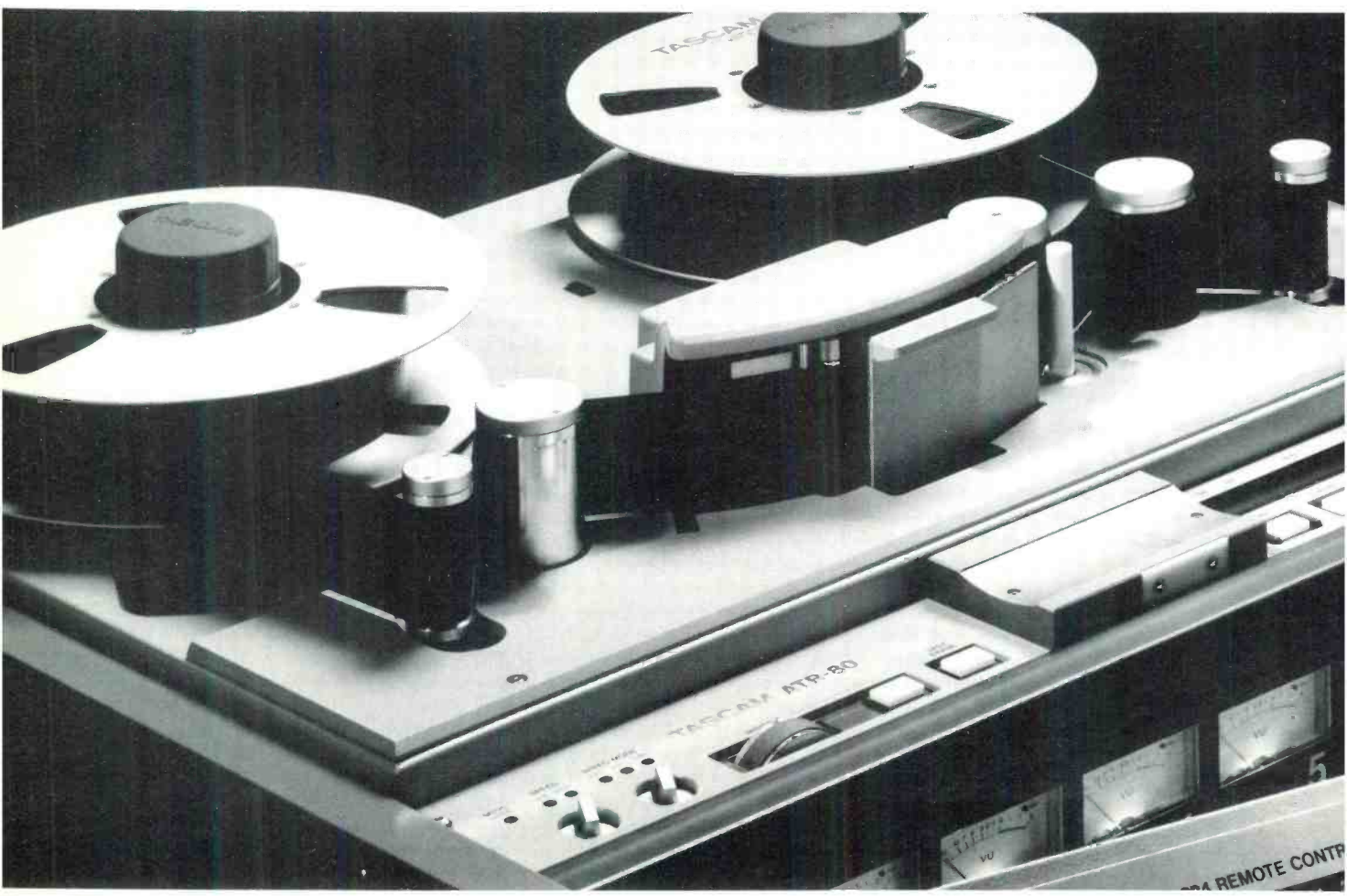
These tests were done with the test dummy in the same position in the acoustical environment. The differences are all a function of the microphone technique. The swept frequency range is 150Hz to 5kHz, providing a good time range of 83ms, or 93 feet. (See the sidebar "Making the Measurements" for more information.)

Even though the display is frequency blind, and we cannot determine the frequency content of the individual reflections, by sweeping through the voice range we can be reasonably sure we are getting relevant data. These reflections are likely to contain some information in the audible voice range and will affect the character of the pickup.

First, look at Figures 3-6. Compare the appearance of each ETC looking at the features we have discussed. Using your own experience with microphone technique, try to decide how the ETC relates to the character of the pickup. Which is the most live? The driest? The most distant? The most intimate?

Figure 3 shows the ETC of the PZM on the floor. Note the direct arrival shown with the cursor crosshairs. The distance between the talker and the microphone is indicated in the upper left margin of the display. The direct sound arrival spike is quite narrow, showing the trademark pickup character of a PZM. Those early reflections from surfaces a few inches from the diaphragm are greatly reduced because the diaphragm is only fractions of an inch from a surface. There is a small hump of arrivals on the next vertical graticule; these are reflections from nearby surfaces close to the floor. The PZM has two ITD gaps, one before early arrivals and one before late arrivals. Early arrivals tend to be late with a PZM, as the microphone sits on a surface that would cause them, eliminating many in the process. This leaves only the longer path reflections with which to be concerned. There are a few significant late arrivals standing out of a fairly high diffuse field.

Figure 4 is a short shotgun located just past 30 inches above the talker. There is a good sharp direct arrival, with a minimum of early reflections, because the microphone and source are both suspended



above ground. The few early arrivals are likely reflections from the body of the test dummy. The single significant late arrival is isolated by a clearly defined ITD gap, and it is a reflection from the floor. The pickup of diffuse reflections is reduced by the microphone's location and its very directional pickup characteristic.

Figure 5 is an omni lavalier on the test dummy's tie. The direct arrival is higher in level and quite clean. The early arrivals are low in level and spread over a longer time. The ITD is not as sharp as the late arrivals. The dummy's body blocks some arrivals and helps reduce diffuse reflection pickup.

Figure 6 is the cardioid lavalier. The direct arrival and the early arrivals are similar to the omni, but the ITD and the late arrivals look quite different. Also, the diffuse pickup is greatly reduced. It's no surprise that this microphone has the driest and most direct pickup character for recording the talker.

OK, so we have some interesting pictures, and they proved what we felt subjectively. Now what? If you are doing dialogue replacement or a radio drama, and you want to create or re-create an apparent space, it helps to know what microphone technique or what reverb and delay settings to use. The ear/brain processor is quite a smart system. People can quickly detect when an audio perspective does not jive with the visual image, such as in ADR or Foley work. In a radio drama or recorded sound effects for live theater, the ear/brain system has to provide the setting for the action. The right attention to detail can mean a more realistic experience for the audience.

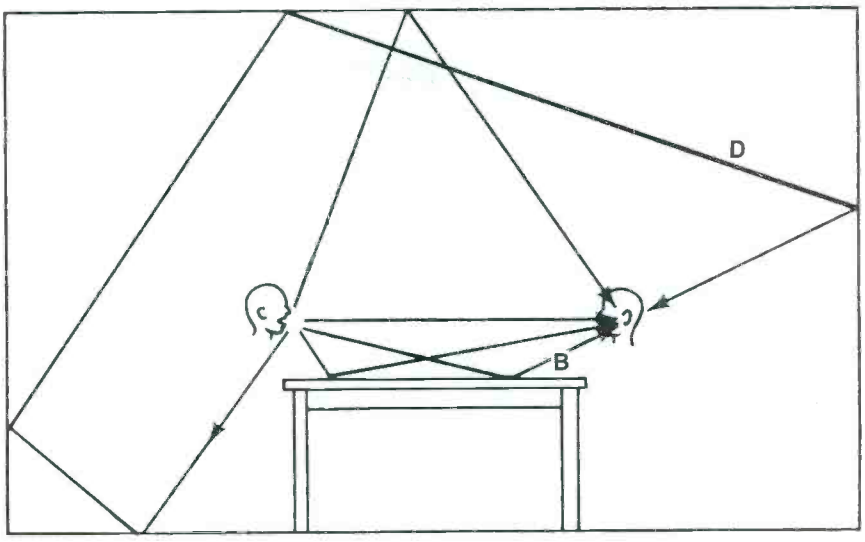


Figure 1. *The relationship between a speaker and a listener in a room. A: direct sound; B: early arrivals; C: late arrivals; D: diffuse arrivals.*

With this in mind, let's consider a few aspects of the ETCs and how the combined factors relate to intimacy, point of view and apparent distance. A characteris-

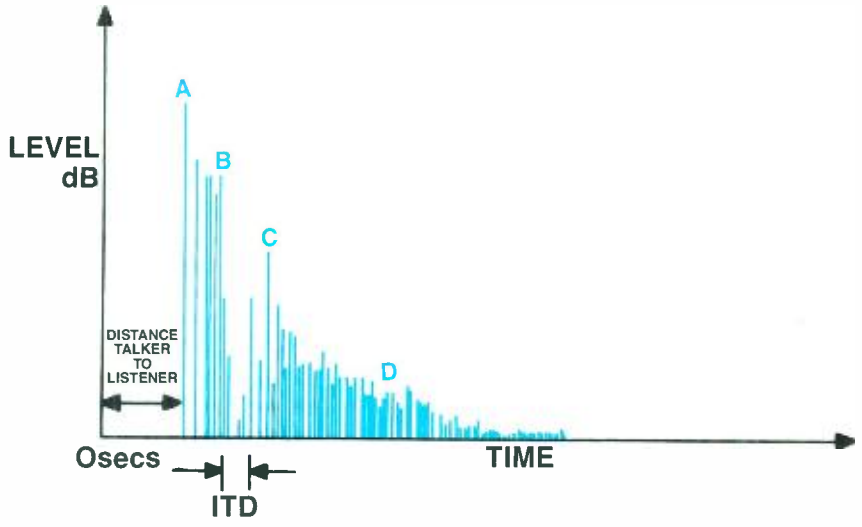


Figure 2. The Energy Time Curve (ETC) of Figure 1, showing the time domain.

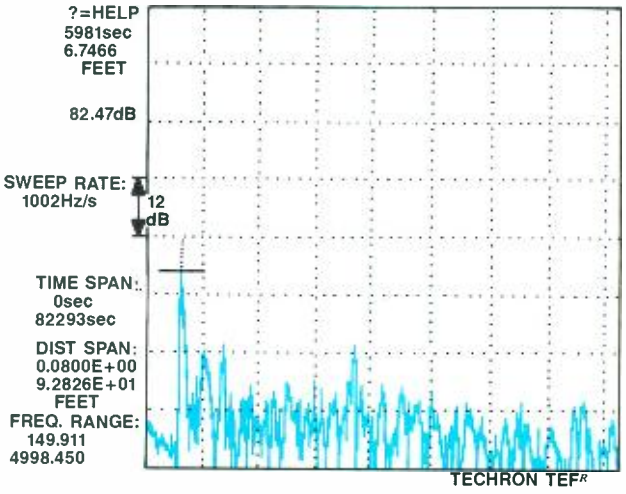


Figure 3. The ETC of a medium-sized live space, using a Crown PZM microphone.

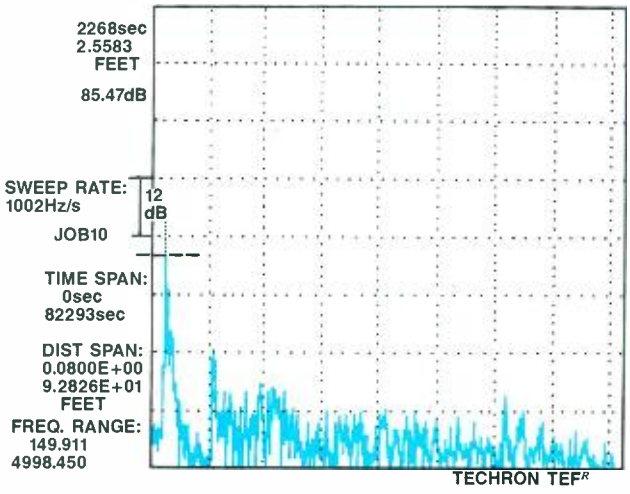


Figure 4. The ETC of a medium-sized live space, using a short shotgun microphone.

tic of the time domain display is a good visual indication of direct-to-reverberant ratio. This is helpful in sound system work, but it also has application in our area of interest.

One of the factors that helps us determine the apparent distance of a talker is the direct-to-reverberant ratio. This would be the apparent intimacy of the pickup. If you look at Figures 3-6, you can see that the direct arrival and the later arrivals have various relationships to each other. The PZM has a higher level of late and diffuse arrivals than the cardioid lavalier. Experience will tell you that a more intimate pickup is had by close miking. The ETC displays in these figures show the relationships you have been hearing in the time domain. Note that there are two ways to improve the direct-to-reverberant pickup ratio: You can move the microphone closer or use a more directional microphone.

The shotgun and omni lavalier have similar looking ETC displays, but note that the distribution of the arrivals is different. That is important, which is why a distant microphone technique will always sound less dry than tight miking, unless you were recording in an anechoic chamber. The distribution of arrivals, both in time and intensity, is decoded by the ear/brain system as one of the indicators of apparent distance.

In ADR and production application, engineers will often try to use a reverb to simulate certain spaces and will find that no matter what setting of reverb is chosen, the effect is not quite right. In Figures 3-6, the diffuse field has various relations to the early arrivals. The acoustical space was the same for all four measurements, yet reverb settings on a single effects device would not duplicate what you see.

Often, two devices will be required. One will deal with shaping the density and decay of early arrivals. The other will deal with overall acoustical space, which will not change with microphone technique (that's what real reverb is all about). Sound falls off in level at 6dB per doubling of distance in an open space. In a space that has a reverberant or diffuse field, such as a gymnasium, the direct sound falls off at 6dB per doubling of distance until you reach a critical distance where the diffuse sound field is the same level as the direct sound. Past this point, the sound field remains fairly uniform. If you are in a gymnasium, in the dark, you could tell the distance between a talker and yourself quite reasonably until the talker gets beyond the critical distance. Unless he approaches more reflective surfaces, you will have a hard time determining distance and direction of the sound.

The same localization methods hold

true for a small, live room. You can quite easily tell the apparent distance to a talker by constantly comparing the direct sound to the early arrivals and noting the ITD to the late arrivals. This is some pretty high-powered math, but, like the seat-of-the-pants calculus you do everyday, it is quite painless. That is, until you try to re-create that space without benefit of being there. The process can require tedious work and painstaking effort to duplicate a space for ADR work or create a space for radio production.

Here are a couple of examples. In Figure 7, we have an ETC of a talker and listener far apart in a small space, across an office. We have a distinct direct arrival, with a separation between the direct and early arrivals because of the distance between the talker and listener. The ITD is almost non-existent because the distance between the talker and source is comparable to the distance to the major reflective surfaces. The late arrivals begin sooner and are high in level compared to the direct sound and the diffuse sound.

In Figure 8, the talker and listener are a couple of feet apart in a large, live space. We see a high direct sound, with a discernible ITD, and the first late arrivals are very low in level compared to the direct arrival. There are late arrivals intermixed in the diffuse field because there are large surfaces far away from the talker and listener. Note that the reverb time of the space is not a factor in the apparent intimacy in this case. Whether the RT_{60} was long or short, the ratio between the direct and reverberant sound remains high. Notice the relationship that is developing between the levels and location in time of the direct sound, early arrivals, ITD, late arrivals and diffuse sound, for the various microphone techniques and situations.

Keep in mind that the time domain is dynamic in the case of moving actors. The ratio of direct-to-reflected sound will change as the actors move in relation to large reflective surfaces and to each other.

As a thought experiment, consider an actor moving toward a mirror as he talks. The reflection path for the sound from his mouth to the mirror will continually shrink. The time for that reflection to travel to and from the mirror will diminish, and the level of the reflection will increase. Consider what you already know about reflected sound and cancellation in microphone technique (this is where PZMs developed their reputation). If you have to drop a word into the dialogue that was picked up by a lavalier on that actor, you can duplicate the equalization necessary with a delay line, the same way it happened. Note, too, that you can't fix this kind of notch filtering with equalization. That sound was canceled at the micro-

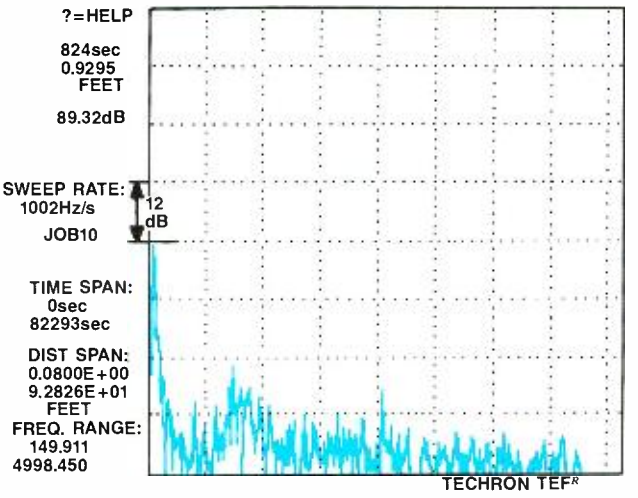


Figure 5. *The ETC of a medium-sized live space, using a omni lavalier microphone.*

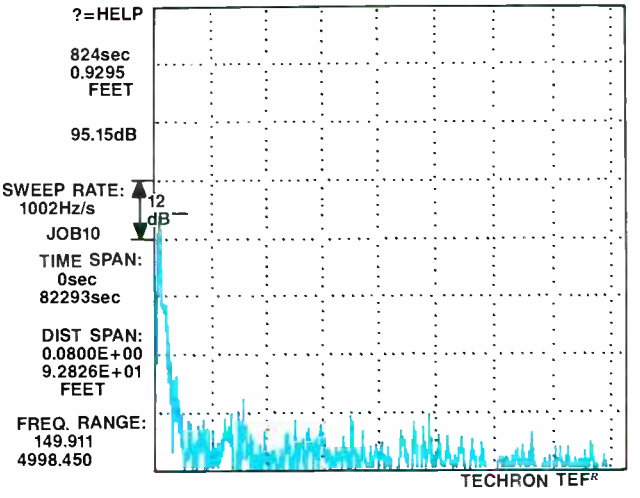


Figure 6. *The ETC of a medium-sized live space, using a cardioid lavalier microphone.*

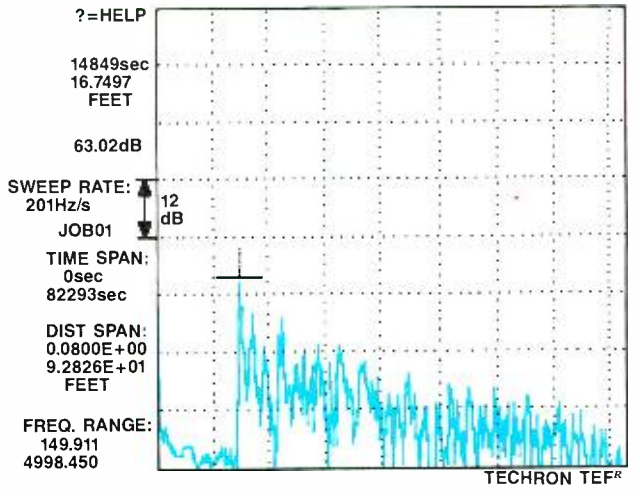


Figure 7. *The ETC of a talker and listener far apart in a small space.*

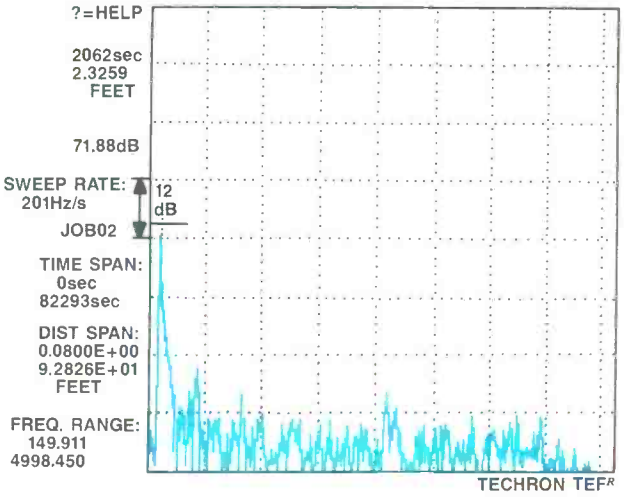


Figure 8. The ETC of a talker and a listener close together in a large space.

Making the Measurements

Some of you may have access to a TEF or Melissa analyzer and may be interested in making this type of measurement yourselves. The measurements are unobtrusive, and you can make them without delaying the entire production.

A test dummy provides an effective test source for this type of measurement, having the same "coverage pattern" as the human voice. In TV or film production, it would be reasonable to dress the dummy in similar costume to the actor and apply the same microphone technique the actor will be using. TEF tests are fairly immune to noise, so the test sweeps can be done while lighting is being placed or when the actors are in makeup. (The sweeps in the examples in the main article are only five seconds long.)

ETC sweeps can be done with the dummy in the appropriate locations on the set, and stored on a disc that follows the raw audio tapes to the post house. The ETC sweeps that I have used here cover the frequency range of 150Hz to 5kHz. This time range, however, is too short from which to draw accurate reverb time measurements. If that information were required, you would need to narrow the swept range to be 1kHz to 2.5kHz, or a number of ETC sweeps centered on octave bands, if you wanted to get specific. It takes a few moments to reset the parameters and get a measurement long enough to calculate the actual room decay. The resulting frequency response of the particular microphone technique can be measured while the dummy is in the location as well. The time domain and frequency domain measurements would make ADR a snap.

If ADR is needed, the applicable ETC curves can be pulled and viewed. Electronic reverbs and delays can be set to match (in fact, they can be measured with the same test device to test the accuracy of the settings), and then the appropriate ADR or Foley work can be added. The cost of this type of measurement device has dropped drastically since its introduction, and the idea of a production house or post house owning one is not farfetched. The ability to quickly and accurately do ADR work can actually be quite a cost savings when production schedules are tight.

phone diaphragm, so there is nothing left to equalize. As the actor moves between two feet and six inches from a reflective surface or from another actor with a microphone mixed to the same audio track, the notch filters will sweep through the voice range. A fixed notch filter would be easier to live with, but the dynamic character of the filter makes it stand out.

This dynamic character of the time domain becomes a factor in a multitude of situations. Another thought experiment might involve a hospital corridor, a common setting for television and film. The corridor has a 10-foot ceiling and is 12 feet wide. Two actors are speaking as they walk down the corridor. What would you expect the ETC to look like, assuming that one actor is speaking and the other is listening? There will be a high direct sound level, because they are likely only two or three feet apart.

You have been in a hard corridor like this sometime in your life, so you know that the flutter echo is quite noticeable. That will show up on the ETC as periodic spikes, quite high in level. If the listener and talker are three feet apart in the center of the corridor, the first two spikes will be at 9ms and 15ms (one from each side of the corridor). There may be another one at 10ms, if the floor and ceiling are hard as well. Because of the hard surfaces, these will repeat with minimal decay at those intervals. If everything has a hard surface, the shoes will make that characteristic "steps in a rain barrel" sound, and the voices will have some trace of that as well.

One hundred percent accuracy is not always desirable. Some very reverberant spaces have a critical distance of 10 feet to 15 feet. If the dialogue took place with the actors that far apart, you would not be able to understand what was said. If you were doing ADR for that situation, you would have to artificially "dry up" the room, improving the direct-to-reverberant ratio. This is actually equivalent to making the voice more directional, and this is why you find very directional speakers in reverberant spaces. You can still tailor the early arrivals to be correct for the point of view chosen in the scene and just "turn down" the diffuse or reverberant field.

These are general cases designed to give you a little insight into how to think in the time domain when you need to do some digital wizardry in production or post-production. By using the apparent or actual background in which the action or dialogue happens, or by laying out a floor plan of the fictitious space, you can create or re-create an accurate and believable sound effect or bit of dialogue using electronic reverbs and delays. RE/P

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In Part 1 of this article, we discussed several techniques and approaches designed to optimize system performance more properly. Many of these involved measuring the system in pragmatic, organized ways in an effort to learn more about its hard performance characteristics, both good and bad. These measurements explained many of the effects heard but not easily understood. We also took a look at how some of the problems revealed could be corrected by simple and obvious means.

In Part 2, we finish measuring the system, correcting for any final problems.

Some of the things that our tests reveal, such as polarity, signal level mismatch, hum or noise, and lack of headroom are so specific that there can be only one or, at most, a few possible corrections. However, one exception is the relative polarity of a loudspeaker system's component sections above and below its crossover frequency.

G.R. (Bob) Thurmond is principal consultant for G.R. Thurmond & Associates, Austin, TX.

SOUND SYSTEM PERFORMANCE OPTIMIZATION

Or, how to EQ without using the EQ.

By G.R. (Bob) Thurmond

PART 2

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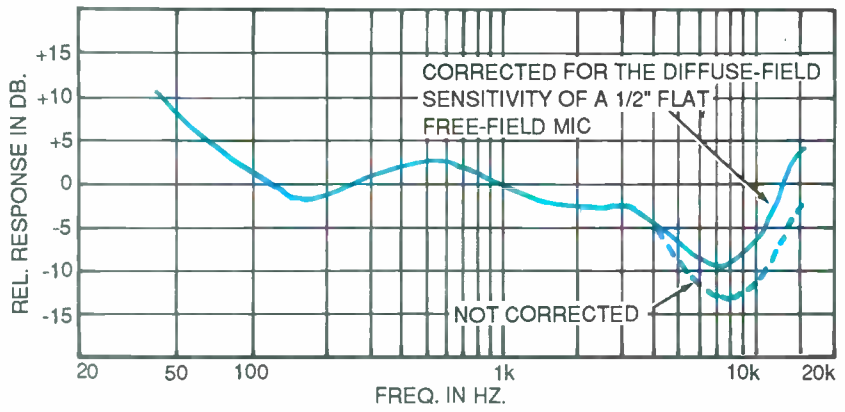

Figure 1. *Desired measured response of a sound reinforcement system (after Schulein). (Courtesy of Schulein/J.AES; copyright 1975, J.AES.)*

Often, the correct arrangement depends on the type of crossover used and its setup. In many speaker systems, the unavoidable physical offset between low- and high-frequency units introduces an additional alignment problem. This effect may vary with the listening angle off of the system's center axis, creating a real cluster of worms.

Keep in mind, however, that the real problem with polarity or phase/time offset errors is in the *response irregularities* they create. If the polarity situation is unclear, probably the best approach is to measure the system frequency response carefully through the crossover frequency. This usually requires a 1/6-octave continuous-sweep acoustic measurement; generally, a 1/3-octave real-time device will be insufficient.

If such a measurement reveals a sharp dip at the crossover frequency, then the relative polarity of the low- and high-frequency units should be reversed, and the measurement repeated. In fact, such a dip is often hard to find. It varies with location and does not change noticeably when the polarity is reversed! Before this leads to fists full of hair, you should realize this reaction in a complex situation.

The polarity or ultimate phase will never be exactly right or wrong.

The polarity or ultimate phase will never be exactly right or exactly wrong, and any residual problems will be confined to a narrow frequency region around crossover. There probably are much bigger fires to fight. Live with the best you can get.

The next important consideration is the low-frequency response. Does it roll off a lot? Some rolloff is difficult to avoid; loudspeakers couple to the air less efficiently as frequency decreases. (The size of the diaphragm becomes smaller than the wavelengths it is trying to reproduce.) A truly flat response requires either a very large cone/baffle area for good low-frequency efficiency via coupling (more in a moment), or a deliberately suppressed efficiency at higher frequencies.

Because high efficiency at all frequencies usually is necessary in sound reinforcement, and because large woofers or subwoofers are expensive, we usually accept some bass rolloff in the loudspeakers and try to compensate for it by equalization.

Be aware that other thieves may be stealing your thunder. At very low frequencies, a speaker is almost omnidirectional, radiating sound nearly equally in all directions. Above a few hundred hertz (depending on the size of the unit) it becomes more directional, radiating most of its energy forward. Thus, more extreme low-frequency energy is lost to the sides and rear of the unit. This loss can be considerably reduced by mounting the speaker systems side by side, coupling the front baffles together. If this baffle area blends into a room boundary, so much the better.

Don't forget that the response of the system microphone probably also rolls off substantially at lower frequencies. Dynamic directional types, in particular, typically roll off rather strongly below about 200Hz. Because our system response measurements included the system microphone, its characteristics could be tricking us into thinking that the loudspeakers are rolling off more than they really are. Even worse, we could be tricked into overcompensating for this effect.

If the actual response of the microphone is not known, it should be measured. This can be easily done by measuring the system response twice in the same location, once with the system microphone and again with a microphone with a known flat response. The difference between the two curves is the response of the system microphone.

Off the wall

A hard surface, such as a wall, near a loudspeaker will reflect sound back to that source. This reflected sound will be in phase alignment with the source at some frequencies, and out of phase at others, canceling part of the energy and causing an uneven response. Allison and others have described this effect, but many sound professionals appear to forget about it at times. Absorptive material applied to the surface is usually not effective enough. The best solution is always the proper aiming of well-chosen devices (for example, one with the right directivity index/polar pattern).

A hard surface near a loudspeaker can also reflect sound in unwanted directions. Occasionally this results in some rather bizarre horn coverage patterns. Here, at least, absorption on the offending surface is often effective. By the way, don't forget that reflecting surfaces have exactly the same effects on microphones as they do on loudspeakers. Such undesirable effects are common and the solutions are the same as those for loudspeakers. However, their implementation is often more difficult or objectionable.

A microphone pointed at a snare drum head might pick up more crash cymbal reflection off the skin than you might think. The same is true of guitar amps leaking into a mic aimed at a kick drum, or feedback off of a singer's glasses.

Other problems, such as obstructions in front of the high-frequency units or misaligned drivers, have solutions that are pretty obvious but are not always possible. In practice, you do the best you can with the circumstances of a particular job, understand what it is you cannot correct, and resolve to do better next time. Sound familiar?

Puzzle pieces

Now that the loudspeakers are arranged as well as they can be, you are ready to see what they can really do. As discussed in last month's installment, you measure the coverage of each loudspeaker and sketch the coverage pattern onto a drawing of the audience area, showing at least the point of highest level and the 3dB contour. Then, check to see whether the various patterns fit neatly together to cover

all of the audience area, with minimal spillover. If they really do, let me know what design technique you used. They usually don't! The results are typically somewhere between not quite right and not even close.

Assuming that this system is in a large venue, and proceeding with the safe assumption that adjustments are needed, start with the unit(s) covering the most distant part of the audience. This is usually the most difficult area. If these are aimed at the rear of the audience area, and the level there is more than 10dB down from that at the hottest location, it is time for sweat to pop out on the brow. Either the vertical dispersion of that unit is too broad, or it is positioned too low. Either way, you have a real problem. What design did you say you used for choosing the components in the system?

It may be that the only solution is to change the speakers or devices to those with narrower dispersion and tighter pattern control, or to locate them higher. (Of course, you could also add fill loudspeakers on delays, but could this really be called just a performance correction?) If this is a low-frequency unit, and the overall dropoff to the rear is not too bad, you can probably just hold your nose (ears?) and live with it. In fact, it may be difficult to do much better. If this is a high-frequency unit, however, you have some explaining to do.

All together now

By readjusting mechanical orientations and remeasuring, individual coverages can be optimized. Introduce and adjust individual additions in sequence and remeasure until you can turn everything on at once and recheck the overall coverage. It will never be perfectly uniform, of course, especially in the off-axis overlaps between individual areas, but the nature of the variations can tell you something.

If the variations are gradual, then the individual areas simply don't fit together quite right (they never do). If they are sharp and repetitive, lobing or "swishing" across the pattern area, then the drivers are probably not aligned properly with each other, either physically or in electrical polarity. But you caught that in the inspection, didn't you?

Often, the correct alignment is just not physically possible. In such a case, if the resulting aberrations are unacceptable, a delay unit (or crossover with a delay internal) may be able to affect the correct acoustic alignment. Of course, we anticipated this possibility and included such a unit in the system, didn't we?

But what is unacceptable? If the error is between units covering different frequency ranges, then the effects will be confined to a narrow range around the crossover frequency. In a complex system with many high-power devices covering the same frequency bands and pattern area, such effects may be practically inaudible and unmeasurable. Great concern here may be hard to justify. In other cases the effects may be highly obnoxious, and correction imperative. If such is the case, then now is probably the best time to carry it out by using measurement techniques suggested earlier to check the results.

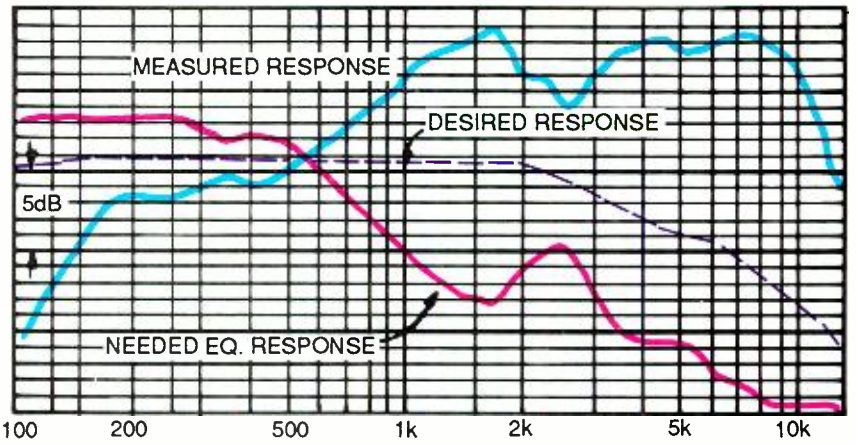

Figure 2. The desired response for a system equalizer, as derived from the measured system response and the desired system response.

The unequalizer

So far, we haven't even talked about an equalizer. Have you been able to stand the suspense, or have you sneaked a look at this final section? If you peeked, you're not alone. Many audio professionals would like to ignore what we've been talking about. They are afraid it might convince them that they haven't been doing an adequate job. But that's always the difficulty with progress — standing still means falling behind.

In fact, we have already done quite a bit of response equalization, without an equalizer. We have boosted the low frequencies by improving the loudspeaker configuration and coupling. We have smoothed the midrange by matching levels and polarities through crossover frequencies. We have smoothed the treble by improving coverage uniformity. And we have boosted the highs by aligning high-frequency drivers.

But exactly what are we trying to achieve through equalization? Perfectly natural sound? Highest intelligibility? House contour? Flat frequency response? If we want flat frequency response, then do we want flat response from the direct energy, flat response from reverberant energy or flat response from the total energy in the room? If so, at what ratio, which means measured from where? With what program material? How many samples? Accumulated? Averaged? In what time domain?

Created equal

Many approaches have been tried, but the one most widely accepted is to make the reinforced sound appear to be an amplified version of the original sound. In other words, natural sound is the primary goal. In many real situations, speech is difficult to understand, even with highly natural, high-quality reinforcement, so a compromise between natural sound and intelligibility is necessary.

But how do we determine this performance through measurements? The naive approach is to measure the system response by some RTA means, then equalize it flat. However, a bit of listening quickly reveals that the resulting sound is much too bright and that the highs must be rolled off to obtain a natural balance.

Sooner or later, most practitioners speculate that it is more of the direct sound we are really understanding (if not hearing, thanks to our psychoacoustic onboard brain/computer), rather than the reverberant energy, which is what a microphone picks up. Therefore, if we equalize for flat direct sound, the system will sound right.

Hear this

Most of us can attest to this equalization approach. Sometimes it works well. Sometimes it works with disappointing results. Clearly, this is not the all-time correct approach either, but what is?

This matter was studied thoughtfully by Schulein, who published the results in 1975. This paper, which should be required reading for anyone who attempts to do equalization, explains a great deal. It establishes that the way we hear reverberant sound coming from all around, as

compared to the way we hear direct sound coming from straight in front of us, is the real explanation.

Our hearing is much more sensitive to high frequencies in a reverberant field, possibly because more of the energy enters straight into the ear canal. This difference accounts for the way we actually hear reinforced sound.

This means we only need to measure this reverberant field with an ordinary microphone, then equalize the system to account for our hearing characteristics, and it will sound right. The suggested proper measured response, as determined by Schulein, is shown in Figure 1.

Add and subtract

Now we know the actual response of the system and the response we want it to have. The difference between these two is the response needed to be added by the system equalizer. If we have the time in a permanent or longer installed system, plotting the original and desired responses on the same graph makes it easy to see the difference, or EQ setting. Just don't get mixed up and plot what would be needed to correct the desired response to the measured response! This mistake is easy to make. Figure 2 shows an example of these plots. Note that the difference between the measured system response and the desired response is the same as the difference between a flat response and the desired equalizer response.

With the desired equalizer response plotted on the graph where you can also plot the actual equalizer response, all you have to do is adjust the equalizer setting until its response matches the desired curve. This should be easy but often isn't. Sometimes the filters simply aren't tuned correctly, and sometimes they are too broad or have the wrong shape.

More likely, several adjacent filters interact in surprising ways. Several mild individual adjustments can add up to a major collective one, or they can all but cancel out each other. Sometimes plotting the equalizer electrical output quickly shows the limitations of real-world equalizers.

That is exactly the value of this technique — you can see much better what is actually happening. If you try to make such settings using only a real-time analyzer, most of the characteristics of both the system and the equalizer will be lost in the analyzer limitations. Trying to set an equalizer by watching what happens to the system response at a single location is even worse.

First, there is no such thing as an "average" location. Each location is different, and none will match the overall average. Second, this means that the readings for whatever location(s) you pick will have to be corrected to the system average, which introduces another chance for inaccuracies.

Third, equalizer corrections may not affect the response in your chosen location the same as elsewhere (really!), so that, fourth, you will have to recheck in several locations anyway, so why bother with this one? The method of matching the responses on a graph works much better.

Close to the line

This response-matching technique may work a little too well, in fact, because it may tell you that your equalizer cannot achieve the response you need for a particular system. This is especially common with equalizers that use fixed-tuned filters, such as most graphic types. The most flexible equalizers use parametric filters, but this is a double-edged sword. The extreme flexibility allows more opportunity to achieve the desired response, but it also is prone to make mistakes. Parametrics demand more skill and more time, but they can often do a better job.

There are a number of good parametric equalizers on the market, but most are generic multipurpose devices rather than specific designs for the noise, grounding, RF-shielding and gain staging/headroom requirements of sound reinforcement systems. After you have dealt with a number of systems, you begin to see which characteristics are particularly useful. We might decide on something like the following:

First, because there is often a severe low-frequency rolloff to be corrected (and concern for the frequency-dependent excursion limitations inherent in low-frequency

Controlling Feedback

If an important function of the design is to control feedback, you must consider the hardware needed to do so. Almost invariably, the most effective and least objectionable device for this purpose is a notch filter, one that significantly reduces the system gain at the feedback frequency and has negligible effect elsewhere. Ah, there's the rub.

Any actual filter will have a finite bandwidth, attenuating more than just one exact frequency. In fact, a certain bandwidth is needed in feedback suppression, because feedback frequencies shift with such variables as microphone placement and air temperature, the latter at a rate of about 0.1% of frequency for each degree (Fahrenheit) of temperature change.

Thus, if we want a notch filter to be effective over a temperature range of 10F, a reasonable range, the filter must have a bandwidth of 1%. This would be as measured at, say, 6dB down to provide good feedback suppression within the 1% bandwidth. That would force the more familiar −3dB bandwidth to be at least 2%. If we allow a little extra for component tolerances and tuning error, a practical bandwidth might be 4% or 5%. This is less than 1/12 octave, which is one semitone.

A true ⅓-octave filter has a bandwidth of 23%, which, if used as a band-stop ("notch") filter, is the bandwidth 3dB up from the bottom. The 3dB down bandwidth is much wider, which is one reason such filters are not suitable for feedback suppression. Clearly, they suppress much more than just the feedback. Furthermore, most such filters are not tunable, so they may miss the target spot.

There is another little-known problem with many filters. Not only will the feedback frequency change with temperature, but so will the notch filter frequency unless we prevent it. Most capacitors used in these filters have a temperature coefficient, which might cause the filter to drift off of the feedback frequency with the temperature changes in the equipment rack. It doesn't hurt to let the rack "warm up" before the final settings.

And doesn't such a sharp filter ring? Yes, but barely noticeable when cutting, and nowhere near as much as the feedback it suppresses. In fact, a good set of properly tuned notch filters will make a system sound noticeably cleaner and less reverberant.

Feedback suppression, like all aspects of performance optimization, is a bit more complex and demanding than we had thought. No wonder casual efforts in this area, as in other areas, often produce disappointing results. Did someone say that nothing worthwhile is ever easy?

devices), we need a protection high-pass filter that can be tuned from about 20Hz to 100Hz and a section that can boost up to at least 15dB, adjustable from fairly narrow to very broad.

Second, because there are often similar deficiencies at high frequencies, we need a similar filter that can be tuned from about 2kHz to 20kHz. Third, because there often are other strong irregularities in the low- or high-frequency region, another filter is needed for each range. However, these need not reach all the way to the frequency extremes and should reach more into the midrange. They will not need so much boost or cut capability, yet they will be somewhat narrower.

Fourth, to deal with the smaller irregularities at various frequencies, we will need probably four other filters that cover most of the audio frequencies in broadly overlapping ranges, have a modest boost or cut capability, and have a moderate to very narrow bandwidth capability. Also, these four filters should be capable of producing a very narrow notch, if desired.

The real thing

Whichever brand we use, after it has been set, it must be inserted into the system and the system response measured again. Almost without exception, the corrected response will not be quite what we intended, no matter how carefully we did everything. The necessary final adjustments must be carried out and a final response measurement run.

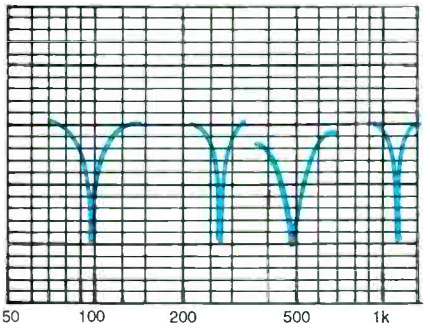

Figure 3. Measured responses of four different notch filters. Left to right, a commercial unit for feedback suppression, a commercial parametric set at minimum bandwidth, a proprietary parametric set at minimum bandwidth, a proprietary filter for feedback suppression.

Into the sunset

So now, after all of our tedious, but all-important, measurement and tweaking activities are done, we can stand back and listen to the fully optimized system. We know that the results were worth the effort. Even though the process is involved, we now have a useful step-by-step technique that can make even the simplest systems sound much better. We know that by addressing and correcting one thing at a time, no matter how simple or obvious, we can make any system work as well as it possibly can.

As we all know, performance is only partly the pile of equipment used. Proper setup and operation are the other half.

References:

R. Allison, "The Influence of Room Boundaries on Loudspeaker Power Output," J. Audio Eng. Soc., Vol. 22, No. 5, pg. 314 (1974).

R. Schulein, "In Situ Measurement and Equalization of Sound Reproduction System," J. Audio Eng. Soc., Vol. 23, No. 3, pg. 178 (1975).

RE/P

LIVE & DIRECT

By David Scheirman

1980s: Rock Sound Matures

Perhaps one of the most obvious phenomena in our culture during the 1980s is the greater impact that the rock entertainment industry made on society.

Forty-year-olds play electric guitars on stage for thousands of fans, many young enough to be their children. There is a 24-hour TV station dedicated to rock music. Corporations use rock to sell everything from shoes to pickup trucks. Even department stores are more likely to be playing Prince over an extended-bandwidth foreground system, than offering Lawrence Welk over the elevator speakers.

There are many identifiable trends during the past decade that can give us clues about today's changing sound industry climate. Being aware of musical trends can give sound system designers and sound company owners a better knowledge of what may be coming next. Often, musical styles or patterns that start with one person or group can ultimately affect the entire entertainment industry.

Remember what happened 20 years ago when noted drummer Hal Blaine pioneered a multi-tom style of playing that led to Ludwig's bringing out the Octa-Plus set (with eight separate toms)? That led to every concert sound company having to triple and quadruple its microphone inventory of Shure SM-57s and then Sennheiser 421s as live sound professionals looked for ways to present the new drumming style to live audiences.

Remember what happened to mixing consoles when groups like Yes and Genesis started bringing multi-keyboard setups to the concert stage as new synthesizer products were developed? No matter how many channels the board had, it wasn't quite enough. Sound companies embarked on development projects for new custom consoles considered quite extravagant for their time, and audio equipment manufacturers like Yamaha, Midas and Soundcraft began to understand what the live sound market required in the way of new mixing equipment.

And remember what happened when groups like the Bee Gees and the Jacksons gave us disco-dance style music that stretched the limits of low bass-frequency reproduction ability in concert sound systems? "Saturday Night Fever," with its bass-heavy soundtrack, created new box-office records at the theaters. Subwoofer systems for sound-system rental companies were proliferating as fast as they could be developed and built.

These few examples show how popular musical styles can often be the creative force that brings about evolution in the concert sound industry. Once the system technology has changed for a few major artists to the extent that new creative avenues are opened, the entire sound industry benefits.

Let's examine some of the landmark events in rock during the past decade, and look at patterns that may affect the professional audio industry.

• On Aug. 1, 1980, Music Television opened a whole new forum for the presentation of rock music videos by previously unknown artists. Showcase club tours then developed. Rock video techniques began to be adopted by mainstream broadcasting networks, as the hard-hitting, creative graphic style was proven to reach a larger, younger audience.

• Led Zeppelin broke up in 1980 following the death of their drummer, John Bonham. One of the most popular hard-rock bands in the world, the style-setting group's breakup symbolized the splintering of rock's 1960-70 aristocracy as many major, established and successful acts decided to "never tour again." Rock fans mourned the demise of favorite groups they had grown up with and eagerly looked forward to reunion tours and the formation of "supergroups" featuring musicians from different known bands.

• The 1981 U.S. tour by the Rolling Stones saw the use of corporate sponsorship to do advance promotion and to underwrite production costs of a major tour, which were beginning to skyrocket.

• In the mid 1980s, artists like Michael Jackson (with "Thriller" becoming the best-selling album in history) and Madonna helped to pioneer the marketing of the entire personality. Multi-million album sales and a string of radio hits from a single album became the new sales goal.

• Movies like "The Big Chill" helped to promote the re-introduction of favorite tunes from artists like Creedence Clearwater Revival, the Doors and Sam & Dave. All of this revived the careers of everyone from Big Brother & The Holding Co., the Jefferson Airplane and the Doobie Brothers to Chicago and the Beach Boys. More oldies-package summer tours were one result of this trend.

• Mega-Events: Some promoters took huge risks to pull off rock festivals larger than the world had ever seen. The US festivals (1982-83) drew together many major artists on one bill and introduced many newer acts to huge crowds. Hundreds of thousands of people attended, while the world watched via satellite broadcast. The phenomenal success of the global marketing of mega-events like Farm Aid, Live Aid and the Amnesty International and Rainforest Benefit tours put the rock industry back on track with social consciousness.

The 1980s were a dynamic decade. We saw many changes in the world around us that affected the live sound industry. The results of these changes? There are more smaller, high-tech specialty rental and consulting firms than ever before. The major sound companies are now larger and have global reach. Sound systems are more versatile. Soundmixers and technicians are more creative, experienced and respected. The equipment to the industry is more specialized and sophisticated.

Realistically, though, the challenges to the sound reinforcement industry are now greater than they have ever been. The use of louder, more powerful sound systems means the prospect of closer examination and possible regulation by public health officials and noise control officers. The increasing complexity of major audio systems means that entry-level workers in the industry need better education and high skill levels.

We can expect to see more changes in the 1990s. Performing artists will continue to surprise us with their musical innovations and changing needs. Our own industry will respond with improved technologies. More markets will open up, as family entertainment packages and major special media events make use of large-scale sound reinforcement. Live music will continue to impact and transform different aspects of our society, and sound systems will be used to get the message across.

So, the marketplace will change. The equipment will change. But the skills that helped create today's sound industry will stay the same: listening to the music, figuring out how to interface existing audio gear with the performer's needs and then presenting that music to the listening public, all the while thinking about an even better way to do it. May your decade be fun, fulfilling and profitable. RE/P

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

STUDIO UPDATE

Northeast

Soundmirror Inc. (Boston) has upgraded to a Sony DAE 3000 equipped with interfaces for 1630, DASH and F1 editing interfaces. *76 Green St., Jamaica Plain, MA 02130; 627-522-1412.*

Crystal Sound Recording (New York) has named Elyse Klein studio manager. *220 W. 19th St., Sixth Floor, New York, NY 10011; 212-255-6745.*

Midwest

Studio A (Dearborn Heights, MI) has installed into its main room an SSL 6000R 40-input mixing console with G-series preamps and a Sony 25-inch XBR Pro monitor. Additions to the Synclavier/MIDI room include a Macintosh SE with Opcode Editor/Librarians, a KMX MIDI switcher/Studio 3 MIDI interface and a Korg M-1 rackmount. *5619 N. Beech Daly, Dearborn Heights, MI 48127; 313-561-7489.*

Southern California

Fred Jones Recording (Hollywood) recently received first place for "Best Humorous Radio Spot" at the London International Advertising Awards. The :60 spot, "Security Blanket," was produced for Moses, Anshell (Phoenix) and engineered by Peter Cutler. *6565 Sunset Blvd., Suite 211, Hollywood, CA 90028; 213-467-4122.*

Leeds/Le Mobile (N. Hollywood) has installed Audio Analysts' professional touring systems in all three of its rooms. Rooms 1 and 2 contain HDS-3 main cabinets; Room 3 has HDS-4 main cabinets. All three feature bi-amped wedge monitors, bi-amped drum monitors, mixing boards, cassette decks, reverb, delay, microphones and DIs. Power is supplied by QSC, Crown and SAE power amps. *11131 Weddington, N. Hollywood, CA 91601; 818-980-7774.*

Buena Vista Sound (Burbank) has named Dave Campbell and Gregg Rudloff rerecording mixers. They are responsible for re-recording and mixing on motion pictures and filmed TV productions for Disney and other outside producers. *500 S. Buena Vista St., Burbank, CA 91521; 818-560-0044.*

Northern California

Music Annex (San Francisco) has named Jon Grier sound designer/mixer. *69 Green St., San Francisco, CA 94111; 415-421-6622; fax 415-391-4995.*

Northwest

Klub Kev's (Seattle) has updated its facilities to include a Neve 8232 console, Westlake monitors and an Otari 24-track. *3177 N.E. 82nd, Seattle, WA 98115; 206-527-2250.*

England

Nomis Studios (London) has signed a publishing and production contract with Warner Chappell Music that gives Nomis four or five projects per year for new artists signed to Warner-Chappell. The new signings are entitled to use all the facilities and services offered by Nomis as part of an "all under one roof" package. A budget is allocated to Nomis for each project in order to rehearse, demo, record and produce three or four mastered songs. *45-53 Sinclair Road, London 214 0NS, ENGLAND; 01-602-6351.*

Manufacturer and dealer announcements

Music Annex (San Francisco) has purchased a **Sony** D-2 Composite Digital recorder. According to Sony, the studio is the first audio facility to incorporate the recorder.

Magno Sound & Video (New York) has purchased a Sony PCM-3324A digital multitrack recorder.

The Coach House Studio (Bristol) now features an **Audio Kinetics** Mastermix II console automation system installed in a DDA AMR 24 console.

Alpha Audio (Richmond) recently took delivery of its third **New England Digital** Direct-to-Disk digital multitrack recording system.

Eel Pie Studios (Twickenham, Middlesex), owned by Pete Townshend, has installed a **Neve** VR 72 sound mixing console with recall facility and Flying Faders automation.

Otari has supplied a custom-designed, automated, 64-input, 3-man film mix console to the Saul Zaentz Film Center (Berkeley, CA).

Although Record Plant Studios has closed, **Record Plant Remote** (Milford, NJ) is in full operation. The truck has a 48-input Trident console and digital and analog capabilities for audio, and audio for film and video. *109 Pinecliff Lake Drive, West Milford, NJ 07480; 201-728-8114.*

Todd AE/Glen Glenn studios (Hollywood) has purchased three additional **Hybrid Arts** ADAP digital audio recording and editing systems, bringing the total number of systems at the facility to six.

Neve has installed its Prism Series rackmount signal processing systems at the following facilities: Roxanne Music, The Castle Recording, Javelina, Rick Hull Productions, D&D Studios and Rhythm Street Sound. Producers Trevor Lawrence, Barry Eastman, Jeff Layton and Bruce Naranan of Gnome Productions have also purchased Prism systems.

RE/P

THE CUTTING EDGE

By Laurel Cash Jones

The First Direct-to-CD Recording

Although this column normally deals with product introductions, I take great pride in departing from the norm to relate a truly historic event in our industry: the first direct-to-CD recording session on Dec. 9 at CMS Digital in Pasadena, CA.

I am aware of some disc mastering facilities doing CD reference discs that are recorded one at a time from taped sources. This was different. This was a live band, all members playing at the same time, with the mix being recorded to a CD-R. Of course, the concept is nothing new; tracking as we know and love didn't emerge until the 1960s, and direct-to-(vinyl) disc was the rage for a brief time in the '70s. But the introduction of a new recording medium, the compact disc, introduces a whole new set of challenges. I was curious to find out the differences in the making of this type of recording and the equipment utilized.

For those of you unfamiliar with the direct-to-disc recording using traditional lacquers (remember the LP?), let me describe a typical session. Unlike most recording sessions of today, the *entire* band plays the *entire* side of the LP live in the studio without the benefit of overdubbing, or going back over a portion of the recording that may have something wrong with it. That's right: no mistakes, no second chances, no "We'll fix it in the mix."

This also means that the engineer becomes part of the performance. No automation to save you. You must capture the sound of the band correctly the *first* time. Imagine how you would feel if the band got it right when you had a fader in the wrong position!

With direct-to-disc LP-type recording, the console is connected directly to the cutting lathe. The mixing engineer must be careful not to send the cutting engineer a signal that the lathe can not handle (too deep or too wide). At the same time, the cutting engineer must cut a lacquer that is loud enough to withstand the rigors of the disc mastering and plating process, and still yield a product that can be reproduced.

Sounds like fun, doesn't it?

In the CMS session, the recording process became much simpler through the use of a single point-source microphone (more about the technical details later). However, the constraints regarding errors remained the same.

Because it was December, the owners decided to throw a party, at which the re-

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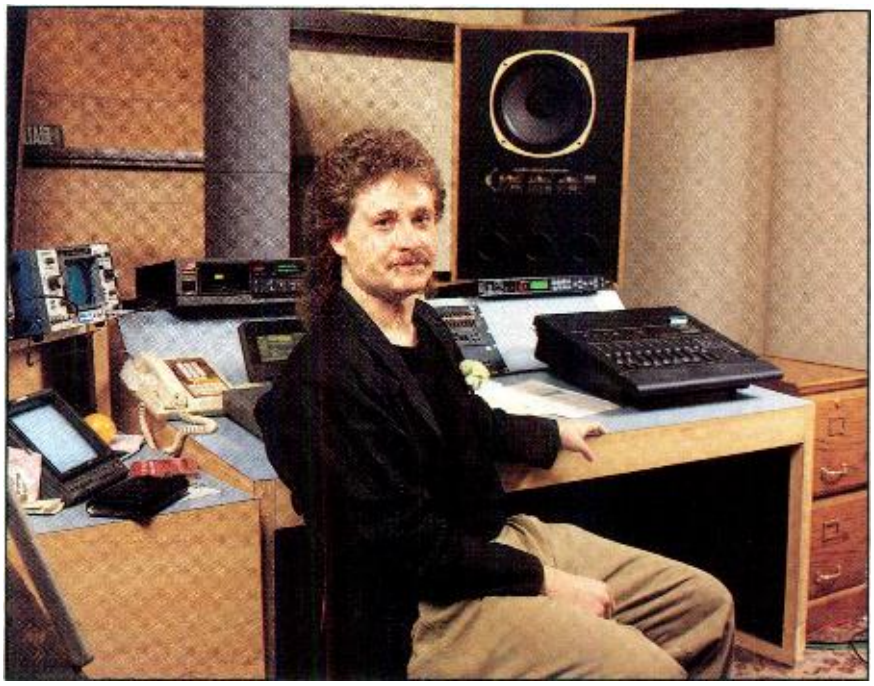
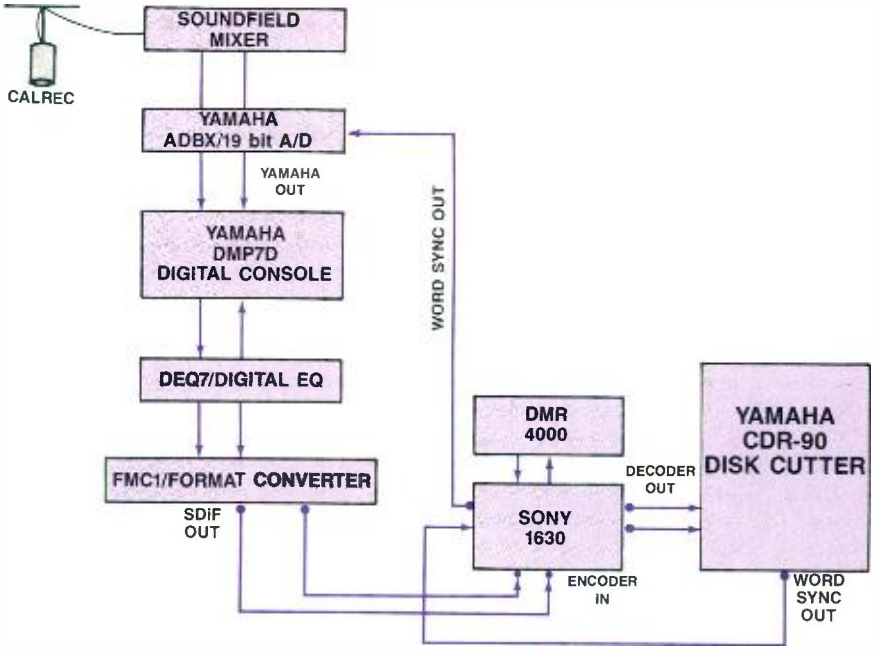


Robert Vosgien, chief engineer at CMS Digital, engineered the direct-to-CD session.



Flow chart of direct-to-CD session at CMS Digital.

cording took place, and jointly host the annual Christmas party for the Audio Engineering Society's Los Angeles section. (Conveniently, this also got them an audience rather inexpensively).

CMS was founded in 1982 by John (Bronco) Cadenhead and Gene Shiveley. The company began strictly as a rental company, then quickly added a digital pre-mastering room as a result of client demand. Shortly thereafter, CMS moved to a larger location and added several digital pre-mastering rooms, as well as a bastion of state-of-the-art digital products for outside rental. Most importantly, CMS began as a digital-based facility when digital still had many unanswered questions and the compact disc was just a gleam in some Netherlander's eye.

The session was engineered by Robert Vosgien, the chief engineer at CMS, who chose a Calrec Soundfield microphone and used the soundfield's microphone mixer to get a stereo image. The microphone signal was converted to digital with the new Yamaha AD8X (8-channel) A/D converter, a 19-bit Delta-Sigma conversion system. The performance was mixed and processed with a Yamaha DMP7D digital mixing processor and a Yamaha DEQ7 digital equalizer. The digital mix was fed to a Yamaha FMC1 Format converter, which provided the SDIF output to a Sony 1630 digital processor. The 1630 provided the simultaneous digital signal for the Sony DMR 4000 and the Yamaha CDR-90 Compact Disc Recorder. Word Sync for the A/D converters was derived from the Yamaha CDR-90 through the Sony 1630 to the Yamaha AD8X A/D converter. Juergen Wahl of Gotham Audio designed and wrote a special PQ code program for the IBM PC, which allowed access to any portion of the CD during playback.

Gotham also provided the CDR-90 Reference Disc System, which includes the Yamaha PDS Programmable Disc Recorder as a write-once CD recording system.

With all of this technology humming away in the other room, MC Roger Layng introduced Chet McCracken's Jazz Rock Orchestra. They played a lively set, which included "Randy's 6/8," "Champaign Brain," "Pair-of-Diddles,, "Big City Chester," "Little Black Book," "The Velvet Touch," "Solos' Samba" and "Windjammer."

Band members were Chet McCracken, drums; Richard Grossman, bass; Chris Pinnick, guitar; and Alica McCracken, keyboards.

After the set was over, the newly recorded CD was removed from the Yamaha CDR-90 and played back using a home CD player over the same P.A. system that the band used moments before. I don't think I have to tell you that everyone was amazed.

At press time, it was uncertain as to the recording's actual commercial release. However, if it is released, this compact disc master will be sent to Disctronics, where it will be used as the master instead of the usual ¾-inch digital tape master. This process may allow a gain in stability and a more accurate CD pressing. We'll see.

RE/P

NEW PRODUCTS

Motionworks' Motionworker

This interface provides integration of console automation systems, tape machine synchronizers and MIDI systems with control from the console automation computer panel. In tapeless studios, the Motionworker virtual machine system controls automation systems from an external time code or MIDI Time Code source. It can also drive a sequencer slaved to the automation system. The interface can communicate with the following protocols: MIDI, Audio Kinetics ESbus, Lynx VSI, Adams Smith 2600 and Zeta 3, SSL G and E series, Amek Mozart and APC console automation systems, and the GML Series 2000.

Circle (100) on Rapid Facts Card

np Portable

The np Portable from np Elecktroakustik AS is now available from Aurora Productions. Standard features include four input channels with transformers, Penny & Giles faders, stereo monitoring with dual PPMs, a phase indicator, a conference/talk-back system, a internal mic, a cue and sine tone reference generator and 48V phantom power supply. Two summing/output amps with limiters are housed in four plug-in modules. Low-noise limiters with optimized regulating parameters may be switched into one or both outputs, with linking control signals to maintain correct stereo imaging. List price is $6,695.

Circle (101) on Rapid Facts Card

More Fun With Computers

Due to a computer glitch, Rapid Facts Card leads for the R-TEC "Practically-Anywhere-Anything Remote" in the November Cutting Edge were not processed, which is why you didn't receive any information if you circled the number on the card.

To recap, the remote has five user-assignable buttons that can control any function or series of functions. It can control any relay or contact-closure operated device or devices. These may be of the pulsed or latched-type closures, as users can select this for each device via internal jumpers.

The system uses a single twisted-pair wire of any time, and can control devices up to a mile away. List price is $750.

RE/P apologizes for any inconvenience. To save time, contact the company directly at 1032 N. Sweetzer, Suite 212, West Hollywood, CA 90069; 213-654-5710.

Circle (146) on Rapid Facts Card

DW Labs Superconductor

The Superconductor combines the punch of a studio buffer with the increased gain and sustain of a high-quality pre-amp. In the low-gain position, the Superconductor adds a 2dB boost to a guitar's signal and sends out a low-impedance signal that can drive hundreds of feet of guitar cable without losing any high frequencies. For maximum bite, sustain and crunch, there are 12dB of noiseless gain. An on-board unit is available for permanent installations.

Circle (111) on Rapid Facts Card

Bryston 280B NR interface

The 280B noise reduction interface for Nagra machines provides 2-channel control and monitoring for Dolby SR, Dolby A and Telcom DM cards, allowing noise reduction to be used in the field. The unit can be powered from the Nagra or from an optional external power supply. When used in stereo, the 280B follows the record/playback function, automatically switching between encode and decode. When used with the mono Nagra 4.2, the unit provides simultaneous decoded monitoring of the encoded recorded tape.

Circle (145) on Rapid Facts Card

ARX Systems Constant Q EQs

ARX has released three Constant Q equalizers: the Multi Q, the EQ30 and the EQ60. The Multi Q features an internal patching system that provides access of up to six channels of parametric EQ, either individually or in multiples, without patch leads. The equalizer is available in a 1-rack unit chassis.

The EQ30 and EQ60 feature electronically balanced inputs and outputs, −98dB low noise, 6dB gain control, cut and boost switchable ±6dB or ±15dB, a 30Hz high-pass filter, and standard ISO frequencies from 25Hz to 20kHz. The EQ30 is two rack units high; the EQ60, three rack units high.

Circle (106) on Rapid Facts Card

Formula Audio monitors

The SM22 and SM25 low-profile stage monitors are designed for high-efficiency SPL. Both models have a D50 2-inch compression driver mounted on a 60×50 controlled pattern horn. The SM22 features an L300FR 12-inch speaker; the SM25, an L380FR 15-inch speaker. All speakers are field-replaceable. List price for the SM22 is $1,400; direct price is $882. List price for the SM25 is $1,600; direct price is $1,008.

Circle (102) on Rapid Facts Card

Otari consoles

Otari's console group has released the first products since the sound workshop acquisition. The Series 54 is designed for production, mixing, music recording, film and audio post-production studios. It is available in 24- and 46-input configurations and features high-resolution meters, dual signal paths, 4-band EQ and 10 aux send buses. Diskmix 3 VCA and Diskmix 3 Moving Faders are optional.

The TC-100 transfer console is an application-specific design for transferring or dubbing in film and video production. It consists of one or more rack-mounted card frames for audio signals, a rack-mounted meter panel, and a separate rack-mounted power supply.

Circle (108) on Rapid Facts Card

Nagra tape timer

This clear plastic card the size of a credit card is calibrated for ¼-inch tape, 7-inch reel, 1,200-foot rolls at 7½-inches per second. When attached to the feed reel hub

of a Nagra audio tape recorder, this card will indicate the time remaining on a roll of tape.

Circle (113) on Rapid Facts Card

The Last Word 2

Available from Titus Technological Laboratories, the Last Word 2 (TLW 2) is a microprocessor-based automatic stereo synthesizer and stereo audio device. Designed for TV and broadcast production, the unit corrects problems such as the loss of a channel, loss of signal or inverted polarity. The unit features user-programmed sequencing and time delays, full metering and audio monitoring. It is also remote-controllable.

Circle (114) on Rapid Facts Card

Spectral Synthesis SynthCard

The SynthCard DSP system is a 2-board DSP "engine" that features a built-in CD/DAT digital audio interface for importing and exporting sampled sound directly from CD sound libraries or DAT in digital format. Sounds can also be recorded directly from analog sources via the optional ADA-2216 converter. The system supports 32kHz, 44.1kHz and 48kHz sampling rates. All sounds are recorded and stored in ultra-low distortion linear PCM format at 16 bits resolution. List price is $1,995.

Circle (116) on Rapid Facts Card

Spectral Synthesis SynthEngine

The SynthEngine Sampler software is a mouse-driven graphics program with multi-tasking capabilities. It includes a recorder screen with tapedeck controls and peak meters for recording and previewing sounds; a main screen; a keyboard screen; up to 16 on-screen mixers for layering instruments and controlling levels; and a waveform 'Scope screen. List price is $295.

Circle (119) on Rapid Facts Card

ProSystems' FP 121M

The FP 121M floor monitor features Fibrelite construction, which makes the enclosures 20% lighter than comparable plywood speaker enclosures, according to the company. The FP 121M 2-way system also features full high-pass and low-pass crossovers, 12dB per octave passive. A jack pan-

NEW PRODUCTS

el provides a full-range input as well as bi-amp inputs. The system includes a 4690 compression horn and a 6012 12-inch low-frequency driver, and it provides a frequency response of 80Hz to 20kHz. List price is $206.

Circle (117) on Rapid Facts Card

Digital Recordings' AD-1

The AD-1, designed for use with DAT recorders, is an analog-to-digital converter that uses "brick wall" anti-aliasing filters. These filters possess the exact linear phase response required to capture soundstage spatial relationships in both music and vocals, according to the company. The converter provides a frequency response ripple of ±0.001dB, a dynamic range of 96dB and total harmonic distortion of less than −88dB.

Circle (118) on Rapid Facts Card

Juice Goose Twelve PAQ

The Twelve PAQ single-rack chassis features six small power outputs that provide 9Vac, 18Vac, 9Vdc and 18Vdc, and six outputs for 120V use. The system has 12 amps of low-voltage current load distributed from six Microports. With the Twelve PAQ the following effects can be powered: Peavey, ART, Lexicon, Valley, Alesis, DOD, Nady, Boss, Roland, Symetrix and Vega.

Circle (120) on Rapid Facts Card

Otari E Series

The DP-4050 E Series in-cassette duplicator, an improved version of the DP-4050, consists of the DP-4050E-C2 (one master, two slaves), the DP-4050E-Z3 (three slaves) and the DP-4050E-Z buffer unit. The E Series features improved cassette transports, rewind capability on slave transports, switchable master tape and detection, short slave tape error deduction and a microprocessor-controlled transport. With the E Series, it is possible to increase the size of a duplication system to 65 slaves.

Circle (121) on Rapid Facts Card

Winsted rack rail brackets

Adjustable rear rack rail brackets are now available from the Winsted Corporation. The brackets allow rear rails on Winsted racks to be adjusted front to back, and can be installed in any position in the rack. They are available in various models to fit 22-inch D and 26-inch D vertical and slope racks.

Circle (122) on Rapid Facts Card

Atlas/Soundolier attachments

The flexible gooseneck attachments, available in 6-, 13- and 19-inch lengths, interface between mics and associated desk and floor stands. Termination is $^5/_7$-inch; 27 male and female threads accommodate all U.S.-standard mics and stands.

Circle (123) on Rapid Facts Card

Ramsa 500 Series

The 500 Series speaker system from Panasonic/Ramsa features compact, high-performance modular components and high-pressure, injection-molded resin enclosures. The WS-A500 2-way component covers a frequency range of 100Hz to 20kHz. It is designed to be used with the WS-A550 low-frequency systems and the WS-SP2A.

Circle (124) on Rapid Facts Card

FM Acoustics FM 1000-1

The FM 1000-1 ultra-high power amp features a real-time computer that constantly surveys the operational stages of the amp. Individual sensors measure all important parameters. The amp will drive loads below 1Ω and will supply power levels of up to 7,000W, 2,500W RMS, with corresponding peak output currents of more than 100A.

Circle (129) on Rapid Facts Card

Polydax PR 17 HR 100 1AK7

The PR 17 HR 100 1AK7 6½-inch midrange speaker features a flat, treated-suspension, exponential-cone, 1½-inch voice coil that incorporates a kapton former, flat-edge wound wire and high-temperature epoxy glues. The results are a driver with 101dB sensitivity and 100W power handling.

Circle (130) on Rapid Facts Card

Vega T-99A transmitter

The T-99A wireless hand-held transmitter is the latest addition to the Ranger line of wireless products. The T-99A features an easy-to-hold contoured barrel, CVX audio processing for lower distortion and the Electro-Voice N/D757 N/DYM mic element. The transmitter provides 50mW of RF power through Vega's patented internal di-pole antennas. A silent on/off audio mute switch allows the transmitter to be turned off while the RF remains on.

Circle (110) on Rapid Facts Card

Audio Cause Reference Signal Source

The latest version of Audio Cause's Reference Signal Source (RSS) generates a composite waveform that enables the sound engineer to accurately verify the frequency response of studio equipment and recordings in much less time than required with discrete frequencies. The waveform creates equal energy levels in each $^1/_3$-, $^1/_2$- and full-octave band from 20Hz to 20kHz, accurate to 0.05dB. RSS can be recorded onto tapes instead of pink noise to provide a precise EQ and azimuth reference. List price is $1,800.

Circle (153) on Rapid Facts Card

AMS ST250 mic

The ST250 stereo microphone, available from Advanced Music Systems, features SoundField technology, which makes it capable of producing phase-coherent stereo recordings. Other features include coincident stereo imaging; remote switching for vertical or end-fire positioning; remote switching for M/S or X/Y stereo formats; independent adjustment of the "M" signal polar patterns from omni through all the cardioids to a figure of eight, and of the "S" signal width; and power from mains, phantom supply or batteries.

Circle (149) on Rapid Facts Card

Sanken lavaliers

Available from Audio Intervisual Design are two new Sanken microphones: the

Aphex Studio Clock

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"A major record company came to me with master tapes that were an absolute disaster. One tune was comprised of several different takes that had been spliced together. The time code was gone and there was no way to sync it to the drum tracks. Short of bringing everyone back in the studio, there didn't seem to be any hope for this project. I put my butt on the line and said I could fix the tapes *over a weekend.*

Monday I delivered the tapes with new time code, new drum tracks. Everyone thought I was a miracle worker ... in fact now they call me the *Rhythm Doctor.* My secret, the Aphex Studio Clock, without it I couldn't have delivered. By the way, that $700 Clock made me $4000 that weekend."

Steve Klong
Drummer/Electronic Percussionist
Session Musician, "Rhythm Doctor"

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Courtney Branch and Tracy Kendrick
Producer/Engineers Total Trak Productions

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Circle (22) on Rapid Facts Card

NEW PRODUCTS

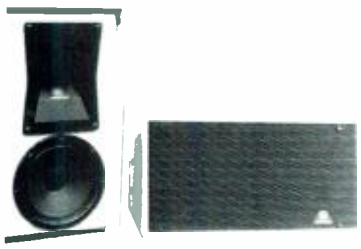
COS-11 tube and the COS-12 flat lavaliers. The COS series is designed for higher levels of transient response, extended frequency range and extremely low sensitivity to mechanical noise. The COS-11 measures 4mm×11.5mm and provides a frequency response of 40Hz to 12kHz and a dynamic range of 93dB. The COS-12 measures 2.7mm×6.8mm and provides a frequency response of 40Hz to 20kHz and a dynamic range of 97dB.

Circle (150) on Rapid Facts Card

Renkus-Heinz CM Series

The CM (Cluster Module) Series consists of two flat-front, high frequency, full-range speakers that can be clustered in several ways (the CM-61 6-inch 2-way and the CM-81 8-inch 2-way), and one 12-inch subwoofer (the Sub-121). The series features a multi-angled box design with two sides at 15° and one end at 30°. The cabinets are made of birch plywood, and the horn coverage pattern can be flipped to maximize the horizontal and vertical coverage options.

Circle (125) on Rapid Facts Card



Sennheiser HD 450 Studio

The HD 450 Studio stereo headphone is a supra-aural open-air model designed especially for applications that call for high load impedance headphones. Accuracy of the HD 450 is derived from the use of neodymium ferrous magnets. The headphone uses all field-replaceable parts and is supplied with a steel-stranded 10-foot cable terminated in a strain relief ¼-inch stereo phone plug.

Circle (131) on Rapid Facts Card

Motorola DSP56001

Motorola's DSP56001 now handles core sound functions in Digidesign's Audiomedia audio board, a sound card that performs compact disc quality recording, editing and playback on the Apple's Macintosh II. The card helps to customize audio tracks to work with visual presentations or as separate audio works. The DSP56001 provides complete control of audio signals on the Audiomedia; users can record and store sound to the hard disk, reconfigure data, mix four recordings simultaneously and emulate a 10-band graphic EQ for customizing tones and pitch. Each sound track can be shortened or lengthened when the 56001 is directed to compress or decompress the data.

Circle (126) on Rapid Facts Card

Tayo Industries Physonic Solution

Physonic Solution is an electrical contact enhancement liquid developed by Eric Data Systems and marketed by Tayo Industries. The liquid is made of a computer-synthesized polymer material that fills the microscopic pores and gaps between metallic surfaces. By increasing the contact area, a more efficient signal transfer is gained.

Circle (127) on Rapid Facts Card

PSC universal power supply

This portable supply, available from Professional Sound Corporation, is designed to power both 12V "T" (AB) and 48V phantom mics. The supply features a 9V battery that provides more than 22 hours of operation, a 3-way switchable pad of 0dB, −10dB and −20dB, a 3-way switchable high-pass filter, LED, a "T" power phase switch and rugged 6063-T5 aluminum extrusion housing.

Circle (112) on Rapid Facts Card

Korg WS Wavestation Synthesizer

The WS Wavestation Synthesizer incorporates 32-voice, 16 bit digital vector synthesis, originally developed by Sequential Circuits for the Prophet VS synth. More than 500 multi-sampled waveforms and sounds are included and can be linked together into user-programmable wave sequences. Dual-programmable stereo multi-effects, 240×64 pixel graphics, LCD, RAM and ROM card ports are included.

Circle (132) on Rapid Facts Card

3M DAT cassettes

Now available in 120-, 90-, 60- and 46-minute lengths, 3M Professional DAT cassettes use a high-coercivity, ultra-fine metal particle coating on a smooth base film, which provides a high-output and a wide bandwidth, as well as a low error rate. A durable binder handles the stresses imposed by high-speed rotary-head scanning and searching at up to 200× play speed.

Circle (128) on Rapid Facts Card

J.L. Cooper Nexus line

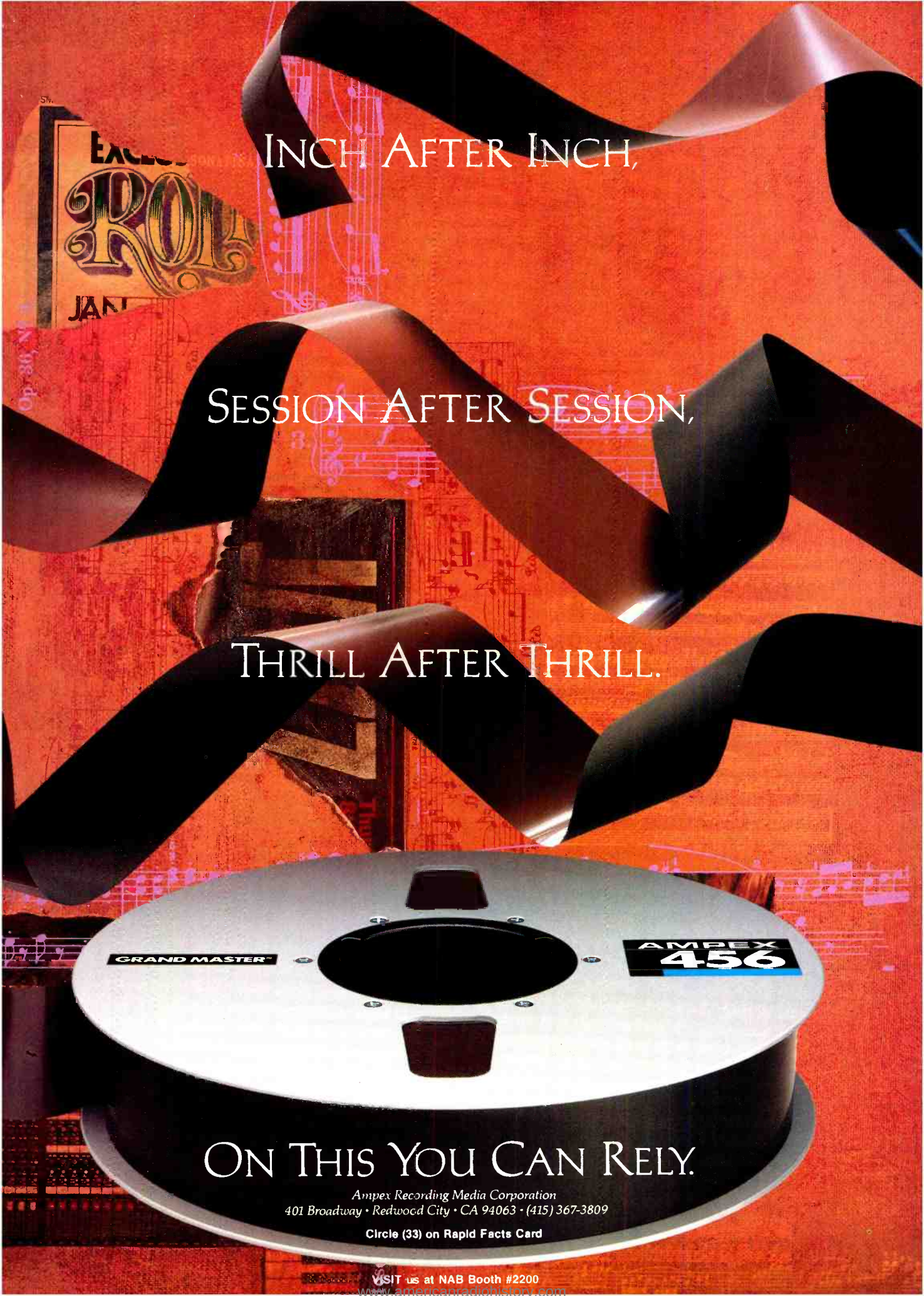
The Nexus line of MIDI switchers and interfaces features lifetime warranties and modular rack-mount capability. Nexus is a 3-input×8-output MIDI Switcher. Nexus Plus is a 2-input×8-output MIDI Switcher that features a panic button, selectable merging, transposition, zoning, channel filtering and auto thinning of MIDI controller data. Nexus M is a MIDI Interface for the Apple Macintosh that features one Mac serial port connector, one MIDI input and three MIDI outputs. The unit does not require a power supply.

Circle (103) on Rapid Facts Card

Hollywood Edge Cititrax effects

The Cititrax sound effects library is a compilation of 10 hours of comprehensive urban recordings. The 10-CD library, which was recorded on DAT in full-spectrum stereo, features more than 300 stereo sound effects, ranging from parade noises to car alarms. Cititrax is offered in CD and DAT, or AMS AudioFile-formatted tapes. Special Synclavier optical discs are also available. A demo disk is offered free of charge. List price for the complete library is $1,175.

Circle (104) on Rapid Facts Card



NEW PRODUCTS

Hardware and software updates

Otari Diskmix 3 Moving Faders

Diskmix 3 Moving Faders is now available from the Otari console products group. The system features menu-driven software; SMPTE time code frame accuracy; high-speed 68000 co-processors; and several off-line mix data editing features.

Circle (160) on Rapid Facts Card

J.L. Cooper Magi Ili

The Magi Ili Internal Retrofit Version console automation system will install into almost any mixing console, and it provides precise fader and mute automation for much less than other systems. Magi Ili consists of internally mounted dbx VCAs, the rack-mount Magi Ili controller unit, the MR-4 remote muting unit and software. The system's software runs on both Macintosh and Atari computers.

Circle (161) on Rapid Facts Card

Ensoniq EPS update

The SLT-8 drum sound library is now incorporated into Ensoniq's EPS drum machine. The library, which consists of 10 disks, was created by Ross Garfield at Music Grinder Studios in Los Angeles. Drums sounds include ambient jazz kits, modern rap, electronic drum, percussion and others.

Circle (162) on Rapid Facts Card

AMX interfacing update

A MIDI interface has been added to AMX's AV Video Remote Control products. MIDI signals can be distributed via any or all of the three SX-DCU output channels to any device that uses MIDI commands.

Circle (133) on Rapid Facts Card

Mark of the Unicorn Performer Version 3.3

Performer Version 3.3, the latest update of the MIDI sequencing software for the Apple Macintosh, features automated sliders for real-time control over continuous controller data such as volume and pan. The sliders allow the user to create custom consoles of on-screen sliders in any of four configurations. The software also features Auto-Scroll and Smart Quantize functions. List price is $495; it is free of charge to all registered Version 3 users.

Circle (134) on Rapid Facts Card

Intelligent Music UpBeat 2.0

Intelligent Computer Music Systems has released an upgrade to UpBeat, a Macintosh MIDI music software program. Version 2.0 is designed as a graphic rhythm sequencer and features a device list that allows drum machine and sampler sounds to be loaded into the program and recalled by descriptive name rather than MIDI not

number. It also includes a real-time editing function and automatic variation features, which make it possible to program drums and rhythms without the repetitive qualities associated with drum machines. List price is $250.

Circle (136) on Rapid Facts Card

Studer A820-24 update

Switchable Dolby SR and Dolby A noise reduction systems on one plug-in card are now available for the Studer A820-24 multichannel recorder. Studer also has introduced software for enhanced operation of the A820-24 in the TV/video post environment.

Circle (163) on Rapid Facts Card

Opcode Systems Timecode update

The Timecode Machine SMPTE to MIDI Time Code converter is now available at a lower price. Previously priced at $299, it recently was lowered to $199.

Circle (135) on Rapid Facts Card

Orion Electronic Blue Books

Orion Research's seven trade-in guides are now available in an electronic format, providing accurate pricing information for audio/video components and musical instruments. Accessible to IBM compatible computers equipped with modems and 20MB to 30MB hard disks, the Electronic Blue Books are available on specially encoded diskettes. A separate Electronic Update Service is available via modem, which will automatically update the information stored on the user's hard disk. A demo disk is available.

Circle (137) on Rapid Facts Card

Perma Power warranty

Perma Power Electronics has introduced the 14-Carat Coverage plan, a lifetime warranty and customer equipment protection guarantee for its Fail-Safe surge suppressor and power control center products. The warranty applies to all products sold after Nov. 1, 1989. All applicable models stocked by distributors are covered also.

Circle (139) on Rapid Facts Card

Hybrid Arts updates

Enhanced versions of the SmpteTrack II and EditTrack II MIDI sequencer programs are now available for $495 and $99, respectively. New features include a Cycle Record mode that allows drum machine-style recording; real-time control of MIDI parameters; time quantization during recording; song sets; and user preferences, which are remembered between sessions.

Circle (140) on Rapid Facts Card

Hybrid Arts ADAP II updates

The ADAP II digital audio recorder/editor now features enhanced software that includes crossfade editing, SMPTE chase lock and the MIDI Performance Page. The crossfade editing and SMPTE chase lock are available to current users at no charge; the MIDI Performance Page is listed at $400. Also available is the Erasable Optical Disk (EOD) option, which makes it possible for audio material to be recorded and edited directly on re-usable optical disks and archived for future use. This option retails for $5,495; optical disks are available for $249. Retailing for $795 is the Digital Audio I/O Module, which allows the ADAP II to record and play back digital audio from several sources, including DAT.

Circle (141) on Rapid Facts Card

Turtle Beach Softworks Proteus XR support

Support of the E-mu Proteus XR synthesizer is now available from Turtle Beach in an update to its Oview/Proteus synth programmer. The programmer allows the user to customize elements of the program and offers icons, pull-down menus and multiple sizable, movable, zoomable windows. A full-featured bank editor is also included. List price is $149.

Circle (142) on Rapid Facts Card

Otari Diskmix 3

Diskmix 3 Moving Faders, the latest version of Otari's Diskmix console automation system, features menu-driven software, SMPTE time code frame accuracy, high-speed 6800 co-processors and many off-line mix data editing features.

Circle (107) on Rapid Facts Card

Pixel Publishing update

Pixel now offers Super Librarian, a storage system for all types of MIDI data, for the Macintosh, IBM PC and Yamaha C1. Additionally, an editor for the MIDI Operating System (MOS) enables users to input system exclusive routines. The Super Librarian also features file compatibility between Amiga, Otari, C1, Macintosh and IBM Super Librarian data files; on-line help files; and automatic routing. List price for the Atari/PC/C1 is $149; for the Macintosh, $199.

Circle (115) on Rapid Facts Card

DAR SoundStation II option

A removable, erasable optical disk subsystem is now available from Digital Audio Research for its SoundStation II. The sub-system has applications in project archiving and sound effects libraries and provides a secure, removable back-up medium. It features the ability to playback and edit mono or stereo segments directly from the optical disk. Back-up copying or retrieval is carried out in background mode. The sub-system is a stand-alone unit that plugs in to SoundStation's processing and storage unit. It uses removable 650MB 5¼-inch magneto optical disks.

Circle (138) on Rapid Facts Card

NEW PRODUCTS

Publications

DIC Pocket Guide to Digital

DIC Digital Supply has released a dictionary of digital terminology, the *Pocket Guide to Digital.* The guide focuses on the hows and whys of digital technology, including compact disc, digital video and DAT. The book was compiled in collaboration with Data Storage Technologies, and is free.

Circle (143) on Rapid Facts Card

SSL SL 4000 manuals

Solid State Logic now offers two operation manuals for its SL 4000 G Series automated mixing console. The first manual provides an in-depth look at the G Series and the applications in each phase of the recording process. Details on signal flow and basic maintenance are included. The second manual covers session management, Intelligent Machine Control, Total Recall, and fader and mute operation. A pull-out automation reference guide is included.

Circle (144) on Rapid Facts Card

"Tonmeister Technology"

This recently translated book, written by Dr. Michael Dickreiter of the German Broadcasting System Technical Training Center, integrates music and musical engineering. A point in fact: A Tonmeister is a person who is usually a graduate of a music conservatory and who has opted to go into broadcast, TV, recording or sound reinforcement engineering rather than music performance. The 142-page book is nearly half illustration, and is an excellent textbook for short courses in audio engineering. Three areas are covered: recording environments, sound sources and microphone techniques.

Circle (164) on Rapid Facts Card

"1990 BBS Bible"

The second annual directory of computer bulletin board systems is now available from Bubeck Publishing. The 1990 BBS Bible lists BBSs by area code and by interest. The listings include specific computer interests as well as non-computer interests. Also, the 1990 BBS Bible includes information of accessing BBSs and low-cost telecommunications service for BBSing.

Circle (165) on Rapid Facts Card

Magi II video demonstration

J.L. Cooper has released its Magi II video demonstration, which provides a complete overview of the automation system. It is available on VHS or PAL.

Circle (166) on Rapid Facts Card

Ranger brochure

Available from Vega, this brochure features the entire Ranger line of wireless microphone systems. Also included are Ranger receivers and two transmitters.

Circle (167) on Rapid Facts Card

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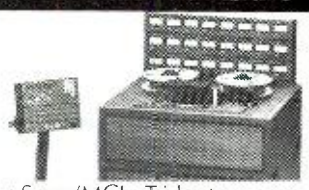

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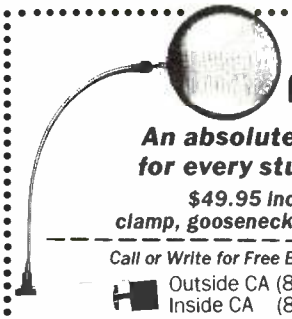

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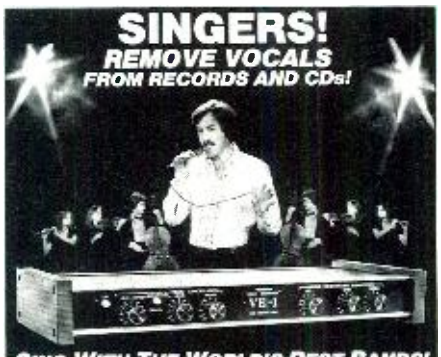

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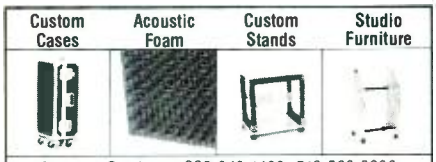

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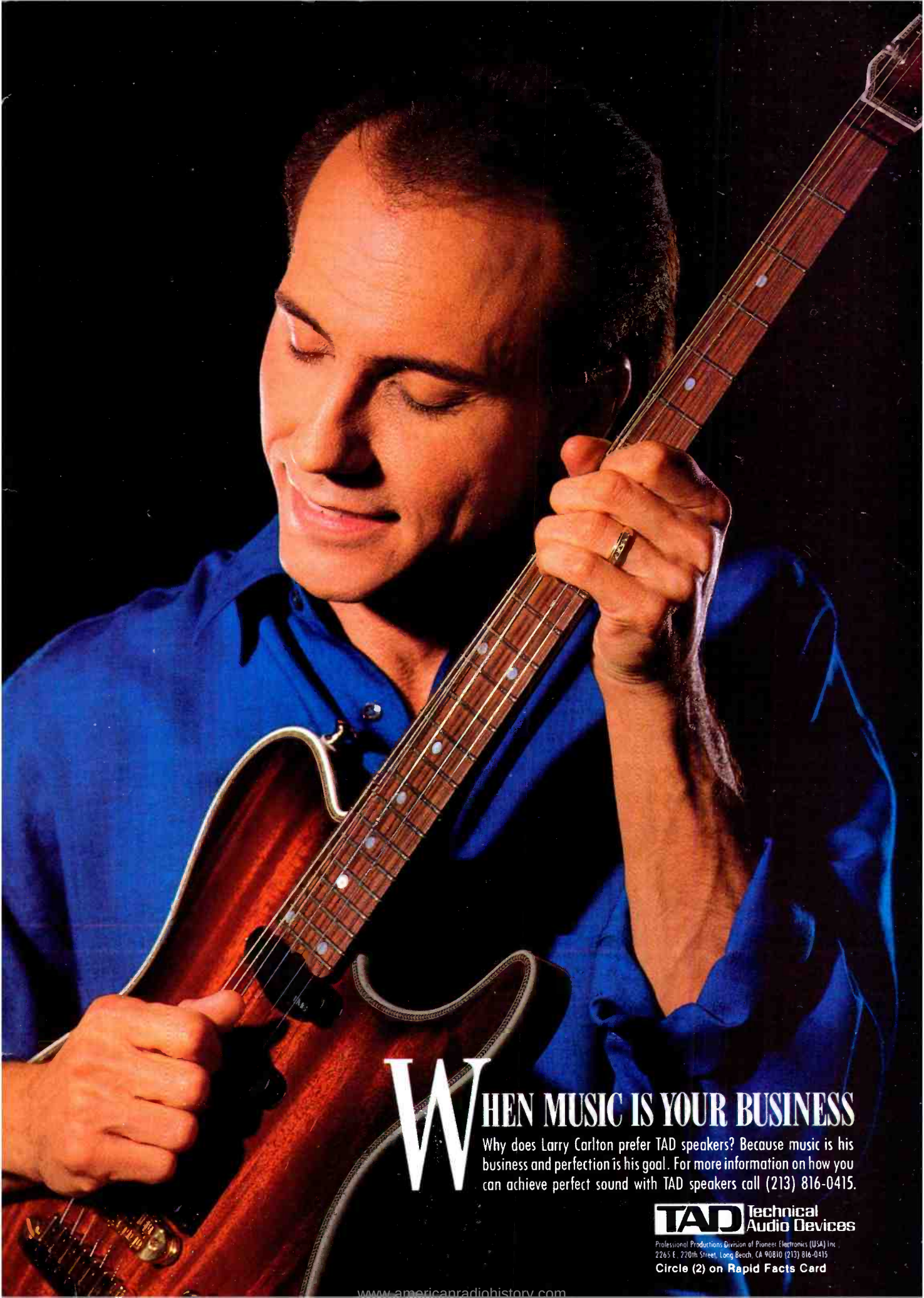

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For a 16 or 24 track studio owner, the future looks very good.

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That's precisely why we've developed the new Series 6000, an evolutionary design that clearly demonstrates the forward thinking of Soundcraft. Behind the classic layout is a revelation in performance and capability.

For one thing, it's equipped with enough busses and routing options to make adventurous productions a pleasure, not a nightmare. The 6000 is a full 16 or 24 buss console with six auxiliary sends per channel. The split format of the 6000 means each of the tape returns will double as extra inputs, with EQ.

We've also provided each input with push-button routing, EQ by-pass, and programmable electronic muting that eliminates the clicks produced by ordinary switches. You even get true solo-in-place, sadly lacking on more expensive consoles.

But it's the 6000's sonic performance that really sets it apart from the competition. Our revolutionary input design gives you 2dB to 70dB gain without a pad and virtually unmeasurable distortion, crosstalk, and noise.

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The Series 6000 input module gives you programmable electronic muting under optional MIDI control, solo-in-place to get a clear picture of your progress, and a patented active panpot with isolation of 90 dB (1kHz).

To give you the subtle control it takes to achieve dramatic results, you also get four-band EQ with mid sweeps on each input channel.

When you specify Soundcraft's Series 6000, with options including 16 to 56 channels, stereo input modules, and built-in patchbay, you'll find it an affordable slice of progress. Series 6000, simply the most comprehensive production console in its class.

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6000

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