RECORDING ENGINEER/PRODUCER

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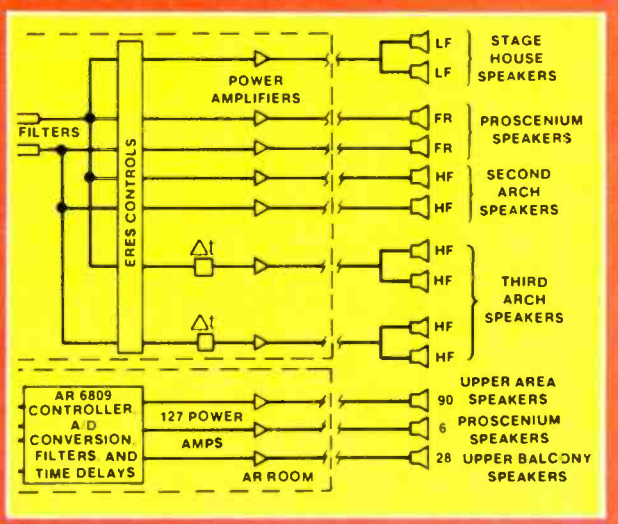

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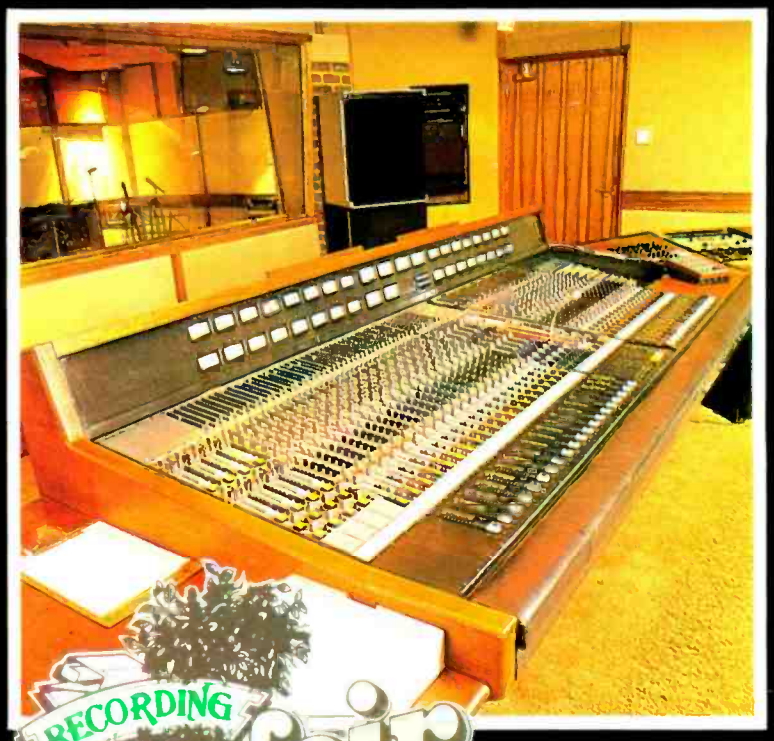

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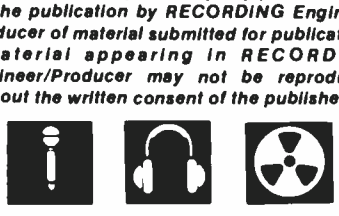

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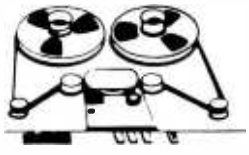

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For additional information circle #3

"It is with that

much difficulty we offer you a better way."

Harrison Systems unveiled its revolutionary Series 4 consoles at the Anaheim AES Convention and Exhibition in October

What is revolutionary is that Harrison has **never built a better console** and **never offered a lower price.** The first question that comes to mind is **HOW?**

The answer is not **"simple"** but, instead, is "with **much difficulty.**"

David Harrison relates the complete story of how the new Series 4 consoles came to be

All of the consoles Harrison has traditionally offered come from a family of consoles that share common design elements. Their amplifier designs, interconnect methods, structural systems, and other mutual elements make the differences between our traditional consoles only skin-deep. They all share a common technology of design and experience.

Two years ago we at Harrison saw the need for more cost-effective consoles to meet the needs of our customers in a changing economic climate.

Others were beginning to offer less expensive consoles, but their cost savings were often brought about by design and construction practices that produced consoles with marginal operating characteristics and questionable long-term reliability. This rather common approach to low-cost consoles has usually been accompanied, even until today, by an outdated and inefficient distribution method—a method of distribution based on a network of dealers, each making a profit of 30 to 35 percent on a typical console sale.

Harrison has chosen a **better way.** Two years ago we made a cold, hard business decision. Regardless of the difficulty created by the task we set for ourselves, we decided we would only be involved in the low-cost console business if we could continue to offer consoles with the qualities we had built our business upon: features, performance, and long-term reliability.

The first step of our "better way" was to carefully trim factory profits and to reduce the profits of our dealers to more realistic levels.

The second step, taken one year ago, was to radically change our method of distribution within the United States. Rather than continue under the old dealer-network mode, we began offering our consoles through Harrison-employed salespeople and selected organizations acting as factory representatives. Reasonable commissions paid through this method of representation are much lower than the traditional dealer profits.

The third step was begun almost one year ago. Our console design technology was not technically dated or inadequate, but it was certainly economically inadequate for a market that was making purchase decisions based on purchase price alone.

I knew what I had to do. I turned over all my normal business duties to other well-qualified associates. I then went back into the lab and, for the first time in seven years, began to design a console **from the ground up.**

The result of that work is our Series 4 consoles. Creative application of today's best technologies has allowed us to actually construct a superior console at a lower price than ever before.

It would have been easier for us to simply lower our standards in order to lower our prices. Others have done that. But **it is with much difficulty that we offer you a better way.**

Others have offered low prices, but now, **you get the price and you get a Harrison.**

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For additional information circle #4

DO SOMETHING GOOD FOR YOURSELF

The creation of the Audioarts Engineering 8X Series console marks a new turning point in the technology accessible to the 8, 16 and 24 track recording profession. This console series affords the features and technical excellence previously available only in larger track formats — features like three-band sweepable frequency semi-parametric equalization, full 24 track monitoring capability, mixdown subgrouping, stereo monitor sends, electronically balanced inputs and outputs, truly flexible effect send and return functions, and fully modular plug-in construction.

The features don't stop here; 8X Series consoles also include super solo sections (giving instant access to pre-fader, post-fader and tape solo), comprehensive slate and talkback systems, a built-in calibration oscillator, and a high speed LED metering array in an easy-to-read meter bridge assembly. Standard module features include XLR balanced inputs (both mic and line), XLR balanced outputs (buss and stereo master outs), continuously variable mic and line input gain controls, switchable phantom power, phase reverse, pad, 12dB/octave high pass filter, EQ bypass switch, channel on button (w/LED indicator), channel peak clip LED, and the exclusive Audioarts Engineering M-104 precision conductive plastic linear fader.

The 8X is an excellent choice for the small studio in need of upgrading performance or expanding format. For the large studio the 8X is an ideal system for your Studio B or 24 track mixdown room. Because it is compact the 8X is also ideally suited to video and remote recording applications.

Whatever your application, the Audioarts Engineering 8X recording console comes loaded with features previously not found on medium format systems. The mixing engineer is afforded maximum control and creative freedom. The technical excellence of this console approaches the theoretical limits of today's technology. If you demand sonic excellence, meticulous craftsmanship and flexible control take a good look at the 8X.

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views letters news

dbx DIGITAL PROCESSOR

from: **Greg McVeigh**
Audio Services Corp.
North Hollywood, CA

I would like to comment on the article published in the October 1982 issue of *R-e/p*, regarding the dbx Model 700 digital processor. I was dismayed that the article failed to represent a full picture of current developments in digital technology.

Contrary to what one might have been led to believe from the article, there *is* an affordable PCM digital processor. The Sony PCM-F1 meets the most demanding specifications, and is available right now. The PCM-F1 is being used by Irwin Kostal (also profiled in the same issue ["Fantasia Digital Re-Recording Sessions," by Larry Blake — *Ed*]), Jerry Bruck of Posthorn Recording, Roger Nichols, and others. The film industry also is taking notice due to the F-1's capability for battery operation. The PCM-F1 lists for $1,900.

The dbx processor sounded great at the recent AES show in Anaheim, and I am very happy to see American technology in the limelight. But I do think that the whole picture of affordable digital should be presented.

CRESCENDO NAME CONFUSION

from: **William Ray, President**
Crescendo Recorders,
Atlanta, Georgia

After reading the October issue of *R-e/p*, I think some confusion may exist as to the location of our facility. As I was reviewing the issue — including my article describing operational trials of the new dbx Model 700 Digital Audio Processor — I turned to page 181, and my heart stopped. It seems that a new facility is being constructed in Puerto Rico, and that they also like the name "Crescendo." They call their facility "Crescendo Audio Productions," and have an ad in the October issue.

Now, for the record, Crescendo Recorders is located in Atlanta, and formerly we were called the Sound Pit. Our facility has two 24-track studios, as well as video post-production. We have more Gold and Platinum records under our belt than any other facility in Atlanta. In no way are we connected with Crescendo Audio Production in Puerto Rico.

I would like to add that I did manage to find Alan Manger, vice-president of Crescendo Audio Productions, at the Los Angeles AES show in October. Apparently, the choice of name was coincidental and accidental. Legally we are at a stalemate, since "Crescendo" is a generic word. However, we mutually agreed to emphasize our locations to avoid confusion in the future.

FILM MIXING FORMATS

from: **Neil Brody**
Hollywood, CA

Because of limitations of space, I was unable to provide a full description of 35mm magnetic mixing formats in my article, "Design and Construction of Film Scoring Stage," published in the October 1982 issue of *R-e/p*. Readers may be interested to know that the most common formats for the 35mm master dubs are: three-track mono, consisting of dialog on track #1, music on #2, and sound effects on #3; four-track Dolby Stereo, with dialog, music and effects /left on track #1, dialog, music and effects/center on #2, dialog, music and effects/right on track #3, and a common surround channel on #4; and six-track Dolby Stereo, which has an identical track layout to four-track Dolby Stereo, but with the addition of tracks #5 and #6 used as either low-frequency "boom" channels, replayed on speakers 2 and 4, or as discrete surround-sound channels.

I trust that the above details will provide a greater appreciation of the various 35mm mag formats leaving a film dubbing stage.

---

news

NEW MR-4 SERIES MIXING CONSOLES AND EVERYTHING AUDIO AS SO CAL REP FOR HARRISON SYSTEMS

In remarks prepared for the introduction of the new line Claude Hill, vice president of Marketing for the Nashville based manufacturer of control consoles said, "Our new MR-4 24-track music recording console and TV-4 stereo teleproduction console have been designed to be the most cost effective products ever offered by Harrison. Under the personal direction of company president, David Harrison, and engineering vice president Stan Force, the all-new console series has been generated from the basic component and circuits level through to the finished system." — *continued overleaf . . .*

news

According to Dave Harrison, "the application of new technology and manufacturing techniques developed during the past few years will allow us to deliver better consoles at lower prices, without in the least compromising Harrison's reputation for quality."

The new Series 4, as well as Harrison's full line of music recording, teleproduction, and film sound consoles are represented in the USA by a select group of Manufacturers Representatives and Factory Sales Personnel. The most recent of these to be announced was, according to Claude Hill, Everything Audio of Encino, California. Hill said, "Everything Audio, its ownership and staff have proven to be the type of dynamic and aggressive audio sales company which directly complements our reorganized domestic marketing structure."

Everything Audio will exlusively represent the Harrison MR-2, MR-3, and MR-4 Series consoles in Southern California, and will also represent the Harrison TV-3 and TV-4 teleproduction consoles on a non-exclusive basis in conjunction with Harrison Factory sales efforts.

Harrison Systems will continue direct support of the film production and post-production industry, the major television networks, and broadcast companies through its Factory West coast office headed by Ken Fay at (213) 622-0331 or (415) 441-4945.

PEAVEY ELECTRONICS ACQUIRES NEW FACILITY

According to company officials, the new integrated R&D Center will combine the electronics, transducer and guitar research departments and their associated drafting departments, and will use the latest in computer-aided design (CAD) systems. This will enable Peavey to be even more efficient in developing new product and will facilitate much shorter "lead times" in getting new products into production, and subsequently into the marketplace.

Company officials also state that this new addition will provide the necessary space for totally new programs that Peavey will be announcing in 1983 and, by relocating the Shipping and R&D Departments, will allow expansions at present locations, such as an additional commercial sound final assembly line, and double the present office space.

Peavey's facilities now include seven plants totalling nearly 700,000 square feet. Recently, Peavey announced another expansion in the nearby city of Decatur, Mississippi, for production of an "unnamed" product line. This anticipated expansion involves some 20 acres of land and at least two factory buildings.

RAY DOLBY AWARDED THE ALEXANDER M. PONIATOFF GOLD MEDAL

The Society of Motion Picture and Television Engineers has presented the first Alexander M. Poniatoff Gold Medal for Technical Excellence to Ray M. Dolby, president of Dolby Laboratories, Inc., in recognition of his contributions to the advancement of magnetic sound and video recording. Dolby is noted for his work in the design of the first Ampex videotape recorder, and subsequently for the design and introduction of audio noise-reduction systems.

Ray Dolby was born in Portland, Oregon, in 1933, and received his BS degree in Electrical Engineering from Stanford University in 1957. From 1949 to 1952, he worked at Ampex Corporation on various audio and instrumentation projects, and from 1952 to 1957 mainly was responsible for the development of the electronic aspects of the Ampex videotape recording system. Awarded a Marshall scholarship, followed later by a National Science Foundation Graduate Fellowship, he left Ampex in 1957 for further study at Cambridge University in England where he received a PhD in physics in 1961.

After his studies at Cambridge, Dr. Dolby turned to audio equipment design. The well-known Dolby-A, -B, and -C noise-reduction systems resulted. Subsequent designs include devices now in widespread use by the television, sound recording and motion-picture industries. The recent introduction of integrated noise-reduction systems for one-inch helical magnetic video recorders is said to be of significance to improved stereophonic performance of these machines.

Ray Dolby is a Past President and a Fellow of the AES, and a recipient of its Silver Medal Award. He is a Fellow of SMPTE and the BKSTS. In 1978 he was awarded the Samuel L. Warner Memorial Award by SMPTE, and in 1979, he and his colleagues received the Scientific and Technical Award, for their work in motion-picture sound systems, from the Academy of Motion Picture Arts and Sciences.

JBL AGREES TO ACQUIRE UREI

Harman International Industries, Incorporated, parent company of JBL Incorporated, has signed a Letter of Intent to purchase United Recording Corporation, parent company of UREI.

In announcing the proposed acquisition, Jerry Kalov, president of Harman International and JBL, said: "UREI is an ideal complement to the JBL Professional Sound Division. The marriage of JBL and UREI will enable sound contractors and pro-sound dealers to design complete top-quality systems from a

. . . *continued on page 117*

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Digital Update

STUDER SHOWS DIGITAL EIGHT-TRACK AND PREVIEW UNIT AT AES CONVENTION

The preproduction model of the A808PCM digital multitrack recorder is based on the A800 transport which has been adapted to quarter-inch tape. The A808PCM features up to eight channels of digital audio at sampling frequencies of 48 and 44.1 kHz, along with two analog channels for cueing and SMPTE code. Maximum playing time is in excess of one hour with 14-inch reels. The new recorder implements the stationary-head digital audio recording format jointly supported by Studer and Sony/MCI.

Digital inputs and outputs are in the interface format proposed and supported by the EBU, Studer and other manufacturers. Also, the A808PCM is said to be compatible with both electronic and tapecut editing techniques, and is supplemented by a family of remote control units.

The DAD-16 digital audio delay (preview) unit for disk cutting can be used with analog masters at normal or half speed. When used with a digital mastering machine, the DAD-16 can be fitted with either standard Studer digital interfaces, or with custom interfaces for any type of digital audio equipment on the market. The unit is compatible with all types of cutting lathes in use today.

SONY INTRODUCES DAQ-1000 CUE EDITOR FOR MASTERING COMPACT DISC

The new DAQ-1000 encodes the digital master tape with various types of information, including copyrighting data. Accessibility for the end user of the encoded data includes music selections, read out of playing time, and play time remaining.

Cue information is recorded on one audio channel of the U-Matic digital master videocassette tape, and may be used in various system configurations: by direct input, in conjunction with the DAE-1100 digital audio editor; or by use of the DAQ-1000's 10-key data pad.

EXPANDED PCM FORMAT CAPABILITY AT DIGITAL SERVICES

Digital Services, of Houston, Texas, now offers true digital duplicating, editing, and disk mastering for the Sony PCM-F1 processor, as well as Technics digital processors and any other 14-bit EIAJ standard digital recorder. Utilizing the Sony PCM-100 14-bit digital processor in combination with DAE-1100 digital audio editor and PCM-1610 16-bit digital recorder, users of the consumer-type F1 processors may achieve the full benefits of true digital editing and disk mastering.

True digital transfer of 14-bit videocassette tapes to full 16-bit PCM-1610 format is available at Digital Services, and recommended to those seeking the fullest performance from their recordings.

Recent bookings for Digital Services have included digital mixdown for Jerry Lee Lewis' next album release recorded in Muscle Shoals, Alabama; remastering of Bruce Springsteen's *The River*, in Nashville; digital recording of the Chicago Opera at Northern Illinois University for Orion Records of West Germany; and the provision of editing facilities for Dionne Warwick's current album *Heartbreaker*, recorded at Middle Ear Studios, Miami.

■■■

Audio/Video Marriage

POLITICS IN THE AUDIO MARKET-PLACE

by Martin Polon

The year 1983 has become the make or break time frame for many segments of the world's audio-visual economy. The slow movement towards economic recovery will be superimposed upon public demand for superior audio entertainment, both alone and as part of visual entertainment. Of foremost importance in 1983 will be digital disk versus professional digital recording; satellite digital transmission; and stereo television and motion pictures for home entertainment.

As has been discussed in this column before, the technology is in place for every category of improved audio for recording, distribution and reproduction. Less well organized are the marketing and distribution politics of the marketplace, which are keeping new products and services from being accessible to either the user or creators of audio entertainment.

That the audio industry can benefit from new technologies seems clear. It is true that phonograph records will gross nearly $4 billion dollars in 1982 sales of singles, albums and music tapes. That is more than theatrical motion pictures, or arcade video games will input in their separate categories. The record business is still moving product, but the profitability and diversity of performance has waned, especially for popular music.

It seems likely that digital disk will reclaim much of the aural excitement that critics and the listening public feel has left the home audio field. The stakes are high, since by 1990 digital players and digital disks could account for sales in excess of $10 billion per year, or roughly three times the current sales of analog disks. The presence of a digital system in the marketplace will slow the sales of analog disks very little initially, with the two systems expected to coexist until the end of this century.

As of the beginning of 1983, the Compact Digital audio disk is undergoing the following indignities. It still is being treated by many in the professional audio fraternity as a secondary standard providing less than professional results with its 44.1 kHz sampling rate. There has been strong rumours of minor dissensions between the major partners in the project, Philips and Sony, over US introduction, European introduction, pricing, the size of Philips' per disk royalty (10¢?), and other issues.

Both partners deny instability, but recent labored statements of harmony would seem to confirm the strain. Whatever the dynamics of the Sony-Philips relationship, the proposed January 1983 introduction to the US (Japan has the product now), will find an $800+ player with perhaps 100 or so titles. That is enough to launch the product as an esoteric tool, but not any way to create a mass market.

For several reasons the digital disk is likely to suffer slow growth initially. It will be costly, with prices dropping only as the numbers of units sold rises. Sony and others have performed manufacturing research to establish a minimum price base, and a $300 retail (subject to discounting) seems likely by 1985-86.

The laser system and digital-analog (D-to-A) conversion chips can be cost reduced further. Some of the 38+ partners signed on for the system may well second-source the costly D-to-A chips in Silicon Valley rather than use

AUDIO/VIDEO PERSPECTIVES

Sony's. But, as a commodity, the digital disk supplants an existing system. Since it is a variation on a theme, it is most likely to fuel a replacement market from conventional players for the listener who has few disks in his or her collection.

The consumer with hundreds of valued records will be slow to adopt the Compact Disk, or to create a second collection of records. The broader the catalog of digital releases, the quicker the acceptance of the whole system. Which is why Philips' royalty is such an issue, since American record companies seem to be resisting the contribution of titles as much because of the royalty costs as anything else.

Satellite transmission of audio was pioneered nationwide on classical broadcasts from Lincoln Center, and on network programming over the last 10 years. The public television and public radio stations in the United States have a system called "DATE," which provides up to four discrete audio channels to every recipient station equipped with satellite downlink antennas. Almost all of the PBS stations are so equipped.

Some PBS and NPR operations have elaborate audio production facilities, like WHBH's (Boston) elegant audio production van. That PBS and NPR produces much of its programming in stereo is a fact, but it is a smaller percentage than, say, the BBC produces, with a similar network of transmission (though not via satellite). American television networks have access to stereo satellite transmission, with its virtual freedom from phase shift and delay found in terrestrial circuits; yet use it so seldom for high-quality audio as to virtually ignore its high potential. The launch in 1983, by Space Shuttle, of two more communications satellites assures the availability of additional channels for audio use.

Audio, unlike video, uses very little transponder space on a satellite. The multitude of satellites to be in the sky by 1984, including several under the Bell system label, make audio usage of satellite transmission an act of simplicity — if only the marketplace would demand such high quality. Chicago's landmark classical music station, WFMT, regularly broadcasts concerts from London produced under the aegis of the BBC. Even though WFMT is offered to millions of listeners all over the United States via cable TV systems, and delivered to each of these systems via satellite and Wegener Electronics Systems, the BBC concerts are "canned" on analog tape. Such issues as musicians' unions and satellite time cost are currently keeping live stereo off the satellites, but consumer and advertiser demand could change all of that.

The growth of stereo soundtracks for theatrial motion pictures has reached the point where 1983 will be the year in which the total breaks the half-a-thousand mark. Of this total, nearly 300 titles will be from pictures using the Dolby stereo system. Production experts from Dolby are working with several Hollywood studios to transfer films made with magnetic stereo to Dolby optical. Many studios have classics dating from the Fifties through the Seventies with elaborate magnetic stereo soundtracks. Despite all of the fears regarding storage of magnetic tracks, these older films have more problems with the fading of their chemically unstable colors than with the quality of the audio. With over 500 titles in the pool, the availability of stereo theatrical motion pictures promises virtual success for stereo TV ventures.

The advent of stereo TV in Japan had to begin dry; that is without a bang of software. Stereo TV in the US could begin with a large stock of films and video productions of a musical nature (from both the networks and PBS) complete with stereo soundtracks. Unfortunately, the process of making a decision on stereo TV has been confused by a lawsuit brought by one of the contenders fearing a choice against its system. The EIA subcommittee, ready to provide a report, now has to stall another six to nine months to clear the legal ramifications.

In addition, Grumman Aerospace has entered the picture with a "Rainbow Sound" system. This system, using a small unseen portion of the TV picture (due to overscan), provides 15 kHz audio response, and meets most of the EIA criteria. Although the Grumman system has merit, and the lawsuit represents an exercise of a company's legal rights, the TV stereo process is beginning to look like the AM Stereo game all over again.

The bottom line in all of these cases is to wonder if the old saying about the weakness of the Democratic process doesn't have some truth. The FCC which, right or wrong, used to provide decision making in some of these instances, has become impotent. The Commission has surrendered, where its input might save the industry from years of intellectual masturbation. So our bright electronic future for 1983 has taken on some of the characteristics of the neighborhood domestic scene: arguements, squabbles, petty court cases, etc. So much for progress. ■■■

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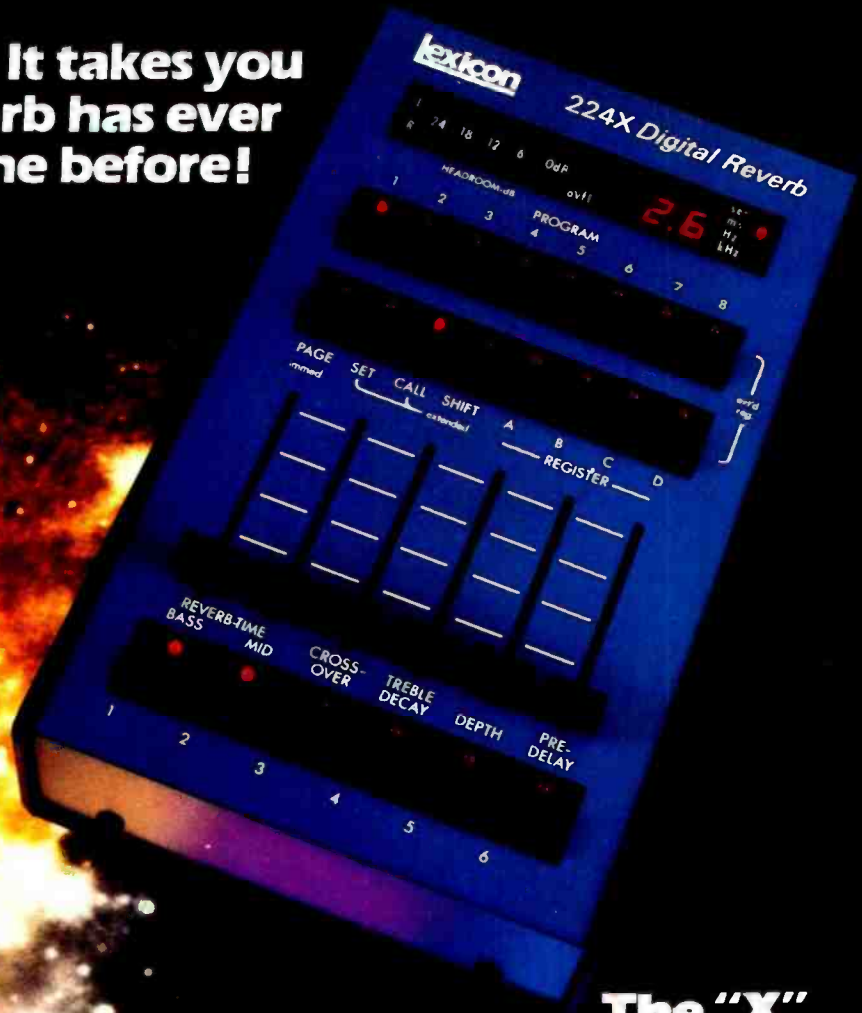

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Production Viewpoint —

ON THE CUTTING EDGE OF COUNTRY

RODNEY CROWELL & BRADLEY HARTMAN

The names Willie Nelson, Emmylou Harris, Bobby Bare, and — more recently —Rosanne Cash are often linked to a new sound in country music, a sound that has also made frequent advances up the pop charts. Intertwined with the recording careers of all the aforementioned artists are producer *Rodney Crowell,* and engineer *Bradley Hartman*. Although the two deny the existence of a "Crowell/ Hartman Sound," critics invariably praise the clean, sparse and forceful sound of their records — a sound that blends the newest rock recording techniques with the best of the "Nashville Sound."

Although both were born in Houston, Crowell and Hartman first met while attending Stephen F. Austin College in Nacogdoches, Texas. Rodney Crowell soon dropped out and headed for Nashville to pursue his budding career as singer and songwriter. In 1974 producer Brian Ahern recognized Crowell's talents and brought him out to Los Angeles, where he was recruited into the Emmylou Harris Hot Band. Crowell first won recognition as the writer of such contemporary country classics as "Till I Gain Control Again," "A Song for the Life," and "Voila, an American Dream." He has released three solo albums on Warner Brothers.

After Crowell departed for Nashville, Bradley Hartman transferred to the University of Texas at Austin, where he earned a BS degree in Radio, TV and Film. After a short stint with Leon Russell's Sheltervision company, Hartman went to Los Angeles for an interview (arranged by Crowell) as second engineer in Brian Ahern's Enactron Truck. He got the job, and soon worked his way up as mixing engineer, before turning independent.

Throughout the following conversation, frequent mention is made of musicians in the "Cherry Bombs" — the crack group of Nashville and LA studio pickers that work on all Crowell/Hartman projects. Key members include Larrie Londin (drums); Hank DeVito (guitar and pedal steel); Emory Gordy, Jr. (bass, arrangements, keyboards, and much more); and Albert Lee (guitarist and solo artist).

The following interview took place between sessions for singer/songwriter Guy Clark at Bullet Recording in Nashville.

. . . continued on page 21

Interviewed by Sam Borgerson

Photography by Beth Gwinn

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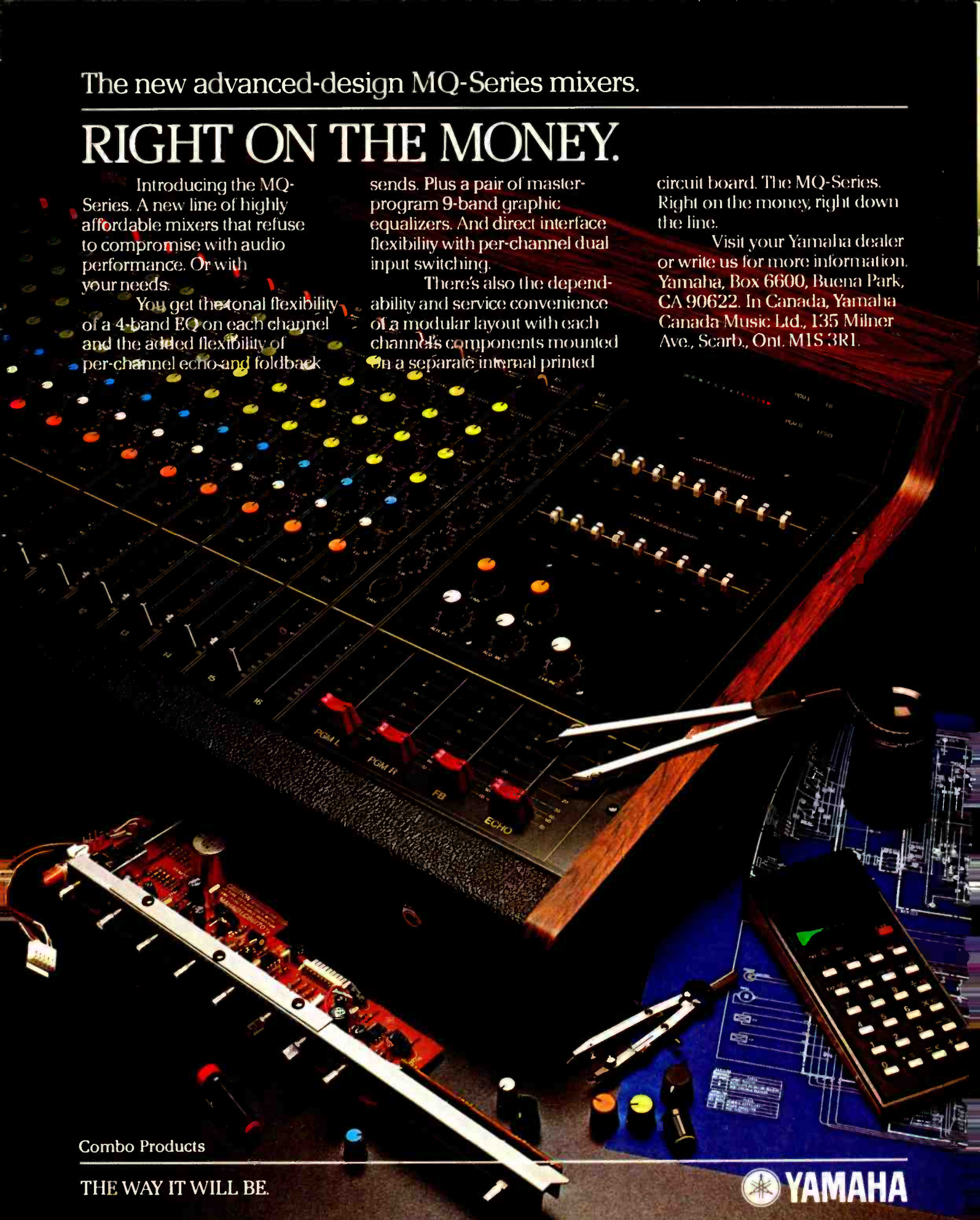

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— continued from page 18 . . .

R-e/p (Sam Borgerson): Rodney, have you been surprised by the critical acclaim you've received as a producer, and by the demand for your services?

Rodney Crowell: Yeah. I didn't start out to be a producer. It was one of those things that just happened. But sometimes those things work out for the best. Rosanne [Cash] really got me started. When she heard the demos we did she said she wanted me to make her records. Plus, I've been trying to do my homework.

R-e/p (Sam Borgerson): In a recent review, Rolling Stone *magazine called what you do "minimalist country music." Is that a fair description?*

Rodney Crowell: If that means we go for understatement, I think that's true. It's a matter of personal style, I guess, hoping that people will be charmed by it. Sometimes maybe it just misses them completely.

R-e/p: Possibly that "minimalist" approach could also apply to the way you keep the song at the heart of every cut. You don't drown it in production.

RC: Right. If there's one thing I believe in this whole music business, it's that it all starts with the *song*. I can't think of ways to make a song work if it doesn't interest me. Bobby Bare once said that if the producer and the artist don't both like the song, you ain't goin' nowhere because one of them will be dead weight!

R-e/p: Were you expecting Rosanne's album Seven Year Ache *to take off, or were you also surprised by that?*

"It all starts with the *song*. I can't think of ways to make a song work if it doesn't interest me."
— Rodney Crowell

RC: Yeah, I was at first, because everything I'd done up until then had just laid there.

R-e/p: When I talked to you back in '79, right after you'd finished your first LP with Rosanne Cash, you made the following statement: "To me there's a cohesiveness in the feel to a song on a record that you have to get by everybody hitting the right licks at the right time, and it bonds it all together, creating a feel and a color. And that should extend to an entire album as well." Have you succeeded in doing that yet?

RC: I think we got real close with *Seven Year Ache*. That has a thread that seems to run through it, both in the sound, and in the performances. It was different with *Somewhere in the Stars*. And here I'd like to say something about Bradley, and one of his great attributes as an engineer. Just about every time we start to make a record we're in a different studio. So it's been a constant thing for us, going into a new studio and analyzing what we have to do to get the sound we want, and trying to get through the paranoia of whether we're fooling ourselves about what's going on the tape. But Brad's been great at keeping the sound together through all those changes. We're getting better at handling it.

R-e/p: I certainly haven't noticed much raggedness around the edges. In fact, if I were forced at gunpoint to make a criticism of your sound, I might say it was too clean, too precise.

RC: Yes, but I think we may be evolving away from that; I like ragged, rough records myself. You see, when Brad and I started making records it was already 24-track, and we had so many different options. There were so many different avenues to follow. But then I remembered that guys were making great records back with three-track stereo, and with mono. I wished I'd had the opportunity to learn the way they had learned; to learn to point the microphone to get the sound you wanted.

So I think we are de-evolving more towards using the room and getting a live sound.

Bradley Hartman: I agree. The whole recording industry went through this thing with dead rooms and close miking and total separation, and then putting ambience back in by using reverb in the mix. Recently we've been shooting for getting more ambience on tape, and being less worried about separation. We're getting leakage in the drums now,

Recording and Mixing of the *Honeysuckle Rose* Film Soundtrack — A Personal Recollection by engineer Bradley Hartman

All the live tracks for *Honeysuckle Rose* were done in Texas with the Enactron Truck. We did seven different remote locations, going from a large concert hall in San Antonio [the Hemisphere Arena] to a little dance hall out in Fisher, Texas. I had recorded Willie's band previously for albums, so I was already used to the way they worked.

The track layout was much the same as in the studio. At the time Willie Nelson had two drummers and two bass players, so I had about eight tracks of drums, two tracks of bass, and then — depending on who was playing with him at the time, since he had several guests on stage throughout the movie — I spread out on down the 24. Usually we had about 20 tracks of live music, plus the two sync tracks — SMPTE on track 24 and the 60-cycle on track 23.

We did scratch mixes for the dailies in the front of the truck, in the small room normally used as a lounge or for overdubbing. I was doing a stereo mix in the control room, and putting that on a two-track for a live reference mix. I sent the dailies mixer a mono combination of that mix, and he'd record that on one side of a stereo Nagra, putting the 60-cycle reference on the out-of-phase center channel, and SMPTE code on the other stereo channel.

They would always shoot an "open picture" reference [or master shot] of each concert segment, which they would go back and edit later, cutting in different shots, close ups, and two-shots, etc. I used the master shot later in the mixing process. After the shot had been transferred to videocassette, along with the mono mix and the 60-cycle and SMPTE codes, I'd lock up the videocassette to the 24-track so I could watch the concert as I was pre-mixing the tracks — more on this later.

After they'd shot the open picture reference, I would then transfer the various sync tracks, and a mono mix without vocals, to the second 24-track. We would play the mono mix over the monitors when they shot the close-ups, and I'd record all of those new live vocals, plus solo instruments if that was appropriate, on the second 24-track as the musicians mimed their playing. That way we would always have a live, on-camera vocal on tape, so we wouldn't have to worry about a lip-syc that was a little bit off. The slave 24-tract was sometimes filled up, since we'd shoot another three or four rounds of close-ups over the same song, often with two or three people singing. (I don't think we used any of the recorded close-up vocals and solos in the end, but it was good to have that option as a back-up.)

Some of the vocals had to be done over in the studio, particularly those by people who were actors or actresses first, and vocalists second. But all of Willie's vocals were live

the cutting edge of **COUNTRY**

[which is] something that wouldn't have been acceptable a few years ago.

RC: The more I listen to what other producers, past and present, have done, the more I realize how I want to make records. I mean, if they're clear and you can hear everything, that's great. But I'd rather go for the extra animation that comes from a good performance with everybody locking in at the same time.

R-e/p: What's at the heart of the "Crowell/Hartley" production team?

RC: Brad knows how to read my mind. We've got a language that requires very few words, so it's very easy to get to what we want to get to right away.

Bradley Hartman: That really extends to the whole band as well. We've all evolved together, so it's a pretty comfortable situation.

R-e/p: How do the two of you work out the sound you'll go for on a particular record. For example, how do you decide when to close mike, and when to use the room sound?

Bradley Hartman: It depends a lot on the song and the artist, but we've established a few ground rules. For example, we never put the drums in a drum booth — to me a drum booth defeats the whole purpose of drums. They have to have room to expand, and the air around them is part of the whole sound. It amazes me to see people put drums in a small room with glass right at the front, with cymbals splashing all over, and phasing off the glass. It just doesn't make sense to me.

R-e/p: But don't you still try to get separation on your tracks?

BH: Lately I haven't been worried much about separation. There's a little drum leakage into just about everything — especially with Larrie [Londin, member of "Cherry Bombs" session band] because he's such a monster. He hits the drums hard! But I like the ambience of having drums leaking in; it gives the sound more *space*.

R-e/p: How do you break down a cut into basic tracks and overdubs?

BH: We always try to get the most out of the basic group of people on the tracks. Based on what you get, some things you keep, and some don't stay. A lot of times we'll overdub on the spot. If we get a good solo track, for instance, we might go back out and double it right away.

R-e/p: Are there some things you always overdub, and some you avoid overdubbing?

RC: We almost always overdub background vocals. Usually we have to go back and do the lead vocals over, although I don't like to. I like to keep the live track because it usually has the energy. Rosanne has several live vocals on *Somewhere in the Stars*. I think at least portions of five cuts on that record are live, although we had to fix some spots. Bobby Bare had just about all live vocals on his LP [*As Is*], and the same with Guy Clark on *South Coast of Texas*.

R-e/p: Are many of the solo parts live?

BH: Several on *Stars* are. If there are no mistakes on a live track, it usually fits the feeling better and has the energy.

RC: Usually working with Brad we'll be equalizing everything, and putting it down in a way the sounds all fit together. That way you get a lot better sound; a thicker sound with more edge from the energy of the people playing. I prefer live solos 10 to 1.

R-e/p: But do you isolate the lead instrument just in case?

BH: Usually. On Rosanne's record we'd put the drums out in the room, and use the drum booth for the guitar amp. We were getting pretty much an ambient sound on that. We'd put Albert's amp on plywood, and mike it back 3 or 4 feet. Most of the time it's just a single mike on the amp.

RC: Hank Devito gets an incredibly live steel sound by going into his amp, and then coming straight out into the board.

BH: He has a little Roland amp with a pre-amp output that he runs direct. He sets up the reverb and the amp EQ, and we pick it up direct.

R-e/p: What else generally goes direct?

BH: Bass almost always goes direct. There are a few times that Emory

Recording and Mixing the *Honeysuckle Rose* Film Soundtrack

. . . *continued*

except for the duets with Amy [Irving] and Dyan [Cannon], which had to be re-cut in the studio.

My job was originally supposed to end there. I had been hired to do the live tracks, but not necessarily the mixing. But Willie wanted my to be there to oversee it all the way through. Jerry Schatzberg, the director, trusted in Willie's judgement, even though I'd never really done a film mix before.

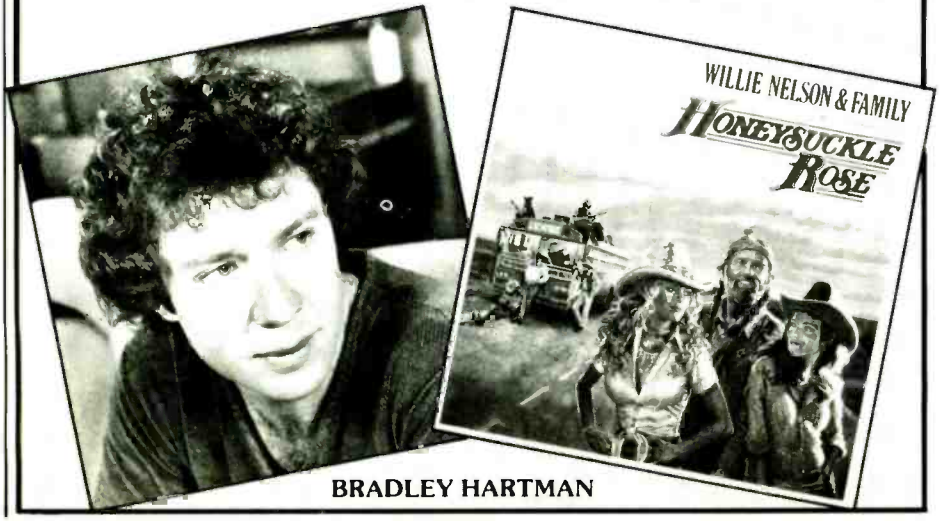
Multitrack Remix Sessions

I started the project at Soundmixers in New York, since they were well set up for film work. From the 24-tracks I usually took the 20 to 22 tracks of music and mixed down to seven tracks of an eight-track machine. I'd do a stereo rhythm track with bass, drums, and rhythm guitars. Sometimes I would leave something like a snare on a separate track, depending on what was going on in that scene of the picture. Solo instruments — lead guitar, Willie's guitar, and harmonica — all had their own track, so I could bring them up whenever there were close-ups. I put all the vocals on separate tracks, and we'd transfer the 60-cycle sync code. The SMPTE code was dropped at that point.

The next step was to transfer the tracks from the eight-track tape to three, three-stripe mag machines. This was done at a transfer facility in New York, and I was not involved in it.

At first we tried mixing the picture [soundtrack] at this place in New York, but it wasn't suitable. I got in there and discovered that the frequency response was set up strictly for dialog. There was no bottom end — 200 Hz down was simply not there. I said we couldn't mix it there, so we moved out to Los Angeles and mixed it at Todd-AO.

At that point we had all the transfer material, along with a stack of charts showing what was where. In addition to the music tracks from the eight-track mix, we had the original



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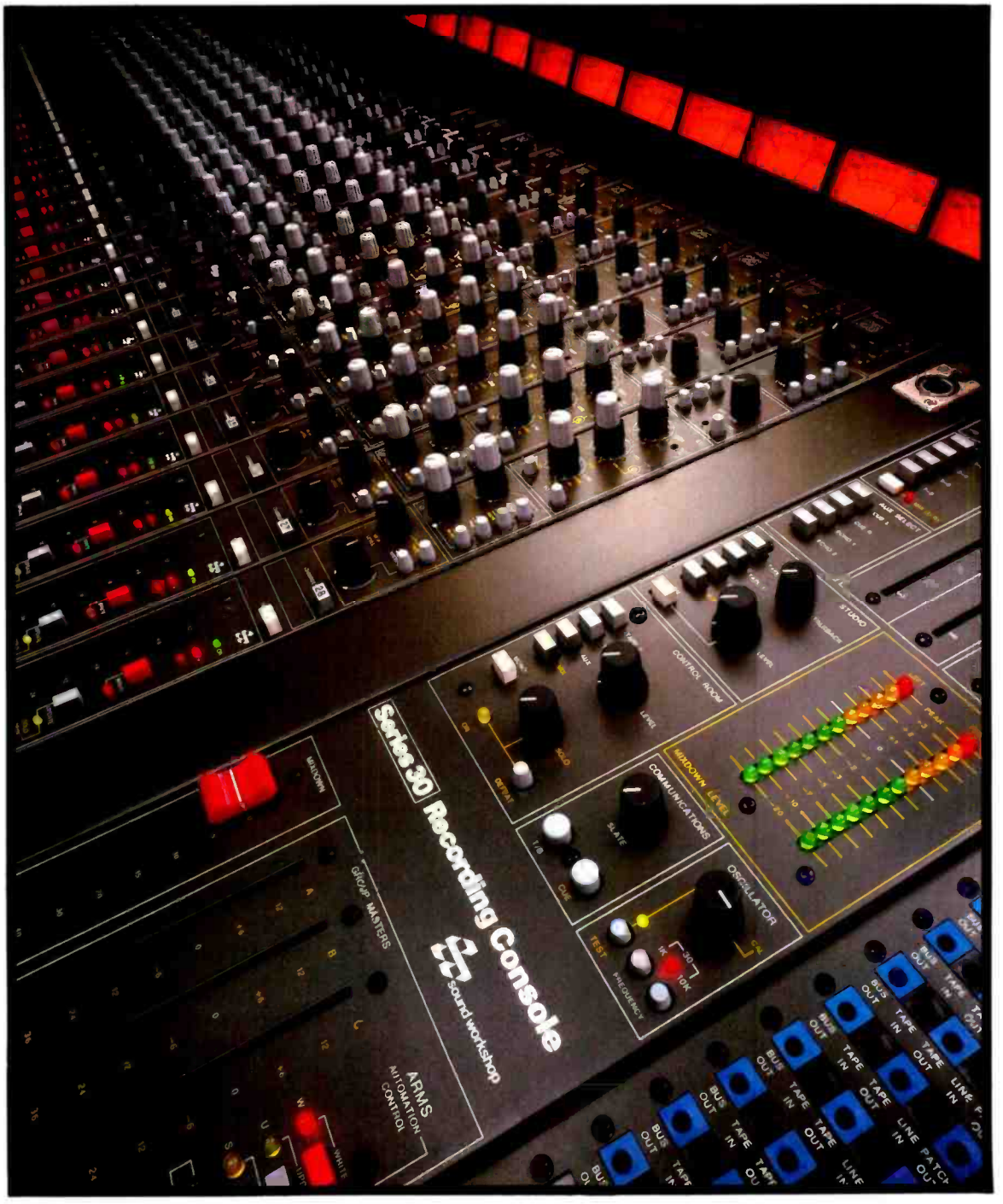

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[Gordy, Jr.] will play a Hagstrom bass, which has a click on the high end. In that case we'll use an amp which we blend with the body of the bass to give it extra "snap." Also, we've been using this interesting direct box with Emory lately. It's a tube-type active direct box. We've used it on the last few records on bass, and I've also used it on synthesizers. It makes them sound more real, more than the cold solid-state sound of a synthesizer.

It's called a Red Box. Donivan [Cowert, engineer at Enactron] actually came up with it. I think he saw it at an AES Show. There were some guys who intended to start mass production of it, and they just had this prototype at the AES. Donivan somehow got that prototype, and they were going to send him a production model when they started making them, and he was going to send the prototype back. But they never got going, so as far as I know we have the only one made. It's an interesting device. It gives the bass a warm sound that you sometimes miss when going direct.

"The guiding principle has always been to do our best to make an artist sound like himself."

— Rodney Crowell

R-e/p: We might as well get down to details. You're moderately famous for your acoustic guitar sound. How do you do it?

BH: Again, it depends on the studio, but generally I'll stick with Neumann. I like the tube 67s if they're available. If not, I like the KM-84 on acoustic. I generally stick to condenser mikes, although I have used a Sennheiser MD-441 in cases where I had to be very careful about leakage. A lot of times I'll use [Sony lavalier] ECM-50s on the inside and blend them in, rolling off the bottom to get rid of the boom in the box. As far as EQ, it'll depend on the track. I'll mold it to fit in, though usually I'll add some upper mid-range or top, from 3 to 12 kHz, whatever blends best with the track.

R-e/p: Do you add reverb on acoustic guitars?

BH: Usually an acoustic rhythm track will sound best without reverb, although sometimes with an acoustic lead I'll put some reverb on it. When we were doing Rosanne's album over at CBS [in Nashville], I discovered this storage room that just had a bunch of junk thrown in it. It was a hard-walled room with a cement floor, and it had a nice decay in it. So we cleared out a little space for Richard [Bennett] in there. The reverb on the lead in "Looking for a Corner" is from that little room, at least most of it. I think I used a KM-84 placed about 3 or 4 feet from the guitar.

R-e/p: What's your usual miking procedure on drums?

BH: Again, it will depend on the studio, and the sound of the room. But usually I like to get away with as few mikes as possible. I use the KM-84 a lot on drums, especially on toms. I usually take one mike for every two toms to keep away from phasing problems. So I'll usually use three tom mikes, snare, hi-hat, and kick. I'm getting to where I only use one overhead mike, and I count on the tom mikes to give some stereo spread on the cymbals. On kick drum I'll use a 421, or [Electro-Voice] RE-20. For the snare I've used KM-84s, Sennheiser 421s, and AKG 412, 414 or 421s.

With Larrie [Londin] we've been using a Synare [electronic drums] with a

. . . continued on page 27

Recording and Mixing the *Honeysuckle Rose* Film Soundtrack

. . . continued

crowd sounds taken off the 24-track straight on to a [35 mm] mag machine.

At Todd-AO they have a Quad-Eight board in a large theatre room with full Dolby [Stereo] matrix sound, so basically we were seeing the movie and mixing to it as if we were in a theatre. I worked in conjunction with Richard Portman, who was the main re-mix person for the movie. When the music segments were being mixed, Richard and I would sit down together, because many times there were dialog and effects intermixed with the music tracks. There were several scenes where there was music going on on-stage, and they'd cut to a scene backstage or outside the hall. The music would keep on going, of course, and we worked on making the ambience just right for those transitions.

Although we had run tape during the close-ups and cutaways, and so had lip-sync sound of these, it was decided to only mix off the original 24-track master, (including the re-recorded parts mentioned earlier), and not replace solos and vocals to match the picture edits.

Special Effects and EQ Requirements

I sometimes used a Lexicon 224 digital reverb during the final mix to enhance the natural sound. I found that the "hall" sounds programmed in the unit could be used to supplement the hall sounds we had on tape. It worked particularly well in one outdoor concert segment. We shot in a big canyon area, where the music was bouncing all around off the rock walls. I found a setting on the Lexicon with a long decay time that worked very well for giving that same feel of spaciousness.

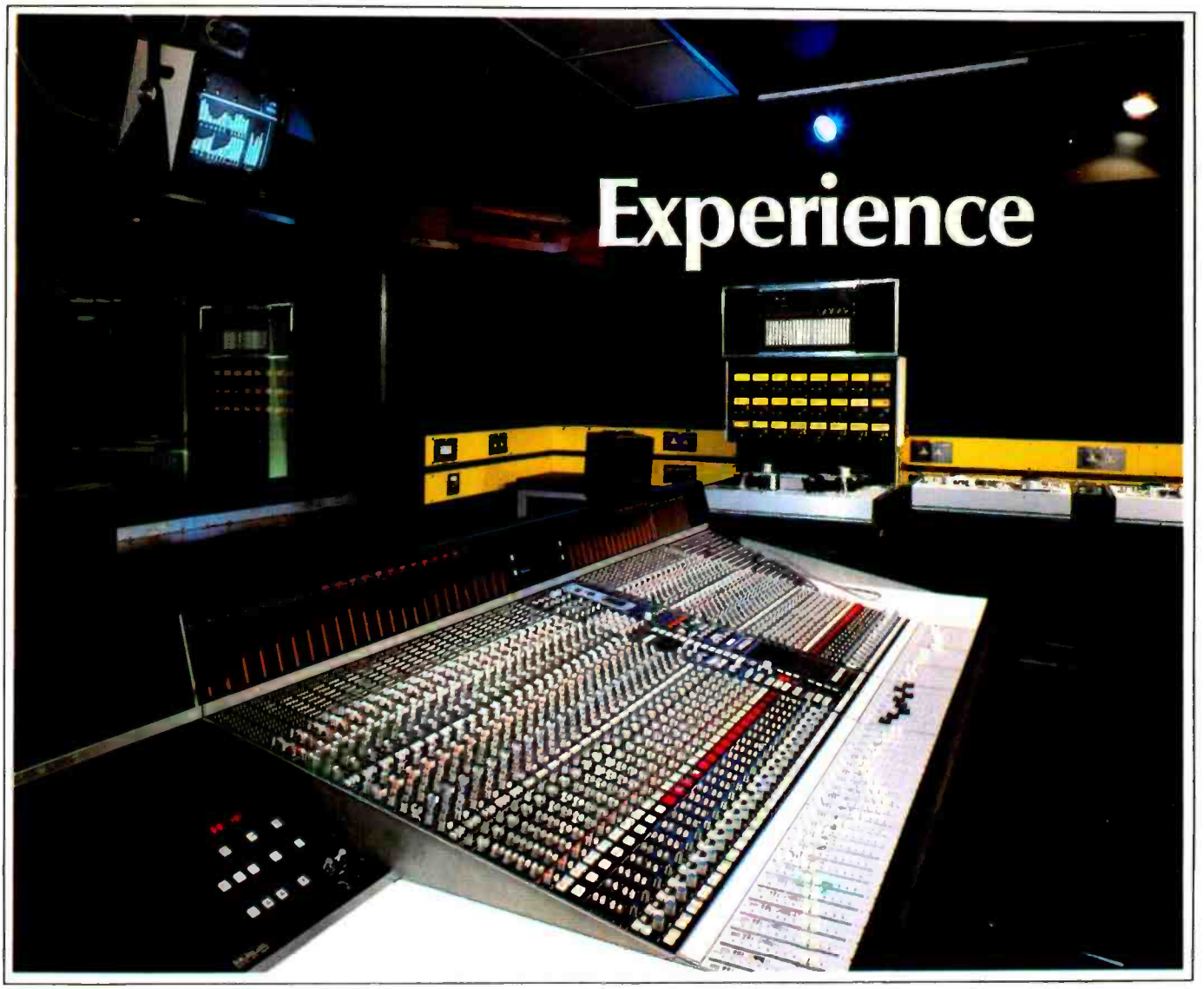
In mixing for film you have to crank the high-end more than you usually do. I cut the tracks pretty bright, and when I transferred to the eight-track I equalized a few things, cleaned up some places. When I got to the film mix, occasionally I switched in an equalizer to touch up the highs, or whatever was needed to make it blend with the picture.

It took me a couple days to adapt to mixing in such a large room with the speakers so far away. It was a drastic change from the [Enactron] Truck, where the speakers hit you right in the face. But Richard [Portman] hipped me to a few things I wasn't sure about, so in the end I didn't find the transition all that difficult.

After mixing we took some test copies out to theatres in LA set up for Dolby Stereo, so we could see what we were getting outside that room. It was a working copy, sound without the picture. So there we were, two or three of us sitting in an empty theatre just listening to the sound!

If I had to do it over again, I would like to try doing it differently. The way we did it worked well for the film editors, but it's all sort of new ground since there haven't been many musicals shot live that way. We did the "trailer" differently, the way I'd prefer to do it. The trailer was done at the Warner Brothers lot, and I mixed directly off the 24-track. We were able to lock up the 24-track as long as the film wasn't cut, and the SMPTE code wasn't interrupted. That's the *best* way to do it, since you have full control, and don't have to deal with submixes and multiple mag machines.

By the time the movie got to the theatres, I think we were about 10 generations down on multitracks, mag machines and optical prints. Fortunately [or unfortunately? —*Ed*] the sound in most theatres is so bad that you can't notice it. In fact most people said it sounded pretty good! ■■■



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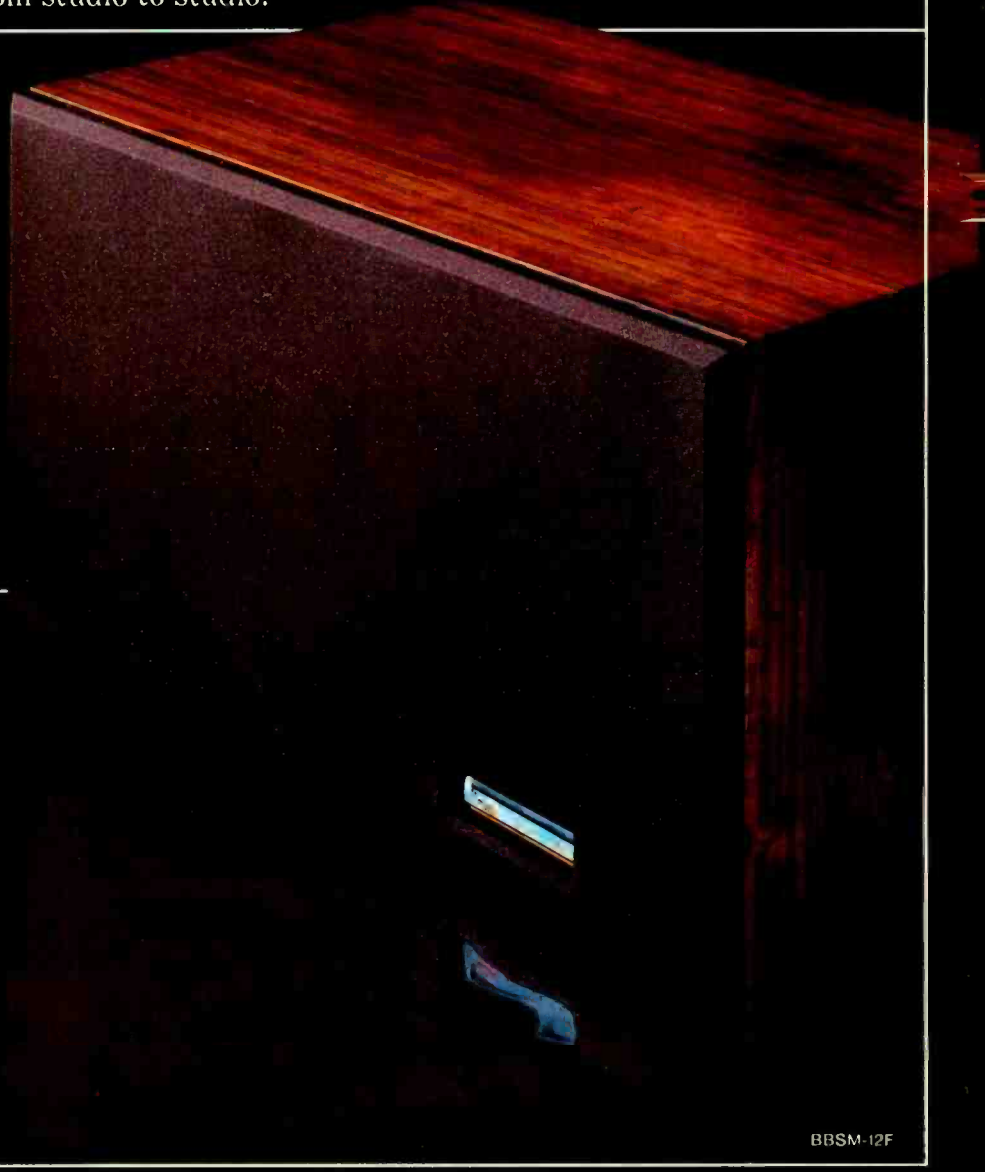

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direct feed. Larrie tweaks it up for the song to get the effect he wants, and I'll blend that in with the acoustic snare mike.

R-e/p: You achieved a fat, full kick sound on some cuts of Rosanne Cash's Somewhere in the Stars. *Did you do anything particular for that LP?*
BH: Well, Larrie got a new kick drum before we started that album. It's a custom drum with a specially made pedal —he's so powerful that I can't even push down his pedal! Anyway, it's a 20-inch drum, but it's deeper than most drums.

I'll move the kick drum mike around for different effects, depending on the type of song we're cutting. For a rock 'n' roll song I'll have it all the way inside, and pretty much pointed dead on to the impact of the pedal, to try to get as much punch as I can; as much bite as possible. Then I'll fill out the bottom-end from there, adding whatever I need.

R-e/p: How do you combine or group the mikes for routing to the multitrack?
BH: Let's see . . . one track for kick. Snare on a separate track, though sometimes we'll use two — [the second] for the Synare, or for a bottom mike. One track for hi-hat, since Larrie does some nice things on that and we'll want to bring it up. Then two tracks for a stereo spread on the toms, and one for the overhead.

R-e/p: What effects do you use on drums?
BH: I really like a good-sounding EMT plate on drums. That seems to be pretty much standard. On some tracks I'll use the Lexicon 224 [digital reverb], and with that you can actually make an artificial room sound. We'll take a short decay sometimes to add room ambience.

R-e/p: Are there any studios with live chambers that you like particularly?
BH: The one we used at United Western [Hollywood] is good. We used that a lot on Albert's record.
RC: Yeah, United's right on Sunset Boulevard, and when we were mixing that album sometimes we'd have the chamber up loud, and we could hear Sunset Boulevard coming through the monitors between takes. Blending in a bit of Sunset Boulevard added a nice feel to that project!

R-e/p: What's your approach to recording acoustic piano?
BH: I'll use a basic stereo setup using two 87s, usually. It's a pretty consistent formula for getting a good sound from studio to studio. If they don't have those, I'll use AKG 414s. I'll bang on the piano to see where the spread is — where the best sound is coming from.

R-e/p: Any effects you like to mix in with the piano?
BH: Aphex [Aural Exciter] works really well on acoustic piano; it's one of the better applications of Aphex. A lot of times people will add top-end to piano to make it cut through, and sometimes that will make it sound harsh and unnatural. Aphex seems to give it more presence, without making it harsh.

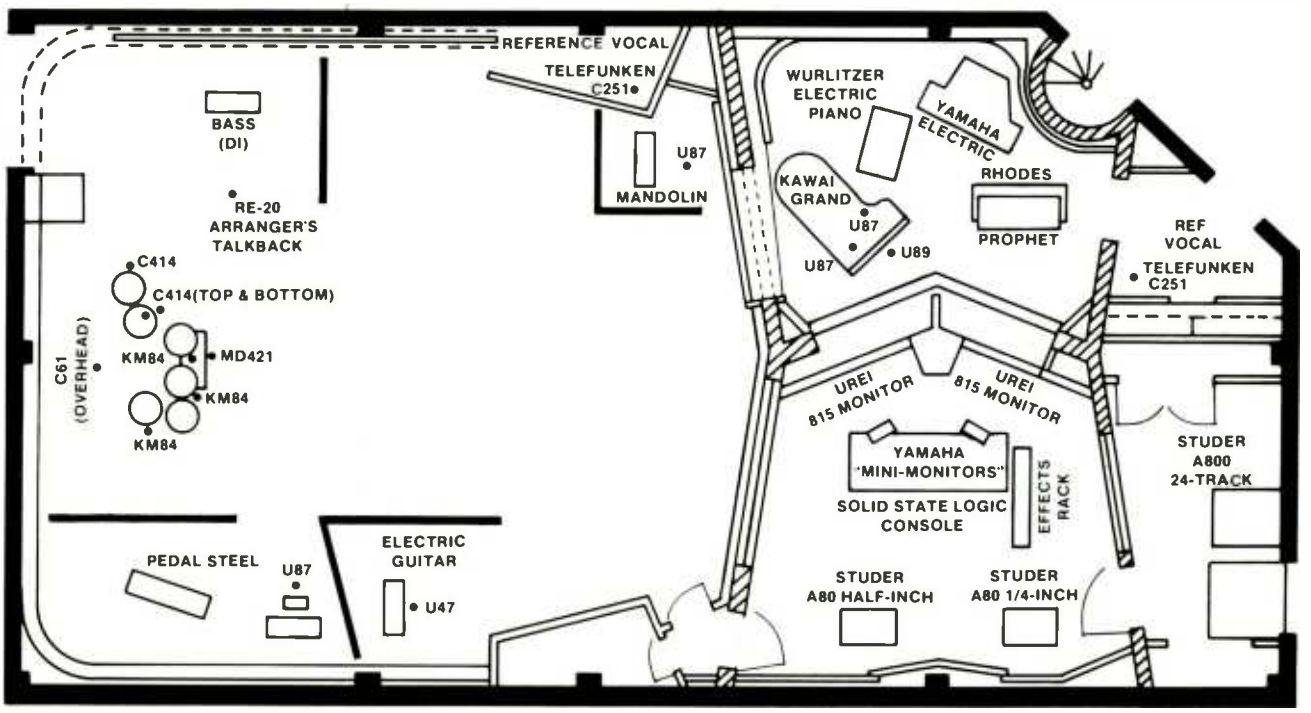
R-e/p: What are your favorite mikes for vocal tracks?
BH: I like tube mikes if they're available. With Rosanne we usually use the U47 or the M49 Neumann.
RC: What's that unusual Shure mike we use sometimes to get that sound?
BH: Oh, that's a SM-7; it's a disk-jockey mike. It sounds great on vocals for real rocking tracks. It has some well-placed frequency bumps that give vocals an "edge."
RC: It makes them all sound like Little Richard records!

R-e/p: Are there particular limiters you favor for vocals?
BH: [UREI] LA-2As if they're available. If not, an LN1176 is my favorite. I try to keep limiting to a minimum; just trying to catch the peaks.

R-e/p: You achieved some nice-sounding sax solos on Rodney's last album. Did you do anything particular on that?
BH: Nothing all that unusual, although we have done some experimenting with playing the saxophone into the piano. We play it right into the lid of the grand, and you can get some interesting harmonics doing that. I don't know if we've ever used any of those tracks, but we've played around with it.

— continued overleaf . . .

Microphone and Room Layout for Guy Clark sessions at Bullet Studio, Nashville. Engineer: Brad Hartman.



R-e/p: So much for tracking. How do the two of you go about putting together a mix?
RC: The best way for us to mix is for me and the artist to leave Brad alone for a few hours. He'll get everything set up, get the levels balanced the way he likes, and then we'll come in. There's usually three or four of us there, and we'll try to mix it live. We'll use automation, but we don't build our mixes on the computer. We used to do that, but we found ourselves running down blind alleys and getting into trouble.
BH: It was taking the spontaneity out of the mix, totally.
RC: Now we go for a manual mix — a live mix — with everybody pushing faders to pump life into it. We run the computer just in case something great happens, and we want to document it.
BH: By the time we get to mixing, everything is pretty well ordered on the tape. I've been trying to get to the point where I never have to switch in an equalizer on the mix. Of course there are times when you absolutely have to, but we're trying to get everything right the *first* time, so you don't have to go back and interject another circuit and some more phase shift into the mix.
RC: Since we've stopped building mixes from the computer and doing it live, we haven't done any remixes. We used to remix alot.
BH: I do like the Total Recall [computer automation] on the Solid State Logic console, though. It's great for getting the console set back up the way you had it. It's flawless. We did it at Bullet once, and we couldn't tell the difference between what was coming off the multitrack, and what was the mix we'd already done.

R-e/p: Before we talk about the studios you're using now, let's go back to the old days of the Enactron Truck. What was it like working in the mobile truck?
RC: It was like being on a submarine crew, except that we didn't have a periscope.
BH: But it was nice in a lot of ways. It was parked by a house in a box canyon in Beverly Hills with 15 acres around it. [In the early days of Enactron's history, owner Brian Ahern had the truck parked beside a house, "Lanya Lane," where essentially it served as a control room for the building's various recording areas. Mudslides have since destroyed the house, and Enactron now handles, in the main, "conventional" remote recording dates — *Ed.*]
To me the truck is one of the best recording tools ever assembled. It has one of those vintage Neve boards in it that Deane Jensen went through and modified; I think it's one of the *best* track cutting boards ever.
The truck is actually a 40-foot semi trailer, from 1962 I believe — the year it was actually built. It consists of three main sections. The back part has two 24-tracks on one wall. It's set up lengthwise, with the board running alongside one wall in the center section. The monitors are inverted Belle Klipsch speakers. They served the purpose, although at this point they're not my favorite monitor.
Then there's a little room in the front which we called "the comfort center." It was used as an overdub room primarily, although on remotes it became a lounge area where you could send people to get all the "riff-raff" out of the control room.
We would cut most of the basic tracks in the house, and then move into the front room for overdubs. The front room was about 8×12 and pretty dead — adequate for the purpose.

R-e/p: Did the house add much to the sound of those records?
RC: Yeah, it was haunted! You could have mikes up and be sitting out in the truck, and hearing conversations, and then go inside the house and there would be nobody there! That's true!
BH: The house did have some good sounding rooms. There was a lot of rock — stone fireplaces — and high ceilings. One room had a tall ceiling with a window stretching all the way up, with a linoleum floor. For strings we used to put about 20 players in that room and keep it live, then we'd deaden it down for tracks. We used just about every room in that house at one time or another. There was a patio out by the pool where we used to stick Leslie speakers and guitar amps, and we'd mike them across the pool.
RD: Yeah, and we'd send guitar amps down the breezeway.
BH: Right, that breezeway was incredibly live. We actually used that on parts of Rodney's first album as an echo chamber for snare and several other things. We also used it exclusively for the little four-track demo studio we had in the house.
RC: It was the sole reverb for "Studio X."

R-e/p: What speakers and mike did you use in the echo chamber?
BH: We had these sort of custom speakers . . .
RC: They were cheapos! Cheap copies of the Klipsch.
BH: It was an inverted horn box with an amp built into it. We put a little AKG omni condenser mike at the other end; just hung it up in the air.

R-e/p: Enactron probably has the distinction of having more hit records mixed in it than any other mobile facility. Was it a good "room" to mix in?
BH: Well, I got used to it having spent so many years in there. It was a very critical monitoring situation. You had to be sitting right square in the center of the console, and leaning right over it. You couldn't move back more than a foot, or there would be bass build-up against the wall. The actual pocket was very small — you couldn't move more than a few inches either way. The way the speakers came out and wrapped around you, it was like having a big pair of headphones.

R-e/p: Both of you learned the ropes of recording while working out of the Enactron Truck. Was that a handicap when you went out to "real" studios?
RC: No, I think it was a great education.
BH: That's the way I look at it. We were making records out of a house under some pretty rudimentary conditions. You could almost look at it like a basic training camp for making records. In a lot of ways it gave us much more room to experiment by using different rooms in the house, and listening to how they sounded. For example, we'd sometimes record harmonica in the shower — things like that, which most studios don't allow for. I've found it easier now working in studios because I'd had to endure quite a few hardships making records out of the Truck. In the studio, we go in and everything's just right — more or less!

R-e/p: Did moving out of the Truck change your approach to making records?
RC: Well, when I was making records at Enactron, that was really Brian's [Ahern] whole domain, and I kind of made records the way he did when I was there. I didn't really start my own thought processes about how to make records until I got away from the truck, and started working in other studios. That made me analyze situations more, and forced me to make decisions. It made me lock into taking a professional attitude, instead of treating it like a great hobby.

R-e/p: You also work in Brian Ahern's new Los Angeles studio, Magnolia Sound. What kind of facility is it?
BH: They have a Sphere Eclipse C console, and Stephens multitracks. They also have a lot of outboard equipment —an old Pultec and several LA-2As. It has a large room, about 30 by 40, and it's very live — probably one of the livest you'll hear in a big studio; it's got teak all over the floor. When we do strings in there we pull up all the carpets and get it as live as possible. String players enjoy it in there, though one contractor said it's *so* live that you can't sluff off and make mistakes, because you'll hear it right away!

R-e/p: What do you look for when shopping for studio time?
BH: Well, a big room to begin with . . .
RC: A couple isolation booths.
BH: And a respectable console. There are a couple I don't like working on.

— continued overleaf . . .

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RC: Give me Neve and Trident.

R-e/p: You've worked quite a bit at Davlen in LA. Do you like the room there?
BH: Yes, it's a spacious room, and the dispersion is very even. When you clap, there's no real ring. It's almost like free field, it's so even.

R-e/p: Which studios do you like here in Nashville?
BH: I liked CBS in some ways. The atmosphere was maybe too subdued for our tastes, but I liked the room and the history behind it. Also, their mike collection was incredible! They had 10 or 12 old tube U47s, and a collection of equipment and mikes stuffed away in closets — just about anything you wanted — old RCA 77s, and things like that. But I guess there's no use in talking about CBS Nashville anymore!
RC: The two studios I like in town are Woodland and RCA [now Music City Music Hall] — the big room that they've just upgraded with the new Studer equipment. I love those basketball gym-sized rooms. To me, that's where records are *supposed* to be made. I don't like a tight, dead room.
BH: I think it's a much looser recording environment in a big room, too. Nobody's cramped for space, and you can get up and walk around without stumbling over things. And for recording drums, the more space you have around them the better.

R-e/p: Let's talk about how you put it all together on Rosanne Cash's Seven Year Ache. *First, those famous handclaps on the title cut?*
BH: They came in at the end. They were the last thing we put on. That was a good example of the ambience of that live room at Magnolia. It was Rodney, Emory and Hank out in the studio, and we miked it about 2 feet away on an omni pattern to pick up the room, so it captured an ambience around it.
RC: I think it really made the record.

R-e/p: Was any signal processing added to it?
BH: Maybe just a touch of reverb, but the basic sound was a room sound.
RC: The truth about those handclaps is this. I had the idea, and at the beginning we put them in about five places in the song. Then we listened to it, and realized they only belonged in two places. So it wasn't such a brilliant idea until we figured out where they should go!

R-e/p: There's one basic riff that starts the song, and goes in and out all through it. What instrument is that? It sounds almost like a synthesizer, but none is listed on that cut.
RC: That's Hank [De Vito] using an E-Bow on his steel guitar.
BH: An E-Bow is an electric device that vibrates the string continuously. It's magnetic I guess, since it's not mechanical. You can't feel it vibrating.
RC: It sustains the note so you can slide up and down with it. There's also a 20-piece string section buried in there behind him.

R-e/p: Yes, I've noticed that the strings are usually mixed way down on your records.
RC: Well, I always hated cascading strings on records. I'm one of those guys that loved Ray Price until he started making records with strings all over them.

R-e/p: You've got so many talented multi-instrumentalists in the "Cherry Bombs" group. How do you determine who will play what?
RC: Emory's always very involved in that. He's really a silent producer in this group. In fact, everybody in the group has a hand in the production. The way it works is that Emory and Hank usually stick pretty close by us. And Emory is so versatile. If you read the liner notes, you'll see he's played just about everything but solo guitar.

He's also very quick and efficient on overdubs. He can come in and, by himself, knock out several simple overdubs on a number of instruments. It's faster and cheaper than if we had to line up three or four guys on each instrument.

R-e/p: Did you do much overdubbing on the song, "Seven Year Ache"?
RC: Well, that started when me and Rose and Emory and Hank went in and recorded some tracks — it's funny, because that's an example of the way I *don't* like to make records. I think the only thing we kept on that was the original drum track. We went back and built everything around that, which gave us a thick, "textured" sound.
BH: Still I think it kept the spontaneity despite the way we did it. Sometimes you'll lose that when you add a lot of overdubs.

R-e/p: My favorite cut on that LP is "What Kind of Girl?" Was that intended to be a heavy rocker from the beginning?
BH: Yeah, it was pretty much an "animal-out" attitude from the start.
RC: I remember we made some adjustments after the basic tracks, though. We replaced the bass line; we made it a bit more "nasty." I'm proud of that cut. I think it has a pretty ballsy rock and roll sound.
BH: I like the vocal effect on that one too. I think the slap effect we added was very effective, using the Lexicon 224. I worked with it just around the vocal, then put the two together back into the tracks.

R-e/p: Was that a live vocal?
RC: No, she worked on that one.

R-e/p: That cut also has a very punchy drum sound. Did you do anything special on it?
BH: I think we used a little bit more of the overheads on that one, rather than sticking with the sound of the close mikes.

R-e/p: Have you ever felt like making a straight-ahead rock album to shun your country image?
BH: I think Albert's [Lee] album may make that statement a bit more than

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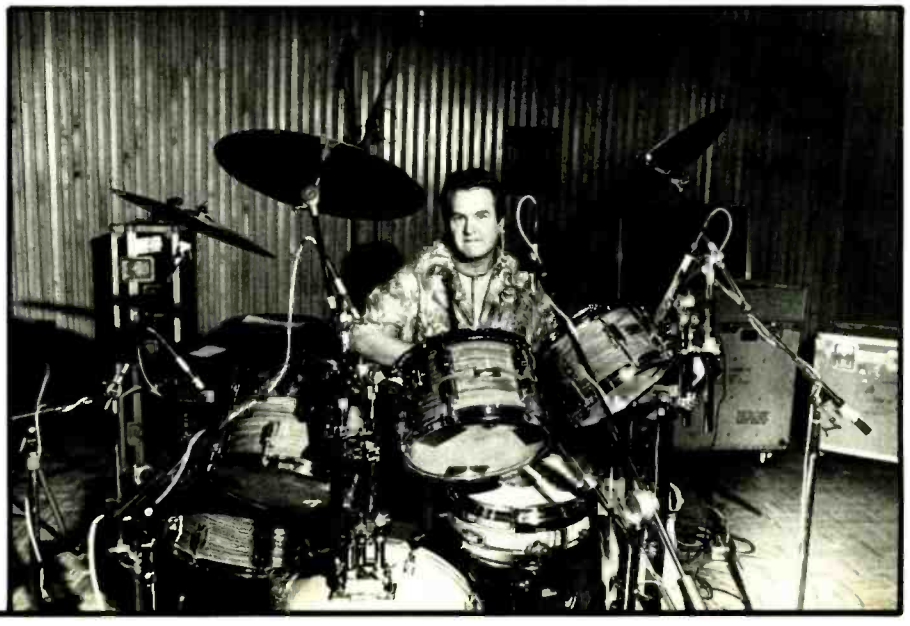
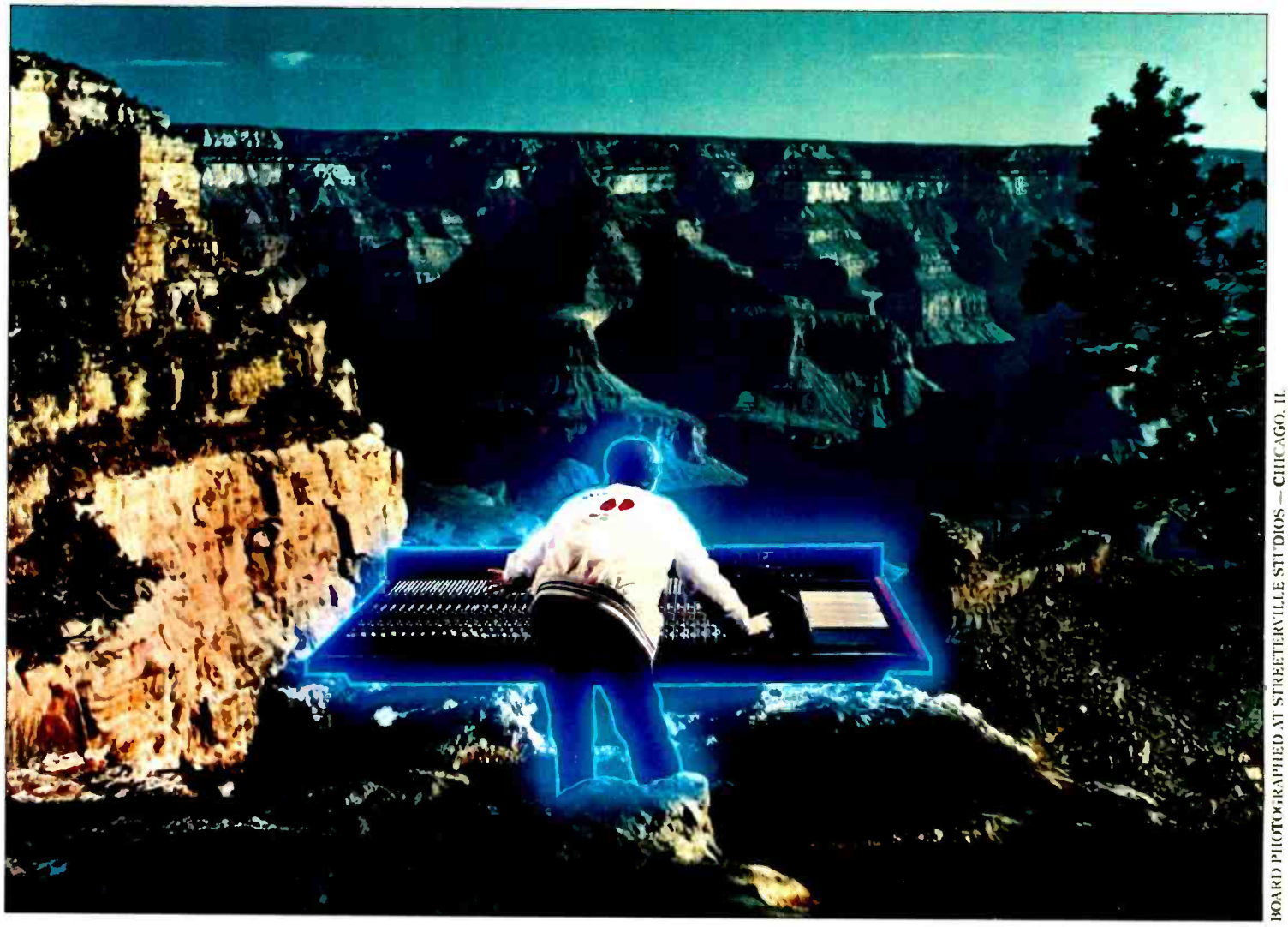
Session drummer and "Cherry Bombs" member Larrie Londin, behind an "open-field" microphone setup





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anything we've done so far. The first thing Rodney told me when we started was that we were going to pull out all the stops. I did some different things on that project, using a lot of the ambience at Magnolia. Late one night we just pulled the guitar amps out in the middle of the room, 10 feet from the drums, and turned it all up. I took everything that was live and put it on a tape slap — which I was feeding Albert's guitar and vocal through — and put it on the 24-track. We actually used it in the final mix.

R-e/p: Rodney, when you finish with Sissy Spacek, Guy Clark, Albert Lee, and whoever else is in line, are you going to make another Rodney Crowell album?
RC: No, not until all of these are out of the way. I find it hard to go right out of the studio as a producer, and get into being an artist. It takes a couple months just to wind down, and get that work out of my system. I think on my last album I made the mistake of coming right off a couple other projects, and going in to do mine without catching my breath.

On certain levels I'm happy with my album. I learned one thing — I don't think I'll ever produce my own albums again. It's just something personal. I think you've got to have one guy who can be of assistance to everybody; who can stay objective and listen and know what everybody is doing. But when you're out there under the headphones trying to sing, and get a performance on tape, you *can't* be objective. I mean, who can be objective about himself? Making a record of yourself is like those times when you go look in the mirror, and you don't know what shirt to put on. You waste your time trying to decide, when it doesn't really matter. You can do the same thing in a studio. What you need is somebody to say, "Hey, just do this!"

[At this point in the interview Rodney Crowell had to leave to continue pre-production work with Guy Clark.]

R-e/p: Let's talk about the possible problems of making the transition from LA to here in Nashville. Did you have trouble adjusting to the studios when you did Rosanne's Somewhere in the Stars?
BH: No, not really. I'd already cut Bobby Bare's album at CBS here, and I'd also done a couple projects with Willie [Nelson] at CBS. And I'd also worked a bit at Woodland.

R-e/p: At CBS they have — had — the same kind of console as Magnolia, a Sphere. How does it stack up?
BH: The Sphere is a really clean console, it's almost like . . . well, it's strange, because the older Neves and Tridents have a lot of color in terms of the way the equalizers work. But the Sphere seems to be cut-and-dried, or maybe black-or-white would be a better way to put it. You can add shades of grey, but you're still getting a very transparent image —which I like in many cases. But it doesn't allow that leeway to add some "color" if you want it. Still, it's a very good sounding console; better than some others which, out of politeness, I won't mention.

R-e/p: Does it help during a session to use the same console, or the same studio? Or do you prefer to move around and try different ones?
BH: In a way it's an advantage moving around, because you learn about more of the possibilities. But you're always a bit unsure in newer situations. You know the sound you want, but you're not exactly sure where to go in order to get it.

"For recording drums, the more space you have around them the better [they'll sound]."

— Brad Hartman

R-e/p: So you're more conservative when you first start working in an unfamiliar studio?
BH: Yeah, although usually I can pick up on a new situation pretty quickly. But there are things I'll do in the middle of a project that I wouldn't have tried the first day in a new studio.

R-e/p: Let's take Somewhere in the Stars *as an example. You cut the tracks in two different Nashville studios, added some overdubs in LA, and then came back to mix in a third Nashville studio. Did you have to futz around more in order to get an even, consistent sound under these circumstances?*
BH: Yes. When you're recording on different consoles — especially with critical parts like lead guitar and vocals — it's sometimes hard to get segments recorded at two different places to sound the same. You have to work at it a lot harder.

I've experienced this problem going from Davlen to Magnolia. At Davlen we were doing some guitar overdubs, and we tried to match the sound at Magnolia. We used the same amp and mike — a U87 — going through an 1176 limiter, adding the same amount of EQ, maybe 6 dB at 5 kHz or something like that. It was a tight-miking situation, so there weren't many variable in room sound. Well, it sounded *totally* different. In situations like that, it just takes more time to get it right.

R-e/p: Do you ever encounter tuning problems when you're overdubbing?
BH: We used to have them all the time, and we'd tear our hair out because Brian is very picky about that. But recently Emory and I worked out a technique to take care of that. He has a Korg tuner which he calibrates to the 440 on the piano, in case the piano is a little sharp or flat. Then everybody tunes from that tuner before they do a track. All the time they're playing we run an output of the Korg into a direct box, and put it on track 1 of the 24-track. When we go back to do an overdub, we'll take the tone off track 1, and put it into a Conn strobe tuner I have at the console, dialed up to the "A." We tune the instruments to that.

It's great to have this constant reference, especially when doing piano overdubs. You calibrate the tuner to the tape, and then speed up or slow down the tape until the pitch matches the piano you're using. It's saved us a lot of hassles in the last year and a half we've been doing it.

It's nice for other tricks, too. One time we were doing piano overdubs for Rodney's album. With the key the song was in, Albert had to play a lot of black keys — which he didn't want to do. So we just took it down a half step, and when we sped it back up it helped brighten the piano.

R-e/p: I also notice you work on several different kinds of tape machines in the course of a project. Does that lead to additional problems?
BH: Yeah, that's another thing I like to avoid. It's better to stick to the same type — to go from one Studer to another, for example, rather than going from, say, a Studer to an MCI or an Ampex. You're guaranteed more consistency if you stick with the same type of machine. Still, each individual machine has its own little idiosyncrasies. It may not seem all that noticeable, but you will be getting subtle phase shifts because the tracking is different.

R-e/p: Would you like to try some digital sessions?
BH: I'm not convinced that it's all that much better, though I would like to mix down to digital and see what it's like —cut some reference disks from analog and digital just to see what the difference is.

— continued overleaf . . .

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R-e/p: You and Rodney were assigned the task of mixing three-fourths of the "Million Dollar Quartet" for the Johnny Cash, Jerry Lee Lewis, and Carl Perkins' The Survivors *album. Were you pegged for that project from the beginning?*
BH: No, I didn't even know they were going to do it. They cut all the tracks over in Germany, and then just came back and handed us the tapes. There was about two hours worth of tape altogether. It was sort of disjointed. There were several different segments. Each one did a few songs, then Johnny [Cash] did a set by himself with the band, and then Jerry Lee [Lewis] and Carl Perkins came out.

R-e/p: Was it a tricky mix?
BH: Yeah, it was real loose musically. It was all unrehearsed, with vocals coming and going, people swaying back and forth away from the mike. We went for a live mix with a live feel, then updated with the computer if we needed to. That was the first time Rodney and I mixed to half-inch, which I'm glad we did because it helped salvage some of the sound from the live tapes.

R-e/p: Did you have to do more with reverb and EQ than you usually care to?
BH: Yes, I did do more of that. Some of the things sounded strange on the master. I've always found it hard to take tapes that somebody else has recorded, and get them to sound the way I want them.

R-e/p: I take it you'd had this experience before?
BH: I did pretty much the same thing with *Willie Nelson and Family Live*, which went on to sell quite a few copies. That was one of the hardest projects I've ever worked on. Showco had recorded the tapes. They had a 24-track that they took on the road with them, and just took a split off the PA and put it straight on the tracks, with no EQ on the tape. They didn't really watch the levels at all. Willie just handed me these tapes, and it was the most awful technical mess. There was so much tape hiss it was *unbelievable.*

I had to live with a lot of it, though I used noise gates wherever I could to keep it down. I had to add some high-end to some things, which brough out even more hiss. I just plowed through it. It took me a couple weeks of very long days.

R-e/p: Was Willie Nelson involved at all in the mixing?
BH: No. The good thing was that it was a straight performance all the way through, and Willie wanted to leave it in that order. The transitions were all natural. There was only one point where the tape ran out, and that was on "Red Headed Stranger." I had two shows from that night, so I cut right in the middle of the song and went to the other set. Luckily the drummer was playing a constant tempo on both songs. I didn't use the full song from the first set because Willie wanted everything from the second set. I never told him I did that, so it's news to him too if he should read this article.

R-e/p: How did you mike Willie's guitar?
BH: In the studio I'll take several feeds off his guitar. Sometimes he'll want to play through his amp, but not always. He has this strange stereo guitar pickup that some guy in Arkansas made for him. He goes into an old Baldwin guitar amp, with the highs going into one side, and the lows into another. We just put one mike in front of the amp. For an acoustic pick-up, I'll usually mike the outside, put [Sony] ECM-50 inside, and blend the two. I'll put them on two tracks, and in the mix use the one that sounds better, or blend them.

R-e/p: How is it different working with artist/producers Willie Nelson, Brian Ahern and Rodney Crowell?

"I don't think I'll ever produce my own albums again."
— Rodney Crowell

BH: Well, Brian's much more technically oriented than either Rodney or Willie. In fact Willie's not technical at all — he just trusts the people he works with. His whole organization is based on that kind of trust. So he gave me all the freedom in the world. Brian owned the studio, of course, and he was technically abreast of what was going on. A lot of the the time he's in playing on the tracks, but he'll get involved in episodes of knob twisting himself.

Rodney's one of the best producers I've worked with in terms of having an overall perception of everything that is going on. He knows enough about arrangements, pitch, engineering — he's able to sit there and not get too involved in any of the areas, and keep a good perspective on all that's going on.

R-e/p: Emmylou has been quoted as saying that "Rodney Crowell is expanding the parameters of country music." Would you care to — or dare to — elaborate on that?
BH: That's a good way of putting it. I think what Rodney and I are doing is taking all the options available — all the techniques used for making pop records and whatever else — and not sticking to the same old country formulas. That's one thing Rodney avoids. Everytime we come in to a session it seems we'll be trying something different, to see if we can get a new and better sound. It applies to the music as well. He's doing things that I don't hear on other country records. In fact I don't even know if you should call Rodney's records "country."
[At which point Rodney took a break from his pre-production work to rejoin the interview.]

R-e/p: There's something I wanted to ask you earlier in the interview, about the problems and blessings of being a "renaissance man" of progressive country music. You wear three hats: those of songwriter; recording artist; and producer. How do these roles complement each other, and how do they conflict?
RC: Being an artist myself makes me much more aware of what an artist feels like when working under a producer in the studio. I have a pretty good idea of how to do things and how not to throw somebody off. It's not that I use kid gloves; it's just that I'm a comrade, rather than an authority figure. I think an artist will clam up sometimes if he's confronted with that high-brow attitude.

As far as conflicts, I think the part that has suffered the most is the songwriting, just because of all the time I have to spend in the studio as a producer. And I feel that I have to have the songs before I can work as an artist, so there's actually a conflict there as well. I could wind up chasing my tail, so I try to concentrate on one thing at a time.

Also, I think my musicianship —my guitar playing — may have slipped as well. When I first started out producing, a lot of the time I'd be out there in the studio playing while we put down tracks. But whenever there were any questions to be answered, I'd have to get up and go in the control room to listen back. Now, I mostly stay in the control room with Brad, and work from there. So, all-in-all the production work has been a problem — especially since my basic impulse is still to write songs.

R-e/p: Is your primary goal when cutting an album to make the best record you possibly can? Or do you also consider things like commercial appeal and radio formats?
RC: Basically I go in to make the best album I can make, but always with an ear out for something you could hear on the radio. But the guiding principle has always been to do our best to make that particular artist sound like himself — or herself.

R-e/p: After you finish Sissy's album, do you have any more outside production projects scheduled?
RC: No, nothing firm right now. I'm putting that aside because I want to make another Rodney Crowell album early next year, and I'm trying to find somebody who can — and will — produce me. I'm looking forward to an opportunity to be out in the studio with the band, and making some music without having to worry if it's all going down on tape properly or not. ■■■

CARE AND REPAIR OF

MCI JH-SERIES TRANSPORTS

Part II . . .

Phase Lock Loop Boards, Interface Lamp Driver, Mother Board, Autolocator, Head Alignment

by Greg Hanks

In Part One of this article, published in the October 1982 issue of *R-e/p*, we looked at the overall components of the Transport System, Power Supply, and Analog Torque Board. In this, the conclusion, it's the turn of the Phase Lock Loop Boards, Interface Lamp Driver, Mother Board, Autolocator, and routine Head Alignment.

Phase Lock Loop Boards

On some of the older JH-100s there can be a sizable start-up loop thrown, especially at 30 IPS and at the end of the reel. This can be diminished greatly by three changes to PC.2500C0089:

1) Change C2 from 4.7 mFd/25V to 20 mfd/25V, which lengthens the accelerate pulse discharge time.

2) Change R39 from 27K to 39K, which slows down the capstan slew-up time.

3) Change R90 on the Analog Torque Board from 12K to 7K5, which increases the gain of the accelerate amplifier.

Together these changes minimize the loop. Sometimes, however, they also cause a bit of trouble, for instance:

A) Capstan takes a long time to come up to speed. To correct this simply decrease slightly the value of R39.

B) Capstan overshoots operating speed; to remedy increase slightly the value of R90.

If it proves difficult to adjust the PLL so that there are not intermittent flutterings in the transport, it may be necessary to adjust the tach pickups under the capstan motor. To perform this adjustment, which requires an oscilloscope, look at TP1 on the tach board, and adjust the cam on the top of the PCB so that the envelope of the tach signal is the most stable. Slow sweep rates on the scope are necessary to do this (50 milliseconds per division or so). If the signal cannot be made stable then it may be necessary to adjust the individual photocells. For proper cell operation they should be matched to within about 10% of each other; they are sold by MCI in matched pairs. To adjust them, ground the white lead from one photocell and note the envelope level at TP1 (Figure 1). Ground the other white lead (removing the ground from the first one), and note the level; the two signals should be the same. If they are not, the photocell with the lowest gain should be reset to match the other. To accomplish this, loosen th 4X40 Allen that holds the photocell, and repositions it so that its gain matches that of the other. When this procedure is complete remove the ground wire and readjust the cam on the top of the PC board.

It should be noted that the 74C00 CMOS IC on the tach board is biased as an amplifier, and is also used to double the tach frequency; every other pulse from this board will have small amounts of jitter.

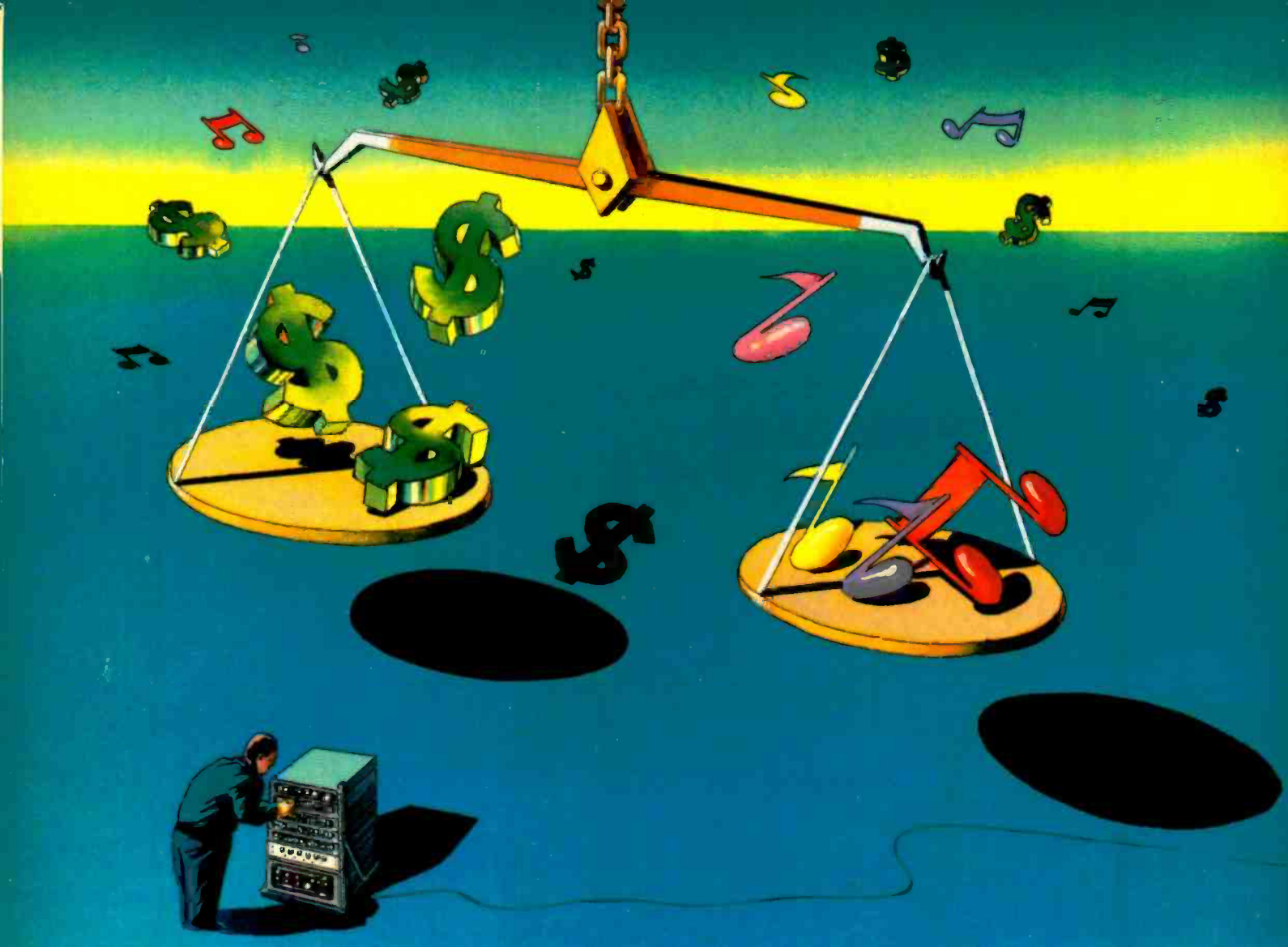
Maintenance: One of the common failure modes of the capstan assembly is bearing deterioration, caused by excessive amounts of alcohol or other cleaners running down the shaft and into the motor bearings. Once these bearings get real noisy, the only real cure is to send the motor back to MCI and have it rebuilt. To clean the shaft apply 409 or Fantastick to a Q-tip and rub this on the shaft; these alkali-base cleaners work very well on the ceramic — they also do wonders cleaning up the Woelke head.

Alignment: Older PLL boards should be aligned with the gain control so that TP3 has minimum jitter. The duty cycle is usually somewhere between 30 and 60%. When the "sweet spot" is found it is usually fairly sharp, but not very obvious, and sometimes difficult to spot.

Preferences: With old style PLLs, the "S" revision is the most desireable to have in your machine. It includes an

— The Author —

Greg Hanks, formerly service manager at Audiotechniques, Inc., recently formed New York Technical Support Ltd., which will specialize in studio installation and service. Previously chief engineer at Wally Heider Studios in Hollywood, Hanks is currently consulting on several film production and editing suites, as well as recording studios.



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CARE AND REPAIR OF JH-SERIES TRANSPORTS

extra drive transistor in the output circuit, along with a little different pattern layout that eliminates some radiation problems.

Pitfall: When utilizing an external drive signal for speed control on JH-100 type PLLs, ensure that the FET control is biased off; without this the VCO and External signal are shorted together. The result is in excessive flutter in the variable and external mode, and catastrophic PLL failure can result. To bias off the FET, tie the gate to ±15V through a 100K resistor. (Actually, this may be done by tying pins 2 and 3 together in the external Capstan Programming plug used with the synchronizer or external frequency source.)

Troubleshooting: The time for capstan re-building is at hand when the pinch roller pressure causes the shaft to shift its rotational axis off from being perpendicular to the top plate. To check for this, you must unplug the capstan, put the machine in play and check the parallelism of the shaft to the outgoing head guide. If it changes between pinch roller engaged and stop, then it is time for a new motor. (The motor may be repair/exchanged with any MCI dealer, or MCI proper.)

Interface Lamp Driver

This cute little guy is almost trouble-free, but provides a good place to find the various signals that are very handy for troubleshooting the rest of the deck electronics. The record momentary and the record hold pulses that go to the power supply are to be seen at P38, pin(s) 1 and 3 respectively; the tach generator pulses can be seen at P26 pin 4; and the MVC firing line appears at P22 pin 2. The MVC circuit, in fact, is one of the two most common failures in this sub-system. The MVC control was originally called U300; however, the company that made this hybrid touch decoder went out of business. The replacement device was a long thin hybrid circuit called IC 3000 (but the documentation designations never changed); alas, this supplier also bit the dust. Now, however, there is a new circuit that uses industry standard JEDEC-type parts that can be repaired, and with parts available through many sources. The only problems that we have encountered with this circuit, whether it be the U300, IC 3000, or the new circuit, is the connection to the motherboard of the twisted pair that goes to the joystick contactor — sometimes the Molex is put on mis-pinned.

Another problem involves the use of 748 op-amps instead of 741s, a device change that sometimes results in oscillation because of the lack of compensation required for the 748. A change to the 741-type device cures this difficulty.

If when you turn your machine on it goes into the MVC mode, or locks into this mode intermittently, we have a fix. On JH-114s introduced in early 1980, there was a ground trace left off the circuit board. To repair this tie a jumper from the upper left-hand corner pin to the top of the upper left-hand resistor.

Mother Board

Strange as it seems, there are a significant number of troubles that can be associated with this PCB. Intermittent tensions, odd servo lock, and intermittent wind and locate functions can usually be traced to cold solder joints on the Molex pins, sometimes because the solder just won't take to them. If you look very closely at the connection between the mother board and the pins, sometimes you will see what looks like a cone of solder building up around the pin, but a thin dark ring around the pin identifies that the solder didn't form a weld with the pin. Bad connections can also be identified by wiggling the pin and seeing if it moves at the board, within the solder connection. When defective joints are found, the best cure has been to file the pin at the solder line, removing the old solder with wick or solder sucker and re-soldering the connection. When all of the suspect connections have been re-done, a thorough cleaning with a flux remover followed by another re-inspection is in order.

Modification: When using the machine with an A/L-3 or the JH-45 Autolock, R1 must be jumpered out. When using the JH-45 Autolock, the minus 15V feed to the transport switches — both on the transport and the remote — must be isolated by a 4K7 resistor, or greater.

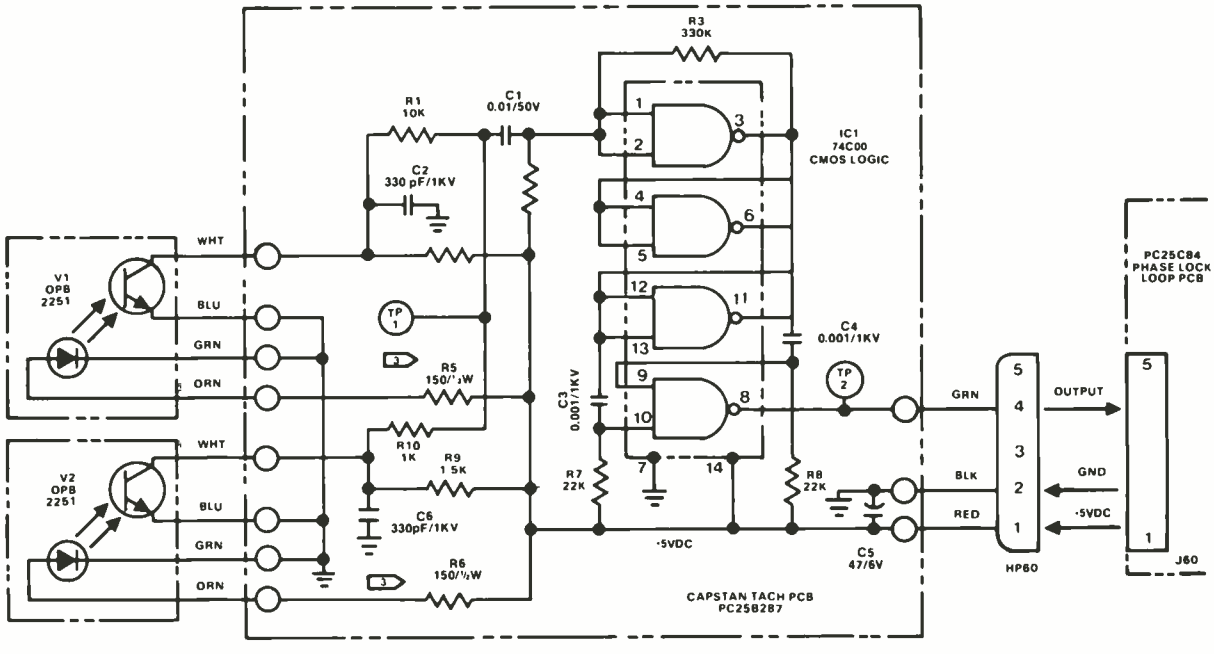
Troubleshooting: When the play mode will not initiate, but both the stop light and the play light are illuminated, and the tape is at a complete stop, look for:

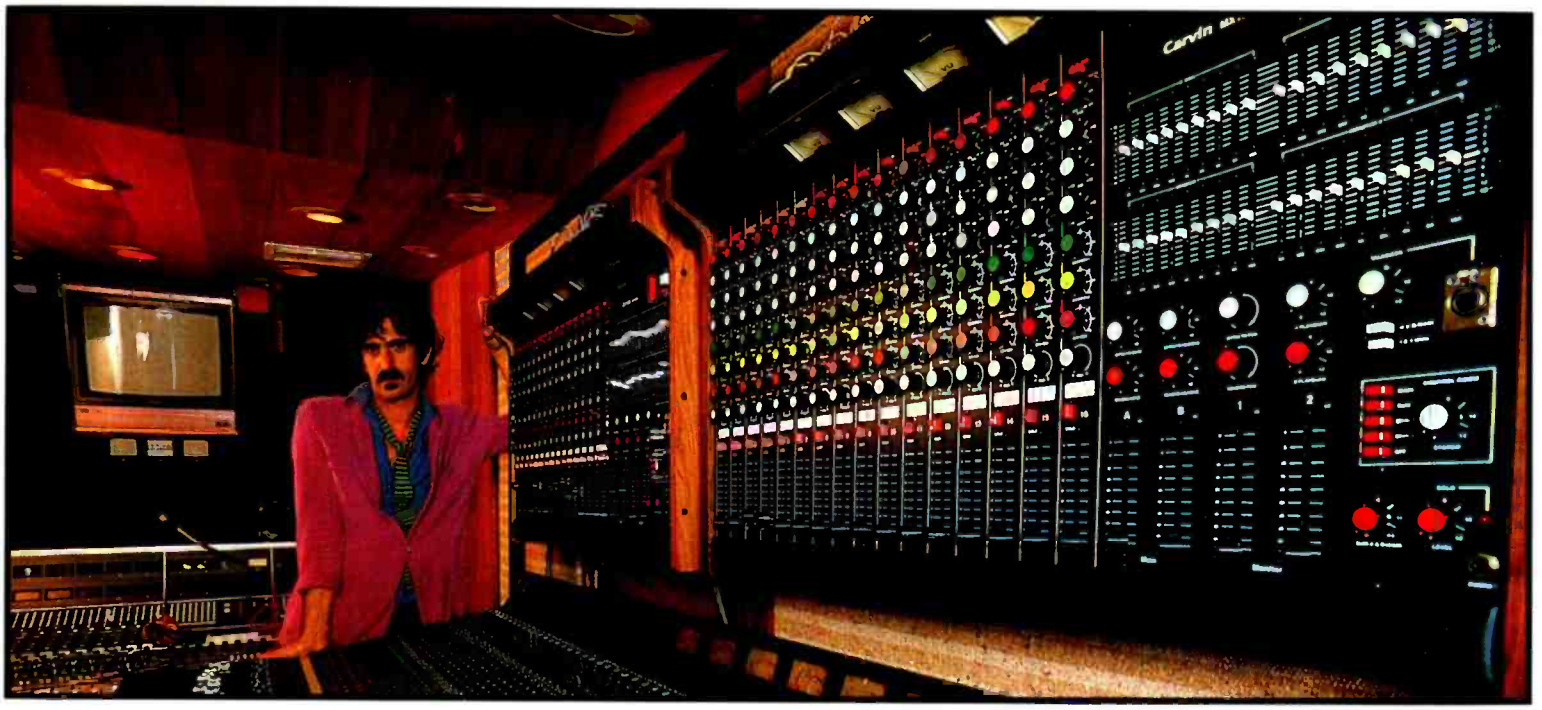
- Lifters not fully retracted (microswitch senses this)
- Motion sense line hung up
- Tape not fully stopped, but creeping sufficiently to hang up a motion sense line

Adjustment: Head shield adjustment on the JH-100 and JH-114 is by means of a brass screw on the rotary solenoid. Sometimes this screw (the bottom one of three) is replaced with a screw like the other two. To properly adjust the head gate, set the screw so that when the shield button is depressed, the shield doesn't bounce when it hits the bottom. Once adjusted, hold the screw's position while tightening up the locking bolt. Double check the operation because locking the bolt often changes the adjustment.

. . continued on page 41

FIGURE 1: WIRING OF CAPSTAN TACH PCB, WITH CONNECTIONS TO TACH PHOTOCELLS





6 Ch CX601 Mono $599 with 200 watt amp (list $1195)
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6 Ch SX602 Stereo $649. Add $150 for 200W stereo amp (list $1295)
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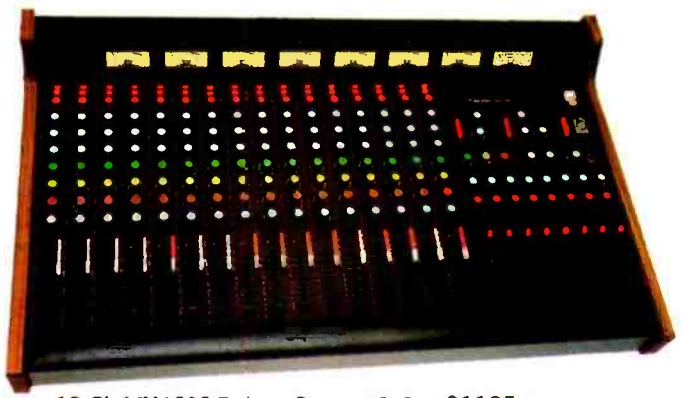

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The Vocoder circuit consists of two major sections: the analyzing section and the synthesizing section. The analyzing section breaks down the voice (or program) input to determine its frequency or content at any given instant, then re-assembles this sound in the synthesizer section using the instrument (or carrier) input as a basic source of building material. The program or voice input is analyzed by passing it through a set of eleven Voice Character Frequency filters.

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Instrument Input (carrier)	1/4" phone jack 100 kΩ 0 dBm max
Guitar Input (carrier)	1/4" phone jack 100 kΩ
Guitar Amp Output	1/4" phone jack 5 kΩ
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— *continued from page 38 . . .*

After considerable use, the Teflon block that guides the shield becomes mired with crud. To determine if this is happening, back off the air dash pot a bit. With no dash pot tension, the gate should move very briskly. If it doesn't try cleaning the bar, and the top and bottom of the block with alcohol. After a thorough cleaning lubricate the bar and the block with something like Tri-Flow, which is a Teflon-based lubricant. Then adjust the shield with the dash pot for brisk, but not violent, return.

The tape lifter is also oftentimes a little out of spec. The adjustments for this are:

- Non-engaged rest position
- Engaged extension position
- Rate of outward travel

These adjustments are located under the deck in the form of screws that retain eccentric disks with rubber peri-

ADDING A FLASHING VARISPEED INDICATOR TO MCI JH-110A SERIES TAPE MACHINES . . . . . *by Roman Olearczuk*

Imagine you have just finished a tricky mix, and now are ready for some fancy tape editing. Horror of horrors, you've just realized that the VSO switch was still on, and everything you've just patiently recorded is at the wrong tape speed! Apparently, this frustrating experience is quite a common occurence, since MCI didn't include any kind of indicator within the JH-110A Series tape machine remote control unit to show the tape operator when varispeed mode has been selected.

The circuit addition described here solves such an ommission in the following ways. With the varispeed mode engaged, the Play indicator (on the machine and the remote box) flashes at a rate of 1 Hz whenever the transport is in the Play mode. In the Stop mode, the Stop indicator flashes instead. With this design an engineer will always know if the tape recorder is in a varispeed mode.

The original partial schematic of the JH-110A's control logic board is shown in Figure 1. Only the ICs and board I/O pins directly involved in this modification are referenced here. An effort was made to utilize the spare IC gates already present on this board, but the number required for a correct logic interface exceeded those that were available. Figure 2, then, shows the external circuit that was designed to provide the features mentioned above.

The flashing circuit basically is added in a logical AND fashion to the existing Play and Stop light commands. The circuit operation is as follows. A Signetics NE555 (or similar device) is connected to run as a multivibrator, by triggering itself for an astable operation. The frequency is set at 1 Hz (33% duty cycle) through the combination of 470 kohm resistors and a 1 mFd capacitor across pins 6 and 7 of the IC. The timer is inhibited whenever pin 4 (Reset) has a voltage of 0.4 VDC, or lower. The Variable Not-Enable flag that originates from the Speed Reference Switch (refer to the JH-110A Manual Interconnect Harness Schematic) is used to activate the oscillator through a logic interface at the Reset pin.

When varispeed mode is engaged, Variable Not-Enable goes Lo (Gnd). This action turns on the 2N3906 PNP transistor, which pulls pin 4 high, enabling the timer. The output pulses on pin 3 drive a 2N3904 NPN transistor that interconnects with the 7408 AND gates. The pin 3 output remains in a low state whenever the timer is in Reset. This logic permits the normal Play and Stop light commands to occur unimpeded through the AND gates.

When the VSO is switched in, then the output pulses appear at the inputs of both AND gates. The appropriate gate will then be activated dependent upon the transport mode selected. As expected, the Rewind and Fast Forward modes deselect the flashing indicator from either Play or Stop lamps.

The circuit can be constructed compactly on a small pre-drilled circuit board. A convenient mounting location is to the right of the transport control logic card. As Figure 2 illustrates, two PCB traces need to be cut and a wire brought out to a spare socket pin, in addition to the expected circuit interface wiring.

The first cut trace occurs at the trace leaving the output pin 4 of IC 11. An incision should be made somewhere on the trace after this point, but before the trace reaches P33-5. Here a wire is added from IC 11 pin 4 to P19-9 (a spare pin).

The next incision occurs on the Transport Mother Board. The PCB trace leaving J33-2 splits off to three pins, J3-3, J14-5, and J27-4, on the same mother board. Only the trace that goes to J27-4 (the Lamp Driver Board) should be cut free from this interconnect —we only want to flash the Stop lamp, and not the torque motors! The rest of the wiring is straightforward. For neatness ribbon cable is recommended. ■■■

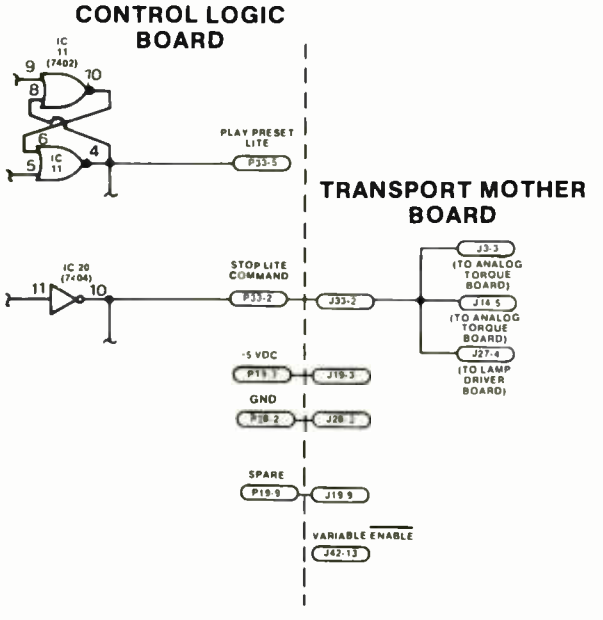


Figure 1 *(above)*: **Partial schematic of JH-110A Control Logic Board and Transport Mother Board, prior to modification.**

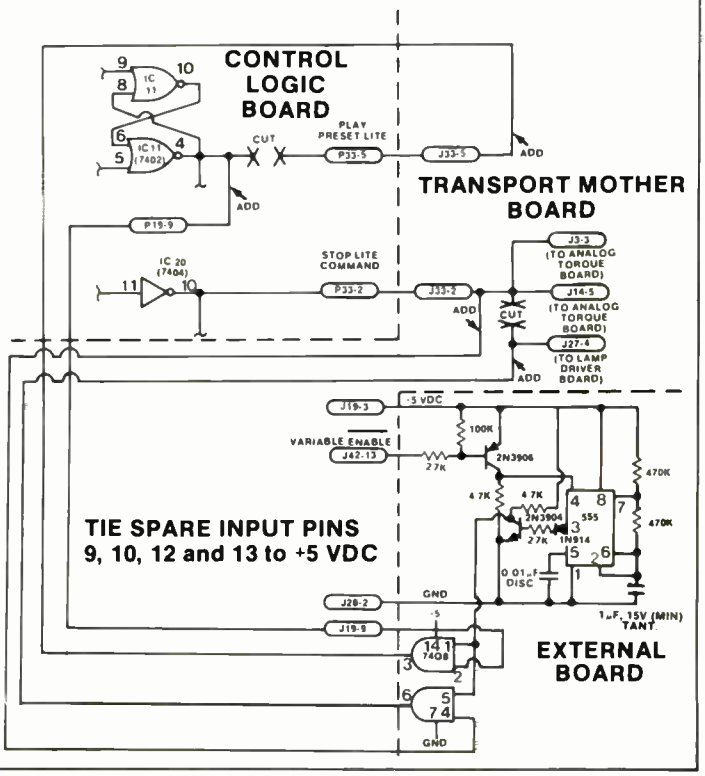


Figure 2 *(right)*: **External flashing circuit interface, circuit modifications, and necessary connections.**

CARE AND REPAIR OF JH-SERIES TRANSPORTS

pheries, and a dash pot located behind the assembly. The inner limit set should be about 0.125 inches behind the tape, while the outer limit should be set so that when in fast wind the tape is not extended beyond the guides — most specifically the guide next to the capstan. The dash pot should be set so that the lifter does not slap the tape, but engages it quickly.

Another adjustment that is rarely "on the nuts" is the azimuth and zenith of the tape lifter; this is an adjustment and mounting method combined. There are three threaded shafts that extend down from the bottom of the transport, and hold the rotary solenoid actuating the lifter. The solenoid is mounted on a plate that is secured in position by a nut and lock washer on the bottom, plus a jam nut on the top of the plate and threaded to the shafts.

To adjust the lifter, you need a third and sometimes fouth hand —plus a machinist's square and a two-inch gauge block. The first adjustment to perform is lifter azimuth. To do this, block the tape-break sensor, tilt the deck up to the first stop, and put the machine in fast wind with no tape threaded. Position the machinist's square to the left of the lifter, and use the two front nuts to set the lifter to perpendicular. Once this is done, use the gauge block to measure the parallelism of the lifter to the erase head and the outgoing head guide, and the rear nut to set lifter zenith. When these adjustments are done properly, the fast wind tape pack can improve considerably.

The smoothness of starting is affected by the rest position of the pinch roller. This can be adjusted on the solenoid itself, by changing the position of the solenoid plunger retainer arm. We suggest that the pinch roller should be as close to the capstan as possible, while still being able to easily thread the tape; this dimension is somewhere between 0.25 and 0.125 inches.

To adjust the newer Analog Torque Boards, the order and procedure are as follows:

1) *Set off-set nulls.* With tape loaded and no tape motion, attach a scope probe to TP1 or 2, picking up ground at the large trace that ties the two electrolytic caps together. Set the off-set null pot so that the DC at these test points turns into a 50% duty cycle semi-square wave. (While making these measurements, if the tape is creeping have someone hold it still.)

2) *Idle.* Remove tape from the transport, block the tape sensor, and set the idle adjustment so that there is 1 volt at the test point(s). For the JH-100 and -114, set this to 1.5 volts.

3) *Supply tension.* Assuming that the dancer arm settings have not been changed, set the hold back tension for a dancer arm position in the center of its travel. If the dancer arm position has been changed it can be checked in one of two ways: with a "fish-scale" pulling the dancer arm to the center of its travel, and noting the tension; or by using a Tentelometer. Check the tension of the tape at the incoming side when it is centered; the tension should be around 8 ounces. My preference is to set the dancer arm 1 ounce above the minimum tension required for consistent head contact and phase stability. Our settings usually end up at around 6.5 to 7 ounces.

4) *Take up tension.* This setting is done after the supply reel because it depends on correct supply tension for its performance. Initiate play and allow a few seconds for the play boost to subside, then push the pinch roller away from the capstan. When set properly tape speed will remain constant.

Older Analog Torque Boards pose one or two problems:

1) *Off-set nulls.* Same as above.

2) *50/60 Hz.* To set this, hook up a scope to the banana jacks next to the side being adjusted. Put the machine in rewind (for the supply reel) and adjust the pot for minimum crossover distortion in the waveform. CAUTION!! The case of the scope is at line potential! For the take up reel, put the machine in fast forward and do the same as before.

3) *Idle.* Same as above for newer boards.

4) *Empty reel/full reel.* The method I use is as follows: Using a calibrated Tentelometer with full reel of tape on the supply side, set the full reel tension pot for 7.5 ounces at the incoming side of the head stack. Roll to the end of the reel and set the empty reel tension to 7.5 ounces. At the same time, set the full reel pot on the take up side for a tension that allows for correct speed. Again, on the supply side reset the full reel tension to 7.5 ounces. Push the puck away, and set the empty reel of the take up side so the speed remains constant. Roll the tape so that the supply side is almost empty, and set the empty reel tension at 7.5 ounces. Push the puck away and set the full reel tension pot on the take up side so that the speed remains constant. *Note:* when starting the above procedure set the initial tensions using the empty reel/ full reel test switch to get the pots in a good starting position.

An alternate procedure that seems promising is to set the tensions using predominatly electronic measurement means:

1) Thread a full roll of tape, put the empty/full operate switch in the full position, and set the full reel pot on the supply for 7.5 ounces of tension at the incoming guide to the heads. Note the voltage at pin 6 of IC24. Switch back to operate.

2) Roll the tape so that there is a full pack on the supply reel. Initiate play and measure the voltage at the test points, noting the full reel tach output for the tape up side. Roll the tape so that the full pack is on the take up reel. Again

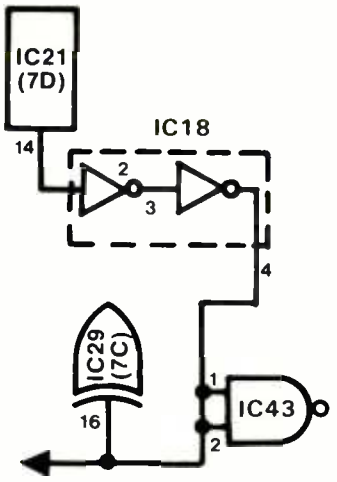


FIGURE 2: BUFFER ADDED TO AUTOLOCATOR LOGIC BOARD OUTPUT

note the voltages at the test points. These will be empty reel supply, and full reel take up tach outputs.

3) Set the test switch to empty reel, and note the voltage at these test points. Devise a method of making these voltages the same as for the actual reel-tach output; this can be accomplished by inserting a multiturn pot in place of the resistor networks currently there.

4) Put the test switch to the empty reel position, and set the empty reel pot on the supply side to give the same voltage as measured in step #1. Now place the machine in operate mode, and set the tension at the take up reel to give constant speed with the puck pushed away. At this time measure the voltage at pin 6 of IC14, and note it.

5) After achieving the same voltages with the test switch at the test points that the corresponding tach in the operate position yields, it is possible to set the tensions without having to go back and forth with a reel of tape and the Tentelometer, but simply to use the empty reel/full reel test switch to bounce back and forth. Care must be taken to ensure that the voltages that appear on the test points are within 5% of the actual tach output values, otherwise this procedure is useless. *Good Luck!*

Autolocator

There are a number of changes that should be made to an Autolocator II that has intermittent problems:

Field Service Bulletin 712 — Install grounding bus from the ground pin on the Autolocator logic board to pins 12 and 13 of IC31, 32, 33, and 34. Also add a pair of 1 mFd Tantelum capacitors from each +5V Molex pin to ground.

FSB 711 — Add a 100 pf cap from pin 14 to pin 12 of IC21.

FSB 703 — Replace IC21, 22, 23, 24, 31, 32, 33, and 34 to *ceramic* 7483J.

FSB 702 — Install buffer on output of IC21 (Figure 2).

This one has no Field Service Bulletin attached, but is from my experiences: closely inspect all IC connections on the soldered side of the PC board for leads bent over traces that run alongside the pins. Trim bent leads carefully so that overlaps do not take place.

For problems with multiple entry of numbers with a single key stroke on the Autolocator, a de-bounce board modification kit is available. A slight problem exists with the de-bounce board, however, that requires modification as well: install a 300-ohm "R" and a 0.47 mFd capacitor as described in FSB 711.

Sometimes the tape counter will seem to gain counts, or lose location, a problem that almost always is caused by the tach pulse generator becoming smooth with use. To verify that this is the problem, attach an oscilloscope to the interface lamp driver board at P26 pin 3, and put the machine in fast wind. Pulses at this location should become closer and closer as the velocity increases. At terminal velocity, if the pulses widen out, then the roller needs replacing; factory repair/exchanges are offered by MCI. The tach pulse generator is the outgoing roller guide.

If, after all of the above fixes have been tried, intermittent operation still plagues the A/L II, remove all of the ICs and wash the sockets with a stream of "Flux Remover." Re-insert the ICs after any residue has been removed with Freon.

Heads

Over a fairly long period of time, MCI has been using AMC record and playback heads. While these units perform well, a large number evidence incorrect crown centering. (What this means in a nutshell is that the gap is not centered on the center of the headface peak, which leads to an asymmetrical wear pattern about the gap.) As a result, higher tape tensions than would otherwise be required are necessary to maintain sufficient tape-to-head contact. The only solution to this problem is to have the heads recontoured. (JRF, Inc. of Hopatcong, New Jersey, is the only facility I know of that can handle such work.) This phenomena has not been noted with the JH-24 type head assembly.

EQ Set-Up: Newer machines have a potentiometer located on the repro board just behind the head transformer, which is used to "critically damp" the secondary of the head/primary/secondary input network. Properly set with a flux loop and a square wave generator, this pot would be set for minimum overshoot or droop. Lacking a flux loop or equalizing network necessary for its use, a practical way to set the pot is to make the playback flat at 16 kHz.

It is necessary to set up the machine first using 500 Hz as level reference, 10 kHz as the HF EQ reference, and use the damping pot for 16 kHz. This control has a lot of interaction with the HF EQ at 10 kHz, so a lot of bouncing back and forth is necessary.

On older JH-100s and -114s a fixed resistor serves this purpose. The easiest

way to determine what value it should be is to replace the "R" with a 2 megohm multiturn pot. Adjust the pot as described in the above paragraph, measure the value obtained, and insert the appropriate fixed value. (This resistor should be of the metal-film type.)

Wrap Set-Up: To perform wrap adjustment on a new machine, or on heads that have just been relapped, run a full reel of tape over the heads and adjust the wrap for a symmetrical scrub pattern on the gap. Run this same reel of tape over the heads two or three times. The pole tips of the heads are often slightly "work hardened" during either manufacture or lapping. Running at least half to a full hour of tape across the heads is necessary to remove the very thin crust of work-hardened material from the gap. Otherwise this material changes the permeability of the gap area, and often results in an incorrect initial wrap adjustment.

Once there is a slight wear pattern on the heads, the wrap is then turned on the playback head while reproducing a 16 kHz tone, to obtain maximum output. The record head is adjusted in the same way, but reproducing in the sync mode. There is about 0.5 dB interaction between the record and the erase heads, so some re-adjustment is necessary — the erase head should be done first.

To adjust the erase wrap and azimuth, plug the head connectors into the record head position on the back of the headstack. Reproducing a 500 Hz tone (or something convenient in the midrange), set the head so that the outputs are symmetrical. Because the erase head uses a staggered track pattern, the setting should be one where the even tracks are the same distance away from their peak as the odd tracks. The azimuth should be measured (on machines with adjustable azimuth!) using the summing method, and even or odd tracks only.

Once the erase head is set return the head connectors to their proper place and do the record and playback. The record head only interacts with the erase setting about 0.75 dB on erase head output, so the latter, once set, need not be touched again.

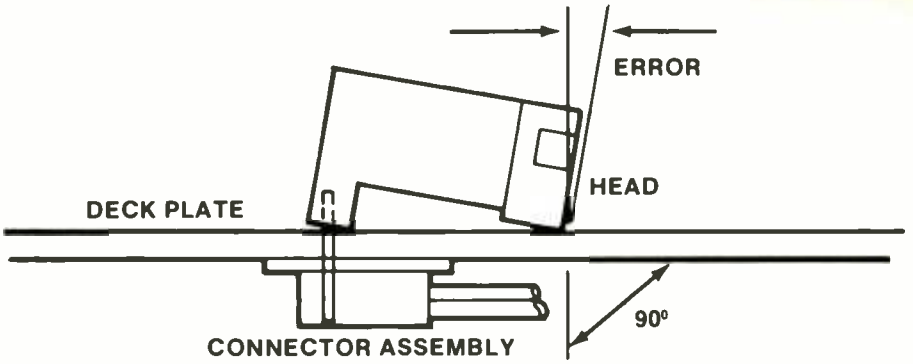


Figure 3: Correct seating of head connector is essential before making further adjustments.

Head Set-Up: Before attempting any head adjustments on the JH-110 recorder ensure that the head connector is not fully bottomed in the connector, and thereby spacing the head stack off the deck. The easiest way to determine if this presents a problem is to simply back off the screws that retain the head plug assembly on the bottom of the deck, and check if it pushes away from the deck plate. If it does, space it off of the deck with washers. Position the plug bracket just a skosh farther away from the deck then it sits naturally (Figure 3).

Update: The 30 IPS playback low-end response on most JH-100s and older - 114s often leaves a lot to be desired. This can be remedied to a large extent by the replacement of R103 (1.3 megohm) with a 0.0018 mFd capacitor. To protect yourself, it is also advisable to put a 200 to 400 kohm resistor in series with this new cap, to limit the range of the pot.

Pitfall: Every so often a tape comes in from the outside world that has far too much signal at 100 Hz. Lining up the machine with the outside tones you find that the low-end EQ has to be turned all the way off. When this is done, however, what often goes unnoticed is the phenomenon of broadband level coming down about 2 dB, which results in a net low-frequency gain. After a "line-up" in this manner, the level is often brought up in subsequent re-alignment, and a complaint of not being able to get 100 Hz down far enough arises. A cure is to inset the series resistor mentioned in the preceding modification, which prevents the 0.0018 mFd cap from being simply parallelled with the 0.0033 mFd capacitor.

Bias: When carrying out bias alignments you've got to be careful of a number of things. First of all, when making any erase peak or bias trap settings, it is imperative that either a low capacitance (10X) probe or an instrument be used that has input characteristics of 1 megohm or more, shunted by 10 pf or less. The reason is that by attaching a set of test leads to the trimmer cap, we in essence have placed a capacitor equal to the instrument/probe between the measured point and ground. It is amazing what a difference 100 pf to ground makes between measured and actual performance!

When setting the erase peak on non-QUIOR erase boards using an old

Woelke head — this is the one with the checkerboard pattern, but without the guard bands between tracks — there are some insidious problems that can be encountered. Most of the time, the original erase board can "ring up" with Woelke heads to 170 to 210 volts, peak-to-peak. This amount of erase current provides an excellent level or erasure, so long as nothing in the circuit is in saturation. However, the erasure is so good that the adjacent erase head track can couple to the energized channel, and start partially wiping the track!

The easiest way to test for this is to record a 10 to 16 kHz tone at zero VU. Rewind the tape to the head of the tone, and erase only one track. Observe the playback on the adjacent track, and note whether and by how much it is attenuated. Repeat this test on the same track and see if the playback on the adjacent track is further reduced in level. If further attenuation is noted it will be necessary to back down the erase level until such "cross-erasure" ceases.

Something to note is that the phenomenon of the first pass, with any usable erase level, will cause an attenuation of somewhere between 0.25 and 0.5 dB of loss. I think that this is due, in part, to a differential in short-and long-term tape retentivity. It is the second and subsequent passes that are of concern. The process takes some time to accomplish the first time around, so it is advisable to log the resulting peak-to-peak erase voltage for future alignment.

Erase Set-Up: With newer Woelke heads (the ones with the guard band) the ideal place to be with the erase peak adjustment is centered on the peak, and below saturation. However, in the real world saturation usually occurs right near the peak, so a good place to set the trimmer is about 5 volts below saturation.

Modification: On a number of late vintage JH-114 recorders, the erase peak cannot be adjusted to provide sufficient erasure without going into saturation. The reason for this is the cans enclosing the coils. The vendor of these cans changed the winding diameter without proper OEM notification, so a number of these units hit the field; replacement cans can be had from MCI.

Repair: After performing the erase alignment procedures on a 24-track machine, it was found that a number of tracks would not bias properly —the tracks would not back down to a readable peak. After much fussing around it was discovered that the pots, after 3 or 4 years of alignment, had decreased to almost half of their original value.

Master Bias Set-Up: One procedure that needs reviewing is the JH-110 and -110A master bias bus level adjustment. This pot is, in effect, a master *erase* level adjustment. To set the pot, first peak the erase capacitor, then adjust the bus level to a point just below clipping. The places to look for clipping are:

1) TP1 on the bias card(s).

2) The other side of the erase peak cap (look for waveform distortion).

3) The output of the first buffer amp on the bias PCB.

Clipping at any of these locations, on any track, will be the cause of "rocks" and other bias noises. The idea is to maximize the erase currents but ensure that none of the circuits clip. The manual gives values for the master bus level that vary with the date of the manual's printing. Given the importance of this step, it is this author's humble opinion that the figures given in the manuals must be disposed of.

* * *

So there you have it. The above hints and tips are the result of many, many years working with all "breeds" of MCI machines, and more hours of trial and error fixing than I really care to think about. MCI stereo and multitrack decks are, without doubt, the most popular brand of tape machine in the United States (probably the world for that matter), and there's no denying that with routine love and attention they can provide years of excellent service to a studio. The information provided in this article is intended to help those who are just beginning to get to know the JH Series of tape transports, or who have been working with them for a while, but want to know more about keeping them in fine fettle. I trust that the information proves useful to you all.

■■■

DIGITAL MASTERING AND EDITING IN THE REAL WORLD

The JVC System in Operation At Masterfonics, Nashville

by Paul D. Lehrman

When does a new technology come of age? One possible answer is when its use is no longer predicated on its novelty, or the sales value of its name, but instead when it gains acceptance because it does the job well — preferably better than the techniques that preceded it. Such is the case at Masterfonics, one of Nashville's busiest disk mastering facilities, where since July, 1981, a complete JVC two-track digital editing and mastering system has been in almost constant use.

Masterfonics is presided over by chief engineer and president Glenn Meadows, and his new partner, Ken Perry, who defected from Capitol Records in Hollywood last spring. The facility contains two cutting suites, one with Neumann equipment, and the other with a Westrex system. Both rooms are busy enough to work two shifts. In between the two mastering rooms is the digital editing facility, and plans are currently in the works for a third cutting room in an unused space at the rear of the building.

Masterfonics is the choice of many of Nashville's top producers and artists, although it is completely free of any long-term label commitments. "The future of our business is totally dependent on the quality of the last record we put out," Glenn Meadows offers. "When people see our name on the back of a record that sounds good or sells well, *that's* what makes them decide to try us."

And a lot of people have made that decision. Meadows proudly pointed to the Billboard Top 100 Country Chart for mid-June of this year — where Masterfonics had a hand in 41 of the charted singles. With Perry's arrival, the company hopes to expand its sphere of influence even further, moving more solidly into pop, jazz, and even classical work.

The Digital Decision

The acquisition of the JVC digital system (which operates under a separate company, Master Technologies, within Masterfonics) is another way of making the firm more responsive to, and useful for, its clientele. Meadows explains how it came about that his company should be the first to have a permanent digital editing facility in Music City:

"About three years ago, we had the first half-inch, stereo Ampex ATR-102 in the country, but because that deck can't easily be fitted with a [second] preview head, we also bought the Ampex ADD-1 preview delay. That was out first experience with digital, and we felt very good about the way it sounded. Early last year, Larry Boden of JVC [Cutting Center, Hollywood] — who has been a good friend for years — contacted us and said he had 10 days or so free on his new digital recording and editing system. And he wanted to bring it out and let someone do an entire project on it, gratis, for demo purposes.

We had been working very closely with Jimmy Bowen, who produces for Elektra, and who's very quality-conscious. He was our main half-inch user, and was very interested in trying digital. Larry's schedule happened to fit into Jimmy's mixing schedule, so we brought the system into Sound Stage Studios here in town, and mixed on it for 12 days.

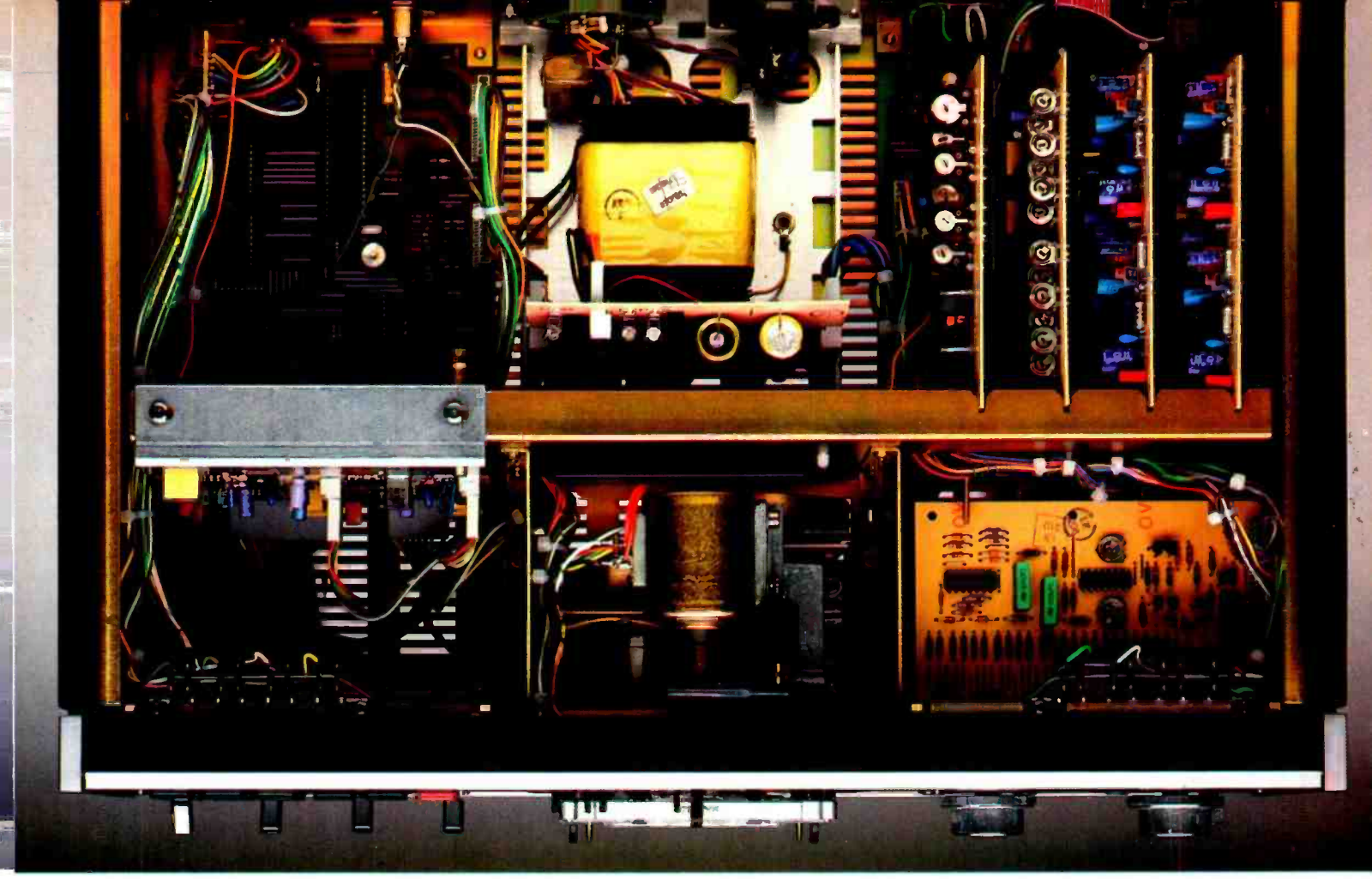
"We set up a quarter-inch deck, the half-inch, and the digital, and did a rough, 'no-fader-moving' mix into all of them simultaneously. Then we lined them all up, along with the 24-track, as close in sync as possible, and played them back, letting everyone push buttons.

"The difference between the 24-track and the quarter-inch was startling. We'd been using the half-inch for a while, so we knew what to expect and, sure enough, the improvement was *ridiculous*. What surprised Jimmy was that the digital was that much better again than the half-inch. At one point he thought he was listening to the 24-track, and he reached out and hit the stop button on the machine, but the music, which was actually the digital playback, kept on going.

"But what really sold Jimmy was when he edited on the system. There was no automation for the mix, and there were some very complex vocal balances that had to be maintained. We ended up doing a lot of one-line, or even one-word, edits from different mixes. He was just blown away by how well they worked. — *continued overleaf . . .*

Neumann cutting lathes in tandem operation.

Photography by Paul Lehrman



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Engineering Fundamentals–The B710 MKII utilizes a die-cast aluminum alloy chassis as a solid base for the four direct-drive motors. (Rubber belts, pulleys, and clutches–which can degrade long-term performance–will not be found here.) The two Hall-effect capstan motors are quartz regulated for exact speed and synchronization; and the two DC spooling motors are microprocessor controlled for constant-speed fast wind, controlled tape tension, and gentle electronic braking.

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Features in Summary–Three head design • Automatic start-of-oxide cueing • Dolby B and C NR • Modular plug-in PC boards • Mic/line mixing • Internal timer for programmable start/stop in record or play • Precise 4-digit LED counter • Headphone volume control • Optional remote • Full microprocessor control of transport modes and audio switching • Adapts to external computer control.

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JVC Digital Mastering System, comprising two BP-90 16-bit processors, CR-8200 U-Matic recorders, and AE-90 editor.

"Boden's system, which he calls 'Processor #1,' is a hand-wired job he takes all over the world to show. We ordered a full-production model in May, and it arrived in July [1981]. In the meantime, JVC was very kind to us and lent us their editor when Jimmy needed it to do singles."

Since the arrival of the system, Master Technologies has done more than 20 album projects on it, and it is being used to nearly 75% of capacity. The company owns two JVC BP-90 two-channel 16-bit digital processors, and three U-Matic videocassette decks: two CR-8200s, which are microprocessor-controlled for interfacing with the editor; while the other, a GR-6600, can be sent out to other studios for recording dates, or even to live video/film shoots.

"We've been real happy with it," says Meadows. "There has been no serious downtime, and no field failures at all. We don't even send a technician along when it goes out. I'll just show the studio engineer how to use it, and how to adjust the offsets for minimum flutter and quantizing noise, and tell him to call me if there's any trouble."

Editing Flexibility

There are a lot of features that Master Technologies likes about the JVC system. One is that the timecode, which follows JVC's own format, is placed in the vertical interval of the video signal, and doesn't have to be prestriped on the audio tracks, as with SMPTE timecode. Each line within the frame has its own address, and this allows for tight resolution. "We can get down to 180 microseconds," Meadows explains, "which is about the width of a razor blade at 30 IPS. "When an edit is made in the middle of a frame, the editing unit reshuffles the data, and fills out the frames."

"I suppose it's possible to do a simple edit faster with a razor blade," Meadows considers, "but if you blow it, you have to try to put the pieces back together. Here, there's no physical cutting or generation loss, and the original is always intact. If the next day you decide you don't like your edit, you can do it over again."

Editing on the JVC Series 90 System is done entirely by ear. The AE-90 editing unit's digital memory holds 12 seconds of signal, six from each of two sources — the cut-in and cut-out points of the proposed digital edit — and the exact edit point is found by turning a control dial which scans the memory. It works, and sounds, just like cueing a reel of analog tape.

"We have video monitors," Meadows says, "but all they show is bits, which don't represent the sound at all. We do use them to time the spreads between the cuts on a disk master — they won't show you waveforms, but they do show silence."

The JVC AE-90 Digital Editor also allows for adjusting offset levels digitally, previewing edits, and four selectable (10, 20, 30, or 40) millisecond crossfade times. The system is made to work with just two VCRs — an original and a master — and the procedure is similar to video assembly editing. Because only two video transports are in use, tapes have to be shuttled or changed to set up edit points between separate takes, but since everything is timecoded, the procedure is virtually automatic. Once the edit "windows" have been read into the unit's memory, however, there is no need to shuttle tape.

"A system with three machines would probably be a little faster," Meadows concedes, "but this editor is not as foreboding as [other systems] which use three machines, and it is much easier to learn."

Teaching the operation of the system is very simple, as Meadows demonstrates during *R-e/p's* visit, but explaining the best way to set up tapes when they leave the mixing studio is a little more complex.

"When we send the system out," he explains, "we always tell the engineer to put an audible slate on the digital tape, and to log the code numbers as they are displayed on the tape deck. People sometimes don't bother, and just jot down the video tape-counter numbers, but an audible slate makes it much easier to keep track of multiple takes. Especially when the engineer makes analog cassette copies for people to take with them, so that they can listen at home and decide what takes to use. And if they give us a list of the code numbers of the various takes, it speeds up the editing process immensely — all we have to do is give the numbers to whoever is editing, instead of having him search around for the various takes and cues."

Audible Effects

Being something of a pioneer in the field, Meadows has a lot to say about the possibilities and problems of digital recording. "I know theoretically that there are problems at low levels," he says, "but we've never been able to hear anything, either on fades or reverb trails."

"At Capitol I did a lot of classical music," Ken Perry adds, "and even on low-level flute solos, for example, we never had any distortion problems. Nor did we experience any of the physiological stress that some people claim to have found." [Capitol's classical label, Angel, has made extensive use of the JVC system to edit digital master tapes — *Ed.*]

"Going to 100 kHz sampling, or a longer word length, wouldn't hurt," Meadows admits, "but, as Larry Boden says, 'Who are we making records for —people or Doberman Pinschers?' Some folks are screaming now about the loss of quality [caused by] dropping from 50 kHz sampling to the new industry-standard 48 kHz, but the JVC is 44.1, and when we did a symposium — what the press called a 'shoot-out' — with the various digital systems in Nashville not long ago, the JVC stacked up very well, even against the 50 kHz machines.

"I think 90 percent of what people are complaining about with digital has to do with the analog portions of the equipment, or the filters. The various systems *do* sound different, although theoretically they shouldn't. The differences are not how they handle numbers

— that technology is very advanced, and doesn't affect the sound at all. It's in the execution of the line inputs and outputs, the cheapest parts of the machines.

"It just doesn't make sense to build a recorder that can go from DC to 20 kHz, flat, with 0.02% or less distortion, and put a transformer on it that has half a percent distortion at 20 Hz, and gets worse as you go lower. We know from console design that you can build active differential circuitry without transformers that can match any load. It's just a question of doing it, and that takes time and money.

"Then there's the question of filters. One of the unexpected advantages of using a 44.1 kHz sampling rate is that JVC had to design its filters really right. We've found that some groups who've mixed to analog like to transfer the tape to digital for cutting, because they like the way it sounds. That's because there's a limit on the top-end which eliminates the signal between 20 and 25 kHz, which can excite the cutter head in an area where there is no real active feedback control. That's where sibilance problems are aggravated, and you get slewing and phase problems which make for a funny-sounding top-end. Nobody has ever though to make a sharp enough filter for analog, and the filters that do exist you can hear. The filters on the digital, however, are so sharp that those frequencies are just turned right around."

The Weak Link

Even though the end result of all of this fancy technology is still a plain old vinyl disk, and Meadows is very quick to criticize many of the major labels for cutting corners at the pressing stage, he still thinks that digital editing and mastering represent a significant improvement.

"Digital tapes do sound better than analog," he maintains. "There's less noise and better transient response. Direct-to-disk records, which we did quite a few of here, prove that vinyl has better sonic properties than the best analog tape, so now we have something that can push the disk medium to its limits. Even when we're going from an analog multitrack, we're subjecting all those complex waveforms to only one generation of analog degradation instead of two, so we don't compound the noise and distortion with a second analog tape."

By promoting digital techniques at the professional level, Master Technologies is not indirectly furthering the cause of digital consumer equipment. A strong case can be made that, by doing so, a mastering lab may soon find itself out of work — the mixing engineer could theoretically send a tape directly to the pressing plant that makes Compact Discs (or whichever form consumer digital finally takes), and be confident that the results will be precise replicas of what he has done in the studio. But Glenn Meadows is not worried.

"I expect it will be at least 10 or 12 years before consumer digital takes over," he considers. "For the time being, we'll still be equalizing and compressing tapes, even digital ones, before we send masters out of here. After a while, things will reverse, and we'll be able to simply edit the tapes and say 'cut this flat,' and process them later for analog disks. But even now we don't do all of what we do just to accommodate the disk medium — the bulk of our work is creative. We supply an additional input to the record-making process. We don't compress music to make it fit on vinyl, we compress it to make it sound better. The tapes coming out of the studios will have the same problems they do now: balancing, level discrepancies, and room anomalies. The limitations that we now face on sibilance levels or mechanical excursions in general will obviously be eliminated, but the creative input won't change.

"The ability to give the consumer 96 dB of dynamic range is just startling, but that much sound won't fit into a home environment — that kind of range is just not 'real world.' Real world has to do with record wear, surface noise, distortion, scrapes, scratches, and whoosh. That's where digital will help, and be important, in the home. But there will still be a need for a good processing facility."

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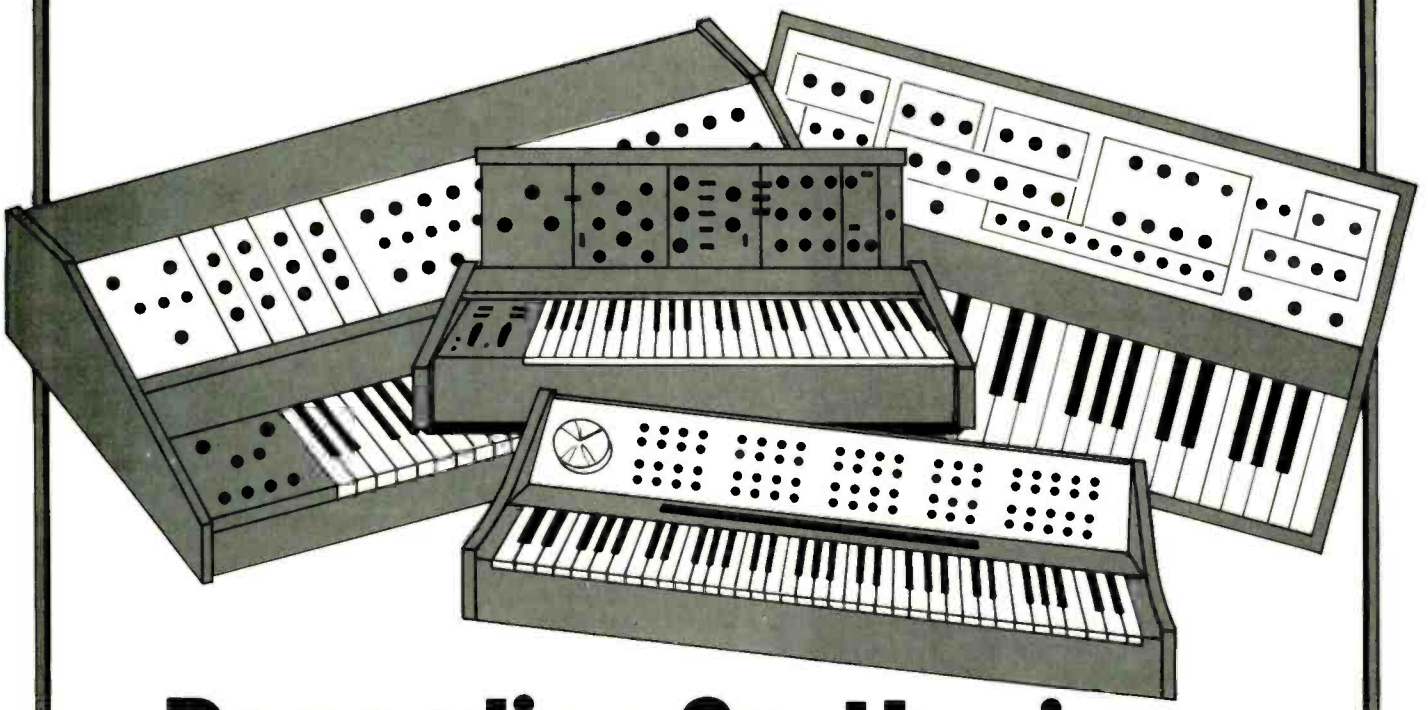

Recording Synthesizers And Drum Machines

by Robert Carr

What's difficult about recording a synthesizer? After all, you just plug it into the console, and play it. In some cases that is all that's required. Most programmer/players are their own engineers, however, and this integration of technician and artist has itself synthesized an approach to recording that penetrates far beyond any simplistic attitudes of "plug it in and play."

Musicians that work with synthesizers usually are addicted to exploring the myriad of aural labyrinths hidden between the instrument's keyboard, and the line-level output. They are a dedicated breed of people, who appear to fit somewhere between the temperamental virtuoso determined to find just the right sonic texture and shading, and the mad electronic genius of the B-grade movies surrounded by blinking lights, miles of patch cords, and two of every conceivable piece of processing gear known to man.

Frank Serafine, whose credits include sound effects for *Star Trek — The Motion Picture*, and Disney's *Tron*, voices the following opinions when asked about the phenomenon of the musician/technician rolled into one: "I do my own engineering, because of the intimacy involved in being alone, especially when composing music and sound effects. Using the BTX [synchronizer] remote system, I'm able to compose right at the keyboards. I just assign my tracks and keep layering. I'm working with machines anyway, so recording is the least of the problems."

Recording synthesizers starts with the original concept of the tone, and a rough sketch of the musical composition. Next comes an analysis of what equipment is appropriate, and how the voice will be created, followed by the actual artistic process of shaping, bending, molding, and breathing life into a delicately-balanced intangible that is capable of evoking an emotional reaction. And, finally, the delivery of that product from the end of an umbilical line-level output. From then on, the primary focus is one of simply positioning in the track, because everything else has been completed before the sound reaches the console.

The procedure is complex, demanding a marriage of numerous skills. So knowing how to record synthesizers entails understanding the evolution of the tone's birth. *R-e/p's* guides through that maturation process are two synthesist/engineers — Frank Serafine and Stephen St. Croix — and one keyboard engineer, Bryan Bell.

Essentially, there are two kinds of tone synthesis: analog and digital. However, the instruments currently on the market don't always fit neatly into one category or the other. A number of models are hybrids that incorporate elements of both technologies to yield their own unique features and qualities.

Digital Synthesis

Digital synthesis employs a computer to generate, modify, store, and control a sound. The big draw of digital instruments is the fact that, in addition to creating voices and storing them in a microprocessor memory, a computer can be used to sample (record) a pre-existing sound, analyze it, break it up into discrete bits for storage, and then puts the parts back together again to reproduce the original sound . . . with amazing accuracy. This basically is the ability to faithfully produce all the individual harmonic components of complex waveforms in real time. (Ironically, simple tones, like a sinewave, are better served by analog means.)

One such digital instrument is the Fairlight CMI manufactured in Australia. Frank Serafine relied on the Fairlight to generate sound effects for *Tron*, but has since replaced that instrument with an Emulator by E-mu Systems, Inc., based in Santa Cruz, California. "The Fairlight has great software, and eight outs," he says, "but the Emulator tends to be a little more 'user-friendly.' It's a synth, not a computer. It doesn't have everything the Fairlight does, but

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for my purposes, the Emulator works better."

Serafine is currently writing and recording the entire score for a set of fairy tales produced by Shelley Duvall for Showtime Cable TV. "My whole band is orchestral in nature," he explains, "where I take real instruments, like flutes, cellos and violins, and store them in the Emulator. Interestingly, there's a lot of equalization needed to make them sound real. Synths are so dry, I have to pinch here and there to get it to sound good, especially on cellos, when I'm simulating real acoustic sounds. I'm using the new TEAC parametric equalizer with four bands of EQ.

How the sounds are digitally sampled is very important, and sampling directly into a digital synthesizer is not necessarily the best approach. In the case of a violin, for example, Serafine records the musician on tape. After the player leaves, the tape is played back, compressed, limited, and then sampled into the synth. "It's easier to work with. The synth has a limit of two seconds for the sample. If the sound is on tape, I can work with it as much as I want or need to, at my own pace. Actually, my studio is not really set up for recording instruments. The best way for me is to record the instrument live at some other location with a Nagra on ¼-inch tape, and then sample that. An Audio + Design Transdynamic [compressor-limiter] is a big help. It appears to boost the volume of the instrument without raising the dB level, which really enhances the dynamics of the sounds I put in the Emulator."

Frank Serafine has been a musician most of his life, playing primarily flute and Fender Rhodes. His recording credits to date include sound effects for "Star Trek — The Motion Picture," "The Fog," and "Tron," and trailers for "Saturn 3," and "Quest for Fire," as well as conceptualizing and developing electronic music behind high-tech commercials for Bell Telephone, Zenith TV, Chrysler, and Suzuki. He is currently writing and recording the soundtrack for a series of fairy tales produced by Shelly Duvall for Showtime Cable TV.

As the president of Marshall Electronic, **Stephen St. Croix** has been involved in research relating to psycho-acoustics and/or synthesis for 20 years. While, by his own admission, he has no formal musical or technical training, his work has appeared on many albums. He's worked with Stevie Wonder doing synthesizer patches since "Songs in the Key of Life"; a large portion of "The Secret Life of Plants" was the result of St. Croix's work. He developed a device that would measure the turgor of plants (similar to a human's blood pressure) and the differences in surface conductivity of a leaf, and translated that into a signal that would control an ARP 2600. By exaggerating his emotions, the reaction of the plants generated the music heard on Wonder's record. His movie credits include "Star Wars," "Alien," and the majority of recent science-fiction films, for which he acted as synthesizer player, and/or supplier of innovative processing hardware.

"What can be done with synths through sampling is pretty outrageous," Serafine continues. "For instance, a fantastic kettle drum sound, or unbelieveable piano sounds; flutes are wonderful, but they will never replace the orchestra. The parameters of self expression on one instrument alone is ridiculous. You can't get all the inflections and innuendoes of how that instrument *really* works — variations from player to player as they're expressing themselves add to the overall sound.

"I have to sample in a pizzacato, then bowing, and so on. I have to get all these different sounds in the memory, and I can't play them at the same time. I can play a bowed section, then switch to another [floppy] disk for the pizzicato. The process is still limited. Eventually we'll see the accuracy and flexibility to reproduce the real-instrument idiosyncracies, but not for another 10 years or so."

Serafine frequently controls a synthesizer with an acoustic flute, which is hard-wired through a FRAP acoustic transducer to the pitch-to-voltage converter of his Sequential Circuits Prophet synthesizer. "I've played flute my whole life, so my technique is pretty good. I run the flute and FRAP through a Sescom direct box to the pitch-to-voltage converter, which controls the voltage and triggers whatever sound I have in the Prophet, like strings. My flute actually takes the place of the keyboard, and the embouchure I make with my mouth defines the envelope of the sound. I used the same techniques in *Tron* for the Lightcycles, but I played a microphone instead of a flute."

Since Serafine developed an interest in synthesizers, his Fender Rhodes electric piano has remained practically idle in one corner of his studio. He's found it much easier to sample the Rhodes into the Emulator, and run the signal through the Auto-Pan module in his ADR Scamp Rack. He reports that the "watery" kind of tremelo effect for which the Rhodes is known sounds a lot cleaner that way.

Analog Synthesis

Unlike digital synthesis, analog synthesizers generate tone with voltages that vary continuously rather than discretely, and are controlled by other circuits and signals. Analog methods arrived on the scene long before "digital," giving it a head start in terms of instrument choice. — *continued overleaf . .*

Bryan Bell runs his own production services company through which he is a concert production manager, sound engineer, and keyboard engineer, all of which involves research and development, computer programming, and recording. While mixing for John McLaughlin in the mid-Seventies, Bell met Herbie Hancock, and has worked with him since in the above capacities. Between recording and concert dates with Hancock, Bell can be found doing special effects control on tour for Earth, Wind and Fire, and sound mixing for Santana.

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The Computer Connection

The Fairlight CMI, E-mu Emulator, and New England Digital Synclavier synthesizers, and others, essentially are computers with musical keyboards. The internal memory of their central processing units store digitized signals that are converted back to analog when played. Practically any voice can be created by, or sampled into, one of these units with amazing accuracy. The addition of a disk drive affords virtually unlimited capacity for information storage, either as individual voices, or entire compositions. and, if necessary, the sound can be analyzed on a video monitor, manipulated via a typewriter keyboard, or with a light pen, and, in the case of the Synclavier, even printed out in the form of a spectral display, or as musical notation. A major shortcoming, however, is the price —which can range from very expensive, to even more expensive.

Despite the cost considerations, computers are infiltrating the audio industry almost as quickly as they have made themselves at home in all the rest. And the applications are boundless. Stephen St. Croix uses an Apple computer to interface his digital synthesizers with a sync track laid down by an Oberheim sequencer. Frank Serafine's E-mu Emulator has an RS-232 input/output port that connects to an Apple II+, to "do all the things that the Fairlight can do with their software," he says. "The computer lets you control all the sound effects, set up sequences, turn the sound backwards, even operate it like a video game. You can write graphics and transpose them into audio. By adding a $700 digital oscilloscope, you can do an analysis of your waveforms. On the consumer level, you can piecemeal a Fairlight together for a lot less money."

While doing the sound effects for *Tron*, Serafine discovered several other applications for computers in The studio. Typically, a lot of production time is spent looking through a stack of tape reels for sound effects, especially when some of the effects require up to 10 or 15 individual elements. An Apple running a Visicalc business software program offered part of the solution. One column from the program was assigned to each of he 15 tracks on the master sound-effects library tape. Under every column heading was listed the names of he effects on that track, and their locations. The 16th column, which corresponded to the SMPTE track on the tape, contained the names of all the elements that made up a particular composite sound effect. With the push of a control key, the printer typed out an up-to-date track sheet.

An Atari computer handled the remainder of the housekeeping duties. Using a File Manager 800 software package, Serafine logged the finished sound effects and related data at random. The computer automatically alphabetized all the names, and sorted the individual files by sound number, category, reel number, track number, clock number,

Stephen St. Croix is president of Marshall Electronic which, according to St. Croix, "makes serious toys for the recording industry." In addition, his work with psycho-acoustics is approaching awesome proportions in terms of the control he exhibits through the use of particular synthesized voices. (By way of an example, see St. Croix's article entitled "Hassling Hass," in the June 1980 issue of *R-e/p.*)

St. Croix attributes his success to 20 years of playing experience, and extensive psycho-acoustics research conducted over the past few years. His background allows him to write music, compose the voice, program the synthesizers, perform specialized wiring, and even invent new equipment to deliver the proper tonal character.

He describes his approach as follows: "For me to build a sax sound, I first study a sax in my mind's eye. It's a tube that resonates in certain ways, and a piece of reed that moves like so. Occasionally I'll take an instument apart, analyze sections with a digital memory 'scope, and process the waveforms. I just logically think out the mechanical process of how the instrument produces its tone. The sax, for instance, must have 20 or 30 resonances. So I build them up and, when I get close enough, I start fine tuning and adjusting by ear, because the gear I'm using has certain limitations."

According to St. Croix, when building a tone, only the first four or five seconds are critical to establishing voice credibility. "Once the mind has identified that you've generated a sax or a trombone, it accepts that fact, and figures it doesn't need to study that voice anymore. It releases its power of observation, and goes on to something else. Then you can slack off, even make the voice quite sloppy, and the brain will never notice it's been changed. The mind takes a great deal of convincing before it will believe something has happened to the tone, and that it's no longer a sax."

But, in order to fool the brain's recognition centers in the first place, the sound must be *accurate*. St. Croix says that he achieves this end through a substantial amount of post-signal processing, with an emphasis on very short time-domain processing in the microsecond range. "You can't tell the delays are there, but the sound is warmed up, fattened, sweetened, smoothed, or more acoustic. I'll use an many as 20 to 25 delay lines on one voice. We're talking about 80 microseconds, 16 microseconds, 1.3 milliseconds, 6 milliseconds, maybe 18 mililliseconds, and back to

Erik Lawton

Frank Serafine's personal-use studio houses an E-mu Emulator, Roland Digital Sequencer, Minimoog, and Prophet 5 synthesizers.

120 microseconds — all very short times, in order to build references."

"I put the instruments in a room," he goes on to clarify, "by synthesizing the walls of a room to make the instrument sound more acoustic. You could try to synthesize a sax for 20 years and it will *never* sound like a sax until you put it in a room. The room is an integral part of the way people are used to hearing saxes. That component *has* to be there. Unfortunately, there is currently no commercial gear that addresses this need. I have a little algorhythm on a Hewlett Packard HP-41 [calculator], where I punch in the room height, width, and depth, and key in how much furniture I want in the room, and how far the instrument is from the nearest wall. Where I place the instrument asymmetrically sets up a series of echoes. Then, within both the millisecond and microsecond ranges, I go for how many cycles or repeats of echo I want. I can determine the number of bounces, or how hard the room is. The very short delays I compute from the HP-41 are what I dial in on a series of delay lines. Basically, I build a room."

The natural extension of such an analytical approach to voice synthesis has led to some startling discoveries in psycho-acoustic phenomenon. St. Croix has reached the point where he can literally guarantee that any voice he builds will produce a specific desired effect — relaxation, fear, anxiety, spacial disruption, etc. — in the listener. This process is accomplished by psycho-acoustically loading the voices with subliminal cues, such as waveforms, certain patterns of waveforms, or harmonically-scanned patterns that are associated with such emotional reactions.

St. Croix's basic approach is to construct a scenario for each voice project. Take "fear," for example. Considering primitive man's environment, an electrical storm would have been of major concern. The components that make up such a product of nature involve an ionic change in the air just before the storm. Cloud layers converge on the area producing subtle high-frequency response changes in the ambience, as a result of changes in air density, and a shifting barometric pressure. And compound the scene with a stalking dinosaur that shakes the earth and provides certain low-frequency pattern. The survivors learn that this package of aural cues mean "danger." This education is genetically transmitted through time, so that today these responses can still be invoked.

"I could play the first two bars of *Peter and the Wolf* for you," he explains, "and you'd go, 'Isn't that cute.' But if I play it again using these loaded voices, you'll think it's *very* heavy. The voices are only slightly different. If you heard the voices an hour apart, you might not even be able to notice the alterations, but the psychological cues would produce completely different reactions."

To make his point, St. Croix presented a hypothetical experiment similar to his previous studios. For a double-blind test, 10 subjects over 16 years of age (old enough to have developed an awareness of social cues) are chosen at random, some being religious and some not. A typical, unprocessed male voice is played for them to identify on the first pass.

The second sample is processed by taking a slightly delayed feed to a reverb, linearly shifting it down in frequency by about 0.75%, and looping that through a reverb five or six times before the chosen RT60 is reached. Every time a bounce comes back, it is slightly more flat. The ear can't hear it, but almost 100% of the listeners will think of a church.

He then mentions that if the voice is shifted up by the same amount, "almost 100% of the people would think of a fresh, *The Sound of Music*-type of scene — sitting up on a beautiful green hill in the sun. The results are reliable. I can put them on an album, and I can slam

— continued overleaf . . .

the audience around with what is almost immoral control."

RECORDING TECHNIQUES

Getting any synthesizer voice, whether analog or digitally generated, on tape is easy. Usually, running the signal directly into the recording console is all it takes. However, on a few rare occasions, a natural room sound is still the only answer for achieving a particular coloration.

"If I'm trying to get an electric guitar-type sound," says Serafine, "I'll run my synthesizer through an amp, and mike it to get the lead-guitar distortion sound from the speaker. I use a Music Man amp with two twelves, or a Fender Twin Reverb. I'll put an Electro-Voice RE-20 on a short stand, in the back of the room [about six feet away] on-axis facing the speaker. I don't do it often, because I usually prefer to create electronically. I just go straight in to the console, and I have no problems."

Bryan Bell, keyboard engineer/live-sound mixer for Herbie Hancock, mentions a similar technique that he picked up from Fred Catero at The Automatt in San Francisco. "For solo sounds we sometimes use a Mesa Boogie, Mark II, to achieve guitar sustain. It's not acoustical feedback, but a certain type of sustain, which produces the same kind of overtone series generated in guitar distortion, when the tubes start breaking up.

"In general, Herbie's amp is recorded the same way I mike Carlos' [Santana] Boogies. A Sennheiser 421 is set up about two inches away, and directly in front of, the amp. A second mike, a Neumann U-87, goes across the room [about 20 feet away] on-axis and approximately four feet off the ground facing the amplifier. The distant mike works best when placed near a reflective surface, like against a glass wall, to get the 'PZM-kind' of feeling."

Drum Machines

The Linn Drum Machine, a life-like digital drum synthesizer growing in popularity as a never-rushing, never-dragging rhythm keeper, is usually taken direct, but sometimes a speaker is effective for recording a bass drum sound. "The bass drum sounds a little too unnatural," says Bell. "We'll use a mike and speaker to actually get the air motion. My favorite bass drum mike is the Shure SM-7, a big vocal mike. As a rule, it's usually not necessary to do that with the other drums on the Linn."

In contrast, Frank Serafine chose to purchase a Roland Compu-drummer, but finds that he's creating most of the drum sounds on the Emulator. "I bring the real instruments — bass drum, hi-hat, cymbals, snare — into the studio, sample them, and it's like the Linn machine. I can get up to 12 instruments on one keyboard, and they're all tunable. The advantage over the Linn is that I can change the drum sounds whenever I want. Using the sequencer, I can compose the drum parts at half speed, turn them up to full speed, and the compositions sound outrageous!"

Steve St. Croix owns practically all the available drum machines, including a Linn, the new Oberheim, and the Roland TR-808. The Roland builds drums by analog synthesis, and "happens to have the finest handclaps and cymbals that money can buy," he says. "That machine is slaved off my Linn, which is highly modified. I doubled the memory [example: a high tom stored in 4 kbytes of memory now has 8 kbytes] and the clock rate, and changed the output filters. I now have twice the frequency response, and no clock whistle."

The output of the Linn and/or Oberheim go directly into the mixing desk, St. Croix explains, with every drum getting its own channel and EQ. These eight drums/cymbals are recorded on eight tracks of the 24-track, just in case he wants to put some processing on any of them for the final master. In fact, these tracks are often thrown away without being used at all.

Simultaneously, a stereo sub-mix is sent to a pair of Altec 604s in a medium dead room, and miked with four microphones. An AKG C451 is positioned approximately two feet in front of each driver, and monitor levels kept down so as "not to splatter the 451s." The other two mikes, Neumann U-87s with full pickup patterns, are located against a rear, reflective surface of the room to capture the maximum amount of space. Because the microphones and the time delay between them are so different, there's no problem with phase cancellation.

"I jockey these four mikes, depending on the type of ambience I want, down to two separate channels on the 24-track," says St. Croix. "Just natural ambience is used, never any reverb, which really slurs the field. But I do use some multiple delays. I leave the kick alone, but I put about 15 milliseconds of delay at around -20 dB on the cymbals, and multiple delays on the toms and snare at like -25 and -40 dB. That really warms up those sounds, because they're pretty edgy in the machine, and need this help to put them in a room.

"The [drum computer] manufacturers were careful to take the sounds out of the room when they put them into the chips of the drum machines. If there was too

The Computer Connection

— *continued*

¼-inch tape source that the various sound elements came from, timecode number, description of the effect, plus any comments concerning how it referenced to the picture.

"We could type in any key word, like 'solar,' and the computer would sort through 500 files to pull out everything containing the word 'solar.' From that abbreviated list we found the correct SMPTE number, typed that into the BTX Shadow [synchronizer], and all the machines would shuttle to the right spot on the tape."

Bryan Bell, too, has found that Apple computers are easy to use, but sometimes they require a little bit of help. Presently, Herbie Hancock owns an Apple II+ and an Apple III. To augment their capability, Bell, along with Michael Larner, designed and built a Z8000-based master computer that tells the sequencers and master clock when to start, and what program any of the computers are supposed to be on. The sequencers, in turn, start and stop the tape machine, drum machine, and everything else. The 16-bit motherboard has the memory, SMPTE interface, Apple interface, and automated ports to "talk" at high-speed to 6502-, Z80- (Linn, Prophet and Oberheim) or 6809-based (Fairlight) computers.

"This is top secret; totally cryptic," Bell confesses. "No address is the same as the one you actually write to, yet all language commands are in simple English. The programs are stored in Applesoft BASIC, and all the telecommunications are stock Apple. And it can talk to the patchbay through one simple channel."

The patchbay is another one of Bell's digital projects, designed to uplift Hancock's entire arsenal of synthesizers and attendant devices to the ultimate in "user-friendliness." The 32-by-32 stereo switch package (64 channels in and 64 out) is a digital interface for computer control that does simple telephone-like patching. When the keyboards — E-mu 16-voice, Oberheim eight-voice, Prophet, Linn Drum Machine, two Minimoogs, two ARP 2600s, Chroma, Yamaha CS-80, and Memory Moog — are plugged in, a change from one keyboard voltage to another can be accomplished in a fraction of a millisecond. Any stock RS-232 output can program the switcher, albeit from an Apple, TRS-80, IBM-PC, and so on.

"Herbie can program the entire performance and control all the patches from one portable keyboard. He can play strings for the first half of the first song, horns for the bridge, switch back to strings, turn on the Vocoder and mike, just by pushing a button for next cue. The color display on the Apple [video screen] tells him what's currently plugged in. If he needs to make a quick change at any time, the computer understands simple English commands, like 'Plug the E-mu into the Oberheim,' and knows what wire numbers to connect. The code is quite extensive. This part of the set up took about two years of research and development." ■■■

Synthesizer Room at Steve St. Croix's personal-use Lightning Studio contains: an ARP 2600, modified for three voices; D6 Clavinet with digital control board to drive Oberheim 8-voice; Wurlitzer modified to drive an ARP 2600; an Apple II+ Computer & Alpha Syntauri keyboard, with Sequential Circuits Digital Sequencer, Roland CR78 Drum Machine, Oberheim Expander Module, & Roland 100 Series Module; Polyphonic Chord Shifter custom-built by St. Croix; Oberheim DS-2 Digital Sequencer; modified ARP 2600 with additional oscillator; Oberheim Modular Eight-Voice; Roland TR808 Drum Machine; Oberheim Analog Sequencer; Roland 600 CompuComputer; Sequential Circuits Envelope & Voice Memory Programmer; Roland Analog-to-Digital Converter, & OB8 interface for Jupiter 8 driver; Oberheim DSX Digital Sequencer/Controller; Jupiter 8 modified to "talk" to the Apple & Roland; and a heavily modified Oberheim OBX. The Electric Dream Plant Wasp monophonic synthesizer provides input for Vocoders.

much room ambience, you'd hear the sound drop off when the ROM [read-only memory] ran out. The drum would hit and turn off, rather than have a natural-sounding decay. In order to reconstruct the room, and soften some of that turn-off that the drum machine has, I use these multiple delays. I've also found that when I use the Linn and the Oberheim at the same time, the different sounds blend together to give me an expertly-miked drum kit."

Sound Depth and Imaging

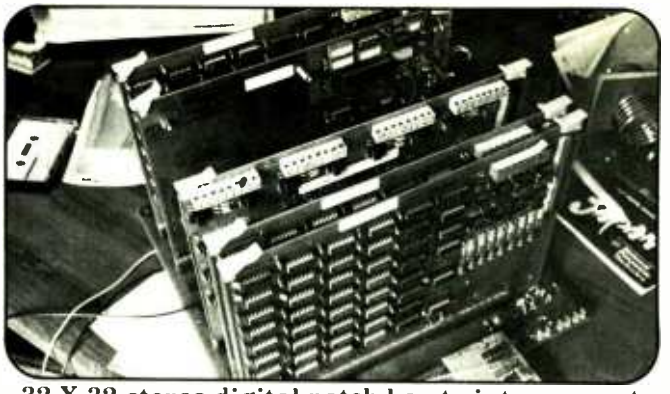
Direct recording requires a dependence on artificial acoustics to achieve depth and distance, and that's especially true where the potentially sterile tones of a synthesizer are concerned. Frank Serafine relies on a Lexicon 224 digital reverb and a Delta Lab DL-2 Acousticomputer DDL/effects unit to provide spacial characteristics. Because the majority of synthesizers have mono outputs, stereo has to be synthesized as well. "The processers are stereo, and I have auxiliary A and B on each one of my synth modules," Serafine explains. "One note splits into two-channel stereo at the synth's output. On the [Tascam Model 15] recording board are 'Aux A' and 'Aux B,' which are for stereo effects. I come out of the reverb in stereo through the console, and spread the voice out."

To add a touch of variance between the two signals, Serafine plays one track off against the other with a little delay difference. "No tracks are dry, or they'd smash right up against your face. They both have a little bit of sound processing. One voice receives a short delay [approximately 50 to 75 milliseconds on the left side of the Delta Lab DL-2] for presence, and the other gets a longer delay [about 100 milliseconds on the right] to give the illusion of depth. There's no problem with imaging through the processors, because I'll just pan each track where I want it in this simulated stereo space. A mono effect can go anywhere, because stereo is derived out of individual placement for the whole band."

Bryan Bell elaborates on the concept of spatial orientation by referring to his experience with Herbie Hancock: "It's not really hard to get good imaging and depth with a synthesizer. We generally record the sounds dry. We don't prerecord echoes, or any reverb, unless it's part of the solo sound, which we get from the guitar reverb in the amplifier. But we do prerecord phase shifting, flanging, and most of the double effects; any effect that composes the 'character' of the sound. Digital delays are a component similar to the effect you'd get feeding a signal through an amp, and blending in the room sound with the original. Those are all pretty much predetermined, and placed in space where we want them to be.

"For instance, we spend much more time on the string tracks and horn

Bryan Bell's custom-designed 16-bit computer controller/interface for the Apple II+.



32 X 32 stereo digital patch bay to interconnect various synthesizers under software control.

punches, getting the parametric equalization, the compression, and the actual synth sound prerecorded correctly, so that later we can choose how to do the stereo imaging — either front-to-back, or left-to-right, or whatever. The Oberheim eight-voice is stereo. A lot of times we'll do stereo layers with the OB8, and then stack horn punches from the Yamaha CS-80 and Prophet on top."

These are all done with fairly standard panning," Bell continues, "Because Herbie's album *Mr. Hands* mostly was trio performances recorded live in the studio, with Herbie orchestrating the rest of the album by himself. His more recent projects have been directed more towards popular music, however, where the lyric and the vocal parts have been as important as the keyboards."

Bell feels that the biggest problem in synthesis arises when the artist doesn't understand the difference between trying to emulate a natural, acoustic sound, and trying to create an original one. "When Herbie is trying to recreate a string section with cellos, first and second violins, right from the beginning he's conscious of the stereo placement of the Oberheim's eight voices, and conscious of the modulation, and the vibrato, as if it were supposed to be *real* cellos. He has the articulation in the voice of the program, and all the engineer needs to do to make it sound like an orchestra is record it in stereo, and put echo on it."

Post-production imaging is not difficult, if you are trying to make a string or brass voice sound like real strings, or brass. The real problems in stereo imaging show up when "you're doing something unique," says Bell, " . . . when you're making a new sound that's not supposed to sound like, let's say, trumpets, but are supposed to fit in the brass region. Then you have nothing to go on. You can't say, 'Well, Mile's [Davis] trumpet used to sound like this,' or 'Wynton Marsalis' horn sounds like this.' You're trying to make a sound that's *never* been heard before. You don't really know what you're doing with stereo imaging, because there's no groundwork there. Herbie does it by feel. As he hears it, he modifies the concept. It's a maturing process that's totally creative."

Steve St. Croix's approach to voice placement is somewhat more unorthodox, but highly effective and accurate, he considers. The result depends on the quality of the gear, and the ear of the programmer. "It's possible to actually control the phase relationships of the waveforms of the multiple oscillators. I've been experimenting with psychoacoustics for a long time, mostly in the area of Hass Modulation. Right now, I can float my voices in space on an album without having to pan them. Panning is not really the right way to relocate a signal anyway, because movement of signals in a soundfield involves a whole host of other effects, amplitude changes being the most insignificant. For example, Doppler shift and the intra-aural delay matrix dominate the localization process. Phase delay manipulation, on the other hand, can do 360-degree image positioning. With two speakers sitting in front of you, I can put a voice anywhere in the world — behind you, above you, inside you, as far left or right outside of the speakers — and all *predictably*."

Several years ago, St. Croix and his

associates were commissioned to develop a backpack for the US Defense Department which, utilizing pyschoacoustic manipulation of a gun blast sound, would mislead the target into thinking the marksman was as much as 100 feet from his actual position. This effect is accomplished by two speakers located in the rifleman's backpack. If there's sufficient time to spread the speakers out on the ground, the shift in distance can be expanded to 500 feet from the source of the blast. Through this research, St. Croix developed spatial techniques applicable to the world of music synthesis and professional recording.

The image shift is done strictly by inter-aural delay cues, as it relates to the distance between the two ears, and the physical folds in the outer ear. St. Croix explains: "There is approximately a 1-millisecond maximum delay potential between what the two ears hear as the sound from a single source travels around the front of the head. Although one millisecond is a very short period of time, the outer ears are equipped with folds, which are actually flangers that put several important notches in the acoustic signal within the microsecond range. The frequency location of those notches is one component in determining where the sound is coming from."

"The second component relates to, and is triggered by, the difference in delivery time of the transient signal between the two ears," he adds. "Within that 1-millisecond domain, the computer in the brain does a search for the source; it computes the difference in time very accurately, and provides the point source illusion.

"And 'illusion' is an important point. The point source is generated strictly in the brain. Once you understand the phase relationships that create the 'keys' that give the computer the information, you can control the image completely — not only across the front of the [sound] field, but by frequency selective inter-aural delays, and by simulating the high-frequency phase notches generated by those little folds in the outer ear."

St. Croix's first successful attempt was accomplished with commercially-available gear, "I used four Marshall Time Modulators, three ARP 2600s and an Oberheim OBS. I got the sound to spin, and the result was not fragile. You could press it on an album; air it on stereo radio and it still spins. The effect can be built into a synth voice so that without touching a panpot, one electric guitar will go all the way around your head. This image shift can also be voltage-controlled, so an envelope can move the sound."

As an interesting side note, St. Croix discovered that people over 45 years of age have much better location capability than younger people. Location is accomplished by inter-aural delay with information up to about 1.5 kHz. Contrary to popular belief, frequencies above that point play little or no part in determining location. Older people, whose hearing rolls off at high frequencies, don't receive as much conflicting information, and can do a cleaner job of locating the point source.

A second important fact emerged from recent work with digital equipment. "About eight weeks ago [as of November, 1982] I finally got into digital synthesis along with my analog work," says St. Croix. "On the machine I'm using, I can digitally command the phase relationship of each oscillator, but its limited resolution and the damage from digital-to-analog conversion is severe enough that I cannot do image location with it. If I dial in the same phase relationships that I know for a fact produce predictable image shifts with analog equipment, the digital gear doesn't produce them. There is only a certain amount of resolution on all axes.

"Digital is quantized, and if one of the values I'm after happens to be between the quantized steps, it's not available to me. The theory to rectifying this deficiency is increase the resolution. Then I may be able to start doing amplitude and phase relationship matrixes that can reintroduce this location work I want. Right now it can't be done."

Sequencers

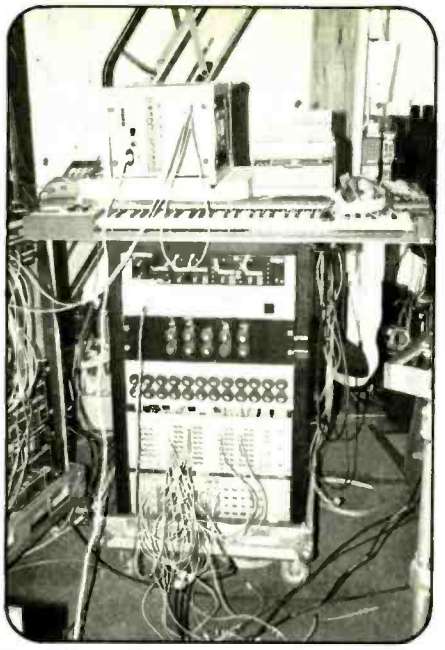
The complexity of musical synthesis

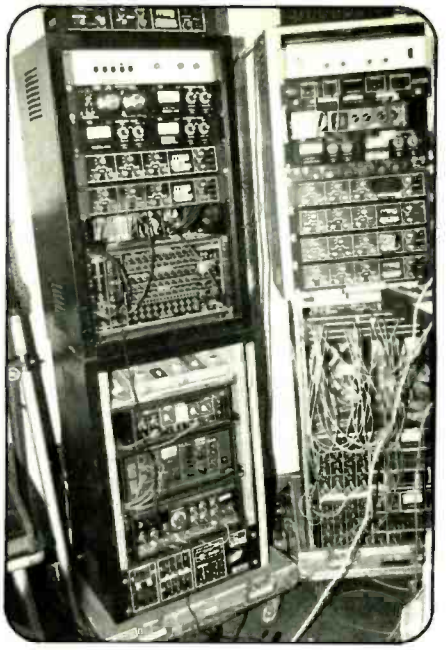
— *continued overleaf . . .*

has already reached the point of being overwhelming for most players and engineers. Yet, with all these complexities come options that programmer/players continue to exploit to the fullest. But manually playing several keyboards at once is impossible, and layering can be tedious, especially when the same part has to be repeated flawlessly and in sync, with not only other recorded parts, but the studio rate card as well. To maintain command, increase efficiency, and simultaneously push forward into the unexplored regions of the aural universe, players turned to controlling devices that stretch the abilities of existing music machines.

The most common means of control are clock pulses and sequencers. The clock pulse is typically a square wave, but can be a low-frequency pulse of a sawtooth waveform that may or may not be voltage-controllable. Sequencers output control voltages and timing signals according to a plan established by the programmer. These outputs are used to control synthesizer modules in much the same way that a keyboard does. However, unlike a keyboard, the sequencer can store the desired commands — such as tempo envelope shape and filter parameters — and release them when triggered by a clock pulse, or some other method.

Analog sequencers fulfill this task via the particular settings of several rotary knobs or vertical attenuators. Digital sequencers employ a microprocessor





The E-mu Rack (left-hand photo) contains an E-mu Sequencer, a Z80-based polyphonic keyboard with disk drive for storing programs; Master Clock; Clavitar 4 by 8 Switching Network; a pair of Tuning Interfaces; 16 discrete E-mu control voltage & gate outputs, and E-mu power supply; and manual patch bay for all inputs to an Oberheim 8-voice, Prophet 5 two ARP Odysseys, two Minimoogs, and ARP 2600. The Signal Processing Rack (right-hand photo) contains a Clavitar Power Supply; Eventide H910 Harmonizer Yamaha Digital Tuning Scope; two UREI LA-4 compressor/limiters; Trident Parametric and 4 Moog Parametrics. The Vocoder Rack houses a Waves custom digital switcher, which accesses all the functions of the Sennheiser VMS201 Vocoder; a UREI 556 Bandpass Filter; three UREI LA-4 compressor/-limiters; two Moog Parametrics; and a Guitar Effects Rack.

that affords a much expanded capacity for storage. Voltages are fed into the device, transformed into and stored as numbers, passed through a D-to-A converter when triggered, and output as voltages.

Frank Serafine's E-mu Systems Emulator outputs a clock pulse, which he transfers to multitrack tape that also contains SMPTE timecode; in this way video tape, audio tape and synthesizer can be locked together. All the devices can be shuttled in sync by his BTX Shadow synchronizer control unit. Layering tracks from that point on is practically effortless.

"For instance," says Serafine, "I can put a cello disk in the Emulator, let the sequencer play a part, replace that disk with a violin disk, and the machine will

repeat exactly the same notes I played with the cello. Then I can go to the Delta Lab DL-5 Harmonicomputer, and change that to a pitch a third up, or down. And all of it is in sync with the video. There's no comparison to the old click tracks."

Steve St. Croix agrees, and has taken the concept a step further. All of his synthesizers, including the analog models, are in some way digitally controlled. "All I'm recording on the 24-track is control information, and whatever acoustic instruments — like bass guitar, vocal, etc. — that I want. A typical tune may be four or five tracks of live audio, and 10 tracks of digital control. The synths are set up in the studio, and are not only driven by click tracks, but by digital code down to the microsecond.

"When I roll the 24-track from the top, the synths follow while I play engineer, and make a mix direct to two-track. I can shuttle anywhere, and as many as 15 synthesizers will pluck the right voices, reconstruct the settings, and always play together in sync.

"Because I'm coming directly out of the instruments, all the transients and delicate harmonics are preserved, and no tape noise is added. The final mix is literally first generation. I can roll my 24-track 500 times, and tape wear doesn't really affect the control information. All I'm doing is playing the synths 500 times, and they don't care."

St. Croix's Oberheim DSX Sequencer generates a sync track that operates within a 500 Hz to 1 kHz range, and is printed on track #1 at -10 dB on the console meter. Five or six synths can be slaved from that one track, including Roland, Oberheim, and ARP equipment. Since the DSX provides a click and a tone, the sync track for another synthesizer is easy to lay down. The original (DSX) sequencer is capable of firing gates which, in turn, cues the rest of the sync tracks and locks them on tape. The digital synths are controlled by an Apple computer with modified Alpha Syntauri software that monitors the DSX track from the 24-track recorder.

System Integration

Herbie Hancock's extensive keyboard system, out of the necessity for having to remain somewhat portable for live concert applications, required a few modifications to make life easier. The activity is based around an E-mu sequencer, and intelligent polyphonic keyboard with a Z80 microprocessor, 48K of RAM (random-access memory) on board, and a custom-installed 8-inch IBM disk drive for increased storage capacity. The E-mu outputs 16 separate control voltages, and gates polyphonically. Bryan Bell customized the system so that the oscillators of all the keyboards can be tuned and stretched from one central location. That comprises all eight channels of the Oberheim OB8, five channels of the Sequential Circuits Prophet, two Moog Minimoogs, the ARP

. . . *continued on page 64*

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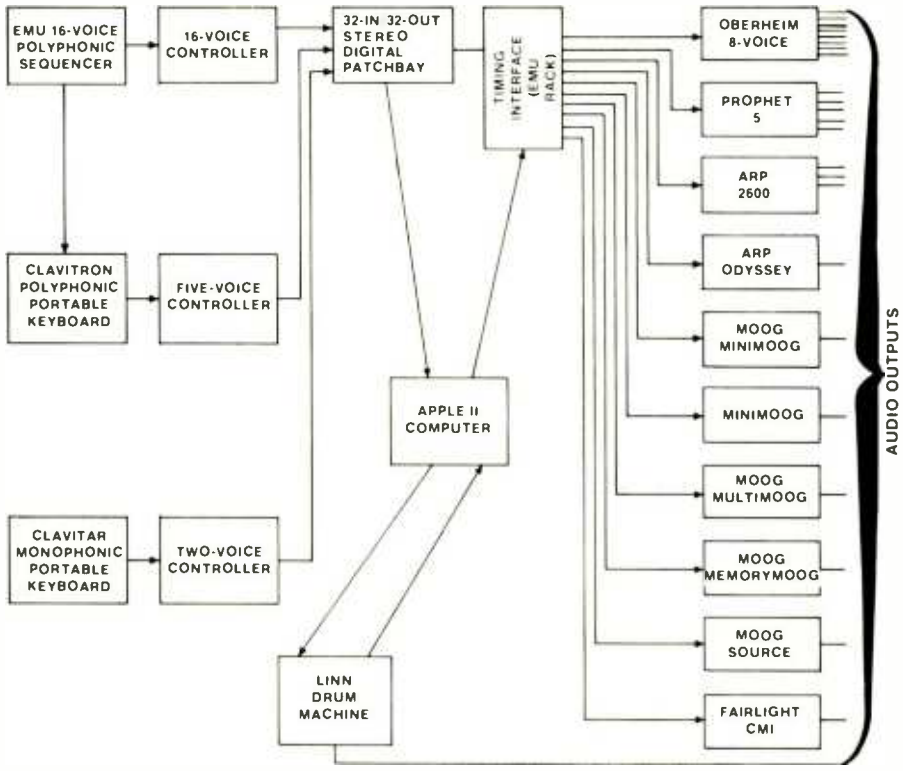
Odyssey, and the two ARP 2600s.

Tuning is effected by control knobs, each one of which displays through a window on its face three numbers representing the pitch frequency of the oscillator it controls (for example, 256 for middle-C). By moving a guitar cord from one channel of the E-mu to one of the Oberheim, the voice changes, but the player can still control the music from th E-mu keyboard or its internal memory.

After dealing with the lack of standardization as it applied to pitch, Bell tackled the problems associated with gates, S-triggers, and various other nuisances attendant in diverse synthesizer designs. "In the same tuning device, we made a trigger conversion unit, so regardless of whether the instrument is a Moog, an [ARP] Odyssey, or Oberheim, you can plug it straight into the system, and it will play. We also have a portable, monophonic keyboard, called a Clavitar, that can be dialed into eight separate instruments through the tuning interface with a simple on/off operation. You can even do multing.



LAYOUT OF HERBIE HANCOCK'S KEYBOARD SYSTEM

"One of the great sounds we had was a Minimoog and ARP 2600 playing simultaneously, in tune, and completely in sync from the portable keyboard. Herbie was doing overdubs on the session from a director's chair in the control room with the keyboard in his lap. He could dial in any of the instruments in the other room, and play them through a screaming Boogie [guitar amp] in the studio."

For compositional applications, Bell's next project was a master rhythm clock capable of driving tape sync, as well as the Linn Drum Machine and the 16-channel E-mu. The E-mu, in turn, drives an entire eight-voice Oberheim, entire Prophet, two Minimoogs, and an ARP 2600 all at once.

The Linn and E-mu Sequencer were made to control other devices rather than be driven. Now the Drum Machine and Sequencer can be used to write music by storing all the orchestration, for the whole song in the sequencer's random-access memory.

"We also developed a master clock divider network," Bell remembers, "that enables Herbie to play 16-part symphonies, plus a live instrument like Clavitar, all at the same time. He can write a song, and play everything live in digital without even using tape. Everything locks up to the drum machine. He can work on tempos, pitch arrangement, etc. Basically [it's] a rehearsal with the whole band, but he's all alone."

To date, the only song released on a Herbie Hancock record with a Linn Drum Machine as the actual "drummer" is "Textures" from the *Mr. Hands* album. The bass, vocals, flutes, strings, horns, acoustic piano, and drums were all done by Hancock. Most other compositions are written into the sequencers with the Drum Machine, and then played back as a click track during the session. Real drummers and bass players are hired to play along with the sequencers.

The Future

Where will all these advancements and innovations lead? Stephen St. Croix has an opinion that seems to sum up the attitudes of many of those involved with playing and recording synthesizers.

"Synthesis will be the music of the future," he offers. "Eventually all contemporary popular music will be synthetic; there still will be sax and guitar players, for example, but they'll comprise maybe 10% of the music. And the reason is strictly economics. A guy can get an album done faster, cleaner, cheaper and more reliably synthetically, than he can with real players. I'm a serious synthesizer player, but I've also been recording my old Les Paul, Strat, or other personal favorites, for 18, 20 years. And I would hate to see us lose the skill of forcing a mechanical instrument into its n'th degree of expression. But it will be lost. The power per dollar of synthetic music is too severe to ignore.

"Programmed synthesizers, like the Jupiter 8 and the OBX, are growing in popularity because you can push a button and a conga comes up; another button and a marimba is there. In a session today, as opposed to even one year ago, the attitudes are changing. The producer would say 'Let's get a marimba for this part.' Now the engineer says: 'In the mix, the OBX will sound the same. It's here now, and we own it; it won't be triple scale.'

"As long as engineers and musicians realize that synthesizers can solve problems, they'll be in the studio. As long as the synths are in the studio, the temptation to use them to fake instruments will exist, too. And as technology becomes more accurate, that's all we'll need."

* * *

Recommended Reading

The Complete Guide to Synthesizers, Devarahi; Prentice Hall, Inc., Englewood Cliffs, New Jersey 07632; 1982. $13.95. (An excellent book, complete with explanations of most analog and digital synths on the market. The text also offers detailed descriptions of the various sections of a synthesizer, as well as diagrams and discussions of suggested patchings for particular sounds and functions — *RC*.) ■■■

REVERBERATION TIME CHARACTERISTICS OF RECORDING STUDIOS AND CONTROL ROOMS

— Quantifying the Selection of Absorbent and Non-Absorbent Materials to Achieve a Flat Reverberation Time Response at All Frequencies

by Michael Rettinger, Consultant on Acoustics

Studios and control rooms are like people, in that no two of them are alike. Yet, to assure a uniform recorded product, it is necessary to employ acoustically equivalent listening enclosures. It is not sufficient, for instance, simply to have a "flat" reverberation time characteristic in a control room, but one in which this can be achieved by employing only highly reflective and highly absorbent materials, for the following reasons:

1. Commercial acoustic products do not produce sound reflections that are frequency-independent. Thus an acoustic tile whose absorptivity is 0.1 at 100 Hz and 0.8 at 1 kHz will, by definition, reflect 90% of the sound energy density at 100 Hz, but only 20% at 1 kHz — which, in turn, is equivalent to a 1 dB reduction in the reflected sound pressure at 100 Hz and a 7 dB reduction at 1 kHz. Figure 1 provides a graphic illustration of this phenomenon.

2. While the point discussed above pertains to random sound incidence, the absorptivity of an acoustic tile is a function also of the angle of sound incidence. Thus, a listener near a wall covered in the acoustic material described above will receive a "boomy" first reflection, heavy in low frequencies.

3. A flat reverberation time characteristic in a room can be obtained with a wide variety of products, usually so chosen that the lack of absorptivity of one product in one frequency region is compensated by the high absorptivity of another product in the region where the first one is deficient. The discrete placement in an enclosure of tuned Helmholtz units with sharp absorption peaks at various frequencies does not remedy the effect of irregularly-shaped reflection characteristics, especially for the first- and second-order reflections.

4. Absorptivity coefficients of many commercial acoustic products generally are not available below 125 Hz, so that frequently much guesswork is involved in designing an enclosure with a flat reverberation time characteristic from 50 Hz to 5 kHz. A highly absorbent product on the other hand, such as a "bass trap," and a highly reflective material, like two-inch thick wood panelling or a concrete floor, is not nearly so much afflicted with an irregular wide-band reflection spectrum.

5. The acoustic design of critical recording and mixing rooms employing a wide variety of products to establish a desired interior decor becomes cumbersome, expensive, and uncertain, and generally does not lend itself to changes in the reverberation time of the room, as might be done by simply covering a highly absorptive area with a highly reflective one, without changing the shape of the reverberation-time characteristic.

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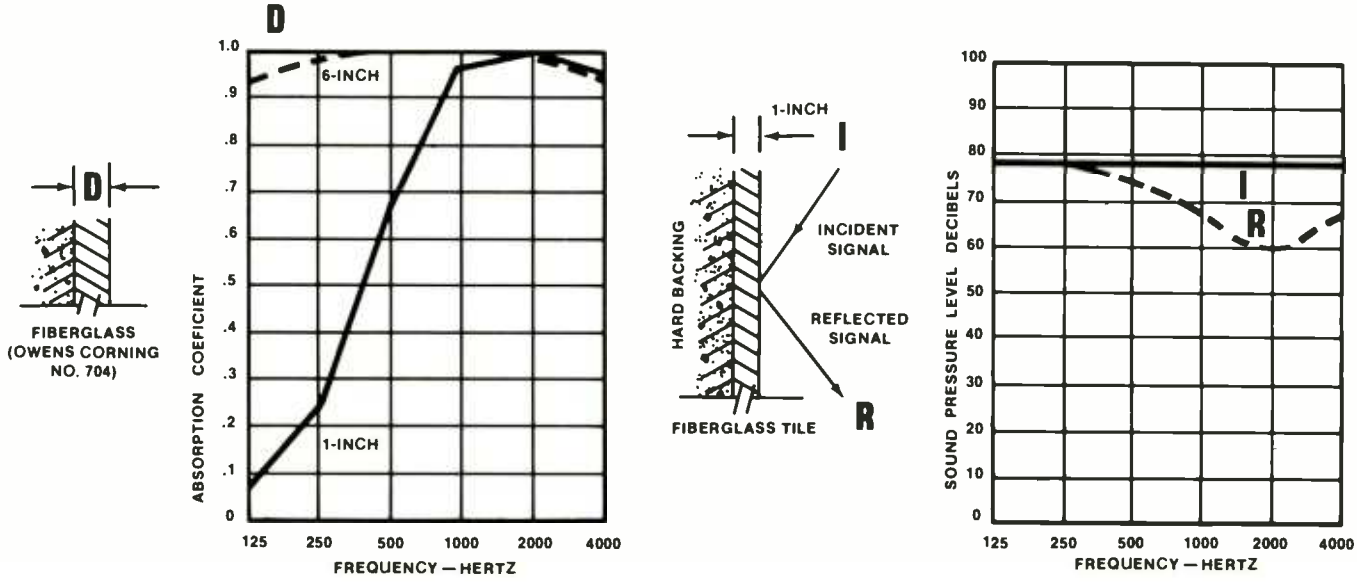
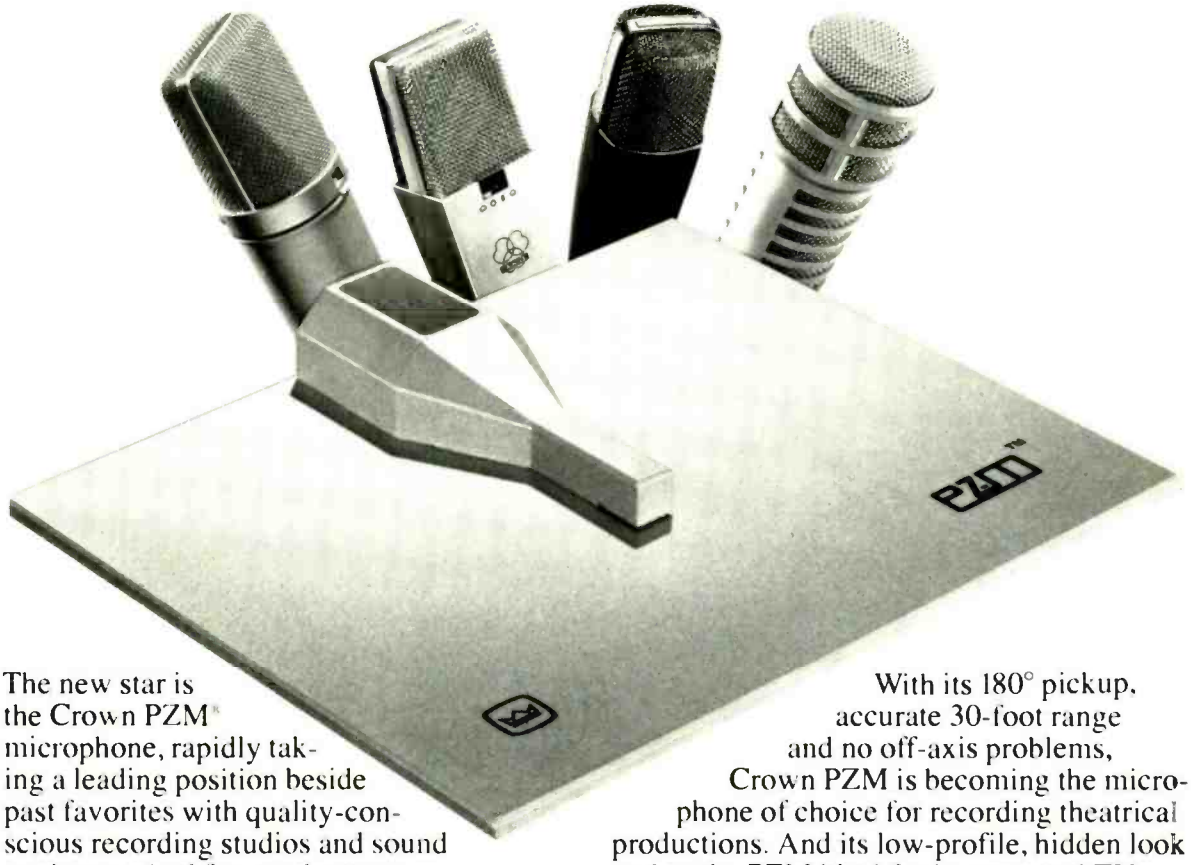
Figure 1: I is spectrum of incident signal; R is spectrum of reflected signal.



STAR PERFORMERS



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Acoustic Design
REVERBERATION TIME CHARACTERISTICS

As an example of this type of acoustic design, consider a control room with average dimensions 10 by 20 by 30 feet; with a volume, V, of 6,000 cubic feet; a total interior surface, S, of 2,200 square feet; and which is to have a reverberation time, T, of 0.3 seconds at all frequencies.

By the Eyring-Norris equation we may write for the average sound absorption coefficient, $\bar{a}$:

$$-\mathrm{Lg}(1-\bar{a}) = 0.05V/ST = (0.05\times6000)/(2200\times0.3) = 0.45$$

From the above equation we determine $\bar{a}$ to be 0.365.

Assuming that for the very absorptive section we achieve an absorption coefficient of 0.95, and for the very reflective section an absorption coefficient of 0.05, we may write this as:

$$\bar{a} = (0.95S_1 + 0.05S_2)/S = 0.365 = [0.95S_1 + 0.05(S-S_1)]/S$$

Where S_1 is the absorptive area of the room, and S_2 is the reflective one, so that S_2 is equal to $(S - S_1)$. By this arrangement we obtain that S_1 is 770 square feet, and S_2 1,430 square feet.

Figure 2 shows a proposed acoustic treatment for the example control room.

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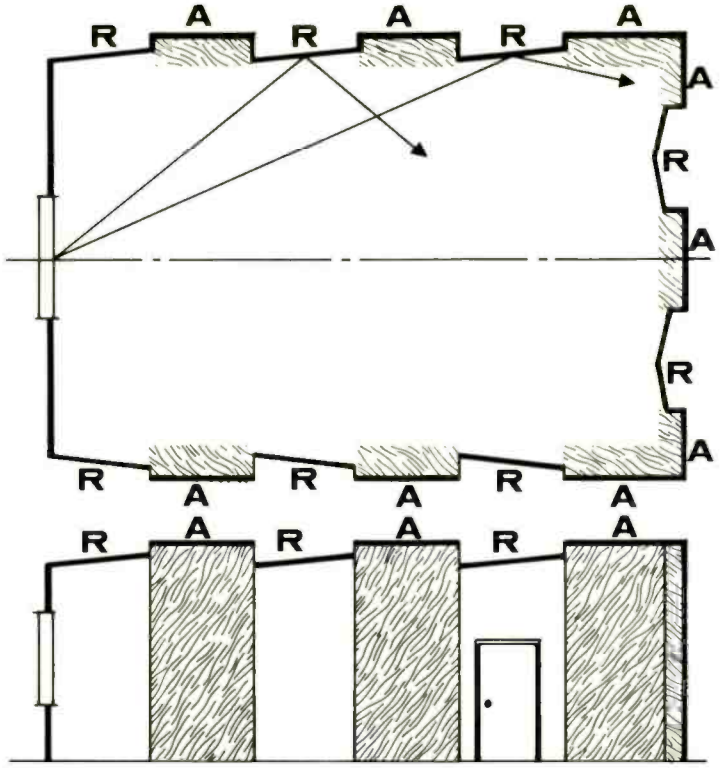

Figure 2: Plan and elevation of control room with highly reflective and highly absorptive wall and ceiling treatment to achieve flat reverberation time characteristic without frequency-discriminatory reflections.

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STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

Northeast:

☐ **MASTER SOUND PRODUCTIONS** (Long Island, New York) has purchased an Ampex ATR-124 to augment its expansion to audio and video post-production. The ATR-124 is locked to an Ampex MM-1200 multitrack and video tape recorder via SMPTE timecode. *Long Island, NY.*

☐ **MARK CUSTOM RECORDING SERVICE** (Clarence, New York) has added a second studio, Studio A, which will feature an AMEK 3000 34 by 32 console with Allison 65K automation in Studio A; and the existing Studio B is now equipped with Auditronics 110B 24/8/16 desk. Tape machines comprise an Otari MTR-90 with 24- and 16-track head stacks, an Ampex ATR-102 and Studer A80 two-tracks. Studio B utilizes an Otari MX-5050B 8-, four-, and two-track. UREI, JBL, and Auratone monitors are powered by UREI and Crown amps, while effects include an EMT 240 Gold Foil, EMT 140S Tube, Lexicon 224X, AKG BX20E and BX5E, Lexicon PCM41, DeltaLab DL4, MXR Pitch Transposer, Cooper Time Cube, UREI parametrics, and an EXR Exciter. Mikes are by Neumann, AKG, Sennheiser, Beyer, Shure, PZM, and Electro-Voice, and the instrument list now features a Kawai grand piano, Prophet 5 synthesizer, Polymoog, and a Rhodes stereo electric piano. *10815 Bodine Road, Clarence, NY 14031. (716) 759-2600.*

☐ **EVERGREEN RECORDING** (New York City) has upgraded its facility with control room re-design, and the addition of two isolation booths to the enlarged main recording area. The new equipment is highlighted by a Harrison MR-3 console, an Ampex MM-1200 24-track, an Otari MTR-10 two-track, 24 channels of dbx noise reduction, and an Ecoplate reverb. **Joel Greenbaum** is the studio owner. *215 West 91st Street, New York, NY 10024. (212) 362-7840.*

☐ **BEARSVILLE STUDIOS** (Bearsville, New York) has completed a new control room for Studio B, according to studio manager **Griff McRee**. The new room, designed by **George Augspurger**, and features a 32-input Neve 8068 console, Studer A80 multitrack and two-track, a B67 two-track, UREI monitoring, an EMT 251 digital reverb. *P.O. Box 135, Bearsville, NY 12409. (914) 679-7303.*

☐ **UNIQUE RECORDING** (New York City) has expanded its MCI JH-636 automated console to 36-inputs and nine echo returns. *701 Seventh Avenue, New York, NY 10036. (212) 921-1711.*

☐ **EASTERN ARTISTS RECORDING STUDIO** (East Orange, New Jersey) has installed a pair of Electro-Voice Century 100 monitors to complement the new APSI 32/24 console. On the personnel front, **Tina Verras** has been named maintenance engineer, and **Marc Marseglia** added as assistant engineer. *36 Meadow Street, East Orange, NY 07017. (201) 673-5680.*

☐ **TROD NOSSEL RECORDING STUDIOS** (Wallingford, Connecticut) has added an MXR Pitch Transposer, along with a UREI LA-5 limiter, dbx Model 157 noise-reduction unit, two RCA BA6A tube limiters, an E-V 666 microphone, and a Shure 315S ribbon mike. *10 George Street, P.O. Box 57, Wallingford, CT 06492. (203) 269-4465.*

☐ **THE REVIEW ROOM** (New York City) is the new recording studio constructed by **Dave Grusin** and **Larry Rosen**, co-principals of GRP Records. The room will be equipped with the new JVC DAS Series 90 Digital Audio Mastering and Editing System. The facility also features a Mitsubishi wide-screen video projection system, JVC ¾-inch videocassette player, and a BTX Shadow System for synchronizing audio to video with SMPTE timecode. A customized twin Ramsa console with 32 inputs is interfaced with such outboards as dbx 106X limiters, Orban parametric EQ, a pair of Pultec tube equalizers, Lexicon PCM41 digital delays, four Valley People Kepex II noise-gates, an MXR Flanger Doubler, and an AKG BX-10 echo device. Recording duties are handled by a Studer A80 24-track, a 500-watt McIntosh MC2500 amp powers UREI 813A studio monitors for analyzing test pressings. *555 West 57th Street, New York, NY 10019. (212) 245-7033.*

Southeast:

☐ **SHEFFIELD RECORDING** (Phoenix, Maryland) has outfitted its 30-foot mobile truck with a new Audiotronics console to feed the on-board 24-track recorder. Also added are two Thompson 601A video cameras, a Chroyn TG-3, and an Ampex VPR-800 one-inch video recorder. *13816 Sunnybrook Road, Phoenix, MD 21131. (301) 628-7260.*

☐ **BEE JAY RECORDING STUDIOS** (Orlando, Florida) now has a Studer A800 in Studio A, and has moved its Stevens 32/24-track into Studio B. *5000 Eggleston Avenue, Orlando FL 32810. (305) 293-1781.*

☐ **OFFORD STUDIO** (Atlanta, Georgia), producer **Eddy Offord's** facility located in the East Point Theatre, has opened for outside bookings, and is now offering mastering in half- and quarter-inch formats. The studio also has added Lawson Echo plate to its outboard rack. *1493 Jefferson Street, Atlanta, GA 30044. (404) 766-5143.*

☐ **ROAR PRODUCTIONS** (Columbia, Maryland) is a new facility that features tripled studio space, separate drum and vocal isolation booths, new control room monitors, and a client lounge. **Gary Zeichner** has been hired as operations manager and staff engineer. *6655-H Dobbin Road, Columbia, MD 21045. (301) 596-0600.*

☐ **STRAWBERRY JAMM** (West Columbia, South Carolina) has added two UREI 1176 compressor/limiters, and a Shure SM-58. The studio has also isolated the electronic metronome (click-track) in the control room. *3964 Apian Way, West Columbia, SC 29169. (803) 356-4540.*

☐ **CRESCENDO RECORDING STUDIO** (Atlanta, Georgia) has refurbished and decorated its two studios, and added a complete in-house video production suite with video and audio interlock. Both studios can act as sound stages, and are equipped with video monitoring. Harrison 32/32 and Auditronics 501 26/24 consoles are fully automated, and complement Otari, Studer, and Ampex 24-track recorders equipped with dbx or Dolby noise reduction. Each room features an array of signal processing units, and an instrument list that includes a Bosendorfer concert grand piano, Musser vibes, various synthesizers, drum kit, Hammond B-3, and a kazoo. *125 Simpson Street North West, Atlanta, GA 30313. (404) 223-0108.*

☐ **DOPPLER STUDIOS** (Atlanta, Georgia) has added **Joel Neil** to its engineering staff. The studio also has opened a complete tape duplication facility with mag transfers, and open-reel and cassette operations. *1922 Piedmont Circle NE, Atlanta, GA 30324. (404) 873-6941.*

Mid-West:

☐ **AUDIOGRAPH PRODUCTIONS** (Haslett, Michigan) has added a Neotek Series II 32/24 console, Eventide H910 Harmonizer, Lexicon 93 Prime Time, Valley People Stereo Dynamites, two Valley People Kepex II gates and two Gain Brain IIs, UREI 1178 limiter, UREI dual parametric EQ, and Orban 526A Sibilance Controller. New mikes include two Neumann U-87s, two AKG 451Es, two more Sennheiser 421s, and two Crown GP30 PZMs. *6285 West Reynolds Road, Haslett, MI 48840. (517) 339-1049.*

☐ **SOUNDTREK RECORDING STUDIOS** (Kansas City, Missouri) has opened its newly remodeled Studio I, designed by **Steve Durr & Associates** of Nashville, with **D.A. Petersen** of Omaha, Nebraska, acting as contractor. Flanner's Pro Audio supplied the electronics, led by a Neotek Series III 28/24 console, UREI 813 monitors, and an Otari MTR-90 24-track. **Ron Ubel** is Soundtrek's owner. *3727 Broadway, Kansas City, MO 64111. (816) 931-TREK.*

☐ **STAR BEAT RECORDING STUDIOS** (Chicago) has purchased a new Otari MTR-10 two-track machine from Flanner's Pro Audio. **Steve Cronen** is Star Beat's owner. *9 East Larksdale Drive, Dearfield, IL 60015. (312) 945-3555.*

☐ **FIFTH FLOOR RECORDING STUDIOS** (Cincinnati, Ohio) has taken delivery of a new MCI JH-114 24-track recorder. *517 West Third Street, OH 45202. (513) 651-1871.*

☐ **DOMAIN COMMUNICATIONS** (Chicago) has taken installation of a new Neotek Series IE 16/8 console custom designed for production use; Flanner's Pro Audio was supplier of the studio hardware. *Box 337, 289 Main Place, Wheaton/Carolstream, IL 60187. (312) 668-5300.*

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STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

Southern California:

☐ **SKIP SAYLOR RECORDING** (Los Angeles) has added an Ampex ATR-102 two-track recorder with quarter- and half-inch mastering capability via interchangeable head stacks. *506 North Larchmont Boulevard, Los Angeles, CA 90004. (213) 467-3515.*

☐ **CONWAY RECORDING STUDIO** (Hollywood) control room has been enlarged and equipped with an Augspurger two-way monitor system. Other recent equipment purchases include a Studer A800 24-track recorder, two Studer A80RC half-inch recorders, two Marshall tape deck eliminators, four AKG C460 mikes, an Audio Kinetics Q-Lock synchronizer, and an Ampex MM1200 24-track recorder. Studio owner, **Buddy Brundo** has been making local news as donator of a new video game system to a crippled child whose home was ransacked and robbed. Conway engineer **Bill Hudson**, who delivered the gift and was "caught" by a local newsman, said Brundo read a newspaper account of the boy's plight and said he just "couldn't stand seeing people like that suffering." [Nice going, Buddy!] *655 N. St. Andrews Pl., Hollywood, CA 90004. (213) 463-2175.*

FUTURE DISC SYSTEMS

☐ **FUTURE DISC SYSTEMS** (Hollywood) is a new, custom-designed disk mastering facility, featuring a new Cybersonics MC2003 disk-mastering console interfaced with a Cybersonics DM2002 lathe. The desk can store as many limiting, EQ, and level changes as necessary in the transferring of the master tape to disk. The pre-cut rundown is stored in digital form by the console, and can be immediately repeated or dumped on to any storage medium for retrieval at a later date. The lathe is outfitted with the new Ortofon "Green-Head" cutting system. Playback machines are by Ampex, in both quarter- and half-inch formats. **Steve Hall** will act as the house engineer, with mastering credits for Pat Benatar, Blondie, and The Crusaders, among others. **Gary Rice** is the facility's general manager. *3475 Cahuenga Boulevard West, Hollywood, CA 90068. (213) 876-8733.*

☐ **FIFTY-FOUR EAST SOUND RECORDERS** (Pasadena) has added an automation package to its API 44/40 console. *54 East Colorado Boulevard, Pasadena, CA 91105.*

☐ **KENDUN RECORDERS** (Burbank) has named **Howard Steele** as chief engineer for the facility. Steele has worked in studios all over the world since 1960, designed and built many consoles and recording studios, and founded Quantum Audio Labs. Other personnel changes include **Matthew E. Vertin** to the posts of assistant sales manager and night studio supervisor, while **Darryl Caseine** has been promoted from the latter post to the positions of traffic manager and assistant sales manager. *Burbank, CA.*

MARS STUDIO

☐ **MARS STUDIOS** (Hollywood) has merged with Spectrum Studios, formerly of Venice, California, with the latter's staff relocating to the Hollywood operation to open a new 24-track analog and digital recording facility. The main music room measures 25 by 35 feet, with a 25-foot ceiling and two large isolation booths. The control room, designed by **Jack Edwards** and tuned by **George Augsperger**, measures 20 by 20 feet with a monitoring system designed by Augsperger. The complex also contains three large rehearsal studios with sound stages, and is equipped with a Harrison automated console, 3M Series 79 recorders, Sony PCM-1600 digital processor, and boasts a staff headed by owner **Stan Gittelman**, manager/chief engineer **Arne Frager**, and engineers **Joe Q. Hall, Scott Skidmore**, and **Michael Boshears**. Scheduled for delivery in early 1983 is a new Otari 24-track, an Audio Kinetics Q-Lock synchronizer, Eventide digital reverb, and the construction of a second live chamber. *665 North Berrendo Street, Hollywood, CA 90004. (213) 660-6334.*

Northern California:

☐ **AUDIO-VIDEO RESOURCES** (San Francisco) has hired **Troy Alders** as director of media services for the newly remodelled eight-track facility, which also features audio/video interlock for ¾-inch post-production. *60 Broadway, San Francisco, CA 94111. (415) 781-2603.*

☐ **BODACIOUS AUDIO** (Sausalito) has relocated its mobile recording operation to the **Harbour Sound** complex in Sausalito. The audio truck features full 24-track remote services, and an Ampex half-inch, two-track ATR-100 for mixdown and mastering. Harbour will still be offering fully automated 24-track recording with expanded services to include digital and half-inch two-track mastering, full real-time tape duplication and transfer and other related services. *301 Harbor Drive, Sausalito, CA 94965. (415) 331-7559.*

☐ **HEAVENLY RECORDING STUDIOS** (Sacramento) recently engaged **George Augspurger** of Perception, Inc. to tune its control room and studio monitoring systems. *620 Bearcut Drive, Sacramento, CA 95814. (916) 446-3088.*

☐ **THIRD EAR RECORDING STUDIO** (Solana Beach) has moved and expanded to the **Belly Up Tavern**, and now offers live eight-track recording from the venue's stage; future plans include a 24-track operation in a separate recording studio. The stage measures 270 square feet, with an 8- by 8-foot drum riser. Recording gear includes a Soundcraft 26/4 feeding an Otari eight-track MX-5050 MKIII, Otari MX5050-B two-track, and Tascam 70-2 half-track. UREI 811 monitors, Delta Lab DL-4 digital delay, MICMIX Dynafex noise reduction, EXR Aural Exciter, and dbx compressor limiter are also featured. Mikes are by Shure, Beyer, AKG, and Sennheiser. *Belly Up Tavern, 143 South Cedros, Solana Beach, CA 92075. (619) 436-4681.*

☐ **CORASOUND RECORDING** (San Rafael) has added a new Otari MTR-90 16/24-track with full remote and autolocator. Sideboards now feature URSA MAJOR Space Station, Lexicon 224 digital reverb, and a Lexicon Super Prime Time DDL/effects unit. *122E Paul Drive, San Rafael, CA 94903. (415) 472-3745.*

☐ **HYDE STREET STUDIOS** (San Francisco) has added an ADC Stereo Sweep Equalizer, and two Electrodyne Limiter De-essers to Studio C. Studio D now features an additional stereo Valley People Dynamite package, while Studio A has added six Aphex parametrics and a pair of AKG C12 mikes. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

The Philippines:

☐ **AD & AD RECORDING STUDIOS** (Manila) is now operating with three fully-equipped recording studios. Studio A features a Neve 28/24 8158 console with NECAM automation, feeding an Ampex MM-1200 24-track, and three Ampex ATR-100s for mixdown. Studios B and C boast Neve 24/16 5316 consoles, and Ampex multitrack and mastering decks. All studios are equipped with UREI graphic equalizers, full Dolby noise reduction, Aphex Aural Exciters, EXR Exciters, MXR and Delta Lab digital delays, ADR Vocal Stressers, Eventide Model 949 Harmonizers, Lexicon 224 digital delay units, and an array of reverb devices. Microphones are by Shure, AKG, Sennheiser, Neumann, Beyer, PZM, and Crown, while the instrument list features three Wittemberg Grand Pianos, two Fender Rhodes pianos, Wurlitzer electric organ, Sequential Circuits Prophet 10 synthesizer, guitars by Fender, Martin, Gibson, Tama, and Ovation, and drum kits by Rogers and Ludwig. Monitoring in the orchestra-sized Studio A is handled by UREI Time-Aligned speakers and Auratones, with Studio B and Studio C offering JBL 4343s and Auratones. All the speakers are powered by McIntosh amps. AD & AD also operates its own Neumann disk cutting lathe, and Worldex record pressing plant. **Aileen B. Dumlao** is AD & AD's studio manager. *304 Shaw Boulevard, Mandaluyong, Metro Manila, Philippines 3119. 780-866.*

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For additional information circle #14

STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

Australian Activity:

□ **YORK STREET RECORDING STUDIO** (North Fitzroy) has added an Auditronics 501 24/16/24 console, and an Otari MTR-90 MKII 16-track, pre-wired for 24-track. The recent upgrade is expected to provide the facility with greater flexibility in production of film soundtracks. **Richard Brobyn** has joined York Street as managing engineer. *Rear 62 York Street, North Fitzroy, 3068 Australid. (03) 481-1627.*

Canadian Activity:

□ **MID-OCEAN RECORDING STUDIO** (Winnipeg, Manitoba) has upgraded from 8- to 16-track operation with the purchase of a Tascam 85-16 multitrack equipped with dbx and full remote controls; a Tascam M-15-24/8 console; and a Studio Technologies Ecoplate II reverb. The system is wired through a tip, ring, sleeve patchbay designed and installed by **D. Grant Jassen.** Eight new pairs of Fostex T-20 headphones round out the system. **Dave Zeglinski** is the facility's chief engineer. *1578 Erin Street, Winnipeg, Manitoba, Canada R3E 2T1. (204) 774-3715.*

MID OCEAN RECORDING

Great Britain:

□ **BRITISH BROADCASTING CORPORATION** (Glasgow, Scotland) has taken delivery of two Soundcraft consoles: the first is a 2400 24/16 Series, which will be used in television post-production work interfaced with a 16-track recorder and video machines, while a 28/24 Series 2400 is to be installed in Glasgow Studio 2 for music recording. *Glasgow, Scotland.*

Venezuala:

□ **RADIO CARACAS TELEVISION** (Caracas) recently completed a 24-track studio with a Live-End/Dead-End control room design by **Chips Davis** with HNE, Inc., of Burbank, California. The new facility features a Sphere 32-input Eclipse console, feeding an ATR-124 24-track and ATR-100 four- and two-track machines. UREI 813A Time-Aligned speakers handle the monitoring, while 24-tracks of Dolby provide noise reduction. Outboards include digital delay, echo plate, and an Eventide H949 Harmonizer. Mikes are by Shure, PZM, and Neumann. **Sherman Keene,** author of the book *Practical Techniques for the Recording Engineer,* has been invited to the studio to train the facility's staff. *Caracas, Venezuela.*

— AUDIO/VIDEO UPDATE —

Eastern Activity:

□ **ONOMATOPEIA** (New York City) provided audio services for the live satellite broadcast of Broadway's *Sophisticated Ladies.* The program's producer, **Oak Communications,** also used Onomatopeia's president **Matt Kaplowitz** as music producer for the show. Kaplowitz worked with music conductor **Mercer Ellington** to adapt the live theatre sound to the television medium. Dolby's DS-4 Multidimensional Sound Enhancement process was utilized on the videocast, as well as Crown PZM microphones substituted for the traditional shotgun mikes used in stage work. Kaplowitz engaged The Record Plant's "White and Black" trucks to handle, respectively, the live transmission and the recording of the event for subsequent videocassette and disk release. **Dave Hewitt** served as chief engineer, while **Bob Liftin** of Regent Sound provided audio post-production. *37 West 57th Street, New York, NY 10019. (212) 688-3167.*

□ **DEVLIN PRODUCTIONS** (New York City) provided one-inch video transfer of the original 16mm film material for Pat Travers' recent music video, *I'd Rather See You Dead.* Devlin's senior colorist, **Robert Lovejoy,** cropped and repositioned film segments from the footage using X-Y-ZOOM in Devlin's Correction Suite. Insert editing was handled in the company's CMX room. The work was produced and directed by **Ken Walz.** *150 West 55th Street, New York, NY 10019. (212) 582-5572.*

Central Activity:

□ **RON ROSE PRODUCTIONS** (Detroit, Michigan) is expanding its 24-track audio operation to include video post-production. The new fully-computerized studio allows producers to mix multichannel soundtracks to one- and ¾-inch video tape. The new gear includes a Neotek 24-channel console, Otari 24-track, Sony one-inch VTR, UREI monitors, and a 78-inch Kloss Nova Beam projection TV. Two sound booths, each with a view of the video screen, allow for live music and vocal overdubbing to picture. *29277 Southfield Road, Southfield, MI 48076. (313) 424-8400.*

□ **CEDAR CREST STUDIO** (Mountain Home, Arkansas) specializes in audio/video production services, and is equipped with modified TEAC and Fostex mixers, linked to a TEAC Tascam 80-8 with dbx DX-8 noise reduction. Control room monitors comprise JBL 4312 and Auratone Sound Cubes powered by BGW and Spec-4 amps. Outboards include MXR digital delay, compressors, noise gates, flangers, time delay, and graphic EQ, dbx Model 161 limiters, TAPCO 4400 reverb, and Univox echo. Mikes are by Electro-Voice, Shure, AKG, Sony, TEAC, and Sennheiser. Video gear includes ¼- and half-inch recorders by JVC, Sony, and Sanyo, with cameras by JVC and Sony. Outboards include a Sony HVS-2000 special effects generator, and a Vidicraft Image Enhancer. *P.O. Box 28, Mountain Home, AR 72653. (501) 425-9377*

Western Activity:

□ **THE COMPLEX** (Los Angeles) provided its soundstage to **Linda Ronstadt** for the production of a 35mm promo film of her album, *Get Closer.* Filming was directed by **Richard Namm,** with lighting and camera work by **David Lewis. Ed Maloney** of The Complex managed the staging for the production of the promotional short. *2323 Corinth Street, West Los Angeles, CA 90064. (213) 477-1938.*

□ **RUSSIAN HILL RECORDING** (San Francisco, California) has fitted Studio A with an Audio-Kinetics Q-Lock 3.10 synchronizer, coupled with a video projector system for use in film and video/audio post-production. *1520 Pacific Avenue, San Francisco, CA 94109. (415) 474-4520.*

□ **SFO PRODUCTIONS** (San Francisco, California) produced the video/animation and music project by **The Toons,** a California-based acappella-pop fusion band. In the piece, entitled *Video Games,* SFO producer **Jeff Daly** depicted the band as "video-crazed" students who disrupt the school work of their fellow fourth-grade classmates. Cameraman **Robin Hirsch** used his specially-developed 4.5mm wide-angle lens mounted on an Ikegami HL-79DA camera inside a video machine to give a games point of view. Incorporating the animation with layered digital and chromakeyed effects was handled by editor **Mike Certic** of Positive Video of Orinda, California. Editing was done on a CMX-340X system linked to a Sony BVH 1100A. *P.O. Box 16035, San Francisco, CA 94116. (415) 621-3434.*

□ **MUSIC ANNEX RECORDING STUDIOS** (Menlo Park, California) is constructing a sound stage and audio/video control room adjacent to its existing Studio C. Acoustical design consultation is being handled by **George Augspurger,** with completion of the project slated for early Spring. The design calls for a central control room, two studio areas — 35 by 45, and 30 by 20 feet — and an isolation booth. The larger of the two studio areas will have a lighting grid, a 20-foot ceiling, and a 180-degree hard cyclorama wall. A 28-input AMEK console will feed an MCI 24-track recorder in the control room. *970 O'Brien Drive, Menlo Park, CA 94025. (415) 328-8338.*

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Glen Rosentein (left), Gerry Block, Jim Dougherty (right), at Sigma Sound

Neve console in Electric Lady's Studio A

Joe Gaswirt in Frankford Wayne's Room F

ON THE STUDIO TRAIL

Mel Lambert at Large this month in . . . NEW YORK CITY

While attending the SMPTE conference and Exhibition in New York during November, I had the opportunity to visit and talk with engineers and production staff at five interesting studios — **Skyline, Electric Lady, Atlantic, Sigma Sound,** and **Frankford/Wayne Mastering.**

Skyline specializes in demo and album work, and has been doing a fair number of sessions for CBS Records and several East Coast independent labels.

Assistant studio manager Dave Young tells me that the single room, equipped with a Neve 28-channel 8058 console, Studer A80 24-track, Ampex ATR-100 and Studer A80 mastering decks (the latter with half- as well as quarter-inch headblocks), billed around a half-million dollars last year, and currently runs full occupancy for between 12 and 16 hours per day, seven days a week.

Questioned about the studio's operating philosphy, Young offered that "Skyline doesn't want technical considerations to stand in the way of musicians getting their performance on tape. We aim to simply help the translation from one idiom to another. When we deal with established or professional bands and artists, who usually are comfortable with working in a studio environment, it becomes a pretty 'industrial' process. Our aim is towards 'informality,' but not at the expense of our technical expertise."

Young concedes that Skyline isn't particularly typical of New York studios. The studio's four house engineers tend to record combinations of instruments in the same area, and not worry too much about leakage between tracks. The room, which features a reasonably live acoustic treatment on the walls and ceiling, was designed so that a performance could be played and captured on tape in a fairly live way, but with sufficient separation to leave options open at mixdown. The majority of the room's acoustics was done in-house, Young recalls, and is designed to enable engineers to successfully record substantial portions of a band without the need for many overdubs. That way, he offers, not only can a band play in much the same way as it would on stage during a live performance — with improved communications between group members — but also save some money in studio time.

And this increasing emphasis on moving away from total separation on each tape track — in contrast to the practice of surrounding each instrument with gobos and screens, or else recording it alone to achieve practically total acoustic isolation — was to occur more than once during my tour of five Big Apple studios. At Electric Lady, for example, chief engineer Sal Greco pointed out that Studio A will be undergoing a major rebuild in the near future. As well as the control-room acoustics being reworked, and a new vocal booth added, the studio area will be made even more live-sounding that it is at present.

While studio staff were investigating the surface acoustic treatment on one of Studio A's long walls, and had temporarily exposed some of the underlying concrete, session engineers discovered that they liked the resultant "larger" sound. As Greco was quick to point out, however, for a live acoustic environment to be workable, any increase in reverberation time needs to be broadband — that is, for all frequencies — and not just the mid- and high-frequency regions. And to increase the RT60 of bass frequencies, there is no substitute for reflective, massive surfaces — in other words, bare concrete walls. To provide local, more adjustable areas of sound reflection, Electric Lady engineers often lay sheets of plywood on the wall behind drums, to achieve a "bigger" sound from the kit.

Each of the facility's three rooms features a popular Neve/Studer control-room combination: Studio A boasts a 32-input 8078 with DC subgrouping, linked to an A800 24-track and two A80 stereo machines; Studio B, which has a smaller recording area and is intended for overdubs and remix, offers a 32-input 8068 tied to a similar complement of tape decks as Studio A; while Studio C, located on the first floor and able to run separately from the pair of basement areas, centers

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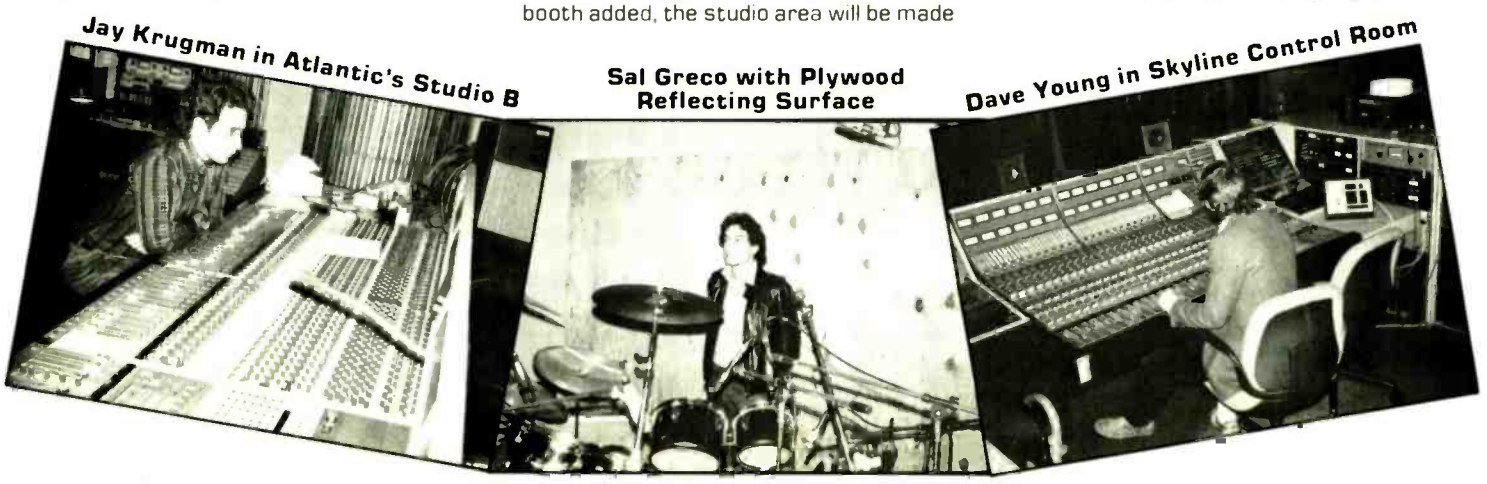

Jay Krugman in Atlantic's Studio B

Sal Greco with Plywood Reflecting Surface

Dave Young in Skyline Control Room

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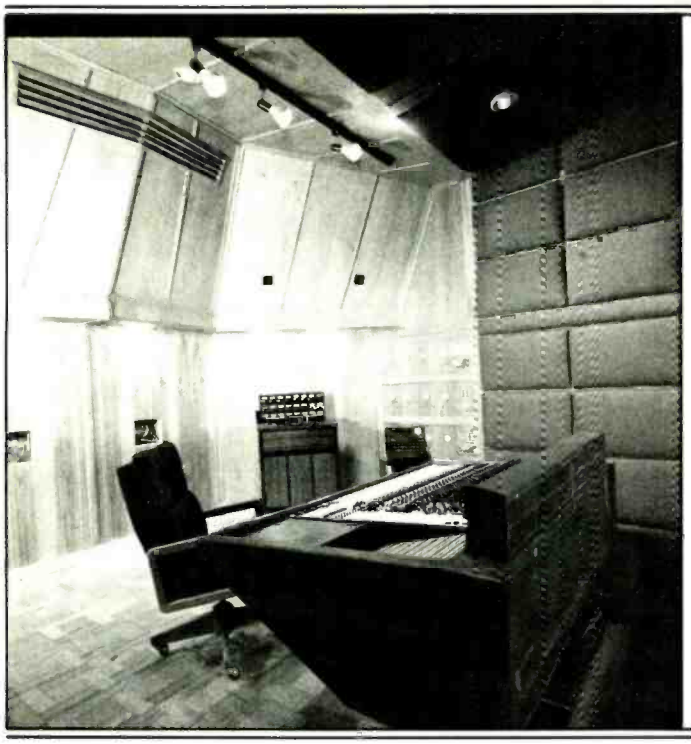

STUDIO DESIGN & CONSTRUCTION

TRES VIRGOS

One of The first Examples of a Complete Live-End/Dead-End Control Room Installation

by Robert Hodas

In an age when many studios are struggling to survive, or going out of business, people that undertake the construction of a completely new project from the ground up must be sure they have something the client is going to like. Even with all of the very best equipment, if the ambience of the studio and control room are not comfortable, musicians and producers are not going to want to come and work with that studio. Tres Virgos has taken a big chance in building a brand new studio incorporating a radical new control room design that claims exceptional transparency and clarity, plus a studio with variable ambience to meet the needs of different types of music.

The control room design the studio chose was Live-End/Dead-End™ (LEDE). (Rather than getting heavily involved in LEDE theory, it would be easier to refer the reader to the February 1979 issue of *R-e/p*, which contains an article describing in great detail the basic design principles.) Quite simply put, LEDE control rooms have the front half of the control room padded and soft for absorption, while the rear half is hard and dispersive. The idea behind this is to avoid any reflections from the front of the room — off walls, speaker cabinets, etc — occurring within 18 milliseconds of the direct sound from the monitors, which can cause comb filtering and phase smear of the direct signal. Because of the psycho-acoustic Haas Effect, rear wall reflections are rejected by the brain as not being part of the direct signal, and hereby allowing, it is claimed, for much more clarity in perception of the signal coming from the speakers. This type of control room design has recently come of age due to the work of Richard Heyser, who developed (and subsequently patented) the equipment for TDS™ (Time Delay Spectometry) measurements. TDS facilitates measurement of the spectrum in the domains of time, energy and frequency.

Studio Origins

In the very beginning, there was an eight-track garage studio situated in the hills of Marin County, California. For a couple of years the studio prospered, until a point when sessions started going beyond the sleeping hours of the neighbors. Since this was a business operating in a residential neighborhood, the zoning commission ruled against the studio, and eviction papers were served. Prior to the final eviction in 1980, plans were developed for building of the new studio, Tres Virgos.

Tres Virgos was mostly a fantasy project at the time. The original concept was to build a small 16-track studio in a location that would not "disturb the neighbors," or have the neighbors disturbing them. The more the studio potential was explored, the larger and more complex the facility became. Its owners realized that they really did not need to have a studio of their own in order to operate their business, and if

. . . *continued on page 81*

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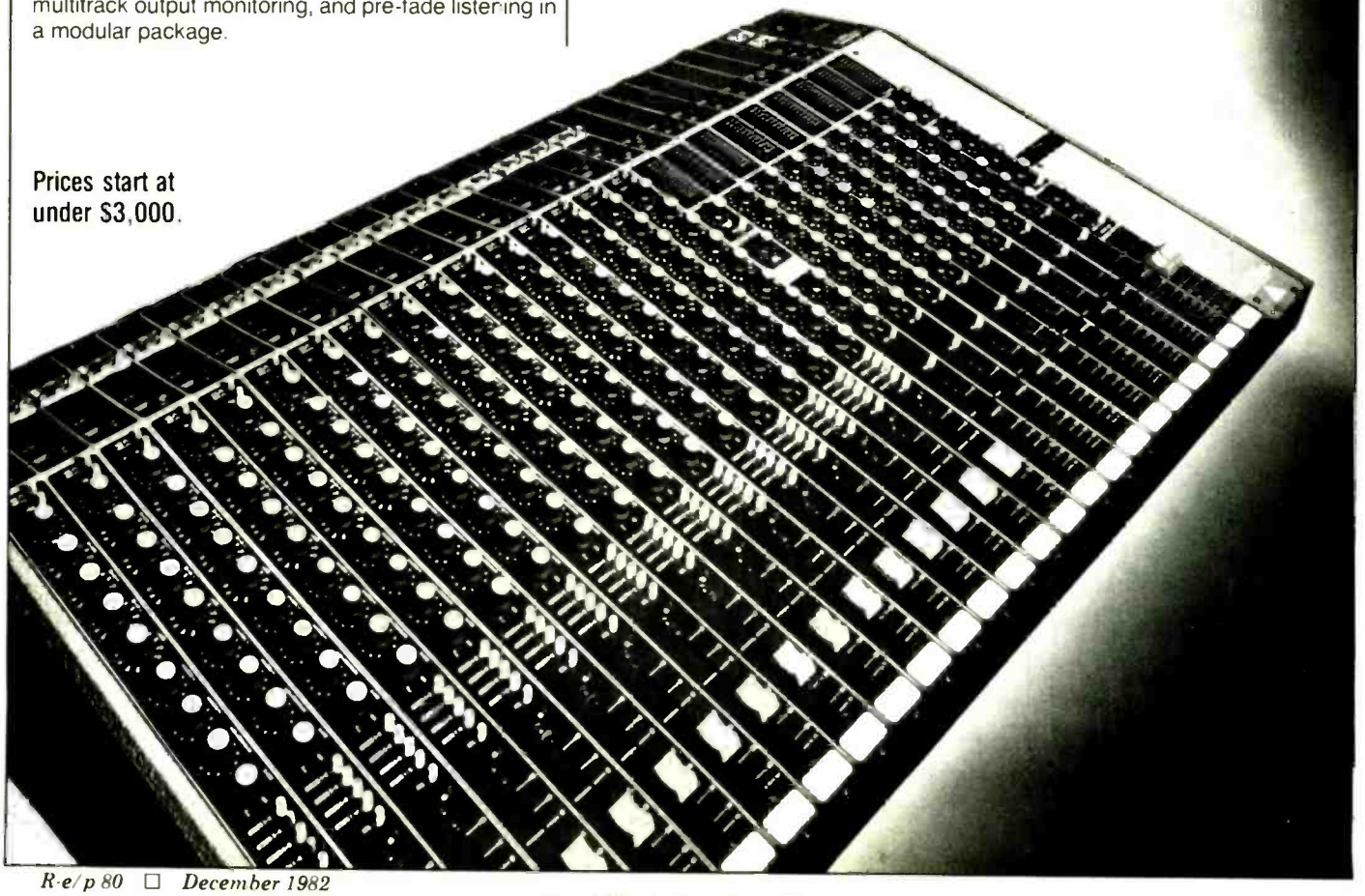
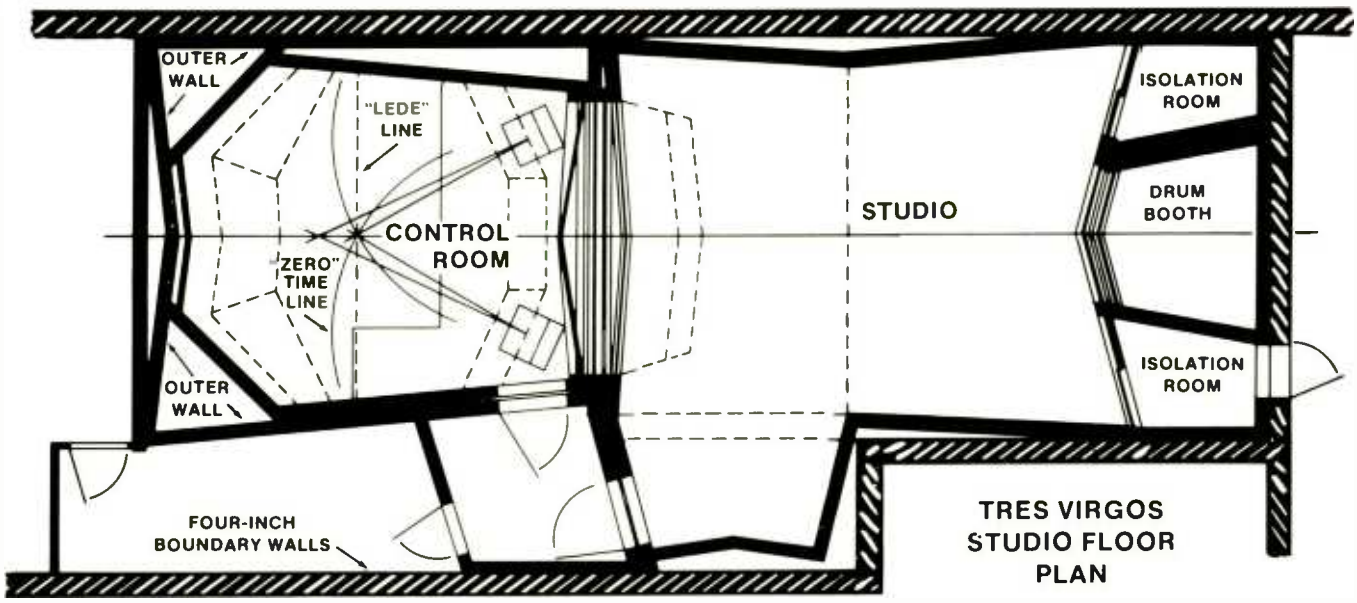




TRES VIRGOS STUDIO FLOOR PLAN

STUDIO DESIGN

TRES VIRGOS

— *continued from page 78* . . .

they were going to get involved in building one, it must be a project that other people would want to come and record in order to support their fantasies.

The original idea was that recording would be done in the studio, and not in the control room. They did not want to design the studio strictly for close miking rock 'n' roll, but for minimum mike technique. The goal was to make a studio room in which any musician would feel comfortable — one with variable ambience, where you have dead sections or live sections, and create any type of sound that one desired. From there, it was realized that in order to accurately *record*, one must be able to accurately *monitor*.

The LEDE idea first came to the eight-track facility after studio partner Alan Rice had taken a Syn-Aud-Con course, and came home excited about the concept. (For the record, Tres Virgos is owned by the partnership of Allen Rice, Robin Yeager, Mike Stevens, and Jerry Jacobs — the studio name, incidentally has its origins in the fact that the first three individuals are all Virgos.)

A pair of UREI 813 monitors were purchased and placed in front of the tiny control room. Absorbent materials were mounted over the front of the control room, and some reflective panels added to the back of the room. The whole approach was totally unscientific, and there were certainly no Time Energy Frequency (TEF) measurements to substantiate or guide the placement of reflective and absortive materials. As unscientific as the whole project was, the stereo imaging tightened up considerably; the bass end was much more clear; and the sound was impressively better. When there was no other choice but to move the studio, the idea of an LEDE control-room design was set firmly in the owners' minds.

The search was then on to find a designer who would be in tune with the goals of the owners. The search led to Chips Davis in Las Vegas, who had retrofitted his own studio with the first LEDE design. Jerry Jacob and Alan Rice flew to Las Vegas to view Davis' room, and were quite impressed with the sound quality of that retro fit. They realized then that by building an LEDE room from the ground up, the result would be a vastly superior monitoring environment. Jacob had only met a few engineers at this time, and no producers, and was the first to admit to being fairly naive as to the workings of the studio business. The excitement generated by his partners about this design had convinced him, however, that this was the way to go. The determination to build the studio right, no matter how long it would take, and to make an integrated system, with no compromises, was the goal.

CONTROL-ROOM EQUIPMENT LIST

Console: MCI JH-528B, modified with Aphex VCAs, and custom interface circuitry.
Multitrack: MCI JH-24 transformerless 16/24-track, with custom interface.
Tape Machines: Ampex ATR-100 stereo mastering deck, and two Otari MX-5050 half-tracks.
Monitor Speakers: UREI 811 and 813s, MDM-4 "Near-Fields," Auratone Sound Cubes, and assorted AKG headphones.
Power Amplifiers: Crown M-600 (with special Delta Omega cards), D-400, D-250, D-60s, and BGW 750s.
Outboard Effects: Marshall Time Modulator, Valley People/Allison Research Kepex and Gain Brains, Eventide H949 Harmonizer, Lexicon Prime Time, Studio Technologies Ecoplate, ADA stereo delay unit, and assorted compressors, limiters, spring reverbs, and equalizers.
Microphones: Assorted AKG, Beyer, Crown PZM, Electro-Voice, Neumann, Sennheiser, and Shure models.

At that point, the "commitment" was made. Commitment is a key word, as opposed to faith, simply because the partners were committed to each other, and the decision that they had made to build it *right*. It was not so much a leap of faith to go with a radical new design, but it seemed to be common sense to these gentlemen who had found, with their own experimentation, that the LEDE principles had improved the sound of their control room. It was also felt that by building an LEDE room, Tres Virgos would have something different to market in the studio scene. If the partners had gone with the standard studio design, it was felt that they would have to invest in nothing less than another high-tech, state-of-the-art studio in order to be competitive with other facilities in the area.

Design and Construction

A budget was put together, both time and monetary, and the building found in a quiet section of San Rafael where the rent was very good. The owners did not hire a construction company, but were the builders of the studio themselves. Alan Rice was the construction supervisor, the crew were musicians who had extensive construction experience, and they agreed to take portions of their pay in time credit when the studio was completed. Ed Bannon, the first licensed TEF user, was chosen as electronics designer for the new studio. His pragmatic approach and meticulous inspection went beyond putting the audio electronics into the studio; he even helped with the design itself to make sure that all integrated systems were kept on track.

The prime job of chief engineer Robin Yeager during construction was to continue working as an engineer for his clients in other studios after the old

eight-track room was shut down. This is primarily because Tres Virgos was privately funded and, not being able to rely on the bank for loans, the owners had to maintain a cash-flow situation in order to finance construction of their studio. The experience, he admits, was a good one for Yeager. He was able to work in a number of studios around the Bay Area to see what other people were doing as far as acoustics went, and how their rooms translated once the music got on the outside. Up until this time, everyone except Yeager had been supporting themselves with conventional jobs. When the building project finally solidified and got started, everyone quit their jobs and devoted full time and energy into the studio itself.

Although members of the work crew had extensive construction experience, none of them were professional studio construction workers. For this reason, the main motto was "attention to detail." The crew was striving to achieve maximum ⅛-inch tolerance on any angles and all walls at any juncture. A lot of time was spent in proper framing to ensure that every piece was cut properly. Studs were rejected if they did not fit in perfectly with a gentle hammer tap. There were virtually no nails used in the constructions of this studio; all of the framing is glued and screwed. Each stud required 12 screws, with another 12 for the fire blocks at four and eight feet.

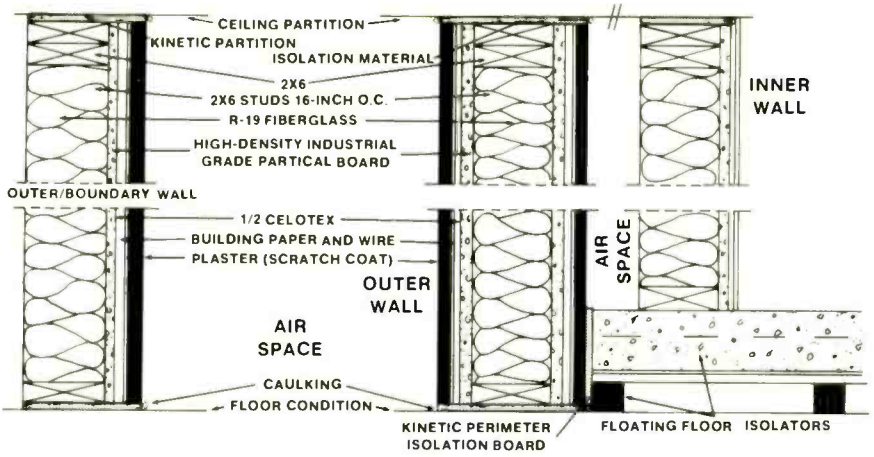


Figure 1: Control room partition detail.

Some of the systems used during construction were developed on the spot. The work crew developed its own systems for isolation with the walls at the ceiling juncture of the building, and used high-tech kinetic isolation materials for the wall systems, and for under the floating floors. A special motor mount system, using several pieces of rubber for isolation, was developed to separate the inner ceiling from the ceiling of the building itself. The same type of construction was used to isolate the walls, so that all of the walls and other ceiling members reacted independently of each other. As each wall is independent, it is possible to beat on one wall and not have any significant vibrations transmitted to the adjoining walls. Both the drum booth and the control room are set on independent floating floor systems to prevent unwanted sound transmission.

Sound isolation is also designed on the principle that the studio walls are built for mass and rigidity. With the wall systems, even people who are in adjoining spaces within the warehouse in which the studio is built have absolutely no idea the studio existed, since absolutely *no* sound transmission is taking place through or via the facility's outer walls. The rooms are designed on a walls-within-walls system. All wall coverings were cut to close tolerances. The sheeting was screwed and glued to the studs, and caulked between each layer and seam. Rigidity was required of the walls, to make sure that low frequencies were not to be transmitted by diaphragmatic action or escape through the walls into other rooms. The walls were made of different combinations of cement, Celotex, plywood, sheetrock, and high-density industrial-grade particle board with USG acoustical sealant between each layer (see Figure 1 for details of exact wall construction).

No conduit made direct mechanical contact with framing or through wall runs, even though the entire system was airtight. All pipes were isolated mechanically by using flex conduit wherever a run went from a wall to a ceiling, or to an AC box. In the control room, audio conduit was isolated from the floor by building floor supports around the completed conduit array.

All of the wall angles and planes in the control-room (which measures 22 feet by 22 feet) were designed explicitly in relation to measurements regarding what happens to sound with time; how the human ear perceives sound; and some of the psycho-acoustic phenomena that contribute to the "listening process." This is all tied in by the TEF (Time, Energy, and Frequency) mea-

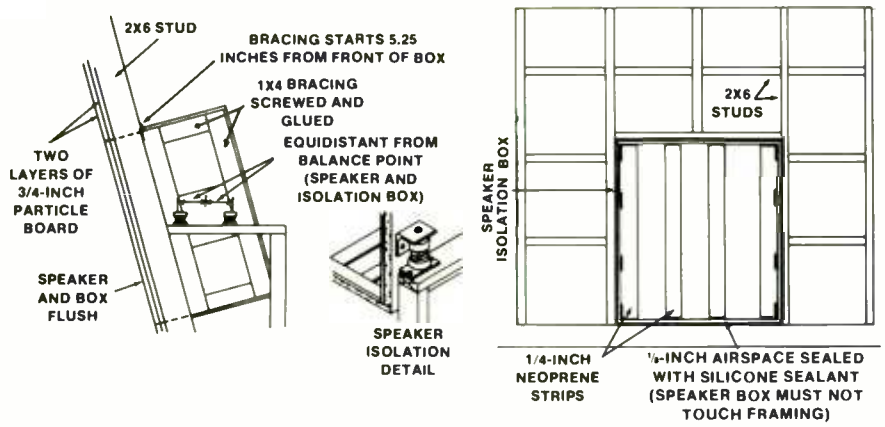


Figure 2: Speaker mounting and bracing detail — not to scale. ©1980 Chips Davis LEDE DESIGNS.

surements, so that the actual construction plans for the final walls of the control room were extremely detailed. For example, the back wall construction is definitely on not just one plane. Disbursive panels were shifted so that they actually form a Mobious Curve, as opposed to a flat surface.

Precision and detail was achieved by the use of cabinet-maker skills and tools, even in the rough framing stages, to ensure that all angles and walls would be as exactly as specified. The entire project was approached from a finished carpentry standpoint. With attention to detail such as it was on the rough work, the finished work was much easier. Angle finders, miters, and specialist tools were all used for the final construction. The finished walls were basically a jigsaw puzzle fit. The use of screws, as opposed to nails, allowed the workers to remove various angles if they were not exact, and prevented wastage of materials. The final trim was routed and mitered to fit by the work crew on the spot, since what they were looking for was not commercially available.

The control room speakers are placed in specially designed sockets that are mechanically isolated from the rest of the framing and construction (Figure 2). Because sound travels faster in denser mediums, it was discovered that low-frequency sound could arrive at the mixing position through mechanical transmission *faster* than the sound coming directly from the speaker. This phenomenon is referred to as Early, Early, Sound (EES), and can cause low-frequency time smear that interferes with mixing perspectives.

The studio itself, which occupies around 900 square feet, incorporates two isolation booths with sliding glass doors at the back of the studio facing the control room, and located on either side of the drum booth. The drum booth and isolation rooms are designed with high ceilings to prevent a restricted "small room" sound particularly associated with a "conventional" isolation booth. The studio also utilizes carpeting in one section and parkay floor in another, to change the ambience and provide the engineers with more variety in developing specific room sounds.

Electronics Installation

As mentioned previously, Ed Bannon was brought in to oversee aspects of the construction, and all of the electronics of the studio to make sure they were first class. He designed and supervised all electronic modes, a system that is stressed as being totally phase and polarity coherent. Very strict attention was paid throughout the installation to the detail of absolute polarity from the microphone transducer all the way through to the speaker and tape machine.

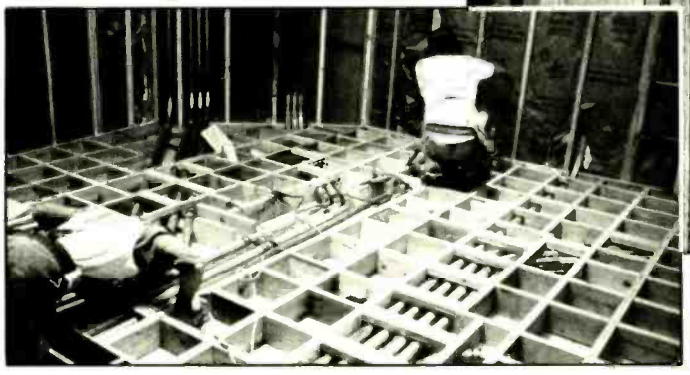
The control-room console is an MCI JH-528-B, which previously had resided in Crimson Sound and Cherokee Studios. Bannon chose the MCI console because he believes it to be the most powerful console ever built. The board features heavy-duty power supply rails and, at worst case, will deliver +29 dBm into 600 ohms at *any* stage, which represents, he offers, a considerable headroom figure. Another factor was the practicality of construction that made the console very "user" servicable. Modifications to the MCI console include the replacement of transformers on the line-level inputs with an OP amp, which is balanced to put out a very low impedance, thereby assuring proper matching for all modes of console function, and maintaining a truer accuracy of the VU meters in all modes. The existing VCAs were also replaced with the Aphex VCA, which Bannon feels is a much more quiet unit.

All master and individual console functions were matched mechanically, so that they coincided with the console number designations beside the faders

and pots. The left and right mix/ balance ratios stay the same, and all faders are linear in function; when the master fader is pulled down, the mix does not change; zero on the console is *truly* unity gain; fader accuracy is no longer plus or minus 1 dB, but plus or minus 0.1 dB. This proved to be a time consuming job of sitting down with a digital multimeter and a lot of 10 kohm resistors, and matching them within 10 ohms as opposed to 100 ohms. It is a lot of drugery work, but the result is truly consistent.

The Ampex ATR-102 stereo mastering machine was modified by removing the input and output transformers, to eliminate any phase distortion. The monitors at Tres Virgos are UREI 813-A Time Aligned speakers, each powered by 1,300 watts. The power amps are Crown M-600s with special Delta/Omega cards, which remove capacitance from the speaker lines, and make the output of the amplifier "see" a dead short as if it were welded to the terminals of the speakers. [The Delta/Omega cards are a special experimental unit made by Crown, but will be available commercially within the near future —*Ed.*]

Left: Wall construction between soundlock and control room. One-inch concrete, tar paper, Celofex, particle board, 2X6 fiberglass filled, particle board, Celofex, tar paper, concrete, air space, 2X4 edgewise, air space, ¾-inch exterior plywood, particle board, and 2X6 fiberglass filled.
Below: Control room floor internal structure with electical conduits running in wooden boxes.



The Time Aligned UREI monitors were chosen as part of the phase-consistency philosophy. Sound coherent phase is stressed as being as important a point of the control-room system as its LEDE acoustics. Phase is frequency responsive, while polarity is amplitude responsive. Phase distortion is often called "nervous distortion," and low levels of phase distortion are tolerated much less by the brain than even large amounts of harmonic distortion. It should also be noted that there are *no* graphic equalizers used to tune the control room. Instead, the room was tuned acoustically and, as a result, no EQ phase shift or distortion is present.

There is no earth ground in Tres Virgos' electrical system; rather, the system is set up on a basis of a zero potential for all equipment and outlets. In other words, everything is referenced to the same point, which is the AC circuit box. All electrical receptacles are isolated ground receptacles. The receptacle system is a 3-wire system where the conduit is used strictly as a Faraday Shield. Each receptacle, even if it is on the same circuit, has its own three wires running all the way back to the distribution box, where they are bussed together so that they all "see" the same point at the same time. Bannon considers that it is important never to daisy chain, because two different devices will be hooked to the same system at *different* points, thereby causing a different potential.

Bannon assembles the electrical system by first powering up the console and speaker amps. The system is then assembled one piece at a time, virtually with the console on. This is done so that if there are ground loop problems with the multipath hook-ups, tie lines, outboard equipment, etc., it is known immediately which piece of gear is causing the problem, and can be fixed instantly. Which saves a great deal of time in troubleshooting once the entire system is assembled. Bannon feels that the majority of time, grounding problems come from DC as opposed to AC paths.

A major problem exists in studios because of a lack of standardization by outboard equipment manufacturers. A good example of this is the recent decision by microphone manufacturers to standardize all mikes so that Pin 2 should be "hot," while most recording consoles are built with Pin *3* hot. At Tres Virgos, every microphone and piece of outboard gear was checked for polarity to ensure that there will be a constant maintained throughout the system, and that no signals are inverted. This was accomplished with a simple impulse generator, and turned out not to be a very time-consuming project. It's one more example, however, of the way in which Tres Virgos was not just a simple studio project, but one that adopted a complete systems approach.

When the control room was complete, and the first TEF plot was shot, the owners called Chips Davis in Las Vegas. Davis, who had never seen the room, pulled out his prediction plot, and it was found to match almost exactly with what was shot in the room at the time. This similarity gave the studio's owners a great deal of confidence in Davis' and Bannon's abilities, and the success of the facility itself. The control room's uniqueness was that the acoustics actually were predictable before the fact, which is a rarely-seen phenomena in studios. Because of the room's listening clarity, other manufacturers beside Crown are now jumping on the Tres Virgos bandwagon as a testing studio for their products. Aphex recently sent its newest Aural Exciter to Tres Virgos for a test run, and Sequential Circuits is now using the studio as a display room for its latest synthesizers.

Above: The control room/studio wall shown from the drum booth, and showing monitors in framing stage.

The studio was brought in pretty much within the budget and, basically, was on the time schedule except for a six-month shutdown when they were looking for more money to finance the construction.

Studio Business

Establishing clientele for a new studio is tough no matter what kind of design parameters are involved. Trying to influence producers and engineers from studios where they have already developed a rapport and are comforta-

ble is difficult. Couple that with the new acoustical concept, and you could have a real problem. The one major problem facing Tres Virgos, at this time, is that of bucking the "LEDE retrofit" image. Many people are skeptical of LEDE because of the number of studios that have been retrofitted with the LEDE concept, and are not performing as planned. Different retrofitted rooms have been facing different problems, but this is not the place for such an analysis. The fact is that most people have *not* heard an LEDE room built from the ground up, such as Tres Virgos. As a result, the studio must not only establish itself as a new studio, but also must establish itself against the misgivings and inaccurate preconceptions caused by some other unsuccessful projects.

This studio though, is not relying simply on the new acoustical design to establish its place in the recording field, but instead is promoting a well thought out, fully-integrated system; a system that is phase coherent, polarity coherent, and acoustically transparent. Such a total design concept is the feature that, its owners consider, makes Tres Virgos a powerful tool for the recording engineer and producer. The ears make the final decision, and one only has to come and listen in this studio to find out.

* * *

As Chips Davis summed up of his Tres Virgos design: "I think what it does is to give you a true picture of what is on the tape. You can take a true picture into different kinds of light over here, or you can look at it in this light over there, and the picture is still the same. The picture has a little different flavor because of the light in front of the picture, but no matter what light you take it into, it is going to look similar, and have the same relationship. If you take the picture that is totally distorted and put it to any kind of light, then the picture still has that distortion." ■■■

COMMENTS FROM TRES VIRGOS USERS ON THE LEDE DESIGN

"The first impression I got was the very pleasant feeling of free space, both physically and acoustically . . . one feels well at ease. The control room can easily accommodate 7 or 8 persons very comfortably.

"There is not the slightest amount of resonance at any frequency; direct field and diffused reverberant field are perfectly balanced. The rear half of the room acoustically disappears, and the front half is very neutral. The room sounds good . . . acoustic instruments and classical music are very accurately reproduced.

"The most interesting feature is surely that this system can reproduce classical music with high accuracy and refinement, while heavy rock and roll and electronic synthesizer will be reproduced with the needed 'punch' and energy."

— Andre Bourget
Manager, Systems Engineering
Studer International

"The studio is World Class. The sounds in the studio were true to life, and everything we heard in the room sounded exactly that way when we got out. Just having one studio allows you to get total focus while you're there."

— Marty Bleckman
Producer, Megatone Records
Project: Mind Warp,
Patrick Cowley

"The separation was really strong, and the monitors were very clear. The room was real smooth, and the mix I went out with was very close to what I had in the room."

— David Kahn
Independent Producer

"What you perceive in sound in the control room went down on the tape. You can sit in the room for 15 hours at very high SPL and not get ear fatigue. Even by setting our drums up in the center of the large studio, we could get tightness and warmth on our tracks."

— Phillip Miller
Producer, To Be Continued Music
■■■

SOUND SYSTEM DESIGN FOR THE EUGENE PERFORMING ARTS CENTER, OREGON

Luke Bandle

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As one approaches Eugene, Oregon, from the air, a feeling of desolation is readily apparent. Nestled within thousands of square miles of timberland sits a striking complex that, on first appearances, seems futuristically incongruous to its surroundings.

Amidst the junction of two highways in Eugene's business district is situated a $52.5 million performing arts and convention center complex. A new Hilton hotel and conference center combine esthetically with the Performing Arts Center, to offer the performer and businessman alike an all-encompassing entertainment and commercial complex.

In 1963, a need was seen by the city fathers to draw non-lumber related industry to the Eugene area, and to create a tourist-based economy. A cultural center was envisioned, offering an urbanized public a "touch of civilization" in an area offering all the benefits of rural life.

The populus of Eugene, numbering slightly over 100,000, managed an amazing feat. A city bond issue financed the $18.5 million necessary to complete the structure of Performing Arts Center. Without any outside state or federal help, the citizens passed the issue yet fell some $4 million short of the final tab.

With some last minute donations from private individuals, (which explains the 11th-hour name change to "The Hult Center for the Performing Arts") the $22.5 million venue opened on schedule, in late September of this year.

Exterior Appearance

Architect Norman Pfeiffer, of Hardy, Holzman, Pfeiffer Associates in New York, was hired to design the center. Walls of concrete, 20 inches thick, keep the road noise from the venue's interior. According to Dave Pelletier, Operations Director for the Center, "11,000 cubic yards of concrete were utilized in the exterior construction. If the same quantity of concrete were laid for a two-lane highway, it would be five inches thick, and extend over 100 miles."

A metal roof, 85 feet high, is supported by wooden columns. Glass panels in the front of the center arranged in an "A" -frame motif reveal the multicolored interior of the front lobby, with its rising spiral staircases. To this writer at least, the Eugene Performing Arts Center is probably the most esthetically pleasing venue built to date.

Interior Layout

Three halls encompass the Eugene Center. The largest area, the Silva Concert Hall, was named after the deceased parents of Carolyn Silva Chambers, who donated $1.5 million to the Center in their memory. The Silva Hall, with its horseshoe shape, 80-foot ceilings, and sweeping balconies, seats in excess of 2,500 people. One immediately notices its massive basket-weave ceilings. The panels (123 in all) are made of the finest

. . . continued on page 89

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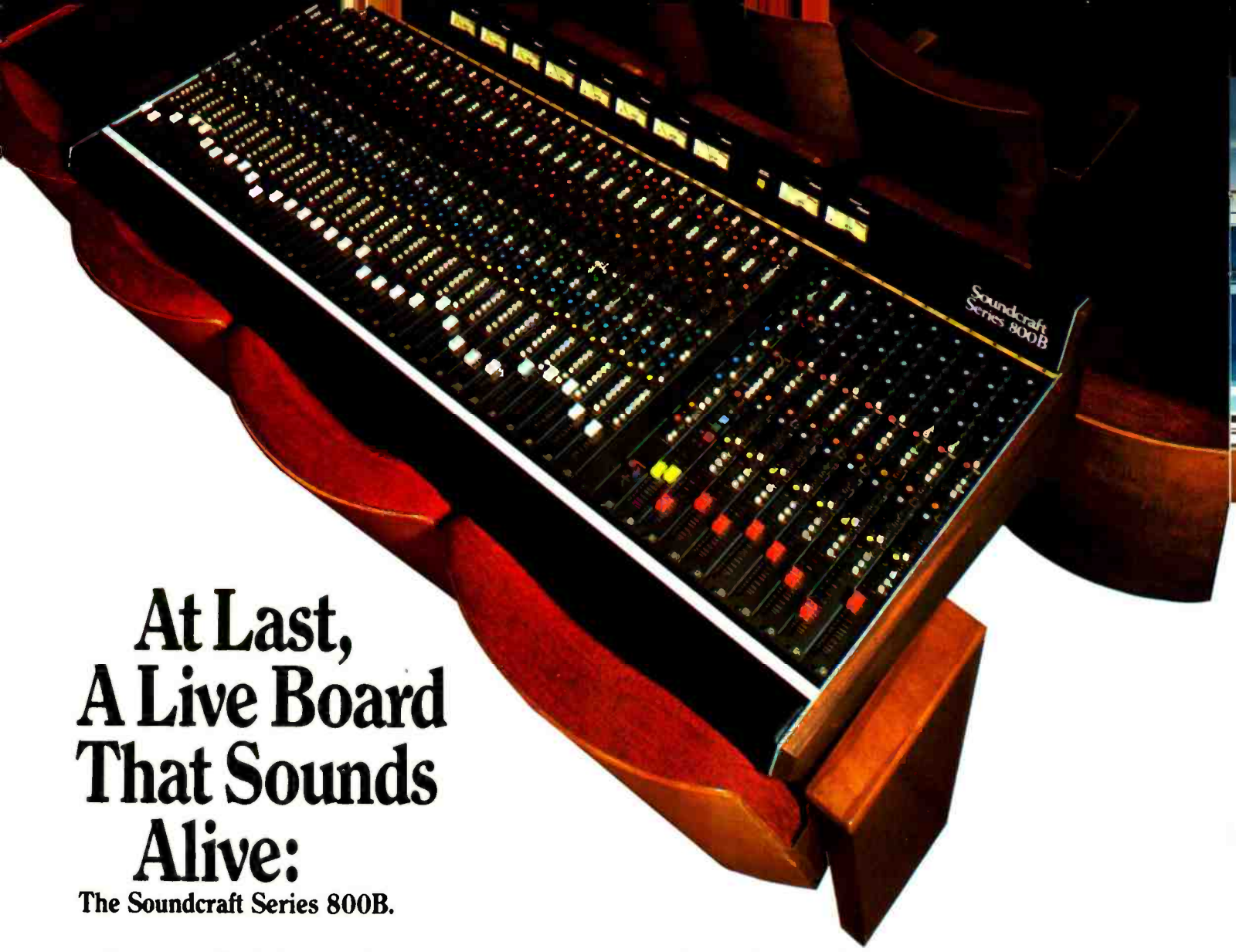

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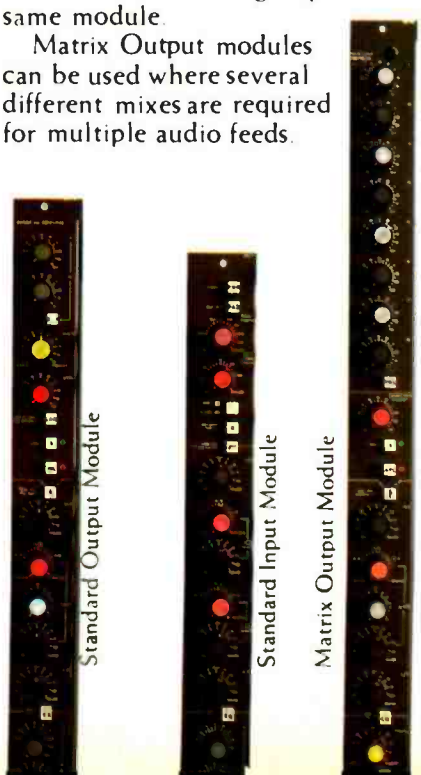

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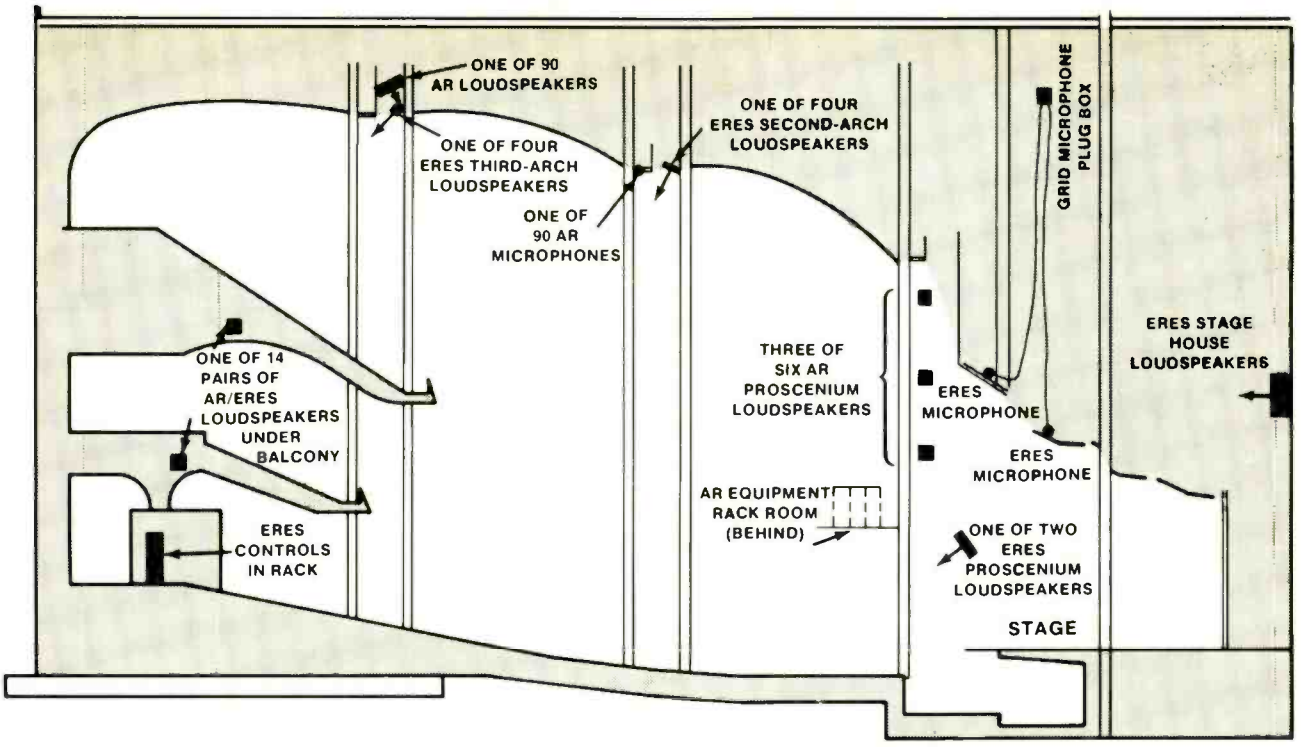


Figure 1: Equipment locations of ERES/AR systems in Silva Hall.

— *continued from page 86 . . .*

Eugene Performing Arts Center

particle plaster available. Each panel weighs over 3,000 pounds each, and were cast in place. To achieve such a process, scaffolding was built to provide access to each of the forms suspended from the ceiling. The panels form an integral part of the acoustical qualities of the hall's interior. The 35-foot high and 59-foot wide Proscenium Arch stage is capable of virtually handling any elaborate production. Because particular care and attention were paid to audience sight-line and acoustic requirements, there is hardly a seat within the Hall that has a bad view of the main stage.

The Soreng Theatre is a 500-seat hall designed primarily for small and varied productions, and is more traditional in physical design than the larger Silva Hall. A sloping orchestra section and tiered loge levels (or front boxes on the lowest balcony level), lend this venue to intimate productions, as well as commercially oriented presentations.

A large rehearsal hall, presently being used by the Eugene Symphony, completes the Center's facility. This hall is adjacent to offices provided for the technical staff that maintain the entire facility.

Acoustic Design

As previously mentioned, the primary decision to build the Center rested on the premise that it would draw tourism and renewed economic interest to the Eugene area.

In order to encompass a varied scope of performance and live-music venues, a multitude of technical difficulties arose. How could one hall be utilized on a given evening for a presentation of Bach, and the following evening La Traviata, with Talking Heads performing on Friday? A great deal of scepticism was felt among the "Symphony Purists" in Eugene, who perceived the Hall as being purely symphony-oriented, and that architecture technical "wizardry" would deter from the "over 2-second" reverberation time almost always required by such purists ears. A conflict of interest between the "Audio Special Interest Groups" was readily apparent at the outset. It was obvious that acoustically the Hall would not be suitable for as varied a format as originally had been perceived; consequently, an electronic solution became a necessity.

Christopher Jafee, a leading acoustician in the field, was called in as a consultant on the project. The "key phrase," according to Chuck McGregor of Jafee Acoustics, "is *multipurpose* venue." In order to accommodate a range of programs, ranging from symphony performances to speech, a variable-acoustic

Interior of Silva Hall during construction stages.

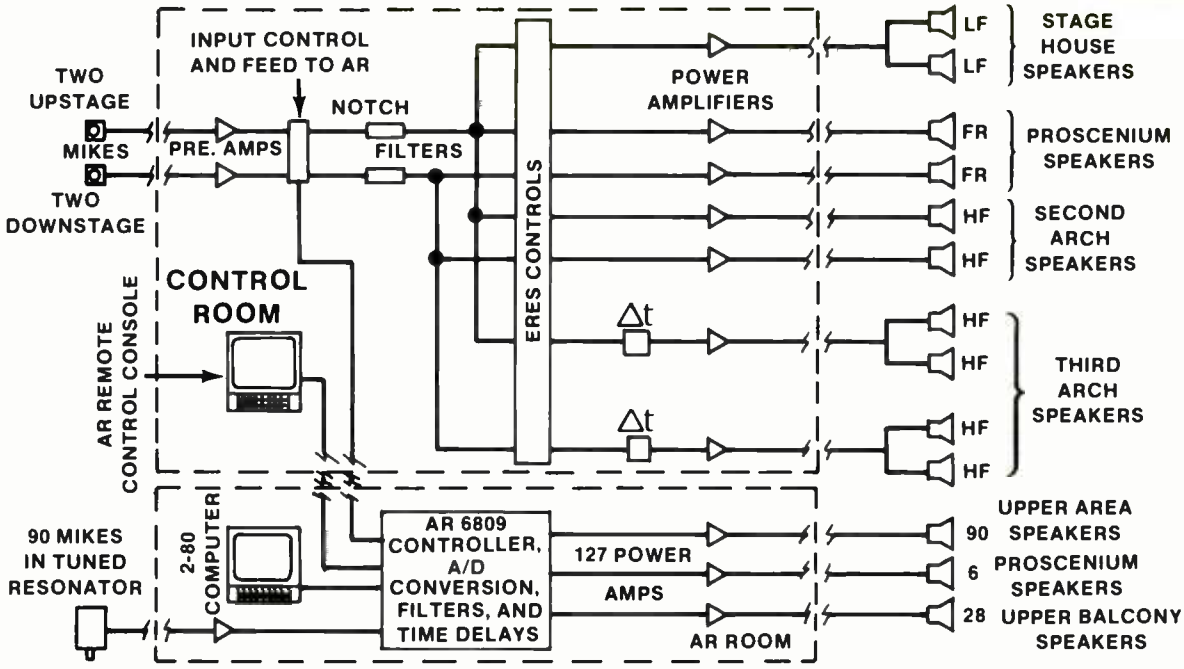


Figure 2: Block diagram of Electronic Reflected Energy System, and Assisted Resonance (AR) System.

environment was deemed necessary. In lieu of "massive moving ceilings, and tons of draperies," it was considered more cost effective to accomplish this task electronically, rather than physically, with a resultant greater degree of control.

Two electronic "systems" of acoustic treatment were employed: what was termed an "Assisted Resonance System"; and an "Electronic Reflected Energy System," to adjust and enhance the center's acoustic environment. (Equipment and electronics layout for the two systems are shown in Figures 1 and 2.)

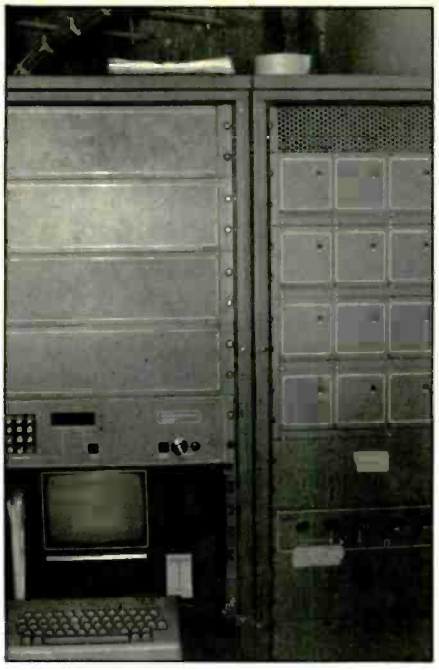
The Assisted Resonance System, in its most basic sense, electronically varies the apparent volume of the Silva Hall, by controlling via a special computer the variable 1.5 to 2.5 second room reverberation times. A network of 90 individually enclosed and tuned microphones, located in the ceiling catwalks, respond to narrow bands of low- and mid-frequencies (40 Hz to 1.8 kHz). The output from each microphone is digitally processed and controlled with predetermined reverberation enhancement, then connected to one of 90 loudspeakers. The end result is room reverberation time that varys as a function of frequency.

Presently, 10 pre-set computer programs are available, inputed by Jafee. However, a multitude of variations in acoustic formats are available to the operating engineers, changes for which are merely a function of software programming.

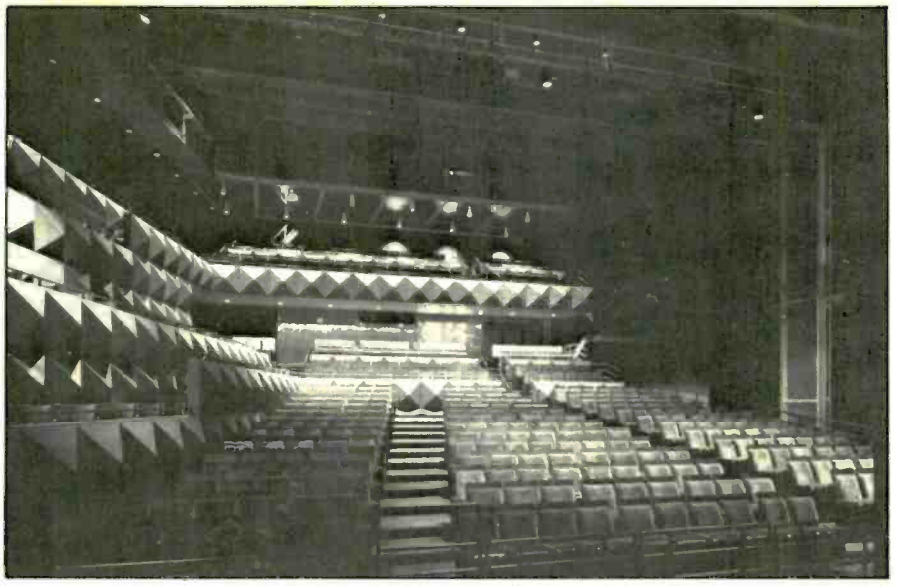
The computer-based system was designed by a London-based company, AIRO, and has been utilized in several other installations, including The Royal Festival Hall, London, England, and The Concord Pavillion, San Francisco. All elements of the systems are proprietary designs of AIRO, with the exception of the loudspeaker arrays.

The second installed system — the Electronic Reflected Energy System — basically affords the architect the luxury of almost uninhibited design flexibilities, by simulating reflected soundwaves from internal architectural surfaces. Functioning as a time delay system, the ERES enhances the first sound reflections, and operates in conjunction with the Assisted Resonance System. A few permanently-mounted microphones are placed in the hall's "far field," so that signal pick up is equivalent from all locations on stage.

The microphone signals are ampli-



Computer controller and rack for Assisted Resonance (AR) System in Silva Hall.



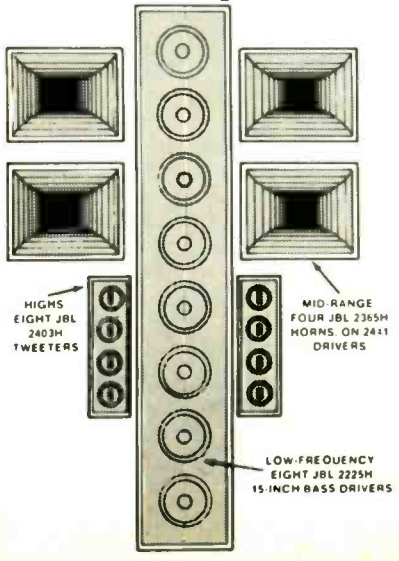
Interior of 500-seat Soreng Theatre, designed for small productions.

fied, equalized, delayed, and then fed to dedicated loudspeakers that represent sound waves eminating from many architectural surfaces with differing coefficients of absorption, and physical orientations.

From an acoustic standpoint, the ERE System enables the engineering staff to electronically "raise" and "lower" the venue's ceiling; move walls in or out; increase room volume; and float non-exsisting clouds and panel space.

The combination of the two systems is said by the Center's staff to provide the operational flexibility necessary for the various performance formats with a naturally-sounding environment, as opposed to other venues often plagued by amplified treatments.

Figure 3: Central Cluster mounted above the stage center of the Silva Hall for balcony coverage



House Sound System

Silva Hall's main house system consists of a cluster array mounted above center stage, and a left and right front-fill arrangement mounted on stage.

The 64-cubic foot bass reflex central cluster cabinet contains eight JBL 2225H, 15-inch low-frequency drivers arranged in a column design. The mid-range crossed at 800 Hz is comprised of four JBL 2365H constant directivity horns, mounted on 2441 drivers. Eight JBL 2403 tweeters are brought in at 7 kHz to cover the high end (Figure 3).

Utilizing a matrix output from a

Eugene Performing Arts Center

Soundcraft Series 800B front-of-house console, the main signal feeds an Orban 672 A parametric equalizer. A Spectra Sonics Model 505 three-way crossover then splits the signal at 800 Hz and 7 kHz. A total of 24 Spectra Sonics Model 701 power amplifiers deliver 2,200 watts RMS into the three-way systems (Figure 4).

Cabinet design for the two front-fill arrays is rather unique, considering the choice of components and varied formats served by the venue. The low-frequency portion of each cabinet contains six JBL 2225H 15-inch, low-frequency drivers, crossing at 800 Hz into a JBL 2360H mid-range horn with 2441 driver, and a 2305 mid horn with lens on a 2441 driver. Six 2403H tweeters comprise the high-end drivers, which are brought in at 7 kHz (Figure 5).

This tri-amplified front-fill system is connected to the left and right remix outputs of the Soundcraft Series 800B console, through an Orban 674 A stereo parametric equalizer, and into a UREI 525 crossover. A UREI power amplifier drives the bass components, and a 6250 the mids and high-end. (Figure 6)

Monitor Cabinets

Four, three-way bi-amped cabinets housing two JBL 2225H low-frequency drivers, a 2397 Smith horn with 2441 driver, and 2403H tweeter are utilized for side-fill monitoring (Figure 7). The remix output from a Soundcraft Series 400 console feeds an Orban 674A parametric equalizer, which drives a UREI 6300 power amp for low-, and 6250 for the HF components (Figure 8).

Eight low profile (12-inch high) floor monitor cabinets are available for on-stage monitoring. Each cabinet comprises a JBL 2225H low-frequency driver, and 2344 horn on a 2441 driver.

Live-Sound Consoles

A total of three consoles are utilized within the Center, a Soundcraft Series 800B having been chosen as the main house board for the large Silva Hall. The 800B has 32 input channels, eight subgroup outputs, eight matrix outputs, and a stereo mix bus. Each input channel features a four-band, semiparametric equalizer, eight auxiliary sends, group assigns, stereo panning, and mix assignments. The subgroup output module features a flexible matrix routing function, allowing connection to any of the matrix outputs for multiple feeds around the Hall, and for external broadcast when required. A parametric equalizer can be switched into the matrix at will. The various matrix outputs feed the central cluster array, while the stereo outputs supply the front-fill cabinets.

The house console is located in the center of the Hall, approximately three-quarters back from the stage, and could be considered somewhat incongruous to the overall installation. It obviously was located with the sound engineer in mind. When asked about the console placement, Pelletier offered that "If

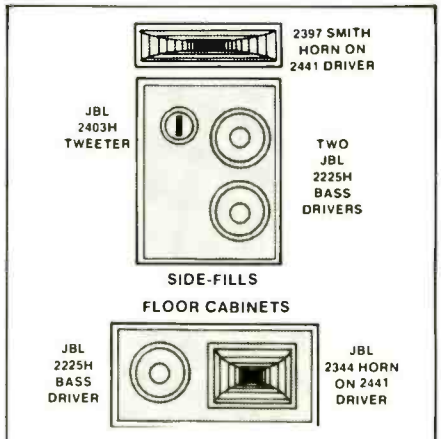
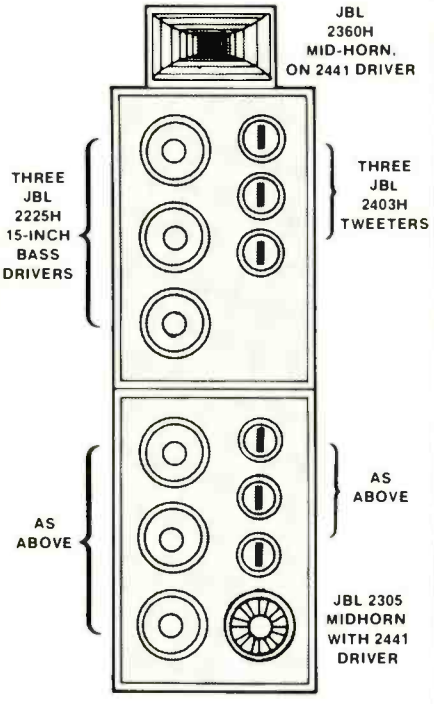


Figure 7: Side-Fill and Floor Monitor Cabinets.

Figure 5: left and right Front-Fill Cabinets, one per side in the Silva Hall. (Designed by Dave Pelletier and JBL.)



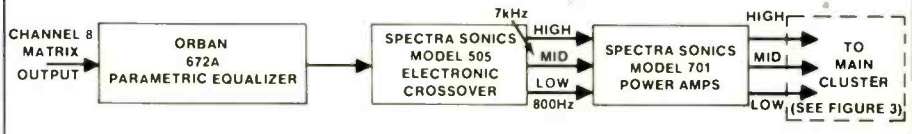


Figure 4: Electronics Layout for Central Cluster in Silva Hall.

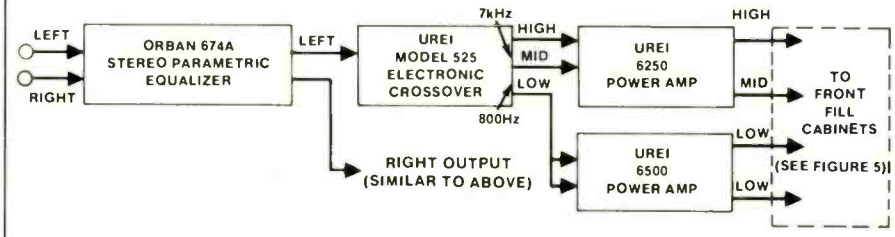


Figure 6: Electronics Layout for Front-Fill Cabinets in Silva Hall

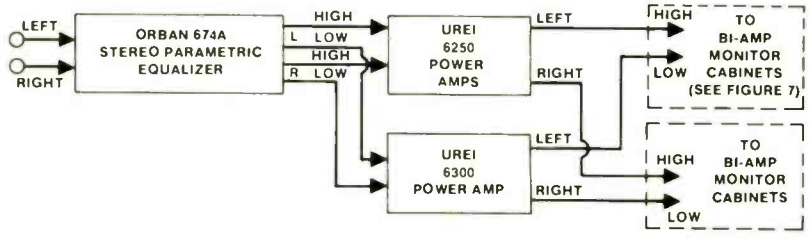


Figure 8: Electronics Layout for Monitor/Side-Fill. Two racks are used for a 4-way monitor mix from the Soundcraft 400 console.

Soundcraft Series 800B 32-input front of house console for Silva Hall.

Soundcraft Model 400 portable stage monitor boards used in both the large Silva and smaller Soreng Halls.

people are paying $20.00 per seat for a powered performance, the console better have a $25.00 seat." Considering the space efficiency offered by the Soundcraft console, only approximately $200.00 in seating was sacrificed.

Doubling as a front-of-house console in the Soreng Theater, and stage monitor console for the main hall, is a soundcraft 26-input Series 400 console. When a performance in the Silva Hall requires extensive stage monitoring, the console is moved from one arena to the other, and placed on stage. The Soundcraft Series 400 features four-band EQ, detented precision potentiometers, and routing to four subgroups. Being as physically space efficent as the Center's Series 800B front-of-house console, the Series 400 combines flexibility and mobility, according to the engineering staff.

When the 400 is being utilized for stage monitoring in the Silva Hall, the Soreng Hall makes use of a Soundcraft 16-input 1S console for its main system control. "We haven't compromised on quality anywhere in the installation," Peltelier says of the Center's selection of Soundcraft consoles for mixing duties. "We went with the best." Steve Hayngebrauk, the Center's house sound mixer, was particularly impressed with operational flexibility, especially the matrix output section.

* * *

"With the Best," seems to be the key in describing the physical and philosophical achievements of the Eugene Center. On opening night some 3,000 dignitaries, VIPs and art columnists from across the country were present, and were heard describing the event as "Eugene's finest hour," and "The beginning of a new era."

Among the more notable comments, music critic Robert Commandey considered that the Center offered "a trend of acoustics which have far reaching implications," while Harold C. Shonoberg, of the *New York Times* felt that "the sound was really good, with presence and definition; a solid bass and all the color one would desire." ■■■

Photography by Steve Barnett

Director Hal Ashby on Todd-AO mixing stage.

"LET'S SPEND THE NIGHT TOGETHER"

Hal Ashby and the Rolling Stones Movie

Synthesis of Traditional Film Production, Video Editing, and Multitrack Techniques

by Steve Barnett

Hal Ashby has directed a number of dramatic feature films, including *The Last Detail*, *Shampoo*, *Bound For Glory*, and *Being There*. He is also a fan of rock music, however, using it to great effect on the soundtrack of the Academy Award-winning motion picture, *Coming Home*. Ashby also has been friends with Mick Jagger for some time through mutual admiration of one another's work, but in all his years in the film business, as an editor and as a director, he has never worked on a "rock and roll movie." Nonetheless, Ashby received a phone call when The Rolling Stones began thinking about a film of their record-breaking 1981 tour.

"I think [the band] had been talking amongst themselves," says Ashby, "and decided that it would be a good idea if they maybe got something done with a film. They were going to be doing something live for television anyway, but I didn't talk to Mick at first. It was Rupert Loewenstein, their business guy, and he said, 'Do you have any ideas about who can do a film?' And I said, 'Well . . . I can.' It was about the second weekend into the [1981] tour."

At this stage of the game, the Rolling Sones' tour was moving across the country like a juggernaut, and the movie makers were forced to play catch up. Ashby looked for assistance from his friend, filmmaker Pablo Ferro. Ferro and Ashby met when the latter was editing *The Russians Are Coming, The Russians Are Coming*. The two began to discuss the style of The Rolling Stones picture.

"I had some pretty definite ideas about what I wanted to do," Ashby recalls. "Basically just to shoot it in a little bit more 'classical' form, if that's the right word. It wouldn't be hand-held, as most rock things are. I wanted to get it locked down a little bit more, and stay in shots longer."

"Everything had to be done non-documentary," adds Ferro. "Everything had to be done slowly. You stay on the shot, and if you move, you move with the performers. If you chose to stay, you stay, so that the performers would bring you in and out of the shot. No fast moves — not all of a sudden going into someone's eye, or something like that. It had to be shot just like a *movie*. So, when the tour came over the the LA Coliseum, Hal and I went over to see it."

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THE ROLLING STONES MOVIE

"What I was looking for was to see how I could make[my ideas] work best," explains Ashby. "I felt very strong about them right off, and everything started to mold itself that way. One of the first things I saw was that all the instruments [and microphones] were wireless, so I knew there was going to be a lot of good movement."

As the Stones moved on to the next stop in their tour, the filmmakers began to assemble their crew. Ashby called cinematographer Caleb Deschanel, and a second director of photography, Gerald Feil.

Filming in Concert Venue

The tour rolled on, and the Meadowlands Arena, an indoor venue in New Jersey, was selected as site of the first two concerts to be filmed. These dates coincided with the arrival on the tour of one of The Record Plant's mobile recording units. Onboard was the Stones' music mixer, Bob Clearmountain, who mixed the group's latest album, *Tattoo You*. Clearmountain, along with David Hewitt of The Record Plant, would handle the live 24-track recording for both the film soundtrack, and the subsequent live album, *Still Life*.

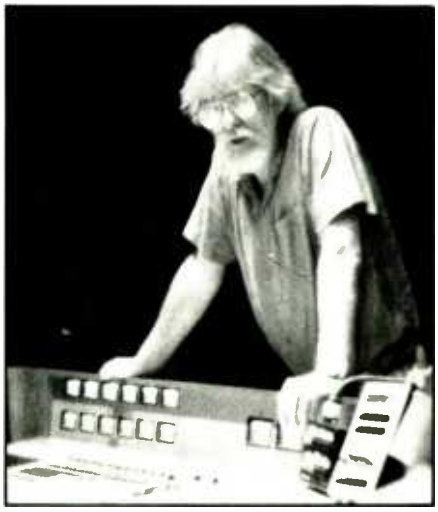
The filmmakers had hoped from the start to capture the spectacle of an outdoor, coliseum-sized concert, but the time taken to assemble the company caused them to miss the last of these dates scheduled for the tour. The Stones, according to Ferro, agreed to add another such venue to their list of stops. The added concert still would be a normal money-making production, and the filmmakers could be accommodated. Arizona State University's Sun Devil Stadium in Tempe, Arizona, was booked for the engagement by tour promoter Bill Graham.

From the film's inception, Ashby also involved Harry Howard, who acts as managing engineer of the director's own Northstar Media, Inc., a Hollywood video and sound transfer house that has been developing video editing techniques for Ashby over several years. Howard would come to serve as a central coordinator for most of the technical matters involving both production and post-production of the picture.

Production sound mixer Jeff Wexler also was signed on to the shoot. Wexler had worked on five concert features, including *Rock Show*, Paul McCartney's film of his "Wings Over America" tour. Acting more as a recording producer than as an actual mixer on this production, Wexler set up lines of communication, helped devise the slating system, and supervised all the recording for the motion picture. Master 24-track recording duties would be handled by the Record Plant mobile truck.

In normal film production, the director and his cinematographer work out how the action is to be filmed with the performers, rehearse the scene, and then shoot it. In this instance, however, 20,000 to 90,000 Rolling Stones fans were not going to sit by patiently waiting for the next camera setup. The operators had to continuously film action with which they were only marginally familiar, and which would not, and could not, be repeated.

To allow Ashby and Ferro input into the camera work during the actual concert filming video splitters were placed on the six main film cameras, and these through-the-lens pictures fed video monitors in Louis Mahler's Location Video truck. There, Ashby and Ferro coordinated the movements of their



HAL ASHBY
— Director —

camera operator in a manner not unlike directors of a televised sporting event. "Except that we weren't doing any cutting," Ashby adds. "When we would see different things, we would talk the camera operators through and help them along with the style that we wanted, so that they wouldn't be whipping all over the place."

The presence of the video truck, with communications to and from the principle cameras, the video feeds themselves, and the audio lines to and from The Record Plant truck presented some large logistical problems for all concerned. "The Record Plant," explains production sound mixer Jeff Wexler, "is obviously real good at this; they just wheel in their truck and work all day setting up. It's easier for them to do this sort of thing. We had to show up with a lot of miscellaneous equipment that had not really worked together before, and just tackle the whole problem from scratch. The hardest job was the huge outdoor shows, and making those long cable runs.

"At one point [there was] the thought of doing the entire film on video. Obviously it could have been handled much more easily, because your communications systems are built right into the camera. And slating is not a problem, because the tracks are being layed down in real-time on the same material as the video. If you wanted to add another camera, all you'd be doing is running one more cable. It would be simple, but it was really just that video has one look, and film another. Hal kept saying, 'You and I both know that the film is going to look better.' And, sure enough, it is a terrifically good-looking film."

Slating Picture to Audio

One of Wexler's major considerations was the development of a slating system for the production. As the image and audio were being recorded on unrelated pieces of equipment, a method had to be devised to mark the soundtrack and film from all the cameras with a common sync point. Later, in post-production, the film's editors would use this common address to synchronize the various picture and audio elements.

On previous concert films, synchronizing visuals and sound has proven a major problem, often holding up post-production for weeks. To help find a workable solution, Wexler brought in Billy Youdelman, since the latter is one of the few engineers in Los Angeles regularly involved with film, television, and music recording.

Wexler had worked with Youdelman on a number of projects, getting his assistance in devising a slating system on *No Nukes*, the motion picture of the MUSE concerts filmed in New York's Madison Square Garden. "With 35 mm film, you basically get 1,000 feet in a magazine; about 10 minutes," explains Youdelman. With a multiple camera show, "you have to have a common marker, like clapsticks, or any number of things. Well that's fine to start, but 10 minutes into the show, when everybody runs out of film, there is no opportunity to reslate. The cameras are all far away, the show is going on, and it's noisier than heck. And The Stones *ain't* gonna stop to let you jump up on stage and yell 'Common Marker,' and all of that!"

There is also the "bloop system." When the operator pushes a button on his camera, a tone of set frequency is placed on the audio tape at the same time that a light flashes in the camera, and is photographed on the film. This technique provides a common sync point on both the film and audio tape. The bloop also may be activated by a camera assistant switching on a flashlight that is shot by the camera as the tone is placed on the audio tape. And which means that a connecting cord is not needed from the camera to the tape machine. Both of these systems allow the operators in multiple camera shoots to start and stop at will, so long as the audio tape is continuously recording.

Both arrangements present different types of problems, however. "Somebody still has to identify which camera is get-

THE ROLLING STONES MOVIE

ting the bloop at that time," Wexler explains. "Some people have used varying frequency bloops — this camera is 400 Hz, this camera is 700 Hz, and so on — and that helped a bit, but it still required that someone actually speak into a microphone, and say that this is, in fact, camera #7 getting their bloop at this time. If you've ever been to a concert, you know that's not easy to do!"

"They used the flashlight [technique] on one picture," adds Youdelman, "and I talked to one girl who cut it. She was telling me of seeing camera assistants faces [on the film] as they're flashing the light on and off going, 'What's wrong with this thing?' And there are all these bleeps on the track. They also put all the private conversations on the *same* track as the bleeps. They called me up, and I chained three notch filters together. Finally they fished out enough of the pops, but you don't know what camera [the audio cues] go to!"

SMPTE Timecode Format

SMPTE timecode was decided upon as the method of synchronization for The Rolling Stones film, since in the past Youdelman had worked extensively with the medium, with much success. It was not, however, the normal 30 frames-per-second timecode commonly employed in television and audio recording. Instead, 24 FPS timecode was utilized on the Stones movie, just as it had been on the seldom-seen motion picture version of *Beatlemania*, and on Bette Midler's *Divine Madness*; Youdelman supervised sync on both pictures.

According to Northstar Media engineer Harry Howard, "24 FPS timecode was selected by Youdelman in part because of past experience, and in part because it would reduce the number of ambiguous frames when it was photographed by a movie camera, which also runs at 24 FPS."

"Thirty-frame timecode makes it real hard to sync to later," says Youdelman. On film, "the numbers are a blur when you're looking at them, or you get a half of one number, or it jumps all over the place, and gets real chaotic. It's useful just to get close, and then you have to trim-sync to a drum track, or something. With 24-frame SMPTE timecode, you still have to trim-sync, but it gets you a *lot* closer."

Needless to say, the Record Plant truck handled the multitrack recordings of the Stones' concert performances for both the film and the live album, *Still Life*. "We also had two stereo Nagras set up in The Record Plant truck," Wexler recalls, "so that we could have overlaps [to avoid gaps in recording for reloading]. On one track we put a mono mix from The Record Plant's console, and on the other the 24 FPS timecode. We were also feeding the same 24 FPS timecode to one of the channels on the 24-track recorder."

"In addition to the timecode on the two-inch and the Nagra tapes," says Howard, "there was also recorded on each tape a common 60 Hz pilot signal, which was in phase lock with the 24 FPS timecode."

While the 24 FPS SMPTE timecode represented a unique address on each tape, it was the 60 Hz pilot signal that provided the reference for later playback at synchronous speed. Additionally, all the cameras were equipped with crystal-controlled motors, which run at exactly 24 FPS. In this fashion, the film, the 24-track, and ¼-inch Nagra tapes all could be played back in sync, using timecode as a common address.

To print this address on the film, Youdelman and Wexler set up a system of video monitors. "From the same timecode generator that fed the tape machines," continues Wexler, "we fed timecode into a video character generator, and sent that to video monitors, the medium which would be photographed."

HARRY HOWARD
— of Northstar Media —

At each of the key camera positions there was a small television monitor, into which was fed a constant readout of the same 24 FPS SMPTE code being recorded on the 24-track, and ¼-inch Nagra tapes. After starting his camera, all the operator had to do was tilt down and shoot the monitor displaying the code. This would record on his film the SMPTE address being recorded at the exact same time on the audio machines.

Audio Transfers

The Nagra ¼-inch tapes then were transferred by Northstar to three-stripe 35 mm magnetic sound stock, which runs in conjunction with the 35 mm film on an editing machine. A mono audio track was placed on one track, and the 24 SMPTE code on another. Using a

SMPTE reader attached to an editing machine, the editor could then locate on the audio stock the same SMPTE address photographed on film, and being displayed on the editing machine's screen. By lining up this common address, the picture and audio track could be played back in perfect synchronization.

"It's the only method that seemed to work," Ferro says. "All the cameramen had to do was go back to the [SMPTE] clock, and that's what was great about the monitors Hal and I were watching. We'd tell them that it doesn't matter if you get it at the front of a roll of film or the back, but as soon as you're running out of film, run for the clock. Sometimes an operator would forget, and we'd be yelling, 'The clock! Get the clock!' "

For SMPTE display, "I use little black and white TV monitors," says Youdelman, "because they're cheap. I also use 30-frame video, so you see this vertical interval — about 21 lines or so — [which is] a vertical bar rolling through a number once in a while because of the different frame rate. So I make the numbers *real* big, and then when the bar rolls through it's very small in comparison. Even with the code running at 24 FPS and the video running at 30 FPS, it comes off fairly well because of the persistence of the [screen] phosphor. I also crank the contrast and the video gain way up so you get a nice hot light.

"One of the problems with using a

PABLO FERRO
— Co-Director —

[conventional] SMPTE reader is that, first, the display is too small and, second, they rent for about $100 per day. Whereas a little 9-inch black and white monitor rents for about $10 a day. Also, one valid frame is all they need to sync, so the camera operators would shoot a couple of feet of it; about a second and a half."

"We also had a big master clock," Wexler adds, "which we set up on the stage for the cameras that were not in a good position to shoot a video monitor — because of a long lens, being on the Steadicam, or something like that. But during set up time the promoter decided that he didn't like the way it looked on stage, even though in fact we did have the clock up there for one whole concert where nobody seemed to notice. But it was taken down."

The lack of an exact sync point for these cameras became less critical in the post-production process, when the editors discovered that they could simply match their images to the images on rolls of film already synchronized to the audio track. At the Meadowlands concert, for example, besides the six key cameras there were two that did not have timecode monitors or video assists feeding Ashby's and Ferro's truck. At the Sun Devil Stadium, there were six such cameras, including one in a helicopter circling the event. Their distance from the stage also made it next to impossible to tell if they were out of sync with the audio.

Syncing Dailies

With completion of filming of the three Stones concerts, the task of synchronizing the dailies began at Ashby's editing facility in Malibu, California. Ferro called in Lorinda Hollingshead to assist in the task, based on her previous experience with Youdelman's 24 FPS timecode on *Beatlemania*; in addition, Lisa Day was called in, and would eventually become editor.

Since there were two Nagras running

THE ROLLING STONES MOVIE

BELOW: Co-director Pablo Ferro in the video suite of Hal Ashby's Malibu editing facility.

ABOVE: Twelve-screen monitor bank and Beta I playback decks, used for simultaneous viewing of material from up to 12 cameras.

during the concert filming, with overlapping segments at the end of the first roll of tape, and at the beginning of the next, one of the first tasks was to prepare a single, continuous soundtrack on 35 mm magnetic stock, and divide that into units of roughly 10 minutes long for ease of handling. The mono scratch mix of the concerts recorded on the two ¼-inch Nagras was transferred to 35 mm, three-strip mag, the 24 FPS timecode going on one of the other tracks. A rough stereo mix would later replace this mono mix.

Soon, however, it was discovered that the SMPTE could not be read from mag stock while running any slower than three frames-per-second. Once the speed past the head fell below this, the SMPTE readout froze in position on the display, even though the mag stock might be creeping slowly ahead. The editing team became proficient at stopping the KEM editing machine as quickly as possible based on the timecode reader.

A way around this problem in the future, suggests Wexler, would be for a synthesized timecode to take the place of the actual recorded timecode on the display. When the speed of the mag stock falls below a readable rate, this synthesized code would begin at the number following the last address picked up by the reader, and maintain a sequential numerical display based on the motor drive of the editing machine. Conversely, when the machine begins running at the minimum speed, the actual recorded time code would resume on the display.

The KEM flatbed editor utilized on the project is capable of running as many as three different rolls of sound and three different rolls of picture, simultaneously and in sync with one another. Each of the two overlapping rolls of 35 mm mag were placed on the KEM's sound plates, and rolled to the approximate location of their overlaps. "What we'd do," Lorinda Hollingshead recalls, "is to use the SMPTE reader to get into the ballpark. I'd roll down and get the two numbers from each roll to match, and then listen to the music. We'd decide where we were dead on, matching drum beats or whatever, and cut off the extra film. Then you'd splice the two together, giving you a continuous soundtrack."

Since this now continuous track would be far too unweildly to handle in its entirety, it was divided into units or KEM Rolls of roughly 10 minutes in length. "We'd always cut off at the end of a song," explains Hollingshead, "so sometimes the roll was 800 feet, and sometimes 950 feet. We'd cut it off and say, 'OK, this is KEM Roll #1,' and then we'd start KEM Roll #2. We just did it so it would be easy to deal with. We did the sound first, because that decided where the songs were going to go, and we never knew where the film [from any one camera] might run out."

Such a procedure was carried out for each of the three concerts filmed for the movie, each of which ran to roughly 2½ hours in length. The next job at hand was to go through each roll of film from each of the camera positions used at the three concerts shot, and log the location of timecode images on each one. With eight cameras in use at the two New Jersey concerts, and 12 at the Arizona event, this added up to nearly 400, 1,000-foot rolls of film; over 6 hours of footage in all.

"It's just like looking for your regular slate," says Hollingshead, "only in this case you're looking for a clock. You'd read out the hours, minutes, seconds, and frames [and its location in footage] on each roll, and write that down. Then we went through our list of slates, and looked for those numbers on our sound roll. We'd mark down Camera A, and write down the hours, minutes, seconds, and frames on the back of the track as close as we though it was."

The editing team would then use a six-gang synchronizer to assemble a rough set of film rolls (anywhere from 6 to 12) for each sound roll, initially utilizing the timecode slating for "ballpark" sync marks. Although it will not permit viewing of the film, the synchronizer or sync

block can run six units of film or soundtrack in frame-for-frame sync with one another. Sound heads usually are installed on the sync block so that audio — and, in this case, the SMPTE timecode — can be monitored.

Matching Image

"Now this is where we'd sit down and go through and check everything on the KEM," Hollingshead continues. "I'd say that we were within a frame of everything anyway, but we had the time, and we were perfectionists enough to put all the rolls up on the KEM and match movement.

"We'd get one camera that we knew was good — say the drum camera — and we'd match the drum beat to be sure that it was exactly in sync with the soundtrack. Then we'd go through and match the image on the other cameras to that one camera; there was always a hand or something that could be used to match."

On one of the screens on the KEM, the editors would have the drum camera matched to a hit on the snare on the audio track, and somewhere in the background Bill Wyman would have his hand on his bass. On another of the KEM's screens, they would then line up another roll of film that featured Wyman's bass, and match it to the drum camera's shot. Perhaps also on the bass camera shot was Keith Richards moving around. The editors would put another piece of film with Richards on the KEM's third screen, and match it to the camera with the bass.

This procedure would progress all the way through the six to 12 rolls of film, the editors giving each roll common start marks in relationship to the single audio track. "We were almost always able to find something," Hollingshead continues. "Even on those cameras that were not able to get the clock, we were able to lip sync them in the same way. We got so familiar with the picture by that point that we could identify unmarked film by what people were doing."

By the end of the process, KEM Roll #1's single reel of soundtrack would have from eight to 12 corresponding reels of picture. Each unit of each KEM Roll had leader at the head and tails, with start and finish sync marks. It was possible to remove the leader from the soundtrack, resplice the 35mm mag to the next KEM Roll's soundtrack, and then run the audio for the concert continuously. The same can be achieved for the different KEM Roll units of picture from a single camera, although leader would fill in sections where the camera operators had stopped filming to reload.

Edge code numbers now had to be added to the soundtrack and picture of each KEM Roll, both for identification and sync purposes. A number, beginning with 0000 is printed at the start mark of the roll of soundtrack, followed one foot later by the number 0001, and

— *continued overleaf . . .*

so on for the length of the reel. The same numbers are printed on each of the corresponding reels of film in the exact same place, so that at any point in the film, the editor can use the numbers to sync up the picture to track. In the same fashion as the audio and the picture of the KEM Rolls, once the leaders were removed, the edge code numbers would run consequtively from one KEM Roll to the next.

The KEM Rolls were also notated by a two-character letter code in front of the code number. Hence, at the start of the audio unit of KEM Roll #1, the edge code would read *AA* 0000; the number at the start of the first picture unit would *AB* 0000, and so on for the rest of the KEM Roll. The subsequent KEM Rolls would have a different first letter.

Video Editing

These synchronized rolls of picture and soundtrack, or dailies, were then ready for transfer to video tape for editing, a process Ashby has been experimenting with for some time. "I've done the last couple of films with video editing," he says. "It's quicker and more through, because you really search the film out more carefully. I'd been leaning very strongly towards this, transferring the film to tape, doing the cutting there, and then going back to the film [for the final cut]."

In designing Ashby's video editing system, Harry Howard and the director selected Sony's Beta I format and its institutional-type editing recorder. "At the time of its selection," Howard recalls, "Beta I represented the fastest scan available. Certainly the one-inch video tape recorders offered very fast scan, but they were also quite expensive. And, although they would prove faster, the broadcast series of U-Matic recorders had not yet been introduced. In terms of competing ½-inch video, the VHS system did not have any decent editorial equipment available at the time."

While the process of editing a film on video is not new, the methods of conforming the film workprint to the video work print — in other words, cutting the film to match the edits made on the video — has varied. At Northstar Media, it is accomplished by "burning" the film's edge code into the video picture. "It also runs in the same 24 FPS/30FPS anomaly as the film," Howard says, "but we have indicated which are extra frames in video, allowing the editors to get within plus or minus two frames on the film, or, if they utilize our chart, to get within plus or minus one frame."

KEEPING YOUR NUMBERS STRAIGHT

A Glossary of Timecode Formats Used in Audio Production

1. 24 frames-per-second timecode. During filming and recording of *Let's Spend the Night Together*, timecode was layed down on the 24-track master tapes and the ¼-inch Nagra tapes, as it was fed to video monitors photographed by the film cameras. In this fashion, sync addresses were supplied on both the picture and audio track. When the ¼-inch tapes were transferred to 35mm three-stripe, the 24 FPS code was layed off alongside the audio, enabling the 35mm soundtrack to be synchronized to the film work print.

When the film and 35mm three-strip had been edited, the 24-frame timecode on the mag stock was no longer sequential. Since the 24-track master recordings also held the original 24 FPS timecode, the cut code on the 35mm mag provided an edit decision list for conforming the 24-track tapes to the edited picture.

2. Edge code numbers. After a KEM Roll of picture and 35mm mag stock had been synchronized, they were given common start marks. Beginning at that point, edge code numbers were printed every foot on both pieces of material. In this way, the editor had sync addresses every foot of the film and soundtrack. When the film was transferred to Beta videocassette for initial editing, these numbers were burned into the picture along with frame numbers. Once the first cut of the movie was finished on tape, the editors used these numbers to conform the film work print and its 35mm mag to the edited video. The edge code numbers had no working relationship to any SMPTE timecode, however.

3. 30 frames-per-second timecode. This code was used to synchronize the videocassette of the completed film to the 24-track tapes that would hold the conformed sound. Since Regent Sound's editing process required 30 FPS timecode throughout, a jam sync process was employed to convert the 24 FPS SMPTE on the original multitrack "master" tapes to 30 FPS timecode. This synthesized 30 FPS code clocked exactly in hours, minutes, and seconds with its 24 FPS counterpart, with only a variation in the frame count.

The same 30 FPS timecode that linked the video to the "edited" 24-track tape also would link the video to mixing engineer Bob Clearmountain's "pre-dub" 24-track tapes, and link those pre-dub tapes to the film chain at Todd-AO for the final mix. ■■■

Beta I videocassette tape library containing film from three Stones concerts, photographed from up to 12 camera positions.

In Northstar's system, not only the edge code footage numbers, but the individual frames within each foot of film are burned on the screen. Once a segment of the film is completed by the editors, an edit decision list is made based on the footage and frames burned into the video.

In addition to the simple convergence editing system, Howard and Northstar put together for Ashby a system linking up 12 Beta I decks in sync. The system allowed the editors to rack up all 12 rolls of picture from one KEM Roll, and view them at the same time. A bank of 12 small Ikegami video monitors were mounted together for this purpose. "We call this feature the search system," says Howard. "It also contains 12 preset controllers or locators, so that you can precue the machines to certain points easily. Our modification for The Stones' picture was to allow those 12 machines to roll simultaneously. The effect was like moving into television's so-called 'line-cutting' situation. The camera images were in sync with each other, and the editors can perform rapid cuts [recorded on a 13th Beta deck], as in a live television switcher situation."

As the creative process began shaping the final product, it became evident that a number of songs would have to be shortened for the film. "We started out with songs that were 11 minutes long," says Hollingshead, "and you just can't run stuff *that* long. 'Just My Imagination' was 10 minutes long, and it came down to just over four minutes."

"I wanted to get in as many songs as I could," explains Ashby, "and the other thing that I wanted to do was make it like a good hour-and-a-half album. It was a 2½-hour show, but I didn't want it to be a 2½-hour film. ■■■

* * *

The conclusion of this article, to be published in the February issue, will consider the music editing stages required to match sound to the edited picture. In particular, part two will look at the various multitrack dubbing processes involved at New York City's Regent Sound, subsequent vocal and instrumental overdubs and "make goods," and the final soundtrack mixing at Todd-AO in Hollywood to a six-track master.

New Products

MICMIX MC-SERIES MODULAR AUDIO PROCESSING SYSTEM

Currently included in the Series is a 5¼-inch rackmount Card-Cage, as well as the MC-101 and the MC-201 signal processing devices.

The Model MC-F Card-Cage will accept up to five individual modules, and is powered by a low-noise toroidal transformer that allows all modules to drive 600 ohms. The MC-F is fully compatible with the dbx 900 Series units, as are all of the individual MICMIX modules.

The MC-101 is a single-channel of Dynafex noise reduction that provides up to 30 dB or more of noise reduction without encoding or decoding. Functions includes a threshold control, in/out switch, and a switch to set the appropriate signal reference level from -10, 0, +4, or +8 dB, allowing interface with a wide range of sound reinforcement, recording, or broadcast equipment.

The MC-201 allows the decay time of virtually any reverb device to be varied, and also provides up to 30 dB of noise reduction for the device. The module operates before the reverb return of any console, and allows the user to shorten the decay time of any plate (without damping), live acoustic chamber, or any spring system by up to 75%, without altering the tonal characteristics of the reverb.

Suggested user prices: MC-F $375; MC-101 $325; and MC-202 $325.

MICMIX AUDIO PRODUCTS, INC.
2995 LADYBIRD LANE
DALLAS, TX 75220
(214) 352-3811

For additional information circle #61

UREI ANNOUNCES NEW 813B MONITOR SPEAKER

The new 813B monitor features the same sound character of previous Time-Aligned monitors, with improved performance and specifications. Key to the 813B's performance is a new proprietary coaxial loudspeaker, the 801B, which doubles the power handling capacity and sensitivity of the 813A (the 813B's predecessor): maximum sound output is 6 dB greater, and high-frequency response is a third-octave higher.

The 813B retains the HF horn with a

diffraction buffer at its mouth for improved acoustical impedance matching, and smooth out-of-band response. The horn also features shadow slots to eliminate the problems of midrange shadowing common to conventional coaxial speakers. In addition, the 813B is equipped with a BNC connector for easy connection to the UREI 6500 power amplifier's Conductor Compensation feature, which extends a secondary amplifier feedback loop to the speaker, and effectively eliminates transient overshoots and other problems.

UREI
8460 SAN FERNANDO ROAD
SUN VALLEY, CA 91352
(213) 767-1000

For additional information circle #62

YAMAHA MODEL RM1608 RECORDING MIXER

The RM1608 has 16 input channels, each with an electronically balanced XLR mike input, and an unbalanced tape input. Two switch-selectable banks of output jacks are provided for the eight main program mixing busses, numbered to 1 to 8 and 9 thru 16, enabling 16-track recordings. Separate two-track tape inputs and outputs are provided for mixdown and monitoring of the stereo master tape.

In "Multitrack" mode, individual inputs may be switched to tape or mike. Another button, "Mixdown," not only selects tape return on all channels, but also simultaneously assigns these channel outputs to the stereo mixing bus. Built-in phantom power for condenser microphones is provided, with individual switches for each channel. The three-band, multi-frequency channel EQ and highpass filter each have bypass switches, allowing direct outputs to be used for recording with the fewest line amps.

A front panel patch bay has insert points for each input channel. The bay also includes 16 auxiliary patch points fed from rear-panel jacks, making it more convenient to link signal processor inputs and outputs to channel insert points. Post insert point levels are monitored by peak indicator LEDs, so signal-to-noise ratio can be optimized and distortion minimized.

In addition to the eight main mixing busses, there are two echo send busses. Echo #1 may be fed pre- or post-input fader, while Echo #2 derives signal post-fader or from the monitor control (for "wet" monitoring of the rough mix during overdubs). To accommodate stereo echo chambers, each of the two echo returns is stereo.

The RM1608 is expected to retail for approximately $6,600, and will be available in April, 1983.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134

For additional information circle #63

LOW FREQUENCY SPEAKERS WITH MATCHED ENCLOSURES FROM ALTEC

Designed as integral systems, the speakers of the new 3000 Series were developed in conjunction with the matching 8000 Series loudspeaker enclosures. The six new woofers include three extended low-frequency models — the 12-inch 3124, 15-inch 3154, and the 18-inch 3182 — and three high-efficiency speakers: the 12-inch 3127, 15-inch 3156, and the 18-inch 3184. The optimally-tuned enclosures range from the 1.5 cubic-foot 8127, to the 24 cubic-foot 8182.

Offering improved long-term performance and reliability, the new speakers are said to provide significant advances in such areas as linearity and power handling. The long voice-coil geometry of the 3000 Series reduces distortion by increasing the diaphragm's linear motion, and boosts power handling through increased heat dissipation.

— continued overleaf . . .

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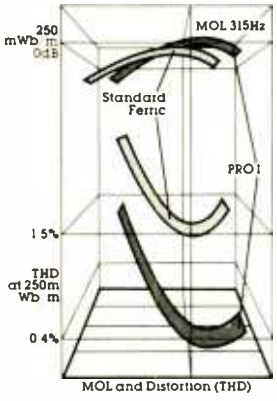

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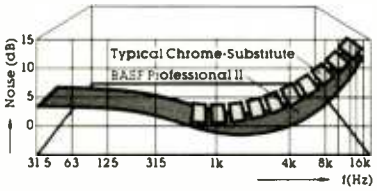
Unlike ferric oxide particles, chrome particles are similar to each other in shape and size. This uniformity and the freedom from physical imperfections provide chrome with electro-magnetic properties for superior dynamic range (signal-to-noise ratio) and lowest noise (tape hiss).



The difference in noise level between PRO II and ordinary high bias tape is greatest where the human ear is most sensitive (2-6 kHz)

When you want a tape that delivers all of the music and none of the tape, Professional II is your tape for the Type II/chrome/high bias position.

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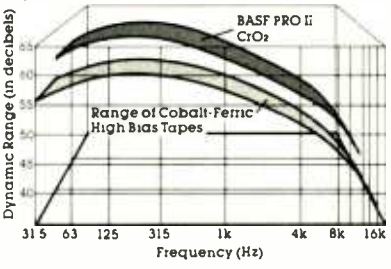

In comparing the Total Dynamic Range of BASF PRO II chromium dioxide tape with ordinary high bias tapes, PRO II is audibly clearer at frequencies above 8 kHz

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For additional information circle #59

New Products

Precision matching of speakers and enclosures in the new Series offers many advantages, Altec claims. The 12-inch 3127, for instance, pairs high efficiency and compact enclosure size (only 19 by 16 by 14¾ inches) with a frequency response smooth down to 70 Hz (-3 dB) when coupled with the Altec 8127 enclosure.

ALTEC LANSING
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For additional information circle #66

FOSTEX MODEL 3180 STEREO SPRING REVERB

The new unit features a multiple spring design, and a unique delay matrix that delays the signal before it reaches the spring transducer. This delayed dry signal is then added to the reverberated signal at the output via a matrix system.

"The distinguishing sound that results from this design is a minimum 24 microsecond delay at high frequencies, unlike typical spring reverbs which have little or no delay," Mark Cohen, VP of marketing and sales, says. "And because of this matrix system, each channel of the Model 3180 is capable of producing a stereo effect from a mono source."

Additional features include LED) overload indicator located just before the drive circuit for effective level matching; built-in limiter; independent reverb and dry mix controls; remote jack for foot switch operation; and both front and rear panel inputs/outputs.

Suggested list price of the Model 3180 is $400.00.

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(213) 921-1112

For additional information circle #67

TASCAM SERIES 50 STEREO AND EIGHT-TRACK TAPE MACHINES

The new two-track ¼-inch Model 52, and eight-track half-inch Model 58, accommodate up to 10½-inch reels and operate at 15 IPS; the Model 52 also operates at 7.5 IPS with NAB or IEC EQ. Full sync recording is standard on both machines, and a choice of optional remote controllers are available.

Both machines have rear-panel accessory connectors that are compatible with most popular SMPTE controller/synchronizers. Unlike some "SMPTE compatible" transports, all three of the Series 50 motors — supply, takeup reel and capstan — support remote commands. This means that manual cueing is not necessary in order to remain locked up during rapid winding and searching.

All transport functions are governed by a microprocessor that detects tape motion by means of five different photo-interruptors. These non-contacting sensors are said to eliminate flutter-causing friction and wear, and continuously inform the computer of the tape direction, speed, tension, and stop or end of tape condition.

A fluorescent tape counter displays postive and negative real time. Coarse and fine speed control sliders are provided for Vari-Pitch mode. The Zero Search button rapidly moves tape forward or in reverse to exactly 00 min 00 sec, and single point Search-to-Cue also is included. Because the search circuitry is built in, a multipoint autolocator is provided in the full-function remote unit.

TASCAM/TEAC CORPORATION
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

For additional information circle #68

SOUND TECHNOLOGY 1510A TAPE MACHINE/AUDIO TEST SYSTEM

With the introduction of the 1510A Tape Recoder/Audio Test instrument, an audio engineer has the capability of performing all of the necessary tests for maintenance, troubleshooting and general check-out of any professional audio device, whether it be tape machine, film machine, mixing board, reference turn-

table, parametric equalizer, or any other outboard device.

The two-channel outputs are electronically balanced and floating; output levels are from +30 to -70 dBm into 600 ohms, with a pushbutton resolution of 0.1 dB. For those facilities with automation in mind, the IEEE-488 general purpose computer interface bus is available.

The 151A will measure and display: AC voltage; azimuth phase error; second or third harmonic distortion; frequency response; channel separation; noise; wow and flutter; speed; drift, maximum operating level; and dropouts.

SOUND TECHNOLOGY
1400 DELL AVENUE
CAMPBELL, CA 94008
(408) 378-6540

For additional information circle #69

MCI/SONY COMPACT JH-800 CONSOLE

The JH-800, a portable 12-input mixing console with four VCA controlled sub-groups, is the company's first product for use outside the recording or broadcast studio.

According to MCI/Sony, the JH-800 is intended for location recording, video production work, and will have applications in remote broadcasting.

Features include dual stereo mix busses; fluorescent bar graph metering; balanced transformerless line and microphone inputs; three-band equalizer on each input; multiway and XLR connectors; and two built-in stereo compressor/limiters. The unit also includes four sends for foldback and effects, or feeding stage monitors in sound reinforcement applications.

MCI/SONY
1400 W. COMMERCIAL BLVD.
FORT LAUDERDALE, FL 33309
(305) 491-0825

For additional information circle #70

TIMEMASTER DIGITAL TIMER UNVEILED BY DOMAIN

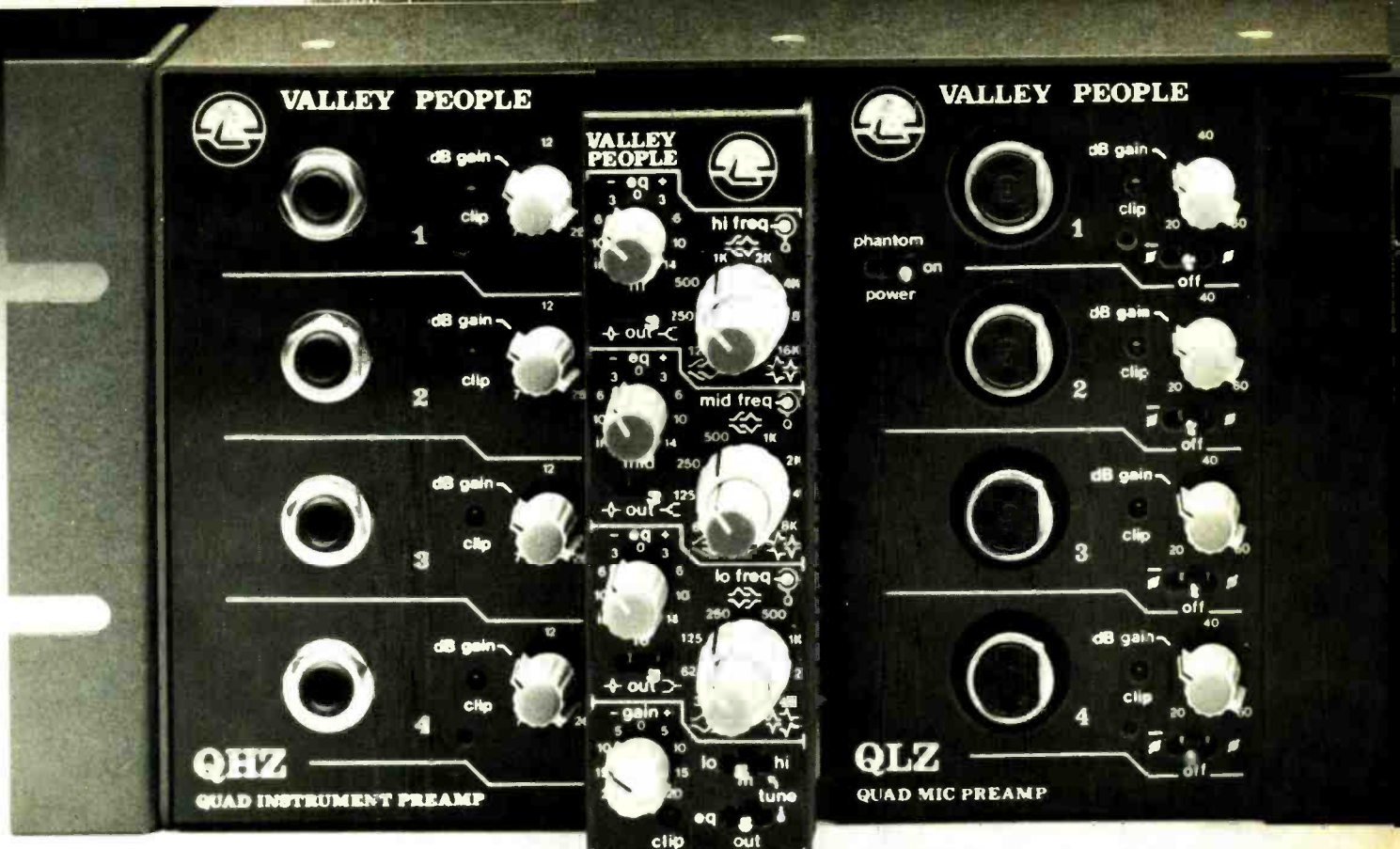
The TimeMaster is a crystal controlled, 100-minute timer with seven different timing modes for practically any studio situation. The device has finger touch controls that "stop," "start" and "reset" from up to five feet away. A remote display with 20 feet of connecting cable can be placed in an adjoining studio booth giving the talent instant readout of the producer controlled timings.

In addition, an automatic warning in the "count down" mode alerts the talent when there's one minute remaining. This warning alert is also available in manual set in all other timing modes.

With simple connections, the TimeMaster can even be started automatically by tape recorders, cart machines or turntables, freeing an engineer's or producer's hands during complex mixing or cueing segments.

DOMAIN COMMUNICATIONS
P.O. BOX 337
WHEATON/CAROL STREAM, IL 60187
(312) 668-5300

For additional information circle #71



QHZ (814) Now, your high impedance equipment is afforded instant accessibility to the superior signal processors found in our '800' series. The full brilliance of any high impedance input source is captured with our transformerless technology, providing a true audible difference.

MAXI Q (812) You'll be hard pressed to find a 3-band parametric equalizer that offers: a 6-octave overlap for more effective spectrum coverage; a bandwidth control active in the shelving mode, for dramatically varying slope effects; non-reciprocal E.Q. curves for less loss of desired frequencies; and series filters to eliminate filter interaction. There's even a 'tune' mode, allowing the operator to 'solo' filter settings, before effecting equalization.

QLZ (813) For those times when noisy transformers just don't make it, we've got the answer. Four independent channels of transparent, low noise, distortion-free microphone preamplification. And, of course, 48 volt 'phantom' powering is provided.

Go for the Gold

New Products

APHEX SYSTEMS INTRODUCES TYPE B AURAL EXCITER

"The Aphex Aural Exciter enhances the signal by providing greater intelligibility, presence and detail while maintaining the natural qualities of music and voice," said Marvin Caesar, president of Aphex Systems. "It adds 'brilliance and clarity' to instrumentals, and 'punch and definition' to vocals," he stated. "The total result is a fuller, more dimensional sound."

The Type B Aural Exciter, which features simpler front-panel controls than previous Exciter models, and retails for $495.

APHEX SYSTEMS, LTD.
7801 MELROSE
LOS ANGELES, CA 90046
(213) 655-1411

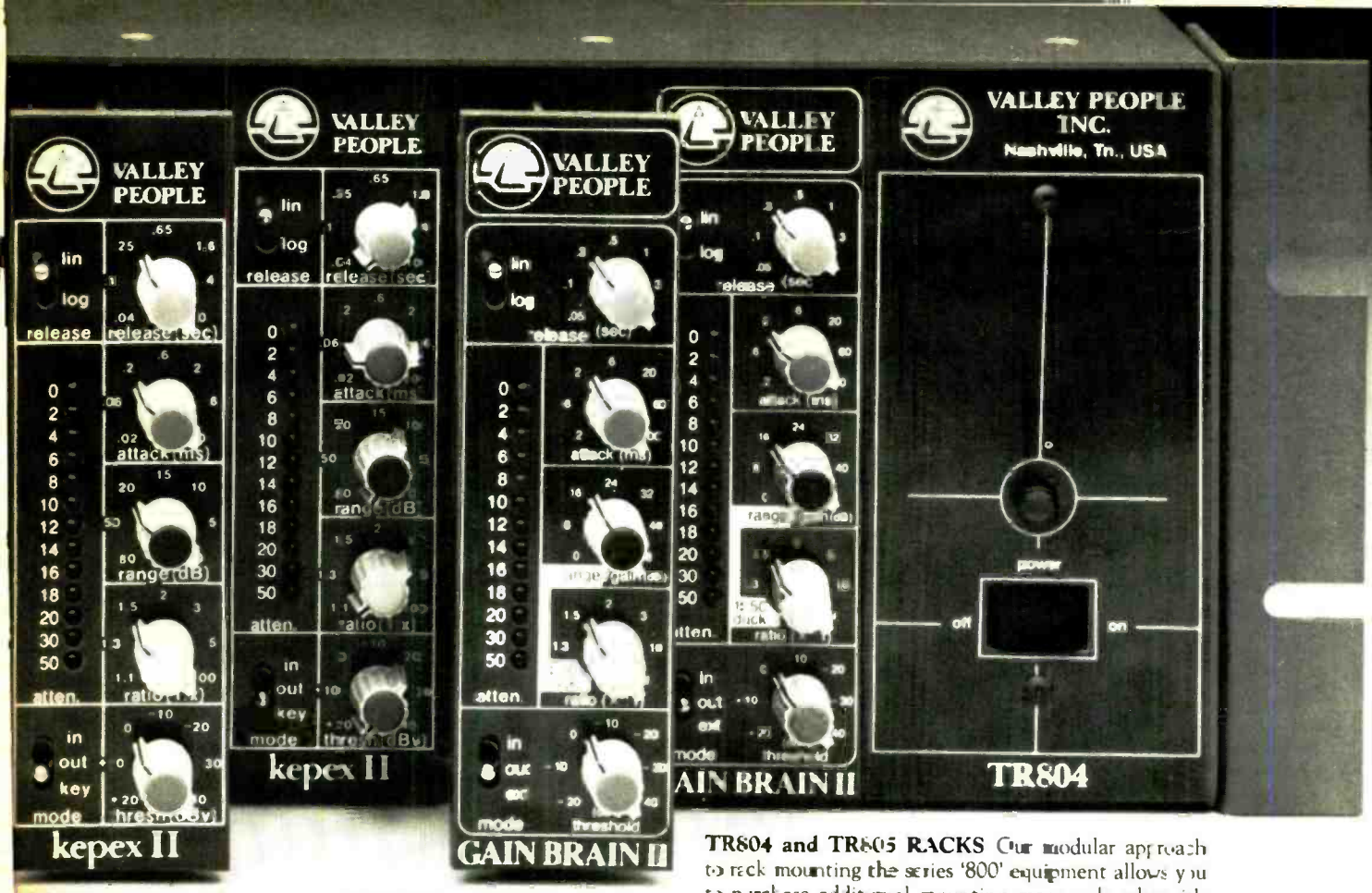
For additional information circle #73

MICRO MIXER IIA FROM IMAGE DEVICES

The new Micro Mixer IIA features electronically balanced input facilities for mini-jack (video) and a ¼-inch phone jack (film). Radio frequency interference has been virtually eliminated, with

frequency response within 1 dB down to 20 Hz.

Weighing just 30 ounces, the IDI Micro Mixer IIA is intended for hand-carrying with a portable video or audio



KEPEX II (810) The name KEPEX is synonymous with noise-gating, as everyone knows. But, if superb dynamic range has eluded you, then the 810's active expansion mode solves the problem. And, in the keying mode, an endless number of electronic music effects are possible.

GAIN BRAIN II (811) The limiter/compressor with a "brain". It's true, the 811 understands music, and processes it the way your ears want to hear it. The 811's response is variable depending upon waveform complexity. "Smart" circuitry is employed throughout to avoid discrimination against low frequencies and counteract excessive pumping and dynamic distortion.

TR804 and TR805 RACKS Our modular approach to rack mounting the series '800' equipment allows you to purchase additional mounting space only when it's needed. The TR804 powered rack holds up to 4 single-card modules and provides power for nine '800' series processing cards. When you require additional rack mounting space, the TR805 unpowered rack holds up to 5 single-card modules and derives its power from the TR804 power supply.

The 800 Series From:

VALLEY PEOPLE, INC.
P.O. Box 40306/2820 Erica Place
Nashville, Tenn. 37204
615-383-4737
TELEX 558610 VAL PEOPLE NAS

Export: COTHAM EXPORT CORPORATION
NY, NY/Tlx: 129269

For additional information circle #72

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recorder. AB mike powering, bass cut and other custom options are available.

With disposable battery pack, the Micro Mixer IIA costs $595.00; with NiCd battery pack, $695.00.

IMAGE DEVICES
1825 NE 149 STREET
MIAMI, FL 33181
(305) 945-1111

For additional information circle #74

NEW RANGE OF ALTEC MID-SIZE MANTARAY HORNS

Approximately half the size of the first series, the four members of the new Mantaray II family include horns with vertical/horizontal coverage patterns of 40/20, 60/40, 90/40, and 120/40 degrees.

Altec engineer Mark Ureda states that the new generation of Mantarays

"has allowed us to significantly improve the low-frequency loading characteristics, while maintaining precise directivity control superior to that found in radial and exponention-type horn designs."

Unlike the earlier two-piece, sheet-metal Mantaray line, Mantaray II horns will feature one-piece, fiberglass construction.

ALTEC LANSING
1515 S. MANCHESTER AVENUE
ANAHEIM, CA 92803
(714) 774-2900

For additional information circle #75

NEW SERIES OF OUTBOARD PROCESSORS FROM TASCAM

Leading off the new series are the MX-80 microphone mixer, MH-40 multi-headphone amplifier, PE-40 parametric equalizer, and RS-20 dual reverb.

The MX-80 compact mike mixer accommodates eight input channels, and provides a stereo output. Mike inputs are on balanced, low-impedance XLRs. In addition to concentric gain and pan controls, each channel has a trim control and a 30 dB pad. Phantom

New Products

power (from an external supply) may be switched independently on each channel.

The MH-40 headphone amplifier accepts any line level mono or stereo signal via front panel phone jacks or rear panel RCA jacks. Input sensitivity is switchable for -20 dBV to +4 dB sources, and a front-panel input control permits overall level adjustment. The signal is then distributed to four stereo phone jacks via independent output amplifiers.

The PE-40 is a four-channel, four-band fully parametric equalizer. Two or more PE-40 channels can be cascaded when maximum control is needed. To reduce rumble, wind noise and feedback, each channel has switchable highpass filters, as well as a lowpass filter to reduce hiss.

The RS-20 dual reverb system occupies just 3½ inches of rack space, and utilizesthree different sized springs on each channel, to produce a well-balanced sound that is said to extend at least an octave above conventional spring reverbs, without the need for direct (capacitive) feed through. Limiters ahead of the spring drive amps prevent "twangy" sounds caused by transients.

TASCAM/TEAC CORPORATION
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

For additional information circle #78

A710 PROFESSIONAL CASSETTE DECK FROM STUDER

The A710 features balanced and floating line-level inputs and outputs, factory set at +4 dBv and internally adjustable over a wide range. Maximum output level is +21 dBv into 200 ohms. Input and output calibrate/uncalibrate buttons are provided on the front panel; in the "uncalibrate" position the level controls may be used to provide an additional 10 dB of gain.

Other standard A710 features include XLR connectors; rack mount flange; connectors for fader start and remote

contorl; Dolby B and C noise-reduction; four-segment LED counter; peak-reading bar graphs; and three-head pivoting head block.

The A710 transport is a four-motor direct-drive design, with two quartz-controlled, Hall-effect motors driving dual capstans, and two DC spooling motors to provide constant speed fast wind and rewind, tape tension control, and gentle electronic braking. All transport and electronic switching functions are microprocessor controlled.

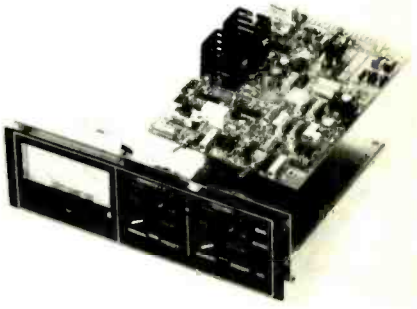
The new A710 has a suggested retail list price of $2200.00.

STUDER REVOX AMERICA, INC.
1425 ELMHILL DRIVE
NASHVILLE, TN 37210
(615) 254-5651

For additional information circle #79

REPLACEMENT AMPEX ATR-100 I/O CARD FROM STRATEGIC SOUND

The new transformerless ATR-100 input/output PCB replacement is said to offer simplified set-up and alignment. A supermatched transistor pair, with 0.01% resistors, maintains very low noise with maximum common mode rejection ratio. The output uses the new OP-37 precision op-amps, in conjunction with VMOS and ring emitter transistors, for minimum distortion.



By using only precision components throughout, SSI has eliminated four internal DC offset and calibration adjustments. Features include: transformerless coupling; no internal adjustments; dual voltage shutdown; power on/off mute protect circuit; and easy PCB replacement.

Improved specifications include variable input sensitivity from -10 dBm to +40 dBm, input CMRR greater than 70 dB; 10 Hz to 1 kHz, with an input clip level in excess of 26 dB above operating level. Frequency response is quoted within 0.1 dB, 10 Hz to of 20 kHz, and within 1 dB, 5 Hz to 100 kHz. Output noise is better than -90 dBA below operating level, and input-to-output system

slew rate better than 20 volts per microsecond.

List price is $1450 per pair.

STRATEGIC SOUND, INC.
P.O. BOX 3148
REDWOOD CITY, CA 94064
(415) 797-7203

For additional information circle #80

YAMAHA UNVEILS PC5002M POWER AMPLIFIER

Designed for use in very large sound systems, or for studio monitor systems where extra headroom is required, the PC5002M delivers 500 watts per channel into 8 ohms, or 750 watts into 4 ohms. By selecting a rear panel mono switch, the amplifier outputs can be

bridged to deliver 1,500 watts into 8 ohms. To allow standard 15-amp AC circuits to be used, the PC5002M has two AC power cords, with independent power supplies for each channel.

The amplifier includes a pair of large peak-reading meters that display power output in watts (into 8 ohms), or decibles. Harmonic distortion is quoted below 0.005% (20 Hz to 20 kHz, both channels, 250W at 8 ohms), and intermodulation distortion below 0.01%. The PC5002M also maintains wide power bandwidth, and has frequency response of 10 Hz to 50 kHz, +0, -0.5 dB; noise is more than 115 dB below maximum rated output.

An LED clipping indicator is standard on each channel, as well as newly designed protection circuitry that senses DC on the output as well as thermal overload, shutting down the amp to prevent speaker or amp damage.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134

For additional information circle #81

DELTALAB EFFECTRON ADM-64 EFFECTS UNIT

Flange ratio of the new unit is said to be twice that of any other digital unit currently available. The ADM-64 provides a full three octaves of flanging (8:1 flange ratio), and includes an internal envelope follower control voltage for enhanced flanging effects.

Doubling and short echoes are other features of the ADM-64. At this setting a range from 16 to 64 milliseconds of high-performance digital delay is available. Unlike most other delay units, the Effectron Series are said to maintain full 16 kHz audio bandwidth and full 90 dB dynamic range at all delay settings.

Suggested retail price for the ADM-64 is $399.00.

DELTALAB RESEARCH, INC.
27 INDUSTRIAL AVENUE
CHELMSFORD, MA 01824
(617) 256-9034

For additional information circle #82

RENKUS-HEINZ UNVEILS MODEL CBH1600 CONSTANT BEAMWIDTH HORN

The new 1-inch throat horn has a cutoff frequency of 1.6 kHz, and is designed for constant coverage in the horizontal and vertical planes. A fast flare rate integrated with a short path length reduces second harmonic distortion, and allows mounting in the low-

On the Studio Trail
this month . . . New York

— *continued from page 76 . . .*

around an 8068 with NECAM servo-assisted automation, and A80 multitracks and mastering machines. In keeping with most of he studios I have visited recently, Electric Lady runs the majority of its sessions at 30 IPS without noise reduction.

Quizzing studio manager Mary Altenpohl on the type and volume of work that Electric Lady had been attracting, I discovered that practically 100% of bookings are for music production, with no jingle sessions. Mary said that the studio was having a very good Fall this year. Studio staff were bracing themselves for a slow period, she recalls, but it didn't seem to materialize.

In contrast with the majority of studios that need to attract business from major and independent labels to stay alive in an increasingly competitive studio scene, Atlantic Studios, located just off Broadway near Central park, would seem able to survive on its in-house label work. As I discovered from general manager Paul Schloman, however, only around 50% of studio time is booked by Atlantic records and its subsidiary labels, the remaining sessions being for a mixture of independent productions, and for other labels.

At the time of my visit, plans were underway to upgrade Studio A of Atlantic's three-room complex (two studios and a remix/overdub room). Sierra Audio will be completely refurbishing the present control-room acoustics, and building a new iso-room, although no major reconstruction of the studio itself is planned for the immediate future. Studio B currently boasts a Neve 8108 with NECAM automation, linked to a Studer A800 24-track.

A separate room houses a Sony PCM-1610 digital processor, U-Matic videocassette recorders, and companion digital editing system, which Atlantic uses to run companion analog and digital two-track masters, and save multiple generation losses when preparing bin masters required for cassette and 8-track duplicating. According to Schloman, Atlantic also is contemplating the purchase of a digital multitrack machine, but staff are still considering "any and all" models currently available before making a final decision.

The key attitude at Atlantic, Schloman offers, is "to make the artist as comfortable as possible, and able to make a better record in that creative environment. We also pay attention to our technical excellence, which should be taken for granted by clients."

Regarding plans for the future, Atlantic intends to offer audio/video capabilities in the near few months, including a post-production facility for remixing multitrack to video.

A couple of blocks down Broadway from Atlantic, Sigma Sound's Gerry Block sees the future of the studio business leading to two, distinct types of facility. The first, which for want of a better expression Block refers to as Type #1, will be a "very cheap, up to 24-track studio with, in the main, used equipment." Such a facility, he predicts, "will attract serious people outside of the mainstream, high-budget market." Such clients will "cut corners where necessary, even to the extent of sacrificing having a good engineer around on the sessions. They just want to be able to make a record, and can probably work without too many frills," he predicts. Type #2 studios, on the other hand, will feature a lot more in the way of recording hardware, and sound benders and blenders; as a result, studio rates will be a lot higher. "But such studios will offer everything a client could need," Block says.

By way of an example, he considers Studio 5 in Sigma's two-floor complex to be a Type #2, "state-of-the-art" room (Neve 40-input 8068 with NECAM automation; Studer A800 multitrack; Ampex ATR-100 mastering deck with quarter- and half-inch headblocks; Altec 604 monitoring; three EMT plates), while Studio 7 is, in his own words, "funky" (Translation: Type #1). The remaining area, Studio 8, is designated a remix and post-production room, and features a 48-in/16-out custom-designed console with Allison Research "Memory Plus," tape-based automation, a pair of 3M 79 Series 24-tracks, and a good selection of outboards, including Lexicon 224 and EMT 250 digital reverbs, three EMT 140 plates, and assorted flangers, DDLs, gates, and equalizers.

With such a selection of rooms, it is perhaps hardly surprising that Block considers Sigma "isn't in the music recording business; we're in the audio recording business." Coining the phrase "A Studio for All Seasons" to describe his facility, clients can work in a variety of recording formats, ranging from 35mm mag stock, through two-track half-inch, to ¾- and 1-inch video. Regarding digital hardware, Sigma has opted for a Sony PCM-1610/U-Matic/Digital Editor combination, since Block considers PCM U-Matic videocassettes "to be the only format that will make any impact on the recording scene. It enables a lot of transfers to be made without loss of quality, and also multiple copies for disk cutting."

Frankford/Wayne is one of New York's more popular mastering facilities, and currently is running no less than four rooms, one of which features two tandem lathes. Studio manager Norvell Miller tells me that the facility has become especially well known for its "disco" cuts; engineer Herb Powers Jr. having earned a reputation around town as being "the" disco cutter.

On the digital front, Frankford/Wayne also has several Sony processors, including a PCM-1610, which are used not only for mastering, but also made available for outside rental. Norvelle says that

between 1 and 2% of masters coming to Frankford/Wayne are in digital PCM format. In addition, the studio encourages clients to make digital copies of a cut, especially if many parts are needed, both to simplify recuts and the preparation of cassette bin master tapes. And continuing the digital theme, most rooms are equipped with an AMS 16-bit digital delay line to provide preview signals for the cutting lathes. In Joe Gastwirt's room, for example, an MCI JH-100 stereo deck with quarter- and half-inch headblocks is linked via an AMS DDL to a Ranstelle console equipped with Sontec and UREI EQ. A Ranstelle-modified Scully console has been fitted with Sontec Compudisc automation which, according to Gastwirt, enables more level and playing time to be coaxed onto vinyl.

As will be readily appreciated, the studio scene in New York would appear to be alive and well, and holding its own. As with most metropolitan areas containing a high concentration of state-of-the-art facilities, competition between studios for the traditional record business music sessions is becoming very competitive. Diversification into ancillary areas of audio recording — such as jingle sessions, audio/video sweetening and, in some instances, film scoring — might enable a facility to better weather the changing nature of the recording industry.

Next issue . . . the small-studio scene in BOSTON.

■■

news

— *continued from page 12* . . .

single supplier." The acquisition is subject to approval by the Harman International board, and acceptance of a tender offer by URC stockholders.

REORGANIZATION OF SOUNDSTREAM ANNOUNCED BY DIGITAR RECORDING CORP

Dr. Thomas G. Stockham, Jr., founder and president of Soundstream for the past 7½ years, has resigned as president and been appointed Chairman of the Board of Soundstream, a wholly-owned subsidiary of DRC. Robert B. Ingebretsen has been appointed president and chief operating officer of Soundstream.

According to Robert DeForest, DRC president and chief executive officer, this change will permit Dr. Stockham to concentrate his strong technical expertise in the long-term technical developments of both Soundstream and Digital Recording Corporation.

NEVE TO SUPPLY TRANSPORTABLE CONSOLE SYSTEM FOR CBS RECORDS

Working to a brief from CBS for an optimum quality recording console that could be easily transported, *and* go through a standard door opening, Neve's designers came up with a sectionalized version of the company's new 51 Series range which, when assembled on site, forms a 36-channel, 24-track recording console.

The session for the new system was a live classical music recording session at the Lincoln Center in New York, with the city's Philharmonic Orchestra. Many more live recordings at major concert halls throughout the USA are scheduled. The sectionalized board will be based at CBS Records' studios in New York City. Transportation to outside venues will be made within 7 custom-built aluminium flight cases with wheels for ease of mobility.

NEW SUPERTRAMP ALBUM ON BASF CHROME TAPE

The new Supertramp album, *Famous Last Words*, will be issued on BASF chromium-dioxide cassette tape, thus making A&M Records the first American company to make a general album release on high-fidelity tape.

While the group was completing plans to release *Famous Last Words*, BASF was in contact with A&M engineering VP Marv Bornstein to explain new methods of translating studio-quality sound to cassettes duplicated at high speeds through the use of BASF chrome, the same tape used in BASF Professional II cassettes.

According to Bob Piselli, national

— *continued overleaf* . . .

news

sales manager for BASF, the unusually large production run required for the new album cassette release made possible special arrangements for A&M.

As a means of testing public interest in improved quality for popular mass market releases, A&M and BASF reached an agreement that, for the first time in the industry, would allow the release of prerecorded chromium-dioxide cassettes at standard ferric tape prices.

CRAMER APPOINTED FIRST US STUDER DEALER

In addition to the A80 and A800 tape machines Cramer Audio/Video of Needham, Massachusetts, already is finding interest in Studer's cassette deck, production consoles, telephone hybrid and especially the new A810 two-track, which features microprocessor control and optional SMPTE head.

The company reports that its first sale was a 24-track A80-VU to Sound Design Recording Studio, in Burlington, Massachusetts.

PEOPLE ON THE MOVE . . .

• **Hans D. Batschelet** has been appointed president of Studer Revox America, effective January 1, 1983. The announcement was made by the outgoing president, **Bruno Hochstrasser**, who has returned to the Studer factory in Switzerland to assume the position of product manager for Professional Recording Systems. Batschelet, formerly vice president of marketing for the Studer Division, will now direct all Studer Revox operations in the US from the company's corporate headquarters in Nashville. Also, **Peter Kehoe** has been chosen to fill the position of field Service engineer at Studer Revox America's New York office. Kehoe previously served as a maintenance engineer at the Hit Factory recording studio in New York.

• **James L. Hartley, Carl S. Jorgensen,** and **Philip H. Sutterlin** recently formed Strategic Sound, a new corporation "dedicated to raise the standard of performance of the analog recording chain beyond the reach of the competition." All three are former Ampex Audio Division personnel.

• **Robert Piselli** has been named as national sales manager of BASF Systems Corporation's new Professional Products Department. The new department will consolidate all sales and marketing of BASF professional products, including duplicator tapes, broadcast cassettes, IEC calibration and reference tapes, Headmaster and audio/video cassettes. The BASF Professional Products sales force currently comprises **Ed Raftery** in the East, **Jerry Shields** in the Midwest, **Mike Ingalls** in the West and **Phil Conforti**, who covers the Southern states. **Paul Kontrimas** has been named professional products manager.

• **Terry M. DeRouin** has been appointed to the newly-created position of domestic marketing administrator for URSA MAJOR, Inc., with responsibility for coordinating US sales of the SST-282 and 8×32 digital reverberators. DeRouin, who will operate out of Whitefish Bay, Wisconsin, has extensive experience as the head of a pro audio products rep firm, and as a former studio owner and manager.

• Studio guitarist **Mike Elliott** has been appointed MXR's Musical and Professional Product Clinician. Elliott will be sharing his expertise in a series of MXR clinics on the use of effects as instruments. He is currently music director of LMJ Recording Studio in Nashville.

• **Lorry Marcus** has been appointed sales engineer for ADA Signal Processors. Lorry previously was sales administrator with Crystal Clear Records, and now will be responsible for developing distribution of the new ADA Professional Products line.

• **Richard Sirinsky** has been appointed director of sales development for Ampex Corporation. Sirinsky, formerly marketing manager of the Ampex Audio-Video Systems Division, will now direct the activities of the training and teleproduction center, and corporate advertising department. Prior to his assignment as AVSD marketing manager, Sirinsky was vice-president and general manager of the Europe, Africa and the Middle East area for Ampex International, based in Reading, England.

• **Tom Carlile** is the new president of Gauss Loudspeakers (based in Sun Valley), while **Hans Freytag** has been appointed European sales manager of Gauss Loudspeakers, based in England. Carlile joins the company after five years as president of New West Audio Marketing Inc., of Woodland Hills, California. Prior to his affiliation with New West, he had been national sales manager of Gauss Loudspeakers, from 1974 to '77. Freytag also has been in the professional audio field for more than 10 years, serving with English companies ATC, PACE, and Eastmill Systems.

■■■

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- *Hamburg, West Germany* — **Salten & Selten**
- *Tokyo, Japan* — **Trichord Corporation**
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book review

THE PSYCHOLOGY of MUSIC

Edited by Diana Deutsch
559 pages; $50.59, including postage

To be in the recording business today, you have to be at least somewhat aware of the science of acoustics — the way that sound behaves in a physical environment. But chances are likely that you've never dealt much with the field of *psycho*acoustics — the way that sound behaves once it reaches the ear, and starts to get processed by the brain. Of course, a few psychoacoustic principles are applied, consciously or not, in the day-to-day work of recording — like the Fletcher-Munson curve, which says that the ear is less sensitive to tones at the low and high end of the spectrum at low volumes. Also, the fields of studio and equipment design use some of these ideas — such as LEDE rooms and, in the world of consumer hi-fi, the Carver "Sonic Holograph," both of which take advantage of recent research in phase sensitivity.

But going beyond these rudimentary applications is a wealth of research that has been done by scientists, far-removed from the recording industry, on the way the brain-ear combination actually hears sound. Many in our field are familiar with "On the Sensations of Tone," by the 19th-century physicist Hermann von Helmholtz. At the time, the book was the most comprehensive look at the subject, and it is still a very useful reference for many applications. But it was written 120 years ago, and the field of psychoacoustic research has travelled a long way since then. Helmholtz, for one thing, was severely limited in his work by the fact that pure tones for his experiments simply were not available. He had to rely on musical instruments for his sound sources which, as we know, produce tones that constantly change — even if two trumpet tones *sound* the same, a quick check with an oscilloscope can verify that their absolute pitch, attack, timbre, and envelope may vary widely. Given these restrictions, it's not surprising that many of Helmholtz's conclusions were misleading, or even downright wrong.

It's only since development this century of electrical sound synthesis that psychoacoustics has progressed into an exact science, and more recent applications of computer technology have resulted in further refinements. That is the point made in the opening chapter of this new book edited by Diana Deutsch, a psychologist at the University of California, San Diego-La Jolla. Deutsch has collected 18 essays on various aspects of the field, written over the last decade by a wide spectrum of European and American contributors: professors of psychology, music, and audiology; researchers in acoustics, electronics, and perception; and musicians.

Each chapter is of value for anyone involved in the recording industry, but some hold extra interest for specialists in certain fields. For example, Chapter 2 — "Exploration of Timbre by Analysis and Synthesis" — makes fascinating reading for anyone working with synthesizers, particularly if they are attempting to simulate real musical instruments. It shows how the synthesis of natural tones is an incredibly complex process, but then offers shortcuts that effectively fool the ear. Chapter 3, "Perception of Singing," is of at least as much interest to singers as it is to psychologists, and helps to explain some of the difficulties in recording singers in a natural-sounding environment. Chapter 4, "Grouping Mechanisms in Music," is full of information for composers and arrangers. It explains, for example, how the ear often takes widely disparate notes and turns them into melodies that aren't really there. Chapter 7, "Timing by Skilled Musicians," makes the startling conclusion that even the finest musician's sense of rhythm is often way off — and explains why some people feel that drum machines sound somehow inhuman.

Other chapters deal with absolute pitch (how do we learn it, and what is it good for?); the role of interaction with other musicians and hearing mechanisms in musical performances (monitor engineers take note); the limits of musical memory; the role of music in social situations (does the *Rite of Spring* cause traffic accidents?); and the mathematics and psychology of musical scales.

Be warned, however — this book is not light reading. It is organized as a textbook, complete with footnotes and bibliographies of books and articles you've never heard of, and the writing style tends towards the stuffy. In several of the chapters, just when things start to get interesting, it bogs down in statistical equations and analysis. Even if you pride yourself on your ability to program microcomputers and compute distortion figures of 23rd harmonics, you'll find that the science of psychological statistics is something else again — quite alien to anyone except psychologists. Don't give up, however. The quadratic equations can usually be skipped right over without losing much of the value of the material being presented.

Unfortunately, the editing is uneven, and while there are graphs and tables aplenty, some of them are explained rather poorly. The ordering of the chapters is not the best either. There is a certain rhythm and sense to the progression of subjects, but the least readable chapter in the book happens to be the first one, "The Perception of Musical Tones." While it is certainly worthwhile reading, you might want to tackle it after you've made it through some of the others.

Someday, we can hope, there will be an easy-to-read treatment of psychoacoustics, but until then this book will do very well. If you're the studious type, you might want to take it along on a vacation — certainly it's a little hard to get through when your mind is on more mundane things like doing sessions. And of course you don't have to digest it all at once. But once you've been through any of the chapters, you'll start looking at the work you do in a whole different light.

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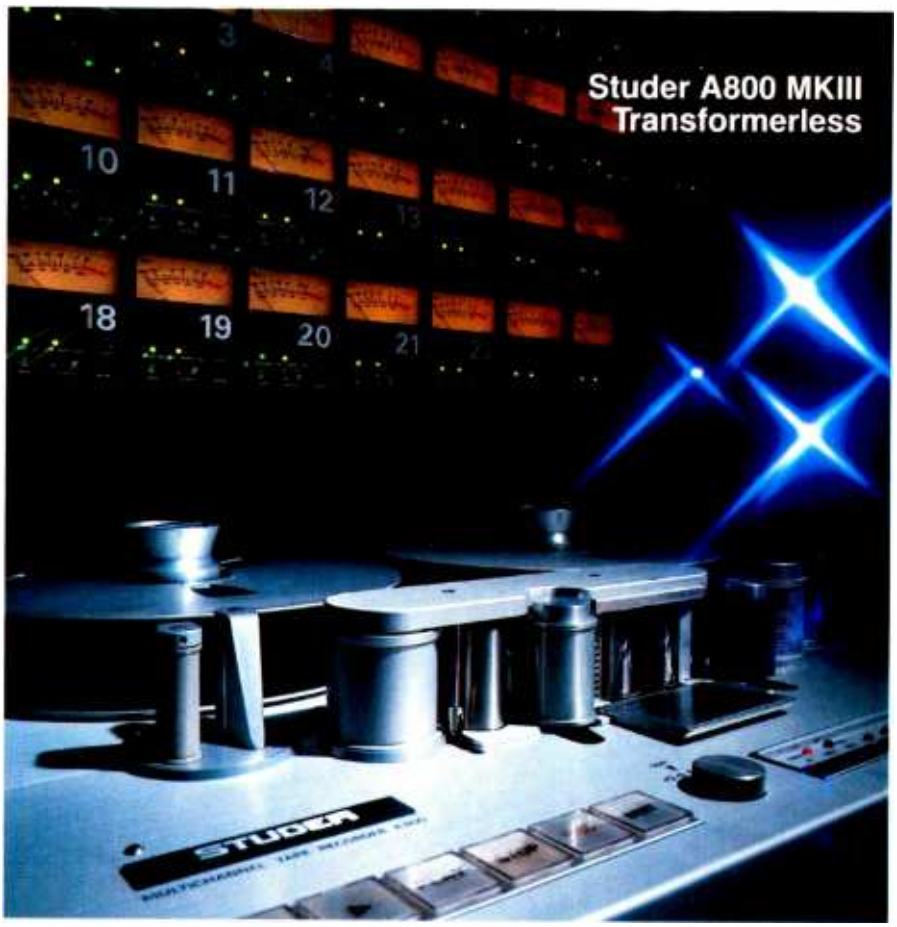

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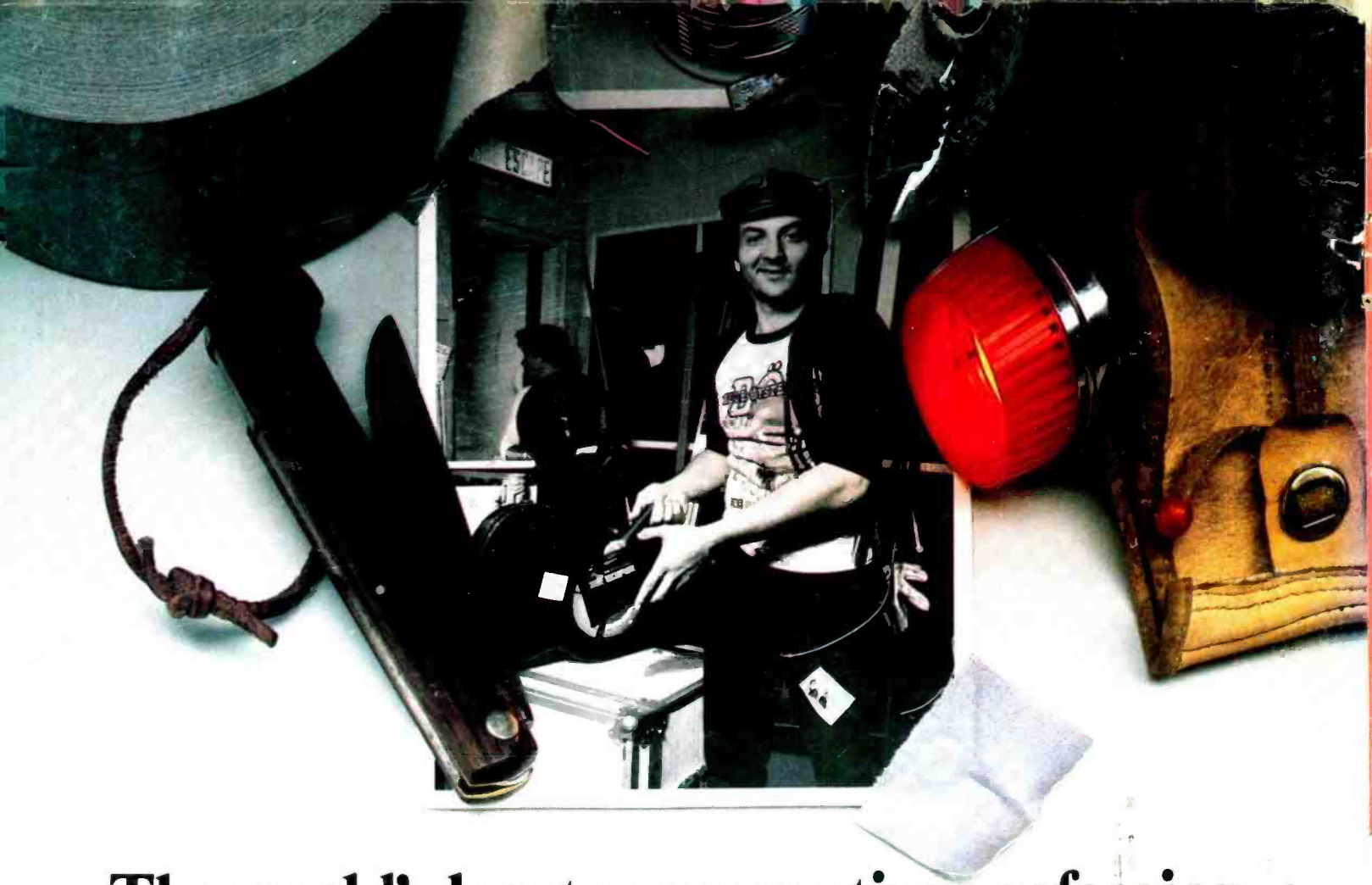

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