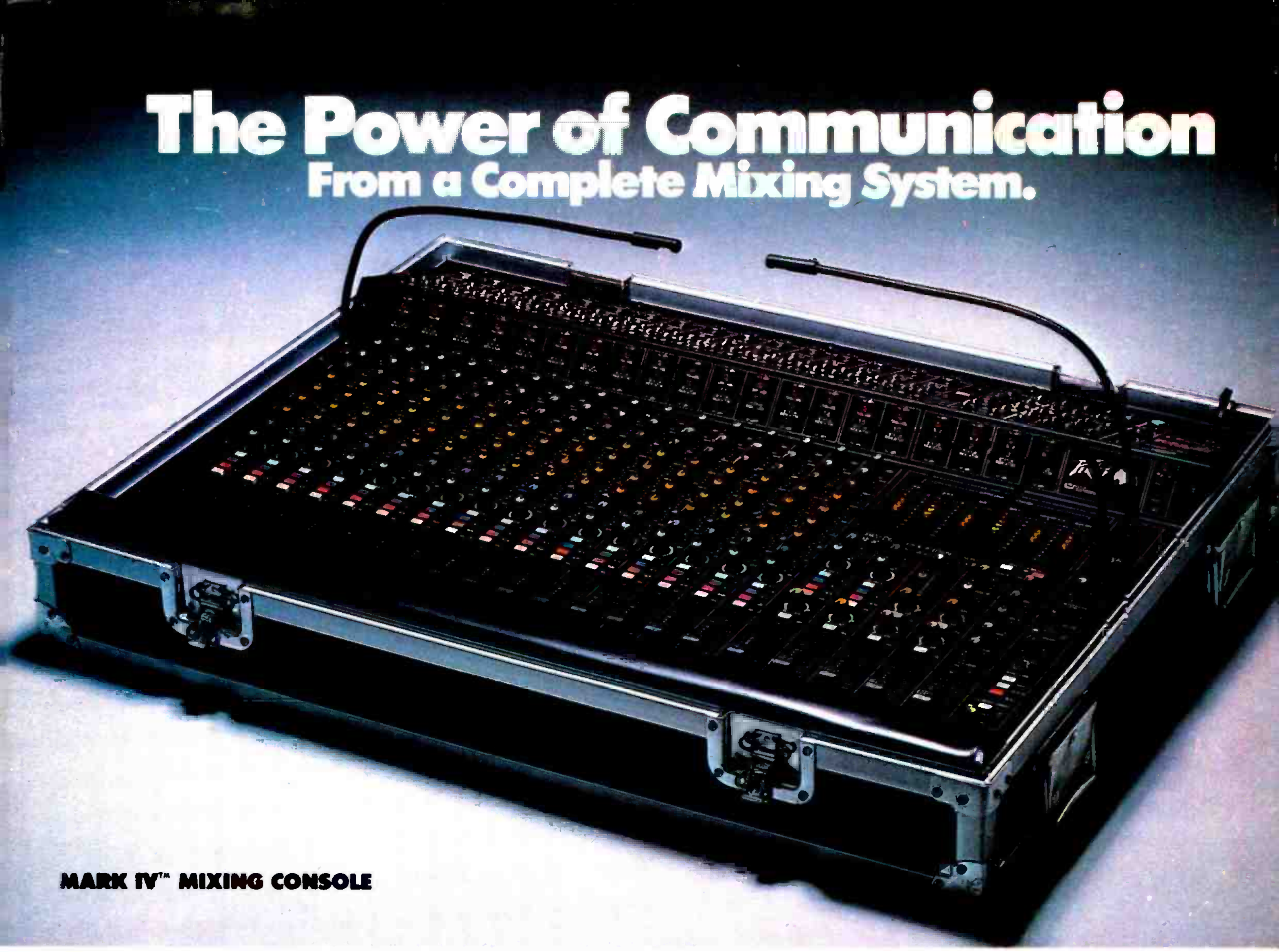
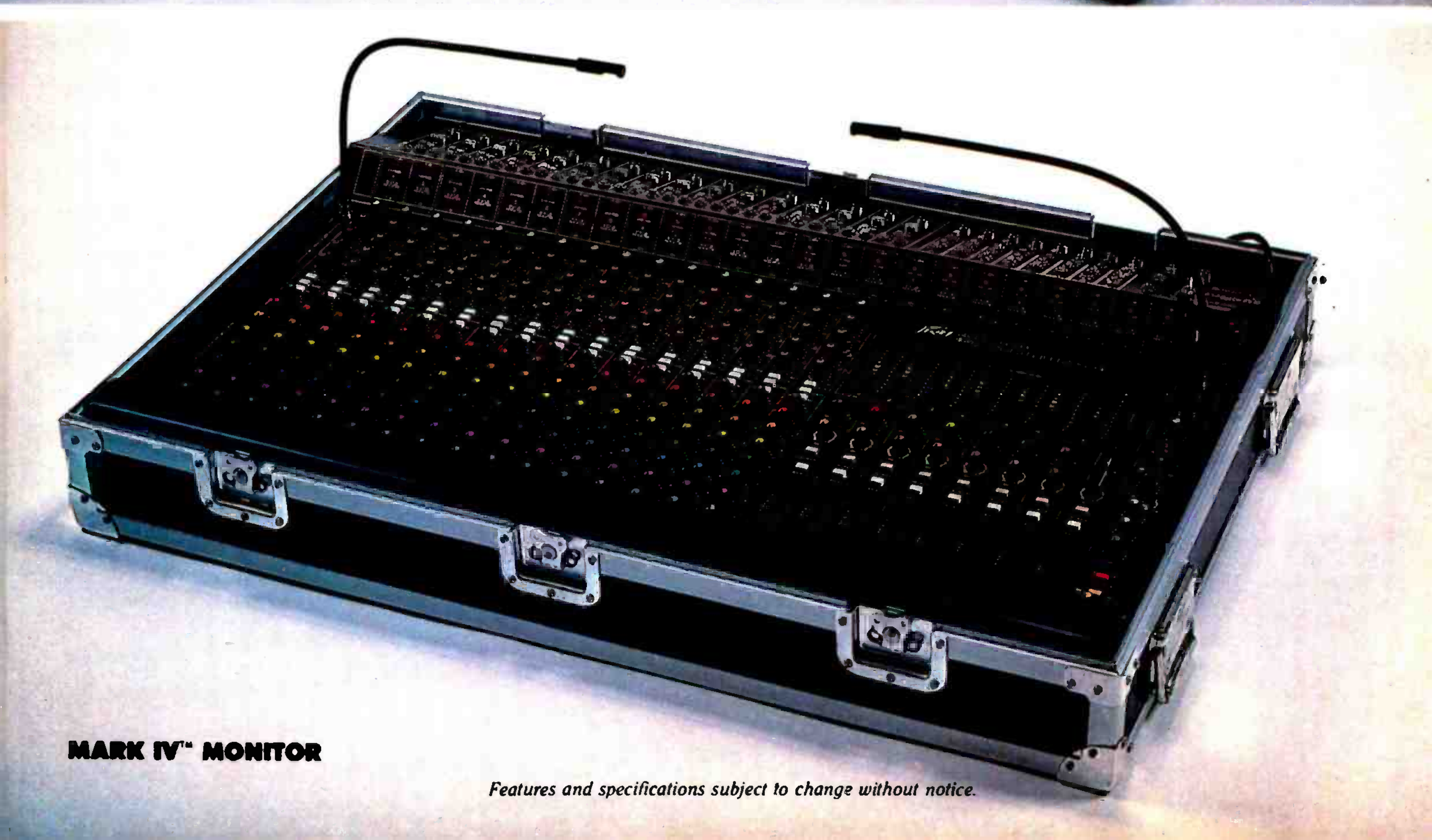

RECORDING
ENGINEER/PRODUCER
Companded Predictive Delta Modulation
BREAKING THE DIGITAL PRICE BARRIER
— See Page 150
$3.00
October 1982
Volume 13 — Number 5
PRODUCING AUDIO FOR • TAPE • RECORDS • FILM • LIVE PERFORMANCE • VIDEO & BROADCAST


TRICI VENOLA

The Power of Communication
From a Complete Mixing System.



MARK IV™ MIXING CONSOLE



MARK IV™ MONITOR

Features and specifications subject to change without notice.

It is the purpose of any musical performance, live or recorded, to successfully communicate with the listener. To attain that goal is often a challenge — even for the most experienced musicians, sound personnel, and stage crew. At Peavey we realize the criteria to be met before this goal can be obtained.

MARK IV™ MONITOR MIXER

First, the musician must be satisfied with the blend and balance of the on-stage monitor mix. In most concert type situations, the musicians may demand anywhere from two to six separate monitor mixes. Our new Mark IV™ Monitor Mixer can supply this need with up to eight individual monitor mixes.

The Mark IV™ Monitor Mixer is available in 16 x 8 or 24 x 8 configurations and features transformer balanced inputs and outputs, 8 unbalanced outputs, PFL/Solo headphone system, 10-segment LED ladder displays for each of the 8 outputs, auxiliary inputs and low-cut controls for each mix and a unique PFL/Solo patch. The PFL/Solo patch is a highly desirable feature that enables the monitor engineer to patch any of the mixes back into the switched inputs so that externally equalized or processed signals can be monitored. This is a feature which is not usually found on custom-made monitor mixing systems costing $15,000 or more.

Each channel of the Mark IV™ Monitor Mixer features LED status indication of -10 dBV and +10 dBV, an input gain control, 4-band equalization, built-in mic splitter, phase reversal switch, PFL and mute switches, and 8 color-coded rotary level controls which correspond to color-coded slider level controls in the output section.

To make the most out of the Mark IV™ Monitor Mixer's capabilities, we have equipped the mixer with two separate built-in communication systems. By utilizing our optional headset or "gooseneck microphone," the monitor mix engineer can communicate with the musicians through any of the 8 separate monitor mixers. This talkback system will help alleviate the problems musicians sometimes have in establishing the proper on-stage mix, especially if a previous sound check was not possible.

A second communication link can also be established by the monitor mix engineer between the stage crew and lighting personnel by utilizing the optional Talk/Comm "slave" units. The Mark IV™ Monitor Mixer's front panel utilizes an LED indicator to alert the engineer as a call function and also shows when intercom is active.

MARK IV™ MIXING CONSOLE

Next, the house (main) system must be able to deliver crystal clear, noisefree sound reproduction to the associated equalizers, power amps and horn/loudspeaker enclosures. For the main PA, our new Mark IV™ Professional Mixing Consoles offer the sound engineer the necessary performance, flexibility and functions to do almost any sound job.

The Mark IV™ Professional Mixing Consoles are available in 16 or 24 channel versions (16/24 x 4 x 1) and feature transformer balanced inputs and outputs, PFL headphone system, 10-segment LED ladder display for all outputs, channel and sub output LED indication (-10 dBV and +10 dBV), internal reverb and effects/reverb return to the monitors. The console also utilizes a 24 volt phantom power supply, variable low-cut controls on each sub (20 Hz to 500 Hz), and in-line patching facilities between the sub outputs and the sum.

Each channel of the Mark IV™ mixing console features an input gain control, two pre-monitor sends, 4-band equalization, effects/reverb send control, pan control, "push/push" channel assignment switches, pre and post EQ, send/reverb patching and PFL (pre-fade listen) switch.

The Mark IV™ Professional Mixing Console has two complimentary communication systems for use with our Mark IV™ Monitor Mixers, headsets, gooseneck microphone and Talk/Comm "slave" units. The Mark IV™ Series intercom system allows communication between the "house" and monitor mix engineers as well as stage, lighting and other associated concert personnel.

Both the Mark IV™ Monitor Mixer and the Mark IV™ Professional Mixing Console feature gooseneck lamp connectors (BNC) with dimmer controls for use with our optional gooseneck lamps. This option allows superb visibility of the mixers in poor lighting situations.

The Mark IV™ Series Monitor Mixers and Professional Mixing Consoles are the successful result of our extensive research and development efforts as well as constant "monitoring" of the needs of professional sound reinforcement companies and soundmen. This outstanding series of mixers represents, we believe, truly exceptional and professional products that will outperform competitive products retailing for many times the price.

For complete information on the Mark IV™ Series write to: Peavey Electronics Corp., P.O. Box 2898, Meridian, MS 39301.

PEAVEY ELECTRONICS CORP.
711 A Street Meridian, MS 39301

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For additional information circle #2

RECORDING ENGINEER/PRODUCER

— the magazine to exclusively serve the **RECORDING STUDIO** and **CONCERT SOUND** industries . . . those whose work involves the **engineering** and **production** of commercially marketable product for:

- **Records** and **Tape**
- **Film**
- **Live Performance**
- **Video** and **Broadcast**

— the magazine produced to relate recording **ART** . . . to recording **SCIENCE** . . . to recording **EQUIPMENT.**

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— The Cover —

"Breaking the digital price barrier," another original illustration by Trici Venola. Two articles describing the new dbx Model 700 Digital Audio Processor begin on page 150


First, again.
The pioneers of 24 track recording make the best even better.
Less than ten years ago, MCI introduced a radical new concept that made other multi-track recorders obsolete. The design was based on a totally servo controlled transport, all new and all D.C. And it made the pioneers of 24-track recording the most imitated designers of professional tape recorders in the industry.
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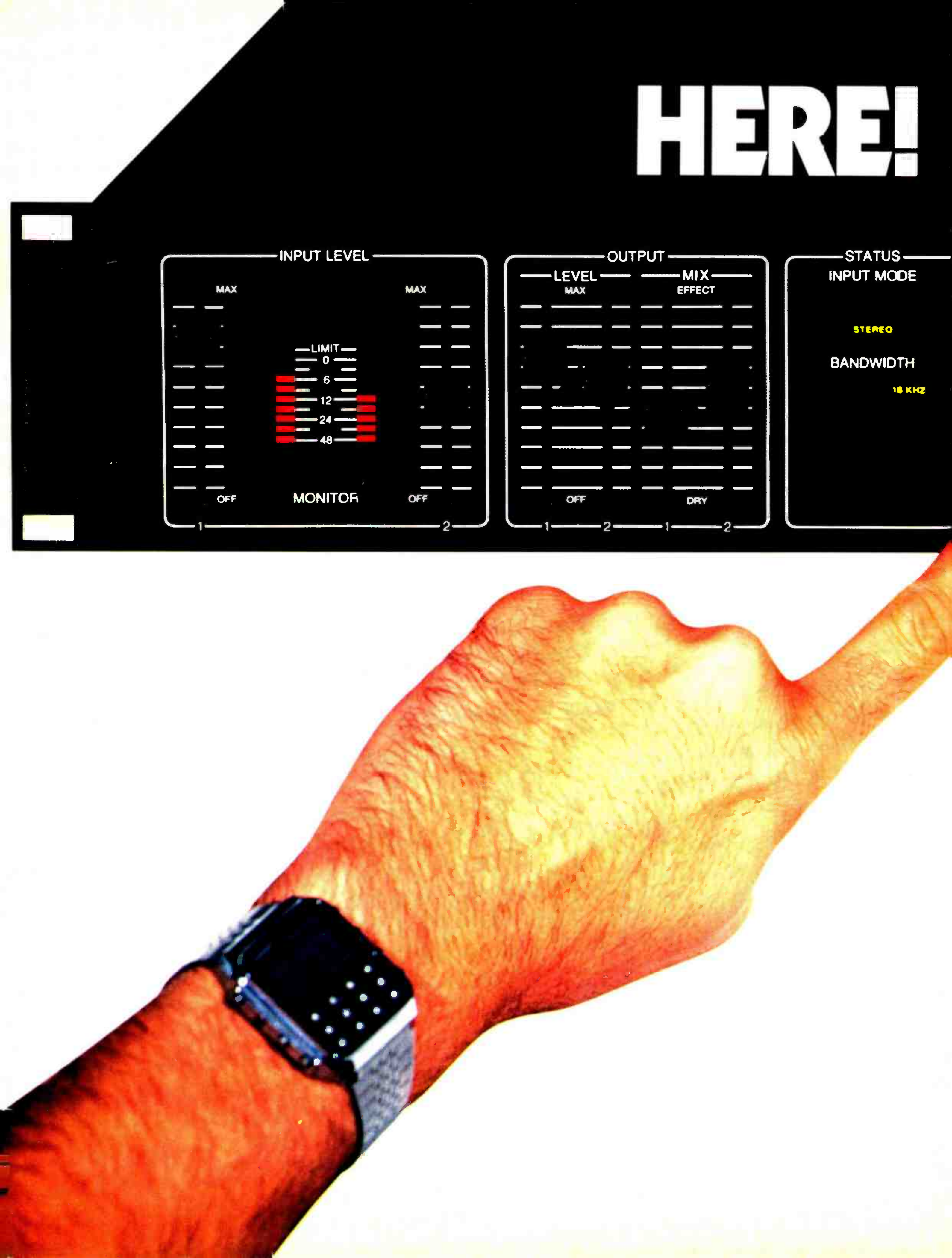
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18 KHZ

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DISK-BASED AUTOMATION

From: **Sid Price, Director**
Melkuist Ltd., England

We write in connection with an advertisment in the August '82 issue of *R-e/p.* The claims made by Sound Workshop for their DISKMIX System seem to be more than a little misleading. In particular, the claim that the system is the *first* of its kind to suit most mixing consoles is plainly untrue, since Melkuist launched their GT800 system some time ago and, in fact, now has more than 20 installations on consoles of many types, including Neve, Harrison, Trident, and API.

We are most surprised that this error was not picked up by yourselves, as the rest of the magazine is notably independent in its reporting and publication of manufacturers' press releases, *viz* our Event Selector press release in the same issue. We trust the lapse is temporary, and that you will continue to produce a vehicle for ourselves and others to reach our markets fairly.

Reply from: **Michael Tapes**
President, Sound Workshop Professional Audio Products, Inc.

I deeply regret that Mr. Price found objection to our ad appearing in the August issue. While his statements regarding the Melkuist GT800 system are accurate, so too is the advertisement in question. The ad states that DISKMIX is "... the first disc-based *editing and storage* system adaptable to virtually all *automated* consoles . . ." (emphasis mine). The critical point is that DISKMIX is not an automation system; it is an automation storage/editing system. As such, DISKMIX interfaces with consoles that are, in fact, already automated with systems by MCI, Sound Workshop, and Allison/Valley People (which represent virtually all automated consoles in use today).

The Melkuist GT800 is a complete automation system which is designed to be retro-fit into *non*-automated consoles. While Sound Workshop also offers a complete automation system (consisting of a VCA fader retro-fit package along with ARMS Automation and DISKMIX) which is in direct competition with the Melkuist, DISKMIX is not. It is being marketed as a stand-alone product to owners of *automated* consoles.

I hope that in outlining the distinctions between Melkuist and DISKMIX, potential customers will have a clearer understanding of which products(s) might be applicable to their needs.

LOSSES IN DIGITAL STANDARDS CONVERSION

from: Dr. R. Lagadec
Product Manager, Audio PCM
Willi Studer

The well-informed article by Martin Polon on the Digital Audio Disk [*R-e/p Augist issue, page 6 — Ed.*] reports an intriguing, and certainly misleading, statement by "experts at Sony" on the loss of signal-to-noise due to standards conversion (or, to give it its correct name, sampling-frequency conversion). According to this statement, the "experts" doubt that sampling frequency conversion can be done with less than 6 and perhaps 8 dB of loss. The article also mentions how there could be signal-to-noise loss approaching 8 dB in real terms for *each* standards change.

A quick remark on this, first. Standards conversion introduces some amount of noise, and thus reduces signal-to-noise ratio. If (see below) the noise component is such that a 3 dB loss in signal-to-noise occurs, 10 conversions will not mean a whopping loss of 30 dB (noise is *added* at every conversion), but rather a more modest 10 dB loss. If one conversion were to cost 8 dB in signal-to-noise (and this would mean lousy, non-state-of-the-art conversion), 10 of them would bring the signal-to-noise down by 18 dB, i.e. from 96 dB (the theoretical value, if and only if the a-to-d's are ideal) down to 78 dB. *Not* down to 16 dB as might be expected if 8 dB were lost at each standards change.

Purely digital sampling frequency conversion between arbitrary ratios (such as the awful 44.1 to 48 kHz one) was reported by Studer at a number of AES conventions, and at the recent AES Conference on Digital Audio in Rye, New York. "Experts" should know about it. The presentations (and the easily available preprints) quoted an increase in signal-to-noise below 5 dB, worst-case ("worst-case" means a 20 kHz digital sinewave at clipping level, a signal utterly without redeeming musical value). With less pathological signals, the system is much closer to the theoretical loss of 3.0 dB, which, incidentally, means that the sampling-frequency converter *adds* a noise component 97.8 dB below clipping level. No digital-audio a-to-d is that good, by 7 dB at the very least. State-of-the-art a-to-d's mean that many state-of-the-art standards changes could be made in cascade before a listener or a measuring instrument would begin to notice, as standards-change noise would be buried in a-to-d noise.

In-between, the SFC-16 (a commercially available digital sampling fre-

quency converter from Studer) has been introduced. At full amplitude, the signal-to-noise ratio is specified as 96 dB or better.

Digital sampling frequency converters tailored to the needs of today's digital audio have been invented, designed, built, measured, and tested in studios. The "experts" mentioned in the article have not invented them, not built them, made no measurements and no tests. Inherently linear-phase, operating very near the theoretical limit of 16-bit systems (and beyond, as our own experience shows), standards converters can translate from (for example) 44.1 kHz to the professional sampling frequency of 48 kHz with a quality far in excess of that of today's a-to-d and d-to-a converters.

Imprecise, uninformed and erroneous, the statement on standards conversion will hardly reflect the views of the experts at Sony headquarters.

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db Recording Studio

In an article describing the modular design technique for db Recording Studios, Florida (August 1982 issue, page 38) we ommited to include a photo credit for the color illustrations. Our apologies to photographer Edward Slater, who was responsible for all the pictures, and also the cover shot.

news

CEO DECLARES A VICTORY OVER SALES TAX ISSUE; BUT WARNS THAT THE BATTLE IS NOT YET OVER FOR ENGINEERS

Due to the unceasing efforts of the California Entertainment Organization (CEO) over the past nine months, on September 10 California Governor Edmund G. Brown, Jr. signed into law AB 2871, the bill that rescinds the retroactive sales tax on all master recording productions. Sponsored by Assemblywoman Gwen Moore, AB 2871 overwhelmingly passed both the California Assembly and State Senate with votes of 53/17 and 40/3, respectively; Marz Garcia had carried the bill in the Senate.

The passage of AB 2871 brings to a close the first chapter of the battle between the CEO and the State Board of Equalization. Due to an interpretation of the law passed by the legislature in 1975, the SBE has been assessing and collecting a retroactive 6% (6½% as of July 1, 1982) sales tax on all "fabricating" costs, or expenses incurred, or personal services rendered in the production of a master recording.

"They taxed us for money we had to spend on studio costs, AFTRA scale, hotels, rental cars, and take-out food," says CEO president David Rubinson. "And, on top of the tax, they tagged a 10% penalty for failure to file, and 20% per month charge."

The SBE made this interpretation depite the law's specific exemption of "copyrightable, artistic or intangible material," Rubinson added. Possibly the most important aspect of AB 2871, which comes into effect January 1, 1983, is that it is not a *new* law, but declaratory of the *current* law.

The pertinent portions of the new bill now read as follows:

Section 1. *Section 6362.5 of the Revenue and Taxation Code is amended to read:*

2) "Amounts paid for the furnishing of the tangible elements" shall not include any amounts paid for the copyrightable, artistic or intangible elements of such master tapes or master records, whether designated as royalties or otherwise; (including, but not limited to, services rendered in producing, fabricating, processing, or imprinting tangible personal property or any other services or production expenses in connection therewith which may otherwise be construed as constituting "sale" under Section 6006.)

Section 3. *The legislature finds and declares that Section 1 of this act is declaratory of, and not in change in, existing law. It is the intent of the Legislature in enacting this act to clarify the existing law and to affect all applicable existing proceedings.*

At a September 25 CEO meeting at the Los Angeles Record Plant, Rubinson outlined the current situation with the State Board of Equalization. "Although we are now at the 10-yard line," he offered, "the CEO still has to make sure that the SBE understands the law, and administers it the way it was written."

Having explained to the meeting that $125,000 already had been spent by the CEO to ensure smooth passage through the legislature — of which $80,000 was donated by the RIAA — the Organization still needs to collect additional monies so that the SBE can be "persuaded" to interpret the new bill in the "spirit in which it was written." The next step, he says, is to work closely with the SBE in the drafting of regulations that will be used by tax officials in actually implementing the new bill.

"We must make peace with the SBE," Rubinson emphasized. "We *have* to make friends with them and write the regulations *together*."

Turning to the question of retroactive taxation going back to 1976, Rubinson offered that the situation was still far from being resolved, but that it was essential for the CEO to act togther in the matter, and establish a correct procedure for claiming refunds. Otherwise, he felt, a precedent might be set in an early case that would affect subsequent appeals. There is also a three-year statute of limitations on tax refunds of this type for those who have been assessed for sales tax on recording services since 1976; the CEO intends to discover whether this period can be extended.

. . . News continues on page 177 —

—AUDIO/VIDEO—
RECORDING

AUDIO VIDEO PERSPECTIVES

PROSPECTS LOOKING GOOD FOR PRO AUDIO/VIDEO IN 1983

by Martin Polon

The fall of 1982 marks the first Audio Engineering Society show since November '81. The past year has been eventful for audio, and for the growing relationship of audio with other technologies, especially video. During the year we have watched the record business sustain grievious losses as record sales plummeted from the highs of the end of the Seventies. The last 12 months haven't been too good for the consumer audio business either, as audio component sales stabilize at a level far below that of the previous three years. Videocassette sales have held pace with previous growth figures, spurred perhaps by fears that copyright restrictions would choke off the sale of recorders. Videodisk has had a sickly birth, with only the RCA SelectaVision system showing any staying power. The studio recording sector, sound reinforcement, TV and radio broadcasting have all mirrored the effects of both a stagnant economy in the middle of a depression, and a lack of consumer interest in audio.

The comparison with forms of electronic entertainment is not flattering to audio on any level. Motion picture attendance is still bigger over the last 12 months than at any other time in its history. Cable television is growing towards a 40% saturation of American homes. Computing, both as electronic games and full-scale computers, has already reached 10% of US homes, and the figure could well climb to 15% by the beginning of 1983. The problem faced by audio is not just the economy, because the sale of VCRs, movies, games and computers appear to have managed to sidestep the recession.

The attitude held by the public towards the quality and quantity of audio in the home does affect the entire marketplace for audio software, and the hardware necessary to produce that software. This last year has finally offered a light at the end of the proverbial tunnel for the audio business, from consumer through semi-pro to the professional audio marketplace.

First and foremost, there has been grudging agreement amongst the various manufacturers on a standard for

AUDIO/VIDEO PERSPECTIVES

professional digital audio recording. A similar standard now exists for consumer digital recording and the Compact Audio Disk, though at a lower bit rate and sampling frequency. Digital consoles are being built and installed, at the BBC in London and in Marin county for Lucasfilm. Standards converters have been successful in connecting units operating on different standards, thereby making digital accessibility, if not compatibility, a reality.

Stereo television has had a year under its PAL belt in West Germany, and nearly five years of use in Japan. An EIA committee has been working over this last year to make a recommendation on a system for use in the United States. The FCC has finally allowed AM Stereo to reach the marketplace, albeit with a myriad of systems.

As much as these and other happenings of the last 12 months can be analyzed and lamented in these pages, the fact remains that progress has been made. The ground rules have been established, and the inexorable pressures of technology are going to push consumer audio and video closer together, and hence create new demands for studio time and studio hardware to supply entertainment for these enhanced home audio markets.

1983 could bring a resurgence in high-quality audio production and listening. Digital audio is moving forward, and the end of 1982 should see the enthusiastic introduction of Digital Audio Disk in the Japanese home marketplace. Initially, 11 suppliers are scheduled to release digital players in the Compact Disk format developed jointly by Sony and Philips: Denon, Hitachi, Marantz, Matsushita, Mitsubishi, Onkyo, Pioneer, Sanyo, Sony, Trio-Kenwood, and Toshiba. These manufacturers are to offer 12,000 units per month to the Japanese consumer market (these figures are for product expected to be shipped during the Fall of 1982). By January 1983, Sony alone expects to make 15,000 units per month. The players will sell in Japan at prices ranging from $646 for a Sony CDP-101, to $923 for the Trio-Kenwood player with 99 selection programability.

The digital disks will be priced in the $13 to $15 range, with CBS-Sony poised to release upwards of 100 titles by the end of 1982. Other releases from Polygram (Philips), Denon and Toshiba-EMI will boost the available catalog of selections to over 150 titles. Exposure on the Japanese market is expected to be followed by release in Europe and the United States, although Sony-partner Philips remains silent about the actual release timetable for Europe. In the United States, units are expected to be available for the second quarter of 1983, with a price tag of $500 per player as production increases, and the learning curve of manufacturing digital players yields cost economies.

Digital audio also will surface in consumer tape recorders during 1983. Hitachi, Mitsubishi, Sansui, Sony and Tecnics all manufacture digital audio units capable of recording and playback. Using the 44.056 kHz standard these units provide the home recordist with virtual studio-quality recording and playback. Although they cost several thousand dollars, prices are expected to drop during '83 as users see the potential advantages of home digital recording. The Hitachi and Technics units are digital audio cassette recorders/players, while the machines from Mitsubishi, Sansui and Sony are processors designed to be used with half-inch VHS and Beta portable VCRs. All of the units be offered in the US during 1983.

1983 also will see the EIA Committee's stereo TV report delivered to the Federal Communications Commission. It is expected that the FCC again will bow to the marketplace for implementation of the decision, and it is possible, though unlikely, that a stereo television system could be implemented in the United States by the end of next year. The market is there, what with worldwide sales of stereo VCRs and stereo prerecorded video cassettes on the upswing, and expected to increase in '83.

Videodisk is also out of its 1982 doldrums, and sales of the RCA system —

SUCCESS BY DESIGN.

The success of Soundtrek's Studio I was not an accident — it was planned, designed, constructed and equipped to be a success.

Working together, Flanner's Pro-Audio and Soundtrek created the reality of the 24-track Studio I from a dream and an idea. Sound quality, versatility, dependability and cost efficiency were the main concerns of Ron Ubel of Soundtrek. With the aid of an experienced designer and a skilled contractor, the staff of Flanner's Pro-Audio achieved these goals.

The most critical choice in equipment was the selection of the master recording console. Based on sonic excellence and professional functions, a NEOTEK Series III 28 x 24 custom console was chosen as providing unequalled value regardless of price. For similar reasons an Otari MTR-90 MkII 24-track Masterecorder was chosen, as well as support equipment from such respected manufacturers as U.R.E.I., McIntosh, Crown and many others.

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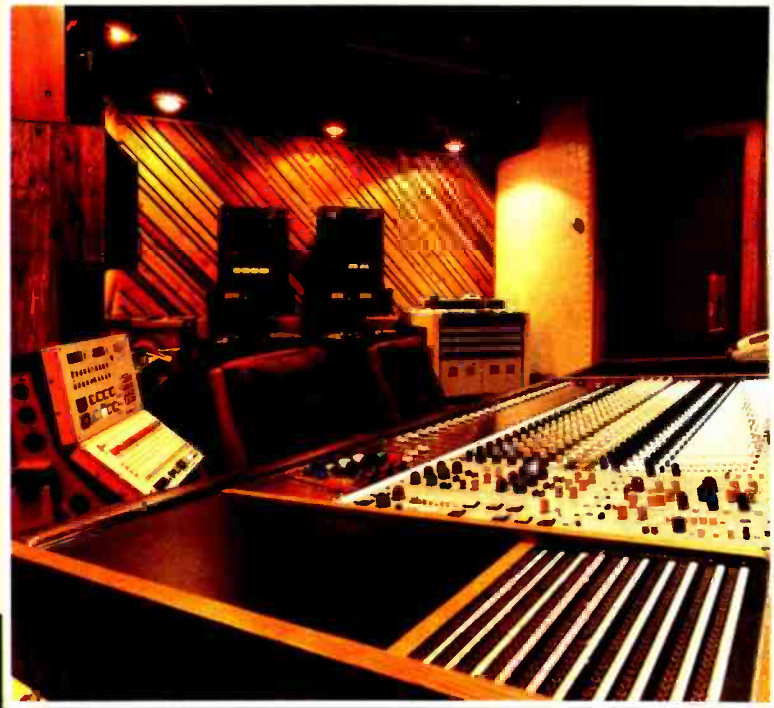

Studio 1

Construction By: D.A. Peterson, Contractors, Omaha, Nebraska.
Design By: Steven Durr and Associates, Nashville, Tennessee.
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AUDIO/VIDEO PERSPECTIVES

now with stereo capability — have progressed from 160,000 units for 1981, to 365,000 units this year, and in excess of a projected 500,000 machines for 1983. What is so exciting about these numbers is that new owners typically buy over 25 titles in the first year of ownership. The Laserdisc is still in the marketplace via Magnavox and Pioneer, who is also producing all of the videodisks. 3M is expected to join the ranks of Laserdisc pressers, and the 1983 outlook for both systems seems good, but with RCA moving at a faster clip.

The interesting phenomenon of an increase in RCA sales seems to be directly related to the concept of the system acquiring stereo capacity. Several consumer studies echo this, with stereo video being given a high priority by most buyers.

It seems certain that 1983 will be the year when all of the planning for direct-broadcast satellite services are finalized here by Comsat, and in the UK by the BBC (oddly enough the custodian of pay services via satellite in England). It seems certain that each service will provide a minimum of two audio channels, and perhaps as many as four, with all of the audio digitized. The provision of direct-broadcast video with stereo could well be a selling point for entry into those US homes not now connected to stereo pay cable.

The final miracle of 1983 will be the emergence of a digital video tape recorder for the broadcast marketplace; that such a machine exists as a working prototype seems assured. The recent adoption of a digital video control protocol by SMPTE makes the prospect of digital video recording that much more likely. The elimination of quality lost in duplication on analog video machines in a strong incentive to both users and manufacturers. The audio implications of such a development are significant. A digital VTR would provide a proven transport and electronic technology upon which to base development of digital audio studio recorders. The presence of four audio channels on a video recording system is guaranteed by current broadcast industry practice, and could provide easier access to tracks for sweetening. Whatever the ultimate disposition of the relationship between digital video and audio (and that relationship is particularly close on the digital disk), the advent of digital video will be an event of value to the entire audio/video industry.

The reality of 1983 is a year in which audio will again be at the forefront of consumer interest, propelled by digital audio systems, stereo video (and perhaps television), AM Stereo and, hopefully, by economic recovery. With this kind of movement at the consumer level, and with a whole new range of high-quality items applicable to broadcasting and semi-pro audio (Compact Disk players, and digital cassette recorders), the studio industry could well find a surge of new release work, sweetening and recording to fill studio time profitably. ■■■

views

THE TRANSITION FROM MUSICIAN TO PRODUCER

RONNIE MONTROSE on the Special Demands of Working Both Sides of the Studio Glass and Moving Into Production

by David Gans

It shouldn't come as much of a surprise that Ronnie Montrose was responsible for producing the first hit record by his band, Gamma. In the past dozen or so years the guitarist has worked with and observed the methods of Ted Templeman, Bill Szymczyk, Ken Scott, Gary Lyons, Edgar Winter, Jack Douglas, and others — and his involvement in the production of his records has grown steadily.

Ken Scott produced the first *Gamma* album, and Montrose co-produced *Gamma 2* with Gary Lyons. When it came time to choose a producer for *Gamma 3*, Montrose says "I asked myself: 'Am I going to go through this again?' It has always seemed to be a compromise, because I have a set way of doing things. If I'd had a really incredible success rate with someone I'd worked with, then my tendency would have been to stay with him.

"I realized that I was capable of doing a lot of the things I wanted to get done, and that I was sort of standing in my own way by thinking it would be easier with someone else producing. It's *not* easier, for me, so I decided to produce the record myself." Ronnie had produced the third Warner Brothers album by Montrose, "but I'm about 10 spirals up the circular pattern now," he concedes. "I have a lot more experience; a lot more studio expertise; and a lot more objectivity."

The first records Ronnie Montrose was involved with were Van Morrison's *Tupelo Honey* and *St. Dominic's Preview*, produced by Morrison and Ted Templeman. "We worked together musically, but I had absolutely no technical knowledge at that point," Montrose recalls. "I was so overwhelmed by the gift Van has of being able to put mean-

ing and depth of feeling into a simple melody, that I just wasn't even interested in the rest. And as far as Van was concerned, I was just a young guitar player looking to blow some fast licks."

Following his tenure with Morrison was a year with the Edgar Winter Group, during which Montrose played guitar on *They Only Come Out at Night*. "I was completely removed from the process," Montrose recalls. "Rick Derringer was producing and Bill Szymczyk was engineering; I literally came in, played my parts, and left — I was for all intents and purposes a session guitar player. This was 1972, and I didn't even know what the board was about. The only thing I insisted on was moving my amplifier into a stairwell, and miking it from there because the studio was too dead." But the album went Gold and spawned two hit singles, "Free Ride" and "Frankenstein," the latter a Number 1 hit and a Gold record.

When Montrose formed his own band with vocalist Sammy Hagar, drummer Denny Carmassi and bassist Bill Church, he remembered how much he'd enjoyed working with Templeman. The band and Templeman co-produced Montrose's self-titled debut album and followup, *Paper Money*.

"I was really shy about working with Ted and Donn [Landee, who engineered

both records], because I hadn't been in the studio much. I thought, 'These guys are LA studio pros — they know everything about the studio, and I know nothing.' "

During the making of the first two Montrose albums, Montrose observed the way Templeman and Landee operated and began to involve himself in the recording process. "Ted thinks in terms of frequency layering," he says. "You divide up the frequency spectrum and know what's going to occupy which sections of it. You know that the kick drum and the bass are going to occupy certain areas of the low frequencies, and that a rhythm instrument will be, say, from 800 to 3,000 Hz; if I know that there's going to be a lead instrument with a lot of bite at around 2 kHz then I'll notch out a spot in the rhythm instrument. The point is to make sure that there isn't a tremendous amount of buildup in one particular part of the audio spectrum, and that things are there — or not there — by choice, and not by happenstance.

"And from Donn," Montrose continues, "I learned the patience to work at it and get it right. When the echo wasn't quite the way it had to be, he'd crawl around the chamber with packing blankets, move the mike around — whatever he had to do, because Amigo [a Burbank, California, studio owned by Warner Brothers] was a totally dead room, and that was all we had to work with." Montrose also notes that the team worked quickly, so staleness was never a factor.

Self-Production — The Pitfalls

Montrose decided to try his hand at producing when it came time to record the third album, *Warner Brothers Presents Montrose*. "I knew I was capable of doing certain things, and in those days there wasn't any worry about having a hit single," he says. "I wanted to try it myself."

He knows now it was a mistake to use an inexperienced engineer on the project. "Neither of us had any firm sense of how to place mikes around a drum kit or bass amp or piano, or how to use room ambience," he recalls. "From the beginning, I knew how I wanted my guitar to sound. I drew an imaginary grid of 1-inch squares and worked in a spiral pattern out from the center, with the mike an inch away from the speaker, and finally found a spot a couple of inches off the center that gave me full


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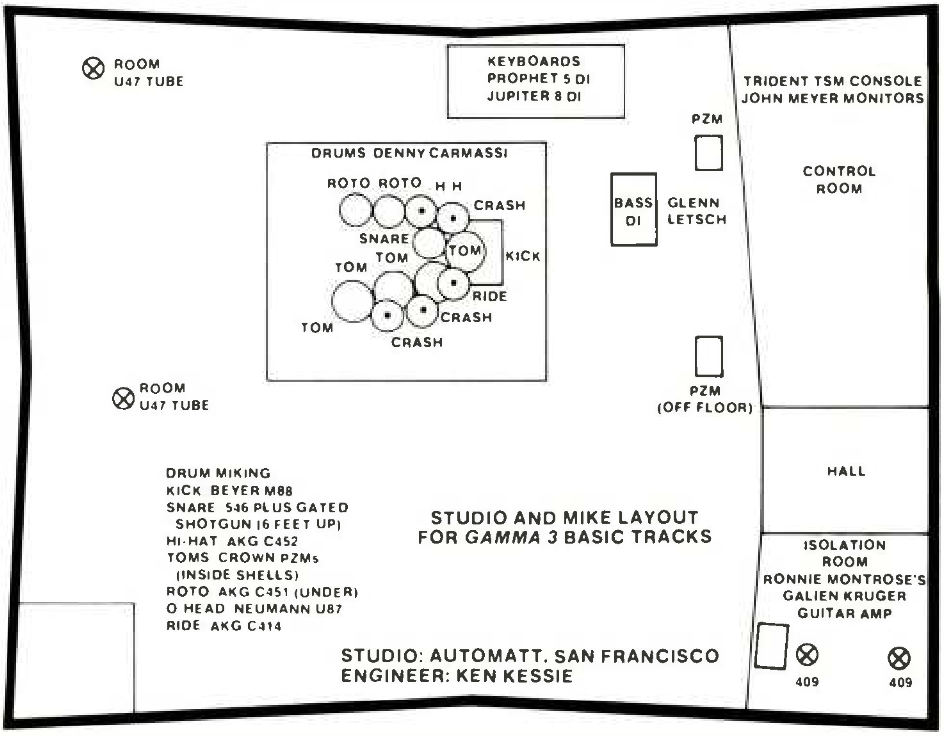
• RONNIE MONTROSE •



frequency response. One inch — especially on a speaker that has quite a midrange beam out the center — can make all the difference in the world as to the way that a microphone picks up what's coming off your speaker, and it varies from speaker to speaker."

Montrose notes also that he failed to acquaint himself adequately with the characteristics of the room and the Westlake speakers on which he mixed the album. "I should have used an A/B speaker setup for comparisons, and I should have listened to albums and things I'd done that I really knew well," he offers. "The Westlakes were so 'bright' that I ended up attenuating all the high frequencies. I did listen on smaller speakers, but I just didn't have the objective ear that I have now."

When *Warner Brothers Presents Montrose* wasn't as successful as everyone had hoped," Ronnie says, "Bill [Graham, the band's manager since just after *Paper Money* was released] suggested that we get a producer who would help us get a hit. The market was starting to become more singles-orientated."

Jack Douglas, who'd had success with Aerosmith and Cheap Trick, was suggested. "He seemed easy to get along with," says Montrose, "and he was a drummer and bass player, so he was able to help with the rhythm arrangements, so I could concentrate on the guitar."

The Learning Curve

By this time, Ronnie understood much more about sound and studios, and wanted to use a more methodical approach to experimentation than did Douglas and his engineer. "They had a kind of haphazard approach that I wasn't comfortable with," says Ronnie. "There is one school of producing that is very cautious, logical and precise — 'get everything on paper' — and think everything out before it's even attempted. Another approach is 'We don't know anything, so let's get all these amplifiers, daisy-chain everything into 'em, put all these mikes up, and lemme just go in and flip some faders around.' And that's just wastin' time as far as I'm concerned."

Montrose found himself closer to the first approach, and Douglas and his engineer closer to the latter. "You're dealing with so many variables that without some scientific way of notating what went on, you can never duplicate a sound," he feels. "You bring up a fader, move a microphone, do this, do that — and four moves down the line you say, 'Something back there was good,' and you've lost it. You're wading through all these variables, and it's very *precarious*."

Nevertheless, Montrose says he prefers the "We don't know anything" approach, "because that's where spontaneity comes from; where wonderful mistakes happen. But I prefer to do at least some calculations and notation to season the approach."

An example of a "wonderful mistake" is the guitar sound on the title track of *Jump On It*. "I described a sound I wanted to get: a kind of 'warbling' sound," Montrose recalls. "Jack said, 'It sounds like you're talking about a bum tape capstan.' He took a piece of tape and wrapped it around the capstan of the 2-track, building up a little wad on one side, and then fed [the recorded

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• RONNIE MONTROSE •

sound] into the board. That gave the guitar *severe* flutter, and it was strictly from improvisation."

Following the breakup of the Montrose band, Ronnie cut a first-ever solo album, *Open Fire*. "There were so many other types of music that I wanted to experiment with," he remembers. "It was vitally necessary that I make that kind of statement at that time." He had planned to work with Ken Scott, he says, "because I liked what he did with The Beatles, and Supertramp's *Crime of the Century*. He gives a lot of separation and presence to the instruments, and even when parts are complex and busy there's an open, uncluttered feeling. It has to do with his use of frequency layering, and echo usage — the spatial distance from full echo to dry." But the timing wasn't right, and Scott wasn't available, for the project.

"I had talked to Edgar [Winter] about writing some songs, and the more I talked to him the more I thought we'd be a good working team," Ronnie says. "The only thing I really needed on *Open Fire* was objectivity, and Edgar knew my playing well enough, and knew what I was talking about. I totally trusted him."

Dick Bogert engineered those sessions. "I call him 'Mr. MOR,' and I mean that in the most respectful way," says Montrose. "He's responsible for all of The Carpenters' hits; he's an excellent orchestra engineer, and he really has a feel for the radio speaker.

"Edgar reinstated the awareness of frequency layering in me. With no vocals, we had a wide spectrum available, and we could bring anything we wanted to the foreground. Edgar would say, 'The kick drum should occupy this area, the cymbals this area, the guitars here.' From low to high, things were as expanded as possible — which was still lacking even as late as *Jump On It*."

Montrose and Ken Scott finally did get to work together when Scott was called upon to produce the first record by Gamma, the band Montrose put together in 1978. "He did The Dregs' *What If*, a phenomenal-sounding record," Ronnie enthuses. "Not only is the music beautiful, but the sounds and the arrangements and the way it's recorded are extremely clean, and to my mind it's a classic album. That and *Crime of the Century* were enough for me."

By this time, Montrose says, "I was approaching each project as a seminar. I knew there were certain things you could learn in the studio with a given talent." He was amazed by Scott's effortlessness in the studio. "It's second nature. For example, Ken might not pay attention to the position of a mike, but he'll instinctively compensate with EQ on the board. Where someone else would go out and make sure the mike is in *exactly* the right position, Ken instinctively understood what he was hearing."

Montrose disagreed with the extremely high listening level Scott favored. "I love listening loud for a little while but pretty soon there's *too much* audio amplitude input and your ears give out — but not Ken's! He's got calluses in his ears," says Montrose with a laugh.

Another interesting aspect of Scott's style, Montrose notes, is that he uses almost nothing but Neumann U-87s. "He doesn't like any other mike," says Montrose. "He'll use 87s on almost everything.

I use a Sennheiser 409 on my guitar. It's a little gold-and-black mike originally intended for use with an echo unit that never got made. The [original] company went out of business, and Sennheiser bought the mikes back and sold them as the 409 — it's not even listed in the catalog.

"Ken wanted to mike my guitar with an 87, and I showed him that at low levels the 409 and the 87 were both fine — but when you turned up the guitar, the 87 softened in the upper-mid range, and the 409 held on." The 409 was used for the recording.

Control Room Chemistry

Ronnie also disagreed with Scott's practice of recording each individual instrument separately. "He starts with the drums, and there's bass in the headphones but never recorded at the same time," he says. "When he proposed this style to me, I said, 'Ken, you've got to be kidding!' He said, 'Listen to the Dregs album.' And it does sound great. So I went along with it, to see what 'Ken Scott's technique' was all about."

Montrose also learned from Scott the technique of recording the drums on a 16-track tape and mixing them on to one or two tracks of the 24 with SMPTE timecode. "That way you're not scrubbing the highs off them every time you run the tape back and forth," Montrose explains. "It's a standard technique, but I hadn't been aware of it before."

Although he says he was not happy with *Gamma 1*, he does not blame Scott's style of recording. He is pleased with what he learned from the experience and does not blame Scott: "That band *needed* spontaneous, live interplay," he says, "so they were the wrong players for Ken's approach to recording."

Gary Lyons and Montrose co-produced *Gamma 2*. "This was another part of my constant, restless search for the right combination of artist and producer," says Montrose. "Gary was diametrically opposed to the Ken Scott way of doing records — he does everything in a live and spontaneous way. I wanted that kind of feeling, to move away from *Gamma 1*, which was really dry.

"Gary was the opposite end of the spectrum from Ken," he adds. "He is of the opinion that things should be spontaneous and sparkling."

. . . continued overleaf —

views

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Lyons, Montrose and Gamma drummer Denny Carmassi (who replaced Skip Gillette prior to the recording of *Gamma 2*) worked very hard on the drum ambience. "Gary really searched for the proper miking positions in the room [Studio A at San Francisco's The Automatt], for which he used [Neumann] 47s. Stereo ambience is a very subtle thing," Montrose says, "and some people say there *is* no stereo ambience. He went back and forth, moving the mikes around and listening to the sound with the EQ flat on the board.

"The miking all around the drum kit was standard, mostly 87s. He didn't use any gates, so there was a lot of leakage and just a big 'wash' of drum sound. He'd really ride the tom and ambience mikes to get a pulsating, live drum sound.

"Gary recorded at extremely hot levels — +6 with the meters buried; there was no audible distortion, but that [Ampex] 456 was completely saturated. There's a lot of tape compression on everything, because he mastered at +6 dB above that. It tightens up the bass and makes everything punchier without losing any top end."

As with Scott, Montrose clashed with Lyons over monitoring levels. It got so bad that at one point Montrose muted the faders, with a white marker and said, "It doesn't go past this level when *I'm* in the room!"

By the time *Gamma 3* rolled around, Montrose knew he had the ability to deliver what was needed. "I was completely comfortable with miking and production techniques," he says, "and I felt that I could be objective enough to do my job as producer. Mitchell [Froom] and I did 95% of the writing, along with the lyricist, before we started.

"We didn't even rehearse; we just went straight into the studio. It's not that we had a huge budget — we worked efficiently. I wanted to be in the studio; in a rehearsal hall, when things get really sparked up, you're not able to do them immediately — especially with the drums and bass. I figured it would actually be more efficient in the long run, and better for the record, to go straight into the studio."

The basic tracks were recorded with Carmassi, Froom and bassist Glenn Letsch, with Montrose only playing guitar when it was necessary to help with the feel. "There's no way you can get a sense of feel when you're playing; no way to monitor the situation," he says. "We kept the drums and bass, and everything else was recut eventually."

Montrose did some experimenting with the drum miking, taking the bottom heads off the toms and suspending Crown PZMs inside. "PZMs are great-sounding, bright mikes, but unless you stick the mike inside the drum, you can't add top-end on the tape because you'll get the snare, cymbals and everything else but the tom-tom. So putting the PZM inside gave us the isolation we needed so we could put all the top-end and midrange we wanted into the toms — and no gates, either." When the PZMs were in use, a pair of U-87s were placed about two feet above the drums, "at a sonic midpoint where I could get the heads of the kit as well as the cymbal ring. Denny hits the cymbals so hard that it masks everything else anyway," says Montrose, "but you still achieve a little natural ambience without having to fabricate it later."

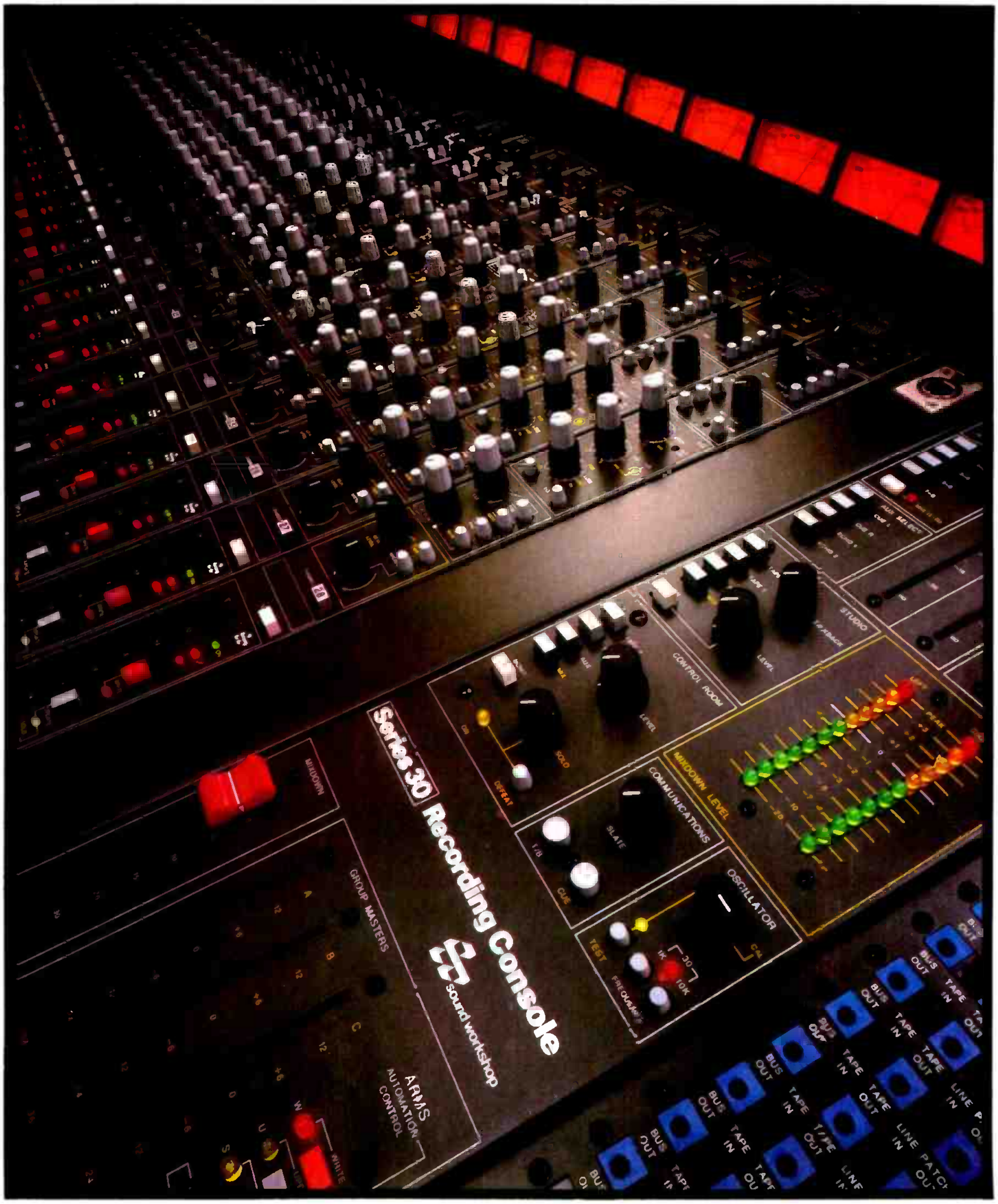
. . . *continued on page 175* —

OMNISOUND/MIKE SHOP

From time to time it becomes necessary to issue a warning to *R-e/p* readers that a company trading in the professional audio marketplace has ceased operation. To this end, it should be noted that from mid-August, 1982, Omnisound, Ltd., and its division, The Mike Shop, of Elmont, New York, went out of business.

This announcement occurs as a matter of record, and to advise readers that still may be responding to advertisements placed in previous issues of *R-e/p*.

For additional information circle #19



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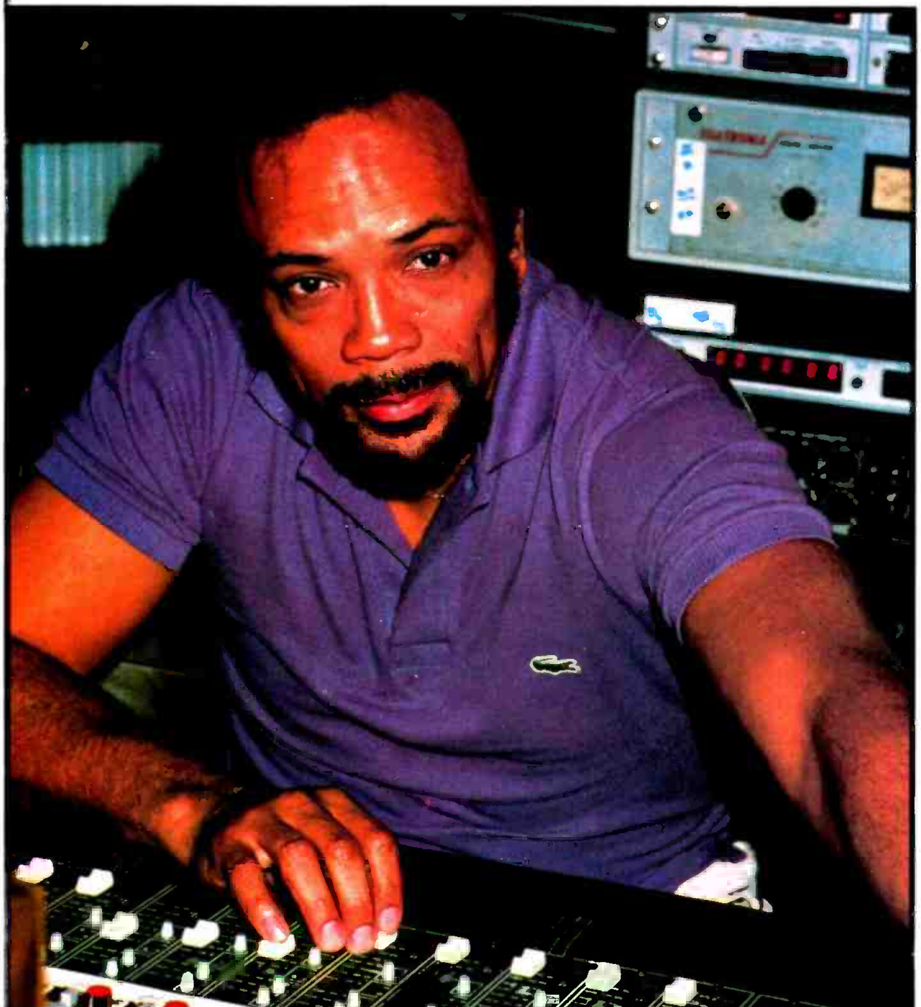
If the word "professionalism" can be epitomized by one of the most successful producers currently working in the recording industry, that man must surely be Quincy Jones. The reports of his humanity, care and response to the needs of his recording "family," and an almost telepathic rapport with his favorite engineer, Bruce Swedien, truly makes Quincy Jones a consummate producer. During the many session hours that *R-e/p* spent with Quincy and Bruce in the studio, it became readily apparent that their complimentary skills — Quincy's proven track record as a musician, composer, arranger, and record producer, married with Bruce's mastery of the recording process — has resulted in a production team whose numerous talents overlap to a remarkable degree. Having worked with Bruce Swedien on so many innovative album sessions, including Michael Jackson's *Off The Wall*, George Benson's *Give Me The Night, The Dude,* and Donna Summer's *Summer of '82*, it came as no surprise to anyone in the industry that Quincy Jones should make such a clean sweep of this year's Grammys, collecting a total of seven awards, including five for *The Dude* alone. The following conversations with this illustrious production team were conducted during tracking dates for Michael Jackson's upcoming album *Thriller,* at Westlake Studios, Los Angeles.



Photography by MARC KORODY

QUINCY JONES

With Engineer Bruce Swedien

THE CONSUMMATE PRODUCTION TEAM

Interviewed by Jimmy Stewart

R-e/p (Jimmy Stewart): How do you first get involved with a particular recording project? For example . . . Michael Jackson.

Quincy Jones: We were working on *The Wiz* together, and Michael started to talk about me producing his album. I started to see Michael's way of working as a human being, and how he deals with creative things; his discipline in a media he had never worked in before. I think that's really the bottom line of all of this. How you really relate to other human beings and build a rapport is also important to me; energy that's a great feeling when it happens between creative people.

I've been in some instances where I have admired an artist's ability, but couldn't get it together with them as a human being. To truly do a *great* job of producing an artist, you must be on the same frequency level. It has to happen before you start to talk about songs.

R-e/p (Jimmy Stewart): Then the important aspect, to your mind, is fostering a faimily feel during a project?

Quincy Jones: Yes. It's a very personal relationship that lets the love come through. Being on the other side of the glass is a very funny position — you're the traffic director of another person's soul. If it's blind faith, there's no end to how high you can reach musically.

R-e/p (Jimmy Stewart): Is the special rapport you establish with an artist based on them saying something unique that triggers off an area in your creative mind?

Quincy Jones: That's the abstract part which is so exciting. I consider that there are two schools of producing. The first necessitates that you totally reinforce the artist's musical aspirations. The other school is akin to being a film director who would like the right to pick the material.

As to what choice of production style I would adopt, your observations and perceptions have to be very keen. You have to be able to crawl into that artist, and feel every side of his personality

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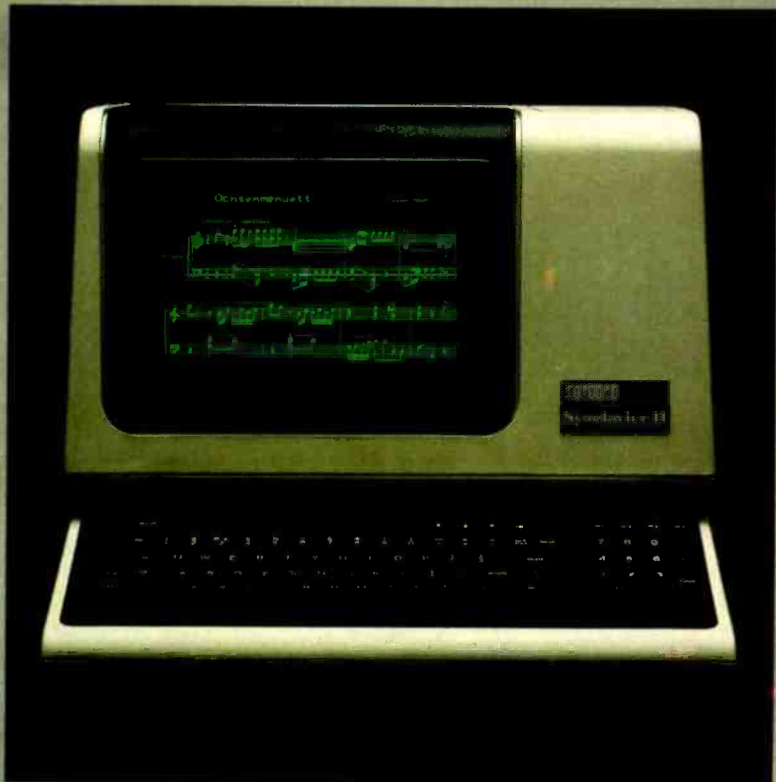

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QUINCY

—to see how many degrees they have to it, and what their limitations are.

R-e/p: Once that working rapport has been established, how do you plan the actual recording project?

QJ: I think you have to dig down really to where you think the holes are in that artist's past career. I'll say to myself, "I've never heard him sing this kind of song, or express that kind of emotion." Once you obtain an abstract concept or direction, it's good to talk about it with the artist to see what his feelings are, and if you're on the right track. In essence, I help the artist discover more of himself.

R-e/p: Do you become involved with the selection of songs for the album?

QJ: On average, I listen to maybe 800 to 900 tapes per album. It takes a lot of energy! I hear songs at a demo stage, and would like to think that the songwriter is open for suggestions. If I say, "We need a C section," or "Why don't we double up this section" the writers with whom I've had the most success must be mature enough and professional enough to say, "Okay, I'm not going to be defensive about any suggestion you make."

R-e/p: So what do you listen for in a song? The lyrics, melody, arrangement, instrumentation . . .

QJ: I listen for something that will make the hair on my right arm rise. *That's* when you get into the mystery of music. It's something that makes both musical and emotional sense at the same time; where melodically it has something that resembles a good melody. Again that's intangible too, because it's in the ears of the listener. So basically I'm saying . . . it transcends analysis. A good tune just does something to me.

R-e/p: Once the songs have been sorted, what runs through your mind prior to the studio session dates?

QJ: I try to get the feeling that I'm going into the studio for the first time everytime. You have to do that because if we started to get to a stage where Bruce [Swedien] and I had a *specific* way of recording it wouldn't work for us. I'm sure some things overlap, because that's part of our personality, but we try to approach it like everytime is the first time; we're going to try something so that we don't get into routine type of procedures.

With Rufus and Chaka Kan I'll do one kind of a thing, where we will have rehearsals at their home, and talk about things. Maybe even come in the session with everybody and do it like "Polaroids." That way you can hear what everything sounds like rough, and feel what the density, structure and contour of the song is all about.

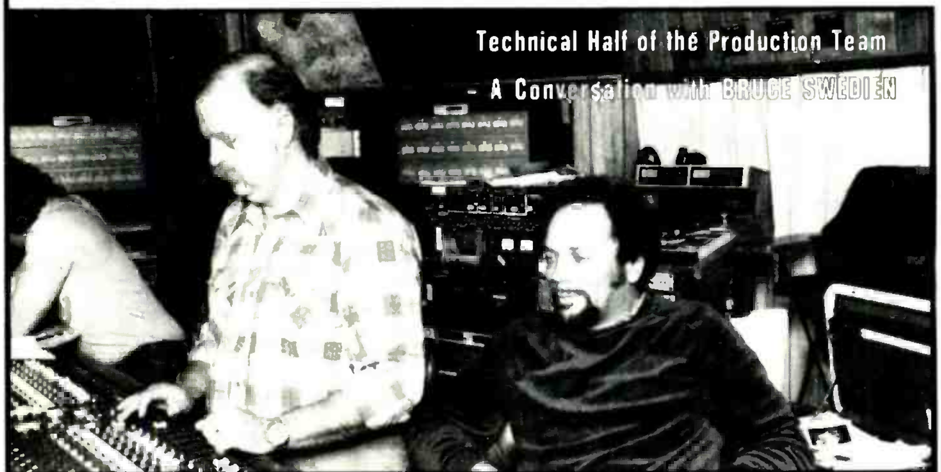
Other times, like with The Brothers Johnson, we used to go in with just a rhythm machine, guitar and bass, and do it that way. We did the Donna Summers album with a drum machine and synthesizer, so that I could really focus on just the material. But with Bruce Springsteen everyone played live, as in a concert. For George Benson's album, Lee Ritenour came over and helped us with different guitar equipment to get some new sounds. At the same time that Lee was there dealing with the equipment, and George was trying it out, Bruce Swedien came over for a whole week to just listen to George with his instrumentals and vocals, like a screen test.

R-e/p: You have obviously established a close affinity in the studio with Bruce Swedien. How do the pair of you interact with one another, and how does he make the moves with you?

QJ: The thing is, what's great about working with Bruce is I like him as a human being. In a funny way, we've the same kind of background. The first record we did together was probably Dinah Washington. During that period of time we recorded every big band in the business. We did a lot of R&B in Chicago in those days . . . a lot of big records.

Bruce's first Grammy nomination was in 1962 for *Big Girls Don't Cry*. He studied piano for eight years, and did electrial engineering in school. Along the way he recorded Fritiz Reiner and the Chicago Orchestra. And a lot of his time was also spent recording commercials. So, from the sound aspect, and the musical aspect, the two of us kind of cover 360 degrees . . . well at least 340. We feel comfortable in any musical environment.

Take, for instance, *The Wiz*, for which



R-e/p: How do you see your role as engineer; working behind the console and handling the technical side of the recording process?

Bruce Swedien: Well, I guess I have to go back just a little bit in my background to really answer that question. Number one, working in Chicago in the jingle field was tremendous training for getting a very fast mix, and being ready to roll because, quite frequently, jingle sessions only last an hour. I recorded all the Marlboro spots, where it wasn't unusual to have a 40-piece orchestra scheduled for nine o'clock downbeat, and literally be ready to roll at 9:05. And when the band is rehearsing, I'm getting little balances within the sections; when the rhythm section is running a certain thing down, I'll use that time to get the rhythm balance ready. It happens very quickly.

I guess I learned a lot about not wasting any movement or motion from the musicians in Chicago. In the early days of the jingles business — about 1958 through '62 — I worked with probably some of the finest musicians in Chicago at that time. They were masters at making the most out of one little phrase. As they were putting the balance together and rehearsing parts, I would be getting it together very quickly behind the console. That was really great training for me.

R-e/p: Too much of an emphasis on the technical side of recording is often said to intimidate an artist in a studio. How do you try and get into the musical groove with the musicians and the producer?

Bruce Swedien: You should be prepared down to the last detail, and get to the studio *early*. Start setting up early; there will always be things to do that you're not apt to think of unless you have enough time. If your session starts at 9:00 AM, be there at 8:00 AM — give yourself at least an hour to set up and prepare the average sized session. Reduce as much of the routine of your work to a regular habit, and always do each job associated with the session in the same order. By scheduling all these mundane mechanical aspects of recording to habit, your mind will be free to think of the creative facets of your work.

R-e/p: What type of musician do you like to record?

Bruce: A musician who gives it up . . . doesn't hold back. Sometimes that's a rare quality. So many musicians go into studios and they kind of tippy-toe around, or they just don't want to commit themselves. I listen for the *real* sound of the instrument and player, not the interpretation.

I like to get to know the player and learn his sound. Ernie Watts, he's my kind of player . . . disciplined! His energy is instant! He never

QUINCY

Bruce handled the pre-recording and shoot. He also designed some of the equipment for the location sound, did the post scoring, the dubbing and the soundtrack album. To do all of it, that's unheard of! Usually there are three to four different people to handle all those facets.

R-e/p: Is there a standard procedure you use for recording the various parts of a song?

QJ: Each tune is different. "State of Independence" for the Donna Summer's album is a good example of a particular process I might use. We started with a Linn Drum Machine, and created the patterns for different sections. Then we created the blueprint, with all the fills and percussion throughout the whole song. From the Linn, we went through a Roland MicroComposer, and then through a pair of Roland Jupiter 8 synthesizers that we lock to. The patterns were pads in sequencer-type elements. Then we program the Minimoog to play the bass line.

The programs were all linked together and driven by the Roland MicroComposer using sync codes. The program information is stored in the Linn's memory, and on the MicroComposer's cassette. At this point all we had to do was push the button, and the song would play. Once it sounds right we record the structured tune on tape, which saves time since you don't have to record these elements singly on tape with cutting and editing. This blueprinting method works great when you're not sure of the final arrangement of the tune.

"It's the song itself that's the *most* important element we are dealing with."

We can deal with between three and five types of codes, including SMPTE on the multitrack. With all these codes, we have to watch the record level to make sure it triggers the instrument properly. Sometimes we had to change EQ and level differences to make sure we got it right.

R-e/p: Do you try and work in the room with the musicians, or stay in the control room with Bruce?

QJ: I like to work out in the room with each player, running the chart down, and guiding the feel of the tune. We will usually run it down once, then I'll get behind the glass to hear the balance and what is coming through the monitor speakers, which is the way it will be recorded and played back. Once I get the foundation of the tune on tape, and know it's solid and right, it is easy for me to lay those other elements to the song. It's the song itself that's the *most* important element we are dealing with.

R-e/p: Any particular "tricks of the trade" that you've developed over the years for capturing the sounds on tape?

QJ: Bruce is very careful with the bass and vocals, and we try to put the signal through with as little electronics as possible. In some cases, we may bypass the console altogether and go direct to the tape machine. Any processing, in effect, is some form of signal degradation, but

BRUCE SWEDIEN... cue/headphone mixes.. the ability to read music... studio rapport with Quincy Jones...

holds back; he'll get it on the first or second take, because he's so used to giving it up. Most of the solos on his album *Chariots of Fire* are first takes with the band. And that's unusual for today . . . really unusual!

R-e/p: Obviously the cue/headphone mix is important to musicians in the studio. How do you help them get into the track?

Bruce: If the instrumentation is small enough, I'll split the Harrison console [at Westlake Studio] and send to the multitrack with half of the faders, and use the rest for returns. In that way you also get the cue mix on the multitrack return faders. It's easier to see what you're doing with the sound using the faders for the cueing mix, as opposed to monitor pots.

R-e/p: Quincy commented that it's important to him that you are able to read music. What do you consider that a young engineer, in particular, should know about the musical side of recording to be a master engineer?

Bruce: I would say the best training is to hear acoustic music in a natural environment. Too many of today's young engineers only listen to records. When a natural sound or orchestral balance is required, they don't know what to do. My folks took me to hear the Minneapolis Symphony every week all through my childhood, and those orchestral sounds have been so deeply imprinted that it's very easy for me to go for an orchestral balance when that's necessary. And I'm talking about the *whole* range — even a synthesizer that is a representation of the orchestra. But, to put that sound in its correct placement in a mix is not easy.

My first advice would be to study the technical end first, so that you know the equipment and what it will do, and what it won't do. *Then* hear acoustic music in a natural environment to get that bench mark in your mental ear.

I think that it is very important for an engineer to understand a rhythm chart or lead sheet. I always make up my own chart with bar numbers on music paper and, as the song develops, I'll add notes and sometimes musical phrases that will be needed for the mix.

R-e/p: Is it important to have a relative sense of pitch?

Bruce: No question about it . . . an absolute must. And I think a knowledge of dynamics is important too. It's not unusual in classical music to have a 100 dB dynamic range from the triple pianissimo to triple forte, and we cannot record that wide a range with equipment. In addition, it's virtually impossible with most home playback systems to reproduce that dynamic range. So, in recording we frequently have to develop a sense of dynamics that does not necessarily hold true with the actual dynamics of the music. It's possible to do that with little changes of level — what I would call "milking" the triple pianissimo by maybe moving it up the scale a little bit. And when you get to the triple forte maybe adding a little more reverb or something, to give the feel of more force or energy.

You see so many guys in studios with their eyes *glued* to the meters. I've never understood that. Take the clarinet, for instance, which can play softly in the sub-tone range; just an "understanding." An engineer would have to know how to deal with a player through the interpretation of the music that would be soft. On the VU meter, which only has 20 dB dynamics, so you don't even see it. In those extremes, your ear is really on its own. You *can't* be the type of guy who has his eyes focused on the VU meter. It's meaningless, absolutely meaningless. The *ear* has to have a bench mark so you know where that dynamic should fall in the overall dynamic range. Quincy is *always* very aware of that, which is a real treat for me.

R-e/p: Having sat in on several of your sessions, I couldn't help but notice that you and Quincy have your own jargon in the studio.

Bruce: You know Quincy and I don't talk much when we work. We spend a lot of time listening: "More Spit" — EQ and reverb; "More Grease" — reverb; "More Depth" — enhance the frequency range, give it more air in the reverb; "Make it Bigger" — beef up the

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"I'm a hands-off producer! When you've got Bruce Swedien taking care of that area, we can't go further. If I can concentrate on other things, I think you can take the plane much higher. To let my deficiency as an engineer be a limitation . . . doesn't make sense."

you are making up for it by adding some other quality you feel is necessary — we always think of these considerations.

Bruce has some special direct boxes for feeding a signal direct to the multitrack, and which minimally affect the signal. With a synthesizer we very often can go line-level directly to the machine, while with the bass you need a pre-amp to bring it up to a hot level.

Lots of times we will avoid using voltage-controlled amplifiers, because there will be less signal coloration. Also, if possible we avoid using equalization. Our rules are to be careful, and pay close attention to the signal quality.

R-e/p: The rhythm section is often considered to be the "glue" that holds a track together. What do you listen for when tracking the rhythm section?

QJ: I listen to the feel of the music, and the way the players are relating to that feel. My energy is directed to telling the players what I want from them to give the music its emotional content, and Bruce will interpret technically the best way in which to capture the sound on tape. And we may try something new or different to highlight that musical character. Because Bruce has a good musical background, he is an "interpreter" that is part of the musical flow.

I like players who have a jazzman's approach to playing. They have learned to play by jamming with lots of different people, and you can push them to their limits. I don't like to get stuck in patterns, so I need players who can quickly adjust to changes in feel. They must also be able to tell a story through their instrument. I look for players who can do it all! [Laughter]

R-e/p: You obviously have a keenly evolved sense of preparation for a recording project. How do you go about planning a typical day in the studio?

QJ: We do our homework after we leave the studio. Bruce will always have a tracking date planned out, with track assignments for the instrumentation, and so on. For overdubbing, he will work out how the work-tape system will be structured, and Matt [Forger] our assistant will be responsible for carrying out that task. I zero in on what my day's work is going to be by listening to the musical elements; how they interact and work in the song in my listening room at home. Bruce does the same by working out in his mind the best method of capturing the music, and structuring these elements so they can be used in future overdubbing and mixing.

I keep a folder for each tune, and make notes as the tune progresses. It may be that changing a stereo image to mono is one way to strengthen an element: stereo for space; mono for impact. If it's a wrong instrument or color it will be redone. Bruce understands the music and the musical balance, and never loses his perspective. Our communication after all these years working together is very spontaneous. This is one of the reasons for our success!

R-e/p: It's obviously important to you that Bruce is able to read music. How does this help you in the studio?

QJ: The way we work with music charts, I can get to any part of the tune. It's fast for drop-ins, and you never end up making a mistake. Bruce will make notes on his music chart to be used later in the mix.

R-e/p: How often do you listen to a work cassette during an album session?

QJ: I'll listen over and over again to a

BRUCE SWEDIEN. . . working with a producer to establish a good vibe in the studio

stereo spread; "More Explosive" — bring the level down, add some reverb, adding a trail after it. Quincy picks the sound or effect; I put the thought into application by choosing the "color," if you like.

R-e/p: While there may be no hard and fast rules in the studio, have you picked up any tips about how to work creatively with a producer?

Bruce: An engineer's important responsibility is to establish a good rapport with the producer. Nothing is a bigger turn off in a studio than a salty, arrogant personality. I have seen this attitude frequently in an engineer, and heard him describe himself as "Honest." It is *very* important for the engineer to know what a particular producer favors in sound. Producers vary somewhat in interpretation of a style, or concept. You must have a mind open to new ideas.

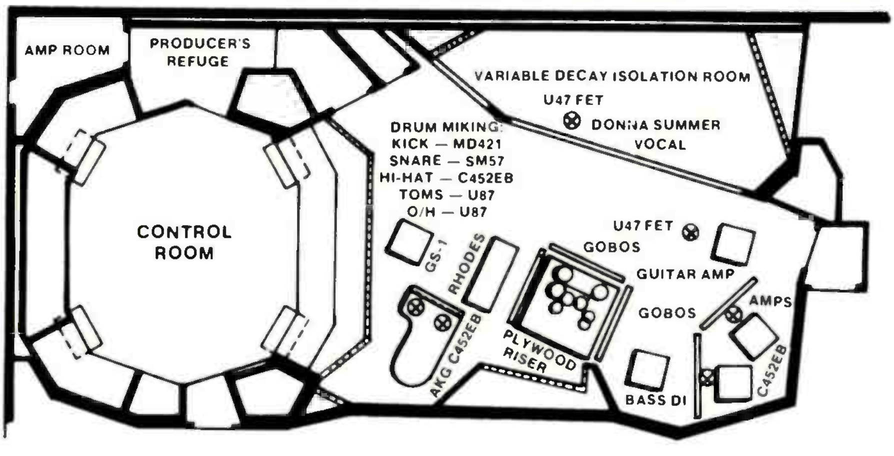
The relationship between the engineer and producer, like any other human relationship, is based on the confidence and trust that exists between the individuals. A bond of trust and respect can only result between professionals working well together to achieve a common goal.

R-e/p: How does the engineer set a good vibe with the producer and the musicians?

Bruce: It's a two-way proposition. I've been in situations early in my career — fortunately I don't have to deal with that any more — where producers were not inclined to allow the engineer to be involved in a recording project. I don't think that's the case anymore, at least in the upper level of the business, because it's a fact that engineers *do* contribute a lot of useful input. Yes, it's absolutely true that an engineer can help an incredible amount in the production of music.

R-e/p: After the tracking and overdubs, how do you set yourself up for the mix?

Bruce: I'll have many multitrack work tapes. For example, on Donna Summer's tune "State of Independence" I had eleven 24-track tapes — each tape has a separate element. Then I combine these tapes into stereo pairs. In the case of synthesizer, horns, background vocals, strings, sometimes I will use a



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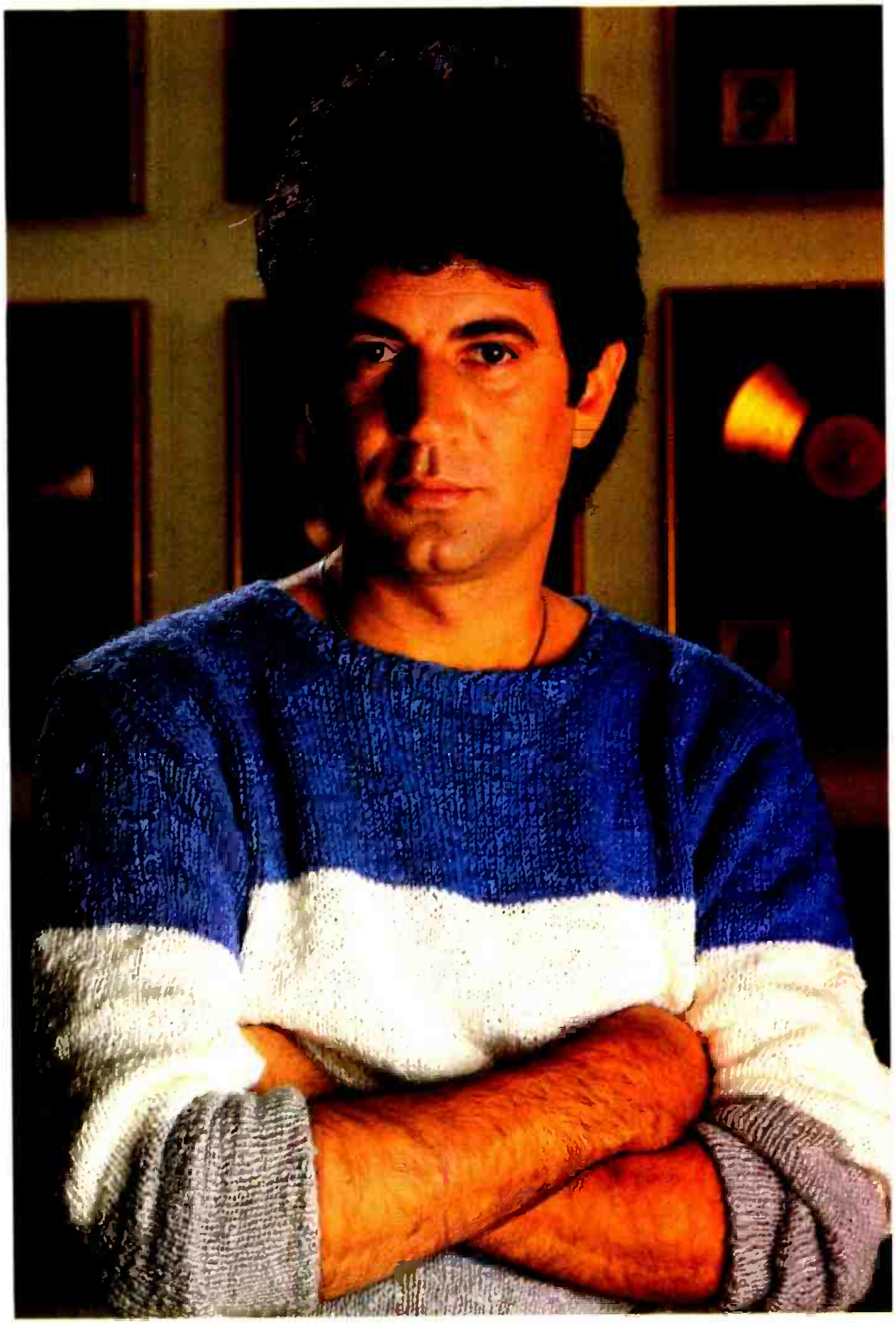

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Output Impedance	100Ω/+4 dBm position 650Ω/−20 dBm position Mixed 550Ω/−20 dBm position Delay
Delay Time	0 to 640 mS in 1 mS steps
Delay Accuracy	+0.5%
Frequency Response	10 Hz to 100 kHz (Direct) 10 Hz to 16 kHz (Delay 0 to 320 mS) 10 Hz to 7.2 kHz (Delay 0 to 640 mS)
Signal to Noise Ratio (IHFA)	90 dB (Direct) 80 dB (Delay)
Dynamic Range (IHFA)	Greater than 12 dB (Direct) Greater than 90 dB (Delay)
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QUINCY

song until it's in my bones. Some songs have just a chord progression and no tune. Others may be a hook phrase and a groove, and sometimes the song may call for a lot of colors. Each song is different . . . when it's played on the radio and *jumps*, I'm happy.

To keep the session vibe up, I use nick names for the guys I work with: "Lilly" for Michael Boddicker; "Mouse" for Greg Phillinganes; "Boot" for Louis Johnson; "Worms" for Rod Temperton. And Bruce has many nicknames; it depends upon the intensity in the control room. If things are going a little rough and I need a hired gun, I call

Bruce "Slim"! [Smiles across room at Bruce Swedien]

And the way I keep in touch with the tracking musicians is to use slang: "Anchovy" is a mess up; "Welfare Sound" is when you haven't warmed up to the track or the tune; "Land Mines" are tough phrases in an arrangement.

R-e/p: How do you gauge that a track is happening in the control room?

QJ: I listen on Auratones for energy and performance at about 90 dB SPL. I'm coming from a radio listener concept. I have two speakers set up in front of my producer's desk. I don't have to ask Bruce to move so I can listen to his set of speakers, and we never play the two pairs of speakers at the same time. When it's a great take you can *see* through it!

R-e/p: With such wide experience over

BRUCE SWEDIEN . . . setting-up for a mixdown session . . . automation in the final mix . . . Donna Summer tracking and overdubs

fresh tape, or there are open tracks on the master tape.

The original rhythm track is always retained in its pure form. I never want to take it down a generation, because the basic rhythm track carries the most important elements, and I don't like to loose any transients. With synthesizer or background vocal you could go down a generation without loosing quality. I call this process pre-mixing, and we use whatever technical tricks it takes to retain sound quality.

We pre-mix the information on two tapes, and bring them up through the console. Having established the balance all the way through the recording process, we then listen to all the elements, and Quincy will make the decision based on what the music is saying. We usually have more than we need. This stage is editing before we master —listening to everything once saves time, and we don't have to search for anything.

Sometimes though, we may have to go back and re-do a pre-mix if the values are not right. For example, a background vocal part may have the parts stacked, and one of these parts might be too dominant. Or sometimes everything sounded fine when we were recording the element, but with everything happening on the track the part gets lost. Then we go back and re-establish a new balance by pre-mixing that particular element again. We also pay close attention to psychoacoustics — in other words, what sound excites the listener's ears. These are the critical things in the mix that will make the difference between a *great* mix and a so-so mix.

Also, we are sensitive to the reverb content. Quincy may ask me to bring more level up on a given element. I may suggest adding more reverb, which will create more apparent level. I establish what the mix will be, and Quincy will comment on the little changes and balances; these are the subtle differences that make for a great mix. We overlap our skills. Quincy becomes the navigator, and I fly the ship!

R-e/p: Does it take very long to get a mix that you both like?

Bruce: Quincy will work with me for the first few days until all the production values are made. Then we close down the studio and I will polish the mix until I like it. After I get it right, Quincy receives a tape copy for the final okay. Because of Quincy's business phone calls we have found that to be the best way to finish a mix. We know the mix is right when we've made the musical statement that we set out to make.

R-e/p: Do you use automation during the final mixing?

Bruce: Yes, because it gives you more time to listen by playing the mix away from the basic moves. Automation is a tool I use for re-positioning my levels. Then I can make my subtle nuances in level changes to get the right balance.

For monitoring the final mixes I am a firm believer in "Near-field" or low-volume monitoring. Basically, all this requires is a pair of good-quality bookshelf speakers. These are placed on top of the desk's meter panel, and played at a volume of about 90 dB SPL or less. My reason for using Near-field monitoring is twofold. The most important reason is that by placing the speakers close to the mix engineer, and using an SPL of no more than 90 dB, the acoustical environment of the mixdown room is not excited a great deal, and therefore does not color the mix excessively. Secondly, a smaller home-type book shelf speaker can be used that will give a good consumer viewpoint. My personal preference for Near-field monitors is the JBL 4310; I have three sets.

Each musical style has its own set of values. When mixing popular dance music, for example, we must keep in mind the fact that the real emotional dance values are in the drums, percussion and bass, and these sounds must be well focused in the mix. Making a forceful, tight, energetic rhythm mix is like building a house and making sure the foundation is strong and secure. Once the rhythm section is set in the mix, I usually add the lead vocal and any melody instruments. Then, usually the additional elements will fall in place.

For mixing classical music or jazz, however, an entirely different approach is required. This is where the mixing engineer needs a clear knowledge of what the music to be mixed sounds like in a natural acoustical environment. In my opinion, this one area is where beginning engineers could benefit their technique a great deal. It is *absolutely* essential to know what a balanced orchestra sounds like in a good natural acoustical environment. Often, the synthesizer is used to represent the orchestra in modern music. A knowledge of natural orchestral balance is necessary to put these sound sources in balance, even though traditional instruments are not necessarily used.

R-e/p: You have provided us with a studio setup plan of the recent Donna Summer sessions at Westlake. How do you plan the tracking and overdubs?

Bruce: I generally record the electric bass direct. I have a favorite direct box of my own, which utilizes a specially custom-made transformer. It's very large and heavy and, to my ear, lends the least amount of coloration to the bass sound, and transfers the most energy of the electric bass on to the tape. From my own personal experience though, active direct boxes are very subject to outside interferences, such as RF fields — you can end up with a bass sound that has a lot of buzz or noise on it.

The miked electric bass technique alone usually does not work very well, primarily because there are very few bass amplifiers that will reproduce fundamental frequencies with any purity down to the low electric bass range. In jazz recording the string bass is always separately miked. My favorite mike is an Altec 21-B condenser, wrapped in foam and put in the bridge of the instrument; I own four of these vintage mikes that I keep just for this purpose. You also can get a good string bass pick-up with an AKG-451, placed about 10 inches away from the finger board, and not too far above the bridge. This placement

BRUCE SWEDIEN . . . microphone and outboard equipment selection and setups

gives a nice, mellow, not too close sound.

Equalization and limiting that I use for bass is to generally boost the 600 to 800 Hz range about 2 to 4 dB, and limit during the recording using a UREI 1176LN set for a 4 to 1 compression ration, but taking off only 2 or 3 dB peaks, maximum.

In a pop recording, the drum set should be treated as a combination of instruments, because many of the desired effects have to emphasized electronically. If the effects were heard in a natural acoustical balance, much of this sound would be lost. Multiple miking on the drum set is the only answer.

I mike the overall set with a pair of high-quality condenser microphones about six feet in the air, in a stereo configuration. These two microphones hear the drum set in a fairly natural acoustical balance, and usually include a good pickup on the overhead cymbals. I place individual mikes on the tom-toms around the drum set, and then these are mixed together with the two overhead mikes to form a left and right stereo image of the drum set. The choice of microphones varies considerably with the original sound of the drum set.

A good choice for overhead is the Neumann U-87 in cardioid pattern. On the tom-toms this varies greatly with the sound of the drums. If the drummer has really good quality tom-toms which are tuned to specific tonalities, usually a Neumann U-87 or KM-84 will work well placed about 8 to 10 inches away. If the drummer has tom-toms that have very little low-frequency content and need a lot of help, the mike is generally placed closer, to make use of the low-end proximity effect.

For bass drum I use a Sennheiser MD421, AKG C412, or an Electro-Voice RE-20. I also use a specially made drum cover with elastic around it and a slot in the middle for the mike.

On snare I use a mike technique that I developed many years ago in Chicago for recording rhythm and blues records, where a hard drum sound was necessary. My current choice of microphone for the snare drum is the AKG C451, with a cardioid condenser capsule and 20 dB pad. I mike the snare drum from the side about 8 to 10 inches away from the shell of the snare drum, being very careful not to position the mike anywhere near the air hole on the side of the shell.

On the sock cymbal or high-hat I use two types of microphones, depending entirely upon the sound. One choice is an RCA 77DX ribbon mike, and the other an AKG C451 (occasionaly I'll use a Shure SM57 dynamic).

While recording drum tracks I never use any limiting or compression, and very little equalization. The only drum mikes on which I would use any EQ would be the kick, usually boosting it about 2 to 4 dB at 1.5 kHz, and maybe a 2 dB peak at 100 Hz. On the sock cymbal I will usually use a highpass filter set at 100 Hz.

The electric guitar has an incredible range of colors; many players are using two amps, and have their own signal processing equipment. I mike each amp with a C452 condenser to capture a close representation of what the guitarist hears from his amp, and then record the sound in stereo. I may add a 2 to 4 dB peak between 5 and 10 kHz, depending on the range of the music.

For acoustic piano I use a pair of AKG C414EBs, positioned about 24 inches above the harp of the instrument in an "X" stereo configuration, to favor the high frequencies of the instrument. Both microphones are hearing the total range of the instrument, but in their position the phase differences of the frequencies provide a true to life and interesting stereo picture. I usually EQ the piano 2 to 4 dB at 10 kHz, and another 2 dB peak at 3 kHz. If the piano sounds thin a 2 dB boost at 100 Hz will help.

In choosing a microphone for a solo or lead vocal, the most important thing to consider is the vocal timbre of the artist. In other words, is it soft and breathy, or loud and penetrating?

Stacking or doubling a lead vocal is often helpful. Frequently I will slightly change the tape speed of the master recorder during recording of the stack. When this is combined with the original, it seems to add a full, rich quality to the lead vocal that makes it more interesting. When mixing a double I keep it at less level in the mix than the basic lead vocal track. This serves to add support to the vocal track, without making it appear to be an obvious trick.

R-e-p: What kind of outboard equipment do you use?

Bruce: I have no set plan, but I can give you some general uses: Allison Kepex II for noise gate on guitar tracks; Lexicon digital delay line to pre-delay the echo chamber; dbx Model 160 as a fast, flexible all around compressor/limiter; a dbx Model 165 Over-Easy works well for vocals; an Eventide H-949 Harmonizer for speed sounds; Eventide FL201 Instant Flanger for solo instruments; AMS digital delay line for fixing the pitch of a given signal that is a little out of tune; Inovonics 201 comp/limiter for pumping the signal on low-end instruments (bass synthesizer, etc.); an Orban De-esser works real good for stacked vocals with sibilants; I always use two EMT 250s with my own plate, Lexicon 224 Digital Reverb gives me another echo "color"; UREI Model 527 graphic equalizer to EQ the echo send; and Sontec Parametric EQ.

If I'm going to be working in a studio for any length of time I'll have them install the two Ecoplates that were made for me in 1977 by Studio Technologies. I always use my JBL 4310 speakers for mixing, and I also carry many of my own microphones; I have a collection of 60 or so. I bought my first mikes —two Telefunken U47s — in 1951 when I was still in high school, and they are still in mint condition today. The best way for me to rely on my approach to sound is having many of my own mikes with me, plus my own custom direct boxes.

R-e/p: On a couple of the albums you've recorded, the liner notes make reference to something called "Acusonic Recording." the years in so many state-of-the-art facilities, it's perhaps surprising that you haven't opened your own studio.

QJ: I'm not at all interested in owning a studio. I've got enough things to take care of, and I like to leave the studio right where it is when I get out of it. I don't want to have to think of what's going on in the studio . . . if the 24-track heads are clean!

R-e-p: So you look upon the studio equipment as just being a production tool?

QJ: Absolutely . . . as an instrument; matter of fact, just like a part of an orchestra. I think even a variable speed oscillator is an instrument.

R-e-p: Do you have a favorite studio in which you feel most comfortable?

QJ: My favorite studio at the moment is Westlake [Los Angeles]. I like the vibe there very much. The people are fantastic — great maintenance handled by Jimmy Fizpatrick and Dave Concord. Matt Forger, the assistant engineer we work with all the time, is incredible; he shares in the vibe. His duties have gone beyond the page. In fact, we have given him a new title: "Technical Director."

Glenn Phoenix, the president of Westlake Audio, has a well rounded knowledge of the whole studio scene. We can give him feedback for new designs. When you walk in the door you feel the flow, and everyone's tuned in. At the technical level in a studio, Bruce is a real perfectionist. If something is not working right he'll find it! We *have* to have good maintenance.

R-e-p: Are you a "hands-on" producer, or do you leave it all to Bruce? Do you ever get behind the faders?

QJ: I'm a hands-off producer! When you've got Bruce Swedien taking care of that area, we can't go further. I'm not the "Do it all type." If you can concentrate on other things, I think you can take the plane much higher. To let my deficiency as an engineer be a limitation . . . that doesn't make sense.

R-e-p: Although you might leave the hands-on side of running the session to Bruce, you seem to have a pretty good grasp of the technology of the recording process. Are there any aspects with which you are uncomfortable?

QJ: Automation sometimes bugs me, because after a while it kind of tells you *how* to mix and, if you want to re-adjust things, it sometimes gets difficult. It happens sometimes that you can get so far with automation, and then want to make a little left turn there; sometimes the system makes it difficult for you to do it.

I found I'm going to wait a little while with digital too. So far, digital doesn't give me what I'd like to hear — it doesn't sample enough of the regularities. It nails the *characteristics* of a sound, but it doesn't get the dirt . . . the build up . . .

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QUINCY

that I think is necessary to have in a record. Digital sometimes gets a little *too* squeaky clean for me. But I know it's going to improve, because it's a wonderful direction.

R-e/p: With album sessions becoming more and more complicated, both technically as well as artistically, do you think a producer has to be a good arranger too?

QJ: I don't know, because everybody produces with his strength. That ability can come from the strength of an engineer, player, singer, instrumentalist, arranger, or a combination of these things.

An illustrious vocal team during Donna Summer overdub sessions at Oceanway Studio, Los Angeles. From left to right: Stevie Wonder, Brenda Russell, Michael Jackson, Dionne Warwick, Lionel Richie, Christopher Cross, Dora Bernard, Kenny Loggins, Michael McDonald and James Ingram . . . producer Quincy Jones.

R-e/p: As founder and president of your own record company, Quest, do you find it hard sometimes to combine the creative ability of a producer, with the business side of running a label?

QJ: Let me give you some background. In 1960 I got in trouble with a jazz band I had on tour, and when I came home with my tail between my legs from Europe I took a job with Mercury Records for about seven years, in A&R, and eventually vice president. During the course of that time I had to understand a whole different area of the record business that I wasn't even aware of before. It was a big company because Mercury merged with Philips, which is now Polygram, and we started Philips Records in this country.

It was an incredible education, because I use to think that all these companies get together once a week to plan how to get new artists on the label. You should be so lucky that you get past being an IBM number on a computer with a profit and loss under your name or code nubmer. That gave me an insight into understanding what corporate anatomy was all about.

Understanding the rules of the game is important for a producer with a huge company like Philips, which is dealing with raw products, television sets, vacuum cleaners, and all the rest, at the that time we were doing $82 million a year worldwide, and music was only about 2% of the total.

R-e/p: So how do you communicate with the business person?

QJ: Somewhere along the way it's got to make sense if it's going to cost money. If you want to go to Africa and make a drum record, for example, you're going to have to figure out how to get it done for the people who put up the money. Somewhere along with your creative process you have to 'scope out what the situation is, get your priorities straight, and don't let that interfer with your creativity. If they put a pile of money right in front of you, there's no way to correlate the essence of what that means, and yet still tie it into the creation of music.

Being a record company president is a lot of responsibility, but it's going to be okay. To become a successful record company president, you have to apply and reinforce your creative side with a business side, but you can't *lead* with the business side.

BRUCE SWEDIEN . . . "Acusonic Recording" and SMPTE interlock . . . going beyond 46-track with multiple slave reels and worktapes

What does that refer to?

Bruce: Quincy came up with the term to describe the way I work — my "philosophy for recording music" if you like. To be more specific, it's really my use of two multitrack machines with SMPTE codes — "Multichannel Multiplexing." Essentially, by using SMPTE timecode I can run two 24-track tape machines in synchronous operation, which greatly expands the number of tracks available to me.

Working with Quincy has given me the opportunity to record all styles of music. With such a variety of sounds to work with I could see that single multitrack recording was not enough to capture Quincy's rich sounds. I began experimenting with Maglink timecode to run two 16-track machines together in sync. This offered some real advantages, but since then I have expanded my system to use SMPTE timecode and two 24-track tape machines.

The first obvious advantages that come to mind are: lots of tracks, and space for more overdubbing. With a little experience I soon found that the real advantge of having multiple machines with Quincy's work is that I can retain a lot more true stereophonic recording right through to the final mix. An additional major advantage is that once the rhythm tracks are recorded, I make a SMPTE work tape with a cue rhythm mix on it, and then put the master tape away until the mix. In this way we can preserve the transient response that would be diminished by repeated playing during overdubbing.

Quincy usually has the scheme for the instrumentation worked out for the song so we can progressively record the elements on work tapes. For example: Work tape A may have background vocals; Work tape B lead vocals; Work tape C horns and strings; and Work tape D may have 10 tracks of synthesizer sounds to get the desired color. All of these work tapes contain a pitch tone, SMPTE timecode, bar numbers cues, sometimes a click track, and cue rhythm mix.

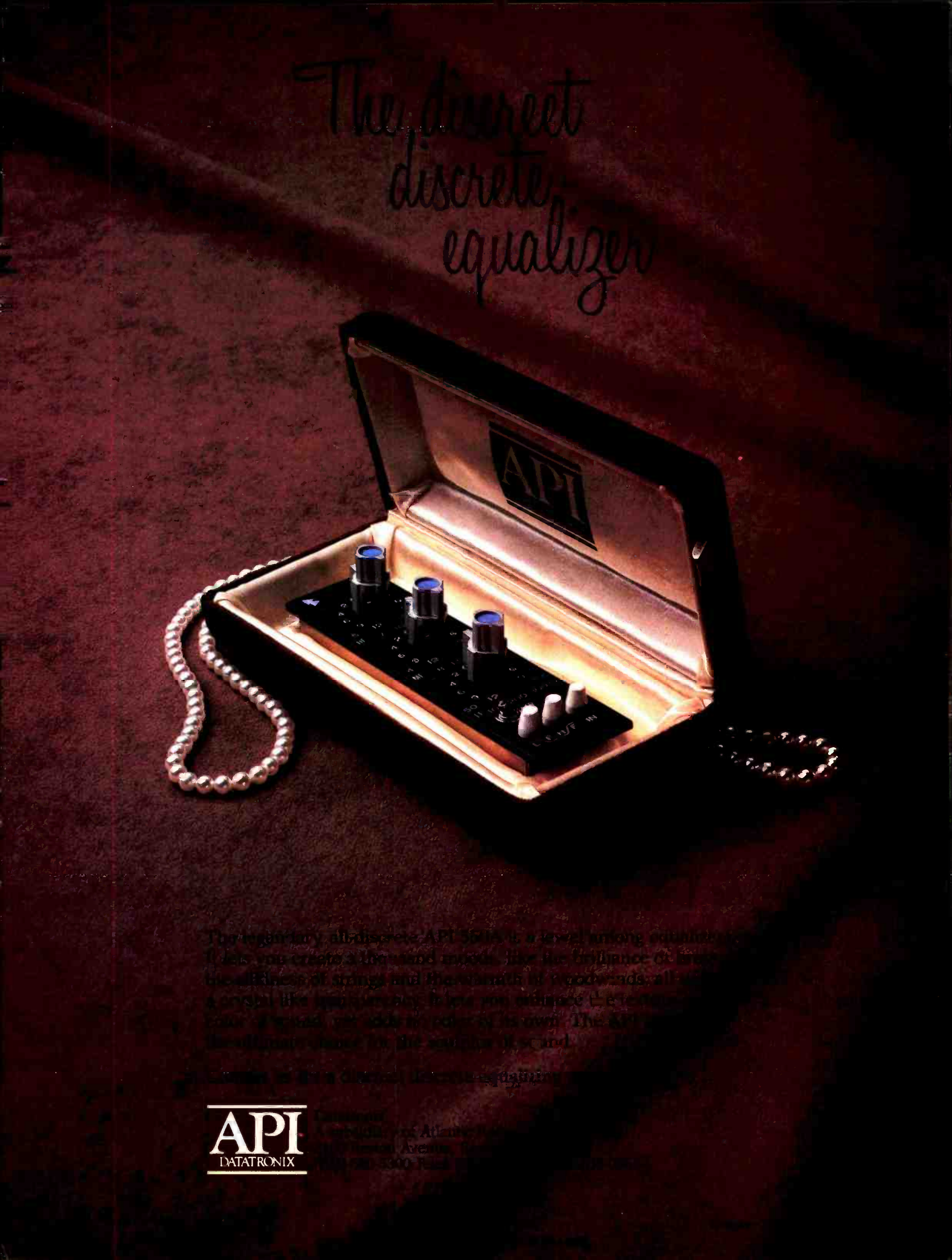
R-e/p: What kind of interlock device do you use to sync the multitracks?

Bruce: We use two BTX timecode synchronizers. A BTX Model 4500 is used to synchronize the two multitrack machines, and I keep the 4500 reader on top of the console in front of me to provide a SMPTE code readout. We work with real time from the reader, and don't depend on the autolocator during the work tape stages.

There are times when I'll use the "Iso" mode on the 4500 to move one element on a tape to a different place in the tune. Say, for instance, you have a rhythm guitar part that isn't tight in a section; I'll find it on the slave work tape and move it to the new location on the master work tape.

R-e/p: How long does it take to make a work tape?

Bruce: We start by adding the SMPTE code, and I'll make a few passes with a mix until I like it. Then we record the pre-mix on to the new work tape. I'm very fussy about the sound, and we'll listen back and forth between the master and the slave tape to make sure the sonics match before we move on to the next work tape. I always want Quincy and the dubbing musicians to hear my best. It takes about three hours per work tape to finish the job. To keep all the tape tracks in tune, we also calibrate the speed of tapes by going through a digital readout. ■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LOGIC TESTERS

Hewlett-Packard HP-5023 Logic Probe Kit

FAULT FINDING AND DIAGNOSIS OF AUDIO PROCESSING AND RECORDING EQUIPMENT

by Roman Olearczuk
Technical Consulting Editor,
and Technical Engineer, Studio 55, Los Angeles

The increasing use of digital logic circuitry in today's professional audio gear requires a different variety of test equipment for fast, successful fault repair. To fill the need for simple yet powerful test instruments for logic fault-finding, in recent years several manufacturers have developed a family of logic testers that make digital circuit diagnosis easy.

Digital troubleshooting basically involves the verification of valid logic levels at all inputs and outputs of the individual interconnected ICs in a digital network. For effective fault location, the technician must have at his or her disposal complete operational data for each IC under test. Truth tables and timing relationships, as well as valid logic threshold information, play an important part in the checking of proper circuit operation.

For analog circuits, a technician using traditional test instrumentation, such as an oscilloscope, multimeter, and signal generator, can apply stimulus-response techniques to quickly isolate an electronic problem. In principle, this same technique can also be applied to digital circuitry. However, the use of the above-mentioned test equipment is, at best, time-consuming, if not cumbersome, when such instruments are applied in complex digital network analysis. While an oscilloscope will provide voltage and timing information, the voltage levels must be read constantly to see if they fall within the valid logic family thresholds.

Meanwhile, short pulses or one-shots are difficult to detect, even with the best triggering oscilloscopes. A multimeter can be useful, but only when static logic levels are prevalent in the circuit design. A signal generator can also be used, but only at convenient signal interjection points; otherwise PCB traces would have to be cut in order to get at specific integrated circuits.

Logic testers alleviate these problems since they are designed to take advantage of the digital nature of the signals generated within the ICs under scrutiny. They were developed to provide a simple indication of logic 1's and 0's, as well as generating and detecting pulses. The purpose of this article is to describe these popular devices, and show their application in the recording studio.

Logic Probes

The most common digital tester is the logic probe. This hand-held device somewhat resembles a large, fat pencil with a pin-tip on the end. It is put into service by simply connecting the power leads and ground of the unit to the most convenient and correct voltage supply points on the card under test. Now, with this active unit the technician can "probe" (with the tip) various test points, IC pins, and circuit traces for logic levels and/or pulses. Most logic probes available today provide the following features:

- HI and LO level indication using either one or two LEDs.
- Non-valid logic level indication.
- Pulse train indication up to 50 MHz.
- Pulse memory to capture single pulses or level transitions.
- 10 nanosecond pulse response.
- Multilogic family test ability.
- High input impedance for minimal loading effect.
- Voltage overload protection.

How these features are designed into each unique probe differ vastly among the manufacturers. For example, the HP 545A Logic Probe uses a single LED near the tip, visible from all sides, that becomes illuminated with a high logic level. Meanwhile, the B&K DP-51, Global Specialties LP-3, Kurz-Kasch LP-770, Non-Linear Systems MLB-1, OK PRB-1, and Radio Shack 22-301 Logic Probes, all use two LEDs to show both a LO and HI level state. The Production Devices 110 probe has an added feature that provides the user with a HI and LO audio tone, in addition to two LEDs for logic level recognition. The Heath kit IT-7410 probe is unique, since it is sold only in kit form. It also uses separate LEDs for level indication, but provides omnidirectional viewing found otherwise only in the HP 545A. Most of the probes surveyed here also detect invalid logic levels that are between the accepted voltage thresholds; these are usually displayed by the absence of a lit LED.

All these units feature pulse detection, as well as pulse memory, except for the Production Devices probe. Measured pulse trains are recognized either through an additional blinking LED, or by a flashing high-level LED at a steady rate between 3 and 10 Hz. The B&K DP-51, Global Specialties LP-3, and Kurz-Kasch LP-770 units provide a relative display of duty cycle, by varying the intensity of HI/LO LEDs while the pulse LED is flashing at a fixed frequency.

The Non-Linear Systems device also provides a gross indication of frequency for the pulses under measurement. Below 100 Hz, the brightness of the HI and LO LEDs is duty-cycle proportional. From 100 Hz to 100 kHz, the HI and LO indicators are programmed to be both on for 50% duty-cycle pulses, while duty cycles below 25% or above 75% illuminate the HI/LO LEDs in an Exclusive-OR fashion. Above 5 MHz, the HI and LO lights extinguish, and only the pulse indicator is operational.

Most units can capture "one-shots" as small as 10 nanoseconds. The probes made by Radio Shack, Production Devices, and Non-Linear Systems detect pulses as short as 50 nanoseconds. When a pulse occurs, the memory "stretches" the pulse and displays it by illuminating the pulse indicator. The OK PRB-1 probe resets the pulse memory automatically, while memory storage in the other products has to be reset by the press of a button.

. . . continued overleaf —

LOGIC TESTERS

A majority of the units operate up to and beyond 50 MHz. (Note: a few products only have a maximum operating frequency of 10 MHz, and this drawback is reflected in an inexpensive price!) The Kurz-Kasch LP-770, for example, has a claimed frequency specification of 100 MHz.

The ability to test multifamily logic states is a fairly common feature among the probes included here. The most standard design is usually a selector switch that provides DTL/TTL testing at one setting, and CMOS/HTL at the other. Generally, voltage thresholds for valid logic detection are set at 48% and 16% (of the supply voltage), or precise thresholds of 2.4 volts DC and 800 multivolts DC for TTL HI and LO states respectively. CMOS/MOS thresholds are typically 70% and 30% of the DC supply voltage. In addition to these families, Global Specialties also manufactures a logic probe (LP-4) for the ECL logic family, which requires more precise logic thresholds and supply voltages. The Kurz-Kasch LP-770 unit also provides a separate switch for HTL logic, with thresholds set at 1.5 and 9.5 volts DC. Finally, the unique OK PRB-1 logic probe automatically changes thresholds; based on the logic family under test it senses the supply voltage and adjusts itself accordingly.

The input impedance of any logic tester should be high enough not to affect circuit performance when put into service. Also, the tester should provide an overvoltage protection for the user, so that the test tool isn't inadvertantly "wiped-out" on a mistaken high-voltage test point. In this survey, the majority of logic probes exhibited an input impedance of well over 1 Mohm. The least expensive units, as a rule, tend to have either unpublished and unknown specifications, or input impedances that are only in 100 Kohm region. Overvoltage protection features also vary considerably in competing products. The maximum protection is provided by the Kurz-Kasch logic probes, which can withstand 140 volts DC reverse voltages, as well as 110 VAC potential.

It should be noted that the model numbers listed in the text do not indicate the complete range of product models available by each manufacturer. Several companies produce low-cost units, and special application devices. Additional information regarding product line, specifications, pricing and warranty can be obtained by contacting the individual companies listed in the accompanying Table.

Logic Pulser

The logic pulser can be thought of as a companion tool for the Logic Probe. While it is similarly shaped and specified, the Logic Pulser is designed to provide the stimulus "signal" for digital circuit analysis. Basically, the pulser is a quality pulse generator without the controls. It is capable of "sourcing" or "sinking" a test point, by providing short, bounce-free, high-current pulses that can overcome stuck nodes without IC destruction. Although not all manufacturers in the table offer pulsers, the ones that carry them provide devices with similar features. All products have selectable single pulse or pulse train models; TTL and CMOS test capability; high input impedance; and overvoltage protection. However, the HP 546A Logic Pulser is unique in that it offers six ROM programmable output patterns; these signals consist of single pulses, pulse streams of 1, 10 or 100 Hz, and pulse bursts of 10 or 100 pulse duration. The pulse-burst feature, when combined with single pulses, enables a technician to obtain any exact number of pulses to check circuits that trigger digital events.

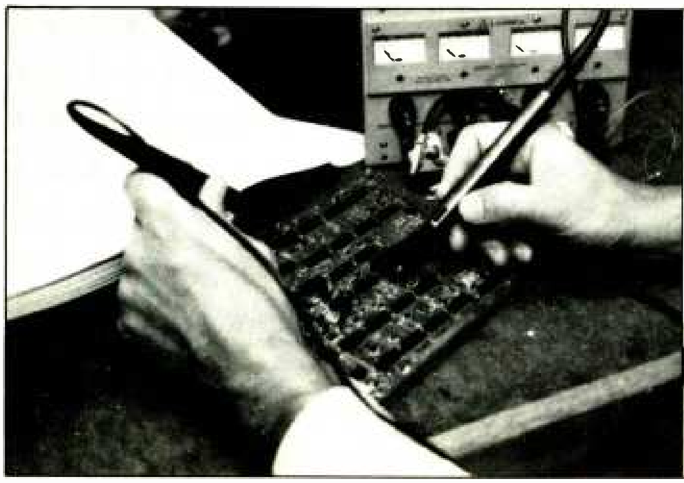
Logic Clip

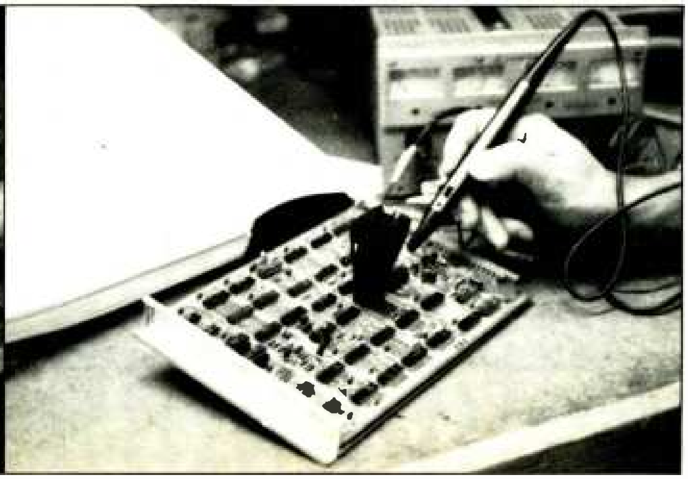
Another logic tester is the easy-to-use Logic Clip. As the name implies, this device resembles a DIP IC clip, yet it also incorporates 16 LEDs in two rows that correspond to the DIP pin numbers. The indicators show simultaneous and independent logic information on any 8-, 14-, or 16-pin logic IC, and is simply clipped on to the device under test. Which ever way the clip is connected, the devices automatically senses which pins are the supply voltages, and thereby draws its power. In one strong application, timing relationships can be viewed easily when the pulsed clock rate is one pulse per second.

However, the logic clip's use, as easy as it seems, does have a few drawbacks. The indicators only light when a single threshold is exceeded. A true valid LO threshold is not really verified by an extinguished LED. Therefore, bad levels are not indicated on this device. Although pulse activity shows up as a dimly-lit LED, single pulses are not captured and displayed as they are in a Logic Probe. Power drain requirements can also tax the circuit power supply if all LEDs are continuously in use.

The HP 548A clip accommodates both TTL and CMOS ICs automatically, while the Global Specialties LM-1 is designed primarily for TTL. The HP unit also provides an auxiliary clip to power the unit from external supplies. The only requirement is that the input voltage does not exceed 30 VDC, and that it is at least 1.5 VDC greater than any pin of the IC being measured. Glo-

Investigation of a faulty Ampex ATR-100 Transport Control Card that is exhibiting a total short across its +5 volt supply rail (as detailed in Example 3 overleaf). In the left-hand illustration, the author has connected a Logic Pulser to the +5 volt rail, and is using a Current Tracer held in his right hand to follow the progress of current flow along vertical and horizontal rows of ICs. In the right-hand photograph, the fault has been located at IC A27, and the chip replaced. Correct operation is now being verified by mounting a Logic Clip on IC A27, and applying pulses to the input pins while checking state changes of individual logic gates.





Photography by: KATHY COTTER

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For additional information circle #41

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Fantasy Studios

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bal Specialties also manufactures a Logic Monitor (LM-2A) that tests TTL and CMOS, and provides fixed and variable thresholds. It is more cumbersome to use, however, since the device connects the DIP clip through a 24-inch ribbon cable on to a small case, where the LEDs are located. The same company also recently introduced a low-power, TTL/CMOS, 40-pin Logic Monitor (LM-4) that shows simultaneous logic information on a 40-segment liquid-crystal display. This unit can be quite useful for diagnosis of microprocessors, ROM, RAM, and other LSI chips.

Current Tracer

At this time, the Current Tracer is a logic tester manufactured solely by HP. This powerful tool locates current source and sink faults by sensing the magnetic field generated by pulse active circuits. To use the 547A, one simply adjusts the reference control (for a sensitivity of 1 milliamp to 1 Amp) near the tip at a known pulsing node. The circuit activity then can be traced from this node, and the changing intensity of the indicator in the tip will show precisely where the current is flowing. Stuck nodes in large fan-outs can be easily diagnosed, as well as complete circuit board supply to ground shorts. The HP unit works with all logic families within its 4.5 to 18 VDC power supply requirements, and will provide an indication for single pulses greater than 50 nanoseconds in width, and pulse trains up to 50 MHz.

Logic Comparator

Finally, there remains one other simple logic tester whose disadvantages outweigh its advantages to the extent that it has not attained popular use. The HP 10529A logic comparator can easily verify the operation of an in-circuit logic IC, by comparing it against a properly functioning reference chip. Any difference in logic states will show up on the display indicators. This useful tool is limited, however, to only DTL/TTL circuitry. In addition, each reference IC must be programmed through the use of either pre-soldered IC reference boards, or through selectable input/output switches on a universal socket, prior to the insertion of the reference IC. As a resplt, to test a wide range of TTL ICs, a large supply of pre-programmed boards must be stocked, or the technician will have to continuously program the universal socket as he moves from IC to IC.

For advanced microprocessor-based circuitry, another level of test equipment — generally in the form of logic analyzers, signature analyzers, and micro-system testers — has been designed by various manufacturers to address the intricate fault-finding procedures required to service these kind of digital networks. Due to their complex nature, a separate article would need to be devoted to the description and use of this type of equipment; the following section will only concentrate on the application of the simple and popular logic testers discussed above.

Troubleshooting Procedure

Prior to troubleshooting digital ICs, one should have a knowledge of what faults to expect during the diagnosis procedure. There are only two types of circuit faults: internal IC failures and external IC failures,. Internal flaws usually comprise of any combination of the following: input or output open circuits; voltage supply or ground shorts to an input or output; short circuit between any two IC pins; and internal circuitry breakdown. External failures usually show up as one or more of the following: open circuit signal path; short circuit between signals; short circuit between any one or more voltage supplies, grounds, and signals; and a breakdown of a discrete component.

Armed with this information, a good source of digital IC data, and an equipment service manual, a technician can easily stimulate circuits with the Logic Pulser, while he checks their response against published specifications with the combined use of a Logic Probe, Logic Pulser, Logic Clip, and Current Tracer. The examples detailed below will demonstrate this fault-finding procedure as it is used in diagnosing of some well-known studio equipment. Once the diagnostic techniques are mastered, and a thorough service manual is obtained for the equipment under test, a service or maintenance technician will be able to solve difficult logic problems with less frustration and time. □□□

Further Reading

"New Techniques of Digital Troubleshooting"; Hewlett-Packard Application Note 163-2.
"Logical Troubleshooting"; *HP Bench Briefs*, June/August 1981.
"Functional Analyzing"; *HP Bench Briefs*, Jan/Feb 1982.
Digital Troubleshooting, by Richard E. Gasperini; Hayden Book Co., 1976; 192 pages; $11.95.
"How to Troubleshoot Digital Circuits," by L. Steven Cheairs; *Radio-Electronics*, November, 1978.

EXAMPLE 1: AMPEX ATR-700 QUARTER-TRACK TAPE MACHINE WITH TRANSPORT FAULT

Trouble or Sympton: Machine does not stop when the Stop button is pressed.

Troubleshooting Procedure: The first thing to check is the transport control modes for correct operation. Will the tape machine stop after any previous command has been engaged? Does the machine enter stop mode initially after the unit has been turned on, and the tape has been threaded? In our example, the transport will stop only after power is turned on! As soon as any other transport command is issued, the transport has to be put into Pause mode in order to stop the tape movement. Also, an audible observation points out that the familiar "clicking" sound of the brakes engaging is not heard during Stop commands.

Based on this knowledge, the first troubleshooting venture should be done on the transport control circuitry. The assumption is made that all power-supply voltages must be correct, since other transport features are optional, and a quick check of the play/record capabilities also looks correct. The transport control schematic and associated information should now be consulted prior to any dismantling procedures. The theory of operation and schematics in this manual shows that the first convenient test point, using the half-splitting troubleshooting technique,* should be the Brake Solenoid Drive Signal of the output of IC11 pin 6 (Figure 1). The useful timing diagrams provided within this manual show that the logic signal at this test point should be a "1" whenever Play, FF, Rewind, or Record Play commands are issued. In the Stop or Pause modes, the output should be "0," which causes the brake solenoid to be not released, and thereby keeps the reel motor brakes still engaged.

Now the control unit PCB is accessed, and the Logic Probe power leads attached to any convenient +5 VDC (TTL supply voltage) and ground. The outpost of this gate is checked for proper logic levels, with the probe tip placed directly on pin 6 of IC11. The transport is placed alternately in Stop and Pause. The probe indicates a logic 1 at all times, except during Pause commands, when a logic 0 is present; the Stop command "0" is missing! (Note: If a logic was correctly displayed during Stop, then the next test point would have been in the middle of the analog control circuitry, located on the power-supply board. This diagnostic procedure would then be consistent with the half-splitting troubleshooting technique.)

The next test point should be any one of the motion flip-flop circuits, which is at a midway point on the just tested faulty half-section. With the logic probe on IC4 pin 6, the device indicates that a logic 0 occurs during Play, Pause and Stop commands. The timing diagram, however, shows that a logic 0 should *not* be present during Stop, so the fault still has not been located.

The third test point is the output of IC2 pin

*In the technique of half-splitting, a check for proper operation is made at the most convenient test point midway between any remaining untested circuitry. The tested half-section that still operates incorrectly is then divided for the next test point. In this manner, the fault is eventually pinpointed. In the above example, the brake solenoid circuit consists of two PCBs — the control card and the power supply card. This first test point (IC11 pin 6) occurs halfway at the interface between these two cards.

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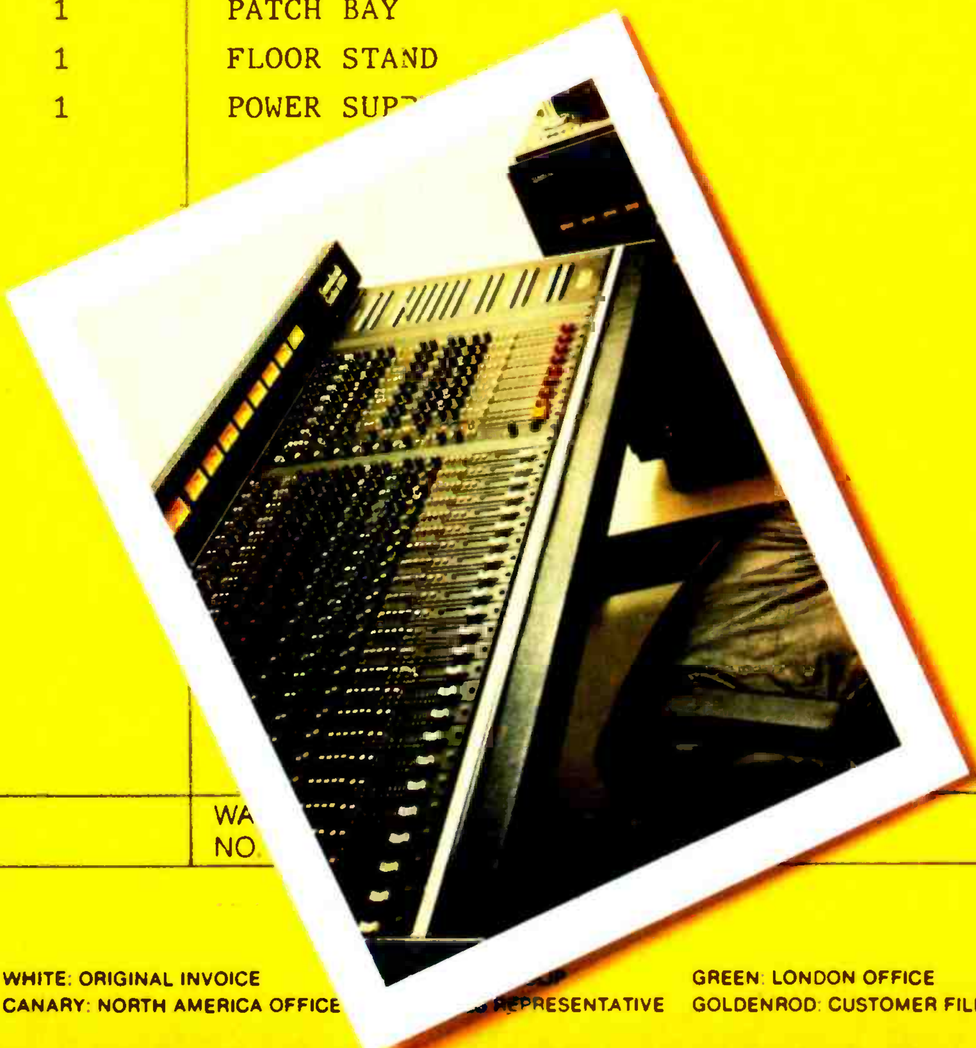

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LOGIC TESTERS

8. This IC feeds all the motion command flip-flop reset circuits. While pressing the Stop button, the probe displays a constant logic 0 at the output pin 8 of IC2. The input to this gate (pin 9) is checked next for valid levels during Stop mode, and this test indicates correct logic levels being fed from the Stop button. Which means that there is either an open circuit at pin 9 input, or pin 8 input.

One more check should be made to completely verify the isolated fault. The Logic Pulser is activated in the same manner as the Logic Probe. The pulser tip is placed at the other input (TP5 of Figure l) to the OR gate, IC2 pin 10, and the probe at the output (pin 8) once again. A single pulse from the pulser provides a logic 1 which immediately resets the motion flip-flops and engages the reel motor brakes.

Conclusion: Pin 9 of IC2 is an open circuit; IC2 (SN7400N) needs to be replaced. ■

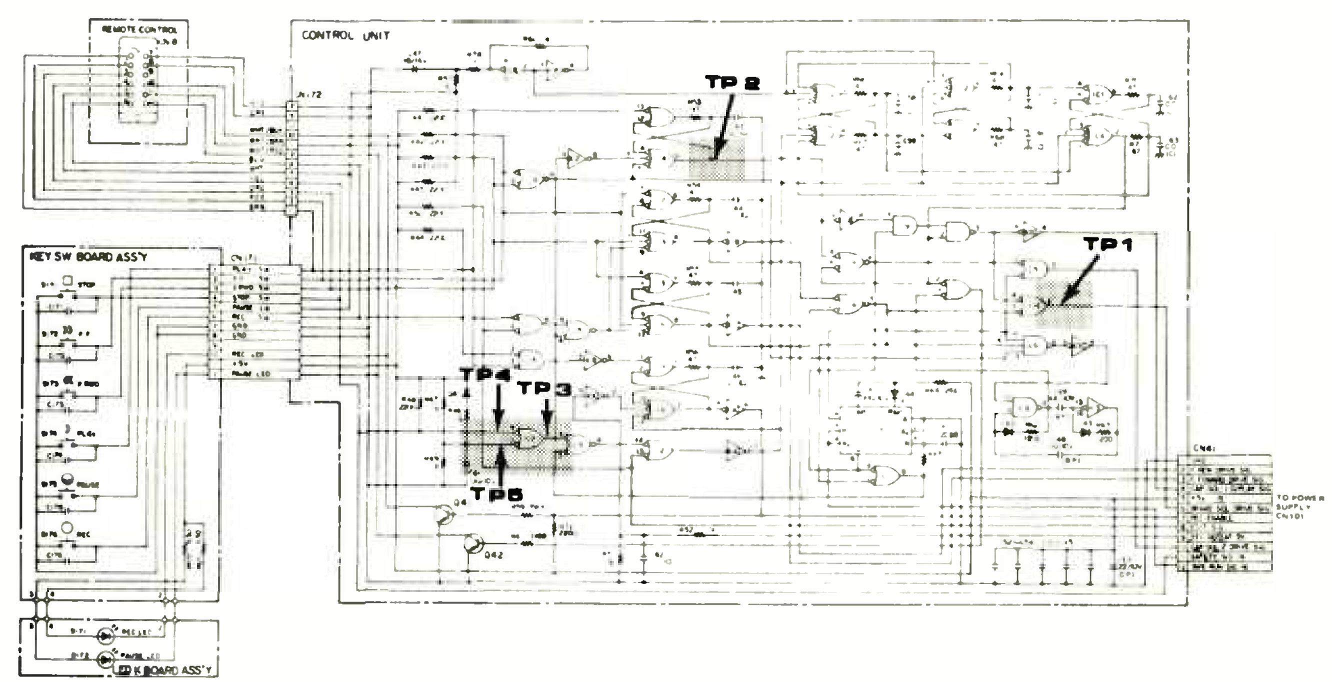

Figure 1: Ampex ATR-700 quarter-track control unit schematic. Test Points are numbered in sequence used to isolate the fault in Example 1.

EXAMPLE 2: EVENTIDE 1745M DIGITAL DELAY

Trouble or Symptom: Noisy, buzzing signal on output at all input levels and delay setting.
Troubleshooting Procedure: One of the first things to check is to see if the buzzing signal is present on one or more outputs, since this information will provide us with a clue as to what section is the most likely to be at fault. In this example all outputs exhibit the same problem. Individual output cards can be ruled out as the problem source. Also, since the noise is not level dependent, we can expect that the input audio processing circuitry is also functional.

An oscilloscope shows that a 1 kHz line-level sinewave at the input to the DDL with a 0 millisecond delay setting produces a clean sinewave output, except for a periodic shifting image that resembles a gated delay of the signal. This symptom is indicative of an addressing control problem. The service manual points out that the addressing and memory circuitry is located on a large board behind the motherboard; easiest access is through the back panel. Once the metalwork has been taken off, the screws that hold the board to standoffs can be also removed.

The memory organization for 320 millisecond of delay requires 16K of RAM in the design. This amount of memory requires 14 address lines, A0 to A13. A0 to A11 directly

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STUDIO MAINTENANCE AND TROUBLESHOOTING

LOGIC TESTERS

interface to the 12 required address lines of the 4K RAMs in the memory matrix. The additional two addresses, A12 and A13, are handled by a CMOS BCD-to-Decimal Decoder, IC23 (74C42), which provides four outputs that select IC sections between four different groups within the two RAM arrays. These four chips select signals, CS1-4, go directly to the arrays without any additional buffering. This is a good initial test point.

The Logic Clip is placed on IC23 (Figure 2), and its power leads connected between +12 VDC (CMOS supply) and ground. As expected, we observe the corresponding decimal outputs, 0 to 3, go L0 as the inputs A and B sequence binarily from 00 to 11. Using the clip again, we probe IC21, 22, 28 and 29 (74C904), which are non-inverting CMOS buffers used to drive the large number of address lines in the arrays. IC21 and 22 control address lines A0 to A5 and A6 to A11 respectively in array #1, while IC28 and 29 command the same lines in array #2.

For properly functioning circuits we should see simultaneously lit LEDs on the inputs and outputs of each buffer in each IC. This test reveals a discrepancy at pin 13 of IC22 in array #1; the level at this pin remains HI at all times.

The Logic Probe is attached next to this same point. The probe, being a dual-threshold device that detects valid LOs and HIs, also remains lit at all times. However, it does dim periodically in sync with the HI signal at the input (pin 12) to this IC.

The pulser is now placed at the trace leaving pin 13 output, and the probe is placed at the nearest A0 address line on pin 8 of any RAM in array #1. The probe gives the proper indication when the pulser is activated.

Conclusion: IC22 (74C904) needs to be replaced, due to an invalid LO fault on pin 13 output. ■

EXAMPLE 3: AMPEX ATR-102 TWO-TRACK TAPE MACHINE WITH TRANSPORT POWER SUPPLY FAULT

Trouble or Symptom: Machine has no display; tape will not thread up.

Troubleshooting Procedure: Since there is no display or any tape motion, the absence of one or more supply voltages is suspected. The ATR-100 service manual shows that the +5 VDC supply line, which is separately fused, powers all the LEDs and transport control circuitry. A quick check of this supply voltage reveals a 0 VDC measurement. A continuity test of the fuse shows that it has blown. As soon as it is replaced, the new fuse blows when the tape machine becomes powered again. Apparently one or more of the circuit cards that contain +5 VDC powered circuitry is shorting out the power supply.

The Audio Control and Transport Control cards are removed first, one at a time. With the Transport Control card removed, the +5 VDC supply remains stable. Reinserting this card once again blows the fuse; the faulty card has been found! A resistance check across the supply lines of the card gives an indication of a complete short. A visual and olfactory inspection does not reveal any burned or discolored ICs and PCB traces, so we are left with the problem of finding out which of the 39 ICs and assorted capacitors are defective.

Typically, the following techniques have been tried to solve this kind of perplexing troubleshooting situation. An external +5 VDC power supply with current limiting is connected directly to the board's supply traces. While the current is slowly increased, individual ICs are checked by touch for an increase in heat. Sometimes the PCB traces discolor, lift off the board, or even burn up. This drastic procedure can definitely pinpoint the defective IC, especially if there's a burn mark surrounding it!

Another common procedure involves removing all the supply by-pass capacitors (large and small) from the circuit. The most convenient way is usually to lift one lead of the capacitor from the circuit board. If the short circuit has cleared after this procedure, then each capacitor is placed back into the circuit one by one until the short re-appears. If the short circuit has *not* cleared after such a procedure, then another frequently-used technique is to remove the individual ICs from the circuit. More often than not though, the board has soldered, non-socketed components, so IC removal will usually involve cutting the power-supply traces first to sections, and then to individual ICs. Such a time-consuming method will eventually isolate the bad circuit but, because trace-cutting technique is a destructive process over the lifetime of a board, it should be avoided as much as possible.

Finally, a last resort solution (besides scrapping the board) is to ship the defective unit back to the manufacturer's repair facility. This method can be quite costly, as the required turn-around time for a repair becomes critical.

All the processes discussed above can be avoided with the use of a Logic Pulser and Current Tracer. With these test instruments, the faulty IC can be located without applying power to the board, since pulser and tracer are powered by an external DC supply.

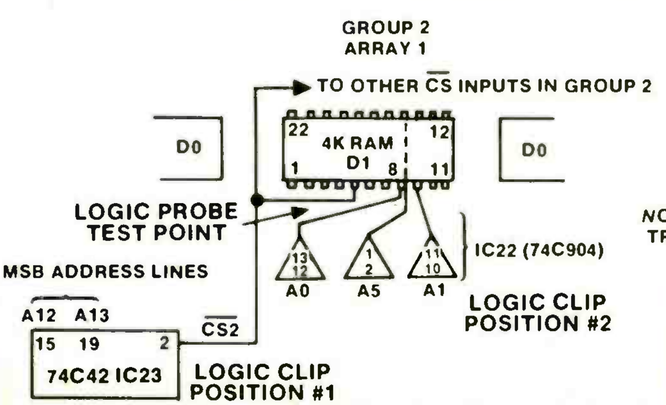
First, all electrolytic capacitors located across the supply traces should have one of the leads disconnected from the circuit. The pulser, set in a continous pulse mode, is placed at the +5 VDC trace emitting from pins 1 and A in the upper right corner of the PCB. (Note: A conventional two-sided logic board usually places the supply voltage traces into an interleaving, comb-like grid on one side of the board, while the other side will usually contain all the logic signals.)

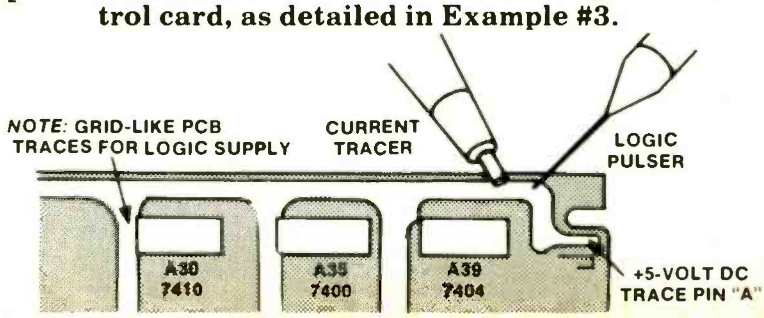
The Current Tracer is then placed near the pulser, and the sensitivity adjusted for a half-brilliance indication of over 100 milliamps. The pulser is moved along the large trace horizontally to the left from this first test point (Figure 3). We notice that the tracer LED is extinguished after the third vertical intersecting trace. This vertical column consists of IC A25 to A30. The pulser is moved to this intersection point, and the tracer now shifted from the new test point down the vertical supply trace.

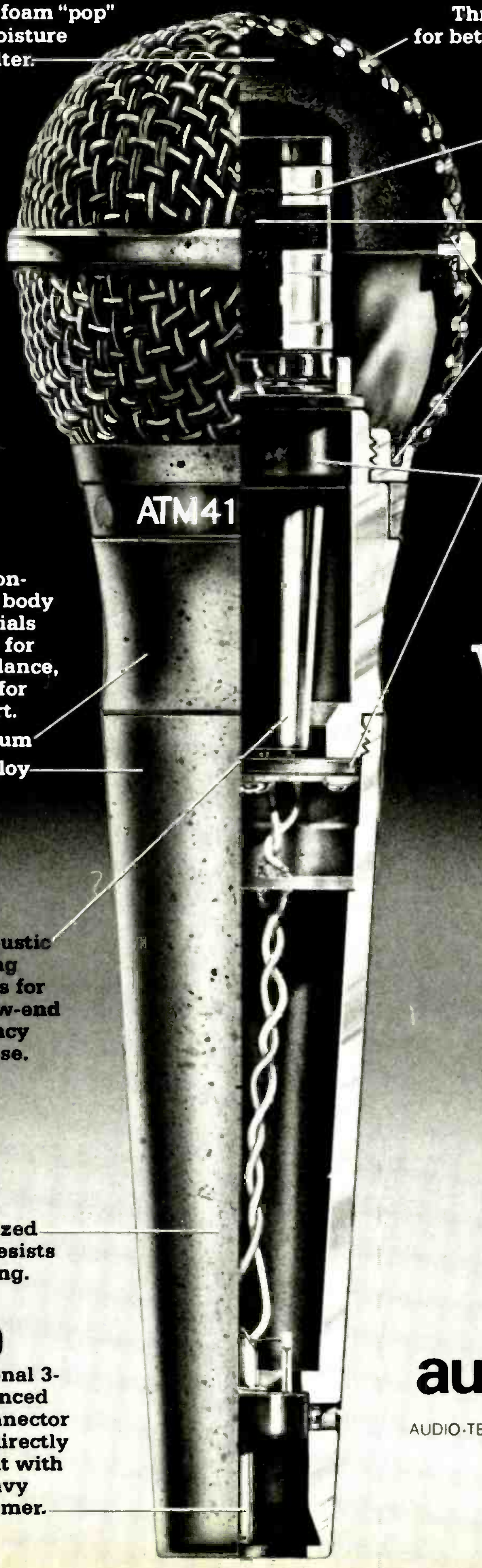
We next notice that the LED is extinguished again when IC A27 is passed. To verify that IC A27 is the faulty device, the pulser is placed on pin 14 (+5 VDC supply), and the current tracer moved away from that point. The LED remains extinguished even when the sensitivity control is reduced to the 1 milliamp range.

Conclusion: IC A27 (SN7400N) needs replacement due to massive internal circuit failure, resulting in short circuit of voltage supply. ■■■

Figure 2 (*left*): Partial schematic of Eventide 1745M DDL, showing test positions of Logic Probe and Clip detailed in Example #2. Figure 3 (*below*): An illustrative example of how a Logic Pulser and Curent Tracer can be used to track down a +5 VDC-to-Ground short circuit on an Ampex ATR-100 Transport Control card, as detailed in Example #3.







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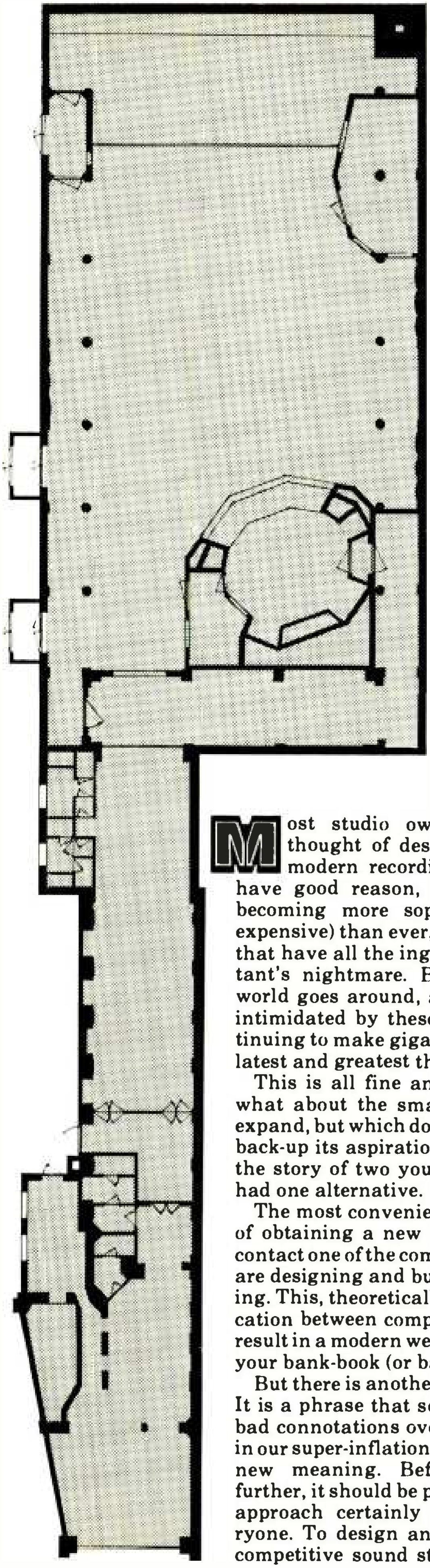

STUDIO DESIGN AND CONSTRUCTION

EASY SOUND STUDIOS

An Unusual, Low-Cost Conversion of a Movie Theatre into a Multi-function Recording Environment

by David Rideau

Most studio owners shudder at the thought of designing and building a modern recording facility. And most have good reason, what with equipment becoming more sophisticated (and more expensive) than ever, and construction costs that have all the ingredients for an accountant's nightmare. But still the recording world goes around, and many refuse to be intimidated by these obvious threats, continuing to make gigantic investments in the latest and greatest that money can buy.

This is all fine and very admirable, but what about the smaller studio longing to expand, but which doesn't have the money to back-up its aspirations? This article relates the story of two young studio owners who had one alternative.

The most convenient and reliable method of obtaining a new recording facility is to contact one of the competent audio firms that are designing and building studios for a living. This, theoretically, with good communication between company and client, should result in a modern well-functioning studio. *If* your bank-book (or bank) is behind you.

But there is another choice; Do it Yourself! It is a phrase that somehow has developed bad connotations over the years, but which in our super-inflationary times seems to have new meaning. Before progressing any further, it should be pointed out that such an approach certainly doesn't apply to everyone. To design and construct a modern, competitive sound studio requires a strong background of acoustics in theory and practical application; a knowledge of construction techniques; and, most of all, the *will* to work.

This last pre-requisite is, by far, the most important qualification. If you ask one of the many who have travelled down the long and lonely road of the self-constructed studio, you are sure to find hard work as the one common denominator. No matter how well prepared you are with your endeavor, you will most certainly come upon the unexpected delay. And, once you're behind schedule, only long hours and/or working through the weekend can save you!

The Germ of an Idea

This writer first met Easy Sound's co-owner Niels Erik Lund when we were both working for Westlake Audio, in Los Angeles. As part of the construction crew that was in the process of building Westlake's existing recording complex on Beverly Boulevard, we were two young aspiring engineers glad for the chance to sweat 10 hours a day. It was, we considered, a small price to pay for a theoretical and practical lesson learned from a veteran studio builder. Following completion of the two Westlake studios, we both continued with the company as second engineers.

Lund and I became good friends during his year-long stay in the US, and he often spoke of my coming to work in the studio that he and his brother Henrik owned in Denmark.

. . . continued overleaf —

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Finally, the day came when he had to leave the States, to get back to help his overworked brother.

A year later I received a phone call from the Brothers Lund. "How would you like to come over and make some Danish hits?" asked the distant and distorted voice from over the line. "Sounds groovy," I said thinking at the same time what life would be without LA's smog, and the Santa Monica freeway during rush hour.

Little did I know that I would still be around two years and one studio completion later.

When I first arrived at Easy Sound, it was a small but well-equipped studio located in the basement of their parent's house in the suburbs of Copenhagen. Situated half a block from the beach, the studio possessed a really charming recording atmosphere; a home-cooked Danish meal often was waiting for you above after your daily session.

But the location did have its drawbacks, limited space being number one. It was comfortable for smaller pop and jazz dates, getting cozy with large group and string dates (Mrs. Lund often forfeited the use of her dining room for the latter operation), and impossible for more than 20 musicians.

Then the inevitable finally happened — time to find a larger, more flexible location. But, with recording in the US and the rest of the world in a decline, any modern studio must be *more* competitive than ever before. The Brothers Lund realized this, and took on a "diversify or die" attitude.

Choice of Site

First, it was decided to stay in Copenhagen. Considered by many to be the "Gateway to Europe" and a really charming, almost fairytale city, Copenhagen is a major stop for groups during more lengthy European tours. It is also home for the two brothers, and they have established many steady clients (a dying breed) that they would stand to lose by moving to another area in Denmark.

The next requirement was a more central location. It's strange for a former resident of LA to hear, but the old studio was considered "out of town" — even though it was only a 20-minute bike ride (a major source of transportation) from the city's heart.

Also, the "shell" for the new studio should not be cost prohibitive. The bottom line was obvious: they could only afford to pay so much.

The one thing that can be dangerously overlooked, however, is what we refer to as "The Rebuilding Factor": how much you will have to invest in basic isolating construction costs. That is to say: what's the noise level of the surrounding environment of the structure, and its ability to impede the noise coming from this environment. Isolation is expensive, and could be the most important point of studio construction. (Of course, it is also equally important that the surrounding environment is protected from the sounds being emitted from a working studio.)

We all agreed that it would be most advantageous at this point in time to invest in a larger structure, for several reasons. First, Video is here to stay. Anyone who won't admit that the visual medium is rapidly having a greater influence on modern society, and on our industry in particular, is not being realistic. A recording studio that has capabilities for handling video sessions and shoots has a much better chance of survival.

Secondly, we were also very interested in accommodating larger session work, particularly of the classical variety. We had done smaller-scale classical recordings in the old studio, and had grown to realize that the close-miking method could never capture the full body and tone of a group of instruments as effectively as the mike placed some distance away, where the sound can be allowed to "develop" (the combination of a balanced sound source, and the correct acoustical environment). Obviously, the traditional small and heavily trapped studio is not the place for this type of recording.

The last point of our studio "shell" was to have adequate facilities for the concert/live recording situation. This seemed to be a particularly wise field of endeavor, due to the fact that it could be useful to many Danish groups who don't have a large enough budget to spend several weeks recording and mixing an album. Instead, they could use two concert evenings, with a few added dates for "repairs" and mixing. The group's share of ticket sales could also be used to offset recording costs.

In theory, all these points were valid, but where does one find such a location out in the real world? The one thing on our side was time. With the existing studio in operation, there was no real time limit. But, although they would never admit it, I'm sure Niels and Henrik's parents were looking forward to getting their house back from the rock and roll world!

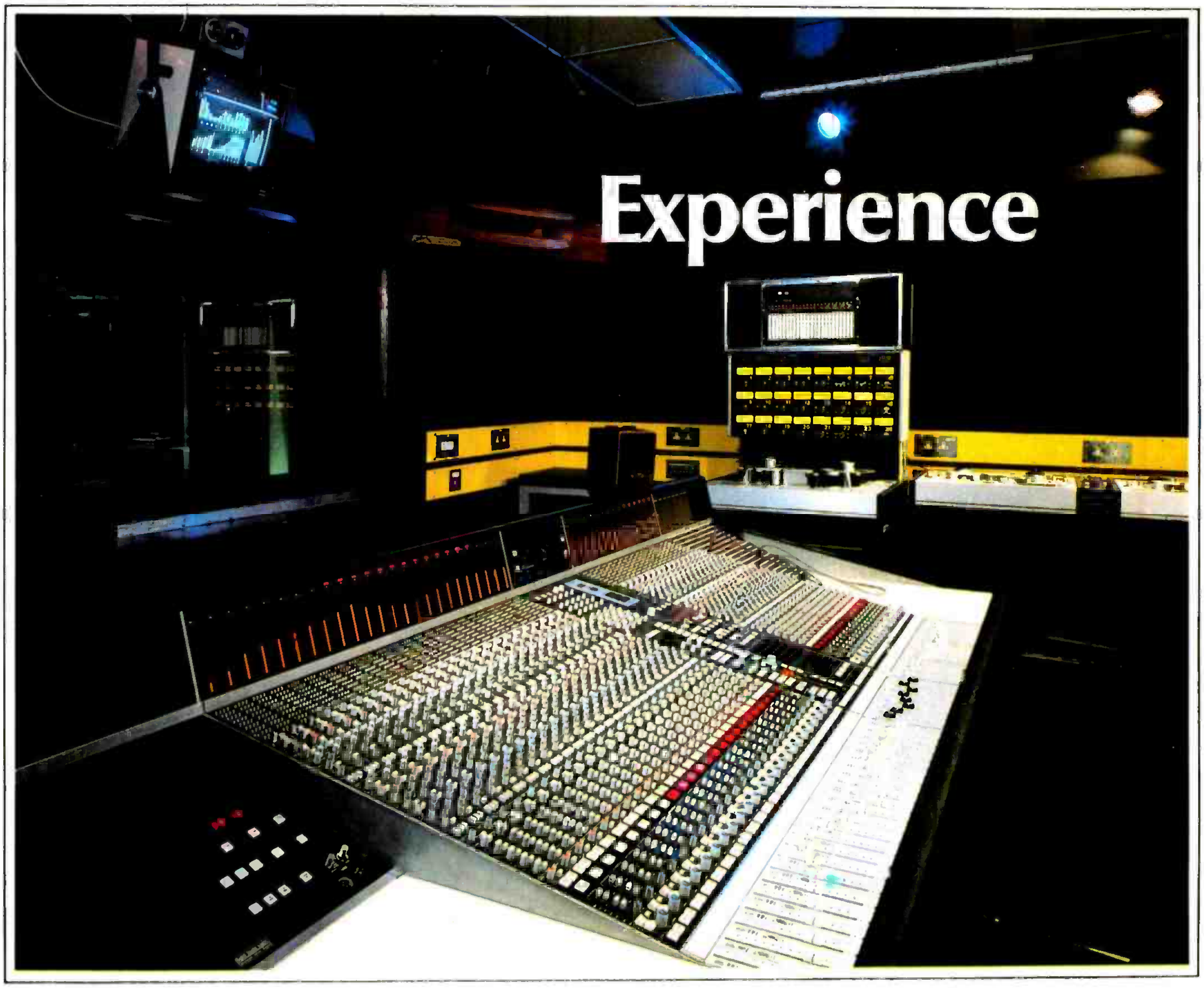
Over the next few months many prospective sites were investigated — large homes, shipyards, a steel warehouse — but nothing that could meet all of our requirements. Then, one day while driving through one of the city's more busy intersections, Niels Erik spotted a "Til salg" (For Sale in Danish) sign on an old film theatre. The location proved to be almost ideal: in the middle of town; close to a major intersection; but at the same time well isolated from traffic.

But what about the cost? The Lund Brothers decided they were interested in buying, as opposed to a long-term lease. Then came the bad news: $430,000, and it seemed like the owner wasn't interested in coming down in price. It was a letdown, because the price was definitely out of the ball park in relation to our projected budget.

The realtor took us to see one other theatre for sale by the same owner, but it was not nearly as nice, and the location

Easy Sound co-owner Niels Erik Lund surveys the half-completed studio, just after ceiling treatment had been completed. In rear can be seen a ½-ton steel I-beam supporting most of the inner control-room ceiling weight, prior to completion of the surrounding walls

Photo: JESPER WANTING



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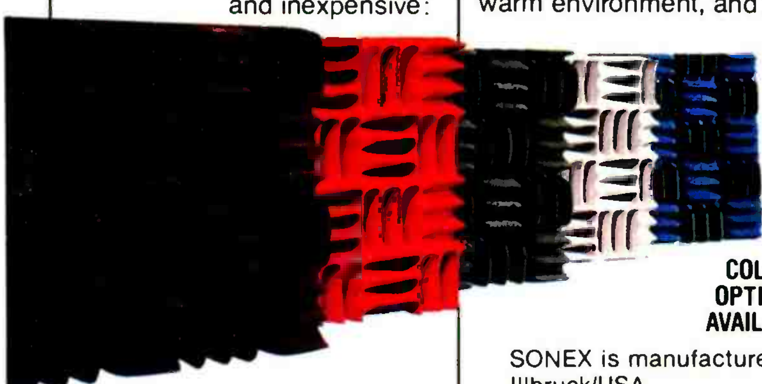

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in town was far from ideal. So it seemed that we were back to square one, checking the daily paper and other sources of realty information. This continued for almost six months, until we received another call from the same realtor who had shown us the theatres. It turned out that the owner was ready to come down in price.

As we found out later, the theatre in which we were interested had been actually sold to a supermarket chain for the previous asking price, only to be rejected later by the county zoning committee. After several weeks of bartering, all parties agreed to a $210,000 selling price; less than half the original asking price!

The building shell was more than we had ever hoped for. A well-constructed theatre built in 1926, and which originally seated 700 people, the facility had only been closed for 18 months and was still in good condition. It was also equipped with a snack-bar, kitchen, four toilets, and a lobby. All of these fixtures, which worked in well with our projected studio traffic pattern, eventually saved us thousands of dollars in construction costs. Unfortunately, the film projectors had been sold in advance, but the wide screen was still in perfect condition, and opened up a possibility that neither the two brothers or myself had even considered — film scoring.

One thing to note at this point is that we are working with a limited construction budget of $50,000, which was also to include the purchase of a new set of studio monitors. To work within this budget, most of the design and construction would have to be handled ourselves.

The Lund Brothers worked out the basics of the control-room design one long night, using their imagination, and two chairs situated in the middle of Henrik's living room. Later their sketches were checked against a few other traditional control-room designs, and it was uncanny how close they had come to details, such as monitor placement, using only common sense and past experience.

They next took the sketches to Thomas Sheel, a friend who was in his last year of study as an architect in the Danish Royal Academy, and had experience in studio construction methods through the various expansion and remodelling projects of the old Easy Sound studio. He would now do the neccesary drawings needed to apply for a construction permit, and for zoning office approval.

While waiting for the necessary clearance to begin construction, we began preliminary reverberation time measurements of the room with the following rented Bruel & Kjaer instruments: 4133 microphone; sound power source type 4205; rotating microphone boom 3923; 4417 building acoustic analizer; and alphanumeric printer 2312. Using this measurement equipment, several points in the theatre were checked. The 4417 is one of the latest additions to the B&K family of measuring instruments, and controls both the sound source and the rotating microphone boom.

The results were very encouraging — an RT60 of 1.3 seconds at 500 Hz, and a smooth "musical" curve going from an RT60 of 2.0 seconds at 100 Hz to an RT60 of 0.58 seconds at 8 kHz. Of course, these numbers would all change as we began the first stage of the studio's transformation, that of removing the theatre seating. It did prove the room had the potential to sound good, however, which was the object of the exercise.

The Copenhagen Connection

(...Or how to get the best use from your programmable desk-top calculator)

Here in Denmark we are very fortunate to have truly the "first in sound and vibration," Bruel & Kjaer, literally on our doorstep.

Soon after the building for Easy Sound Recording was purchased acoustical measurements began, using exclusively B&K instrumentation, Determining the overall frequency characteristics of the proposed studio area down to 100 Hz was a relatively simple operation. But what was happening below this frequency was vital information, especially for a room of this size.

The room could be tested with a fair amount of accuracy using a method that utilizes a digital real-time parallel analyzer and a desk-top calculator, but the technique is both a lengthy and complicated operation. It then occured to us that it might be possible to calculate the needed information using the wave theory of room acoustics.

This method, based upon the motion of sound waves within a three-dimensional enclosure, is characterized by establishing boundry conditions that describe mathematically the acoustical properties of the walls, ceiling, and other surfaces in the room. Of course, the more irregular the room dimensions, the more difficult it is to calculate, but we were basically dealing with a (large) rectangular room.

When employing the wave theory, a room is considered as a complex resonator containing many normal modes of vibration, each mode having a characteristic frequency of damped free vibration. These modes can be exited by introducing a sound source in the room. The acoustic energy supplied by the source can be considered as residing in the standing waves established in the room.

The characteristic frequency of vibration of the standing waves depends principally in the room's size and shape, whereas the damping of these waves depend mainly on the boundry conditions. Using the simplest possible boundry conditions of rigid walls and no damping, we could calculate the room's characteristic frequencies. The acoustical treatment could then be prescribed accordingly.

This all sounded just wonderful to me while doing the research. But what about application? I made it as far as calculating the axial modes (one-dimensional), and realized I would commit Hari-Kari before correctly calculating the oblique (three-dimensional) modes.

Conceding that maybe I had gotten in over my head, I decided to talk to someone else more qualified. I called Eduardo Fayad, a Brazilian-born studio designer and owner now based in Los Angeles. As somebody who has designed and built studios in the US and Brazil, and has written specific computer programs employing the wave theory of room acoustics, Fayad was truly "what the doctor ordered."

It should be borne in mind that we were operating on a strict budget, and round-trip airfare, not to mention the fee for a man of his talents, were out of the question. But Eduardo is a good friend, and he had made up his mind to help us. Even though we felt it was too much of a personal investment, we were glad to have the help.

On the first day of Eduardo's stay we decided to take him out to the B&K facility. Over the past months we had come to know many of the company's staff that were always more helpful in showing us how to get the best results from their measuring equipment. In turn, they have used our studio for field testing prototype microphones.

After discussing Eduardo's programs over lunch with Stiv Sommer and Ole Brosted, two employees in the B&K research and development department, Brosted suggested "we should pass by and say hello to Holger"; he was speaking of Holger Larsen, one of the most respected names in his field.

The meeting with Larsen proved to be brief, but profitable. For 10 minutes he and Fayad compared computer programs and spoke the same language. As it turned out, Larsen had a program quite similar to the one Fayad planned to apply to our room, but with one degree more of sophistication (for further details, see Reference 2). Larsen immediately gave him a copy of his program lay-out, and offered us the use of his calculator for our application. Larsen furnished him with the neccesary data, and 50 hours of calculation later we picked-up the results.

Eduardo Fayad then took the results, formulated where our problem areas were and, through the use of polycylindrical diffusers and slot resonators, designed a workable solution to our problem areas. ■■■

The Conversion Process

Both brothers had made the habit of checking the classified section of the daily paper for items that might aid out construction process. Not long after we received the go-ahead to begin construction, Neils Erik spotted a lumber sale ad. An apartment building was being torn down on the other side of town, and the demolition crew was offering great discounts on used lumber. There was one catch, of course, — you had to remove all the lumber from the building yourself.

What first seemed to be the hard way to get "B" grade lumber turned out to be a good deal. The general quality of the wood was very good and, since we were literally tearing it out ourselves, it didn't have nearly as many battle scars usually associated with used lumber. And, best of all, we could select for ourselves the type and size of wood we wanted.

After four days of hard labor we had nearly half of the wood we would eventually need to complete the project. Most of these were larger beams that would be very expensive to buy new, and were much easier to remove from the building without their being damaged in the process.

The general plan of attack basically was to build a control room within the theatre, which would then become the studio area. Niels Erik and myself would be the main construction crew, augmented by friends and family when they had the time and will for hard work. This left Henrik to run the existing studio alone, which meant double sessions regularly. Still, he often would come by to help us.

Seat removal was the first step, which proved to be quite a chore. Each chair was bolted to the floor in four places, and had to be loosened from the small crawl space underneath. After the seats were removed, the reverberation time was measured again. As we had more or less expected, the RT60 was about 1 second longer over the entire frequency spectrum.

At this point we were in a slight dilemma. The ceiling had a series of plaster beams with ornamentation that were spaced evenly apart, and spanned the width of the room. We wanted to

Photo: DICK NEIL

View from performance stage area along the length of the studio recording area to the corner control room. Clearly visible is the acoustic treatment applied to the unusually high and ornate ceiling. The treatment consists of V-shaped slabs of Rockwool "Rockfon" insulation suspended on cotton belts to save the cost of contruction a supporting frame.

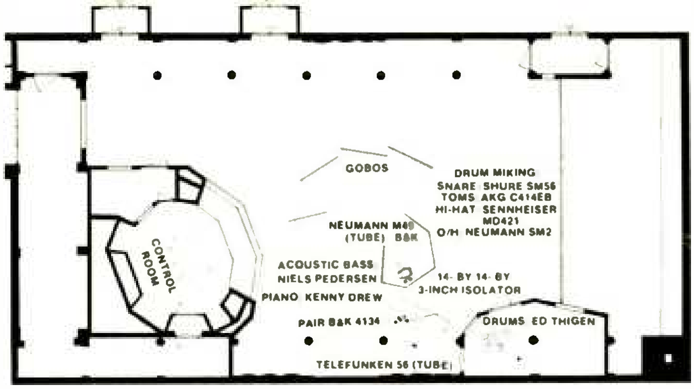
avoid covering this ornamentation with the lowering of the entire ceiling, also a very costly operation. Rockwool International was consulted on the various possibilities of its insulating materials, and the company offered us the service of in-house acousticians to calculate the results of any treatment that we might have in mind.

After two weeks of deliberation, in which time we also constructed a scale model of the ceiling to experiment with the visual effect of the various applications we had in mind, we reached a workable solution. "Rockfon", a compact, rigid-type insulation with 4½ pounds per cubic foot density, was selected, and hung between the plaster beams in a "V" configuration, leaving the ornamention exposed.

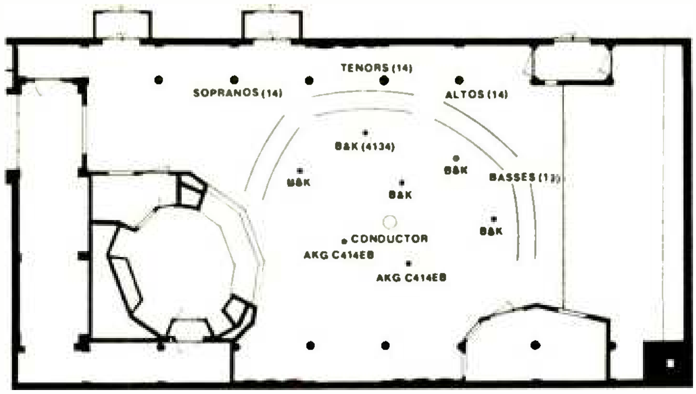
The cotton "belts" supporting this free-hanging configuration could be attached to the beams easliy, thereby saving the cost time of constructing a frame over the entire ceiling. The insulation material could also be ordered in several colors, but its natural state was chosen. This turned out to be much cheaper, and a nice contrast to the newly painted white ceiling.

After application of the acoustic treatment for the ceiling, our measurement gear was brought in for the third time. The results were fine, with an RT60 of 0.8 seconds of 500 Hz, and the overall curve relatively free of dips and peaks.

Isolating the studio area from the outer environment was one of our more painful endeavors. Although its main entrance faces a well-travelled street, the theatre itself is 130 feet from this source of noise and vibration. The ceiling is also well isolated with two separate concrete layers, and enough air space for someone to stand between the two. Running the length of the theatre are two air channels enclosed between these two ceiling layers. These in turn are connected at mid-point by a third channel, creating an "H" design. In the middle of the connecting third channel is a vent that opens to the roof. This vent



KENNY DREW TRIO SESSION LAYOUT
EASY SOUND STUDIOS, MARCH 1982



ROYAL THEATER OPERA COMPANY
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is fully adjustable, and for the most part only partially open. But, as an added precaution, the air channels have all been lined with a semi-rigid insulation.

The basement area is especially well isolated and, as we found out later, was an ice factory built before the turn of the century.

All of these factors added up to a background noise level of 22 dB before we had even touched one roll of insulation material!

What was more our concern, however, was the noise (in the form of music) we could be sending out to all our neighbors. Transmission-loss measurements through the walls proved that we could eliminate the problem by further insulating the side doors which served as original exits for the theatre. This would be done through the construction of external sound locks.

Control Room Foundation

Part of the art of building a studio yourself is knowing when to seek outside help, but at the same time keeping expenses down. An engineer was found to calculate the support requirements of the foundation, and the major supporting beams of the control room. The engineer had extensive experience with special construction problems, and wasn't intimidated by our unusual requirements.

The theatre floor was stripped down to its original foundation at the point in the structure where the control room foundation should be laid. The main supporting structure was then built resting entirely on thick rubber (somebody came up with the idea of using old pieces of car tire for this function), on which a tongue and groove floor was constructed (Figure 1). Several layers of compact insulation were then laid upon this floor, with heavy plastic covering the final layer.

After constructing a form in which to pour the concrete, several layers of steel reinforcing rods were laid upon one another, to avoid cracks and fissures that can occur when concrete is resting on this type of isolation pad.

. . . continued overleaf —

EASY SOUND STUDIO, Denmark

The first isolation room directly adjacent to the control room was also poured within the same form, and "floating" in the same manner. But, at the same time, there is also a complete break in the foundation between the two, with a double layer of heavy rubber ensuring discontinuous construction.

Soon after the concrete had set, the control room wall construction was started. The bottom plate of every wall also rests on a thinner piece of machine rubber, as the final guarantee of attenuating the transmission of unwanted acoustical disturbances.

The control room actually consists of two completely independent structures: one resting on the studio foundation; and the other, enclosed within, resting on the "floating" foundation. The majority of walls consist of a single stud construction, both sides being covered with a sandwich of gypsum board, ½-inch, two pounds per cubic foot density insulation, and gypsum board again (Figure 2). In addition, the inner walls were filled with six inches of insulating material of the same density. The exception were several outer walls, which followed the same formula with the addition of staggered studs.

The second isolation room is also of the floating variety, and has a special double-door construction to allow for passage of the studio's concert grand piano.

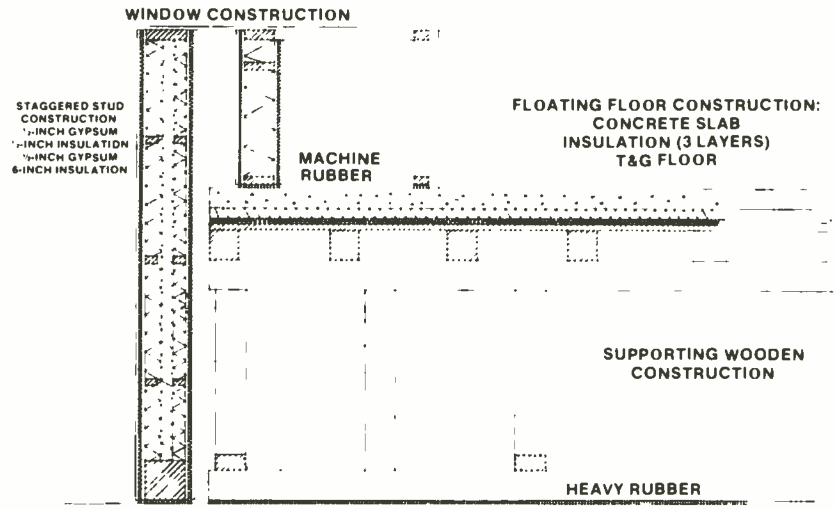

FIGURE 1

When Niels Erik did the layout of this room, he actually had the stage wall following the sight line from the projection booth windows to the screen. As a result, when we show a film using the entire screen, it will be just inches from "hitting" the isolation room's stage wall, utilizing the maximum space available. The same room, divided by a supporting pillar, has a heavily trapped area and a more live side with minimal trapping. The third and smallest isolation room doubles as a sound lock to the door, opening outdoors in the stage area.

The stage, which lies directly beneath the screen, offers the control room an excellent view of its entire width. The supporting structure also rests upon heavy rubber, isolating it from the studio floor. Beneath the stage is a large storage area designed to contain the seating for concerts when not in use.

Just prior to beginning to finish work, Henrik joined us full-time. Seeing completion of the new facility drawing nearer, it appeared that all of our clients had stopped booking time into the existing studio, choosing to postpone their respective production schedules to coincide with the grand opening. This was fine, of course, since we would be guaranteed work upon completion. Unfortunately, it also stopped cash flow at a critical time.

It was a clear case of "time is money", so the boys decided to hire two professional carpenters for the final push. It was also the perfect time too, with such delicate tasks as door frames, stair construction, and finish woodwork ahead. Though by this time we were all "semi-respectable" carpenters, there were still some things that were better left to skilled hands.

One final used purchase saved us time, and hundreds of dollars. It was another ad spotted in the paper, where a bananna warehouse was selling a set of six freezer doors. We went down to see them and it was just what we needed —a set of six doors complete with jams and hinges. Although they were old, the basic design, in which a large handle is closed and gradually tightens until you have a perfect seal, was perfect for our application.

Once the doors had been taken to the theatre, we began the process of giving them more mass. Tearing off the inside panel to expose the original loose insulation, this material was disgarded leaving only the hollow door. Layer after layer of gypsum board and ½-inch insulation was then added, finally ending up with three layers of each. The door was then recovered with a sheet of

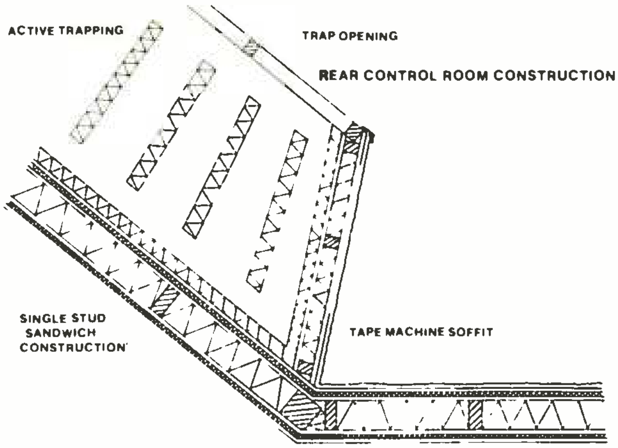

FIGURE 2

½-inch plywood. This made for a *very* heavy door, but the heavy-duty hinges could easily stand the load and, when properly mounted, could be opened easily by even a small child.

Monitor Selection

By now the new control-room monitor system had arrived. This was one new piece of electronic equipment to be purchased for the new location, having most everything else we needed, and the decision of what make and model to use took some time.

The first requirement in selecting the monitor system was accuracy; not only at medium and high volume, but also at lower levels. The second consideration was power. We wanted to be able to play reasonably loud, and still have a respectable amount of headroom. We finally agreed on the Westlake HR-1 Phase Coherent, Four-Way Monitoring System. Not only did it meet the previously mentioned requirements, but Niels Erik and myself had both worked extensively with the system.

We had now been working intensively for five months, and 14-hour days, or longer, were becoming more common. But the end was in sight. Niels Erik has begun to start wiring, while the rest of us forged ahead with finish work.

Glass: A Critical Detail

Glass is another critical point in the studio that nobody should ever attempt unless they have extensive experience in that area. It was a heavy job for us in particular, since we chose to have three layers in the main control-room window. The layers varied slightly in thickness (0.31, 0.39 and 0.47 inches) to avoid any common resonant frequency.

Of all the construction dilemmas heard from studio owners, the installation of glass is somehow the most mentioned. We were lucky to find a fast, efficient company that installed a total of 25 pieces without a single complication. Being the second most expensive single investment ($5,000) after the control-room monitors, it was quite a relief to have the job done right the first time.

We had come to the point of no return. It was just two weeks until our first session. This was a very brave move, booking so soon, since we had just begun moving equipment in from the old studio, and hadn't even begun wiring the cue system. Nonetheless, we all felt we had a decent chance.

Voicing the Control Room

The last few days were spent voicing the control-room monitors. We first tried to use the system without equalization, which looked and sounded very good in the mid- and lower-frequency region, but was unfortunately too bright overall. A White Instruments sixth-octave equalizer was then inserted in the monitor signal path, and used to create a gentle high-frequency rolloff beginning around 3 kHz, and continuing at a rate of about 2 dB per octave. The B&K 2131 Real-Time Digital Frequency Analyzer

Photo: DICK NEIL

Control room equipment centers around an API 2488 console, Studer A800 and Lyrec TR532 multitracks, Studer, MCI, and ReVox mastering machines, Westlake HR-1 monitors, and dbx, UREI, Eventide, Lexicon, and Orban outboard signal processing units.

was used for these adjustments.

The First Sessions

The first thing we noticed in the new locale was how much musicians enjoyed playing in the big room. Many times I've heard musicians complaining that he can't hear or "feel" his instrument. This comes from playing in an environment where the tones are literally sucked into surrounding traps; in the new room it is quite the opposite.

Modern studio design has learned to depend on active trapping, especially in the lower frequencies, to provide isolation between instruments and the acoustical comfort of the musicians. At Easy Sound we feel that we have created a more *variable* work space, where through the careful and considerate planning of a session — such as the judicious use of carpets and gobos — an engineer can create a more interesting, yet still intimate recording atmosphere.

You can almost always achieve a "tight" recording in a reasonable sounding environment through the use of close-miking techniques and gobo placement, but it is very hard to manufacture a genuinely *live* sound.

Since completion of the studio, Easy Sound has been fortunate enough to keep fairly busy and, at the same time, had the chance to do many different types of sessions. The first project was a popular Danish pop-rock group that Niels Erik was producing and engineering. I soon started with another Danish act that ran in parallel with Neils' project. These first sessions gave us both an opportunity to experiment within this new acoustical environment. Soon after, Henrik began a series of dates with the Royal Theatre Opera Company. This was pure excitement, with a choir that at times was as many as 60, and a completely different approach to using to room.

Copenhagen has been a major jazz center in Europe for many years. It has also been the home of many international artists, including Stan Getz, Kenny Drew, Ben Webster, Thad Jones and Niels Pedersen. Easy Sound has always done a great deal of recordings within this idiom, beginning with its first release that Niels Erik recorded on a ReVox machine some eight years ago. I now find doing jazz sessions in the new room very enjoyable, with the opportunity to incorporate a natural reverberation into the recording. It seems that the "classical" miking technique of finding a point where the balance between direct and reverberant sound is optimal, at times can be also applied to jazz and pop dates.

Since its opening, Easy Sound has been the venue for the recording of film music, the room lending itself well to classical instrumentation. The largest of these film dates, with 45 musicians, proved that the room responds very well to a larger string session.

Video and Live Concerts

There have been several video recording sessions, with the band hiring an independent company, and the studio usually recording the tape for subsequent lip-sync playback. These sessions have worked well with the large stage, which provides plenty of room for the most complex scaffold and lighting setup. Several playback tapes have also been done for live television broadcasts.

Our first concert at Easy Sound was coordinated with a private party given by a major record company for local record distributors. This functioned very well, with the guest mixer reported to be quite pleased with the acoustic situation. There are also several groups interested in doing live recordings, which we hope to arrange in the near future. Capacity, with a concert seating arrangement, will be between 350 and 400 people.

Towards the Future

When speaking with Niels Erik and Henrik Lund about the future of Easy Sound, they both maintain a positive and progressive tone. They're interested in buying a larger console with automation, but concede that an SMPTE synchronizer would be a more practical and realistic purchase at this point. With such an addition, engineers will be able to get more out of the studio's second 24-track machine, and we will be able to utilize a method of film and video dubbing that Niels Erik has been developing.

With 1,500 square feet of available basement area, there are all ready plans for a small "B" studio, and serious talk of establishing a video facility. Disk cutting equipment has also been discussed, and it's one area in which we all are interested.

All of these things will take time and money, but Easy Sound's new location is standing proof that anything, with the right combination of determination, skill, and hard work, is possible. ■■■

References

1. "Architectural Acoustics", by K.B. Ginn.
2. "Reverberation Process at Low Frequencies", by H. Larsen; *B&K Technical Review, Number 4, 1978.*

STUDIO DIMENSIONS AND CONTROL ROOM EQUIPMENT

Main Studio Area: 4,500 square feet.
Ceiling Height: 30 feet.
Stage Area: 475 square feet.
Isolation Room "A": 200 square feet.
Isolation Room "B": 100 square feet.
Isolation Room "C": 65 square feet.
Control Room: 300 square feet.
Basement Area: 1,600 square feet.

RECORDING EQUIPMENT:

Console: API 2488, with auxiliary Neumann 0023.
Multitracks: Studer A800 and Lyrec TR532 24-tracks.
Tape Machines: Studer A80, MCI JH-110, and ReVox stereo machines.
Noise Reduction: 24 and two-channel Dolby racks, two-channel dbx, and 2-channel Teletronix.
Monitor Speakers: Westlake Audio HR-1, Tannoy, JBL 4311, Yamaha and Auratones; powered by Crown amplifiers.
Echo: EMT 140 and 240 plates, MICMIX Master Room, and live chamber.
Outboards: includes dbx, UREI and Teletronix LA-2 compressor-limiters, Eventide, Lexicon, and Orban digital delay and effects units.
Microphones: and extensive collection that includes several "classics," such as a Neumann M-49, KM56, SM-2 stereo, and six U47 tube models, plus five Bruel & Kjaer mikes, described as being a "personal favorite" of the author. ■■■

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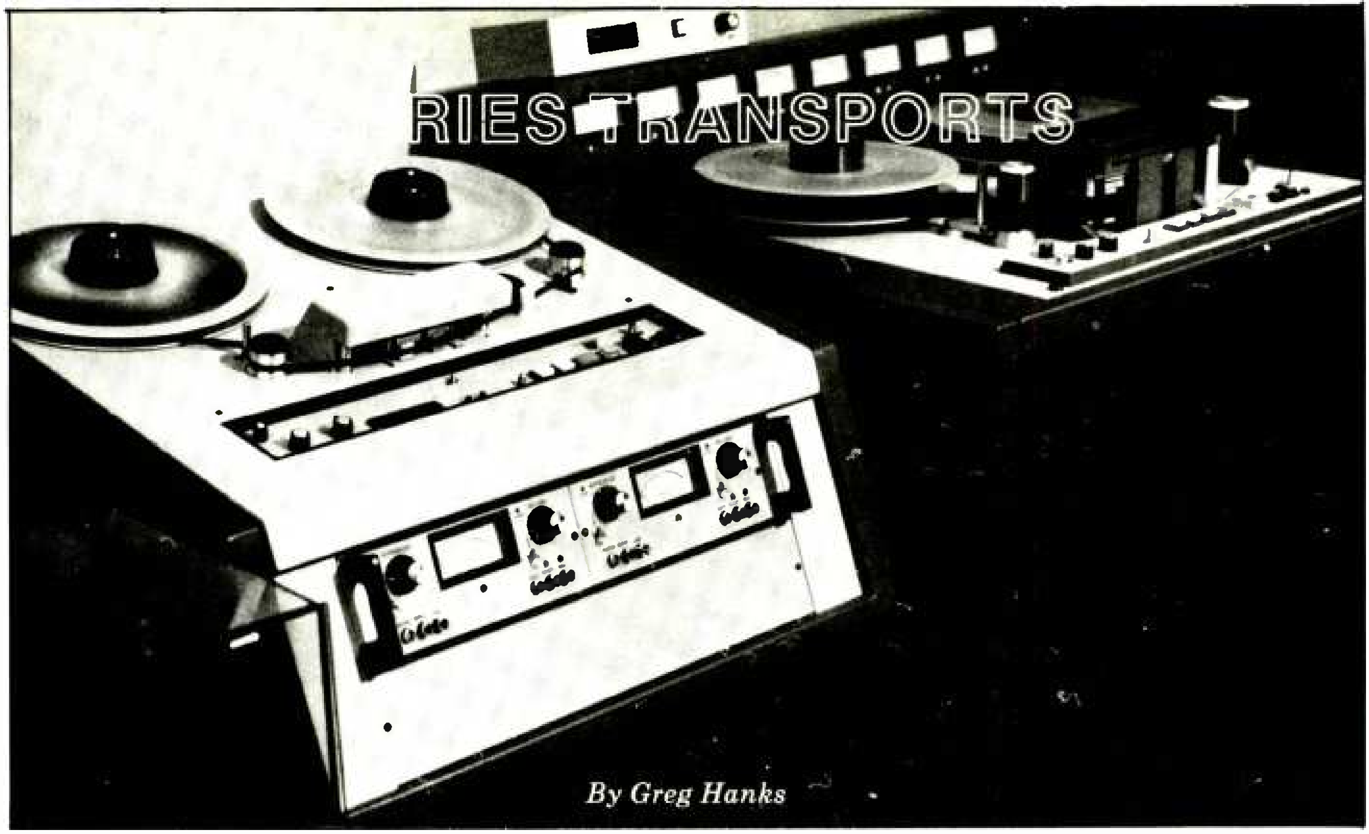


Over the last 10 years MCI (now MCI/-Sony) has established itself as a leading figure in the manufacture of professional multitrack tape recorders. This has been accomplished (in this author's opinion) through technical innovation and budget pricing. A number of MCI/Sony features have set the trend for modern day recording, including:

- Monitoring logic of the JH-100, -114, and JH-24 Series
- Autolocator systems
- Constant tension reel servos
- Wrap and azimuth adjustable heads
- Ceramic Capstans
- One button punch-in and -out, etc.

MCI/Sony machines are modestly priced and provide very impressive results, in addition to being a service person's dream in terms of accessibility. The number of hits made on MCI machines is very large but, due to the price structure, and, I think, technical "intimidation," MCI equipment has been somewhat maligned much of its life. As a maintenance person, this writer has cursed the company many times myself. (But, there again, I would be sorely pressed to name a brand that has not at one time or another been blistered by my frustrated ravings.) Having been involved in the care and feeding of MCI machines for more years than I care to count, I have been able to watch the designs evolve and mature. The following article details some of the modifications, set-up procedures, and repair practices that I have gleaned.

Before launching off into technobabble, however, I feel you should be warned — all machines created equally are actually different! In pragmatic terms, while the basic theories of magnetic recording are applicable to all tape machines, each and every one, of whatever make, is distictly *individual*. The information presented herein is intended to assist you in case your problems are similar to those described, or for intellectual perusal. This is *not* intended to be a how-to modification manual.

Getting Started

A good starting point is to assure that you have the proper documentation for your machine. MCI in Fort Lauderdale has most of the schematics applicable to your recorder (unless you own one of the original JH-10 prototypes). To see if your manual is complete, match your PCBs to the schematics by assembly number. When all the boards, including power supplies and chimneys, have corresponding prints add an "Interconnect Harness Diagram" (the transport wiring) and a "Wiring Diagram" (the audio wiring), and you are complete. Once this is done the next step is to acquire a full set of service bulletins for your machine. These can be obtained through your local MCI dealer, or from MCI directly. Always include your machine's date of manufacture when communicating with MCI.

— 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.

It is my belief that the proper approach to tape recorder optimization begins with obtaining tape position in travel that is perpendicular to the plane of tape travel. Tape travel will be at a fixed and uniform height in a series of parallel planes, and should be accomplished with minimum tension sufficient to provide intimate tape-to-head contact. This tension should not exceed 9 ounces, and the tape reference to the bottom of the tape guides. Hard contact should not be maintained. However, the "low-side skew" of the tape (as induced by slitting tolerance) will be bounded by the bottom guiding edge.

The center line(s) of the bottom tracks of all heads should be the same height above the reference surface of the deck, and parallel to tape travel. Once these characteristics are obtained, the heads should then be adjusted for optimum wrap, followed by a rough azimuth alignment. The playback system should then be conformed to the studio's slowest speed test tape. The bias system should then be optimized for erase and bias, at which point standard playback and record alignments can be performed.

At this time it might be wise to tell you a little about the material this article is supposed to cover. The best arrangement is to break the topics down to subsystems, and discuss each. So here goes.

Transport Systems

The MCI Transport System is comprised of the following components:

1) Take Up and Supply Motors, with their associated Motor Drive Amplifiers (located in the Transport power supply)

CARE AND REPAIR OF JH-SERIES TRANSPORTS

2) Capstan Motor and Tach Assembly
3) Analog Torque Board
4) Control Logic Board
5) Phase Lock Loop Board
6) Interface/Lampdriver Board
7) Solenoid Driver Boards
8) Transport Mother Board
9) Autolocator and Remote Control Assembly
10) Associated Whirling Widgets and other Gizmos that help the machine work, and further confuse the issue.

Before discussing the transport system, a brief overview of the operating theory is in order.

Reeling tensions (in all operating modes) are controlled by the Analog Torque Board, and are selected in the various operating modes by the FET packs (IH 5011): on older JH-100s designated IC21, 22, and 23; on newer JH-114s these are IC16, 19, and 25. The output of the analog torque board drives the associated motor drive amplifier(s) located in the transport power supply. The FET packs are driven by the control logic card by levels that go high for an *on* condition. Inputs to the analog torque board are:

A) The supply and take-up motor DC tachometers, which provide a DC output directly proportional to the rotational velocity of their respective shafts, and which are directly coupled to the motor shafts.

B) Analog capstan speed, which is a DC signal directly proportional to the rotational speed of the capstan, whether it is a fixed, variable or external drive.

C) A/L analog velocity, which is the signal generated by the Autolocator to drive the tape to the desired position.

D) 15 Volts, which provides "hard" drive to the reel motors for Fast Forward and Rewind.

Together these input signals are conditional, modified and adjusted to provide the proper operating tensions for the mode selected.

Older analog torque boards, such as that used in the JH-100 and JH-110, utilize a voltage-controlled oscillator, chopper, and active filter to provide for the tension controls, rather than the quad multiplier found in the newer units. The documentation shows ripple voltage for the analog capstan speed line, rather than the DC voltage actually necessary. The DC voltage determines the time for which control of empty reel is active. The shown ripple should be the *maximum* ripple on this line.

The PLL or Phase Lock Loop board controls the capstan speed. This system is composed of:

A) Crystal-controlled Oscillator and Divider

B) Voltage-controlled Oscillator and Control Voltage Scaling Amplifier

C) Digital Reference Selector (applicable to new style PLLs only —crystal on lower left of PCB)

D) Tape Velocity Generator, which provides the DC voltage proportional to capstan speed

E) Phase Comparator

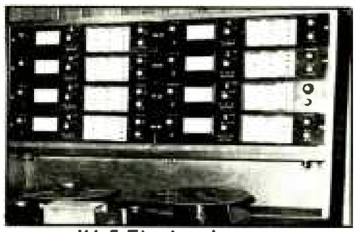
F) Motor Starting Circuit and Output Amplifier.

The system in the fixed-speed mode works very much like any PLL, *i.e.* the reference is divided down, and the speed selected via different dividers feeding one side of the phase comparator, the other side of the phase comparator being driven by the motor tachometer. The time difference between the two signals varies the duty cycle of the signal being fed to the active filter, which in turn provides a DC signal to control the drive circuit that powers the motor; the Tachometer Buffer also feeds the tape velocity generator. When the variable speed mode is selected, the voltage-controlled oscillator is selected through the digital reference selector. The VCO's frequency is determined by a DC voltage, varied by a potentiometer applied through the "Vari/Fixed/External" selector switch.

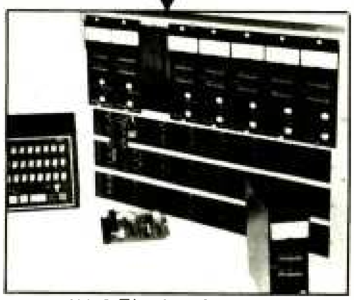
When in "Ext" mode the source of the reference may be either a DC voltage, or a nominal 19.2 kHz external signal. The digital reference selector will select the appropriate line through the use of the

MCI JH-SERIES MULTITRACK FAMILY TREE

ELECTRONICS

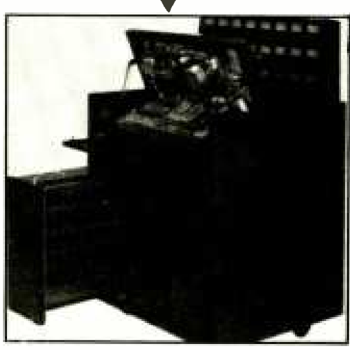


JH-5 Electronics
(For Ampex Transports)



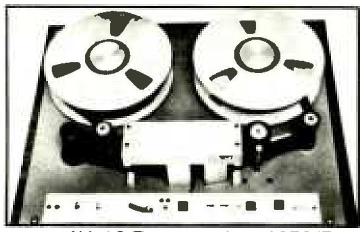
JH-8 Electronics

JH-16 Electronics
(Advent of Stripboard Concept)

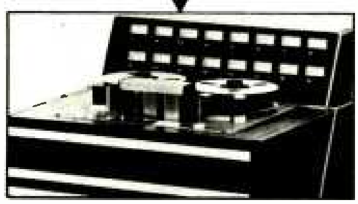


JH-24 Electronics
(Transformerless Circuits)

MULTITRACK TRANSPORTS



JH-10 Transport — 1970/71
(All AC Motors and Synchronous Motors)



JH-10 in JH-16 System + Autolocator I

JH-100 Multitrack Transport — 1973
(AC Reeling Motors, DC Capstan & Impedance Roller)
+ Autolocator II

JH-114 Transport
(Dancer Arm & DC Reeling Motors)

THE PRESENT GENERATION . . .

JH-24-24 WITH AUTOLOCATOR III

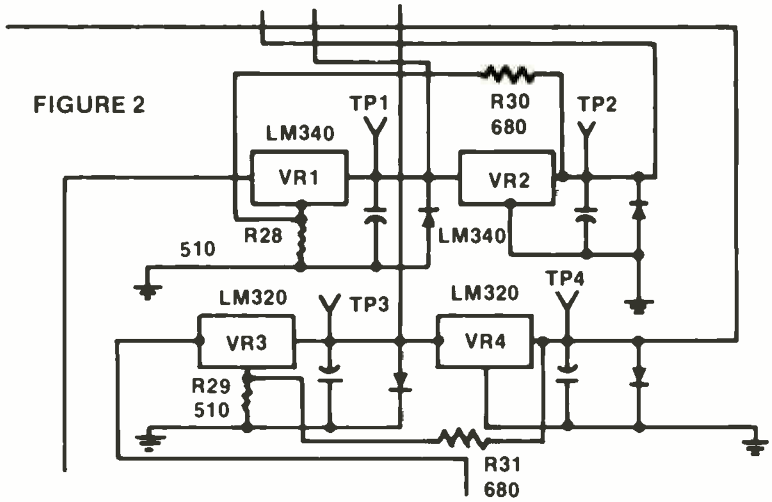
JH-110B-2-VP STEREO MACHINE


V800 MOS-FET
High Performance Professional
Power Amplifier
thermal
power

re-triggerable one shot, IC18, and NOR gates, IC16, 17, and 19. When an external frequency source is applied it is selected through the one shot; in the absence of this signal the VCO is selected. When there is a DC control signal applied, it varies the voltage that controls the frequency of the VCO; in the absence of a DC voltage, the VCO runs at a frequency determined by the "Cal" pot located on the PLL board.

The start time of the capstan is determined by the "slew up" amplifier held at a negative voltage, which forward biases CR10, and shuts off the motor. For start up, the left motor play command applies a positive voltage to R39 and C24, driving the output of this amplifier positive, back biasing CR10, and turning the motor over to the motor amp.

FIGURE 2



SHADED AREAS DENOTE MODIFICATIONS TO JH-110 POWER SUPPLY BOARD

The Control Logic Card does exactly as its name implies — it receives commands from the push switches, and logically converts these to control the transport and record functions. The main problem with this card has been dirty IC sockets, so we do not really have to get into this area.

The Interface Lamp Driver provides five different functions that are necessary to the operation of the machine. These are:

1) Lamp Drivers that provide a logical low for the lamp to be illuminated.

2) Manual Velocity Control sensor, which provides a logical low to the control logic board, when the MVC is engaged.

3) Autolocator enable line that goes to the Autolocator is sent from here; if the "start" button does nothing, check IC6.

4) Tach Generator (or tape counter if you prefer) buffer, is comprised of IC7 and 8. When you begin to loose counts with your 'locator, the top of R11 is the place to look with your scope to see if the pulse frequency is decreasing.

5) Record Momentary Pulse, and the Record Hold lines are both buffered by this board.

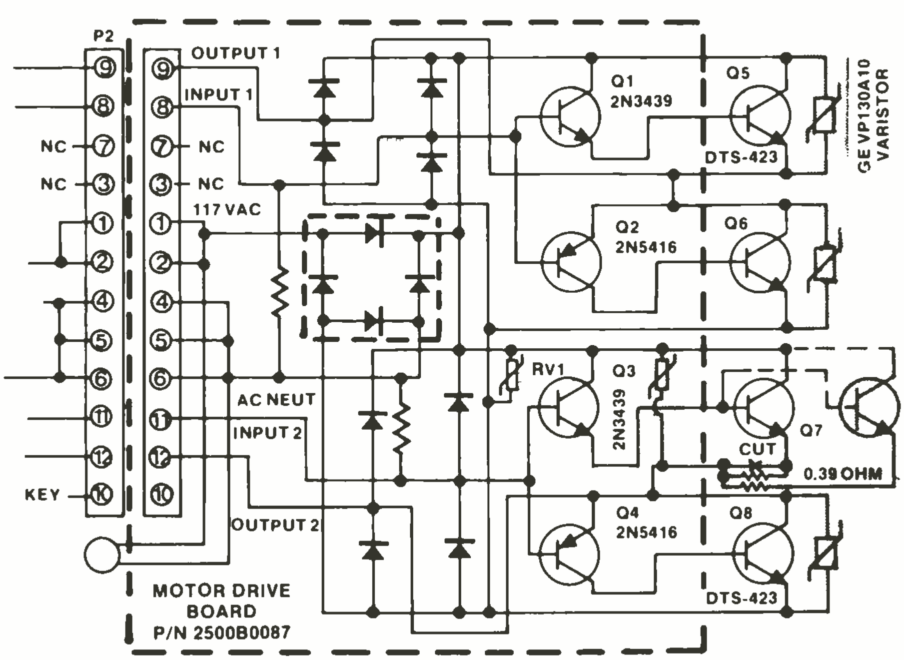
The Solenoid Driver Boards provide the machine's power switching functions, including pinch-roller solenoids, head gate rotary solenoid, brake solenoid, lifters, etc. All of these outputs go low when they are active.

The Mother Board ties these previously mentioned sub-systems together. The interface to and from this board is done with Molex connectors, and they are the only problem this board has. In newer MCI machines there are two buffer amplifiers on this board located under the PLL. Not shown on the interconnect harness drawing, IC2 — the bottom one — buffers the VCO of the PLL, and this signal goes to the capstan servo programming plug, for use in SMPTE timecode systems. The top IC1 provides buffering for the capstan tach pulse, which is also sent to the programming plug for use with SMPTE. Also on the mother board is the Run-Time Indicator, which provides an indication of the amount of time tape has been running across the heads. The timer only works in the play or record modes, and it is also under the PLL. The lifter command is buffered by Q1 on the mother board.

Something to note at this time is how to work your way around the MCI schematics. The easiest way to tie everything together, is to realize that the interconnect harness drawing shows what goes to/from the mother board, and all the connectors that live on the transport. So, to find out what drives what, first find the connector and pin numbers on the board in question, go to the mother board and locate the path, and then see where it goes with the interconnect harness diagram.

At this point in time we should address some of the specific circuit boards and the problems that they possess.

FIGURE 1

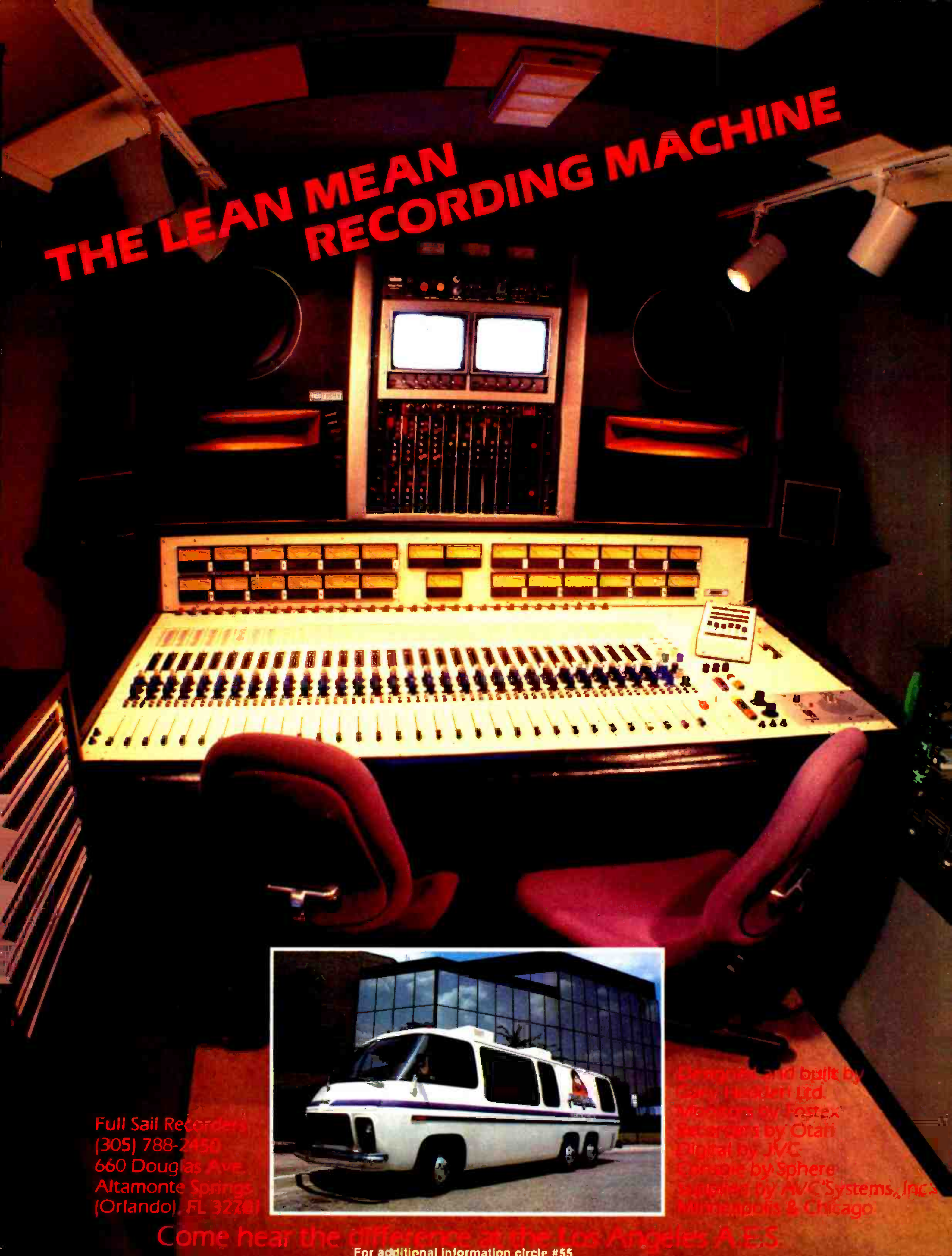


SHADED AREAS DENOTE AREAS OF MODIFICATION. NOTE THAT FOR CLARITY PARALLEL DEVICE IS ONLY SHOWN ACROSS Q7. MOST NEW MOTOR DRIVE BOARDS HAVE THIS MOD IN SITU

Power Supplies

1) Older JH-100s:

Problem: Reel motor transistors



For additional information circle #55

CARE AND REPAIR OF JH-SERIES TRANSPORTS

keep blowing up.

Fix: Update the circuit to the schematic of Figure 1, and replace the DTS 411 transistors with DTS 423s, made either by Delco or Solitron. Remember not to subject yourself to the circular failure game; meaning, when the output transistors blow they are usually accompanied by the driver and the pre-driver.

2) JH—114s:

Problem: Heat is the enemy of any bi-polar power circuitry. One of the common failure modes of the JH-14 and -24 is when the "chimney" fan conks out, the motor drive transistors overheat, and one or more of them short. This turns the corresponding motor fully *on*.

Fix: The easiest diagnostic procedure is to remove the chimney, turn the machine on, and unplug the power transistors until you get to the dud. When you pull the plug on the bad device, inspect the insulation for brittleness and cracking. For maximum reliability, this device should be either an RCA 2N3055*H* or a JAN 2N3055(A). When changing transistor types, its mate should be changed also.

3) Note: When changing any of the TO-3 type negative regulators, the case is the V- input and not ground. If care is not taken to ensure that the insulating washer and heat compound is used, traces that feed these guys can easily vanish into smoke.

4) Odd: After many hours of chasing an increasing number of erratic and intermittent problems, one of my clients found that the wire that energizes the flywheel solenoid on his JH-100 was rubbed through. A lesson that we relearned was that when there is a problem with the deck, or electronics, check the integrity of the power!

5) Noteworthy: On some older JH-110s, the ±15 and ±24 volt regulators were both LM320-15V and LM340-15V devices. Not shown on the schematic are additional diodes and resistors on the ground sensing leg of the 24-volt regulators. Those familiar with these devices are aware that the voltage on the output can be increased by removing the ground leg, and re-inserting a portion of the output voltage into this port. This is exactly what MCI has done on these machines. Therefore, if your machine has a 15-volt regulator in the 24-volt socket, measure the voltage before using a 24-volt regulator in this position, or else you could inadvertantly end up with 32+ volts on your 24-volt line!

Figure 2 shows the power supply schematic following modifications detailed in MCI Bulletin #204, dated July 22, 1976, and which was intended to improve reliability of the 24-volt regulator circuit on JH-110 machines; added components are the pair of 680 and 510 ohm resistors connected across the output and reference legs of the four voltage regulators.

7) Problem: Punch in clicks on older JH-100s.

Fix: This problem as it relates to the power supply is rather simple to fix. There are other causes of punch in/out noises on this vintage machine which will be covered a little later in the article. Apparently, some JH-10 power supplies found their way into some of the early JH-100s that had a different grounding scheme. I found a service bulletin, dated August 13, 1975, that covers some additional wiring in the power supply chassis that I will share with you:

"For JH-16 and JH-24 cabinets using one or more JH-8 supply, the following modification will be needed. Supply is identified by the one 12-pin Cinch Jones sockets on the top of the supply, and three internal relays.

1) Power off. Remove each JH-8 supply one at a time. Remove bottom panel. [*Refer to Figure 3.*]

2) Connect jumper between pin #4 and pin #10 of K3. Connect 22 gauge insulated lead from pin #6 to lug of terminal strip, 24V common.

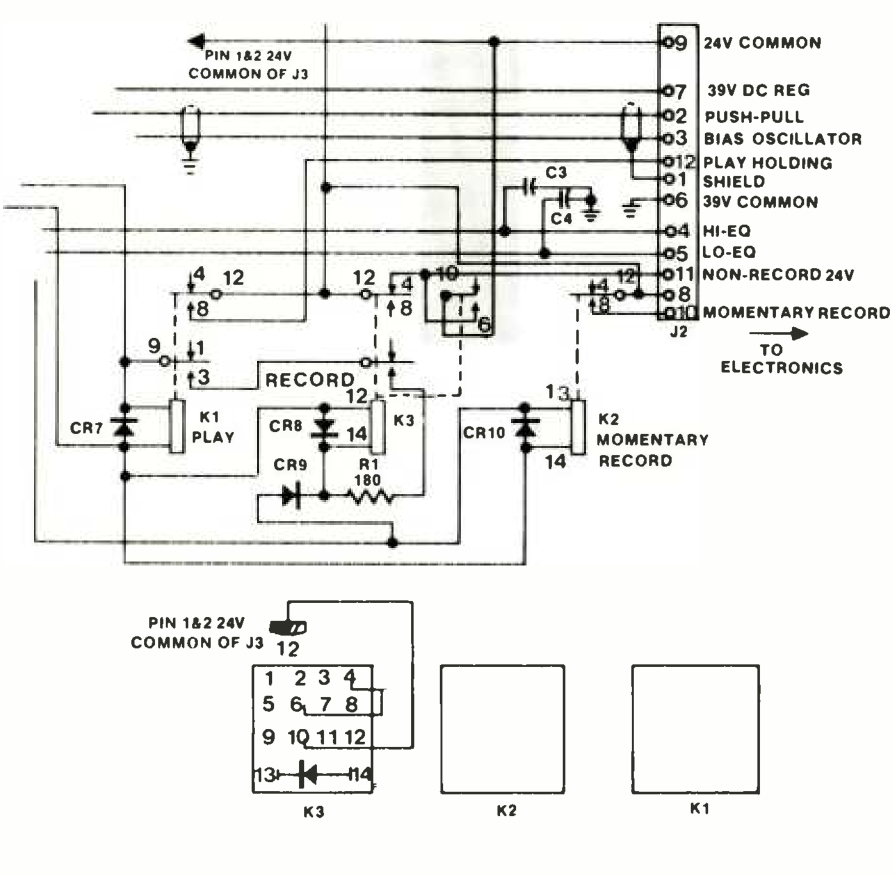
3) Replace bottom cover and return supply to same position."

8) Problem: In some installations, utilizing the Melco "Glide Mount" and a JH-114 with A/L 3, motor noise can appear on all outputs unless the Autolocator case is electrically isolated from the mount. The chassis on the 'locator is at 5 volt ground, and the console chassis is at Audio ground. If the tape machine is now brought to Audio ground, a *big* loop is formed that puts digital chatter on the audio.

Analog Torque Board

Sometimes on the JH-114 and -24 (through rarely), tension inconsistency problems crop up. The supply reel tension can be observed using the centering of the dancer arm; note its position at the beginning, end and middle of the reel at both speeds — it should remain constant at all times. The take-up side is checked by pushing the pinch roller away from the capstan in Play (allowing at least 10 seconds after initiation of Play, to allow the start boost to subside), and observing that the running speed does not change. Again, this test is performed at the beginning, middle and end of the reel in both speeds. If the tensions are deviating, it must be determined if the problem is with the tachometers, the analog capstan velocity, the quad multiplier, or off-sets within the system. These elements are checked as follows, but set the off-set

FIGURE 3



MODIFICATION TO JH-8 POWER SUPPLY BOARD TO PREVENT PUNCH—IN CLICKS ON OLDER JH-100 TRANSPORTS

nulls first!

Tachometers: Measure the offset voltages at test points one and two in low-speed play at a given point in the reel, then switching to high speed. If working properly, the voltages at these points will double.

Analog Capstan Velocity: Measure the voltage at P11 pin 2 on the analog torque board in low-speed play. Again, switching to high-speed play should double this voltage.

Quad Multiplier (on newer machine): Measure the output voltage in low-speed play, then switch to high-speed play. This voltage, unlike the previous tests, should *not* change. To thoroughly examine this chip for proper operation some math is necessary. The output of the quad multiplier is a percentage of the quotient of the analog capstan voltage divided by the Tach output. These voltages are to be measured and noted from the beginning of the reel. Also measure and note the output of the multiplier. Then perform the same measurements at the end of the reel.

Perform the following calculations: Output V/(Analog Capstan V/Reel Tach V)=X. Beginning of the reel "X" should equal end of reel "X," or be within 10%. If they do not, replace the chip.

Modulator Circuit (on some of the older machines, without the multiplier): This circuit is treated much the same way as above, with — for the take-up side — Analog Capstan V measured at pin 6 of EC2, Reel Tach at pin 6 of IC13, and the Output at pin 6 of IC14. For the supply, the tach is measured at pin 6 of IC11, and the output at the junction of R63 and R64. If the tensions cannot be adjusted to provide the same "X" that is given in the above math, then the modulator must also be checked. Between low- and high-speed the duty cycle of the signal at the base of Q3 should change by 50%.

Pitfall: When measuring the off-set nulls of the Analog Torque Board with an AC-powered instrument (scope, DVM, etc.) be careful to use one of the capacitor-minus terminals at the test point as ground, and *make sure* the third pin on the AC cord of the instrument is lifted. Otherwise large DC offsets are introduced into the reading!

Repair: Sometimes at the end of the 30 IPS reel, you hit play and nothing happens. Well folks, it could be one of two things:

1) Too high a hold-back tension setting;

2) The puck/pinch roller tension isn't high enough.

I have seen a few occasions where replacement was the only cure, but usually a quarter to half a turn on the adjustment nut is enough.

Troubleshooting: When chasing down tension abberations, it is sometimes necessary to look directly at the output of the FET pack. This can only be accomplished by removing the driven op-amp, since the FET pack feeds the summing junction, and without reading this point with a current probe you just won't see anything. If you do get a reading, however, the op-amp is defective.

Also, if in fast wind modes the machine acts very sluggishly, or throws loops or performs other nasty behavioural things, check the notes on the schematic, and make sure that the board is configured to the required track format. It is *very* disturbing to discover that your two-inch 15/30 IPS machine is running with a torque board configured for ¼-inch 7½/15 operation.

Repair: On some of the older JH-100 analog torque boards there were some problems with the plated-through holes. Installing a jumper (or "z" strap) and soldering both sides of the board has solved many intermittent tension problems on several machines. Another useful tip for these older boards is to install 68X-type top turn, multiturn pots in place of the full-reel/empty-reel pots. At the same time replace the off-set null and 50/60 Hz pots. When you find an older board requiring "z" straps, also perform the same connection confidence modification to the mother board, PLL, and control logic boards.

Part Two of this article, to be published in next month's issue, will cover Phase Lock Loop Boards, Interface Lamp Driver, Motherboard, and Autolocator. ■■■

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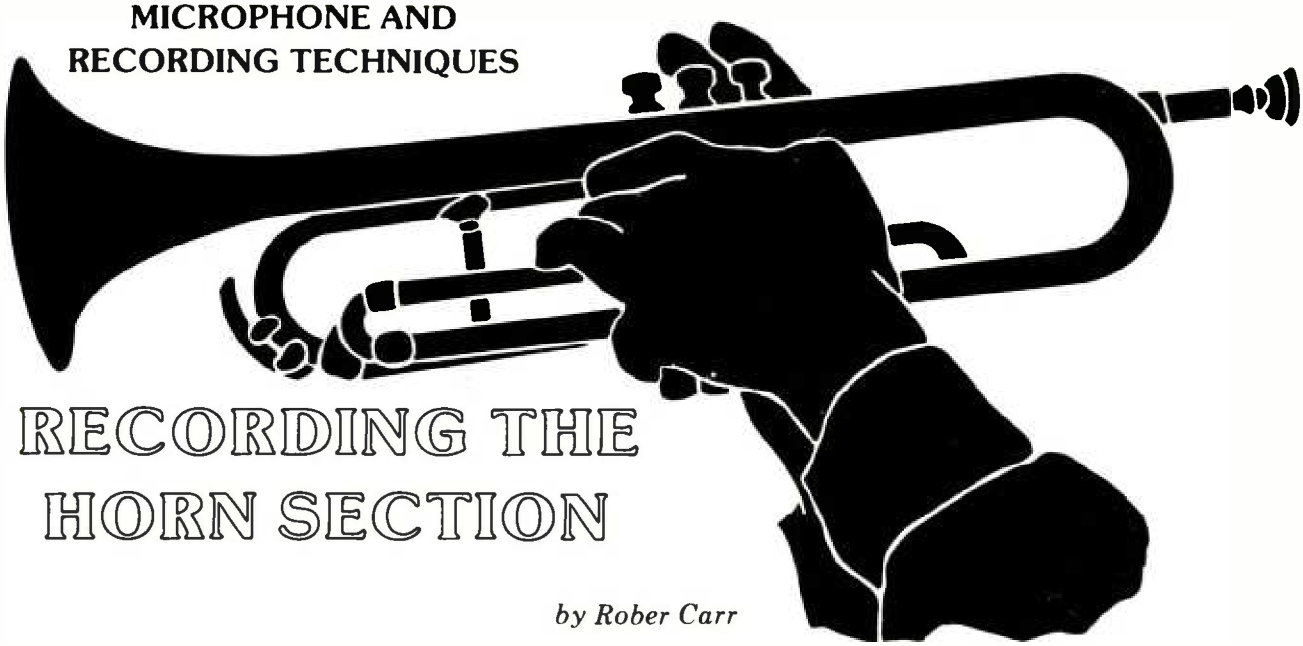


RECORDING THE HORN SECTION

by Rober Carr

Aside from being among the staples of classical and pseudo-classical works, wind instruments have found a comfortable niche in show music and jazz, both as solo and ensemble instruments. But their use as a "section" in R&B and rock and roll currently is limited to the styles of primarily three well-known groups: Tower of Power, Earth, Wind & Fire, and Chicago. Each of these bands has maintained its respective success for more than a decade, because they developed a distinctive sound that caught the public's fascination in spite of, or in conjunction with, the onslaught of electronic keyboard technology and electric guitar "acrobatics."

A look at the methods of recording these three ensembles can go a long way towards explaining the current "state-of-the-art" in section miking. One point should be considered first though — how the instruments function in a general sense.

The wind instruments encompass two families of the orchestra: the woodwinds (clarinet, saxophone, English horn, oboe, bassoon, flutes, recorder, piccolo, fife, bagpipes, etc.); and the brass, itself being comprised of two subgroups. First, there are brass instruments that have a relatively large amount of cylindrical tubing in their center section, and a bell that flares abruptly, such as the trombone, trumpet, and French horn; and, secondly, "conical" horns, including the flugelhorn, alto horn, baritone horn and tuba, whose tubing increases in diameter from the mouthpiece to its smoother flaring bello. For immediate convenience, the focus of this article will remain primarily on the common members of a horn section — trombones, saxophones, trumpet, and flugelhorn.

Throughout both of these two families are uniformities and conformities, which led to the classifications in the first place. Yet the number of operational variables is still overwhelming. For an in-depth study of coloration versus design, and an analysis of the various methods of sound production, "The Physics of Music," a compilation of reprints from *Scientific American*, is highly recommended.

The audio engineer can only benefit from possessing a basic understanding of the mechanics behind these instruments, especially when called upon to capture a unique horn impression, or straighten out an incorrect blend that has resulted in harmonic clashes. Any attempt to distill the information contained within that scholarly publication would do justice to neither you, the reader, nor its writers.

Therefore, what follows represents a slight departure from the procedure established in previous miking articles in this series, and a refrain from addressing the principles responsible for producing tonal characteristics of given wind machines. Instead, it seems more appropriate to concentrate on the conditions that an engineer should be aware of in recording horn sections.

A Soloist's Viewpoint

Electra recording artist John Klemmer has been making albums with his saxophone for approximately 15 years, and is now also producing. From his unique position of working on both sides of the studio glass, he's able to provide some valuable insights into the world of the horn player as an individual.

At the present time, Klemmer is vacillating between two similar tenor saxophones — a gold-plated Selmer Mark VI, which he uses most of the time, and a standard, brass Mark VI. "The gold-plated horn is very brilliant," he offers. "It's not so much that there's a lot of highs, but the gold plating gives a really powerful sound. We've had difficulty miking it.

"Even though I've been producing myself for quite a while, I still go by how the horn sounds, and how it feels. We look for a microphone that will tone down that brightness. I can pick out certain ones that I like, but each engineer has a different way of using that mike. So I go by sound and feel."

For a full, open sound, John Klemmer likes to have distance between his horn and the microphone. "But I've developed a bad habit from playing so many joints with one of those cheap mikes," he confides. "I used to play with the microphone right in the horn. So what we've done over the years is set up a dummy mike that I can stick the horn into, while the real microphone is placed at whatever the accurate distance is for the day; it depends on how hard or soft I'm playing, the nature of the tune, and what register I seem to be concentrating in. The sound is also affected by the physical condition I'm in for any given day, and even whether I'm sitting down, or standing up while playing."

The sound depends a lot on the reed, too. In a single three- or four-hour recording session, Klemmer might go through as many as 10 reeds. "I play so hard and so intensely that I may blow a reed out on one take. Changing reeds can drive engineers nuts, because a new reed will give the horn a completely different character — either a little 'fluffier,' 'buzzier,' or whatever. Plus, as I'm recording, the reed starts breaking in. The sound can change every 10 minutes, and the engineer has to be aware of adjusting to that."

Likewise, brass instruments are equally as susceptible to a number of variables, such as the bore and shape of the mouthpiece, and the size, shape and coating of the horn.

Jerry Hey is a first-call player for all the Los Angeles sessions. Most of his dates involve overdub, section work. He uses a Calicchio trumpet, and a Flugelhorn by Couesnon. "I prefer a brighter trumpet, and a Calicchio is the brightest around. For what I do, which is pretty much hard, loud playing on dates for Earth, Wind & Fire, or Michael Jackson, a 'dark' trumpet doesn't seem to print on tape as well. Even if the soft-trumpet sound is turned up loud, it still doesn't get that same *intensity*. The Calicchio generates that intensity with no problem. I've found that if I play another instrument in a section with two guys playing Calicchios, I can't balance with them. The Calicchio is not my favorite horn for solo work but, in a section, it's the best."

For ligitimate solo work, Hey recommends a Bach C, and for jazz soloing, a Bach B-flat. "Choosing the right horn for a particular application gets pretty involved. Just the mouthpiece alone will really change the sound. As an example, Bob Reeves, here in LA, can make any style of mouthpiece to any dimensions that will give you any sound you want."

Obviously, with all the possible combinations of instruments, mouthpieces, players, and musical styles that an engineer can encounter, even generalities are hard to come by. Every date is an individual, problem-solving situation, yet there are several basic ways to approach a horn section that lead to a solid starting point.

Humberto Gatica — Chicago

Chicago represent an excellent example of a small horn section with a definitive sound, and Humberto Gatica offers the following ideas culled from his years of recording all kinds of horn combinations, and explains how he applied them to the recent *Chicago XVI* sessions.

Most of Gatica's horn dates are done as overdubs, he says, which affords him flexibility in terms of player placement, and using room ambience to enhance the tracks. "I never like having all the horns face one end of the room; that never worked for me. I like to have the trumpets and the trombones facing each other for two reasons. First of all, eye contact is very important. Usually the trumpet player is the leader, and he gives all the cues. Secondly, by blowing towards one another, a natural combination of the instruments' harmonics creates a warm sound, even when they are blowing really hard. Taking advantage of that leakage can improve the section sound."

JERRY HEY is a first-call trumpet/flugel horn player who came to Los Angeles from Hawaii about six years ago with a group called Seawind. After four albums he started doing dates in town, and now gets regular calls from producers and artists including Quincy Jones, David Foster, Earth, Wind & Fire, Elton John, Donna Summer, and Giorgio Moroder.

Because musicians like Jerry Hey and Chicago's Jimmy Pankow have incredible power when they play, Gatica looks for miking techniques that will give him a "thickness" to the horns. "I've found that the harder a player blows, the thinner the horns sound. If I use the wrong mike, all you hear is sort of a high distortion."

Distance is critical in these cases, too, Gatica offers. A microphone positioned too close or too far away will cause individual problems that need to be overcome. "I like to let the horns 'breathe.' The closer the mike is to any brass, the more the amount of space within the brass section tends to be limited, and the overall effect will sound very dead, when you put it in the mix. Then you have to add echo, and/or EQ, and that's when the sound gets lost completely.

"On the other hand, if the microphone gets *too* far away, the sound gets spread apart, and it's gone. You get such a 'roomy' sound that there's nothing to work with. I like to be able to control how far the sound travels without losing it. You have to pick that optimum distance where the sound is going to retain all the fullness and richness."

JOHN KLEMMER started studying classical alto sax at the age of 11, but soon switched to tenor to play the music of Elvis Presley, Little Richard, and Fats Domino in rock and roll bands. By the age of 19 he had his first record deal with Chess Cadet. His third album for that label, *Blowin' Gold* in 1969, is still regarded as one of the premier jazz/rock fusion albums, due in no small part to Klemmer's use of electronic effects, especially Echoplex, on saxophone. To date, he has 22 albums to his credit as a solo performer, and has recorded or performed with such names as Don Ellis, Oliver Nelson, John Lee Hooker, Nancy Wilson, and Steeley Dan.

Usually, the standard section that Humberto Gatica works with is three trumpets, two saxes, and three trombones; his set-up and microphone selections are shown in Figure 1.

The choice of a Neumann U-47 tube mike for the trumpets is placed a good nine or 10 feet from the source. Regardless of how many horns there are in that trumpet section, Gatica uses only one microphone. "Sometimes the gain is extremely huge, and I have to add a 10 dB pad at the mike input. Then I can open my pre-amp any way I want, and bring the faders up to where I have the cleanest, most punchy sound. If there is any balancing necessary within the trumpets, I request that the first or second trumpet move farther back, forward, or whatever, depending on the parts."

Session trumpet player Jerry Hey presents a musician's perspective: "I generally prefer one mike for the trumpets. It's much easier for the players to balance themselves, than it is for the engineer to set up one mike per person, and try to achieve a balance in the control room. Trombones are a little harder to play into one microphone, because the slides, when they're in the extended position, tend to hit each other! And looking across their bells to the music gets a little more difficult for the players. They usually prefer one mike apiece, and the same with the saxes, because they don't play at the same volume as the brass. One mike for all the saxes would be a little too much room."

Humberto Gatica sets up one Neumann U-87 with a 10 dB pad per trombone, the bell of which is usually about an arm to an arm-and-a-half distance away. "I have to be careful with the distance. The farther away I get from the 'bones, the more trumpet leakage I get in the 'bone mikes. But the individual microphones give me good control, because each trombone is usually playing a different part."

The saxes, too, get separate microphones — Neumann U-47 FETs about 18 inches from the bell, again to avoid trumpet leakage. If this isn't sufficient separation, Gatica will put a baffle — dead side towards the saxes — approximately an arm's length in front of the bell. This is not an absolute rule, but a possible solution should the need arise.

Gatica is concerned about separating all the parts within one track. "I should be able to hear a lot of space between the trumpets, 'bones and saxes, and also be able to pick out each trumpet, let's say, when they play different parts. That's very hard to get when they play hard and aggressively. If the mike is too close, or I choose the wrong one, the sound begins to choke. It seems that there's this huge amount of energy that gets almost compressed, and the sepa-

ration disappears. When the mike is overloaded, the overall horn sound is small within the track.

"The room can be a factor, too. The studio changes with changes in temperature, and so does the horn. Even with good control of the environment, it's important to go into the studio and see how the room sounds. A lot of times you could be dealing with sound that is thin in the studio. Then you have to move people around to take advantage of the acoustics. I've been in rooms that are *too* dead, and trying to create that separation in there is extremely difficult. I like to put players in a place where the ceiling is high, again so the horns can breathe."

To add fullness to the sound, Gatica has found that an additional microphone, such as a Schoeps CMC5-U, hung in the center of the section about eight or nine feet above the floor, and recorded on a separate track, helps capture all the harmonics that are ringing together. By putting that track on top of the overall mix, the technique can create more depth in a small, dead room.

He tries to avoid any equalization when recording horns, feeling that EQ generally takes away from the real sound. Equalizing the instruments one by one creates an undesirable harmonic distortion once they are all blended together again, he feels. At the most, Gatica may add a 2 dB boost at 800 Hz to the trumpets, but only if the sound is still thin, and he can't get what he wants with mike techniques. He will, however, add "just a hair" of limiting on the overall blend, in order to maintain a very consistent peak. Using a UREI LA-2 limiter, he sets the threshold to the loudest passage in the music, which lets the power of the brass through without any "dangerous" fluctuations in level.

Gatica prefers to commit himself to the appropriate mix during the recording process, and usually chooses to keep the trumpets a little up front. "Those are the instruments that have the bite; the initial attack. The 'bones are down, and the saxes are in the middle in terms of relative levels. That's the balance we use to create depth."

But he does warn that "If it's an R&B tune where the brass lines have a lot of movement, like an Earth, Wind & Fire or Tower or Power style, I like to keep the section in the track by raising their level just a hair above where I normally keep them. Instead of mixing the trumpets out in front, I try to keep all the levels more even. That way, wherever I put the brass in the mix later on, I can still hear all the parts within the section. If I leave the trumpets a little out in front, that's all I hear, and the rest of the instruments get lost."

HUMBERTO GATICA started his career as a janitor and goffer at MGM Recording Studios a little over a decade ago. During that time he's accumulated at least two dozen Gold and/or Platinum records with artists like Hall and Oates, Kenny Rogers, Average White Band, Leo Sayer, the Tubes and, most recently, Chicago.

Chicago XVI Sessions

Chicago has only three horn players — trumpet, trombone, and sax — but Gatica still used three mikes (a U-47 tube, U-87, and a U-47 FET, respectively). Their physical arrangement in the studio also remained in a U-shape, with the trumpet and trombone facing each other, and the sax in the base of the "U" (Figure 2). What did vary was the distance between the trumpet and microphone — now four feet instead of nine or 10.

The band features the trombone, meaning that the horn mix is slightly 'bone heavy. The other aspect of their sound is the unison-line arrangements on, for example, "Waiting for You to Decide," and "What Can I Say." Humberto Gatica would let the players balance themselves in the studio, lay them all on one track, and double the same part for a spread and fullness.

Where the trumpet, for example, temporarily switches to flugelhorn in the middle of a tune, he recalls, "I'd rather punch in those sections on the same track later on. That way I get a chance to adjust the mike — maybe bring it closer, because I like close-miking on the flugelhorn. So we go through the entire song, double the parts, change the mikes, and punch in the flugelhorn or flute overdubs on the same tracks. When I mix, I just bring in those two faders, and I know I have a perfect left and right with my horns.

"There are three tracks on the [*Chicago XVI*] album where we recorded three separate mixes of horns — "What Can I Say," "What You Missing," and "Bad Advice." One track was the way it

FIGURE 1

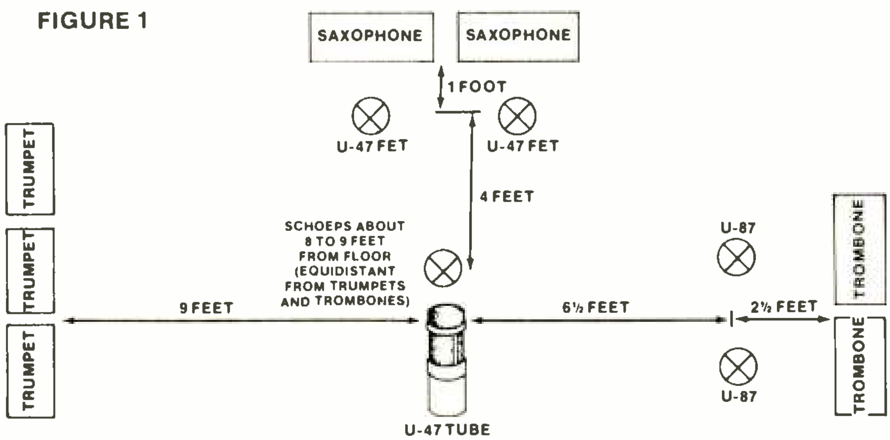
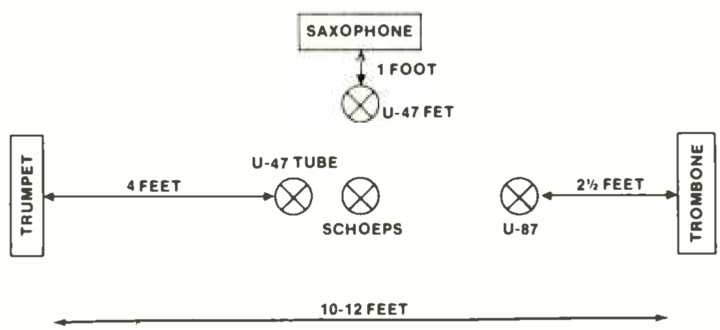
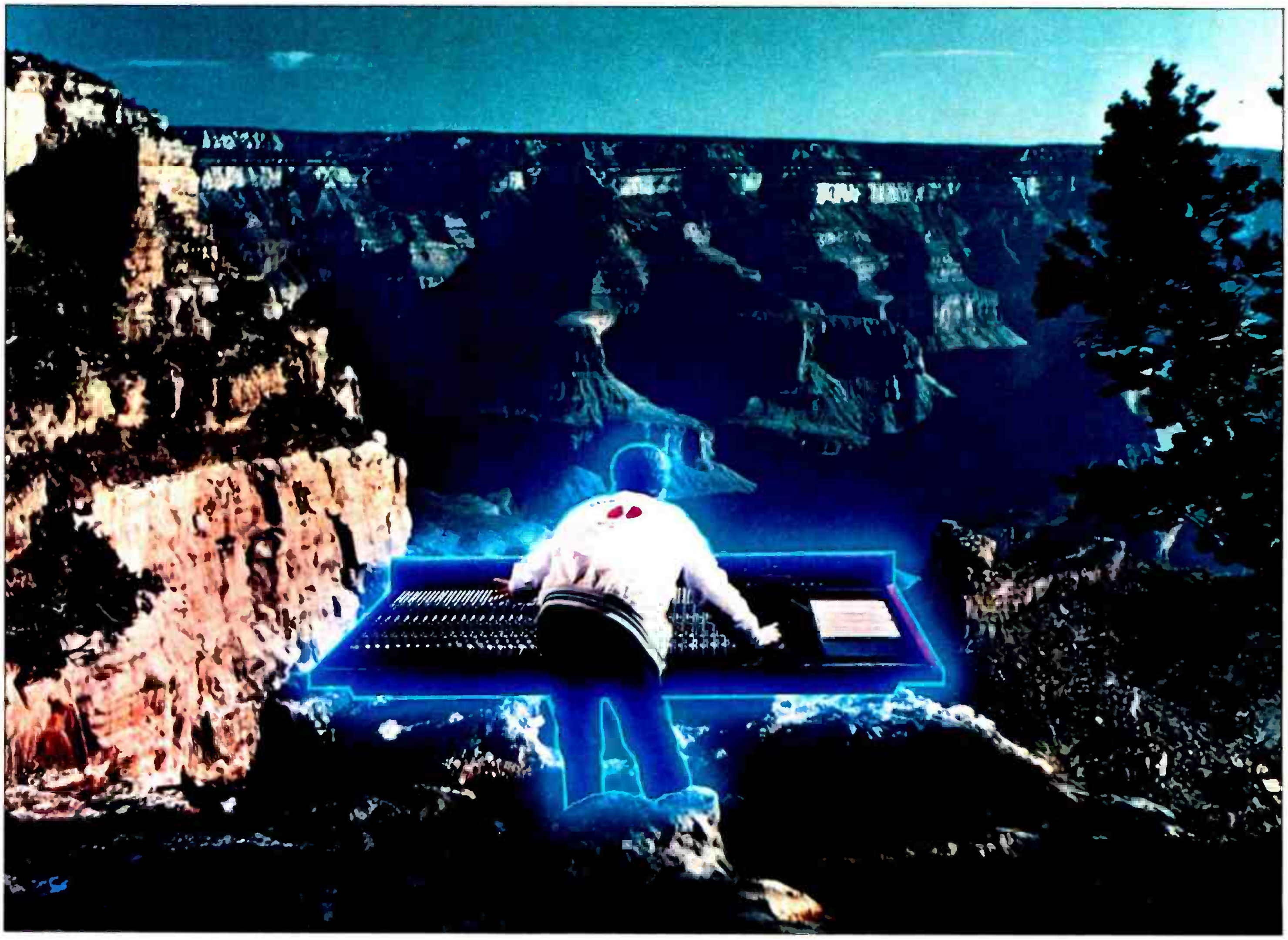


FIGURE 2





BOARD PHOTOGRAPHED AT STREETERVILLE STUDIOS — CHICAGO, IL normally is — a little 'bone heavy; the second track I featured the trumpet a bit more; and the third pass was just a hair more sax. If I needed more of any instument later on, I was protected, since I could bring up whatever track was lacking. The blend that seemed to work well was [achieved by] placing the first track on the left, the trumpet on the right, and a little of the 'sax-mix' track in the middle."

On songs where the arrangements were quite involved, doubling and tripling yielded such a bright section sound that it resembled a big band. In those instances, Gatica recorded either two stereo tracks, or just each instrument on a separate track. He still took advantage of the leakage by putting the overhead room mike in the mix to marry the tracks together.

Monitor levels he keeps very low: "By monitoring at loud levels, everything sounds great. You never hear if the mike is breaking up, or the pre-amp is a little too hot. Then it shows up in the final mix, or on the radio. That's too late."

Mick Guzauski — Earth, Wind & Fire

Mick Guzauski completed the horn and string overdubs for Earth, Wind & Fire's *Raise* album, and is presently recording their new LP, due for release in November '82. All of the horn parts were cut in a large room (48-foot wide by 70 long by 32 high) at The Complex, in west Los Angeles. Two separate sets of thick theatre-type, multiple curtains — one each for the top and bottom half of the room — are capable of completely covering the concrete block walls. The curtains can be opened or closed independently to acoustically control the reflection and absorption characteristis of any area of the room.

At the age of 16, **MICK GUZAUSKI** built a recording studio in his basement, which later became PCI Recording of Rochester, New York. In 1971, he began working with Chuck Mangione, and continued as engineer through the *Fun and Games* album that continued the official theme song of the Lake Placid Olympics. By 1978, Guzauski decided to go independent in Los Angeles, and has since completed the last two projects for Earth, Wind & Fire, as well as work for Cher, Lani Hall, and Peter McIan.

"Without the curtains, the mid-range reverberation time is very long," says Guzauski. "The low-end reverb really isn't a problem with the horns. Any standing wave in a room this size is far below the fundamental of anything we're concerned with. The high-frequency decay time is not really very long, because of the concrete block's porous surface. So what we're really talking about controlling is reverberation from about 200 or 300 Hz, up to about probably 5 kHz. With the absorption of the drapes on 50% of the wall surface, we can keep that time down to about a second and a half."

The horn section for the EW&F sessions consisted of three trumpets, three trombones and two saxes. Like Humberto Gatica, Guzauski set up only one microphone for the three trumpets, but his choice was a Neumann U-48A set to a cardioid pattern. "Using two mikes — unless they're in a X/Y pattern — would cause phase problems," he says. "Because the sound is going everywhere, I didn't want to pick it up from two different instances that are pretty close to each other. The section was doubled on this album, and the final mix is two stereo pairs mirror-imaged to each other. There's really no reason for X/Y stereo on the trumpet section."

The three trombones played into three Neumann U-67s, and two AKG C414s were set up for as many saxes. All of the musicians played towards the north wall for one hard reflection. The 12-foot distance let them hear pretty well, because the first reflections came back at them in a very short amount of time (Figure 3).

In addition to the close miking, a pair of cardioid-patterned AKG C12As in an X/Y configuration were hung facing out about a foot from the wall, and at a height of approximately 10 feet. Guzauski points out that the mikes rejected a lot of that first reflection, "but picked up the reverberation from the opposite wall — actually the whole room sound, or the full one and a half seconds of reverb. The ambience was pretty controlled, because the close mikes were all cardioid too, and facing towards the players. They rejected a lot of the first reflection, and didn't get that much of the room reverberation. The C12As, at the 10-foot height, got very little direct sound. We didn't have a phase problem between the incident and first reflection, because the C12As were so close to the first reflection, and rejecting it anyway. Basically, they picked up just the room ambience."

Even though Earth, Wind & Fire is known for their fast, intricate horn lines, a clean and distinct sound was maintained by not letting the ambience

mikes get too hot in the mix for most of the fast passages. The major advantage was the depth that the room added to the trombones in the lower registers.

Mick Guzauski choose primarily tube-type mikes, because he finds they have a tendency to reject a good deal of the "spitting" harshness that is characteristic of close miking. "Effectively, they provide more distance between the horn and the mike when you have them up close, but you maintain the advantage of keeping the room sound under control with a close mike."

The U-48A on trumpets is similar to the well-known U-47, but has a slightly newer capsule. According to Guzauski, the two models sound similar, and neither have a switchable pad. But the pre-amp in the 48A is set for 12 dB less gain, which translates into 12 dB more headroom, and no problem with overloading.

He avoided compression and limiting during the recording process, because the horn section was so consistent; all the levels were easily controlled through manual gain riding. Likewise, very little EQ was necessary either, but may be required while mixing, he offers, "depending on the rest of the tracks and how the horns fit in the mix. For the trumpets I probably added 2 or 3 dB of 'peaking' centered at about 200 Hz for a little warmth. The U-67s for the 'bones are very mellow-sounding mikes. I did add some top: a 2 dB shelf starting at around 7 kHz. I boosted the bottom a little bit, too: a peak at around 100 Hz. I think I shelved the saxes, and added about 2 dB on a 10 kHz shelf. The room mikes were flat."

The new album was recorded simultaneously on three reels — a master tape and two slaves, although mixing will be handled with just two machines. Some tunes have over 50 tracks (many of them vocal overdubs), so there was plenty of room for horn tracks.

Track assignments on the first pass breaks down to one track each for trumpets, trombones, and saxes, and two tracks for the stereo room mike. All five will eventually end up as one stereo pair. The double was mixed to a stereo pair on two tracks during recording with the same perspective.

"I had all the mikes coming in through separate inputs, and bussed to five tracks," Guzauski explains. "Those busses returned through five faders and, in turn, were mixed to two more busses. I could record the second pass of five tracks at any level I wanted, and then mix them to a pair of stereo tracks while recording. If we need more of any individual instrument, we have control of that through the first tracks, using the second stereo pair as the approximate mix."

In 1968 **LARRY BROWN** served as engineer for the first modern direct disk record for Sheffield Lab. Again in 1971 he engineered Sheffield's second release, *The Missing Linc*. Over the last several years Brown's reputation has grown, not only as an engineer-producer, but as one of the finest drummers in Los Angeles, an excellent keyboard player, as well as a composer. In 1977 Brown and his partner opened The Pasha Music House, Hollywood, which comprises two state-of-the-art 24-track rooms.

Larry Brown — Tower of Power

Over the past decade, Tower of Power's horns have grown to be probably one of the most sought-after sections in the music business. Larry Brown, equally well-known around Los Angeles as an accomplished musician, and as an engineer, especially for his direct-to-disk sessions for Sheffield Labs, has worked with Tower of Power on two separate projects. One was a direct session with the entire group [see October 1981 issue of *R-e/p — Ed*], and the second, an overdub horn date for Peter Noone's last album project.

"I'm a purist when it comes to recording horn sections," Brown concedes. "I feel that 'less is best,' and try to use as few mikes as I can. I'm *not* one to put a mike in front of every player, unless the guys are reading music off stands, or don't know how to play as a section. To me, tight miking a horn section just doesn't sound real. If you get guys, be it horn players, a rhythm section, or whatever, who listen to each other, and don't try to outplay each other, you can get away with a minimum amount of miking, and a maximum amount of sound. If the guys don't listen to each other, and don't balance, you have to resort to heavy miking, and a lot of balancing in the control room.

On the Tower of Power direct date, Brown used just one stereo microphone: a Sheffield lab, line-level mike in an M/S configuration, which was positioned, basically, equidistant from all the horns (Figure 4). "We went back to the old days of radio miking, when they put marks on the floor," he remembers. "At rehearsal we found a place where the tenor player, for instance, could stand in order to get the right balance. When it came time for the solo, he would move to his mark, and then move back into the section when he was finished. I was trying to avoid riding the faders, or doing anything unnatural. I wanted the session to sound as much like a live performance as I could."

Sheffield's studio on the MGM lot in Los Angeles is quite large — at least 85 feet long according to Brown — so additional reverb and echo were not necessary. He placed two ambience mikes

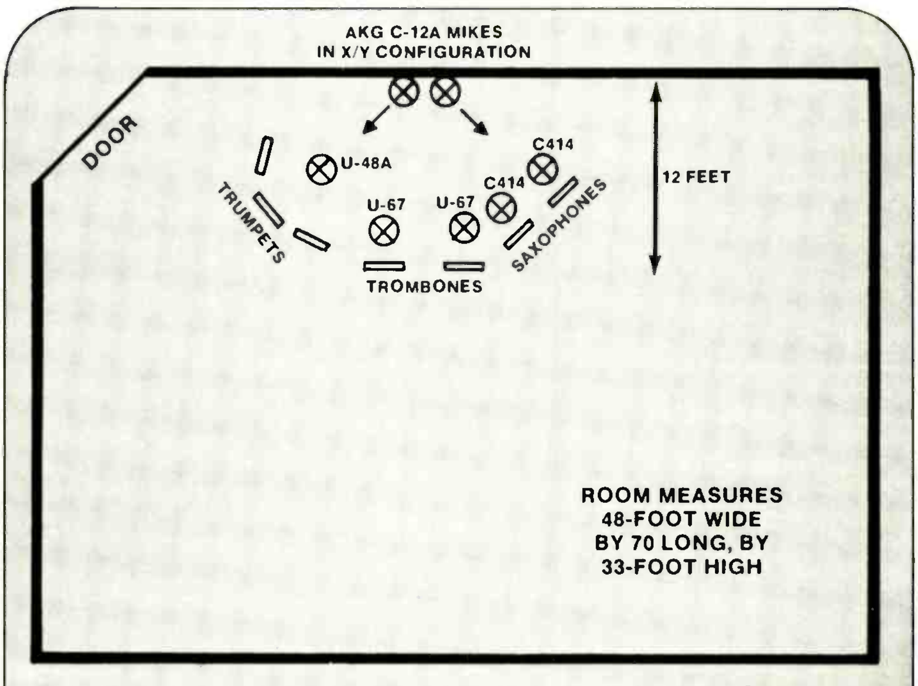


Figure 3: Room layout for Earth, Wind & Fire horn overdubs at The Complex, Los Angeles. Engineer: Mick Guzauski. The three trumpets are located four feet from their single U-48A microphone, while the pairs of trombones and saxophones are positioned 18 inches away from their respective individual U-67 and C-414 mikes.

about 60 feet away from the section. However, he did take advantage of the EMT 140 plate that was there, by putting a 30 to 50 millisecond delay (not enough for a slap) on the horn send to the chamber.

"There's a lot of space around the horns," says Brown. "You feel the width of the room. The date was R&B, and we didn't really want that 'total presence.' We wanted a natural, live sound where the horns felt like part of the band. The main objective was to catch the band sounding exactly like they do in that room, without fabrication of a 'studio' sound."

Brown didn't use any limiting or compression on the date either, and reports that he had no trouble with transient spikes or overloads. "I don't have anything against limiting and compression. But a horn section like Tower of Power plays so well that you don't really deal with that kind of problem. There isn't that one guy who honks through the section, and sticks out like a sore thumb. No matter what they do, they always play as a *section*. If it was a normal session, I probably couldn't have gotten away with this."

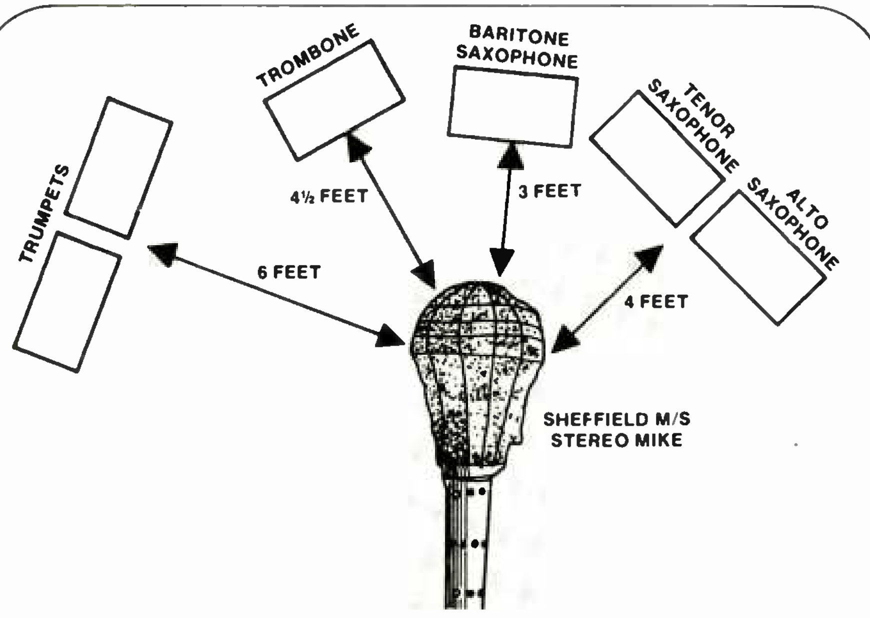


Figure 4: Instrument positions and orientations for Tower of Power's horn section during overdub sessions at Sheffield Studio, Culver City. Engineer: Larry Brown. The centrally-located Sheffield M/S tube microphone — shown here in profile, rather than side-on to the instruments — utilizes the same C12 capsule used in the AKG 251 mike, linked to a custom-designed tube pre-amplifier.

Also like his colleagues, Brown employed very little equalization. He attributes this to the quality of the Sheffield microphone [a joint development between AKG and Sheffield Labs, and which uses the same C12 capsule found in the C251, mated to a transformerless, tube pre-amplifier designed by Sherwood Sax; specs are within 0.2 dB from 15 Hz to 17 kHz, 116 dB signal-to-noise ratio, and under 0.009% THD at +10 dBm — *Ed*].

"The live record doesn't sound 'hype-y' or sound processed. We were dealing with two modified [Quantum] QM-8 boards with no EQ — just 16 faders. We brought in some UREI 500 Equalizers as outboard gear, but the Sheffield mike is, for the most part, virtually flat from 20 Hz to 20 kHz. We ended up rolling-off a teeny bit of top end — about 2 dB at 15 kHz — because the sound was just a little 'bite-y.' We didn't add any bottom at all."

The Peter Noone date, an album for Johnson Records with producer Spencer Proffer, was basically a 24-track overdub session that involved just the Tower of Power horns. Larry Brown's choice for microphones was a pair of MILAB LC-25s, which also feature a line-level output like the Sheffield, but are factory-produced. One mike is attached on the top edge of each side of a four-foot-high foam baffle (Figure 5). Both microphones exhibit a cardioid pattern facing the band. Brown refers to this as a 'pseudo-binaural' technique, because the width of the baffle approximates the size of a human head (one foot).

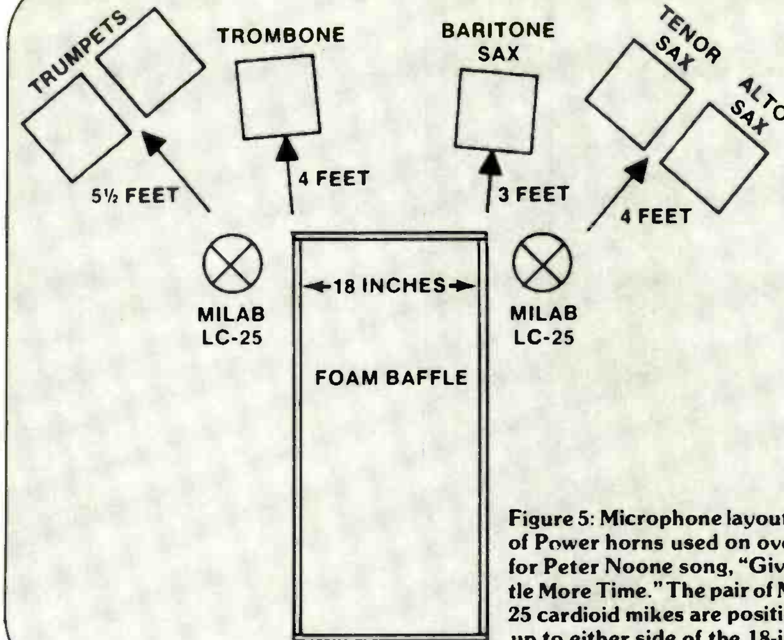


Figure 5: Microphone layout for Tower of Power horns used on overdub date for Peter Noone song, "Give Me a Little More Time." The pair of MILAB LC-25 cardioid mikes are positioned close up to either side of the 18-inch baffle. Engineer: Larry Brown.

"I had the horn section set up as they normally would for a live performance," he says, "and I let them balance within themselves. Sometimes that's not possible, like on a heavy electric session, where the engineer needs some acoustic control. In a way, I like the imaging on this set-up better than the M/S stereo, because it's a little more 'Hi-Fi.' The only complaint I got on the direct album was that, because it was recorded M/S and very natural, some people really didn't feel it was quite 'hype-y' enough as far as the left and right spread. By going with this technique, you get a really nice spreads, especially through a pair of headphones."

In the mix, the horns are panned hard left and right. The baritone sax and 'bone end up being a little mid-right and mid-left, respectively, with the biggest spread between trumpet and alto sax.

There were no distant mikes at this session at The Pasha Music House studio. "I was dealing with a pretty dead studio environment," Brown recalls. "Because of the baffle, this is a real up-front, 'hit-you-right-in-the-face' horn sound; it's like they're sitting in your lap. There's a good amount of air around the section, but it's not a big 'roomy' air — it's like a 'presenty' air. This is basically a rock and roll date, and I needed the horns to cut through a pretty popping rhythm section."

As a purist, whenever feasible he eschews as much electronics as possible by plugging directly into the tape machine. If not, he will at least eliminate the EQ and other non-essential sections of the console. "Whatever I can avoid while recording helps me get the horns back a little more real. Both the MILAB and Sheffield microphones are transformerless. And without transformerers in the board, the transient response is better. I think it's much more realistic, unless you want that softer sound, like a jazz date, or a flugelhorn solo. The specials I've done for CBS Cable, and the musicals outside of rock and roll and R&B, tend to demand a little softer horn sound. Instead of a condenser mike, I'll go to an RCA 77 or 44 ribbon on brass. The sound is a little smoother, and more legitimate. The ribbons won't clip on top like a condenser mike."

Of real concern was the fact that the condensers would clip on these Tower of

Power sessions. The level at the front of the microphone was measured at 125 dB SPL. Fortunately, Brown didn't have to attenuate the LC-25s, but the Sheffield on the direct date required the use of its internal pad.

Although Brown has used the techniques before, doubling and tripling didn't seem appropriate for Tower's tracks. "In both cases, Tower wanted a loose, live horn sound, as opposed to a tight double-horn sound. I always hear a kind of funny phasing sound from double tracking that takes away some of the realism — not so much with the reeds as I do with the brass. It's especially noticeable when the same musicians use the same horns. To get around that, I'll either have players change positions a little bit, or change horns, if they can. Maybe have the trumpets go to flugels, or whatever. A lot of people like that sound. Quite often, for rock and roll, you *look* for that phasing."

Odds and Ends . . .

Occasionally, a section will call for a blend with a flute, or a clarinet. While it is possible to record these instruments with the brass and saxes, it may not always be the most practical method. A couple of miking suggestions are submitted for approval:

In an overdub situation, Mick Guzauski positions the microphone — a Schoeps CMC-5U, if it's available, or an AKG C452 — at approximately a 45-degree angle above, and 18 inches behind the flute.

Humberto Gatica recommends a Neumann U-47 FET. The placement is also a 45-degree angle above the instrument, but about a foot in front. "In most arrangements," he says, "any time the flutes or clarinets are used, the music in the section is really soft. If they are playing in conjunction with the brass, I would still mix them all together on one track, as though they were sax parts. I'd rather go through the struggle of doing the mix at the time of the recording. Single tracks of parts never give me the same sound as if the instruments are recorded as a section."

When the arrangement calls for a clarinet and flute double on the same track (not to be confused with double-tracking), Gatica places the microphone about 30 degrees below the player's mouth, and 18 inches away. The mike is far enough from the player's embouchure to reject any wind noise, and spaced appropriately to pick up either instrument. "The arrangement will probably require a lot of space," he cautions, "and you don't want a microphone any closer than that. It helps add more depth to the music."

For television mixing, Larry Brown finds he has to go a little overboard with equalization in order for the sound to come across as being natural. "In direct-to-disk work, you're dealing with a really pure sound source, where 'less is more'; the subtleties are obvious and audible. The television medium is unnatural, unless you've lost the hearing in one ear, and you can't hear any bass or treble with the other; nobody hears in mono. The limited response means that both ends of the frequency spectrum will get lost, and the final broadcast copy will be down a few generations. I go for a little more top-end — 10 kHz and above — and boost the bottom. Echo also gets lost, so I add more of that." ■■■

* * *

About the best that any article of this nature can accomplish is to prove conclusively that audio engineering is somewhere between in inexact science, and an extremely sophisticated art form. All engineers have "tricks" that work for them and, in most cases, will readily admit that they couldn't do their job so well if it wasn't for the quality of musicianship they deal with.

Extract from this piece the information that is applicable to your individual situation, and experiment constantly. And above all, listen to the music, and your own feelings. If a track feels good, chances are it will touch someone else. That's the only secret.

STUDIO UPDATE
FACILITIES EQUIPMENT PEOPLE

Northeast:

☐ **SKYLINE STUDIOS** (New York City) has installed a Neve 8058 console and a Studer A80VU Mark III 24-track recorder, with half-inch two track mastering machines. The Neve includes custom modifications providing 40 inputs and 10 VCA sub-groups, according to studio owners **Paul Wickliffe** and **Lloyd Donnelly**. Other equipment includes 28 channels of Dolby noise reduction, UREI 813 Time Aligned™ monitors, JBL 4311 and Auratones, EMT 140 stereo plate, and EMT 240 Gold Foil reverberation, and Neve, UREI, and dbx compressor/limiters. Microphones are by Neumann, Sony, Shure, RCA, Electro-Voice, and AKG, including some old tube models. The instrument list boasts a Baldwin Grand piano, Gretch drum kit, Fender, Marshall, and Ampeg guitar and bass amplifiers, a Hammond C-3 organ with Leslie. *36 West 37 Street, New York, NY 10018. (212) 594-7484.*

SKYLINE STUDIOS

☐ **RCA RECORDING STUDIOS** (New York City) has a new MCI JH-636-36 automated console in Studio A. Purchased through Audiotechniques, the new 32-channel desk is equipped with dual microphone pre-amplifiers, which will allow for up to 72 active mike inputs. Also installed was a new MCI JH-24 multitrack. Both console and tape machine utilize differential technology for transformerless balanced inputs and outputs. Installation was handled by Audiotechniques. *New York, NY.*

☐ **LE MOBILE** (New York City) mobile recording has restructured its US operation by re-incorporating in Nashville, Tennessee; original headquarters were in Montreal, Canada. "We're working all over the country," says owner/engineer **Guy Charbonneau**, and Nashville's central location is convenient. The business address will be Nashville, but the truck itself won't be permanently based there." All booking and scheduling responsibilities will be kept in New York. The Le Mobile facility is housed in a 35-foot GMC truck, and boasts two 24-track Studer A800 recorders interfaced with a Neve console. A Studer TLS 2000 SMPTE synchronizer links the two tape machines, while the two-tracks are two Studer B67 decks. Two EMT digital reverb units are also featured, as are over 90 microphones and a video monitoring system. *Olympic Entertainment, 211 West 56 Street, Suite J. New York, NY 10019. (212) 265-1979.*

LE MOBILE

☐ **BEARSVILLE** (Bearsville, New York) is nearing completion of its Studio B. With studio design by **George Augspurger**, the new room will feature a Neve 8068 console, Studer A80 tape machines, and a UREI 813A monitoring system. *Wittenberg Road, Bearsville, NY 12409 (914) 679-7303.*

☐ **GRAMAVISION STUDIO** (New York City) features a Harrison MR3 automated board, Studer 24-track A80 Mark III with transformerless outputs, and an A80VU half-inch two track for mastering. Other equipment includes and EMT 240 Gold Foil echo plate, Teletronics LA-2A tube limiters (among others), UREI Time Aligned™ monitors bi-amped with Bryston and Audio Research amplifiers, and Shoeps and Neumann microphones. The studio was designed by **Alan Fierstein1 of Acoustilog in New York, with systems design by Michael Salafia** of Visionsound. Gramavision affiliated engineers are **David Baker**, **Alec Head**, and **John Kilgore**. *260 West Broadway, New York, NY 10013. (212) 226-7057.*

☐ **BEAR TRACKS** (Rockland County, New York) is the new facility recently opened by Crosseyed Bear Productions. The studio is equipped with an automated Solid State Logic 48-channel console, feeding two Studer A80VU MK III 24-track recorders, synchronized by an Audio Kinetics Q-Lock system. Mastering is handled by A80VU half- and quarter-inch decks. Live chambers, including a three-story stone silo, supplement three reverberation units, and a full complement of outboard signal processing gear. The facility was designed by **George Augspurger** of Perception Inc., built by **Jerry Salveson** of JLS Interiors, and equipped primarily through Visionsound Professional Audio. Bear Tracks and Crosseyed Bear Productions are owned by **Richard Calandra** and **Jay Beckenstein**, producers of the jazz-fusion group, Spyro Gyra. The studio will be limited to in-house work for the first year, with outside projects anticipated in the future. *P.O. Box 239, Tallman, NY 10982. (914) 362-0447.*

☐ **WIZARD RECORDING STUDIOS** (Briarcliff Manor, New York) has acquired a new Studer A80 Mk III 24-track recorder, giving the facility 48-track capability. *P.O. Box 25, Briarcliff Manor, NY 10510. (914) 762-3015.*

Southeast:

☐ **CRITERIA RECORDING STUDIOS** (Miami, Florida) has appointed **Chris Joyce** director of engineering; his responsibilities include overseeing the audio-maintenance department, and maintaining overall technical standards. *1755 North East 149th Street, Miami, FL 33181. (305) 947-5611.*

☐ **WOODLAND SOUND STUDIOS** (Nashville) has replaced its older Studer A80 VU Mark II recorders with the new A800 24-track machine. Additionally, a new A80VU Mk III 24-track recorder has been installed in Studio B, while the A800 will go into Studio A. According to president, **Glenn Snoddy**, there are 11 Studer decks currently in operation at Woodland. *1011 Woodland Street, Nashville, TN 37206. (615) 227-5027.*

☐ **OAK VALLEY RECORDING STUDIO** (Nashville) has added a new Studer A80VU Mk III 24-track recorder to its roster of equipment. *105 Oak Valley Drive, Nashville, TN 37207. (615) 227-9404.*

Midwest:

☐ **TRAX 32 RECORDING STUDIO** (Mequon, Wisconsin) has remodeled it facilities, including the installation of an MCI JH-652 console and an MCI 24-track machine. The new control room has 575 square feet of space, and features JBL 4343, 4301, Auratone and Advent monitors powered by SAE, Yamaha, QSC, and Sansui amps. Other outboards include a Lexicon Prime Time, Eventide Harmonizer, CBS stereo Volumax, a Systec Flanger, and dbx noise reduction. Mikes are by AKG, Shure, Sennheiser, Electro-Voice, RCA, and Audio Technica. Other tape decks include 3M 8-, 4-, and 2-track machines, and a Scully 2-track. Construction was done by **Frank Greefkes Jr.**, with installation by Milam Audio. **John Walsh** is the studio's chief engineer. *11249 North Riverland Road, Mequon, WI 53092. (414) 242-9010.*

☐ **THE CHICAGO RECORDING COMPANY** (Chicago, Illinois) has opened its third 24-track studio, with the new room slated for use in commercial, album, and audio/video sweetening. CRC's new facility is equipped with a Neve 8068 console, MCI 24-track (soon to be complemented with a Studer 24-track), a Sony BVH-1000 Type-C, one-inch video recorder for video sweetening sessions, and a BTX Shadow SMPTE synchronizing system. Also found in Studio B is a 102-year-old Bechstein grand piano previously owned by Pete Townshend of The Who, and fully renovated by CRC. *528 North Michigan Avenue, Chicago, IL 60611. (312) 822-9333.*

☐ **HUFKER RECORDING** (St. Louis, Missouri) has acquired a Sony PCM F1 Digial Audio Adapter, reported to be the first unit of its type in the Saint Louis area. Specifications for the unit include a signal-to-noise ratio of 90 dB; frequency response within 5 dB from 10 Hz to 20 kHz; distortion of 0.005% at maximum output level; and unmeasurable wow and flutter. The 16-bit binary computer code stored on conventional videotape cassettes allows two hours of continuous recording. Rates will be on a par with those charged for analog recording. *4561 Whisper Lake Drive, Saint Louis, MO 63033. (314) 741-7829.*

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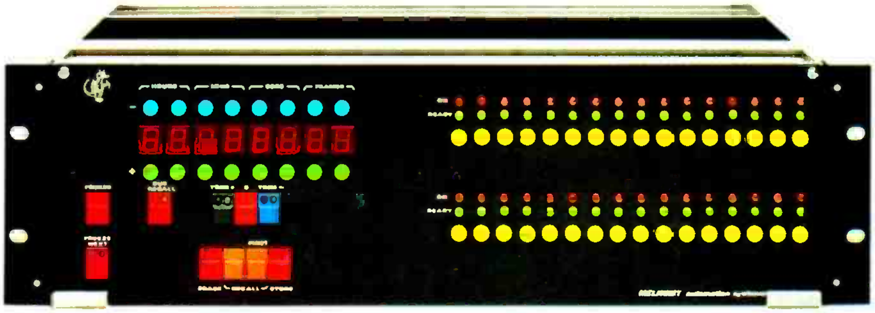

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STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE


WILLOW WIND

☐ **WILLOW WIND PRODUCTIONS** (Bartonville, Illinois) has completed construction of its new Milam Audio-designed studio. The music room features a 14-foot ceiling, an isolated drum booth, a vocal isolation booth, and live and dead areas. the control booth keys around a TEAC Tascam Model 15 24-input console feeding an MCI JH-24 16-track recorder with Autolocator, and a Tascam 80-8 with DX-8 dbx noise reduction. Outboards are by Lexicon, UREI, Sound Workshop, and Omnicraft, while monitors throughout are by JBL. The mike complement includes units by AKG, Beyer, Crown PZM, and Audio Technica, and the instrument list boasts a Kroeger Grand Piano, Ludwig drums, and guitars and amps by Gibson and Fender. The facility also features living quarters, a kitchen, three offices, and a hot tub. *P.O. Box 4189, Bartonville, IL 61607-0189. (309) 697-2434.*

☐ **SOLID SOUND RECORDING** (Hoffman Estates, Illinois) has installed a Publison DHM 89B2 stereo digital audio computer and pitch shifter, and a series of Symetrix signal gates. The instrument package has also been reinforced with the addition of a Kawai grand piano, and Ludwig drum kit. *2400 West Hassell Road, Suite 430, Hoffman Estates, IL 60195. (312) 882-7446.*

☐ **AJAX RECORDING TEAM** (Fort Wayne, Indiana) has upgraded with a Sound Workshop Logex 8 console, as well as an EXR Exciter, Valley People Dyna-Mite, and four more channels of dbx noise reduction. Also included in the update was an expansion of the main studio to include a "live area", with oak parquet flooring and curved Lexan windows. The new area incorporates Helmholtz absorbers, polycylinders, and Sonex foam. **Mike Gemmer** is the facility's new chief technical engineer. *902 West Wayne Street, Fort Wayne, IN 46804. (219) 423-3479.*

South Central:

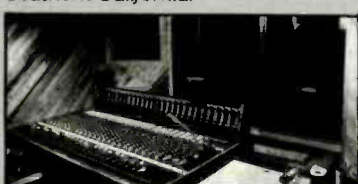
☐ **SUNRISE SOUND STUDIOS** (Houston, Texas) is the new name adopted by Sundance Sound Studios, to avoid confusion with Sundance Productions of Dallas, Texas. The current equipment list at Sunrise includes an Otari MTR-90 24-track with dbx noise reduction, an Otari 2-track mastering machine, Tangent 32-16A 28-channel console, Lexicon 224 Digital Reverb, Studio Technologies Ecoplate II reverb, Orban De-Esser, Delta Lab Acousticomputer, JBL and Auratone monitoring. Mikes are by Shure, Crown, Neumann, AKG, and Sennheiser. The instrument list includes a Yamaha grand piano, Sequential Circuits Prophet 5 synthesizer, ARP String Ensemble, and Tama drum kit. *3330 Walnut Bend Lane, Houston, TX 77042. (713) 977-9165.*


SUNRISE SOUND

Mountain States:

☐ **COMMERCIAL SOUND STUDIO** (Las Vegas, Nevada) has added an MCI transformerless 24-track with Autolocator III. Also installed are a Lexicon 224 digital reverb unit, and ¾-inch pre-and post-production video facilities. Producer **Mark Harmon** has been added as staff engineer. *2010 East Charleston Boulevard, Las Vegas, NV 89104. (702) 384-1212.*

Southern California:


THE SOUND CHAMBER

☐ **THE SOUND CHAMBER** (Pasadena) has upgraded from 16 to 24 tracks, and remodeled both the control room and the lounge area. New gear includes UREI 813 Time Aligned™ monitors, Stephens 821 24-track, MXR Digital Delay and Flanger/Doubler, and mikes by Neumann, Sennheiser, and AKG. **Randy Farrar** and **Dick McIlvery** are the studio's owners. *27 South Molino, Pasadena, CA 91101. (213) 449-8133.*

☐ **SOUND CITY** (Van Nuys) has added a new Studer A80VU Mk III 24-track, and a half-inch conversion kit for its A80 2-track mastering machine. *15456 Cabrito Road, Van Nuys, CA 91406. (213) 787-3722.*

Northern California:

☐ **HEAVENLY RECORDING STUDIOS** (Sacramento) has installed an ADR/Scamp SO-23 Auto-pan module and Lexicon Model 97 Super Prime Time. *620 Bercut Drive, Sacramento, CA 95814. (916) 446-3088.*

☐ **INDEPENDENT SOUND** (San Francisco) is now equipped with a Sound Workshop Series 30 console with VCA sub-grouping, and a TEAC Tascam 85-16 16-track. Outboards include an Eventide H949 Harmonizer, ADR Scamp Rack, and Lexicon Prime Time and 224 digital reverb. The instrument package boasts a modified Linn Drum Machine, plus Yamaha CP-70, CS-80, and Sequential Circuits Prophet 10 synthesizers. *2032 Scott Street, San Francisco, CA 94115. (415) 929-8085.*

☐ **PATCHWORK STUDIOS** (San Rafael) has expanded its 8-track facilities to include full 24/16-track recording and two-track mastering. The new equipment includes a modified Soundcraft Series Three 32/16/16 console, an MCI JH-114 2-track, AKG BX-10/II spring reverb, Lexicon Prime Time, Sound Workshop stereo reverb, Ashley parametric EQ, UREI LA-2A limiters, and 32 channels of dbx noise reduction. Microphones include models by Sony, Shure, AKG, Sennheiser, E-V, Audio Technica, and Neumann, as well as a collection of classic tube microphones. *2111 Francisco Boulevard, San Rafael, CA 94901. (415) 459-2331.*

☐ At **HYDE STREET STUDIOS** (San Francisco) the control room of Studio C has been completely rebuilt and is slated to go 24-track. Design and construction are under the direction of Hyde Steet co-owner **Michael Ward**. Also a highly modified API console, featuring sweep and graphic EQ, formerly installed in ABC's Studio A in Los Angeles, is to be installed in Hyde Street's Studio A. Other new gear includes an Eventide H949 Harmonizer, and Lexicon PCM 41 DDL, a Lexicon Prime Time, four UREI limiters, two MICMIX XL-305 reverb units, and a Lexicon 224 digital reverb unit with all updated software programs. Tape machines installed include an Ampex ATR-100, and Otari MTR-10 two-tracks. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

☐ **CORASOUND RECORDING** (San Rafael) has replaced its older multitrack with and Otari MTR-90 16/24 track machine with full remote and autolocator. Corasound plans to complete an upgrade to 24-track in the near future. *122E Paul Drive, San Rafael, CA 94903. (415) 472-3745.*

— AUDIO/VIDEO UPDATE —

Eastern Activity:

☐ **UPSWING ARTIST MANAGEMENT** (New York City), in conjunction with John Scher's Monarch Entertainment, has co-produced a tour and video project of the reunion of **John Mayall's Bluesbreakers Band.** Production services were provided by Monarch's Performance Video division with **Len Dell'Amico** directing. Unitel Video's Odyssey I unit handled the video remote at the Capitol Theater in Pasaic, New Jersey, while the Record Plant mobile truck did the sound recording. Initial release will be on RCA SelectaVision videodisk, with other markets to follow. *156 Bank Street, 2A, New York, NY 10014. (212) 242-0783.*

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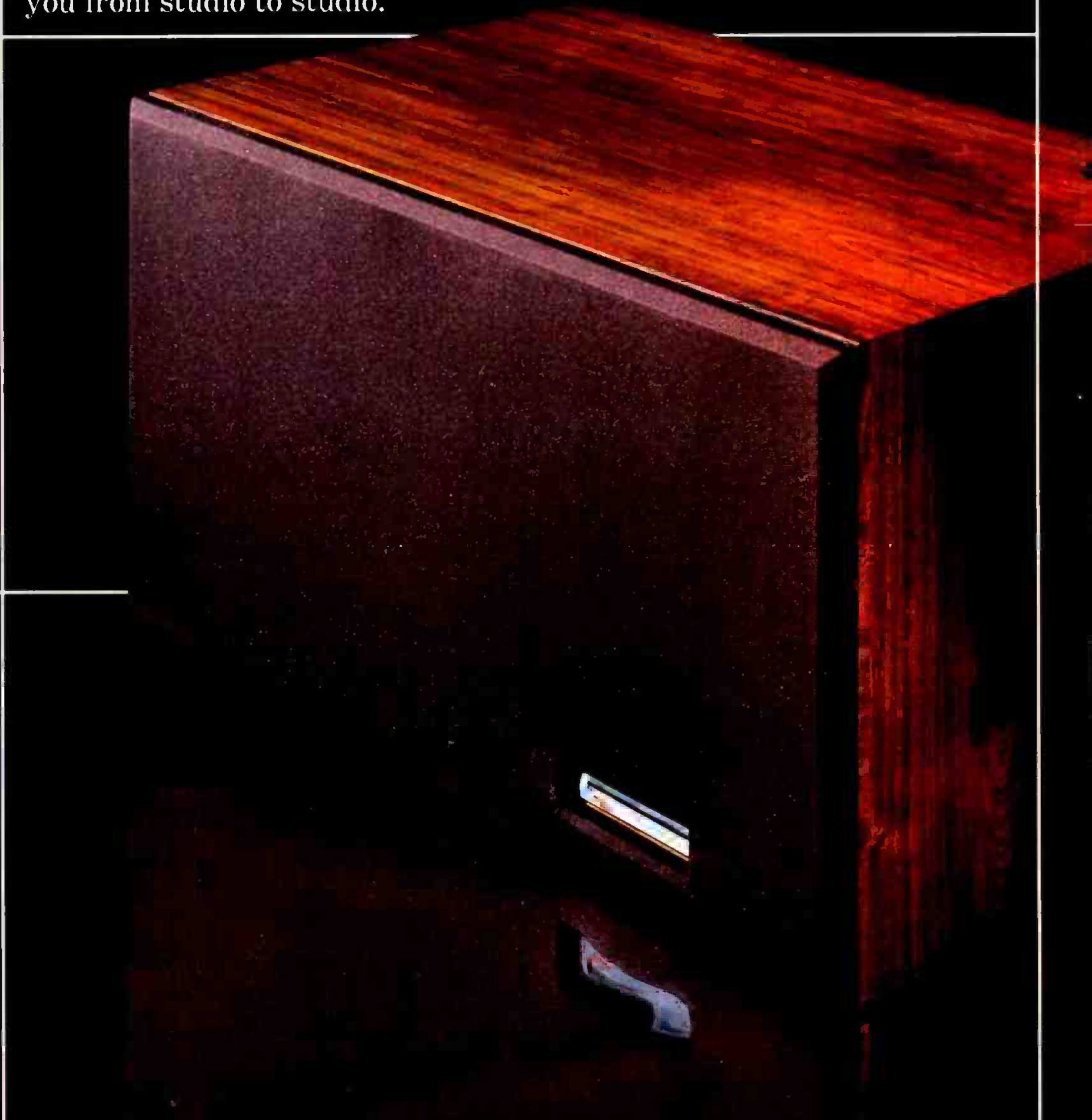

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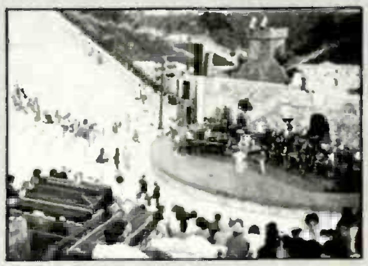
STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

☐ **SCHARFF COMMUNICATIONS** (New York City) supplied its mobile audio truck to Bill Siegler Productions during a recent two-night engagement of **Lena Horne's** "The Lady and Her Music." During production of the event for cable television's "The Entertainment Channel," duties in the mobile unit were handled by chief mixer **Blake Norton** and mix engineer **Aaron Baron**, using Scharff's modified-for-video Harrison MR 3 desk coupled with a second sub-mix console to handle a total of 48 mike inputs. Inside the house, Scharff supplied Shure SM-81, AKG 451, and Schoeps microphones for the band and back-up singers, as well as lavelier and hand-held HME wireless microphones for Lena Horne.



SCARFF IN DOMINICAN REPUBLIC

Scharff also recently sent its gear to the Dominican Republic to supply audio services for the video taping of **Frank Sinatra** and **Buddy Rich** and, two nights later, **Heart** and **Santana** at Altos de Chavon's new 5,000-seat ampitheater. Both shows were produced by Imero Fiorentino Associates for Paramount Video, and will be distributed through several pay-television services. To meet the demands of this operation, Scharff dismantled the entire contents of its truck for transport to the Caribbean via Miami, where it was flown to the Dominican Republic with the video gear. The system was reassembled on site by Scharff chief audio engineer **Gary Rotta** and audio stage manager **Bob Aldridge**. **Don Worsham** was the chief mix engineer for the projects. Concert-sound equipment was supplied by **A-1 Audio** of Hollywood. *200 West 51 Street, New York, NY 10019. (212) 247-2159.*

☐ **NATIONAL VIDEO CENTER/RECORDING STUDIOS** (New York City) has completed post-production on a two-hour **Allman Brothers Band** video project to be released as a stereo RCA SelectaVision videodisk. The program features the band in hotel room jam sessions, studio rehearsals, and in concert in Gainsville, Florida, and at the Capitol Theater, in Passaic, New Jersey. Following video editing, National's engineer **Brent Hahn** employed the facility's Q-Lock SMPTE system to transfer the 16-track final mix to one-inch video. **Lenn Del'Amico** directed for Performance Video Productions, with co-producer **Amy Polon** and **John Scher** executive producer. *New York, NY.*

☐ **ARTISAN RECORDERS** (Pompano Beach, Florida) was on hand in Montego Bay, Jamaica. to supply audio services at the video taping of the Fifth Annual Reggae Sunsplash for Synergy Productions, Ltd., and the KSR Group. The MCI-equipped GMC Motorhome was transported from Miami in a Hercules L-100 aircraft. Artisan provided simultaneous live mix audio feeds to **Trilion**, the London-based video facility, and the Jamaica Information Service for radio broadcast. **Peter Yianilos** and **Jim Fox** engineered with **Stan Strawbridge**, and **Rey Monzon**. *1421 Southwest 12th Avenue, Pompano Beach, FL 33060. (305) 786-0660.*

Central Activity:

☐ **SCENE THREE** (Nashville) has produced a video promotion piece for **Ronnie McDowell** based on his single "Step Back." The concept video piece will screen on national, regional, and local cable television, as well as in night clubs and on college campuses. The project was shot concert-style with twin Ikegami HL79 cameras and RCA TH50 one-inch video recorders, and then married in Scene Three's video editing suite into a multi-image performance, integrated with live action inserts and traveling freeze frames. **Joe Askins** edited the piece for Scene Three director **Marc Ball**. *1813 8th Avenue South, Nashville, TN 37203. (615) 385-2820.*

☐ **CELEBRATION PRODUCTIONS** (Nashville) has completed a video-music piece for the **Johnny Van Zandt Band**. The work, a promotional piece for their new album, *It's You*, commissioned by Polygram Records, was shot on location in the Agora Ballroom in Atlanta, Georgia. Production was done on 16mm film to capture dramatic lighting effects, and then transferred to one-inch video tape for editing. Release to MTV and other video-music outlets is scheduled. *16 Music Circle South, P.O. Box 24459, Nashville, TN 37202. (615) 244-5766.*

☐ **OMEGA AUDIO** (Dallas, Texas) has been busy in the video world of late, supplying audio services for the video taping of **Earl Turner** performing at the Celebrity Theater of the Le Bossier Hotel, in Le Bossier City, Louisiana. Omaga provided 24-tracks of SMPTE locked recording for the cable television special, which Omega will also sweeten in its studio. Clearwater Teleproductions of Dallas provided video facilities. Producer for the special was Earl Turner; while the director was **Giles McCreary**. Audio engineering was handled by **Paul Christensen, Ken Paul,** and **Russell Hearn**. *8036 Austin Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

Western Activity:

☐ **THE COMPLEX** (Los Angeles) recently played host to Schulman Video and **Rick James** to video tape several tunes from James' new album, *Throin Down. 2323 Corinth Street, West Los Angeles, CA 90064. (213) 477-1938.*

☐ **TELEMATION MOBILE PRODUCTIONS** (Salt Lake City, Utah) supplied video services to record the *Conway Twitty: Delta King* television presentation. The concert was recorded on the football field of Clarksdale High School, in Clarksdale, Mississippi, while a separate video crew followed the country singer around during the day of the concert to record reactions to the visit to his home town. Nashville's **Fanta Sound** interfaced its 24-track audio facility directly with Telemation's 32-foot Unit 4. The Multimedia Productions program was produced by **Jim Owens Entertainment; Steve Womack** directed. Following this operation, the Telemation crew traveled to Canton, Michigan's Center Stage Concert Hall to video tape a **Ted Nugent** concert. The video unit was again linked to a 24-track audio truck for high-quality sound. The concert was produced by **George Salovich**, and directed by **Chris Bolton**. *2117 South 3600 West, Salt Lake City, UT 84119. (801) 973-7700.*

☐ **ALCON VIDEO/FILM PRODUCTIONS** (San Francisco) has completed two music-video projects for 415/Columbia Records. Both songs were taken from Translator's new album, *Heartbeats and Triggers*. The video single, "Everywhere That I'm Not," has been added to MTV's rotation playlist, and is currently being distributed nationally by the RockAmerica Organization. The work was shot in San Francisco's 181 Club. The second piece, "Sleeping Snakes," is a composite of archive, news, and documentary footage. **Nigel Paul** and **Vinton Medbury** produced both pieces for Alcon. *950 Battery Street, San Francisco, CA 94111. (415) 397-0490.*

☐ **THE VILLAGE RECORDER** (Los Angeles) has expanded into television audio post-production and motion-picture scoring. In the film area, The Village has installed Studer and other synchronizers, which lock together video and audio recorders for scoring sessions. **Charles Bernstein** scored a 20th Century Fox film, *The Entity*, at The Village, with a synthesist and a 30-piece string section playing while the musicians viewed the action on video tape synchronized with the 24-track recorder. **Robbie Robertson**, late of The Band, is scheduled to record the score of Martin Scorsese's *The King of Comedy* in the studio. In video, The Village has linked with **Canyon Recorders**, a video post-production house owned by **Ed Lever**. The Village has leased its Studio C to Canyon on a long-term basis for the syndicated television series, *Jack Smith's You Asked For It*. "The unique aspect of this project," says Village studio manager **Joel Fein**, "is that we have installed five high-quality 'phone lines which link us to Editel, a post video facility located 12 miles from us in Hollywood. Editel is putting together the SMPTE completed half-hour programs." *1616 Butler Avenue, West Los Angeles, CA 90025. (213) 478-8227.*

☐ **SOUND SMITH STUDIOS** (Portland, Oregon) supplied 24-track recording with interlock for the video taping of a number of Portland jazz performers at the Lung Fung Dragon Room. The program was produced by **David Tower** and **Jack Santry**. Among the bands video taped were **The Rythmn Method, The Ron Stein Trio, Abrupt Edge,** and **The Keith Werner Big Band**. *426 North West 6th Avenue, Portland, OR 97209. (503) 224-7680.*

☐ **CHATON RECORDING** (Scottsdale, Arizona) had its mobile recording truck, "The Cat" in Telluride, Colorado, to interface with Visual Marketing of Denver, for the video taping of **The Rastafarians** at Telluride's annual rock and roll festival. Interface capabilities for the 24-track mobile unit are supplied via SMPTE and BTX Shadow systems. The same services were provided in Rui Doso, New Mexico, for the video taping of **Michael Murphy's** upcoming special, *What's Forever For?* The program was produced by **Mikael Arc** of One World Productions, with **Steven Moore** handling engineering duties for Chaton. *Scottsdale, AZ.*

☐ **WESTWOOD ONE** (Los Angeles) provided its mobile recording facility to handle the audio end for the video tapings at the recent US Festival in San Bernadino, California. Working in conjunction with personnel from The Record Plant, Westwood One interfaced its audio truck with the Greene-Crowe mobile video unit. SMPTE timecode was used to synchronize video tape machines with Westwood's two Ampex MM-1200 24-tracks, and an Otari 24-track in the Greene-Crowe truck. This latter deck was used for crowd reaction picked up on 12 different audience mikes. **Ed Greene** acted as chief mixer for the three-day festival, assisted by **Paul Saunders, Doug Field,** and maintenance engineer **Dave Faragher**, all from Westwood One. The bands appearing at the US Festival were video taped for possible use in a television special, or on a video cassette or disk release. *9540 West Washington Boulevard, Culver City, CA 90230. (213) 204-5000.*

☐ **THE POST GROUP** (Los Angeles) has completed post-production for **Millany-Grant Productions** on two rock and roll videos. The first stars **Billy Joel**, and features three songs from his new album *Pressure*. The second was produced for **Kim Carnes** and her new single, "Voyeur." Director on the pieces was **Russell Mulcahy**; the producer was **Jackie Adams**; and Post Group editor **Doug Dowdle**. *Production House, 8335 Homewood Avenue, Los Angeles, CA 90028. (213) 462-2300.*

Canadian Activity:

☐ **COMFORT SOUND** (Toronto, Ontario) recently made its first recording for US pay-television. The New Zealand group **Split Enz** was recorded on 24-track and video at Hamilton Place for an upcoming one-hour special on the Warner-Amex MTV Network. Other video projects include **The Motorhead** concert at The One for a CITY-TV/CHUM-FM simulcast. *2033 Dufferin Street, Toronto, Ontario, Canada M6E 3R3. (416) 654-7411.*

ON THE STUDIO TRAIL —

Mel Lambert at large this month in . . . *San Francisco*

An all too brief day's visit to the Bay Area resulted in tours of four particularly interesting studios: **The Automatt**, Pat Gleeson's **Different Fir, Russian Hill**, and **Hyde Street.**

After a conducted tour of the revamped CBS Studios that form the basis of The Automatt's Folsom Street facility, studio manager Michelle Zarin arranged a demo of the ADC monitor system recently installed in Studio A. The Meyer system, which comprises an outer pair of mid-range and high-frequency drivers, and a center pair of bass drivers (plus some of John Meyer's magic in the amplifier, crossover and phase-alignment circuitry), results in a very clean and well-controlled sound. Drums and percussion, in particular, maintained a crispness and clarity at both moderate and high monitoring levels. Automatt engineering staff also report that the room's new 40-input/32-bus Trident TSM board is particularly easy to set up and operate — due in no small measure, they consider, to the TSM's "split" rather than in-line design, which enables a producer working on the left-hand monitor section to set up control-room mixes without getting in the way of an engineer. In fact, everyone seems pleased with the Trident/Meyer combination, in terms of signal clarity, and transient response.

THE AUTOMATT

DIFFERENT FIR

Different Fir has now re-equipped with Studer A-80 Mk III multitracks and A-80 Mk III half-inch two-track deck for improved signal-to-noise and dynamic range during mastering. Engineering staff that I met during my visit were particularly impressed with the signal quality, speed of operation, and gentle tape-handling of the new A-80 multitrack transport, and comment that most sessions at Different Fir are being run at 30 IPS with no noise reduction. The facility's Harrison 40/32 console also has been modified to provide additional effects busses. The level control for monitor outputs now features a built-in push/pull switch section that enable an engineer to select the normal monitor function, *or* by engaging the switch connect the output directly to the routing section. In this way, during mixdown up to 32 additional effects sends are available from each input strip. (The next problem facing the studio's technical staff, I understand, is how to provide sufficient *returns* from each of these supplimentary output busses!). Other recent mods made to the Harrison include tri-color LEDs to indicate "plus," "minus," or "null" on each channel strip's automation display; NE 5535 op-amps at all summing points; and push/pull switches on the HF EQ section's cut/boost pot, which select shelf rather than peak equalization.

Jack Leahy, co-owner and chief engineer of Russian Hill Recording, appears to have cornered the market in jingle and commercials work, if the amount of traffic pasing through his Jeff Cooper-designed tworoom facility is anything to go by. Studio "B," which features a 28/24 Neotek III console, MCI JH-114 multitrack, and UREI 813 Time-Align monitoring, looks into a long and narrow recording area that can easily accommodate a reasonable size band or vocal group. Studio "A" offers a larger recording area, complete with drum and piano traps, and variable-acoustic side louvres. The associated control room features an L-shaped 48-in/24-out Helios console which, according to Leahy, has been sitting in storage for the past several years. It wasn't until a recent visit to San Francisco by Helios designer Dick Swettenham that life could be put back into the various modules. Besides practically having to turn away advertising business, Leahy tells me that his plan to open up in the evenings for demo sessions — at a bargain basement rate of $35.00 per hour, including engineer — has been pulling in work from far and wide.

Across town at Hyde Street Studios (the old Heider facility), Dan Alexander must have one of the finest — as well as extensive — collections of vintage tube microphones in the country. You name it, Alexander seems to have not one but two of them: Telefunken/Neumann M49, M50, M269, U47, U48, U67, KM-254 and stereo SM-2; Sony C-37A; AKG C12, C12A, stereo C24, and C61; and numerous other rare exotica. As well as offering them for sale through his used-equipment side line, Alexander also regularly makes the mikes available for sessions at Hyde Street. The vintage mike flavor carries over into the studio's recording equipment, with several elderly — but perfectly respectable — Ampex and 3M stereo and multitracks at various locations around the facility, as well as LA-2A and Pultec MEQ5 tube equalizers, and vintage consoles. Incidentally, while wandering past the maintenance department I couldn't help but notice a bread-boarded project under construction on the bench. It turns out that Don Kruse was just putting the finishing touches to a spectrum analyzer circuit culled from one of Ethan Winer's articles in a recent issue of *R-e/p*, and planned to use it to test out some equalizer modifications he is currently working on for the studio's Trident A-Range console. Kruse says that the circuit powered up okay, and works just as predicted; he also plans to use the analyzer to check room EQ. ■■■

PEACE SUNDAY FESTIVAL PASADENA ROSE BOWL

by David Sheirman

All photography by David Scheirman

MSI-Northwest's Cooperative Sound System Design for Stevie Wonder, Jackson Browne and Friends

Sitting in the audience at an outdoor rock festival used to be quite a gamble. The sound system *might* function correctly all day long; the set changes would *maybe* take less than an hour; and it was even *occasionally* possible to hear the music at the rear of the seating area. In recent years, improvements in concert-sound system technology, coupled with the ever-increasing expertise of sound companies and their technicians, have all helped to make outdoor shows an enjoyable experience.

This writer recently observed what justifiably could be called a "state-of-the-art" outdoor festival; two sound companies cooperated to provide audio services at the Rose Bowl, Pasadena, last June for 94,000 people who listened to a dozen rock groups, three choirs, and countless guest speakers at an event known as Peace Sunday. the well-orchestrated production featured three separate stage areas, rolling risers and duplicate house mixing positions.

The Event

This particular festival offered benefit performances by an array of well-known musicians, including Stevie Wonder, Jackson Browne, Linda Ronstadt, Crosby, Stills & Nash, Dan Fogelberg, Stevie Nicks, Joan Baez, and Bob Dylan. Special guest appearances were put in by Dave Mason, Donovan, Tom Petty, and Gary "U.S." Bonds, along with Timothy Schmit of the Eagles. With such a large number of notable performers on hand, each of whom is used to being given preferential treatment when travelling with his or her own show, it was important that the stage setup be arranged to allow quick set changes, and maximum flexibility. Several groups had rather complex sound requirements. Staying on schedule meant a lack of time for the usual sound checks and preparations, however. The situation presented an interesting set of logistical problems, particularly in the area of stage monitoring.

To solve these problems, and to help ensure that the show would go smoothly, the event's producers retained Richard Irwin as sound coordinator for the festival. Irwin is a Los Angeles-based audio engineer with many years of touring experience behind him, mixing such acts as the Eagles and Boz Scaggs. It was his job to assemble sound requirements for the constantly-changing lineup of guest performers, to put together a master audio staging plan, and to make arrangements for procuring the sound system.

Richard Irwin chose the newly-formed coalition known as MSI-Northwest to be audio contractor for the event. Bob Goldstein, owner of Maryland Sound Industries and Bob Stern of Northwest Sound recently pooled some of their technical resources specifically to serve the outdoor show market for the 1982 touring season. Both companies have experienced tremendous growth over the last decade, and the mutually-beneficial arrangement makes available a large amount of audio gear for outdoor events. Last June's show at the Rose Bowl in Pasadena was to be the first such event done on a cooperative basis.

With less than a month's advance notice, the two companies began their initial planning for the Peace Sunday event. For the production to be cost-effective, both companies planned to rely on touring systems that were already in the general vicinity, additional truckloads of gear being brought in from warehouses on the East and West coasts if needed.

Northwest Sound would cover the Rose Bowl with its time-tested four-way house system, and provide stage monitors for the main and thrust stages. Maryland Sound would provide the extensive stage monitoring facilities required by Stevie Wonder, and also bring in a duplicate house mixing setup with which to handle Stevie's set, and the auxiliary performance area in the stands that was to feature three large vocal choruses.

System Set-Up

Initial load-in for the event began on May 31, six days prior to the show. United Production Services brought in one of its modular outdoor stages, and provided high-rise scaffolding for the sound wings. The main staging area was approximately 60 feet wide and 40 feet deep, with side extensions for case stor-

age and monitor boards. An additional 30-foot area was set upstage in the center to provide storage space for a rolling stage that was to be filled with gear for Stevie Wonder and his band, Wonderlove. The main stage area would be set with gear for Jackson Browne and Crosby, Stills & Nash, and plans were for most other groups to use the same space and equipment.

Additionally, an 8- by 20-foot "thrust" stage was assembled in the downstage region, and placed 3 feet below the main stage level. This stage was to be used by a seemingly endless parade of guest speakers, announcers, celebrities, and film stars, along with the "acoustic" acts.

The stage and sound wings were completely assembled and secured by Thursday evening. Scaffolding on the wings was erected with adequate height to allow for the vertical stacking of speaker cabinets by chain-motor hoists, which would be attached to steel beams laid across the top of the scaffolding towers. By Friday morning, the Northwest Sound and MSI gear had arrived by tractor-trailer truck from various points of origin. Half of the house speaker system was placed, while set-up of the balance of the system had to wait for the truck to arrive from Dan Fogelberg's concert being held in Irvine, California, the night prior to Peace Sunday. Maryland Sound's gear came in from shows with the Village People in Las Vegas, and from the Al Jarreau Tour.

Power for the event had to be supplied completely by on-site generators; the Rose Bowl was never intended to host outdoor rock concerts, and no provision has been made for massive AC power hookups when the facility was built. Two truck-mounted diesel generators arrived and began paying out their heavy copper cable. The units were parked out of sight in the tunnel behind the backstage area, so many yards of 600 MCM wire were required. Breaker panels were placed directly behind the upstage edge of the stage, and Northwest and MSI engineers each did a separate hook-up of their respective power distribution systems.

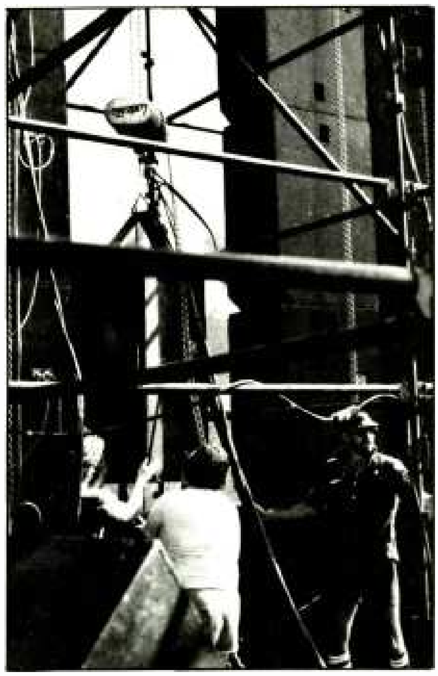


Figure 1: Northwest speaker cabinets being hoisted into position through the 35-foot high scaffolding towers using CM-Lodestar chain hoists. Heavy-duty nylon straps were used to link cabinets together in situ.

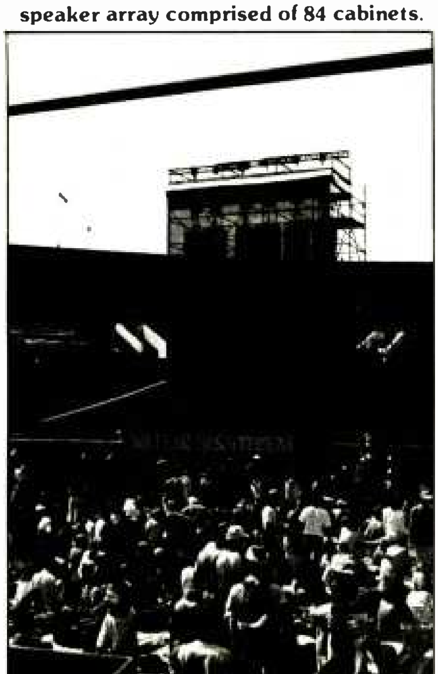


Figure 2: Each sound wing housed six identical vertical columns of six Northwest cabinets. Scaffolding towers were draped in decorative blue fabric. The entire speaker array comprised of 84 cabinets.

Northwest Speaker System

The Northwest cabinets were pulled up through the 35-foot scaffolding towers by CM-Lodestar chain hoists (Figure 1). As each cabinet is lifted off the deck, another is rolled into position beneath it, and secured to the first cabinet by means of heavy-duty nylon straps. Hooks on these straps are fastened to steel plates built into the sides of the cabinets. The speaker boxes were hung in vertical rows of six, and each sound wing housed six such columns (Figure 2). An additional group of six cabinets per side was stacked two high on the stage, bringing the total number of cabinets in the house system to 84.

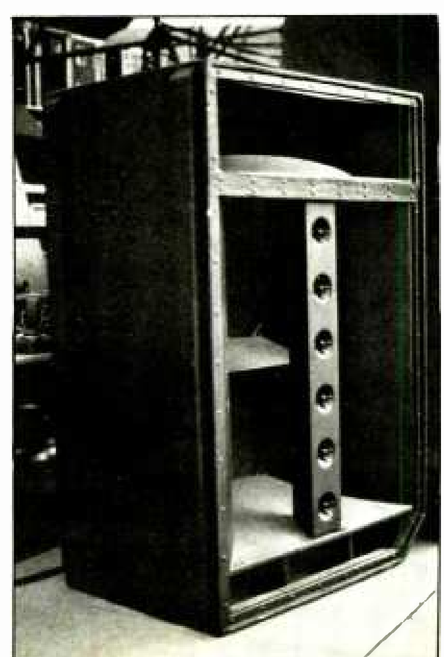
The vertical columns of cabinets were hung in pairs, with two chain motors being secured to each steel beam. The two inner pairs faced straight out into the audience seating area, the outer pair of columns being angled away from the stage towards the sides of the bowl at a 60-degree angle. Low-end supplementary cabinets, known as "Bass-Aug" boxes, were stacked five per side in one vertical column to create a strong line array of 30 15-inch drivers on each sound wing to carry the low-bass signal. All cabinets were built basically to the same dimensions, and all horns and tweeters were contained in the cabinets. No additional long-throw horns were placed.

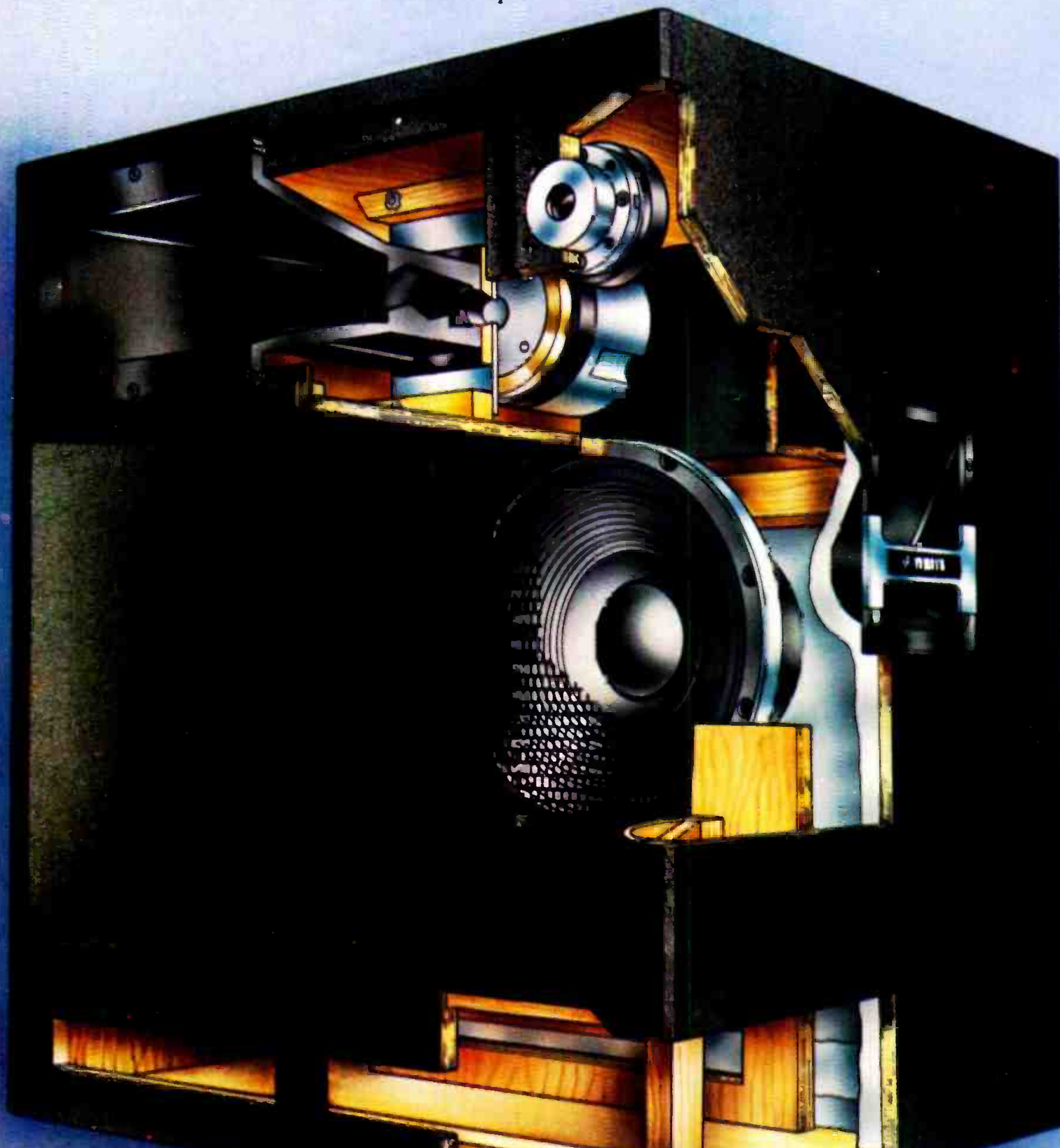
Three basic types of cabinet were provided. The three-way Model 590 (Figure 3), which houses two 15-inch JBL 2220-B 16-ohm cones, a JBL 2440 driver mounted on a Northwest Model 350 fiberglass radial horn, and a vertical column of six Motorola piezo-electric tweeters, is intended for use as a full-range cabinet, or as the high-end complement to the Bass-Aug boxes. The Model 590 weighs 225 pounds, and features a fiberglass horn insert for the low-end section. The Bass-Aug has essentially the same 4- by 3- by 2-foot dimensions as the 590, but is 31 inches deeper due to its increased internal volume. It contains six front-mounted, 15-inch TAD speakers loaded in an infinite baffle.

The third type of Northwest cabinet is, to this writer at least, an exceptionally smooth-sounding two-way cabinet loaded with a pair of TAD 15-inch cones, and a TAD high-frequency driver with a two-inch throat mounted on a Northwest fiberglass 90-degree radial horn with balsa-wood phase plug inserts. These components are housed in a cabinet with the same dimensions as the Model 590 three-way box.

All of the Northwest cabinets have one very practical feature: two built-in wheels mounted on the bottom side, with the box made so that the wheels do not touch the floor until the box is tilted

Figure 3: Model 590 three-way box, which houses two 15-inch JBL 2220-B bass units, JBL 2240 driver on a Northwest Model 350 fiberglass 90-degree radial horn, and six Motorola piez-electric super tweeters.





back at a 45-degree angle (Figure 4). A solid, easy-to-grip handle is built into the top panel to provide ease of handling, and a recessed connector plate set into the back panel.

Power Amplification

The Northwest system is powered by three different models of Marantz amplifiers. Low-end is pushed by the Model 510, which produces 450 watts per channel. The system is set up so that each side of a 510 is powering two of the 15-inch speakers in a cabinet. The Bass-Aug rack consists of five Model 510s, whereas the standard rack contains two Marantz Model 240 amps to power the horns, and a Model 140 to run the tweeter units, along with three 510s. The system used to cover the Roser Bowl contained 11 amplifier racks, capable of producing an estimated total of 40,000 watts.

The amps were housed in a practically indestructible rack (Figure 5). In fact, some of the ones in this system were 10 years old, and still looked sturdy. As Northwest engineer Steve "Lance" Mazy put it, "We feel this is a very good design. The racks we are putting together now are the same as the original design from 1972 or so. Some of them have suffered unbelievable abuse, fallen out of trucks and stuff, and they've held together real well."

A rigid metal frame is covered with impact-resistant panels, and the removable front and back covers are secured by recessed cam-lock latches. (The side panels are also easily removed

Figure 4: Two wheels built into the bottom of each cabinet, combined with a handle cut in the top, simplifies movement of the cabinet during loading and set-up.

Figure 5: Northwest amplifier racks contain Marantz Model 510, 240, and 140 units housed in a rigid external metal frame, with impact-resistant front, side and rear panels.

for cooler operation in hot, outdoor sun.) Each rack is mated to a lightweight aluminum, two-wheeled handcart that has a nylon strap on a roller mechanism. A ratchet-type crank secures the strap around the rack for easy removal. It is a rather heavy unit, but the two-wheeler allows one person to easily roll a rack from the truck across the stage to the sound wing. The combination makes up what is probably the best-protected amplifier rack I have ever seen, and it may be the fastest to set up.

Northwest Front End

The main system was overseen by Ed Wynne, a long-time Northwest veteran of such tours as The Grateful Dead and Joni Mitchell, and who has been using it in various evolutionary forms since it was first conceived. Probably one design idea unique to the system is the fact that the crossovers and equalizers are mounted in the amp rack. According to Wynne, "Every single amp rack has its own electronic crossover and third-octave graphic equalizer. This greatly simplifies wiring up the house mix position, and reduces signal loss due to distance between the crossovers and the amplifiers. It also obviously cuts down on the number of lines running back to the stage from the house console."

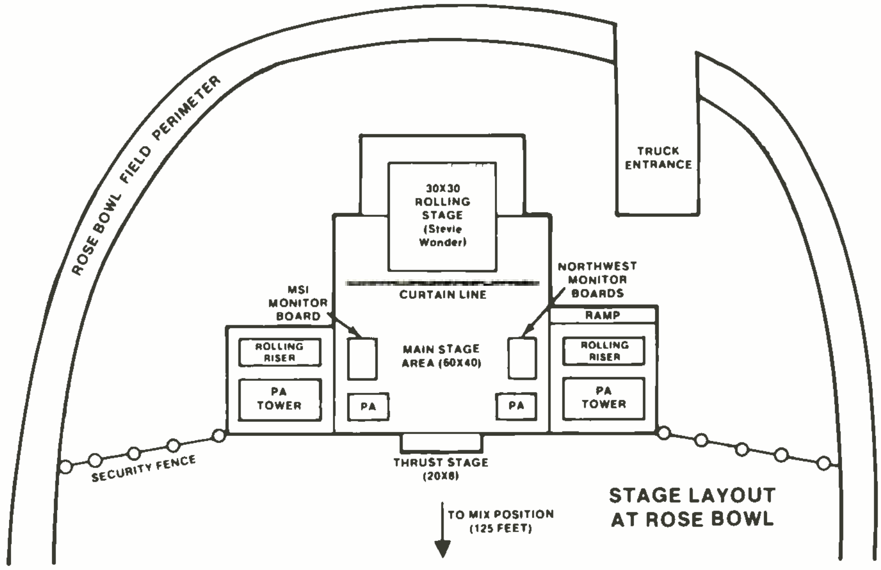
Another advantage is that the system has a built-in fail-safe mechanism: instead of relying on only one or two crossover units for the entire system, there are many; should a crossover ever fail, the output signals from the adjacent rack can quickly be jumpered over to the rack with the faulty unit.

System crossover points are 140 Hz and 1.2 kHz. The Bass-Aug cabinets receive the lower three octaves or so of signal, and the horn-loaded pairs of 15-inch speakers carry the signal up into the vocal range. The radial horns are given a signal which runs all the way on out from 1.2 kHz, and the piezo-electric tweeters are passively crossed-over. They are protected from all frequencies below 9 kHz or so by in-line capacitors.

Outboard processing devices included a Lexicon Delta-T digital delay, a Lexicon 224 digital reverb unit, an Eventide Harmonizer, and a Publison DDL/pitch-shifter and effects device. Situated within easy reach on top of the two Yamaha PM-2000-32 consoles were eight API limiters and four API noise gates, the limiters being patched into the individual vocal channels, and the noise gates used on the kick drum and tom channels to help reduce stage noise leakage.

Northwest was an early supporter of the PM-2000-32 console, and worked closely with the manufacturer when it was still in the prototype stage. For this show, the two consoles were tied together, giving an input capacity of 64 lines (Figure 6). Two input snakes carried the microphone signals out to the console position, which was located relatively close-in at 125 feet. The left and

Figure 6: A pair of Yamaha PM-2000-32 consoles provided a total of 64 mike channels for the main-stage sound mix; house engineer Ed Wynne at the controls.



STAGE LAYOUT AT ROSE BOWL

right main output signals were processed through dbx Model 160 compressor-limiters; Yamaha Q-1027 third-octave graphic equalizers were also placed in-line at the board for use by the engineers. (The EQ units in the amp racks are of the "set it and leave it" type; the few system "peaks" that are present in most any loudspeaker system are tuned out, and day-to-day equalization for a particular show can be done from the mix position).

The house-mix consoles were set to handle all program material coming from the main stage area. With nearly all artists using the same basic equipment set-up, channels on the console labelled for a particular instrument remained constant for the entire show, which helped reduce the time-consuming process of resetting levels and equalization each time a new group came out. Oftentimes the same musicians were on stage with many of the guest artists, so the sound of the show was actually quite consistent from act to act.

House Engineers — Main Stage

Much of the consistency of the show's sound was due to the fact that Northwest engineers had already handled many of these performers as clients in the past. Jesse Colin Young and Crosby, Stills, & Nash have had a long-term relationship with the company. Dan Fogelberg was in the middle of a tour with Northwest, and had his regular engineers there, including "Snake" Reynolds, who also had logged many shows with Timothy Schmit and the Eagles. And Jackson Browne and Linda Ronstadt are both taken care of by Buford Jones, an independent engineer from Dallas, Texas.

Jones offers some interesting insights into how he approaches his job as a concert sound engineer: "Usually, I will have about a month's notice . . . the preparation is very important beforehand. After a group starts working on the new album, I'll receive a copy of the mix, and start thinking about how to approach the material. With Jackson [Browne], I was very fortunate on the last album to be able to sit with him in the studio in Los Angeles for the album mix before the tour started."

Sharing the system as he was with other engineers, Jones had to make do with a few compromises here and there, but basically he was able to pre-set his primary input channels at sound check, knowing that they would remain untouched until he gained control of the consoles. With 64 input channels at the mix position, each of the major acts had some input modules which were not used during any other set. For Jackson Browne, one thing to which Jones paid particularly close attention was the acoustic piano: "Picking up a grand piano outdoors can be difficult at times," he noted. "Here at this show, we are using the Countryman set-up . . . I find that it gives a good, 'full' sound. The Helpinstill is also a fine pickup, but takes longer to get set up. I like to try to place mikes as well, putting them inside the piano directly over the hammers to get that true percussive attack. Lately, I have been using an Aphex [Aural Exciter] unit on the microphones to increase the brightness, and help the piano really cut through the mix. I bring that signal in over the full-bodied sound of the pickup, and it works really well."

Main Stage Set-Up

During the course of the 12-hour show, the center stage area received the most use. Due to the extremely short intervals allowed for set changes, all heavy pieces of gear such as the keyboards and amplifiers were placed on rolling risers, which could be brought into position from the left and right wings. The stage set for Jackson Browne, Linda Ronstadt and Crosby, Stills, & Nash consisted of a drum kit, percussion set, two keyboard positions, including grand piano and Hammond organ, six different guitar amplifiers including pedal steel and bass, and five vocal mikes. This basic set-up was used by nearly all of the groups and artists on this stage. Guests such as Tom Petty, Dave Mason and Bonnie Raitt often walked out practically unannounced, causing the engineers to occasionally scramble for unused channels.

Northwest Monitors

With such a fluctuating guest list on this stage, the basic monitor strategy chosen was to set out plenty of boxes, most on separate mixes. According to Northwest's monitor engineer Sandy Battaglia, "I have 12 slants out there. For Jackson Browne we are using eight mixes: his downstage mix; piano mix; one each for the guitarists and bassist who sing; one for the keyboard position; drum mix; and an open one for the guest artists."

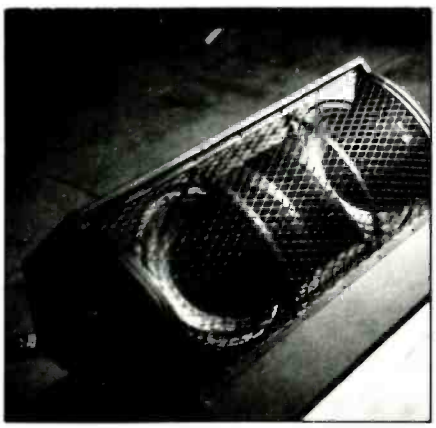


Figure 7: Model 671 floor slant houses a pair of TAD 15-inch speakers and a TAD high-frequency driver in a custom-designed cabinet. Frequency response is said to be flat enough to not require outboard EQ.

The Northwest monitor system was centered around a Yamaha PM-2000-32 console, which was "stock, right out of the box . . . ," according to Sandy. "However, we do one minor modification," he offered. "We remove a small jumper inside, which changes the foldback sends to post-EQ, as opposed to pre-EQ. And we normally run our mixes without any outboard EQ . . . our boxes were designed not to need it. However, today I am supplying third-octave graphics [Yamaha Q-1027s] on each mix . . . it makes the engineers who have never used this system a bit more comfortable."

The slant monitor used by Northwest is known as a Model 671, and comprises a unique box housing two 15-inch TAD cone drivers. These are front-mounted, and set close on either side of a custom-engineered, fiberglass bi-radial horn backed with a TAD high-frequency driver. The cabinet has a very distinctive appearance due to the curved metal housing placed over the front of the box to protect the speaker components (Figure 7). Reinforced steel bands help support the metal grill, and the box has hand-holds cut into the sides.

Northwest engineer Dennis Darby attributes the design of the 671 slant monitor to a cooperative effort by the company's engineers. "It was basically a state-of-the-art development in the fall of 1978," he recalls. "We first put these boxes out on the Eagles tour, and they loved them; it is an extremely flat box. As a matter of fact, it was on that tour that we decided we did not really need the additional stage noise that comes from sidefill monitor stacks. These slants are so clean that sidefills just are not necessary."

As Darby pointed out, no sidefills were being used for the Rose Bowl main stage. Bear in mind that this was outdoors, with reasonably loud groups playing on a large stage. The 671s did indeed put out a very clear, high-level undistorted sound, and there seemed to be no complaints from the performers.

The monitor mixes were all bi-amplified using Yamaha crossovers set at 1.2 kHz. dbx Model 160 compressor-limiters were available for patching to input channels, and stereo lines were returned from a Lexicon 224 digital reverb unit to get a sweetened vocal signal out to the stage.

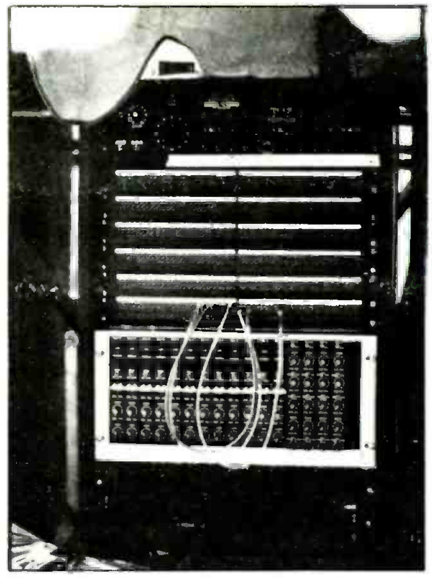


Figure 8: Maryland's monitor electronics rack contains crossover/limiter circuit cards for bi- and quad-amp mixes.

Microphones used on the main stage included Shure SM-57s for vocals, guitars and snare. An Electro-Voice RE-20 was placed in front of the kick drum, while phantom-powered AKG C-71s covered the toms, hi-hat, and overhead cymbals. The percussion set-up was picked up with more 57s, and all electric keyboards were taken by direct boxes, as were the acoustic guitar and

Syn-drums.

Stevie Wonder's Stage

The rolling stage which held band gear and monitors for Stevie Wonder was a completely separate entity from the main stage. Hidden from the public's view by a large, bright blue curtain that remained closed until time to bring the rolling stage out, all cables for the rolling stage were coiled at the edge, ready to be connected quickly. According to MSI's "Jeep" Parker, the entire changeover from the main stage to the rolling stage had to be accomplished in a matter of minutes.

Microphones selected for this stage were of a higher quality than is often seen at an outdoor show. Stevie Wonder's personnel had originally requested Shure SM-58s for the vocals, but MSI engineers successfully substituted one of their favorite vocal microphones: the Sennheiser 431. According to Jeep, "We find this mike to be an excellent choice for vocals. It's good and bright, and has excellent background noise rejection characteristics. I always try this mike first on featured vocalists."

For the horns, Electro-Voice RE-20s with shock mounts were put up. Guitar amps were fitted with Sennheiser MD-421s, as was most of the percussion setup. On the drum kit: Beyer 88 for kick; Shure SM-57 snare; Beyer 201 hi-hat; Sennheiser 421s on toms; and AKG C451 set for overheads. Twelve Countryman direct boxes were used on electric keyboards, which included a Yamaha CS-80, Stevies's Clavinet, a Fender Rhodes, and a host of synthesizers. The acoustic piano was given a Helpinstill pickup, and also a relatively new pickup from England known as the "C'Ducer." In addition, a Crown PZM was mounted inside the lid.

Figure 9: MSI provided Stevie Wonder with Meyer Ultra-Monitors at his on-stage keyboard position. A custom-designed phase-correcting crossover and self-analyzing line amplifier with overload circuitry is supplied with each pair of cabinets.

MSI Monitors

Having supplied a system for Stevie Wonder in the past, MSI engineers were not unfamiliar with the situation, which has ranked as one of the more complex monitoring systems used by any touring group. The original set of sound requirements called for 24 discrete mixes, 64 channels of input, and 21 separate floor slants. After many preproduction meetings, the decision was made to reduce the system somewhat in the interest of keeping this particular show on schedule.

"We had five days of rehearsals with Stevie's band a week before this gig," says Jeep Parker. "By the time we finished rehearsals, we were down to 18 mixes and 60 inputs. For the show here we are actually using 14 discrete mixes plus stereo sidefills, and we're taking 54 inputs."

The Maryland Sound stage system used two Yamaha PM-2000-32 consoles, the pair having been modified to put out 14 discrete mixes. The sliding fader in each input was replaced with eight miniature, rotary volume pots, so that each channel went directly to eight mix outputs rather than being assigned to submasters. The two boards were ganged together, and each mix processed by a Yamaha Q-1027 graphic equalizer. Twenty such units were available in the system, spares being patched into individual channels if extensive EQ was needed.

The equalized mix signal was sent to an MSI-designed line-driver circuit card that housed a limiter and electronic crossover (Figure 8). The card has an internal variable-threshold adjustment for the limiter, and the crossover point may be altered by substituting different chips. This system used a 1.6 kHz crossover point for the bi-amplified mixes. Twelve bi-amp cards were available, and eight quad-amp cards for drum monitors and sidefills. Mixes for Stevie Wonder were as follows: two horn mixes; two guitar mixes; bass vocal; bass amp (used in place of a stage amp); drum; percussion; two keyboard mixes; background vocal; Stevie's vocal; and Stevie's piano.

Most mixes received the MSI 2X12 floor slant, which is a ported cabinet containing two rear-loaded 12-inch JBL or Gauss speakers, and an Emilar HF driver mounted on a JBL perforated lens. The four background singers each had their own single-twelve ported monitor cabinet, which had an internal passive crossover with a variable high-frequency level control.

For Stevie Wonder, MSI engineer Harold Blumberg had placed a Meyer Ultra-Monitor on either side of Stevie's seat at the keyboards (Figure 9). Each pair of Ultra-Monitors comes with a phase-correcting crossover and self-analyzing line amplifier with overload-sensing circuitry. As the output of the cabinet approaches its maximum safe level for a clean signal, the circuit senses overload and begins to attenuate the low-end and the very high frequencies. This allows the engineer to keep a very strong signal going into the box without encountering a loss in audio quality.

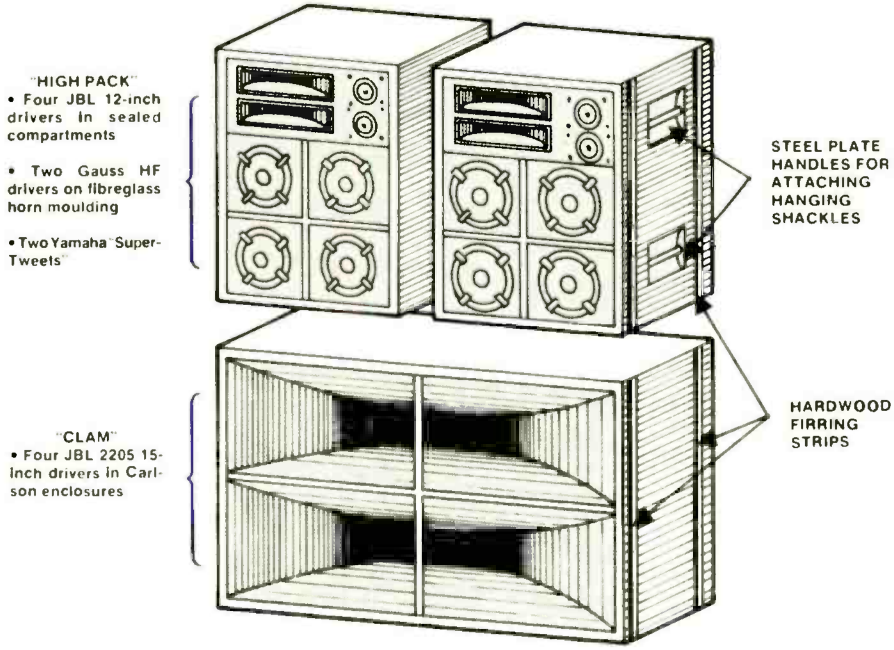


Figure 10: Sidefill monitor stacks for Stevie Wonder consisted of two "High Packs" and a single "Clam" bass bin/Carlson enclosure set up on either side of the stage area.

Meyer cabinets are said to have finally allowed Stevie Wonder to hear his music live as he has wished to; with an extremely accurate reference monitor onstage, he has begun to experiment with playback tapes and exotic effects devices in the monitors to give him the sense of actually recreating the sounds of his albums live in a concert setting.

Unlike the main stage set by Northwest, Maryland Sound did utilize sidefill monitor stacks, which consisted of an MSI "Clam" enclosure (so named because the elliptically-curved front baffle-board in each section of the cabinet is reminiscent of an open clamshell), and two "High-Packs" on each side (Figure 10). The Clam houses four separate Carlson-type speaker chambers, each loaded with a single JBL 2205. The chamber is adapted by MSI from the original Carlson speaker cabinet that was first designed in the Thirties for home hi-fi use, and which contained a single 12-inch speaker.

The High-Pack is loaded with four 12-inch JBL speakers, and two Gauss HF compression drivers loaded on fiberglass 90-degree radial horns; next to the horns are two Yamaha "Super-Tweets." Set up to run as a four-way system, these sidefill stacks were crossed-over at 180 Hz, 1.6 kHz, and 9 kHz.

Amplifier Racks

Both the sidefill stacks and the floor slant monitors were powered by the standard MSI amp rack: a heavy external metal frame with polished oak side panels containing three Crown PSA-2s to drive the cone speakers; a single Crest Model 3500 for the mid-range drivers; and one Crest Model 1000 for the HF units (Figure 11). The amplifier rack is equipped with a unique patch-panel on its rear access plate. Patch points are provided with Telex mini-connector jacks so that any input signal may be switched to any amplifier channel with Switchcraft patch cables. The patch-points are normalled so as to operate in the standard multi-signal mode until the line is broken by inserting a cable.

Figure 11: Amplifier racks for the four-way monitor system housed three Crown PSA-2s, a Crest Model 350, and a Crest 1000. A mini-connector patch panel is also provided on the rear access panel.

Figure 12: Joan Baez and Bob Dylan on the forward thrust stage, with Northwest 2×12 floor monitor slants. A rear curtain shielded the main stage area during set changes.

Additional Staging Areas

The thrust stage in front of the main area was treated as a separate production environment. A curtain was suspended in front of the main stage work area while the thrust stage was in use to hide set changes. Here guest speakers introduced groups, and solo acoustic acts performed. A six-pair snake carried lines over to a Yamaha PM-700 mixer, which supplied a submix to both house consoles, and a foldback mix for the Northwest 2X12 slant monitors placed on the front edge of the thrust stage (Figure 12). This area was difficult acoustically for both performers and engineers, since the platform was actually set up in front of the main stage line.

Another area apart from the main stages that required microphones was a large section of the seating region which had been roped off. This section hosted three different 50-voice choirs at various times, and an 8- by 12-foot riser had been installed for a Hammond B-3 with Leslie cabinet and a bass guitar amplifier. Six Sennheiser 441 dynamic microphones were suspended above the choruses; condenser mikes were tried at first, but the engineers yielded to excessive wind noise and put up the dynamic cardioids instead.

MSI House Console

Directly next-door to the Northwest house mixing position, MSI had set its 64-channels of inputs on the same riser.

From the stage, a 54-pair and a 24-pair snake cable brought lines out to a couple of Harrison Alive 32-channel boards (Figure 13). Rather than having a massive front metering panel protruding above the console, Harrison utilize unobtrusive LED displays in the center of the unit above the eight VCA group faders. The mix of Stevie Wonder's set was done in stereo and sent directly to the Northwest PM-2000-32, after having been passed through a pair of Yamaha Q-1027 third-octave graphic equalizers and a dbx Model 165 Over-Easy compressor-limiter.

Output functions of the main Harrison were set up as follows: mix group "A" left and right went to a dbx Model 162 for processing the vocals; group "C" left and right were assigns to the main output feeds; and group "D" left was sent to an Eventide H910 Harmonizer, which was used as a stereo synthesizer. Stereo returns from the Harmonizer were put into two channels on the console, and panned left and right respectively. A small amount of delay (30 milliseconds or so) added to the input signal gave an effect that greatly increased the spatial dimensions of whatever channels were assigned to this bus. Additionally, the console has eight auxiliary outputs through which each channel can be routed in a pre- or post-EQ mode. Several of these busses were used to get into the URSA Major SST-282 Space Station, the MICMIX Master-Room XL305 rack-mount spring reverb unit, and an EMT Model 250 digital delay.

Figure 13: A pair of 32-input Harrison Alive consoles with VCA subgrouping provided 64 channels for Stevie Wonder's stage.

The MSI equipment racks were built in the same fashion as the amp racks; they had sturdy metal welded frames with solid oak side panels (Figure 14). The racks interconnect with the console and each other by means of heavy-duty, 52-pair multicable with quick disconnects. The front panel of the rack contains a mini patch panel that enables any device in the rack system to be bypassed; this also provides easy access to all lines coming and going from the rack. In addition to the above effects units, the racks contained a TEAC cassette deck for playback, and a Technics for recording. Also housed were Audio-Arts Model 1400 electronic crossovers; the MLE-4000 limiter-EQ panel built by MSI, which contains four sections of variable compressor-limiting and four-band equalization; and power supply for the Harrison console.

The Maryland sound house position was watched over by engineer Al Tucker, who seemed to be perhaps emotionally involved with the Harrison consoles. "This is the only way to go," he exclaimed. "I love it. The difference in audio quality when using VCA grouping as opposed to ordinary submasters is really noticeable. The board is extremely versatile, and it's a joy to use."

Similar sentiments were expressed by Paul Devilliers, a Canadian engineer hired by MSI specifically to mix Stevie Wonder at the stadium shows this summer. "I am quite used to the Harrison consoles due to my work in the studio," he says. "I feel that the sonic quality of the board is excellent, and am quite pleased to see them here. Just like Harrison says in their advertising: 'No Compromises.'"

Devilliers commented on his approach to satisfying Stevie Wonder's desire to reproduce the sound of his albums in concert: "Basically, whatever it takes to get the sound right, that's what Stevie is into," he offers. "For instance, a Sony PCM-10 digital audio processor is [used] for playback of a pre-recorded soundtrack, which contains tracks of a 15-piece horn section, and some extra percussion which we use on one of the tunes; I send this signal up to the stage also, and the drummer gets in sync with it by hearing the count-in on the track. Stevie also supplies an EMT-250 digital reverb unit, which I use constantly on the horns, persussion and vocals to get a 'plate' sort of sound . . . a particular reverb effect similar to what Stevie does in the studio."

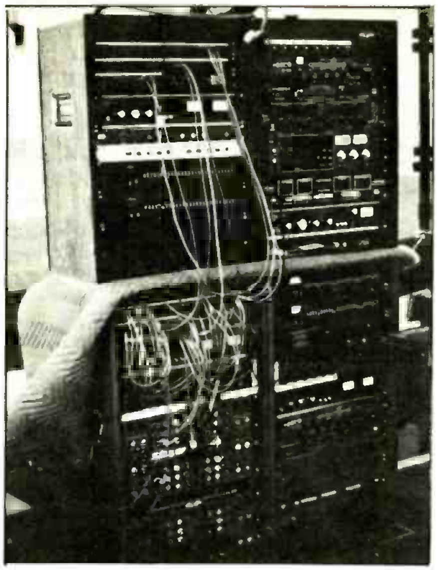


Figure 14: MSI house electronics racks housing a variety of outboard equipment.

Showtime

After several days of set-ups and rehearsals, a few of the crew members seemed to find the actual event almost anti-climactic. The thousands who had bought tickets, however, were quite eager. The sun rose the day of the show, and shone down on a miniature city of tents and sleeping bags in the parking lots outside. By noon, an estimated 94,000 people had found their way into the seating area, so many, in fact that an overflow crowd spilled into the area behind the speaker towers that was not covered by any direct sound.

As the concert was about to get underway at 12:45 pm, I spoke with Bill Geremenz, Outdoor Facilities Administrator for the City of Pasadena. Geremenz was holding an RCA sound-pressure level meter, and had come up to the mix platform to make sure that the engineers understood the rules: *90 Decibel Limit at the Rim of the Bowl.* [The maximum allowable limit has since been revised to 95 dB — *Ed.*] The City Board of Directors of Pasadena had decided to pass and enforce this ordinance to try to pacify residents of the neighborhood surrounding the Rose Bowl.

"I don't expect any problems," Geremenz told me. "The fellows know the rule, and have agreed to comply with it. It just plain cannot be louder than that." I asked Bill if his meter was set to take A-weighted or C-weighted SPL readings. He replied that he did not know.

Buford Jones viewed this requirement to hold the show's volume at a given level as perhaps a necessary evil. "Sometimes we just have to work with this sort of thing," he noted. "With groups like Jackson's and Linda's, it is not so bad . . . I don't have to run it at killer levels. It's not that we want it to necessarily hit a certain decibel level . . . we just want to preserve the dynamics of the music and make it sound right; whether that takes 90 dB or 120 dB will depend on the type of music you are dealing with. What we have on stage here today shouldn't really be a problem.

Bill Geremenz, the "Man With The Meter," looked quite concerned at the start of the show when he found that he was getting a reading of 92 dB just on the opening speaker's remarks. As Tierra, a Los Angeles-based Chicano band struck out the first chords of the afternoon, the meter showed that the system was operating at about 15 dB over the limit. However, by the third tune, the level had been reduced, the program material was relatively well-balanced, and the crowd had settled into the groove. The show flowed extremely well; no feedback; no sudden changes in level. The only audio-related problem occurred when a fault developed in a mike cable. By my count, at least two dozen different individuals had already handled this microphone and removed it from its stand. A spare mike was quickly brought out.

Subjective Analysis of Sound Systems

The Pasadena Rose Bowl is a large area to cover with audio program material. If this had been a regular, full-blown rock concert, perhaps twice the number of cabinets would have been required. As it was, the music was generally well-projected to all areas of the audience seating area, except for the extreme sides near the stage and, of course, the seats which filled up behind the speaker stacks. SPL meter readings were quite consistent as one walked around the rim of the stadium. As mentioned before, the overall sound of the show was quite consistent; this could have been due, in part perhaps, to heavy limiting. Most members of the audience agreed with each other in calling it "good sound," although few isolated individuals in the very rear (750 feet or so from the stage area) did yell "Turn it up!" throughout the day.

The frequency reponse of the system was fairly smooth except in the corridors directly on-axis with the center of the speaker stacks. Here, strong low-frequency standing waves seemed to occur. This effect was particularly noticeable during a tune in Gil-Scott Heron's set, which featured an Alembic bass guitar line taken down an octave with a pitch-shifter. The effect was stunning at the mix platform, and on-axis with the Bass-Aug cabinets, but was barely discernible up in the stands. Perhaps a spreading throughout the stacks of these cabinets might tend to correct the linear "beaming" of the bass frequencies. One thing can certainly be said . . . the TAD 15-inch cones do move some air!

Perhaps the only other personal note I might add is that occasionally the announcers' voices sounded a touch unnatural. While the sound of the music was quite pleasing, a single voice tended to bring out a peak in the vocal region, perhaps at around 600 to 800 Hz. The peak was not noticeable at the mix platform, but became evident at a distance of perhaps 300 feet from the speaker stacks.

The MSI-Northwest system did make Peace Sunday enjoyable, and it did what it was designed to do. 94,000 people heard a dozen different acts outdoors, and the groups sounded good. And no over-heated amplifiers. No hour-long delays. As Stephen Stills said to the audience at the opening of his set, "We've come a long way since Woodstock!" ■■■

IN-HOUSE CONCERT SOUND

SOUND-SYSTEM DESIGN FOR THE UNIVERSAL AMPHITHEATRE

Stanal Sound Loudspeaker System at this Remodelled Venue

by Gary D. Davis

Just under seven years ago, in the December 1975 issue of *Recording Engineer-Producer*, I first wrote an article describing the Universal Amphitheatre which, at the time, was an outdoor music venue with just over 5,000 seats wrapped in an arc of nearly 180 feet that spread 350 feet at the stage, with about 140 feet from stage center to the last row. The setting, on a hill in Universal City, California, five minutes out of Hollywood and overlooking the San Fernando Valley, helped make the Amphitheatre a premier showplace where major talent put on their biggest and best shows.

In 1975, I wrote: "Although the stage and seating area is completely surrounded by high walls, there is no ceiling; performances are done under the stars."

To reduce sound leakage into adjoining neighborhoods, the Amphitheatre's original lower walls already had been built up higher by 1975, and acoustic "clouds" (horizontal baffles) placed atop the speaker stacks. These measures were of some value, but ultimately a combination of politically loud "noise" complaints from the neighboring community, plus susceptibility to weather, and a limited season, led to a complete rebuilding of the facility to make it an *indoor* "amphitheatre."

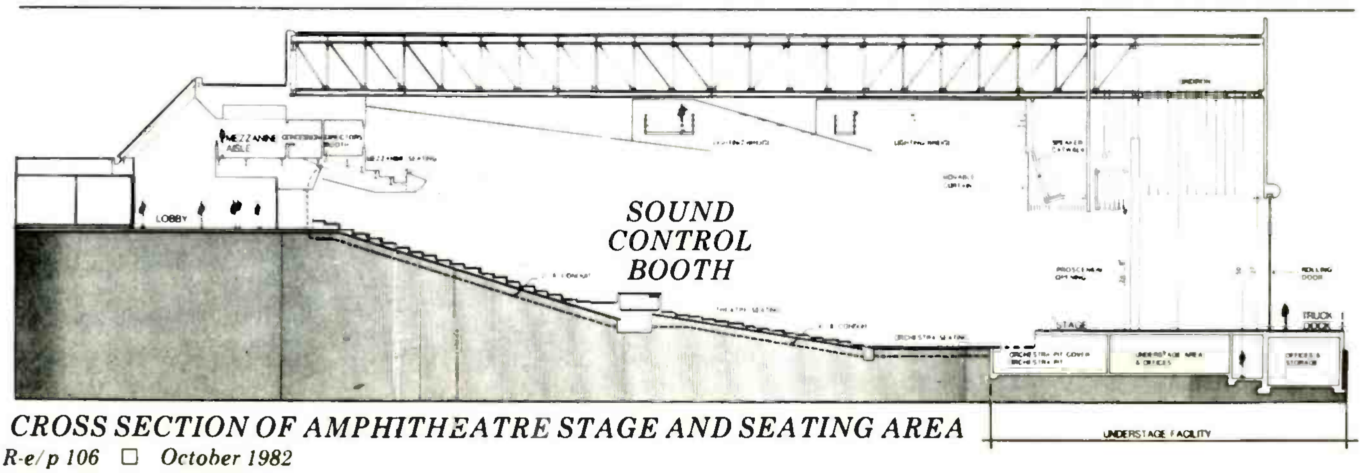
The original concrete slab for the seats was about the only part of the old structure that remained the same. A balcony was added, increasing the seating capacity by 1,000 or so to 6,251. Of course a roof was added with new supporting walls (ceiling height about 40 feet above the stage), and an all-new stage complete with full theatrical curtain and lighting equipment.

New House-Sound System

Sound, as always, is a key element to the success of the Universal Amphitheatre, and the task of designing and installing a new house-sound system fell to Stanal Sound.

An unusual move perhaps? After all, Universal might be considered by many to be perfectly capable of purchasing and operating its own sound system. So why, we asked Stan Miller, president of Stanal Sound, had Universal gone to an outside contractor?

"In this kind of theatre with varied shows, it doesn't make sense to install a 'house-owned' sound system," he offers. "The needs of shows are so varied that the theatre can't, on a practical basis, afford enough equipment to meet them. Stanal can do it because we run lots of shows. [Stanal also handles sound at the nearby Greek Theatre.] Even then,



CROSS SECTION OF AMPHITHEATRE STAGE AND SEATING AREA

the new UNIVERSAL AMPHITHEATRE

we have to rent sometimes. Even if it were practical to own all the equipment one might need, and fill in with rentals, there's something else we can bring to it . . . qualified operational personnel."

Looking back to *R-e/p*'s original story, when Stanal also handled the sound for Universal, the same basic philosophy was true then. Miller told us that, as originally conceived when the Amphitheatre first opened, performers were expected to supply their own audio systems. That situation soon proved unworkable for a number of reasons, both financial and practical. (It was, and still is, practically impossible to learn the acoustic requirements of a house as complex as the Amphitheatre, critically adjust the system, and do a polished mixing job with only a day of preparation time . . . typically the time available for rehearsal.) So, for the last seven or so years, Stanal has been making a "house" sound system — in reality, one of Stanal's systems specifically set up for Universal — available to performers at the facility.

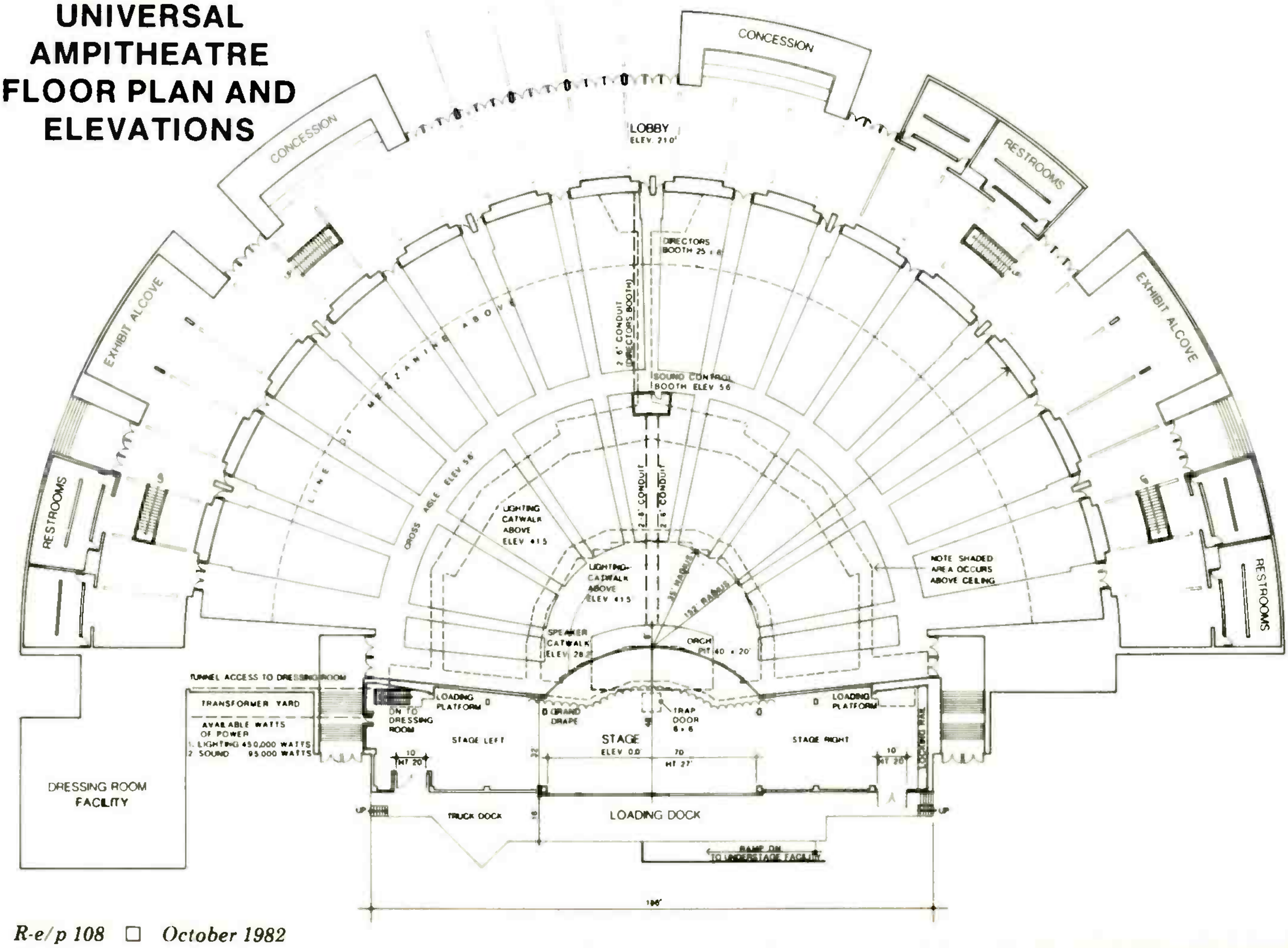
Mixing booth setup for Linda Ronstadt Tour, September 1982. Main sound contractor for the tour was Showco, who supplied a custom-designed 30-in/8-out stereo console, Kelsey 12/2 keyboard mixer, Sound Workshop 12/8 effects subconsole, and various effects units. Stanal Sound also furnished an outboard rack containing a Yamaha Q1027 graphic equalizer, dbx Model 162 compressor-limiter, Clear-Com KB-115 intercom remote station, Yamaha M406 submixer, and crossover units.

Old Versus New Sound Systems

In 1975 the sound booth was centered at the far rear of the seating area — hardly an ideal location. Being centered between two stacks of speakers, it was subject to phase cancellation between the speakers, so that the sound mixer could easily be judging the mix while listening to an inaccurate representation of what the majority of the audience was hearing. Furthermore, since the booth had a low roof and was located up against the rear wall of the Amphitheatre, there was some bass build-up. Nonetheless, a semi-permanent sound system was installed, with two Yamaha PM-1000 16-in/4-out consoles, several

. . . continued on page 112 after System Block diagram —

UNIVERSAL AMPITHEATRE FLOOR PLAN AND ELEVATIONS

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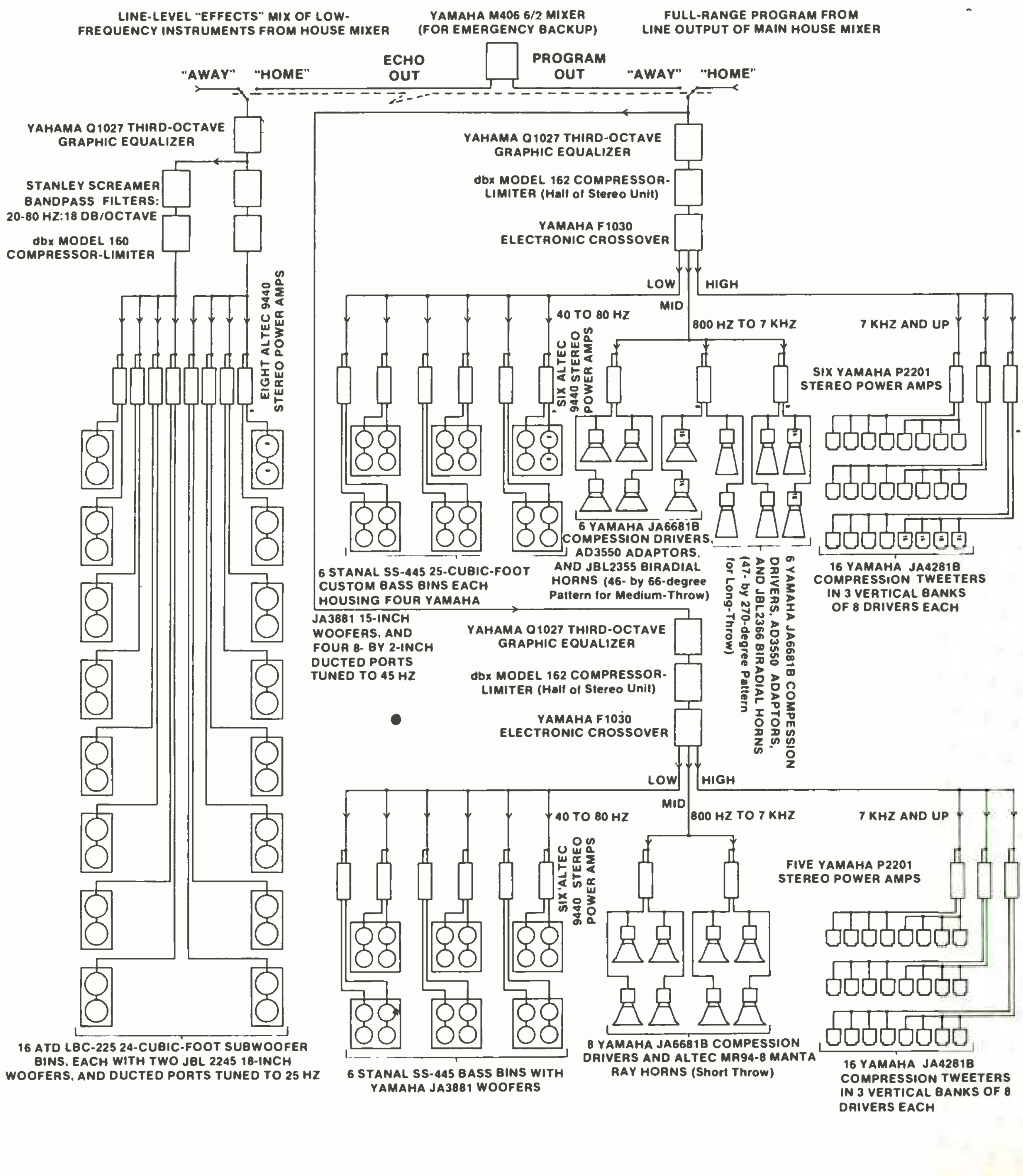
SCHEMATIC OF STANAL SOUND SYSTEM FOR THE UNIVERSAL AMPHITHEATRE, COMPRISING A PAIR OF PRACTICALLY IDENTICAL THREE-WAY, FULL-RANGE SYSTEMS, AND CUSTOM-DESIGNED SUBWOOFER SUB-SYSTEM



MILAM AUDIO

Mr. G. C. "Jeep" Harned
MCI Inc.
A division of Sony Corp. of America
Ft. Lauderdale, Florida

Dear Jeep,

Fifteen years ago, the difficulty I had convincing most equipment manufacturers that enough pro audio market existed in the Mid-West to warrant one dealership, seems impossible to believe today.

The Mid-West has grown into an industry giant, with stability and professionalism second to none in the world. I am proud that MCI and Milam Audio have played major roles in its growth and development, in conjunction with the multi-talented owner/operators who make it all happen.

1982 has been another very successful year, with increased MCI sales and installations of 14% over 1981, with indications for even higher growth before year end. The upgrading of present facilities accounts for most of the major sales, but new rooms are also on the increase.

Of the many great Mid-West cities where professionals profitably use MCI products, one of our Chicago clients has recently ordered his 50th major piece of MCI equipment. His incredible testimony is representative of the vast number of our clients that own MCI products in quantity.

Information regarding your "Next Generation, High Technology Products" is being requested. We are making everyone aware that many of the new designs take giant technological steps into the future, and are not being rushed onto the market.

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Sincerely,

Jerry Milam

Jerry Milam

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For additional information circle #54

the new UNIVERSAL AMPHITHEATRE

smaller PM-400 submixers, and a variety of Altec and Auditronics equalizers and UREI compressor-limiters. The booth could accommodate 64 mike/line channels, while the stage monitor system had an additional 20 mike and two line inputs, plus submixed feeds from the main booth.

In the new Amphitheatre, the audio booth is an open area located in the middle of the audience, not at the back of the house; there are 4-foot walls around it, and nothing to change the soundman's impression of the mix relative to what the audience hears. As previously stated, this time Stanal Sound has not installed a total, permanent sound system at the Amphitheatre. However, due to the tremendous size of the speaker system needed to cover the house, and the fact that it was installed on a catwalk some 25 or 30 feet above the stage, the power amplifier racks and speaker clusters are semi-permanently installed.

The catwalk follows the curvature of the front of the stage, and is aimed down somewhat toward the audience. If visiting artists want to bring their own sound systems, fine; they are welcome to use their own mikes, mixers, signal processing gear and stage monitoring, but are encouraged to tie into Stanal's amps and speakers for the main house feed. That's no handicap when you realize what's involved.

Miller considers his personnel to be equally important to the equipment Stanal provides: Consoles, amps, etc. are tools of trade — the doctor's scalpels — but having them doesn't guarantee a successful operation. Finding qualified personnel for an in-house system is *extremely* difficult. Even if there's a qualified person running the system at one point, people change, and then the system is usually changed to meet someone else's ideas. Which can get very expensive.

"We offer not only a particular quality, but a philosophy — a consistency for solving problems as they develop. We also are able to respond at a moment's notice to the needs of people here."

No matter how skilled the personnel, or how sophisticated the sound equipment, the environment of a live-performance venue will always play a major role in shaping the ultimate sound; indeed it affects the overall experience for the performers and audience. These factors were carefully considered before construction began. Architectural design for the new Amphitheatre was handled by Skidmore, Ownings & Merrill, and additional acoustics by Jack Purcell, of Purcell and Noppe, the joint goal in the new design being an acoustically neutral hall.

One tricky aspect to the design is the nearly 180 degrees seating arc around the stage. Regarding the acoustics, Miller says, "We wanted to add a roof, but to still have it sound like outdoors. I think it's very close." Coverage of such a wide area makes true stereo sound impractical; only a fraction of the audience would ever hear the stereo effect. Therefore, the system is set up in mono — redundant mono, as shown in the accompanying speaker/amp system block diagram.

Speaker System — The Key to Good Sound

Given that basically neutral acoustic environment, Miller has installed a speaker system that he says is capable of delivering "virtually flat response across the audio spectrum, and with no more than 3 dB variation at 4 kHz (our ear's most sensitive frequency for speech and music) between any two seats in the house."

That's a marked improvement over the original 1975 system which, while good by most standards, permitted about 6 dB level variation between the front and rear seats. How had Miller achieved such results in the new Amphitheatre, which basically was covering the same seating area with the addition of a balcony? The design, he offers, was "defined by mathematical calculations for SPL [sound pressure level], which told us the number of components needed, and the amount of power . . . to drive them. Also, it was based on 'gut feel' that comes from doing this kind of thing for a number of years."

The availability of a high catwalk to distribute speakers instead of scaffold-supported stacks, and the provision of rear structures to support time-delayed fill speakers, certainly aided the quest for uniform sound distribution.

Apparently, the calculations, "gut feel," and equipment met the challenge, judging by subsequent reviews. In the August 3 issue of *Orange County Register*, in a story about Frank Sinatra's grand opening concert, Gary Lycan wrote:

"Which brings us to the Amphtheatre's best asset, its sound system. You don't have to be an expert on tweeters and woofers to know the music and vocals coming through the speakers were full and free of distortion. [Sinatra side-man] Tony Mottola's hauntingly elegant guitar solo, for instance, would have satisfied the most critical ear."

In an August 6 *Los Angeles Times* story, titled "Fine-Tuned, but Southern California Sky is Lost," John Dreyfuss described the blending of the acoustics and the sound system. He quoted Bob Kiernan, Sinatra's sound manager, who ran the mixer for the opening show:

"The quality of sound seems more controlled than it did in the open amphitheatre. You hear more sounds. You hear the finer, softer sounds that got lost before."

In that same *LA Times story*, Frank Sinatra was quoted as saying the new theatre was "acoustically sensational", and that it gave him a feeling of being "connected" with the audience. "As large as the building is," he continues, "I feel it has a smallness about it . . . I get a feeling it's a small room. It's not cavernous."

In a space that measures about 375 feet wide by 160 feet deep by between 20 and 50 feet high, such statements can be taken as a compliment to the sound system, as well as the acoustical design.

The accompanying block diagram

Front view of Stanal loudspeaker system mounted in a gentle arc above the stage area, and angled downwards to provide coverage to all areas of the auditorium. The black scrim that normally covers the speaker stacks has been raised here to improve visibility. Access to individual cabinets can be gained by means of a special catwalk running along the front of the arrays.

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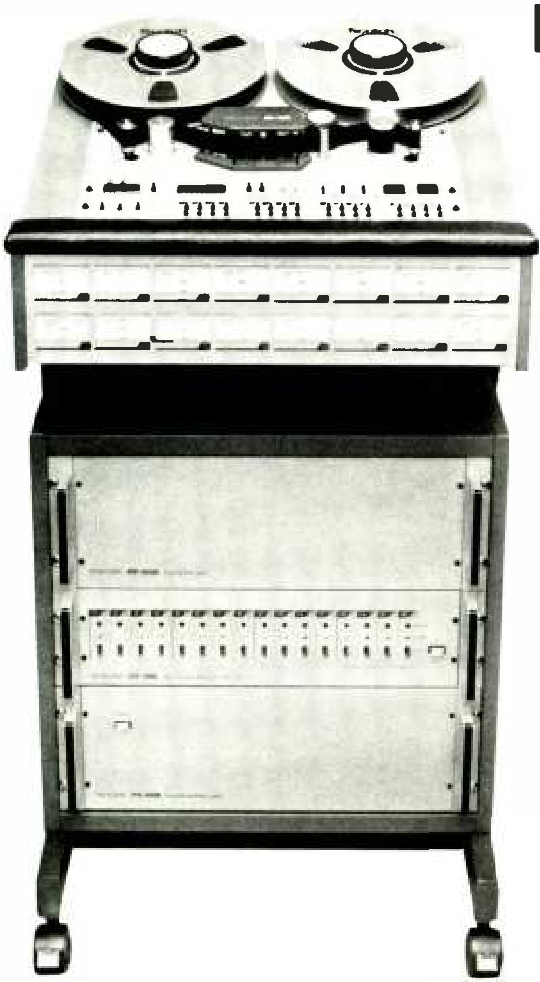

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details Stanal Sound's speaker system. It was assembled with a combination of components, some of which Stanal has used for many years, and others that are new, including a large number of Yamaha speaker components and ATD subwoofer enclosures.

The speaker system, while mono, is divided roughly into three portions: two nearly identical three-way, full-range systems; plus a special subwoofer system. In one of the full-range systems, the high-frequency compression drivers operate into a combination of mid- and long-throw horns aimed well out into the house. In the other full-range system, the HF drivers operate into short-throw horns for near-stage coverage.

Subwoofer Approach

The subwoofer system is something that Stanal set up in a different way from most other low-end rigs. Instead of using four-way electronic crossovers, and simply driving the subwoofers with a very low frequency portion of the full-range program, Miller utilizes Yamaha F1030 three-way crossovers, plus a separate subwoofer system to enhance certain instruments. This is achieved by assigning those input channels with bass-heavy instruments (for example, kick drum, bass guitar, or synthesizer) to two busses: the main program mix, and a subwoofer mix.

The full-range system's bass bins take those instruments down to 40 Hz (via the F1030). The ATD LBC-225 subwoofers then overlap this bass by covering the region between 20 and 80 Hz. The subwoofer feed is bandpass filtered and, as a result, very clean. It also keeps the main speaker system "cleaner", because when more bass is needed, Miller can punch it up in the 18-inch subwoofers without creating muddiness in the 15-inch woofers.

Failsafe Design Elements

The use of redundant graphic equalization (Yamaha Q1027), compression (dbx Model 162 and 160), electronic crossover networks (Yamaha F1030), power amps (Altec 9440 and Yamaha P2200), and speaker components means that a failure in any one major component cannot bring the show to a halt. The possible exception is the failure of whatever console is being used to mix the house, but even here Stanal has provided a backup: a rack-mounted Yamaha M406 6-by-2 mixer, with an emergency switch that swaps it for the house console. (The soundman keeps a local mike plugged into the M406 so that the crowd can be controlled in the event of an emergency).

While the graphic equalizers, compressors, and crossovers are located in the sound booth at a mid-audience position, the power amps are located in locked, 77-inch tall, steel Soundolier racks up on the catwalk, thereby minimizing losses in the amp-to-speaker cables, and providing added security. Black Melotone grille cloth covers the entire catwalk area, including the speakers. Patching between the amps and speakers is done with two-conductor #10, #12, or #16 wire, depending on the specific components, and plugs are used so that changes can be made quickly. Should a component fail during the show, an engineer can move out on the catwalk and disconnect any one bass bin, high-frequency driver, or compression tweeter, without affecting the others.

It would probably take a major power failure to stop the show, and even here Stanal has taken extra care. An AC distribution system has been set up with five Topaz Ultra-Isolation transformers from the main building feed. (In 1975 Stanal was using three of these special transformers, so it must be that the seldom discussed spec of "audio kilowatts per acre" has increased about 160%.) Non-standard outlets are used on the main AC feed boxes so that nobody but sound personnel can plug in; lighting or stage set-up equipment will be forced to use a separate AC system. To minimize hum and maximize safety, the ground path is very carefully thought out and meticulously installed. Even so, the AC ground system was undergoing a few last minute adjustments when *R-e/p* visited the venue, prior to the gala opening show.

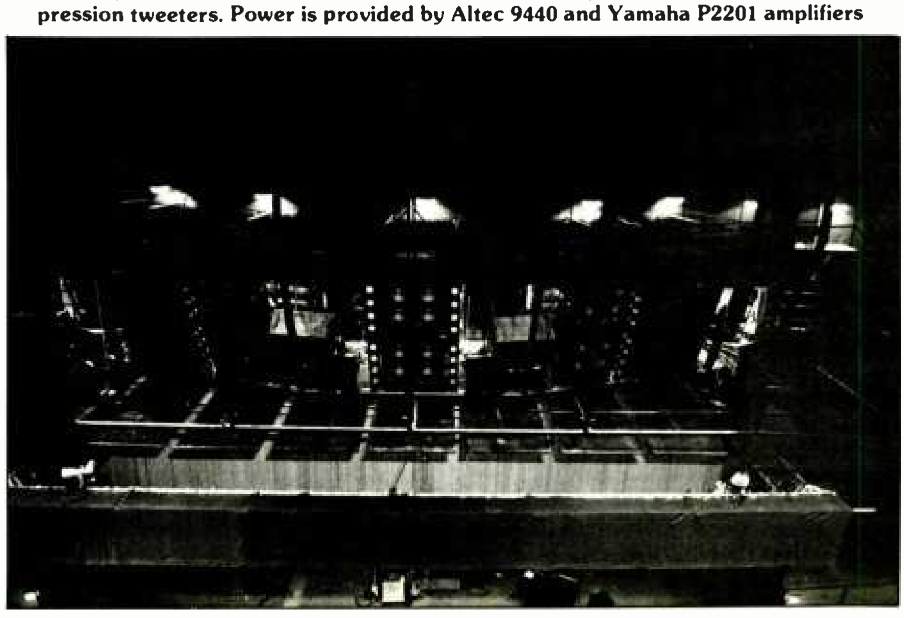
Concert Example

While no permanent mixing system has been installed at the Amphitheater, for one of the opening shows, a charity benefit performed by Frank Sinatra, Stanal used several Yamaha consoles: a PM-2000 (32-in/8-out); an MQ-1602 (16-by-2); and an MQ-802 (8-by-2) that jointly covered the main house and the monitors. The PM-2000 provided most of the required inputs with lots of EQ, subgrouping, and a mix matrix to combine the subgroups into house feeds, recording feeds, etc. The smaller MQ Series mixers were chosen because they provided the necessary number of extra mike/line inputs, yet did not take up a lot of space in the booth. One of the PM-2000's eight main program busses was used for the special "bass-heavy instruments" subwoofer mix.

There were nearly a dozen monitor cabinets on stage for Sinatra, mostly Yamaha S2115Hs with a few Yamaha 6115 systems for side-fill (S6215 double bass bin, 6215H mid-range horn, 6115T-3 triple compression tweeter). As mentioned earlier, the monitor mixing for this particular show was done with the same PM-2000 that fed the main house. Regarding choice of monitor-mix consoles for other acts, Stanal has a number of Yamaha boards specially modified as monitor mixers by Windt Audio, Inc., including one or more of the following: an M1532 32-in/8-out; an M1516 16-in/8-out; a PM-1000 32-in/12-out; a PM-1000 16-in/6-out; and an M916 with 16 inputs and six outputs.

The complement of mixers, signal processing, amps, stage monitor speakers and microphones all depends on the needs of the act. Which points to one of the key benefits Universal realizes by working with Stanal — that of service. The monitor system, as well as mikes, auxiliary equipment and mixers, can be changed based on the artists' needs. Stanal does not send each artist a comprehensive list of available equipment. Instead, it asks what the artist requires, and then, after receiving a reply, Stanal sends back a list of available equipment to do the job. If the artists or groups want to bring in their own console, signal processing, monitors, etc., that's

Close-up detail of Stanal three-way loudspeaker arrays, comprising SS-445 custom bass bins, Yamaha JA6681B drivers on JBL 2355 horns, and Yamaha JA4281B compression tweeters. Power is provided by Altec 9440 and Yamaha P2201 amplifiers



fine, too.

Audio Cables

One learns from experience... at least Stanal and Universal have. As always, balanced lines with XLR-type connectors are used throughout the venue. In the original Amphitheatre, however, conduit poured in the concrete slab carried a dedicated set of cables between the mixing booth and the area below the stage. The conduit was not continuous because the Amphitheatre seating had been expanded at one point, so there were connectors in the cable below the concrete. When the concrete slab was washed down with a fire hose after each performance, water could leak into the cables, corrode connectors, and generally cause all manner of problems.

At considerable expense to Universal, that original set of conduit was torn out in 1975, and new conduit installed with continous cabling for 64 mike/line inputs to the board, plus line feeds to the stage, eliminating the water and corrosion problems. Still, the cabling setup was not easily changed.

Now that the Amphitheatre has been completely rebuilt, and the mixing booth relocated, 81 mike/line input cables are provided between the stage and the booth, routed through four, 8-inch diamter PVC conduits laid in the concrete slab for just that purpose. The large diameter PVC conduit is not only waterproof, but also permits pulling new cables easily, as many as are ever likely to be needed, with adequate physical separation between mike and line signals to prevent crosstalk.

How Does the New Facility Compare to the Original?

Whether you like the new Amphitheatre more than the original will depend on how much you thrive on being under the stars. The new roof is lit to look something like a star-lit sky, but of course you won't see the orange hills as the sun sets. There is a certain feel to an outdoor concert, yet the acoustics of the new Amphitheatre come reasonably close to the real thing. Certainly the sound is not like a typical narrow-walled concert hall; it is open yet intimate. This assessment is all pretty subjective and we've heard comments favoring the old and the new.

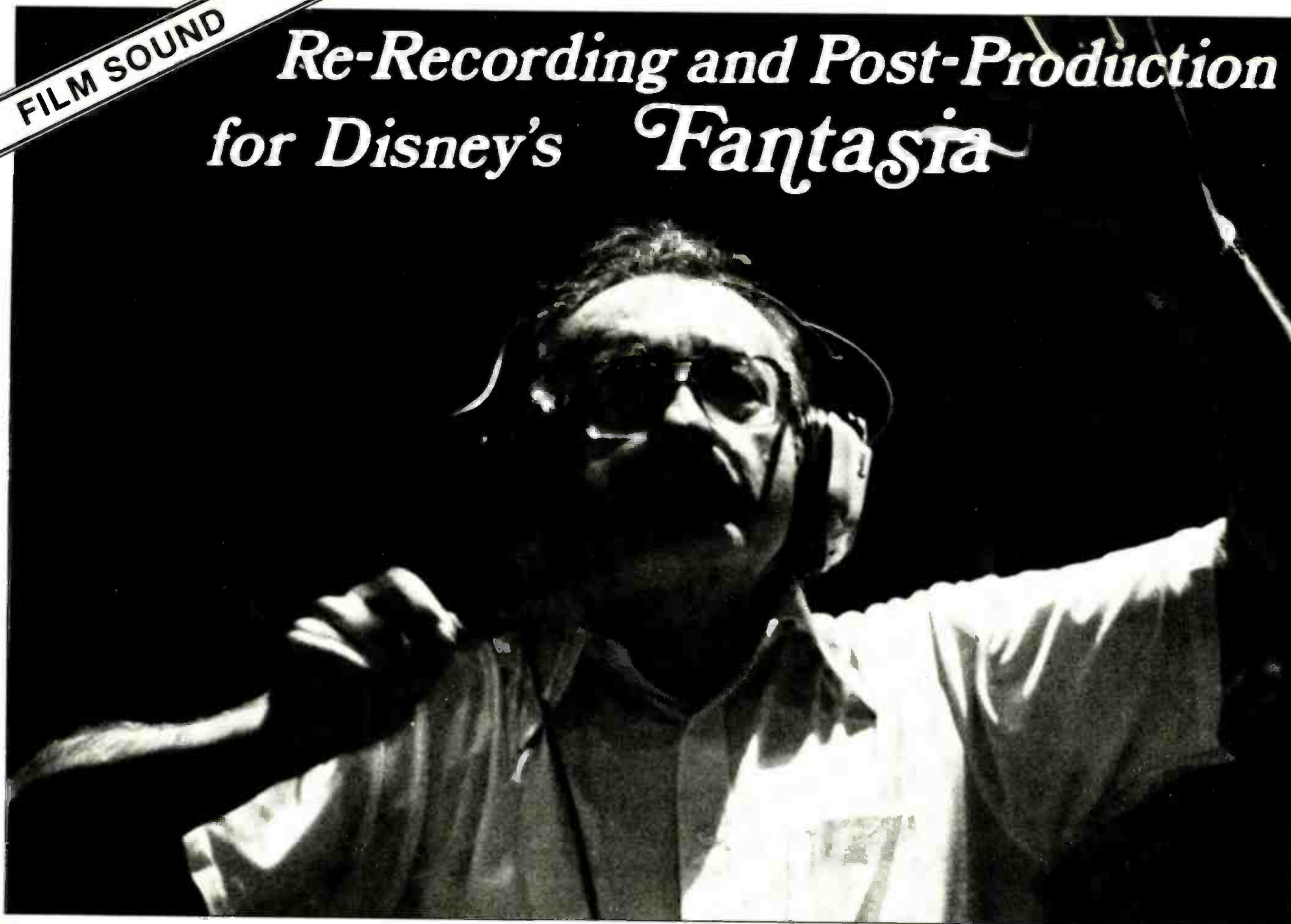
There are other factors, however, that lead us to agree with Universal's basic decisions. For one thing, complaints about "noise" from concerts will no longer force high-energy rock groups to throttle down the SPL. That's not to say that an audience will be subjected to a continuous 120 dB in the back row, but levels used to be restricted to about 95 dB —and Universal staff came around with meters to avoid those nuisance law suits. Also, because the temperature and humidity are controlled, the audience will be more comfortable. You may not see the sun set, but neither will you be rained out, covered with dew, or chilled by the mountain air as it slides from day to night.

Stable environmental conditions will also help keep the musical instruments in tune, as well as the sound system. Remember that the most careful third-octave tuning can go down the tubes as soon as the temperature changes 10 or 15 degrees Fahrenheit. Previously, if a breeze blew across the audience, it could effectively re-aim the sound from the speaker stacks, and wreak havoc with the coverage patterns; this is no longer a problem. In fact, the Amphitheatre will now offer year-round entertainment, not just a summer schedule. Yet there's more.

From an overall production standpoint, the new facility is light years ahead of the old one. There's a covered 65-seat orchestra pit; an overhead gridiron and 45 full stage pipe capability; a speaker catwalk; multiple lighting bridges at 45 and 60 degrees from the primary playing areas, with 450,000 watts of lighting; a director's booth behind the mezzanine; a 6-channel by 27-station matrixed intercom system; a huge rolling loading door at the rear of the stage; and lots more, including ample office and dressing room space beneath and adjacent to the stage area. There's just about everything needed to stage a major production. We'll speculate that, in the future, musicals and theatrical presentations will be attracted to the facility. ■■■

FILM SOUND

Re-Recording and Post-Production for Disney's *Fantasia*



by Larry Blake

A chance meeting in 1937 between Walt Disney and conductor Leopold Stokowski marked the beginning of an association which resulted in the appearance over three years later of *Fantasia*, the first film to be released in stereophonic sound. Though stunning for 1940, improvements in the state of the sound recording art over 40 years diminished the marquee value of the stereo mix. Ever mindful of the re-issue potential of *Fantasia* — it is the only Disney film which is in perennial release, and not just dragged from the vaults every seven years for the next generation of youngsters — executives at Walt Disney Productions decided to bring the soundtrack up to date, and then some, by digitally re-recording Stokowski's score note-for-note.

The Sorcerer's Apprentice

Disney's original idea was not to make the feature-length film we see today, but simply to create a short that would fuse animation with Paul Dukas' scherzo for orchestra, "The Sorcerer's Apprentice." Stokowski was so eager to handle the music chores that he is said to have offered to work for nothing. He saw this as an opportunity to fully exploit a system that had been used to record his score of the 1937 Universal film *One Hundred Men and a Girl*.

Music for that film was recorded on eight optical recorders — remember, Les Paul's medium, magnetic recording, didn't come to Hollywood until the early Fifties — allowing sections of the orchestra to be handled separately during mixdown. Six channels were devoted to individual sections, one to a distant mike, and another to a balanced mix.

However, the film was released with a standard monophonic soundtrack; for his work with Disney, Stokowski wanted to bring multichannel sound into the theater. Stokowski's involvement with multichannel sound dated back to the early Thirties, when he began his research with the Bell Telephone Lab on stereophonic sound. Bell's most famous demonstration came when the Philadelphia Symphony Orchestra was transmitted over three telephone lines to an astonished audience in Washington's Constitution Hall.

"The Sorcerer's Apprentice" was recorded at Pathe Studio in Culver City in January, 1938, and during the animation it became apparent that there would be no way for the short to make back its negative cost of $125,000.

The Concert Feature

That fall, Disney, Stokowski and Deems Taylor, the noted music commentator, met to choose the selections for the rest of the "Concert Feature," as *Fantasia* was referred to during production. It was decided to record the remaining selections with Stokowski's Philadelphia Symphony at the Academy of Music, because of the better acoustics that facility would provide. Recording in Philadelphia began April 7, 1939, and consumed almost a half-million feet of sound film during the mammoth 42-day session.

For the *Fantasia* sessions a ninth recorder was added to the *One Hundred Men* system, giving a click track to animators as a guide in timing. The freedom offered by the multitrack recording allowed for overdubbing to replace a few bars of an individual section, if needed.

Mixing occupied many months at Disney's then new facilities in Burbank, with much of the time spent figuring out ways to move the sound around the auditorium. It was Disney's intention to have the music comment upon the action in both spirit and placement. Purists would hesitate before calling *Fantasia's* soundtrack "stereophonic," since a true stereophonic recording captures an acoustic event in proper perspective; in *Fantasia*, the mixers, and not fast-footed violinists, made the violins move across the screen in sync with the animation, courtesy of probably the world's first panpots.

Sound was recorded on three tracks, feeding speakers to the left, center, and right of the stage behind the screen. The same double-width (200 mil) push-pull variable-area system utilized on the original recordings was used in the

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FANTASIA DIGITAL
FILM SOUND RECORDING

final mix. A fourth track contained control tones which could increase the loudness of the other three tracks by up to 20 dB each. In the theater the four tracks ran on a separate piece of film in interlock with the picture, which had a standard mono track as emergency back-up.

Fantasia opened in New York on November 13, 1940 at the Broadway Theater, and played there for over a year. The sound system was named "Fantasound;" the original installation cost $85,000 (1940 prices), and included 54 speakers placed throughout the auditorium.

Beneath the contents in the programs sold during the original run was the notice: "From time to time the order and selection of compositions on this program may be changed." Disney wanted to re-release *Fantasia* every year, adding and subtracting selections in the process. It would not then simply be a question of where and when *Fantasia* was playing, but what was playing in it.

However, *Fantasia* was not successful at the box office in its initial engagements, and the plan to alter the selections for future editions was never carried out. In 1942, 43 minutes were deleted, most notably the entire Bach segment. The film was now in mono and would not be heard in stereo again until 1956, when it was restored to its original length and released in the four-track magnetic format.

Irwin Kostal at the podium conducting the hand-picked orchestra at CBS Studio Center, Los Angeles, for the *Fantasia* re-recording sessions. At the monitor console Dennis Ricotta (left); and foldback mixing engineer Andy Bass.

Fantasia in the Eighties

The decision to re-record the score was made in late 1981. Chosen for the job of supervising and conducting the new recording was Irwin Kostal, who has won Academy Awards for his scoring adaptations of *West Side Story* and *The Sound of Music*. Working closely with him in "spotting" the score was Jimmy Macdonald, Disney's veteran sound effects expert who, incidentally, can be seen playing timpani in *Fantasia*.

Kostal says that working on *Fantasia* was one of the hardest things he has ever done, basically because "Stokowski's freedom was my straitjacket." He found that in many instances he would arrange a selection differently than on Stokowski's *Fantasia*, because of what he believed were cuts that had been made without Stokowski's participation.

"In 'The Sorcerer's Apprentice,'" he says, "there were points where I think somebody arbitrarily cut eight bars out, when it would have been better to only cut seven and go one bar sooner, or one bar later. I tried to make the cuts consecutively more like Dukas wrote the music."

One of the more unique problems Kostal was faced with concerned Beethoven's Sixth Symphony. In the original version of the film there was a scene where black centaurs could be seen shining the hooves of the white centaurs. It was not until the Sixties that this scene was regarded as offensive, and at that point deleted; the accompanying music was also removed, resulting in an abrupt cut. This "across the board" cut of the music was done for convenience's sake, since everything before and after the cut would remain in its original sync.

For this reason, and because of cuts made by Stokowski himself, Kostal prepared 50 versions of the first two minutes of the Beethoven segment. "I worked very hard to stick with the original Beethoven as much as I could. It was really difficult because to preserve the order of the beginning meant that, in our version of *Fantasia*, *all* the music of approximately the first minute and 45 seconds was different than Stokowski. The reason for this was that the cuts were so *bad*. Some of this had to do with his cuts, but most of it was terrible things, like the across the board cuts, and cuts which removed the downbeat, or the upbeat. It was ruined."

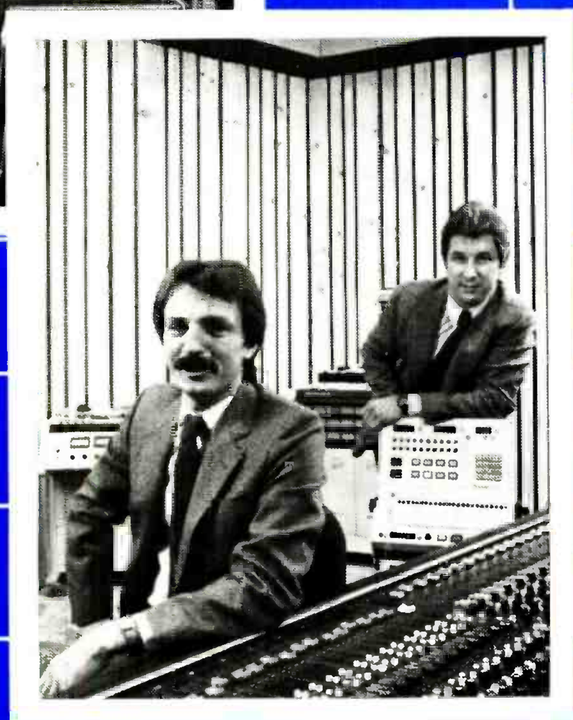
Scoring Sessions

Recording in January 1982 occupied 18, three-hour sessions at CBS Studio Center's Studio One, chosen over other LA film scoring facilities for its size, since Kostal would be leading a 125-piece orchestra.

In 1939, multiple tracks meant multiple recorders, and nine were needed to capture the music for *Fantasia*. The 1982 *Fantasia* sessions also pulled nine recorders, although the reasons were archival in nature.

The primary recorders were, of course, digital: two 3M Digital Mastering System 32-tracks, recording at the 50 kHz sampling frequency. Receiving a similar feed were two Ampex MM-1200 24-track analog recorders, Ampex ATR-104 and ATR-102 four- and two-track machines being used primarily for reference purposes.

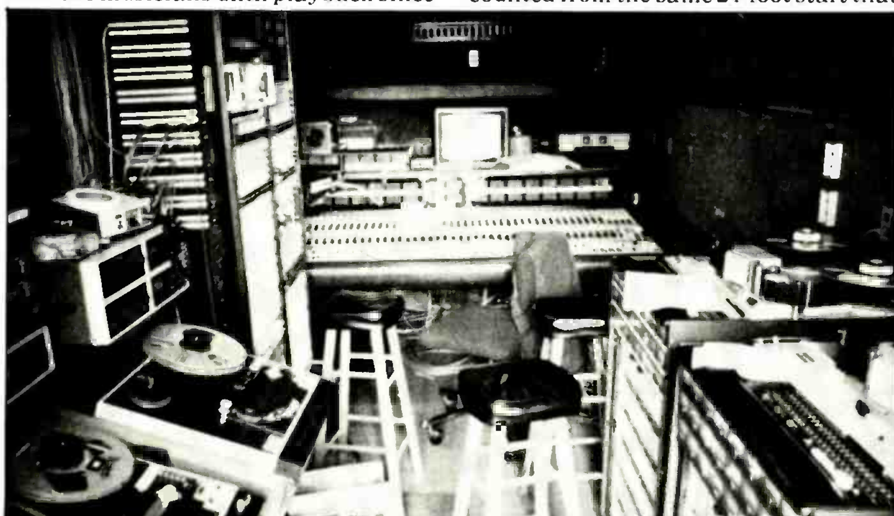
The three-track monitor mix made by CBS Studio mixer Shawn Murphy during the sessions was recorded on 35 mm magnetic film, and on three tracks of the digital machines, in addition to a Soundstream four-track digital recorder, also using a sampling frequency of 50 kHz. Murphy was not only responsible for recording the sessions, but the final dubbing as well. In addition, he



FANTASIA DIGITAL FILM SOUND RECORDING

was a key figure in setting up the synchronization scheme. The digital recorder interface was done by Dave Spencer at WED (Walt Elias Disney), the design and engineering arm of Walt Disney Productions.

Before the sessions, 11 reels of 35 mm magnetic film were striped with SMPTE timecode to be regenerated and fed to the multitrack machines during recording. Also on 35 mm mag running in sync to the picture were the original Stokowski score and a click track, prepared by music editor Dennis Ricotta by punching holes in magnetic film, and played back on an optical sound reproducer. Most of the time Kostal didn't hear the musicians until playback since he was listening to "Stoky" and/or the clicks in his headset during recording.



The ninth recorder on hand for the sessions was Keith O. Johnson's three-track FM analog recorder. Johnson was recording with just three omni microphones, using Schoeps capsules and Sennheiser amplifiers. Murphy set up three Neumann "wide" cardioid U-89s for his overall pickup, and he ended up using them for most of the final mix, although Johnson's were used on a few selections.

At Evergreen Studios in Burbank, Murphy recorded over the three-track monitor mix, and made a master mix of selected tracks for use during final dubbing. This mix was taken primarily from a three-mike stereo set-up, with no spot mikes added.

Music Editing

After the session, the selected takes were "printed" on to three-track 35 mm magnetic film for editing by Dennis Ricotta on a standard Moviola. Track one contained a mono combine of the monitor mix; track two the clicks; and track three SMPTE timecode. Ricotta checkerboarded his cuts over between two and four rolls, and for all practical purposes cut without regard to the electronic "sprocket holes" occupying track three.

Ricotta's cuts were translated to the original digital master tapes on a soundstage at the Disney lot that was being used for the mixing of various films for EPCOT (Experimental Prototype Community of Tomorrow), the new expansion to the Disney World theme park in Orlando, Florida. All work for EPCOT is being recorded and mixed digitally, and this stage is equipped with two 3M 32-track recorders, and one 3M four-track recorder.

The first step, as shown in the accompanying illustration, was to transfer the timecode numbers from the 35 mm rolls used in editing, to a pair of 32-track digital submaster rolls. First, 35 mm roll A was duped on to a pre-striped, timecoded roll on digital submaster Machine A. This pre-striped timecode counted from the same 24-foot start that played back at every take during the scoring sessions. The start marks for *Fantasia* were 24 feet from the first clicks, instead of the conventional 12 feet, because of the limited speed-variation capability of the digital machines, and hence their long sync-up time. One track was recording regenerated SMPTE timecode from track three on the mag film, recording the in-out points of each cut. Next, the same procedure was used to transfer mag roll B to digital submaster B. If there was a 35 mm editing roll C, then its timecode would go to Machine A, and so forth.

Because the timecode numbes given to each take of a selection during scoring were fed from the same SMPTE-striped 35 mm mag film, the timecode on track three of the editor's roll was always close to the master timecode with which the digital submaster rolls were striped during transfer of the 35 mm editing units.

After this procedure was completed, the master recording of the digital roll was threaded up on one machine, and the digital submaster to which it would be duped was on the other. Using as a start-stop reference the timecode which was transferred from track three of Dennis Ricotta's edit reel, all the tracks on the master roll were transferred to the submaster roll.

Once the transfer to the digital submaster rolls was completed, the master rolls to be used in dubbing were created by simply duping each roll to another 32-track digital machine, transferring only the tracks Murphy would need for the final mix. Since the submasters contained basically all 32 tracks of information from the original recordings, there was no space on them to bounce the final mix.

Approximately 11 tracks were copied from the submasters to the dubbing masters: the three-track mix made at Evergreen; the X-Y AKG stereo C-34 woodwind mike (in case Murphy needed "extra beef from the rear"); the AKG C-422 M/S stereo mike; the bass mike; and the harpsichord and keyboard mikes, if those instruments were used.

Inside of Best Audio mobile truck, with *Fantasia* sound engineer Shawn Murphy at the Sphere console. All tape machines aboard the truck were analog, simultaneous feeds being provided to various digital multitracks housed in the CBS studio Center.

A potential problem arose during the transfer of tracks to the B submaster roll. Since only two 32-track machines were available, there was no way to have the A submaster running in sync to check the edit; the moment of truth came when the A and B rolls were played together in the dubbing theater. The system worked so well — and Dennis Ricotta's cuts were so accurate — that only a few cuts had to be "slid." A

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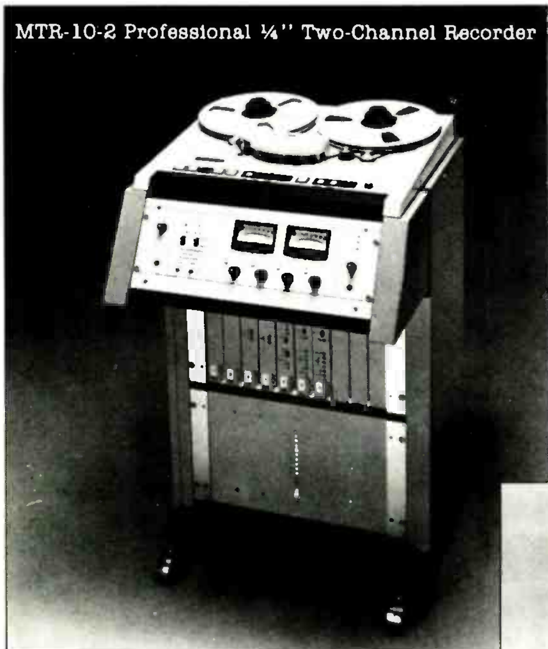
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FANTASIA DIGITAL FILM SOUND RECORDING

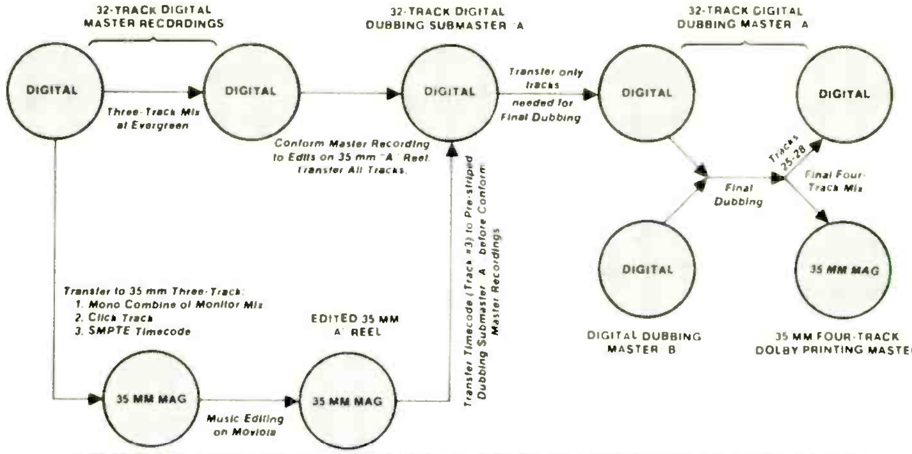
few reels, however, had to be re-transferred because of noise problems.

Vocal Recordings

The narration for the *Fantasia* re-issue has also been recorded digitally, thus forever removing the familiar voice of Deems Taylor from the film. The new narrator is Hugh Douglas, who was recorded in stereo with two Schoeps CMC-341 mikes on both Sony PCM-1600 and 3M digital recorders.

The famous moment when Mickey runs up the podium to thank the Maestro is still there, although since Stokowski died in 1977, his voice had to be dubbed. Mickey will never die, of course, and his voice is currently supplied by Disney employee Wayne Allwine.

On reel #5, when the narrator talks to the soundtrack, the narration was kept separate from the four-track music mix to facilitate foreign-language dubbing.



APPLICATION OF DIGITAL MULTITRACK MACHINES DURING THE RESCORING OF *FANTASIA*

On all the other reels there are overlaps only during tune-ups, which will be no problem to re-mix.

Final Mixing

During the final dubbing sessions in the Disney main theater — built for the mixing of Stokowski's *Fantasia* — the four-track master was recorded simultaneously on tracks 25 to 28 of the A digital machine, and on four-track, Dolby-encoded 35 mm mag film, which would be used as a printing master for the Dolby four-track magnetic prints. According to Shawn Murphy, "We interfaced the digital recorder and the film recorder with the dubbing console so that whenever I punched-in on the digital machine, I punched in on the film machine also."

Many readers will ask a semi-obvious question at this point: If there is no generation loss in digital recording, why go through all this trouble, and why not just make transfers when the digital mix was locked down? Here's Shawn Murphy's explanation:

"The real benefit in a digital storage medium, for our purposes, was the ability to transfer the intermediate steps — the mixed master, the dubbing submaster, and the dubbing master — in the digital mode, without going outside the machine, and with little loss. Not with *no* loss, but with little loss. If you would have had magnetic film tranfers at each of those steps, you would have suffered some loss.

"Every step that you go outside this machine you have a build-up of noise and some type of distortions which are different in digital — but are still there. And when you record on a digital system, you have to abide by its rules in terms of the amount of level — there is no forgiving saturation factor — and how it handles low- and high-frequency extremes, which is different from the way analog handles them.

"The 3M machine was the only multi-track digital machine available to us at that time which would lock up to a dubbing console and [film] distributor system, and which would have a good percentage of reliability in getting the show done."

What were the advantages, then, of using digital recording? In other words, and it's a question often asked in the recording and film world: How different would the project have sounded had an all-analog system been used? Again, Shawn Murphy:

"If we had taken great care to make sure that the analog machines were lined up optimally at every step of the way, and had taken great care with the transfers, though there would have only

DIGITAL SOUND
FOR MOTION PICTURES

In the areas of sound quality and flexibility, the improvement over today's state-of-the-art magnetic film offered by digital audio will be at least as great as that medium's improvement was over optical sound techniques.

The largest improvement undoubtedly will be in the area near and dear to the hearts of film producers worldwide: speed of operation. When integrated with a video editing system, digital recording will be to sound editing what word processors are to writing. An integrated sound/picture editing system will remove forever the bane of sound editors' existences, picture cuts.

Limitations of today's technology and *modus operandi* dictate that sound editing not take place until the picture is locked down. At this point black and white copies are made of the editor's workprint, and sound effects layed in sync with the picture, counting from the Academy leader start mark 12 feet (8 seconds) from the first frame of picture. Which goes to say that if the editor decides to eliminate one scene, which runs for 90 feet (one minute) at the beginning of the reel, the sound editors have to be told to remove this scene from their prints, lest the previously in-sync dog bark occur exactly one minute late, in the middle of the love scene.

Not only will video editing greatly improve the speed it takes to get the picture to "fine cut" form, but the computer-based nature of the beast makes irrelevant such distinctions regarding the status of picture editing. Sound editing can proceed apace from the first cut of the first scene shot, since the computer, and not the sound leader, will keep everything in sync.

The end result will be that, except for perhaps the music, all but the most complicated films could be made ready for final dubbing —and not just initial sound editing — as soon as the reels are "locked." In addition, procedures which today have to wait until dozens of reels are ready for mixing — primarily dialogue and effects pre-mixes — will have been dealt with by the sound designer during production. The formal re-recording session (with the meter counting off Big Bucks) will then be limited to the final blending of dialogue, effects and music, and not to getting car-door slams in sync, or watching the dialogue mixer trying to remove camera noise from the track without removing the actor's voice.

Computer and test console for digital audio processor currently under development at Lucasfilm

While all of this equipment will not come cheap, to paraphrase J.P. Morgan, while some will have to ask how much the new hardware will cost, they will find a way to afford it.

Undoubtedly there will be more responsibility given to less people, and the distinction between sound editor and re-recording mixer will blur along with the differences between recorder and console, creating a new breed of sound designer. . . *continued overleaf —*


ACME
SOUND

DIGITAL SOUND FOR MOTION PICTURES
— continued . . .

Digital Sound At Lucasfilm

A version of the dream system described above is currently on the drawing boards at Lucasfilm, Ltd. in San Rafael, California. Putting his money where it will give him a better product, George Lucas has given a staff of four, headed by Andy Moorer, the time and money needed to design and build a digital sound processing system. Work is going on concurrently on digital scene simulation and video editing.

Although a working prototype has been in operation since April, the first two systems will not be turned over to production until early next year. As is the general policy of Lucasfilm, the fruits of their R&D are available for license agreements.

The consoles that will be used in sound editing and re-recording essentially will be identical, and differ only in computer-controlling software. Like the new Neve DSP digital console, the Lucasfilm boards will be reconfigurable and open-ended, with alphanumeric readouts to tell the mixer what knob is controlling what.

"In our scheme of things," says Moorer, "firstly, the traditional division between the

ACCESS System at Neiman-Tillar

cutting and mixing stations would go away: it's all the same machine. You would have a different program to do mixing rather than cutting, but it's all the same system. The only difference will be in the software.

"The idea is to carry along the sound in the digital domain from the earliest possible time to the latest possible time, until when the six-track master is made. The actual storage and selection of sound effects will be a very, very small part of what this system is designed to do. The largest part will be the processing itself: equalization, filtering, reverberation, pitch-shifting, ring modulation, what have you. We would eliminate the need for outboard equipment. Our machine can do all the operations necessary for music synthesis, too."

Moorer and his staff are currently looking into the possibility of using a sampling frequency of 35 kHz, both for economy reasons, and because "in a good theater the highest measured frequencey response starts to roll off at 15 kHz, which would suggest that a 35 kHz sampling rate would be quite adequte. This is not a hard and fast thing, and we have the capability of running at any rate we like, so we'll deliver whatever can actually be appreciated in the theater."

While many might say that on-set noise precludes the need for anything better than the ubiquitous Nagra, location sound for future productions at Lucasfilm will be recorded digitally, and remain in the digital domain up to the printing master. By starting with digital recordings on location, Moorer feels that the primary gain will be the reduction of "total chance of error or misalignment."

"We have schemes for cleaning up set dialogue in a semi-automated fashion," he says. "But since that would require very, very repeatable calibration of equipment, digital [location recordings] seems to be the only way to get this right now."

ACCESS/Neiman-Tillar Associates

Unbeknown to many people in motion picture sound, a digital sound editing system has been in daily use since early 1977: the ACCESS system at Neiman-Tillar Associates in Los Angeles. Living up to its name, ACCESS (Automated Computer-Controlled Editing Sound System) gives the sound editor the ability to audition over 250 hours from the Neiman-Tillar sound effects library at the touch of a button. Each computer disk can store 40 solid minutes of effects, and many disks can be on line at a time.

Working with a ¾-inch video cassette of a cut reel, the sound editor "cuts" the sound effects by entering into the computer the desired start-stop times. Usually the only control a sound editor has over the track while editing is the scraping or wiping of the magnetic emulsion. Any processing — slowing down, reversing, etc. — cannot be executed in real time, and must be done elsewhere.

The ACCESS editing system, on the other hand, can raise or lower the volume and pitch, and change the equalization of the sound effect. Most sounds, as stored on the computer disk, are less than three seconds long, for obvious space/economy reasons. By the automatic looping of a sound, and creative use of the pitch and volume controls, movement of a car or a helicopter can be accurately simulated. The editor can listen to two cut tracks at a time, and lay a maximum of 19 tracks per reel.

The start-stop times of a sound, and its "mods" — the position of the volume, pitch and equalization controls away from "unity" — are recorded on the disk for each reel. If the film will be mixed in-house then the effects are transferred to 16-track tape with SMPTE timecode. All other films are transferred to standard 35 mm mag stripe. The computer automatically prints out a cue sheet for dubbing, noting the name and start-stop times for each effect as it appears in the film.

Should there be a picture cut after the sound editing is completed, the editor need only enter in the new start-stop times for each sound. This feature can be taken a step further if the film is being mixed in-house; if the director doesn't like a sound effect during the mix, the ACCESS system can be patched into the re-recording console, and he or she can audition all other, say, bird tracks doesn't like a sound in the library. When the right one is found, the system can be told to replace all of the old sparrows with finches, as the case may be.

A recent addition has been the dialogue editing program, allowing ACCESS editors to speed up this most tedious sound-editing procedure. In addition to the A-B rolling of

FANTASIA DIGITAL FILM SOUND RECORDING

been one transfer before we loaded everything on a magnetic [film] machine, I think the difference in terms of frequency response and distortion that you would hear would be negligible, if not zero. I think we gained in terms of signal-to-noise ratio, and wow and flutter stability because we were not at the mercy of a film distributor, which is synced to a 'rubbery' line frequency. The distortions you would hear with the digital medium would not have been any better or worse than that you would have heard from analog hiss or modulation noise.

"We gained a lot of versatility in dubbing, because we were able to bring in all of the tracks that we wanted to use in the final theater version, and still have a neatly edited package on two reels, instead of 10 or 12 dubbers. We were also able to maintain much better sync accuracy, because our synchronizer was accurate to a hundredth of a SMPTE frame, which means in terms of phasing accuracy we were much better off. We were able to bring better quality to some areas, and make for a much more flexible dubbing situation.

"But if you take a single shot at the matter, and say if you had done it 24-track, and taken that to the dub, would it have been any worse — the simple answer is no."

Dolby Stereo Release Prints

The re-issue of *Fantasia* will be in both Dolby-encoded, four-track discrete magnetic, and two-track matrixed stereo optical prints. Digital playback outside of the Disney main theater will happen . . . eventually [see sidebar — *Ed*]. The large number of expensive magnetic prints — over 100, at three times the cost each of a standard print with an optical soundtrack — was felt justified by Disney, because of the compromises implied by the encoding matrix.

The four-track master mix was monitored discrete; that is, without having a Dolby DS-4 monitoring unit simulate the effect of the matrix. To make the two-track printing master for the stereo optical negative, the four-track mix was monitored through the 4-2-4 matrix. While the Left-total and Right-total (Lt-Rt) information was being bounced up to two tracks on the A Machine, a 35 mm magnetic printing master was made, as had been done with the four-track printing master.

"We manufactured the surround information for the discrete four-track mix by delaying the M/S mike 60 milliseconds through a digital delay, with the matrix set to box 85 to 90% 'M' content.

. . . *continued overleaf* —

For additional information circle #89

FANTASIA DIGITAL

We tried taking a mix of the three cardioid mikes, and the M/S seemed to give us the most complete picture to put into the surrounds. We didn't put a lot of reverberation in the surrounds."

This surround information was not used for the Dolby Stereo optical mix, and Murphy let the matrix create the surround information. The surround delay in the Dolby Stereo optical format is individually adjusted at each theater.

Archival Storage

Walt Disney Productions is noted for its 20-20 foresight in the care and keeping of its films: all feature and television color negatives are protected against fading and damage by expensive black and white separation positives (which is one reason that *Snow White* will not be "Snow Magenta" the next time the film is reissued.)

Similar care, of course, always has been taken with the master sound elements, although it was not until digital recording came along that these could be protected without going another generation. A program to protect the sound effects library — consisting mostly of mono recordings — was begun last year, and is now almost finished, a Sony PCM-1600 digital processor being used for this project.

Decision to protect the multitrack master elements for feature films has not yet been finalized, pending an engineering study on the standardization of the multitrack digital format.

The archival function of the digital recording medium was a prime reason for its use in *Fantasia*, as Shawn Murphy explains:

"I think that where we bought a lot was in the ability 10 years from now to take these tapes out and make a new-sounding version of this picture in a different format. We will not have to worry about storage problems and losses which inevitably would occur on a magnetic master. Even in the best temperature- and humidity-controlled vault, a magnetic master would not hold information as well as we *think* the digital master will. We haven't had that much time and experience in digital recording.

"But if it works out correctly that the error correction is good enough to fix any errors that happen in regard to level on the tape, then we have gained a lot. It has already been edited so the editing process will not have to be repeated. If someone said, 'Let's do a version with stereo surrounds,' which is something that I would like to see, then we could easily do it." ■■■

* * *

The work of the Walt Disney Productions Sound Department in the digital recording of *Fantasia*, while not quite the landmark effort of its predecessor, has once again set a precedent. No other studio has undertaken the task of recording and mixing a feature film using multitrack recorders, much less using the digital medium. Armed with this experience, and what has been learned from working on the EPCOT films, one can only expect that Disney will apply this knowledge to a greater extent in future films.

DIGITAL SOUND FOR MOTION PICTURES

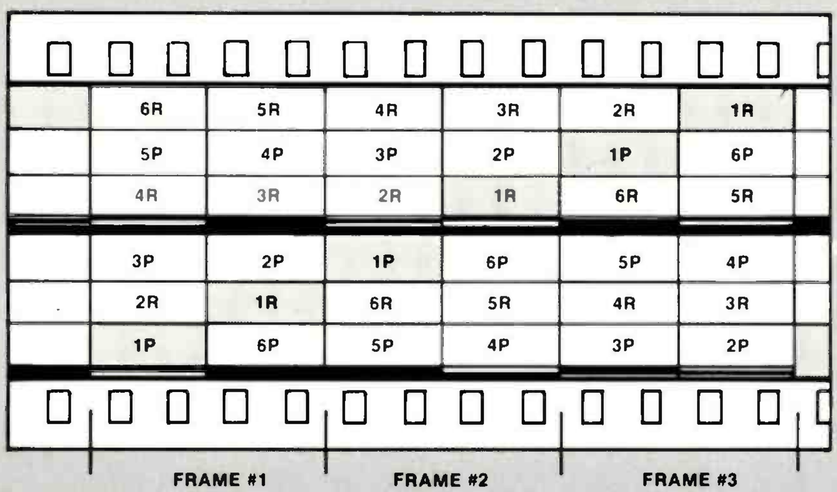
— continued . . .

tracks and "scraping" of noises, ACCESS dialogue editing allows the editor to quickly create background fill from location tracks. This procedure formerly entailed finding segments between words, making a transfer of a small section until a loop could be obtained. Again, this results in considerable time savings.

Digital Fluorescentsound

Regardless of where the digital domain begins and ends in the hybrid systems of today, many re-recording mixers would sleep better knowing that their handiwork was being released in a digital format. The leading (and possibly only) contender in this field is Digital Fluorescentsound, developed and patented by Peter Custer and Dr. George Bird. Instead of abstracting information from available material, we will reprint an explanation of the system in their words from a synopsis of the paper they plan to deliver at the SMPTE Convention in early November:

"Digital fluorescent soundtracking is a high-data density digital sound record multiplexed with the picture. Six discrete channels of 100% redundant soundtracking, requiring a data rate of 10.5 million bits/second, are electrostatically printed, using a high-resolution liquid toner, over the entire 25 mm space between the sprocket holes on 35 mm release prints as a colorless and transparent, brightly fluorescent data image. The soundtrack is invisible on the screen, but when excited by long-wave ultraviolet light in a readout stage that retrofits into the magnetic track penthouse position on the projector, the surface of the film emits bright blue visible light — the data image that is 'counted' by a 2048-pixel line scanner photodiode array [is] error-corrected and converted back into an analog signal, reproducing the source with great accuracy, and without distortion or noise. Recording the sound on the film as numbers that are used to reconstruct the analog waveform . . . separates the sound record, and gives it immunity from the noise inevitably imposed on an analog film soundtrack by the recording medium . . . Film is damaged by both wear and abuse. The error correction possible in a digital sound record, because it uses computer technology, excludes noise caused by grain, splices, dirt and scratches. The fact that the data is buffered —stored for an instant — and precisely clocked out of memory, electronically eliminates distortion of the signal caused by mechanical wow and flutter. These and other properties inherent to digital sound make digital soundtracking uniquely advantageous to motion picture film"



Digital Fluorescentsound Format for six audio channels on standard 35 mm film. Of the 25.4 mm (1 inch) space between the sprocket holes, 24 mm is used for Fluorescent Digital Soundtrack Imaging. The Format Pattern repeats every three frames. (Key: P = Primary Data Patch containing 36,000 bits; R is its identical Redundant Data Patch.)

The data-recording format was designed by Dr. Thomas G. Stockham of Soundstream, Inc., and is shown on this page.

At this time, the cost per theater is expected to be in the neighborhood of $20,000, with half of that going for the electronics, and $5,000 for the soundhead on each projector. The first theater demonstration of working prototype is expected to be made in two years.

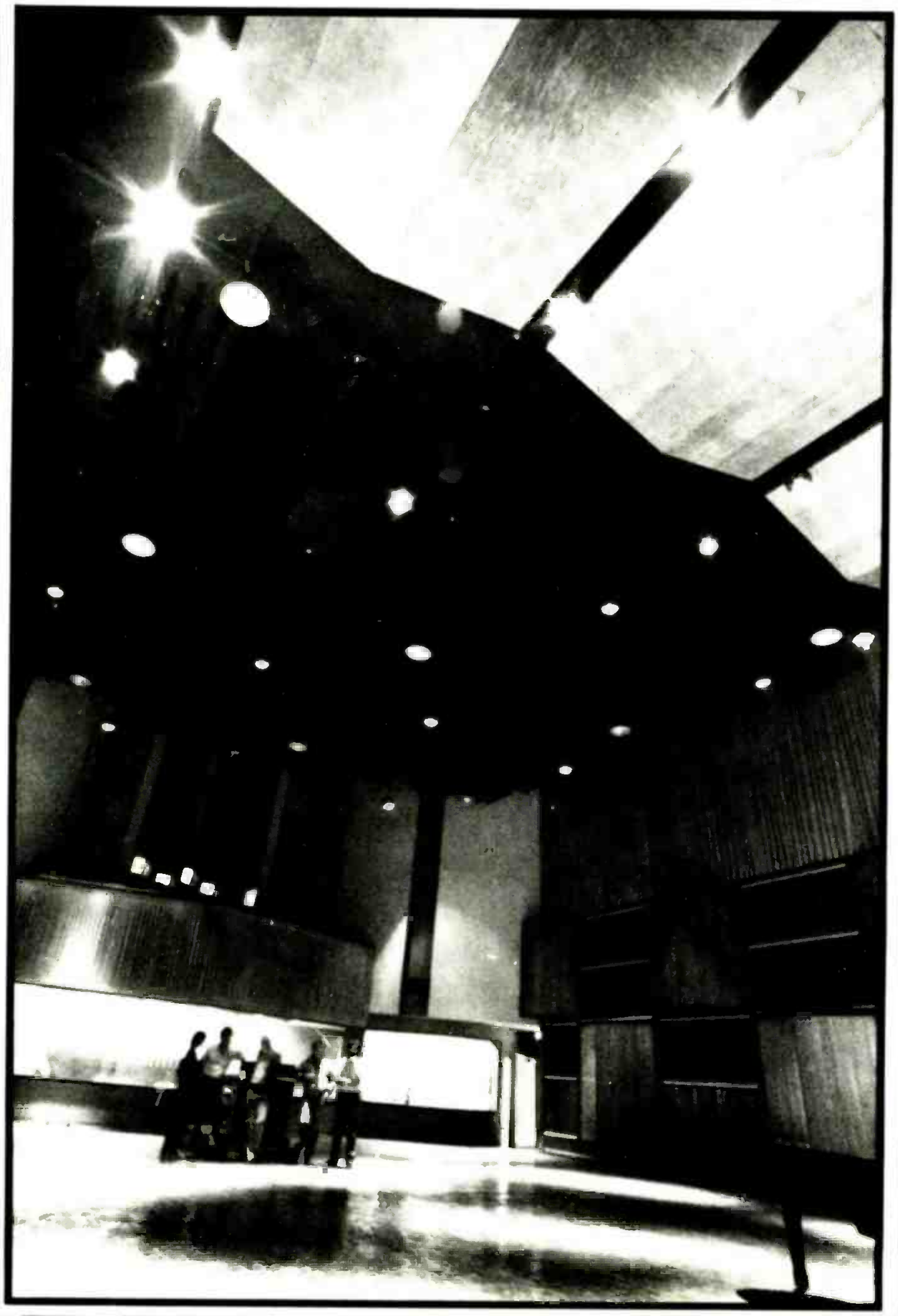
A practical advantage of the proposed format, outside of the potential sonic benefits, will be that for the first time the state-of-the-art soundtrack will be obtained without the price premium of a 70 mm print, thus saving distributors millions of dollars annually. Also, the allotment of the six tracks has not yet been decided — options include stereo surround information, five behind-the-screen channels, and low frequency "boom" information. ■■

MUSIC RECORDING FOR FILM

DESIGN AND CONSTRUCTION OF FILM SCORING STAGE

Record Plant Scoring's Conversion of Glen Glenn Stage on the Paramount Lot, Hollywood

by Neil Brody



Studio M, Record Plant Scoring's newly completed motion picture scoring stage located on Paramount Picture's Hollywood lot, is the product of a unique intra-organizational arrangement involving Record Plant Scoring, Glen Glenn Sound and Paramount Pictures. Since 1968, Glen Glenn has provided all of Paramount's on-the-lot audio services, and currently operates six additional sound studios at Paramount, as well as its own Hollywood studios, and others at the Sound Center in the San Fernando Valley.

Record Plant president Chris Stone explains the new collaboration: "There's a lot of excitement in this town that's been created by this marriage. What Record Plant Scoring has done is to sublet the stage from Glen Glenn with Paramount's consent. Record Plant Scoring has renovated and redesigned the stage, and will run it and book it under the auspices of Paramount and Glen Glenn."

"Now for the first time," Stone adds, "the filmmaker can make one package deal for the entire film of any scope or size. We're coming in here and bringing state-of-the-art [technology] from the music business, and Dan Wallin — who is, we think, the permiere orchestral film scoring mixer in the world."

"The conclusion we've come to," states Joseph Kelly, president of Glen Glenn Studios, "is that the music businsess is very different from television and film. In the past, sound departments and some of the sound services companies have tried to do both, because scoring is a part of producing the soundtrack for a movie. It's never really been successful, because the music business is different . . . a different approach and a whole different field."

Design Considerations

The complete redesign and renovation process for the new scoring stage, which cost in excess of $750,000, and now accommodates over 100 musicians, was based on ideas put forth by Record Plant's scoring mixer Dan Wallin. Wallin's many film scoring credits include soundtracks for *Star Trek II*, *Rocky III*, *Annie*, *The Best Little Whorehouse in Texas*, *Altered States*, *Woodstock* (re-recording only) *Finian's Rainbow*, and *Deliverance*, as well as extensive work in television, radio and records. Wallin has been associated with the Record Plant since February 1981, when it opened its first scoring stage — Studio C — at the Third Street studios in Hollywood.

"The outstanding feature of the new stage," Wallin concedes, "is the acoustics of the recording room. It's the fact that we can totally tune the room for any kind of orchestra of any size, or any kind of rhythm section. We can tune the walls perfectly, and I don't think that that's ever been done before to quite the extent that we have. [The room] is

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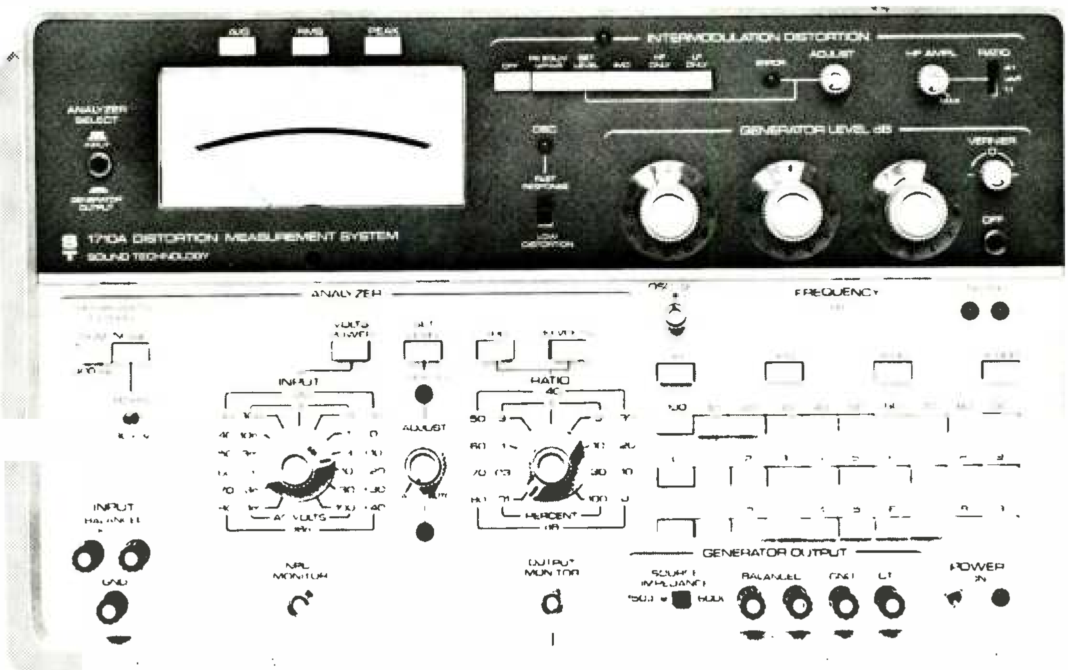

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trapped behind the walls with real true traps, and all the way around the bottom of the floor, which takes care of any room rumble.

"Basically, the acoustics are an accumulation of a whole lot of years of being in different rooms, and finding out what works best — that's what I went for. There's no magic like slide rule mechanics or anything; it's really all based on my experience."

Side walls of the spacious 5,000-square-foot room are composed of several wall-mounted, double-tiered resonating wood panels which, when closed flush to the wall at a 45-degree angle to one other, form a convex reflective surface. When the panels are hinged out from the wall, they expose an absorptive underlayer, which can be regulated, depending on how far the panels are opened. These variable panels are alternated between several double-tiered, rotating panelled areas, constructed of flat wood resonators. These rotating slats can be closed flush to the wall for maximum reflection, or variably rotated to diminish reflection, depending on the desired sound quality.

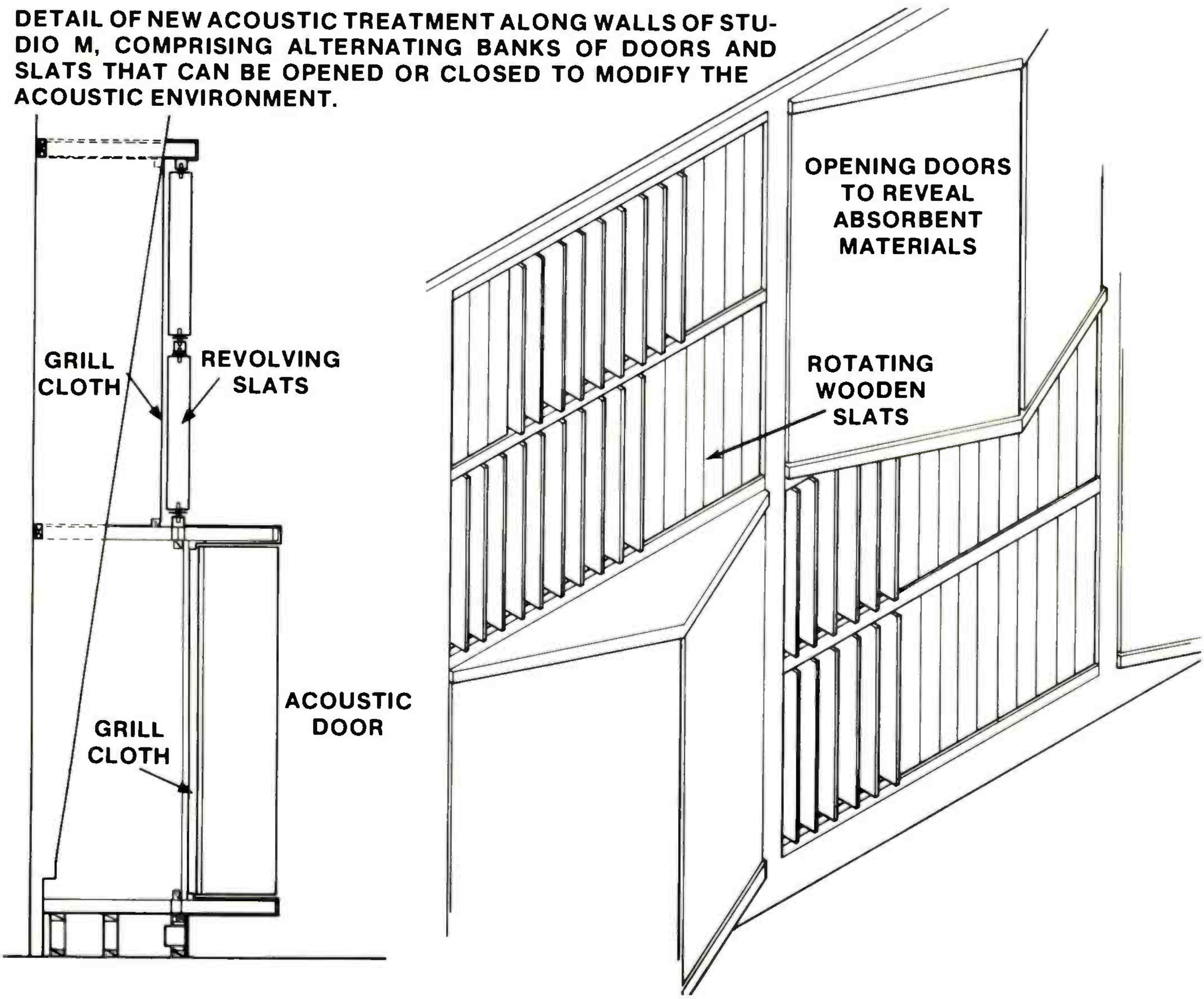
The ceiling, originally 50-foot high, has been dropped to a height of 32 feet, with the front half acoustically trapped to absorb any unwanted random low frequencies. The rear half of the ceiling houses three suspended wooden resonating wedges, designed to break up any direct floor to ceiling "bounce." The 65-foot wide by 75-foot long recording stage utilizes five, four-way Klipsch M-1900 monitors for studio playback, mounted behind the projection screen on the rear wall of the room.

"We left ourselves the option to make the room more live," Wallin continues. "Behind the ceiling treatments are traps that can be opened. We can do whatever we want with them, like put in more 'hardening' surfaces. The stage was designed with the front-end fairly dead, and the back-end more live. It's not live *per se*; it's more 'directed,' actually. Normally, in a room this size, the ceiling can give you a *lot* of problems, so we trapped all the parts that could cause any trouble. The trapping prevents the brass, for example, from coming back out of phase across the ceiling, and cancelling [the in-phase sound]. That's the reason for it being absorptive. The leakage is minimal, and it's amazing how much separation we get. I left the back end of the room fairly live, because that's where I put the percussion, and it sounds much better in a more live environment."

Control-room Hardware

The control room is equipped with a Solid State Logic SL 4000E, 40-input/32-output automation-ready console, which has individual four-band parametric equalizers, limiter-compressors and noise gates on each input strip. Since the monitoring, mixing and headphone requirements for film are considerably different from record work, an additional sub-mixing and monitoring sub-console was designed and built by Record Plant technical staff. This outboard unit interfaces directly with the SSL, and handles up to 40 inputs and up to eight outputs. It contains

DETAIL OF NEW ACOUSTIC TREATMENT ALONG WALLS OF STUDIO M, COMPRISING ALTERNATING BANKS OF DOORS AND SLATS THAT CAN BE OPENED OR CLOSED TO MODIFY THE ACOUSTIC ENVIRONMENT.



monitoring and mixing capability from either the SSL bus outputs, and/or any of the various film formats, as well as simultaneous mono or two-track mixing outputs.

The sub-mixing console has its own bus/track switching, and sends back to the SSL a logic signal to continue the bus or tape selection. The custom-built unit also provides six discrete headphone feed channels, since musicians in various sections of an orchestra usually require different headphone mixes. In addition, the console enables selection of any one of six separate echo *sources* from each channel. Since generally in film scoring the orchestra is mixed *live* directly to a three-track 35 mm magnetic recorder (a 16- or 24-track tape being recorded simultaneously as a back-up), the echo for each section of the orchestra must be kept discrete. In other words, should a violin string section, for example, later be omitted, or the part rewritten, the echo must not have been recorded on any other tracks of the three—track, other than the violin track, so that they can be re-recorded without echo leakage appearing anywhere else in the mix.

Talkback and slate systems are also interfaced between the two consoles. Both originate in the main SSL console, and are linked over to the outboard section. Along with communication with the studio, the talkback system permits independent communication between the mixer, conductor, music editor, machine room, projectionist, and producer's room. The slate system allows slate cues to be made directly from the control room, the machine room, or the music editor.

The control-room monitor system is a five speaker (left, left-center, center, right-center, and right), two-way custom setup utilizing JBL 2235 woofers and 2441 compression drivers, operating with an 800-Hz crossover network. The monitors are equalized with White 4003 third-octive equalizers, and are powered by bi-amplified BGW 750-watt amplifiers, as are the Klipsch studio monitors.

Pen Stevens, chief engineer for the Record Plant, explains the reason for choosing a two-way speaker system: "We felt that with a two-way system we would achieve fewer phase problems, because of the fewer 'points' of sound. The new JBL horn driver goes out pretty far on the top-end, and we decided that we didn't need any additional tweeter. The 800 Hz crossover is about as low as one can go on a two-inch horn driver, and not blow them up at the sound pressure levels that we produce. If you go lower, it's less efficient, and the life of the speaker is much shorter. We are using the same crossover at our Third Street studios, and it's worked out very well."

The machine room at Stage M is equipped with 35 mm magnetic recorders: one three-track, and one mono; along with one non-sprocketed ½-inch four-track recorder, and one ¼-inch, mono recorder equipped with pilot-tone. Because of their increased track width — 300 mil, compared with 70 mil for 16-track and 43 mil for 24-track machines — 35 mm magnetic recorders have a greater signal-to-noise ratio. Since they enable first-generation recordings to be edited for the final mix, 35 mm transports are generally preferred over multitrack recorders for the master film mix; very often the only use of multitrack is for safety back-up, and to provide a record album mix.

In addition to the 35 mm mag machines, clients at Studio M have the option of selecting either digital or analog multitrack format recorders. The room will be equipped with an, as yet unconfirmed, multitrack digital system, one or more synchronized 3M 24-track recorders, and Ampex ATR-100 four- and two-track machines.

"The control room is really versatile," Wallin considers. "The SSL console is very quiet and [has] everything a mixer could desire. The modules are really useful; with the parametric equalization, gating and limiting it requires only a small amount of outboard gear.

"The outboard mixing panel is unique, in that it allows you to punch up any combination of film mixing formats — six-track, four-track, three-track, two-track, or mono. The six separate headphone feeds for musicians are great. Having the ability to discretely separate the echo sends and returns is really important, because a lot of times during dubbing the client might want to re-do part of the orchestra, and you can't leave a trace of echo on any of the other tracks from the original, so you *must* be able to keep all your echo separate. It works out very well because we have several echo sources to choose from, including the live chamber, several EMT 140s, one EMT 250, one EMT 251 [both digital reverbs], and the [AKG] BX-20s.

"I really believe that we are the only scoring stage that is truly state-of-the-art. In terms of the acoustics, electronics and everything else that we have done, it's really way ahead of everything else."

Custom-designed 40-input sub-console interfaces directly with the main SSL board, and provides separate monitor and fold-back mixes. Six discrete headphone send busses are available.

View across the sub-mixer to the control-room's main 40-input/32-output Solid State Logic SL 4000E automation-ready console, equipped with compressor-limiters on each channel.

Digital Recording

As for the role that digital will play in the future of film audio, "I'm really impressed with the digital recordings that we've done like *Star Trek II* and *Annie*," Wallin says, "and overall digital is really superior to analog . . . it's the next generation of recording. I like the dynamic range, the quietness and the fact that you don't need to use noise-reduction. You have to if you're using multitracks, or the noise will kill you — even at elevated levels at 30 IPS, and especially with double 24-tracks. Even though for film you eventually have to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SCORING STAGE DESIGN AND CONSTRUCTION

go to mag, and then to optical, if you improve the sound to begin with you don't have as much degradation."

Joel Sill, vice-president of music at Paramount, sees digital sound as playing a prominent role in the future of film audio: "Digital sound for film is definitely another step foward in the state-of-the-art," he offers, "especially for the larger-budget films. Studio M can facilitate over 100 musicians and record to digital.

"There are more films being made by filmmakers who have come up through a more contemporary musical environment. Consequently, they are more aware of contemporary music and the musicians themselves. The blend of a contemporary musician into the film areas has always been a bridge that both the film and record industries have wanted to have crossed. With Studio M, we are providing an environment that is state-of-the-art, *and* sensitive to a record musician. It's the ideal marriage."

Obviously, economics play an important role in any endeavor that requires significant advance money. Record Plant's Chris Stone cites economics as the deciding factor in the decision to go ahead and renovate the new room.

"When Record Plant first got into the scoring business," he recalls, "someone said to me. 'In film, music is an unfortunate necessity,' and it has been treated that way. Now, with the economic success of such high-quality soundtrack albums as *Chariots of Fire*, *The Jazz Singer* and *Urban Cowboy*, music has demonstrated that sometimes it can save or promote the film, and therefore has become much more important. That realization, coupled with changes in the state-of-the-art of sound, have become much more of a factor in the filmmaker's mind. The picture is one thing and the music is the other; it takes *both* of them to make the film and economic success."

"I believe that digital sound will even do more for that," Stone continues. "Record Plant Scoring would not be spending the money to rebuild this stage if we didn't totally believe that film sound was going to be a major factor in our growth. We're looking for virtually all our growth to come from scoring and from remotes. And much of our remotes will be for the visual medium — television and film."

* * *

Currently, the film industry is coming off its most profitable summer season ever. Several of the major motion picture studios already have committed to stepping up film production substantially for 1983. A film studio's recognition of potential profits and sales promotion brought about by the marketing of soundtrack albums from successful feature films — along with a growing public enthusiasm for quality sound leaves the audio future open to an optimistic forecast for growth in the motion-picture industry.

It may also be seen as a positive trend that many of the people previously associated with the record industry, and which is now experiencing a period of substantial decline, are pooling their audio talents and joining forces with the film industry . . . as Record Plant Scoring, Paramount Pictures, Glen Glenn Sound and others have demonstrated.

■■■

MICROPHONE AND RECORDING TECHNIQUES FOR FILM SCORING

— A Conversation with Record Plant's Danny Wallin

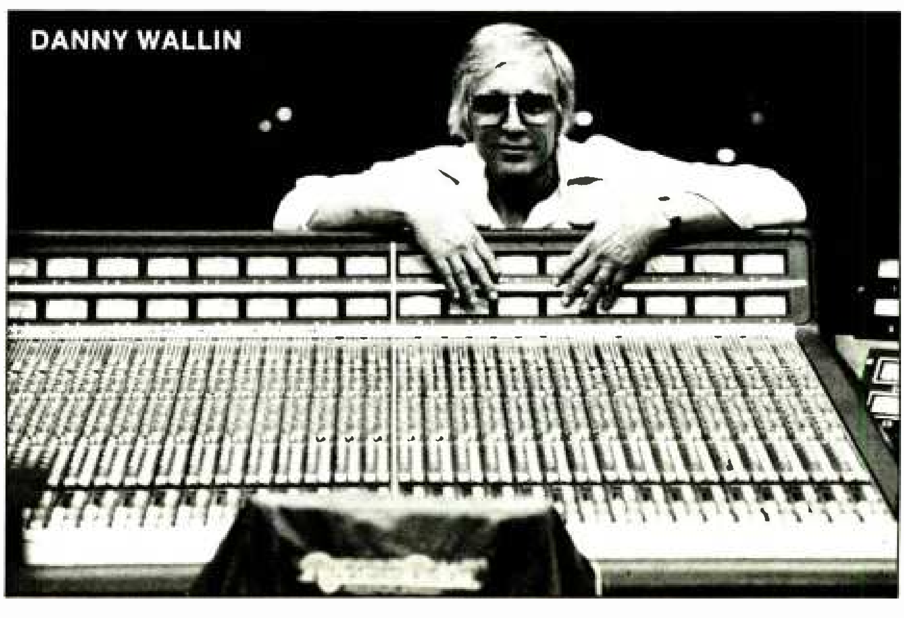
Compared to "conventional" record sessions, recording the music of a live orchestra for use in a motion picture, or any of the other visual arts, shares much of the same basic responsibilities, procedures, and equipment. Many of the same microphones, tape machines, consoles, monitoring and communication systems are used, as well as the studio environments necessary to accurately record and evaluate the music. In addition to sharing common technical components, both processes rely on the engineer-mixer's talents, know-how, and ability to interpret and blend the separate audio elements into an overall cohesive sound.

Moreover, music scoring for the visual arts demands some additional requirements for an engineer to consider, which are particular to recording sound for use with picture. In music scoring, since most of the visual music "cues" are used as "underscoring," dialogue clarity must always be given the highest priority. Consequently, the music must be recorded and mixed in such a way that it is always clear and audible, but never competes with the dialogue.

Since moving into film-music in the late Fifties, scoring mixer Dan Wallin, whose career in audio began in 1946 as a symphonic mixer for the CBS Radio Network, has worked on soundtracks for over 700 motion pictures.

"The most significant aspect to keep in mind when recording music for film," he stresses, "is [to ask] how the music is going to play against the dialogue. In film, dialogue is 'king' and should be regarded as the lead vocal. What the scoring mixer must acheive is to record the music in such a way as to allow it to exist on a slightly different acoustical plane than the dialogue. This is the reason why a typical scoring sound is conceptually closer to the classical sound, rather than the contemporary pop record sound. There's so much more equalized presence on a typical record than a film score, because if you treated the film sound with that much unnatural presence it would overpower the dialogue. You would have to play it so low against the dialogue that its true characteristics — a thin, 'pinched-down' sound, which only sounds good if it's played loud — would start to show up.

"I always bear in mind when I'm mixing that the music is *under* dialogue, and I try to keep my dynamics in such a range so that no matter how low they play, you're still going to hear everything in the orchestra. If you listen to the old Warner Brothers' pictures, they've got the music just roaring away,


DANNY WALLIN

and every single word of the dialogue is *perfectly* clear—because it's in a whole different space. The trick is knowing the right mixes, and where to put them . . . and your conceptual feeling for the orchestra.

"In film 99% of the time we do everything at once, and we generally don't overdub. The basic difference between film scoring and record making is the approach."

Direct to Three-Track

In practice, Wallin relies primarily on his knowledge of room acoustics, and microphone selection and placement to achieve a desired sound. Film scoring, unlike record making, requires that all the aesthetic decisions, such as balance, panning, EQ and echo, be made at the time of recording. Usually the instruments are mixed "live" to three-track 35 mm sprocketed magnetic tape, while utilizing a multitrack recorder (usually 16 - or 24-track) as a back-up should the music cue later necessitate, for whatever reason, overdubbing, replacement or re-mixing, or for a soundtrack record release. Track designation of the three-track mag music score is as follows: track 1 mix left, track 2 mix center, and track 3 mix right. For this reason, everything must be decided on the spot, and there is little room for error or indecision on the part of the scoring mixer. Critical evaluation and correction of possible problems within the orchestra, such as phase cancellation, intermodulation distortion and mono/stereo compatibility, must be resolved in the first few minutes of a scoring date.

"I use very little EQ," Wallin says, "and roll off some bottom when necessary, except on the drums which needs to be punched-up a bit. I give some 'pop' to the kick, some 'snap' and bottom to the snare, and some brightness to the cymbals. This room [Studio M] has so little low-end problems in it, I can pretty much get the sound I'm after with the right michrophone, rather than the equalizer.

"When an orchestra is recorded correctly, there's no cancellation, no intermodulation distortion, and no phase-shifting. When you go mono from the stereo [mix], everything adds up and it sounds just as loud; the instruments don't disappear back in the mix.

"With regard to intermodulation distortion, for most things tube mikes sound better to me than transistor mikes, and they have a lot more forgiving latitude. Plus the fact that with tubes you have tremendous headroom. What I try to do in the first few minutes of listening to an orchestra is to work out any intermodulation distortion problems that may exist. I know that most engineers understand the technical meaning of intermodulation distortion, but I don't know how many of them are aware of how much that actually occurs in, say, a string section. I listen to the physical positioning of the mikes themselves, and then check them against every other microphone in the section.

"I'll start off listening to the inside pair, and then the outside pair. If I hear any IM, I'll adjust the position of that mike. If that doesn't solve it, I'll ask the concertmaster of the section to perhaps change the 'divisi' — the way the instruments are physically positioned in the section — because they are acoustically 'rubbing together' and creating intermodulation. It's the job of the mixer to place the mikes so that there is no trace of intermodulation distortion, and everything is in phase and sounds clean and pure.

"Using a lesser number of mikes reduces your chances of phase-shift, and you begin to hear more of the harmonic structure of the whole section.

I'm convinced that you can get just as hot a sound by using a couple of mikes, as you can by using four or eight; of course, it all depends on the size of the section.

"Placement of the overhead stereo room mike is also critical, and you have to be careful of the intermodulation distortion relationship between it and the closer mikes. Once I get the IM worked out of the closer mikes, I'll listen to the whole section against the overall microhone. If I still hear IM, I move the positioning of the room mike until all the roughness or impurities are gone in the sound.

"IM is the one thing that gets worse every generation and every transfer. Every section, be it strings, woodwinds, brass or rhythm, is subject to IM, and it has to be worked out in order for everything to sound good and be in phase."

Session Setup

Setting up a stage for a large orchestra scoring session is no simple matter. Microphones, chairs, music stands, reading lamps, room lights, headphones, baffles, cables, and so on, must all be arranged, adjusted and checked-out prior to the downbeat. Many times, one or two run-throughs of the first music cue may be all the time that a scoring mixer has to get the sound balanced, and to check that the headphone mix is adjusted to the musicians' satifaction.

The high cost of recording a large orchestra imposes a tremendous expenditure on a film maker, and there is very little tolerance of wasted time. Technical problems must be kept to a minimum, since the session musicians are being paid regardless of whether or not any recording is being done. Communication between the scoring mixer, conductor, musicians, machine room opera-

. . . *continued overleaf* —

tor(s), projectionists, director and producer, and music editor is of utmost importance, and easy access to all participants is crucial, Wallin considers. Speed, flexibility and steady nerves are important attributes for the scoring mixer to possess, since the responsibility for coordinating everythings is theirs. Because of these restraints, a mixer must be able to shift gears effortlessly between technical, aesthetic and political judgments, without breaking stride and throwing off the momentum.

"There are many other aspects involved here beyond good engineering and good acoustics," he confides. "For one thing, the comfort of the musicians is very important. If they are required to listen to clicks and so forth, you have to have good, *clean* headphone busses, and enough to go around. We have six separate busses, with the availability of eight. Each player might need something a little different — especially if there is a unison line and the players across the room have to be able to hear each other quickly and efficiently.

"Colors mean a lot . . . it all affects the mood. Wood is very important for the resonance factor; for the 'warmth' of it. It's closer to nature, and that's what the instruments like. They are little Helmholtz resonators; the violins and everything all respond better to wood.

"I don't like concrete floors even if they have wood on them. We have a wood parquet floor that is slightly resonant. I didn't want it to cavitate, but I did want a little something happening to it . . . so that when a musician bows the bass or cello he gets back a little vibration, rather than deadness. It's the same with brass or woodwinds — it deadens their sound if you have real absorbent baffles around them too much. I wanted to create an environment where I didn't have to use a lot of gobos and flats; where there would still be great separation . . . knowing that I'd still have to put the drums in a booth.

"I wanted a room where you can't actually hear the walls; where you don't have any long-term leakage coming back to you out-of-phase. For example, if the brass are blowing and they are positioned correctly, whatever leakage you get in the strings is direct leakage, and not coming back to them from the walls behind the string. Since the back of a Neumann U67 microphone, for example, is down in level from the front by some 40 dB, that kind of direct leakage is really negligible. What *kills* you is the out-of-phase long-term leakage; that's what muddies everything up and makes you look for more isolation.

"Due to the bass and other trappings, and the way everything is moved around, it's one of the quietest rooms that I've ever heard. I've had 40 microphones open and, until somebody does something, it's extraordinarily quiet sounding in there."

"If you walk into the room when the orchestra is playing, there's a very warm string sound. The players love to play here; if you put a flat next to them they hate it, because it sucks up their natural sound and they start to overblow, since they can't really hear themselves. If they blow harder, especially in the brass, the instrument starts to saturate.

"If you can achieve a real good, pure sound, which sounds like the musicians playing in the room, then you've accomplished what you've set out to do — which is to reproduce the orchestra in its most natural or enchanced form."

Track Assignments and Panning

Once the music score has been recorded, and the other necessary post-production work completed, the film sound is then ready to be mixed by the re-recording mixers on a dubbing stage. There, while watching the scene-to-be-mixed on the picture screen, the dialogue, music and sound-effects mixers rehearse balancing together the playback of individual 35 mm mag sound "units," until a satisfactory balance is achieved. Once this is accomplished, they proceed to record the master mix on an additional 35 mm mag recorder.

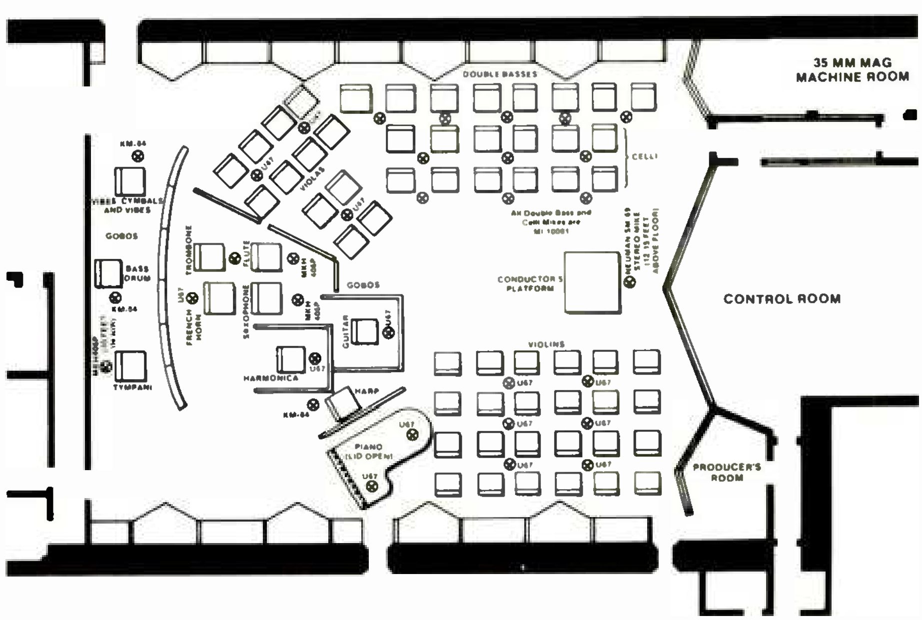
The track format of this mix is determined by how the film is going to be released in more than one format, which will necessitate additional mixes. For example, different versions may be mixed for mono, fourtrack Dolby Stereo, 70 mm six-track Dolby Stereo, television, foreign release (mixed the same but with the exclusion of the dialogue, which must later be re-dubbed into the language of the exhibiting country), and sometimes even separate versions for in-flight airplane screening.

Since there are so many formats to be considered — and it's not always known to the mixer at the time of the scoring session which format(s) will be used later in the dubbing — the scoring mixer will supply the three-track 35 mm mag tape in a format that can be adapted to all applications. Therefore, the panning of the orchestra in the stereo perspective is of the *utmost* concern, since the film may be released as both stereo and/or mono, and the music must be compatible in either format.

"If I'm doing a feature film where a great deal of care is given to the overall balance, I divide the orchestra as follows: high strings on the left; woodwinds, rhythm, keyboards, bass and percussion in the center; and the viola and celli on the right. I usually split the horns, if I can, with the first on the left, and the seconds on the right. I put the trumpets in the middle, the trombones on the right, and the tuba in the center. I always try to put the bass in the middle, and divide it up that way, so that if you put [the mix] up as stereo it really sounds like *good* stereo. I also put the overall stereo mike left and right.

"If I'm doing a television show, however, I do it differently. I put all the strings on one track; rhythm, keyboards and percussion on a second track; and all the brass and woods on the third

ROOM LAYOUT FOR *FRANCES* SCORING SESSION AT RECORD PLANT STUDIO "M", AUGUST 1982. ENGINEER: DANNY WALLIN



track."

Microphone Selection

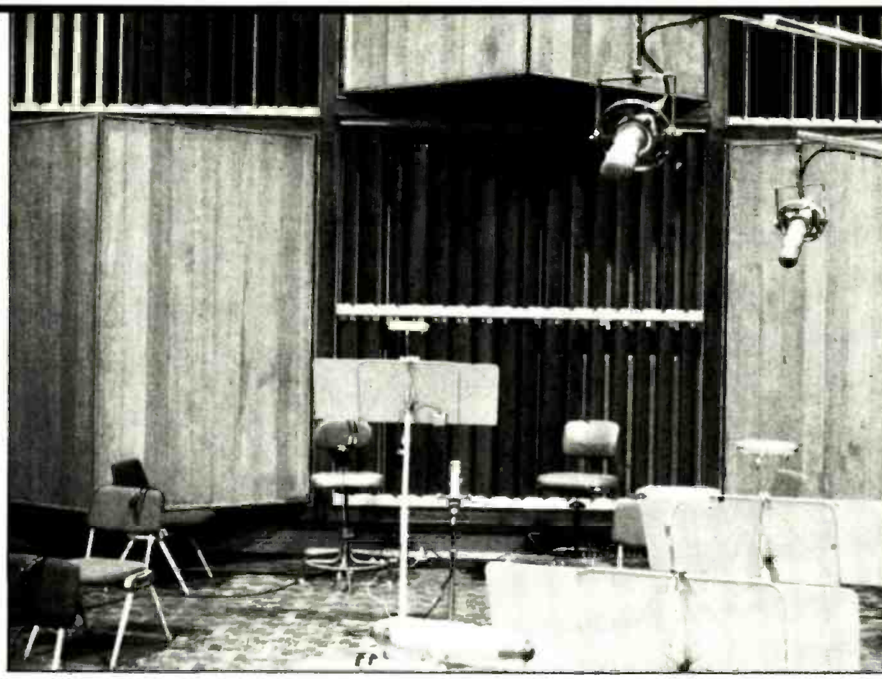
As for microphones, Wallin uses a great deal of tube mikes which, he feels, are the most compatible with the aesthetic requirements of film scoring. "I like the sound of Neumann U67," he says, "and I like the pattern; it has an exceptionally clean sound. If you compare a 67 and a U87 on, say, a vocal, the first pass through it seems that the 87 has more presence, because of the sound of the transistors. But if you hear that same vocal several generations later, the 67's presence is still there, and the 87's presence is already gone, due to odd-order harmonic distortion of the transistors.

"I like the 67 on violins, the U47 on violas, and the RCA 10001 ribbon mike on bass and celli. I also like the Sennheiser MKH406 on woodwinds. I use the KM-84 on percussion, because I'll get a nice bright, clean sound with very little or no clipping — if I'm careful with the padding to the pre-amp. I tend to use very few dynamic mikes, except on the drum kit.

"For recording brass, I like to get the 'bell-tone' sound of the brass that you get with a good ribbon mike. I particularly don't like that 'crushed-down,' small brass sound that seems to be popular in record recordings. They use condensor microphones, and the voltage comes out of the thing so high that all the transients are crushed, because it usually overloads the pre-amp. I like that nice big, 'open' bell-tone because it sounds fat; you can only get that with a ribbon mike that's not too close to the horn. But, you've got to be in a good room too. The room is the *key* to everything."

As far as the rhythm section is concerned, Wallin points out, "Today everyone expects a 'record sound' rhythm section, because the record people did more of that than we did. The 'California Drum Sound,' as it evolved, was something I was aware of all the time. I mike the kick, snare, toms, hi-hat and overheads separately, like anybody else would. Generally the snare and kick would be in the center, and I do mike the toms and cymbals in stereo — so if they want to split it up that way they can, and the toms can travel from left to right. For ambience on the drums, today they seem to want a more dry, up-front, direct sound. But we also use more room sound, depending on the circumstance. Guitars are treated basically the way you would on a record date.

"There's a lot of talented people in the record industry," Wallin concludes, "especially those that have been around a long time, who have begun to recognize that there is more than one kick drum, or Fender bass sound. There are other really good sounds that are just as important as the 'smashed-down,' brash sound and the 'skinny-distorted' string sound, with the rhythm section mixed



so far forward that everyone else might as well have stayed home. It *could* be better if it all was contributory. Instead, it sounds like a mistake; like crosstalk. You could erase the tape completely and it's still hanging in there.

"There's a reason for everything I do — every microphone, every move, every balance. I haven't guessed at a sound in over 10 years. Up to that time, I pretty much knew what was going to happen, but I wasn't able to identify it as quickly and readily when a problem occurred, or how to cure it instantly — leakage, phase-shift, or whatever. When you are recording a film score and you've got 80 musicians out there, there are *no* second chances. So, for your own self-protection, you had better do it right the *first* time." ■■■

COMPONENT EVALUATION

SPECSMANSHIP AND THE NEW GENERATION OF VCA

What a Manufacturer's Published Specifications May Not be Telling a Potential User

by Paul C. Buff
Technical Consultant, Valley People, Inc.

For those of you who don't know what the term "specsmanship" means, let me cite an example: A manufacturer has a new tape machine design that he wishes to sell in the marketplace. While being very inexpensive, it is not a very good design — it is very noisy, and has poor frequency response. Now, the manufacturer knows that nobody will buy it if it has poor specs, or no specs. He also knows he will be sued if he publishes false specs. So he creates a method of specification which indicates excellent performance, stated in such a way as to be technically correct and verifiable.

By manipulating the test conditions to suit the product, a picture of superior performance may be projected, even though the product itself might be inferior. Thus, in its worse case, "specsmanship" is a clever series of manipulations and omissions designed to fool the average user into thinking the product is acceptable for the purpose, even though it might perform terribly in actual use.

There are but two methods known to combat the ill effects of specsmanship: legislation and standardization of terms and methods; and the cultivation of a higher order of comprehension on the part of the user. The user who relies on the former method may have a very long and rocky road ahead for himself, since legislative bodies are bureaucratic — i.e. agonizingly slow and ineffectual — and easily outpaced by the specsman. Thus, the only real defense against such tactics is education of the user to a point where he can more fully comprehend the technical material presented to him for analysis.

At present, the specsmanship situation in pro-audio could probably be called moderate. There are a good number of manufacturers that exert every effort to project the truest possible image of the performance of their offerings. They are generally successful, since there are a fair number of consumers that are able to interpret the specs, and buy accordingly.

In the middle ground, there are a large number of manufacturers who, while basically honest in their specs, are unashamed to perform a certain amount of specmanship — adjusting what is presented to coincide with what the consumer *wants* to see. The success level in this group is excellent, since there are plenty of customers who don't really know how to comprehend the specs, and who may never come to the realization that the equipment does not perform quite in the manner indicated by the specs; if it makes music, it's okay.

At the bottom of the heap are a certain number of outright specsmen. At times, such manufacturers are not fully dishonest since they, themselves, may not even understand what they are specifying. Fortunately, there are few professional users gullible enough to swallow very much of this. Nevertheless, the latter exist in numbers sufficient to allow the sporadic success of a few blatant specsmen and their companies.

VCA Specsmanship

Voltage-controlled amplifiers were introduced to the pro-audio industry about a decade ago, and play an essential role in modern signal processing. They allow a degree of signal manipulation far beyond what was possible in the past, in areas of console automation or programming, and in dynamics control — limiting, compression, expansion, filtering, etc. As with many new technologies, VCAs gained some early proponents and opponents. While the benefits of the VCA structure were immediately recognized from a standpoint of increased system usefulness, there evolved a mistrust for the "sound" of VCAs from a point of pure audio transparency.

The problem here may be traced primarily to the newness of the technology. Neither the manufacturers that employed these devices in their equipment, nor the

SPECSMANSHIP
AND THE NEW GENERATION OF VCA

users who bought same, fully understood the relationships between the numbers presented and the actual in-use performance. Only when these circuits were analyzed and measured under actual use conditions was it realized that rather alarming amounts of IMD (Intermodulation Distortion), modulation noise, and thermally-induced distortion drift were evident.

This author continues to be amazed at how little industry understanding prevails regarding VCAs and their implementation to this day, and how VCAs, in general, continue to be viewed as incomprehensible little cubes that defy conventional measurement techniques. Also, I am concerned that, while certain present-day VCAs offer completely transparent performance and understandable documentation, new products are appearing on the market that may be less than transparent, but which are specified in such a manner as to indicate excellence. In short, this writer fears a return to another era of mistrust for all VCAs out of specsmanship and non-comprehension.

The Requirements

If we are to understand what level of performance is required for VCA transparency, we must first review how they are used. We must also get some handle on what constitutes "transparency" in this decade of pro-audio.

The most prevalent use of VCAs also happens to be the most critical use: the employment of a VCA in the main signal path of a console in place of the conventional fader. All audio signals pass this point at least two times — once during tracking, and once during mixdown; any VCA anomolies are thus injected into the signal twice. We also need to realize that the signal level at the fader point of most consoles is not confined to a tidy "nominal level," but is very apt to cover quite a range, as influenced by the position of mike pads, sound pressure levels in the studio equalization used, and the gain or loss of peripheral equipment inserted.

Thus, VCA inputs need to be able to handle a very wide range of input levels, while producing very low levels of distortion. From a noise standpoint, it should be recognized that, in console use, the outputs of a number of VCAs will be summed together in the mixdown busses, thereby placing stringent demands on the VCA to pass signals while producing extremely low noise levels. Unfortunately, these two requirements (low distortion and low noise) are difficult to achieve at once in VCAs, since the techniques employed to reduce noise tend generally to increase distortion, and vice versa. Such a difficulty was primarily responsible for the shortcomings of early VCAs, and for their acquired reputation; one could achieve either low noise or low distortion, but *not* both at once.

Now, let us get a bit more specific and try to define, in a pro-audio usage, what constitutes "low noise and distortion." We must first consider that both are cumulative anomolies — they build up from multiple sources. It has long been considered that, in a professional system, the console should not add significant amounts of noise or distortion to that already induced by the storage medium, the tape machine. In the past, this was an easy task, since non-noise-reduced analog tape machines were notorious for the production of unwanted noise and distortion. Even then, one could find any number of "professional" consoles which, in fact, did seriously add to the non-transparency of analog tape mediums, through just plain poor design. With the advent of tape noise reduction, console requirements increased, and the desired criteria was unsatisfied with increasing regularity. In this decade, we are beginning to see digital recording techniques and, if the non-degradatory-console-criteria is to be met, or even approached, some fairly serious consideration must be given to the console. It is no longer practical to state that 0.1% distortion and 70 dB SNR will do the job, without some real study concerning the conditions under which these numbers are derived.

In a previous *R-e/p* article [*Now You Hear It! Now You Don't. Perceiving Audio Noise & Distortion*, June 1979 issue], this author attempted to pinpoint what values of noise and distortion might be needed to assure effective transparency (no audible degradation whatsoever) from a direct signal source, considering modern multitrack console techniques. Rather than reiterating those considerations here, it is recommended that the reader refer to this article for more details regarding how the stated numbers were evolved.

In its final analysis, the paper indicated that a good criteria for transparency was a maximum distortion of 0.03% as measured by THD, SMPTE IMD, Twin-Tone IMD and other representative means, at *any signal level that might be encountered in use*. It was also shown that an *actual signal-to-noise ratio* of at least 85 dB (nominal signal level to noise floor) is needed, and that sufficient headroom exist above the nominal signal level to accommodate signal peaks, EQ, etc., without an increase in distortion above the 0.03% figure mentioned. While these specifications may appear stringent, they are in fact achievable, even in VCAs, and are truly realistic in view of the potential offered by digital recording.

Speaking of digital recording, let us take a look at what sort of noise and distortion we can expect here, and how it compares with the criteria just established. In a 16-bit digital encoding scheme, we can expect a total signal-to-noise ratio of 96 dB, measured from clipping to noise. However, we cannot expect to encode signals right at the clipping point, but must leave a certain amount of headroom; at least 16 dB is required to assure non-clipping of signal transients — 20 dB is preferable. Thus, we can expect the nominal signal-to-noise to be in the range of 76 to 80 dB. As far as distortion is concerned, we can expect figures around 0.01% for signals near the clipping point, increasing with *decreasing* signal levels. Thus, when taking the 16 to 20 dB of headroom, we might expect distortion figures in the 0.03 to 0.06% range. Much work has yet to be done on the specifics of digital-induced distortion forms, thus these numbers are presented as rough approximations. One fact is clear though: the criteria we set out for the console are not excessive, as they are just barely capable of handling the digital source signal without significant degradation. To ensure a margin of safety, however, it would be ideal for the console system to exhibit somewhat lower levels of noise and distortion than those stated.

Now, let us look at the input headroom requirement of the console VCA. Not only must it accept high crest factor transients, such as found on percussive material, but it must also give some leeway for gain variations caused by EQ, mike trim settings, peripheral devices, etc. Thus, it is desireable to have 24 dB or more headroom between the "nominal" signal level, and the point where serious VCA distortion sets in. Output headroom is not nearly as critical, since the operator will adjust the VCA gain to a relatively fixed output level at the console busses. Here, a 20 dB or so headroom figure is accepted.

SMPTE IMD

There is one more thing we should look at very carefully when analyzing any piece of pro-audio gear — SMPTE intermodulation distortion. When signals are subjected to the SMPTE IMD distortion form, the result is a modulation of one set of signals by the excursions of another. Most typically, this effect is manifest in a modulation of the delicate high frequencies by high magnitude low-frequency signals, and the SMPTE method is designed to be most sensitive to this specific effect. We have all heard the effect as a "buzziness" of high frequencies during strong low frequencies. Most VCA designs are particularly sensitive to this distortion form, especially when operated at relatively high input signal levels. Specifically, it is almost gospel that, in any VCA design, the measured SMPTE IMD will be from *three to four times* the measured THD distortion, under any given set of signal/gain conditions. It is also near gospel that for each 6 dB rise in signal level, a doubling of SMPTE IMD products will result. Thus, even though a VCA might be specified as having 0.01% THD at *nominal* signal level, the same device has a good probability of exhibiting 40 times this amount (0.4%) of SMPTE IMD at a signal level 20 dB above nominal.

Parameters Unique to VCAs

We should be aware of the importance of certain performance parameters that are unique to the VCA structure. First, the gain control characteristics. Since the

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purpose of a VCA is to vary gain in accordance with an applied control voltage or current, we need to know how the device responds in this respect. It has become pretty much accepted that a VCA which responds on an antilogarithmic or "dB vs Volt" basis is most useful in audio systems. Essentially, all VCAs available have this sort of response. Over what gain control range the device can operate, and at what degree of control accuracy, is usually a function of which general family the VCA is derived from.

Devices derived from the log/antilog family are characterized as having a superior range of gain control ranging from attenuation through gain, and usually exhibit a very precise control versus gain law over the entire control range.

In contrast, devices derived from the transconductance multiplier family are inherently restricted to operating primarily in the attenuation-only quadrant, and have a tendency to develop nonlinearities in the control versus gain curve near the top of the gain control range. While this may not be serious for many uses, it can be disasterous in applications requiring close tracking, such as in grouping use.

Thus, the user needs to be aware of both the gain control range and the degree of tracking accuracy which might expected be in his application.

1. Attenuation-only Devices versus True VCAs:

It is often thought that whether or not a device is capable of forward gain is academic . . . that an external operational amplifier can supply forward gain to an attenuation-only device, and cause the total network to be capable of producing either gain or loss. While this is true on the surface, it causes important tradeoffs to develop that are not present in a true VCA design. Let us say that it is desireable to enable 40 dB of forward gain to occur, using a VCA device which is inherently capable of a gain range of -100 dB to unity. This would require a 40 dB post amplifier. Now the overall network has a gain range of -60 to +40 dB.

In order to achieve the desired forward gain, all noise parameters have been elevated by 40 dB, and the device shutoff has been rendered insufficient for most applications. In addition, other important parameters, such as control rejection and power supply rejection, have been worsened by 40 dB. While indeed the network is now capable of both gain and loss, it has been rendered rather unsuitable for most professional applications. By contrast, a typical true VCA might exhibit a gain control range of from -100 dB to +50 dB without any of the tradeoffs.

2. Control Rejection:

In a perfect VCA, one could change gain under no-signal conditions, and have no change in the output signal (except a change in noise level). In the real world, however, small internal DC voltages are amplified by the device in varying amounts, depending on the VCA gain. Thus, gain changes can result in a changing DC level at the device output. If the gain is changed rapidly, this changing DC output level can be heard as a "click" or a "thump." Depending on the VCA design and quality, such an error signal can be as small as 3 or 5 millivolts (essentially inaudible), to as high as several hundred millivolts (very audible). Thus, the user needs to know what the control rejection is, in the application for which he will be using the device.

3. Modulation Noise:

In one specific category of VCA (the Class A/B log/antilog type) it is very easy to achieve extremely low noise levels under quiescent conditions with *no input signal*. However, when input signals are applied to such a device, a dramatic increase in noise levels can occur (as much as 30 dB). Thus, the noise floor actually rides up and down with signal excursions in a fashion much like that which occurs in tape noise-reduction units. In such a device, the application of near subsonic signals can cause a very audible noisepumping effect. Even with normal audibility signals, the result can be a dramatic increase in noise, with respect to that which might be expected by the static noise level specs or test results. Thus, the user needs to be fully aware of what the noise performance will be *during* signal, since this is the condition under which he will ultimately use the device.

4. Thermal Drift:

Essentially, all VCAs presently available employ the inherently log transfer characteristic of the bipolar silicon transistor as a means of obtaining voltage variable gain, or multiplication. While the characteristic is exceedingly well suited to the task, another characteristic of the transistor is that it is very temperature sensitive in its log transfer scaling. It should be noted that this temperature sensitivity is not based on the absolute temperature, rather that it is a function of temperature *differences* between the various transistor elements used in the design.

Depending on the VCA structure chosen, temperature differences between the transistor elements can be manifest as either temperature-induced

. . . continued overleaf —

TABLE 1: AUTHOR'S ANALYSIS OF VALLEY PEOPLE ECG 101 VCA

Parameter	Units	Conditions	Result
THD 1 kHz	%	Unity Gain; 0 dBv Input	*
THD 1 kHz	%	Unity Gain; +10 dBv In	*
THD 1 kHz	%	Unity Gain; +20 dBv In	0.0032
THD 1 kHz	%	-20 dB Gain; +10 dBv In	*
THD 1 kHz	%	-20 dB Gain; +15 dBv In	0.003
THD 1 kHz	%	-20 dB Gain; +20 dBv In	0.0045
THD 1 kHz	%	-20 dB Gain; +26 dBv In	0.011
THD 1 kHz	%	+20 dB Gain; -10 dBv In	*
THD 1 kHz	%	+20 dB Gain; -5 dBv In	0.004
THD 1 kHz	%	+20 dB Gain; 0 dBv In	0.006
SMPTE IMD	%	Unity Gain; 0 dBv Input	*
SMPTE IMD	%	Unity Gain; +10 dBv In	0.002
SMPTE IMD	%	Unity Gain; +15 dBv In	0.0032
SMPTE IMD	%	Unity Gain; +20 dBv In	0.009
SMPTE IMD	%	-20 dB Gain; +10 dBv In	0.002
SMPTE IMD	%	-20 dB Gain; +15 dBv In	0.0075
SMPTE IMD	%	-20 dB Gain; +20 dBv In	0.013
SMPTE IMD	%	-20 dB Gain; +26 dBv In	0.031
SMPTE IMD	%	+20 dB Gain; -10 dBv In	0.003
SMPTE IMD	%	+20 dB Gain; -5 dBv In	0.007
SMPTE IMD	%	+20 dB Gain; 0 dBv In	0.014
Static Noise	dBv	20-20k Hz; Unity Gain; No Input	-87
Static Noise	dBv	20-20k Hz; +20 dB Gain; No Input	-78
Static Noise	dBv	20-20k Hz; +30 dB Gain; No Input	-73
Static Noise	dBv	20-20k Hz; -20 dB Gain; No Input	-96
Dynamic Noise	dBv	20-20k Hz; Unity Gain; DC Input**	-84
Modulation Noise	dB	Noise Increase With Signal**	+3
Control Feedthu	mV	-30 dB to +30 dB Gain Sweep; 60 Hz	15[1]
Control Feedthru	mV	As Above With Applied Heat***	190[2]
Thermal Change	%	Unity Gain SMPTE IMD With Heat***	*
Crossover Dist'n	?	Objectionable?	None

Notes:

*Below Test Equipment Residual.

**A DC input current equivalent to a +20 dBv peak AC input signal is applied to facilitate noise reading during signal conditions.

***Thermal gradients are induced by applying the output of a 1,500-watt heat gun from a distance of 18 inches for a duration of 2 minutes. Device is then allowed to cool. Worst readings obtained during this process are recorded.

1. 6 millivolts for EGC 101A and TA101.
2. 18 millivolts for EGC 101A and TA101.

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distortion changes (in Class A/B designs), or as temperature-induced control rejection drift (in Class A designs). In either case, a good deal of thermal management is important to successful implementation, and the user needs to be aware of what to expect from a device in various thermal environments.

Valley People EGC 101, EGC 101A and TA 101

Around the turn of this decade, Valley People introduced the EGC 101 gain cell, described as being fully capable of meeting the criteria outlined above. From a technical standpoint, the EGC 101 is described as a Symmetrically Balanced and Cross Balanced Class A Log/Antilog Multiplier.

In the form of the original EGC 101 gain cell, the device had one parameter that required close attention on the part of the user — namely the control rejection when it was subjected to strong thermal gradients. Newer versions of the device are now available in which the temperature sensitivity of the control rejection term has been reduced by a factor of around 10. These versions are known as the EGC 101A (direct replacement for EGC 101), and the TA 101 (minimum package with all transistor connections accessible).

Fifteen pages of documentation and specifications accompany these devices and their employment in various forms as VCA circuits. The most commonly recommended general professional use implementation is designated the EGC 205M.

Test results: As further verification of the expected in-use performance of the Valley People EGC/TA Series VCA circuits, the following tests were performed at the time of writing this article. The data presented is the average from four production units tested at this time, and is typical of the performance to be expected from non-selected production units.

In preparing this article, the author's preference was to conduct side by side measurements on the two other VCA devices mentioned later, and to publish the results of all three devices together for comparative analysis. In fact, the author has performed such specific comparative testing. However, the ethics of professional journalism preclude the publication of such comparative test results, since the tests were conducted by a biased party (the author).

It is the author's opinion that the methods of testing, and the parameters tested for, form an excellent basis for the potential user's evaluation of VCA devices, and the potential VCA user is urged to conduct for himself similar tests on those VCA devices that he might contemplate employing in studio equipment.

Table 1 lists the author's tested parameters and measurement results obtained for the Valley People EGC 101 VCA, while Figures 1 thru 4 show grapns of various test results.

Based on the information given in the manufacturer's literature, and verified in the author's test results, it should be clear what the device capabilities are in actual-use situations. It should also be clear that the device performs well below the established criteria in terms of noise and distortion, while providing 26 dB of input headroom above a nominal O dBv input signal level.

Unless the author has lied or submitted false test data, the user should be able to form an accurate and confident opinion regarding the suitability of the product for its intended purpose.

Other VCA Products

The remainder of this article will be dedicated to the author's analysis of two other VCA devices that recently have been itroduced by other manufacturers, and which are being promoted for sale to OEM and other users. In reading the published literature for these products, the

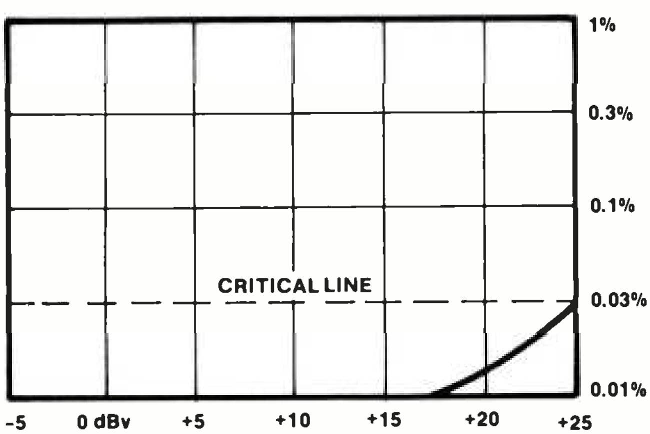

Figure 1: SMPTE IMD versus Input Level for EGC 101 (re: 0.775 VRMS, -20 db gain).

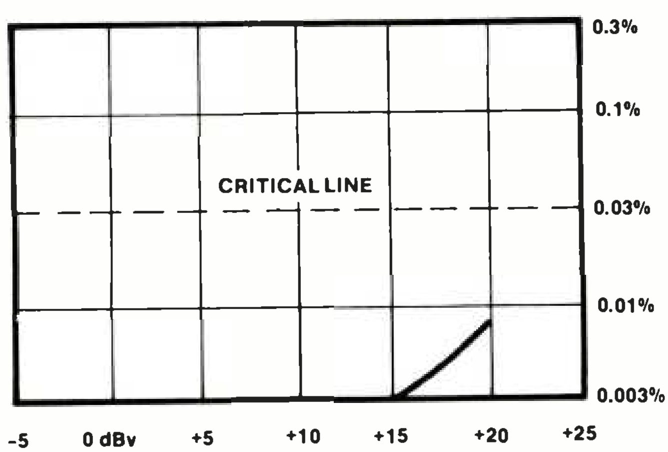

Figure 2: SMPTE IMD versus Input Level for EGC 101 (re: 0.775 VRMS, 0 db gain).

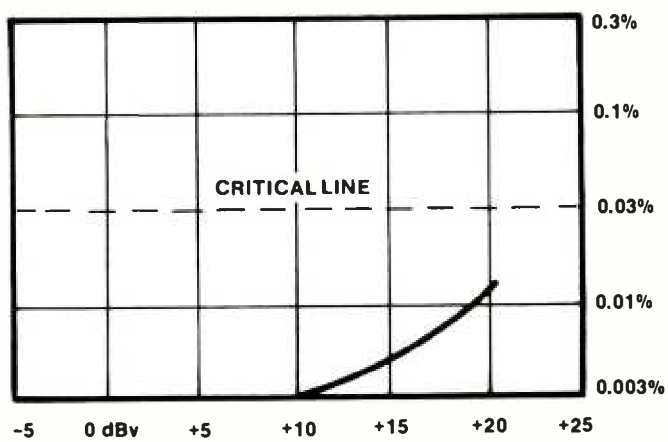

Figure 3: SMPTE IMD versus Input Level for EGC 101 (re: 0.775 VRMS, +20 db gain).

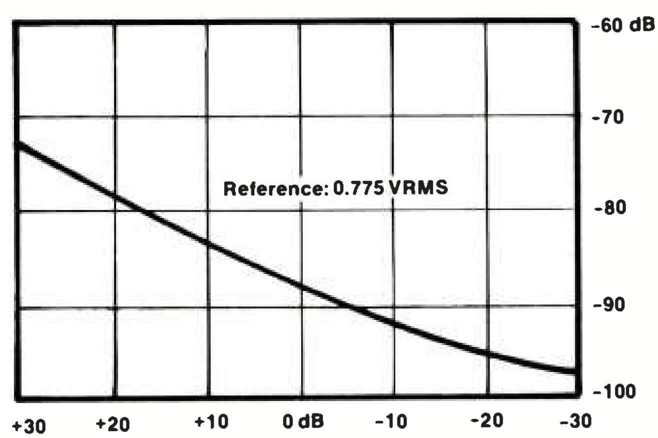

Figure 4: No-signal Output Noise (20 Hz to 20 kHz) versus VCA Gain or Loss.

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author's impression of what is being said is, essentially, that these two devices offer more or less the same performance as does the Valley People product, and that they can be used in similar applications with similar results.

Again, in the interest of literary ethics, the author cannot make statements regarding how these devices will actually perform in use, but must limit his comments to his personal opinion of certain poetic license he feels has been taken in the presentation of device specifications. It should be noted that the manufacturers' names and device model numbers are not listed, since the purpose of this article is *not* to suggest the superiority of one device over another but, rather to educate the potential user of the importance of comprehending the underlying meanings of what is specified, as it pertains to actual use. The opinions expressed are solely those of the author, and an endorsement of these opinions by the magazine or its publisher or editorial staff is neither implied nor expressed.

Excerpts From Published Data for Device "A"

Description: Device "A" . . . is a low cost, high performance linear antilog voltage controlled amplifier with full Class A performance. The device has a 100 dB signal-to-noise figure at 0.01% THD. The current inputs and outputs make possible wide bandwidth, easy signal summing, and minimum external component count. Inherently low control feedthrough and second harmonic distortion make trimming unnecessary for most applications. In addition, [Device "A"] has more than 12 dB of headroom at the rated specifications, and can be configured to give up to 40 dB of gain . . .

Features:
- 100 dB signal/noise (20 Hz to 20 kHz)
- 0.01% THD
- 0.025% IMD
- 12 dB of headroom (at rated specs)
- 40 dB gain capability

Specifications for Device "A" are listed in Table 2.

TABLE 2: PUBLISHED SPECIFICATIONS FOR DEVICE "A"

Parameter	Min	Typ	Max	Units	Conditions
Signal-to-Noise (20Hz-20kHz)		-98		dB	Ve Gnd, I1=100ua
THD (Untrimmed)		0.025		%	" "
THD (Trimmed)		0.01		%	" "
IMD (Untrimmed)		0.075		%	" "
IMD (Trimmed)		0.03		%	" "
Current Output Offest (Untrimmed)	-5	0	+5	microA	" "

Excerpts From Text: . . . A peak input signal of ±200 microamps will give the rated specifications and allow for 12 dB of headroom. The device can handle upto ±800 microamps without clipping or other radical increases in distortion. If the average peak input signal is 4V peak-to-peak, a 10K input resistor will yield the proper input current . . .

Figures 5 and 6 show published output noise and IMD total harmonic distortion measurements for Device "A", versus current gain attenuation.

Author's Analysis of Device "A"

The author believes this device to be a derivation of the Class A Transconductance form of multiplier. As such, performance above unity gain needs to be scrutinized, particularly from the standpoint of control rejection and distortion. An indication of rising distortion above unity gain is indicated on the graphs (Figures 5 and 6). It is noted that in the manufacturer's specifications no meaningful information is given as to what the control rejection is versus various operating gains. It is only stated that the DC output offset current is ±5 microamps at "V_c = Gnd, I_1 = 100 microamp." (After careful scrutiny of the data, the author has determined that "V_c = Gnd, I_1 = 100 microamp" indicates device unity gain.)

It is noted that all distortion specifications are "typical," and are stated without definition of the signal levels, and are stated at 0 dB gain only. Thus, in effect, these specifications mean literally nothing in defining the actual-use distortion. It is easy to achieve 0.01% THD at low signal level unity gain in any VCA design, but it is an entirely different matter to achieve low distortion levels at higher signals and varying gains. The actual-use distortion of a device could be 1% or higher, and still meet the type of specifications offered here.

With respect to noise performance, confusing information is presented. Under the "Features," a 100 dB signal-to-noise ratio is indicated, with an additional 12 dB of headroom. However, on the noise graph (Figure 5), a unity gain noise of -98 dB is referenced to "300 microamps RMS current." In studying the text, it is seen that 300 microamps RMS is 5.5 dB below the rated input clipping point, *not* 12 dB. If we were to assume the graph to be correct, and configure the device for 25 dB input headroom, as done with the EGC 101, we would find the true nominal

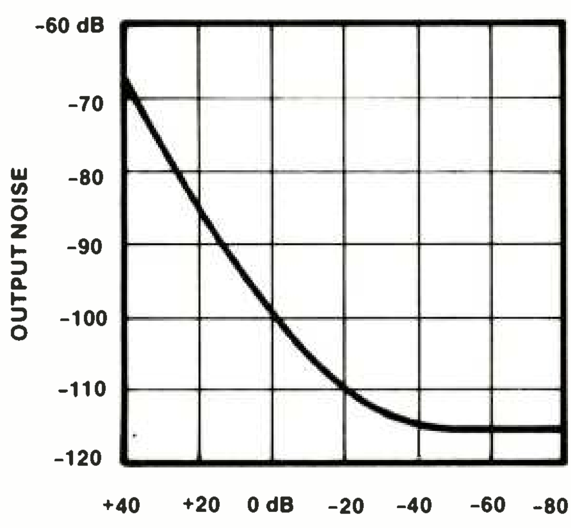

Figure 5: Published Current Gain/Attenuation Output Noise (bandwidth 20 Hz to 20 kHz) for Device "A" (re: 300 microamp RMS output).

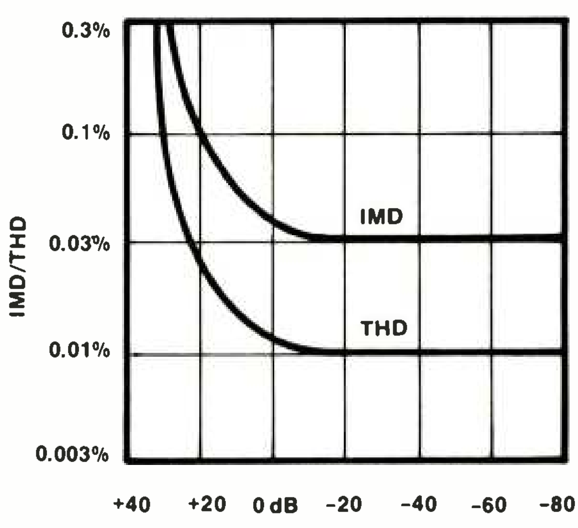

Figure 6: Published Current Gain/Attenuation Distortion (bandwidth 20 Hz to 20 kHz) for Device "A."


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18-90 mm

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SNR ratio to be around 79 dB.

While the "Features" indicate a gain capability of 40 dB forward gain, the text admits that it is necessary to add 20 dB or more of post gain to achieve this in a practical circuit. As described previously, the cost of doing this is to dramatically degrade the network noise, shutoff, and control rejection parameters.

With respect to high-frequency distortion, crossover distortion, and modulation noise, this author does not expect to find serious anomolies in these parameters in this Class A sort of design. Still, no information is given in this respect.

Athough this author has done a complete evaluation of this device, literary ethics preclude the publication of his findings. Suffice it to say that the preformance of this device could lie anywhere between great and absolutely pitiful, and still fall within the realm of meeting the published specs. One can only find out by buying some, and testing the device.

Excerpts From Published Data for Device "B"

Description: Device "B" . . . intergrated circuit voltage controlled aplifiers are high performance current-in/current-out devices with dual polarity, voltage-sensitive control ports. They require little external support circuitry, and are housed in a plastic 8-pin SIP package, thereby affording unusually high PCB packing density. Combining a high gain-bandwidth product with low noise, low distortion, and low input bias current, these devices offer similar performance to discrete or modular VCAs with the economy of ICs.

Features:
- Extremely wide gain/attenuation control
- Low distortion
- Low noise
- 20 Hz to 20 kHz response even at high gains

Specifications for Device "B" are listed in Table 3.

Figures 7, 8 and 9 show published output noise, output offset and THD measurements for Device "B."

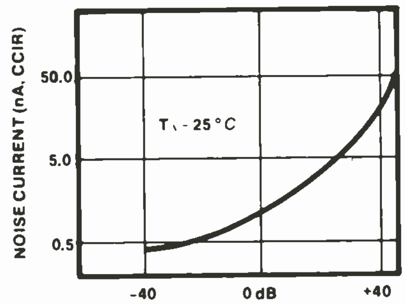


Figure 7: Published Output Noise Current versus Gain for Device "B."

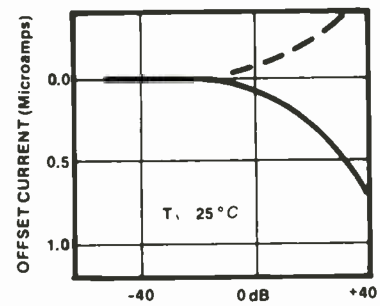


Figure 8: Published Output Offset Current versus Gain for Device "B."

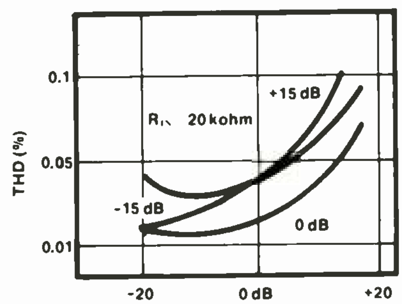


Figure 9: Published THD versus Input Level for Device "B."

Author's Analysis of Device "B"

This VCA offers considerably better documentation than does Device "A," but still requires the reader to be able to comprehend it if he is to discover the real performance capability.

Device "B" is known to be a Class A/B log/antilog multiplier, packaged in monolithic form. As such, there are certain characteristic traits that the author would expect of the device. On the plus side, the author would expect the VCA to yield a wide and accurate range of gain control, covering both attenuation and forward gain. He would expect good thermal stability, due to the monolithic design, and extremely low quiescent (no signal) noise levels because of the Class A/B bias structure. On the negative side, however, the author would expect to find a dramatic increase in noise when signal is applied (due both to the Class A/B structure and the small geometry monolithic design), as well as significant amounts of distortion (specially SMPTE IMD) at the higher signal levels, and at gains other than unity. (This is expected because the Class A/B structure does not provide the inherent distortion cancellation effects found in Class A designs. Also, the small geometry needed for monolithic fabrication invites higher amounts of error at the higher input signals).

The specs tell us quite a bit, but must be deciphered. As far as noise, even though it is stated with "CCIR Weighting," instead of the usual 20 kHz flat bandwidth, and is stated in "dBV" instead of the more usual dBv re: 0.775V RMS, the spec is not misleading. Noise in the -90 dBV region is to be expected from this structure, *without input signal.* However,

TABLE 3: PARTIAL PUBLISHED SPECIFICATIONS FOR DEVICE "B"

Parameter	Min	Typ	Max	Unit	Conditions
Gain Linearity		±1	±2	%	-60 to +40 dB
Output Noise (CCIR Weighting)		-95	-90	dBV	0dB gain, Rout=20K
Output Offset Voltage		±1	±3	mV	0dB gain, Rout=20K
		±2	±3	mV	15 dB gain, Rout=20K
		±10	±15	mV	40 dB gain, Rout=20K
Intermodulation Distortion*		0.01	0.02	%	+15 dB gain
Total Harmonic Distortion					Itotal=175 microamp/1 kHz
		0.01	0.02	%	0dB gain
		0.035	0.045	%	±15 dB gain

*Measured with 10 kHz and 12 kHz mixed 1:1 expressed as (2 kHz product/sum of 10 kHz and 12 kHz) × 100%.

this quiescent noise level tends to belie the noise that will be present during signal. From experience with this class of device, the author would expect to see a noise increase of 25 to 30 dB when high-level input signals are applied, thus yielding a dynamic noise floor in the -60 to -65 dBV region. (In order to clarify any reader confusion, the term "dBV," using a capital "V," refers to an RMS voltage of 1 volt, while the term "dBv," using a lower-case "v," indicates a reference voltage must be stated, as in the author's preferred usage "dBv re: 0.775V RMS." For further clarification,the term "dBv re: 0.775V RMS" is the correct statement for what many people mean when they state "dBm" erroneously as a statement of voltage rather than power.)

The output offset voltage spec (control rejection) is clear, and indicates a maximum DC level shift of 15 millivolt for gain changes to +40 dB, but does not mention attenuation, where it is likely to be no problem.

Getting down to the IMD spec, it is this writer's opinion that this one was written by a specsman. Firstly, the IMD spec is clearly *not* a SMPTE IMD spec, rather it is a twin-tone IMD spec. While the twin-tone spec does indicate potential high-frequency performance, and is a valid and useful test, it does not have the extreme sensitivity of the SMPTE method, and thus does not indicate this common form of high distortion. All of this is really academic, since the IMD spec given shows no *reference* signal level. Thus, it gives the user no information whatsoever regarding real world IMD, and serves as a pacifier.

As for the THD specs, the user must go through a number of calculations to determine what operating level is indicated by the stated "175 microamp I_{total}" signal current. The author has performed these calculations, and has determined the input level for the unity gain spec to be 17 dB below maximum input, while the ±15 dB gain spec appears to be taken at a level 11 dB below maximum input. Only those very versed in VCA design are apt to be able to make these calculations based on the information presented. If this device behaves at all like other VCAs of similar design, the author would expect to see measured SMPTE IMD figures of three to four times the magnitude of measured THD figures. Thus, the SMPTE IMD could be suspected to be in the region of 0.10 to 0.18% at the calculated specification point 11 dB below maximum input, ±15 dB gain. It would also be suspected to double for each 6 dB increase in signal level, thus possibly reaching the region of 0.3 to 0.6% near the upper range of input signals. This is supported when one looks at the THD versus Input Level graph (Figure 9), and converts it to a SMPTE IMD graph by multiplying all readings by between 3 and 4.

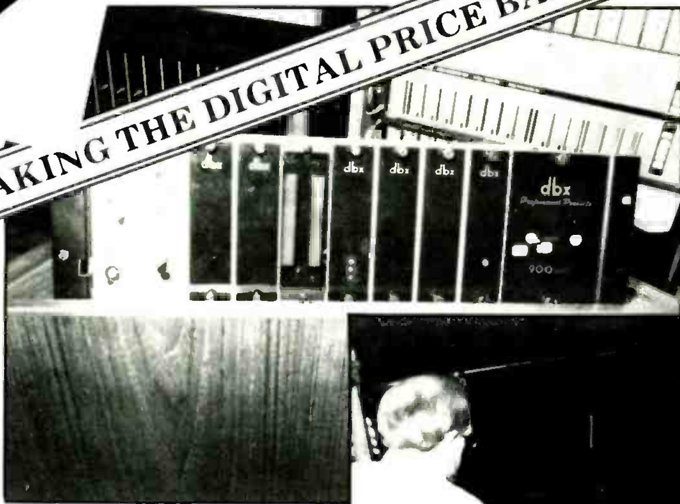
Finally, headroom must be considered. If the desired headroom criteria of 24 dB were to be established using this device, it would be necessary to reduce the operating signal levels by about 6 dB. This would effectively reduce the nominal SNR ratio by an equal amount, to the range of 84 to 89 dB, under no-signal conditions. ■■■

Conclusions

The information given here should not be taken as statements of what sort of performance has been observed with respect to the two VCAs analyzed. It is recommended that the reader make such observations for himself, based on intelligent and educated test and evaluation methods. The information presented here is intended to show what sort of performance could possibly be measured from these devices, while still falling within the realm of what is specified by the manufacturers.

It is the author's personal opinion that the specs make it extremely difficult, if not impossible, for the average user to calculate or otherwise assess the true performance of either device under actual-use conditions. Perhaps this is intentional; perhaps it's due only to hasty spec writing. In either case, the only way the user can accurately assess the suitability of the product for his application is to go and by some, then spend several hours (or days) at the test bench to find out for themselves.

BREAKING THE DIGITAL PRICE BARRIER



Prototype Model 700

Trials at Crescendo

The dbx Model 700 Digital Audio Processor

Design Parameters and Systems Implementation

by Robert W. Adams
Senior Project Engineer, dbx, Inc.

Just about everyone who has heard an original digital recording has been impressed; most are enthusiastic. The virtual absence of distortion, noise, and wow/flutter makes the sound far superior to that of analog. But, because of the tremendous cost involved, owning a digital recorder is not exactly commonplace in the world of professional audio. For the "semi-pro" studio and serious recording musician, owning a professional quality digital recorder is an impossible dream. Although the cost of these digital machines will fall somewhat over the years, their complexity will make them more expensive than analog recorders for some time to come.

Apart from the expense of their respective recording equipment, there is a gulf separating the digital and analog engineer. The former lives in a world of numbers, bits and bytes, while the latter is more at ease with the use of one op-amp and a handful of resistors and capacitors in a tone-control circuit, than with incorporating 30 digital ICs.

Since dbx has considerable experience in analog audio R&D, it was decided to take advantage of the best of both worlds. By combining analog techniques with available digital technology that had never before been applied to the recording process, digital sound could be made affordable to every studio. The result is the dbx Model 700 Digital Audio Processor.

Evolution of the Model 700

The first goal in the design process was to find a form of analog-to-digital and D/A conversion that was both high-quality, and inexpensive. Ruled out because of their high cost were 16-bit linear PCM converters; 14-bit processors were somewhat cheaper, but didn't have the dynamic range needed

... continued overleaf —

Operational Assessment at Crescendo Recorders, Atlanta

by William Ray
Crescendo President

Ever since the invention of the audio tape recorder, deficiencies of magnetic tape as the storage medium have been a major stumbling block in the recording industry's never-ending quest to perfectly reproduce an audio signal. To be a little more specific, tape hiss, print through, limited dynamic range, high-frequency dropouts, head bumps and other frequency response related problems in the past, have seemed like insurmountable problems. Over the years, however, one by one these problems more or less have been dealt with.

Given that ours is an industry staffed by creative engineering types, it's hardly a surprise that so many products and ideas have materialized to deal with the limitations of the magnetic tape medium. One of these "creative engineers" whose work has enabled us to scale the insurmountable "Mount Tape Hiss" located at the beginning of the "limited dynamic range" of mountains is David Blackmer, president of dbx, Inc. Blackmer's development of the voltage-controlled amplifier forms the heart of dbx noise reduction, which provides the reduced tape hiss and enhanced dynamic range we've been looking for (on paper anyway). But, it's still a Band-Aid solution; most of the problems associated with the tape and the format are still there.

The Digital Answer?

Now, this brings us to "Digital Audio." Surprised? I was. You may ask, as I did, what does dbx have to do with digital audio? Read on.

Crescendo Recorders (formerly the Sound Pit) is a multi-24-track recording complex with a complete in-house video post-

... continued on page 157 —

For additional information circle #104

for professional use. Adaptive Delta Modulation (ADM) was attractive because of its cost, but after critical listening to material of very wide dynamic range, it was felt that the overall sound was not good enough for digital-recording applications.

After months of study and deliberation, a system was conceived and devised that offered several improvements in audio performance over ADM. This system was dubbed "Companded Linear-Predictive Delta Modulation," and will be described in detail shortly. The results of listening tests over this system were very encouraging, and convinced us that we had found a low-cost alternative to 16-bit PCM for professional digital recording.

Next we had to choose the storage medium. As is well known, the bandwidth requirements of digital recording are much higher than can be accommodated on an analog tape recorder. The design of a special set of tape heads to be used on a conventional transport was considered, but we decided that this would be too expensive, and take too long to implement. Finally we settled on videocassette recorders, which have adequate bandwidth, are readily available in several formats, and are produced in sufficient quantity to be comparatively inexpensive.

After these decisions, the first prototype was built. Initially no error correction was used because we found that our method of A/D conversion was fairly insensitive to bit errors. In fact, during normal program material, errors of up to 50 bits frequently were inaudible. But we also found that the largest of the dropouts on video tape would indeed cause clicks to be heard during low-level passages. Thus the next prototype was built with full digital error correction. Although this additional circuitry increased the cost, the unit could still be priced far below competing 16-bit PCM systems. This second prototype was used to record a wide variety of instruments and musical materials, both in studios and in concerts. It passed all tests with flying colors.

A/D Conversion: Companded Predictive Delta Modulation

Delta Modulation has been used for years as a low-cost means of A/D conversion. In this digital process, the numbers derived in the A/D represent *differences* between sampled voltages, rather than the instantaneous voltage produced in a "conventional" PCM audio processor. ("Delta" is the mathematical term for change or difference.)

Because it is based on changes in level, rather than absolute values, the dynamic range of Delta Modulation is restricted at the loud-end by slew-rate limitations — the signal slope becomes too steep for the A/D to track — and at the soft-end by the familiar quantization noise inherent in all digital recording systems. At high frequencies the dynamic range is especially limited, but even at lower frequencies it is not sufficient for serious audio applications. To extend Delta Modulation's dynamic range, Adaptive Delta Modulation (ADM) adjusts the step size to suit the dynamics of the input signal.

The analog-to-digital conversion process in the Model 700 differs from that used in normal ADM in two important respects. First, rather than vary the step size to follow the signal, in the dbx converter the signal is varied with a voltage-controlled amplifier to avoid overloading the fixed Delta Modulator. Second, to lower the quantization noise, the fixed Delta Modulator uses a "linear-prediction filter," which relies on the history of the audio signal to predict its future. These two differences between AMD and CPDM result in substantial performance improvements. To demonstrate, we have to go into detail. First, let's look at the high-precision compander (compressor-expander) used in the Model 700:

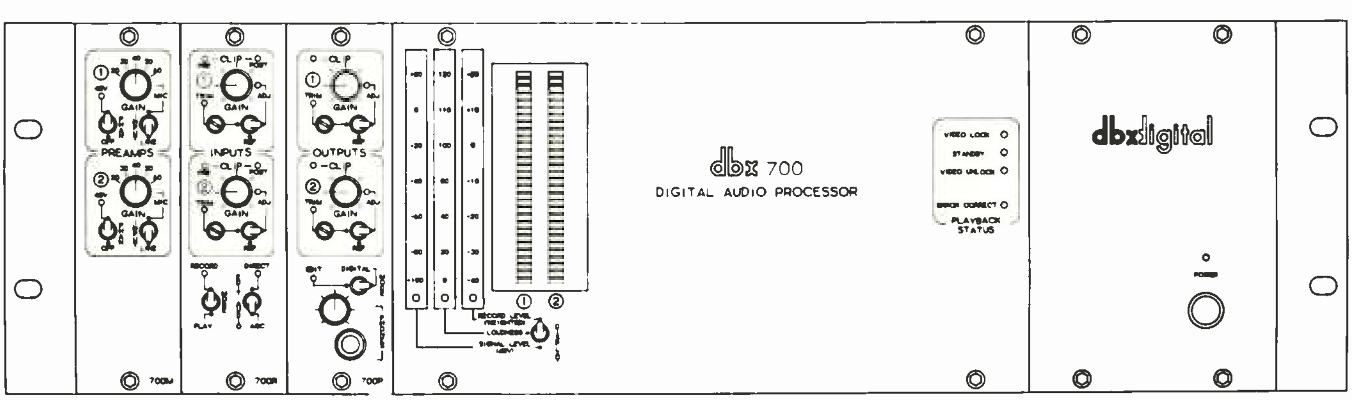
• *Companding versus Adaptive.* In ADM, step size is varied according to the average slew rate (speed of change of the input signal). A burst of high-frequency, high-level input signal requires a large step size, so that slew-rate limiting can be avoided. The problem with doing this, however, is that the range of practical adjustment of step size is limited to around 500:1, and at the smallest step sizes the comparator may not operate ideally, or even close to it. Also, the lack of dither noise can result in the noise floor being non-white (equal intensity for all frequencies), and signal-dependent.

The dbx system overcomes these problems by using a VCA in front of a fixed, non-adaptive Delta Modulator (Figure 1). When a large signal with a high slew rate is present, VCA gain is reduced, which lowers the slew rate of the signal passed on to the Delta Modulator. Thus, the input is adapted to the fixed step size of the Delta Modulator, rather than *vice-versa*. In playback, signals are decoded complementarily: the output of the fixed Delta Modulator is applied to a VCA whose gain is the exact inverse of the encoder's VCA gain.

The range of gain available from the VCA is beyond 120 dB, or voltage ratios of more than a million-to-one, which is a great improvement on the range available from ADM. Furthermore, using the fixed-step-size Delta Modulator lets the comparator have enough signal to operate properly, which also increases the available dynamic range. Finally, dither noise can now be added at the input to the fixed Delta Modulator, to eliminate any noise-floor anomalies ("birdies" and other such tonal effects) that are possible with ADM.

The signal that controls the gain of the VCA comes from a sophisticated level-sensing circuit that uses information present in the Delta Modulator's digital output. Being quite complex, this circuit cannot be fully explained in the space available here. Suffice to say that the VCA gain now can change very quickly to follow musical transients, but will change slowly for material that has slower dynamics.

It should be noted again that this level-sensing circuit obtains its information directly from the bit stream in both encode (record) and decode (play). Since these bit streams are identical in each case, mistracking (non-complementary VCA gains) *cannot*



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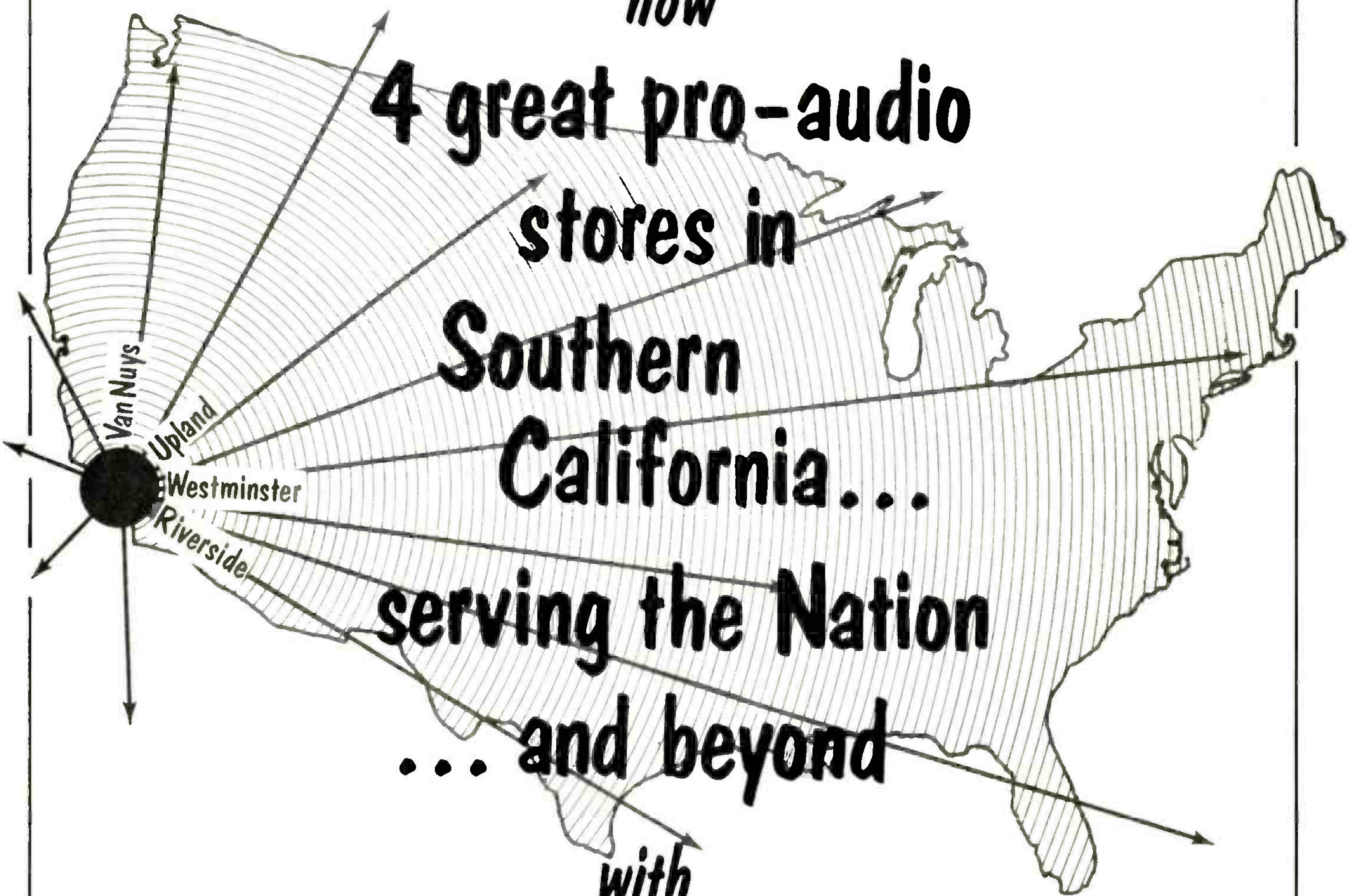

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occur.

• *Linear Prediction.* One of the problems affecting both ADM and companded DM systems is that the noise floor can change with signal level. This occurs because the step size is changing to follow the input, and step size is what determines the level of quantization noise. Generally, if the changing noise floor is far enough below the signal, its modulations are inaudible. Linear prediction is a method of increasing the dynamic range of a fixed Delta Modulator by more than 10 dB, and this increase is sufficient to eliminate any possibility of hearing noise modulation.

By way of illustration, let us assume a situation where the Delta Modulator has a fixed step size of 10 millivolts. Therefore, if its last "guess" at the input level was too high, the next will be 10 mV lower. Now, let us assume that of the last 10 guesses about signal voltages seven were too low, and three were too high. We might reasonably infer that the signal level was increasing. We could then shift the step sizes from ±10 to, say, +15, -5 millivolt, which is in line with our expectation (based on the recent history of the signal's behavior) that the signal is more likely to change in a positive than a negative direction. Note that doing this does not change or lower quantization noise: the difference is still 20 millivolts between +10 and -10, and +15 and -5 mV. But it *does* increase the maximum slope (steepness, or slewing, or speed of change) that the modulator can follow without slew-rate limiting. Hence dynamic range is increased, as well.

In practice, this alteration in the balance of "plus" and "minus" step sizes is achieved by a "linear-prediction filter." This filter is substituted for the simpler filter (integrator) normally found in a Delta Modulator, and is designed for maximum dynamic range. A comparison between linear-PCM converters and the dbx Model 700 system is provided in Table 1.

Memory

The dbx 700's memory has 16k bits of random-access memory storage for wow/flutter absorption, data interleaving and de-interleaving, and video requirements. During recordings, the A/D converter produces a steady stream of bits. The video format, however, has several areas where data can't be recorded (described below), so the memory is asked to store the data bits during these times. Of the 16k memory, 8k is for data interleaving (time scrambling), and 4k for storing data during the video-sync intervals.

During playback, the memory must supply the D/A converter with a steady stream of data while receiving the data from the VCR. But the VCR introduces wow and flutter, which makes the bit rate sent to memory variable. The memory absorbs these variations with the last 4k bits of storage; this results in a very low flutter in the decoded signal (less than 0.01%).

Error Correction

The dbx A/D conversion method, unlike linear PCM, is inherently tolerant of errors. In linear PCM, single-bit error may cause the most significant bit (MSB) to be in error. This MSB error might produce a disastrous full-scale spike in the audio output.

In the dbx Model 700, there's no such thing as an MSB; all the bits have a value just large enough to keep up with the signal's sample differences. For this reason, errors of 30 bits or less are usually inaudible during normal program material played over the dbx system. Professional U-Matic VCRs typically have very low bit-error rates, due to the high quality of the tape used, and the greater head-to-tape velocity; dropouts greater than 300 bits are quite rare. Consumer VCR formats often have longer dropouts, up to about 600 bits.

While it is recommended that the dbx 700 be used with a U-Matic-type machine, Beta and VHS units, being less expensive and offering longer recording time, may be used in situations where economizing is called for.

The dbx error-correction circuitry works by adding one extra parity bit for every three data bits. The parity bits are mathematically derived from the data bits, so that any bit errors on playback will produce a unique error pattern in the received parity bits. This error pattern is decoded to find exactly which bits are in error, and the offending bits then corrected. This correction circuit works in conjunction with the memory interleaving in such a way that a long burst error is presented to it as a series of short errors separated by good data.

Video-Format Encoder and Decoder

The format generator, or encoder, produces all the necessary synchronization, blanking, and equalizing pulse signals required to make the digitized audio signal look like the standard NTSC video signal, and thus acceptable to the VCR. It also controls the memory so that data bits are recorded only in the allowable video intervals.

The dbx video format records 128 bits per horizontal scan line and uses 224 lines per video field (out of a possible 262.5, the NTSC standard). The remaining lines are left blank to allow for the video-synch interval, for the special timecodes used for editing, and for the synchronization of several VCRs.

The decoder extracts the data from the video waveform on playback, and writes them into the memory. To do this, it must separate out the synch and data information, and decide which horizontal lines contain valid data. Unusually extensive protection is employed so that VCR noise and tape dropouts, which can easily look like valid synchronization signals, don't fool the processor.

Analog Display and Control Functions

Extensive metering facilities provide information about both the dynamic range and level of the input signal. The display is a column readout with 30 LEDs for each of the two channels. A peak hold with slow decay is also incorporated. The display can serve three selectable functions:

a) *Record-Level Indicator.* This has a range of 60 dB (2 dB per LED) and is pre-emphasized to follow the headroom characteristics of the A/D converter. Brief transients that exceed the maximum record level (+20 dB) will not clip because of the transient-speedup circuit in the level detector. Continuous operation above the maximum indicated

. . . *continued overleaf* —

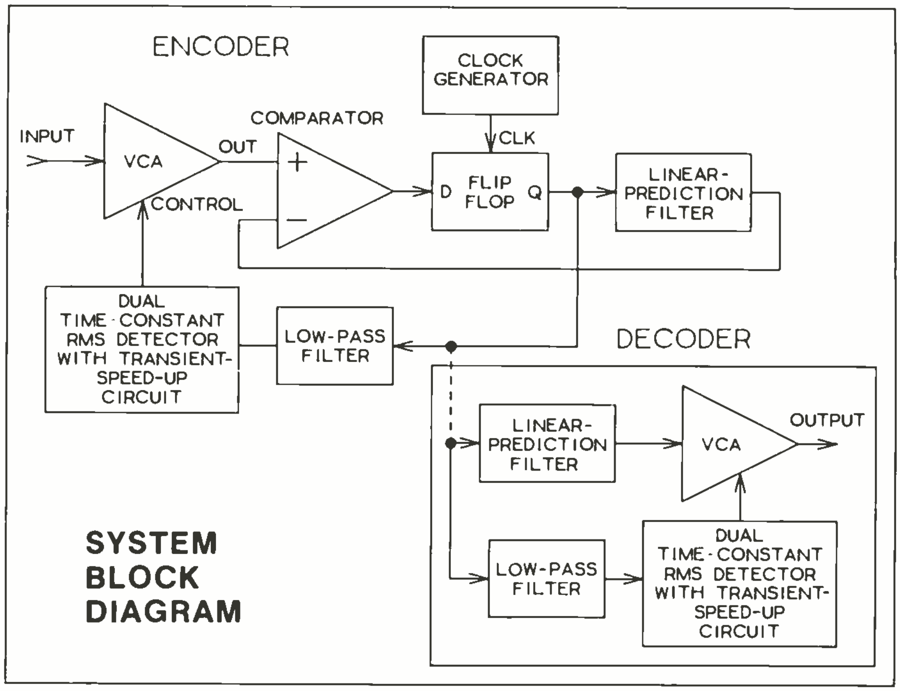
TABLE 1: A Comparison between Linear-PCM Converters and the dbx Companded Predictive Delta Modulation

	16-bit Linear PCM	dbx System
Cost	Very High	Low
Dynamic Range	90 dB	More than 110 dB
Sensitivity To Bit Errors	High	Low
Bit Rate	Approximately 770k per second, plus error-correction overhead	Approximately 700k per second, plus error-correction overhead
Anti-Aliasing Filters	Complex, hard to build, large phase shifts	Simple; small phase shifts
Maximum Level Flat With Frequency?	Yes	No
Distortion	Low	Low
Frequency Response	Depends on anti-aliasing filters; usually very good	Very good

record level is not recommended, however.

b) *Signal-Level Meter.* This is an RMS-responding, non-weighted indicator with a total range of 120 dB (+20 dBV to -100 dBV). In record mode, this meter reads the line input or output of the mike pre-amp, if one is used. In play mode, it reads the unit's line output level.

c) *Loudness Meter.* This incorporates complex dynamic-filtering circuitry that simulates the equal-loudness contours of the ear. It follows the Stevens curves (the modern version of the old Fletcher-Munson curves) to within 2 dB over the entire 120 dB range of the meter. This feature is invaluable in making dynamic-range measurements, where a "flat" meter often will give too high a reading because of low-frequency room noise. The inputs to the loudness meter are switched in the same manner as the signal-level meter. Sensitivity of the microphone is used to set the reference level.



SYSTEM BLOCK DIAGRAM

The analog-input circuitry contains all necessary control functions. A three-position switch selects a front-panel level-control pot; a trim pot adjustable through a hole in the front extrusion; or an internal non-adjustable reference level. This last position is provided so that the unit can become a unity gain device from record to play, which makes it easy to play back a recording at the same sound-pressure level as the original, if the sensitivity of the microphone is known.

Clipping LEDs are provided both before and after the level-control stage. In a device with such a large dynamic range, the gain structure is quite important; if the front-panel LED is set too low, for example, dynamic range may be lost.

The analog-input section also provides a signal for recording on the VCR's audio tracks for use during editing. This is necessary because the digital audio information cannot be recovered when a VCR is put into slow-motion to search for an edit point. A 2:1 compressor may be switched in so that wide-range material can be successfully captured on the VCR audio tracks.

The analog-output section contains two output buffers capable of driving 600-ohm loads to +24 dBm; a stereo headphone driver; clip LEDs; and another three-position switch to select among front-panel pots, screwdriver-accessible trim pots, and an internal non-adjustable reference calibration. Low-noise circuitry is used throughout, and all electrolytic capacitors in the signal path are paralleled with smaller, non-electrolytic caps for good audio quality. Electronically balanced outputs are standard, and may be defeated if unbalanced ones are desired.

An optional low-noise mike pre-amp module can be plugged into the last slot in the frame. Each channel has controls for gain (20, 30, 40, 50, and 60 dB), 48 volt phantom powering (on/off), and Line/Mic source select. Our low-noise circuitry adds less than 1 dB of noise for microphone impedances from 100 to 1k ohms.

Construction

The dbx 700 is completely modular, all circuitry being contained on 10 printed circuit boards that plug into a backplane. The complete power supply, including transformer and AC input, is also modular and plugs into the backplane. High-quality XLR connectors are used on the rear panel for all audio input and output connections, and BNCs for connections to and from the VCR.

* * *

It was through the marriage of analog and digital design that dbx hoped to spread the benefits of digital sound among those to whom they might otherwise have been delayed. For the first time, a studio owner can purchase a digital recording system — the dbx 700 Digital Audio Processor and a professional-quality VCR — at a price comparable to that of a good two-track analog recorder. This feat was accomplished by innovative circuit design, which is most apparent in our A/D converter, wherein a unique combination of analog and digital technology provides extremely high performance at remarkably low cost.

The main goal in designing the dbx 700 was to lower the cost of digital processing sharply in order to bring digital-recording capability to every engineer and studio that could afford a top-quality analog recorder, not to mention the associated processing equipment. The low (under $5,000) price tag meets this goal. We also believe that the dbx 700 sounds as good as, if not better than, the finest digital equipment currently available from the major manufacturers. Delivery is targeted for next summer. ■■■

AN OPERATIONAL ASSESSMENT *by William Ray*

— continued from page 150 . . .

production facility. Our clientele is diversified, and has included such acts as Ted Nugent, Cheap Trick, Lynyrd Skynyrd, and Kansas. As with most studios of our size, the ability to attract new clients — as well as keeping old ones — depends upon our ability to maintain a "state-of-the-art" facility. With microprocessors raining down on the public from every direction, "Digital Audio" has finally entrenched itself in our clients' vocabulary, and has become one of the more popular buzz words. The pressure had been mounting for us to commit to a digital mastering machine.

Normally, I would not question the validity of a particular piece of equipment that so many clients had requested. (Heaven help the studio that interferes with its clients' creative pursuits!) However, we had definitely been procrastinating on this purchase. My reasons for procrastination centered around several very fundamental concerns.

First, was format. Mother technology has been *very* generous in recent years, by bestowing upon our industry many technological breakthroughs. However, man's nature being what it is, we have managed to significantly slow the implementation of many of these technological breakthroughs by spending years on debating the best way to proceed. A classic example of this phenomenon is EQ and alignment for analog audio tapes. We've had to deal with NAB and IEC for years, and only recently was a "standard" agreed upon. There is still no recognized standard tones or levels on a master tape for set-up and alignment purposes. (I find it ironic that a resolution will finally be made at the same time that we discover a technology to make analog tape obsolete as a format.)

As a studio owner preparing to make a significant investment ($30,000+), I have to ask some very important questions concerning the establishment of a new recording format:

- What will the storage medium be? (The options now are audio tape, and video tape.)
- If the storage medium is video tape, will it be ¾-inch or ½-inch — Beta or VHS?
- If the standard is video tape, how will I edit? Will I be able to use a low-cost video editor?
- What will be the standard format for analog to digital conversion? PCM, ADM, or something entirely new?
- Assuming we can standardize on an A-to-D format, what will be the sampling rate, since this has a significant

OPERATIONAL ASSESSMENT AT CRESCENDO RECORDERS

effect on signal quality. If it is too low, it causes significant technical problems as it is increased (especially with PCM A-to-D).

- Will the cutting facilities I use be able to decode my digital masters?

Until some effort is made to answer these questions, purchasing a digital recorder is kind of a "pig in a poke." A $30,000 or more investment (over $60,000 for the Sony PCM-1610 processor with its editor) could be a complete loss in a year if a non-compatible standard were to be adopted. Despite digital's obvious audio attributes, the lack of economic security so far has kept us (and many others, I'm sure) from purchasing a machine.

My second concern is price. I've seen the price of all digital related technology come down drastically in recent years; studio-quality digital delay lines, for example, have gone from $3,000 to $499.00. I assumed (correctly) that digital audio recorders would follow suit.

Despite attempts to enlighten my clientele to these problems, requests for digital audio have continued relentlessly. We were losing the battle. I agreed to appropriate funds for a digital recorder, and began researching.

It looked as though the storage format would be ¾-inch video for two-track masters. A few phone calls proved that although *none* of the mastering facilities we used had digital replay machines, they did have access to a Sony PCM-1610 and video recorders on a rental basis of $500.00 per day — one-day minimum — when available. We were told it would cost us $29,500 for the Sony PCM-1610 processor. In addition, we would have to spend between $4,000 and $8,000 for a ¾-inch U-Matic VCR.

The dbx Alternative

Randy Fuchs, my partner and fellow owner of Crescendo Recorders, in a conversation with his long-standing friend, Lance Korthals, mentioned our decision to purchase the Sony system. Korthals, dbx pro sales director, felt that he had to let a good friend like Randy in on a "little secret." Well, his little secret may well be one of the most significant advances yet in our industry. As you must have guessed by now, dbx was developing a digital audio processor.

If it seems odd that dbx would enter into digital audio, think for a minute. This company has made one of most significant developments in reducing tape hiss and expanding dynamic range. Given that innovation, it has probably reached the limitations of analog audio. Where else could the company turn, but to digital?

In less than two weeks, the studio arranged for the prototype dbx 700 Digital Audio Processor, as well as a Sony digital machine and two ¾-inch U-Matic VCRs, to be installed in our Studio "A" for serious evaluation over a four-day closed session. (Special thanks to Tom Semmes and Associates for the loan of the Sony digital system.) Although this would be the first time we've had a digital recorder in our facility, I am no stranger to digital recorders. I have been to every AES and NAB show in recent years and, as I said before, we have been evaluating digital audio for quite some time. I am well aware of the attributes as well as the deficiencies of the different formats on the market, as well as fundamental A-to-D problems. I have to admit that Crescendo primarily was looking for potential problems or deficiencies in its evaluation.

Systems Evaluation

Our evalutation was set up as follows: We would eliminate the multitrack, and cut "live" straight to the mastering machines. This eliminated any analog tape link. Identical two-track mixes were fed to the Sony PCM-1610, the dbx 700, and to an analog Otari MTR-10, it being felt that the Otari represents the "state-of-the-art" of analog tape machines. The MTR-10 has adjustable phase compensation, and a unique head design that make it audibly superior to everything we've evaluated — in other words, the ideal "analog reference." All levels were calibrated for each machine's optimum performance. No signal processing was used, since limiters, gates, etc., would only mask deficiencies.

With the help of Dr. Robert Manchurian, a prominent Atlanta arranger-producer, and Albert Coleman, of the Atlanta symphony, Crescendo proceeded to book the most diverse and challenging sessions we could. These included a classical pianist, rock drummers, jazz percussionists, acappella vocalists, string sections and soloists, horn sections and soloists, plus jazz, fusion, and rock bands.

In light of the magnitude this evaluation was taking, it was decided to involve as many ears as possible. At dbx's request, we did not identify to anyone that the company's prototype was here. Our engineers, producers, and performing musicians listened to each cut, while the musicians auditioned only what they cut. After each cut, all three machines were played back, and simply identified as A, B and C.

Considering the diversity in listeners, I believe that we compiled some significant data. After all, who knows better what a violin should sound like? An engineer or the performing concert violinist? On the other hand, however, it's the well-tuned ears of an engineer that notices abrupt cut-off of long-fading resonance (due to error correction circuitry in some digital recorders).

When the results were in after an exhaustive four days, they were, to say the least, "interesting."

No one *ever* chose the analog recordings; the limited dynamic range was immediately apparent. The Consensus between the PCM-1610 versus the dbx 700 was split equally. Everyone agreed the difference was minimal. However, the more seasoned ears could ascertain between the two most of the time. There seemed to be no peer grouping as to preference. The engineers were split, but the musicians seemed slightly to prefer the sound of the dbx 700.

I have to admit in this "blindfold test" I did choose the Sony PCM-1610 most of the time. However, just when I though I could tell the difference, I chose the dbx 700, insisting it was the Sony. But my partner, Randy Fuchs, consistently picked the dbx unit as his preferred choice. Our engineers, Will Eggleston and Jim Boling, could identify which was which after about 20 seconds. They disagreed, however, as to which they liked better.

The slight differences in the two digital machines were most noticeable in the high-frequency transients. The Sony PCM-1610 seemed to be more "piercing," for lack of a better term. Depending on your perspective, our evaluators defined the Sony as harsh (bad) or brilliant (good). The dbx 700 was described by the same evaluators as slightly dull (bad) or smooth (good).

The noise floor was non-existent on both units (below the noise floor of our mikes and boards).

The low-frequency response was incredible on both machines. Low frequencies, I might add, are one area that analog machines can't touch digital — with or without signal processing.

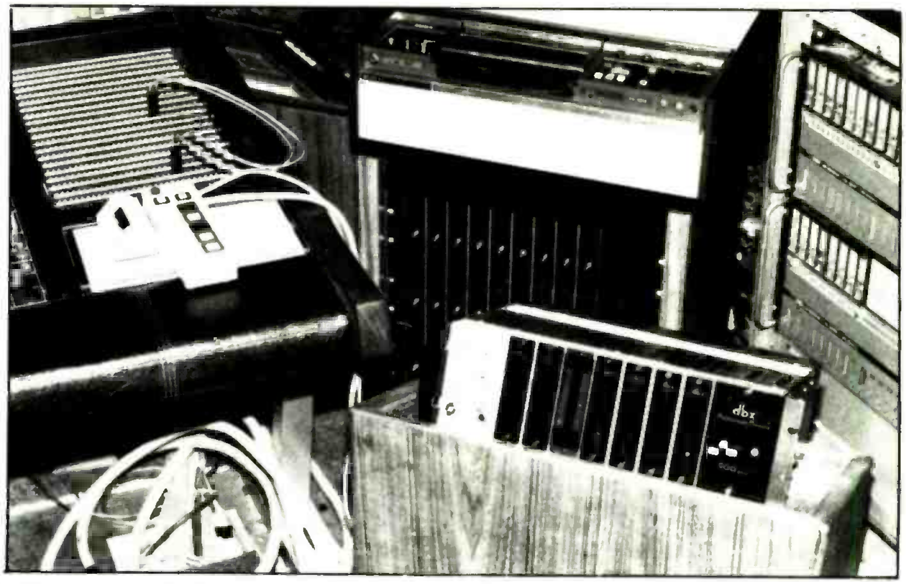
There have been claims that PCM-based digital recorders have a tendency to chop off a signal that falls below a certain SPL, in much the way that a gate would. It is my understanding that error-correction circuitry is responsible for this. dbx informed us that its unit was *not* a PCM system, so we did listen for this anticipated problem. We were not able, however, to get either unit to "chop" any part of even the longest and softest fades.

Cost Advantage

One thing I've refrained from mentioning until now is the cost differential between the two digital recorders we listened to. The dbx 700 is priced between 1/6 and 1/7th the cost of the Sony PCM-1610. While the Sony is truly an excellent machine and certainly *cosmetically* much more impressive to look at, we are purchasing the dbx.

Performance-wise the two machines are on a par. There are some packaging features I think show excellent fore thought on dbx's part. They've constructed the unit in a lightweight, vertical rack package that is similar to their 900 Series modular signal processing frame. The new 700 system is modular, and gives the user the option of tailoring a unit for his particular needs. The available modules are input, output, and mike pre-amps, which permits a mastering facility to purchase a playback-only unit, for example. It also gives the "live" performance "direct-to-deck" user an extremely high-quality mike pre-amp. This pre-amp will be essential for esoteric digital recording, since most available mike pre-amps and consoles that have acceptable noise

Crescendo Recorders' co-owners Randy Fuchs (left) and William Ray during evaluation of prototype Model 700 against a "conventional" PCM digital audio processor.



Close-up detail of prototype Model 700 and U-Matic ¾-inch videocassette recorder used to record digitally-encoded material.

floors for analog use will not cut it for digital.

An interesting observation at this point is that what has been until now one of the quietest links in the audio recording chain will now be the noisiest — you guessed it, the microphone.

dbx has been successful in overcoming some of the objections (the biggest being cost) we've all heard about digital. However, there are a few problems that remain.

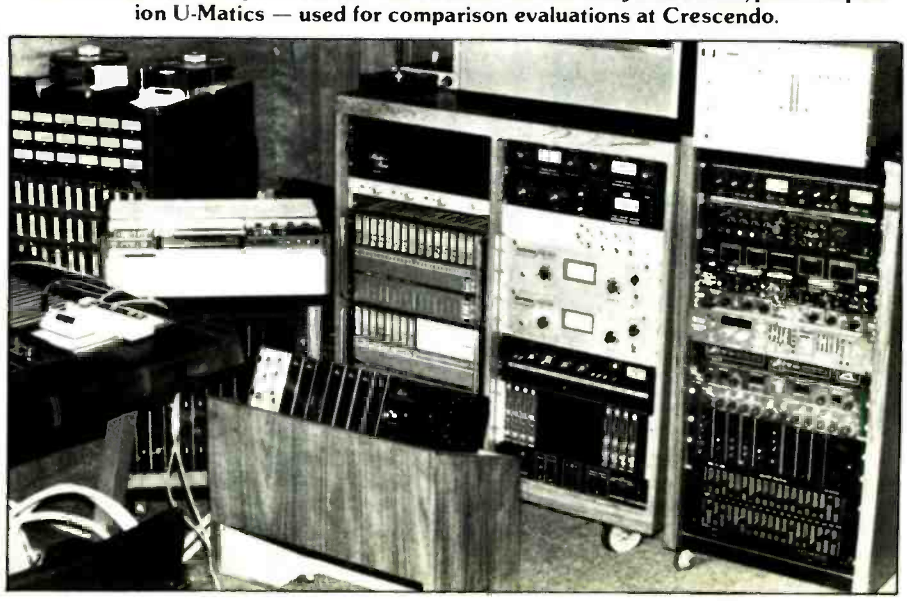
The dbx 700 uses a VCR and videotape. For editing, this means you either need two units and a video editor, or access to a video editor. However, on a positive note, any video editor that will interface with your VCR will suffice. While dbx recommends that it be used with a ¾-inch U-Matic, the 700 processor produces excellent results with ½-inch video tape recorders as well. The Sony and all others must use ¾-inch tapes. There is at least a $3,000 difference in the cost of a ¾-inch U-Matic and a ½-inch consumer VCR.

In our opinion, what dbx has accomplished with its digital audio recorder is certainly going to rock the industry. Facilities competing for album projects will certainly be forced to purchase a digital machine, or lose their business to the competition who has. Considering the cost of the dbx processor and ¾-inch VCR is roughly in line with a good analog recorder, price should certainly not be an obstacle.

Other users of high-quality half-tracks may be interested for other reasons. One very important issue that lies on the positive side of videocassettes is that as a storage medium they are very compact and easy to handle, and you don't have to worry about record/replay EQ, or tape speed.

Another plus with the dbx 700 unit is that a 60-minute ¾-inch U-Matic cassette costs $20.00 each in quantity. If you add up what 60 minutes of tape costs running at 30 IPS, you'll find

The two digital audio processors — dbx Model 700 and Sony PCM-1610, plus companion U-Matics — used for comparison evaluations at Crescendo.



yourself with four, 10-inch reels, or approximately three to four times the cost, with a considerable increase in bulk. To users with extensive tape libraries — for example, radio broadcasters and radio post production — this alone could be reason enough to go to dbx's digital format.

Towards the Future

Before closing, and while I have the chance to "put it in print," I'd like to share some observations of the past and some projections for the future. As mentioned earlier, our inudstry has had to deal with a lack of standardization. Perhaps one very appropriate example to cite would be the Dolby and dbx noise reduction systems. Dr. Ray Dolby was first to come up with a system to significantly reduce the noise floor of a tape. However, dbx would soon be introducing an "alternative." And, as you all know, a triumphant victor did *not* emerge; our facility has both Dolby *and* dbx, and our clients swear by one or the other (or both).

In this case, had a format been established as a "standard" for noise reduction, we would have to give up audio integrity in some applications. Both systems have their attributes, as well as deficiencies.

As much as we'd all like to see standards set for a digital recording format, realistically I don't believe it will happen. Perhaps a by-product of "Yankee Ingenuity" is a common consensus that there is *always* a better way. This, coupled with healthy capitalist competition, will certainly lead innovative manufacturers, such as dbx, into alternative ways of manufacturing a digital recorder. The performance difference of going away from a PCM format, in the way that dbx has, is virtually beyond this listener's ability to perceive (hear). The cost advantage of going to dbx's encoding format is significant. The technology involved is simpler to execute than PCM, thereby enabling dbx to make significant reductions in component count, as well as size and weight.

Given that most studios probably do not have in-house personnel to repair digital recorders, I believe that dbx has a big advantage over its competition in that its new processor is less complex, and completely modular. With a few spare "cards," a studio should "theoretically" never have any downtime.

The dbx digital approach is, to our mind, certainly the most viable and well-thought-out yet. However, PCM-type recorders have already gained a viable foothold in our industry. Although current technology will not permit a PCM-based recorder to compete economically with dbx's approach, I think we will continue to see PCM-based recorders. And so — alas — we will, once again, have multiple formats, and no standardization. The only consolation may be that with the money we've saved on our recent purchase from dbx, I will be able to buy other innovative and new products. ■■■

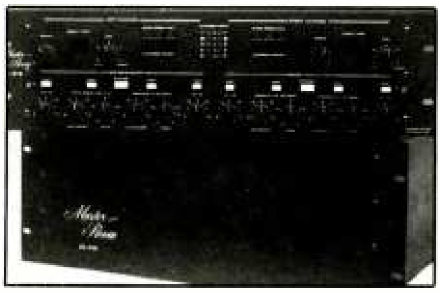
New Products

MASTER-ROOM XL-515 REVERBERATION SYSTEM FROM MICMIX

The Master-Room XL-515 is said to provide the user with virtually unlim ited versatility, along with unprecedented performance from a spring reverberation system. The new unit offers three operational modes in full stereo tosynthesize the reverberation characteristics of a live chamber, plate, and concert hall. Any of these three modes can be easily selected from the main control unit.

The Plate mode offers a bright, clean sound with the high echo density, and the instantaneous diffusive qualities of a plate-type reverberator. The Room mode incorporates the most desired characteristics found in some of the most popular live chambers currently in use. Finally, the Hall mode provides the reverberation characteristics of a concert hall that can be varied in apparent size and sonic qualities.

Control parameters allow a large number of variations in each mode to specifically tailor the sound of the reverberation environments. The continuously variable Decay provides variations from 1 to 6 seconds; importantly, MICMIX claims, no change in tonality occurs when the decay time is varied. The Decay Time is displayed by a two-digit numeric display. An equalization section contains low and high fixed con-



trols, along with two mid-range sweepable controls, all with 12 dB of boost and cut.

The main control unit is housed in a 5¼-inch rack-mount package with all control functions located on the front panel. The remote chamber unit is housed in a 7-inch rack-mount package that can be mounted up to 200 feet from the main control unit.

Suggested user price is $3,990.00.

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

For additonal information circle #135

DIGITAL DISK MASTERING CONSOLE FROM NEVE

According to Neve, the new 9202 Digital Disk mastering Console represents the final component in the recording chain. Digital signal processing and digital quality can now be secured from the studio to the final disk master.

The 9202 DDM console accepts and provides digital *or* analog inputs and outputs, and therefore can be used with a conventional analog disk-cutting lathe, or as a tape-to-disk transfer console for digital disks.

Digital technology allows the new Neve disk mastering console to be completely self-contained with integral delay facility and total memory capability. Full dynamic range control facilities and equalization can be incorporated in each signal path, and these processors may be switched either before or after the delay circuit. Delay is variable up to a maximum of 1.33 seconds at 48 kHz (1.45 seconds at 44.1 kHz), and can be extended to a maximum of 2.66 seconds at 48 kHz (2.9 seconds at 44.1 kHz) if required.

Many of the features found in the new Neve multichannel digital DPS consoles are incorporated in the 9202 DDM, including 4-band equalizers with memorised control settings, and automated motor driven faders. All signal processing circuits are housed in a single 19-inch rack cabinet, and a modular system of building blocks enables the control desk to be configured to allow freedom of layout.

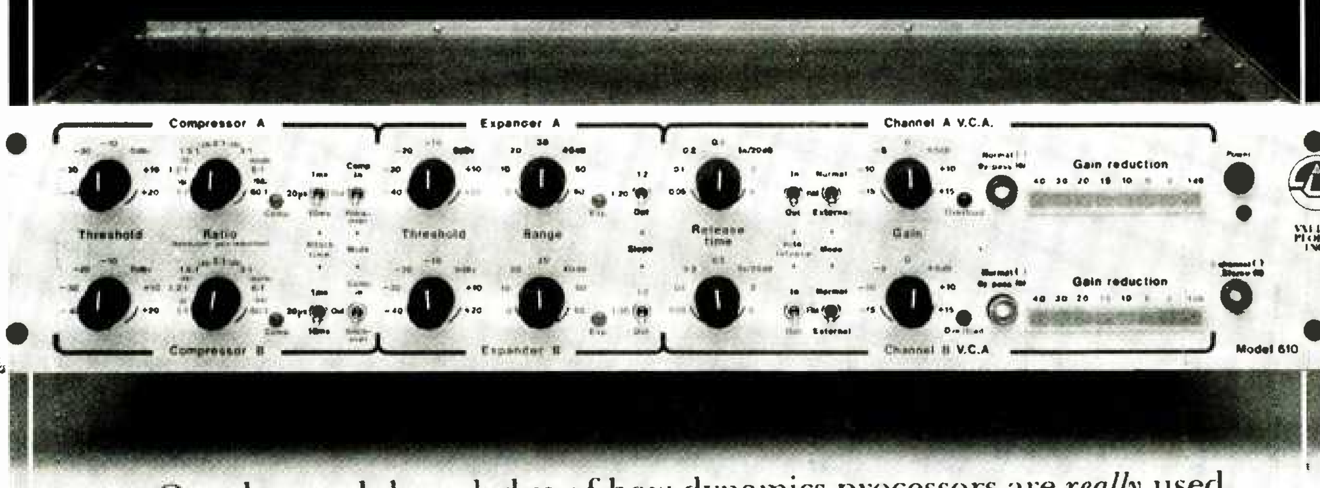
RUPERT NEVE, INC.
BERKSHIRE INDUSTRIAL PARK
BETHEL, CT 06801
(203) 744-6230

For additonal information circle #137

LEXICON LAUNCHES PCM-42 DIGITAL PROCESSOR

The new PCM-42 has a maximum delay of 2.4 seconds, and memory option

Desired Results Delivered.



Our thorough knowledge of how dynamics processors are *really* used, and the performance demanded of them, is reflected in the flexible operating functions of the Model 610 Dual Compressor/Expander.

For example, the desired result of compression should be a controlled dynamic range. All too often, this is achieved at the expense of emphasizing noise. The Model 610's interactive Expanded Compression mode offers precise dynamic control and unobtrusive reduction of residual noise.

By definition, an expander should increase dynamic range. Unfortunately, most don't know when to stop. The Limited Expansion mode of the Model 610 has great brakes, allowing precise control of maximum output level after expansion, thus avoiding distortion.

Constant nominal output level under varying combinations of threshold and ratio settings is maintained automatically, eliminating the need for dual metering.

The VCA employed in the Model 610 offers stable and distortion-free performance under *all* operating conditions. No small claim, since the gain control device is the heart of any dynamics processor.

The Model 610.
Results Delivered.

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

For additional information circle #138

New Products

of 4.8 seconds. The unit's crystal-based delay timer accurately tracks all changes in delay times, including time modulation sweeps. A metronome indicator and clock can be programmed to a precise fraction of the delay period, enabling long delay loops to be used to generate tightly woven, multi-layed,

rhythmic beds, and completely new sound-on-sound effects.

The PCM-42 has 16 kHz bandwidth, input overload protection, and Lexicon's proprietary digital encoding system. Time modulation controls include an envelope follower that can be used alone, or blended with either a sine or square wave sweep for enhanced doubling sounds, "talking flange" effects, unusual "trills,' and pitch twisting effects. An optional foot control enables infinite repeat, by-pass, delay sweep, recirculation and output mix functions.

LEXICON, INC.
60 TURNER STREET
WALTHAM, MA 02154
(617) 891-6790

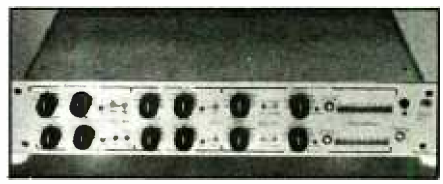
For additonal information circle #139

VALLEY PEOPLE MODEL 610 DUAL COMPRESSOR EXPANDER

The 610 contains two independent channels, consisting of a compressor section and an expander section, both of which control the channel VCA. The channels may be operated independently or coupled for processing stereo program material.

Each of the compressor sections feature continuously-variable thresholds and compression ratios, with a threshold/ratio/output coupling scheme which computes the amount of additional output gain required to maintain a constant nominal output level under varying combinations of threshold and ratio settings. The compressor sections use Valley People's Linear Integration Detector to preserve program dynamic integrity during passages of heavy gain reduction, and Peak Reversion Correction circuitry to lessen "pumping" in the presence of low frequency information.

Each of the two expander sections features selectable slopes of 1:2 or 1:20, and continuously variable thresholds. The expanders also use the proprietary linear Integration Detector.



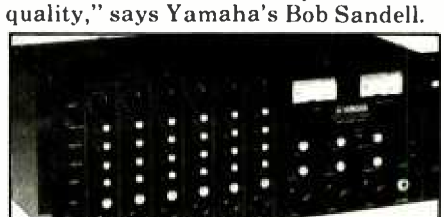
Proprietary coupling of the release circuitry in the channel compressor and expander sections makes possible a unique mode: interactive expanded compression. The nominal release time is set by the VCA release time control, and may be modified through use of the Auto Release function. This combination is said to provide an imperceptible transition from compression to expansion, resulting in noise reduction without adding the adverse effects associated with "hard noise gating."

VALLEY PEOPLE, INC.
P.O. BOX 40306
NASHVILLE, TN 37204
(615) 383-4737

For additonal information circle #141

RACK-MOUNTABLE SOUND MIXER FROM YAMAHA

The M406 is a six-input channel, stereo output sound reinforcement mixer of rugged construction. "Yet, overall the M406 has the attractive appearance, smooth control feel, and superior sound quality," says Yamaha's Bob Sandell.



"Its straightforward features and superb audio performance make it an excellent choice as the sole mixer in a

Why Beyer microphones give you more extraordinary performance for the most ordinary applications.

Beyer M 88

Beyer M 201

Beyer M 160

There are other microphone alternatives when high sound pressure is a factor.

As Sennheiser claims, the MD 421 undoubtedly stands up to extremely high decibel levels and has other features that have contributed to its popularity. But if you're already using the MD 421 to mike loud instruments or voices, we suggest that you investigate the Beyer M 88.

The Beyer Dynamic M 88's frequency response (30 to 20,000 Hz) enhances your ability to capture the true personality (including exaggerated transients) of bass drums, amplified instruments and self-indulgent lead vocalists.

The Beyer M 88 features a matte black, chromium-plated brass case for the ultimate in structural integrity. Beyer microphones are designed for specific recording and sound reinforcement applications.

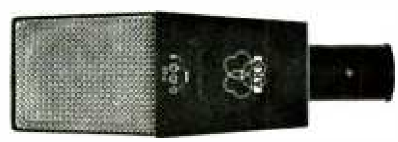
When you need a rugged and versatile microphone, consider the alternatives.

For over 10 years, engineers have used mics like Shure's SM57 for the widest variety of applications in the studio. And we feel that one of the main reasons more engineers don't use the Beyer M 201 in this context is simply because they don't know about it. Those who have tried it in the full gamut of recording situations have discovered how it can distinguish itself when miking anything from vocals to acoustic guitar to tom toms.

The M 201's Hyper-Cardioid pattern means that you get focussed, accurate reproduction. Its wide and smooth frequency response (40 to 18,000 Hz) provides excellent definition for the greatest number of possible recording and sound reinforcement situations.

Each Beyer Dynamic microphone has its own custom-designed element to optimize the mic's performance for its intended use.

You may not always need a condenser microphone for "critical" recording applications.



Some engineers prefer condenser microphones like the AKG C 414 to accurately capture the subtle nuances of a violin or acoustic piano. But should you have to deal with the complexity of a condenser system every time this kind of situation comes up?

The Beyer Dynamic M 160 features a double-ribbon element for the unique transparency of sound image that ribbon mics are known for. While its performance is comparable to the finest condenser microphones, the M 160's compact size and ingenious design offers significant practical advantages for critical applications.

Beyer Dynamic microphones offer state-of-the-design technology and precision German craftsmanship for the full spectrum of recording and sound reinforcement applications.

The Dynamic Decision **beyerdynamic**

*Documentation supporting specific comparative claims available upon request.

Beyer Dynamic, Inc. 5-05 Burns Avenue, Hicksville, New York 11801 (516) 935-8000

For additional information circle #142

New Products

small club, meeting room, church, and similar applications," Sandell added. The M406 may also be used as a submixer for larger mixing consoles in complex sound systems.

The compact package (19 by 7 by 11.6 inches deep) features +24 dBm 600-ohm balanced XLR outputs, three-band EQ, six-position input level controls, and phantom power for condenser microphones. Other features include: echo and effects send bus with master send control; two effects inputs, each with level and pan controls; and front-panel power switch for easy rack mounting.

One of the dual illuminated VU meters and stereo headphone output are switchable, enabling the user to monitor the program or echo output. The VU meters feature LED peak indicators.

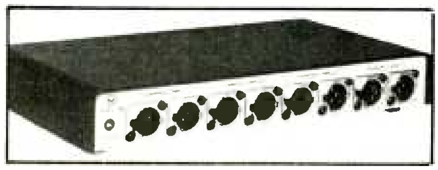
The M406 carries a suggested retail price of $995.00.

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

For additonal information circle #143

MICROPHONE PHANTOM POWER SUPPLY FROM CROWN

The PH-4 system supplies 48 volts of DC phantom power for all types of microphones, according to Clay Barclay, product development manager for Crown International. The system consists of a master unit (PH-4) with connections for up to four microphones, plus slave units (PH-4S), each of which adds capability for another four microphones; slaves are daisy-chained with cables supplied by Crown.



A master PH-4 unit will supply up to 100 milliamps of current, enough to power up to about 12 condenser microphones, or up to about 20 Crown PZM models. Both master and slave units are contained in rugged but light-weight aluminum chassis, finished in a non-chippable urethane. Optional "ears" are available for standard 19-inch rack mounting.

Suggested list price for the PH-4 is $179.00.

CROWN INTERNATIONAL
1718 W. MISHAWAKA ROAD
ELKHART, IN 46517
(219) 294-5571

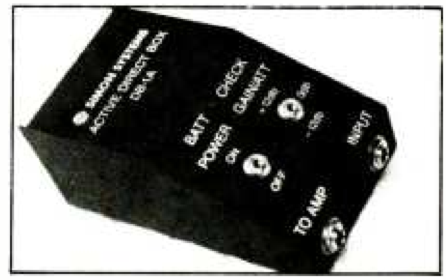
For additonal information circle #145

SIMON SYSTEMS MODEL DB-1A ACTIVE DI

Stand-out features of the new Model DB-1A include high current output capable of driving capacitive loads; stable circuit design to prevent RF pick-up or oscillations; automatic battery check circuitry; three-way power scheme (battery, AC, or rechargeable battery and 3-position gain/att switch for input attenuation of low-Z, high-level sources, "no loss" instrument input, and direct into a line input.

Frequency response is a quoted 10 Hz to 150 kHz, +0, -0.5 dB; distortion less than 0.005%; and dynamic range 10[illegible] dB.

Also available from Simon Systems: the CB-4 Headphone Cue box which allows up to four pairs of stereo head-



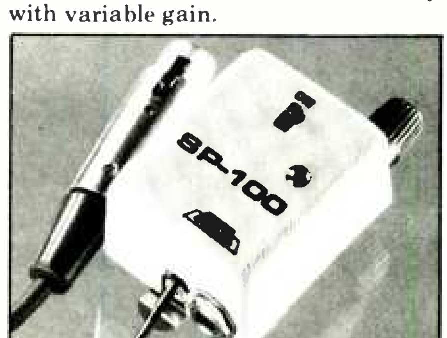
phones to be driven from the same amplifier.

SIMON SYSTEMS
20224 SHERMAN WAY #23
CANOGA PARK, CA 91306
(213) 716-7905

For additional information circle #146

ARTISTS X-PONENT ENGINEERING UNVEILS SP-100 HEADPHONE AMP

The new SP-100 is a belt pack headphone amp, described as being invaluable for monitoring mike or line level signals, as well as general audio system troubleshooting. The unit's high input impedance allows for minimum circuit loading, making it ideal for tuning wireless microphone receivers, setting up and balancing piano pickups, quality testing microphones, and as a "listen only" intercommunication headset amp with variable gain.

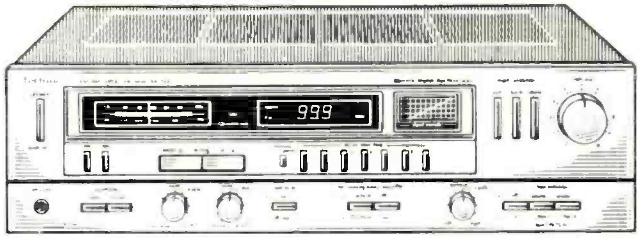


The unit weighs just 4 ounces, is priced at $74.95, and features long battery life, low noise, wide frequency response, and can accommodate almost any high-or low-impedance, balanced or unbalanced, signal source.

AXE
P.O. BOX 2331
MENLO PARK, CA 94025
(415) 365-5243

For additonal information circle #147

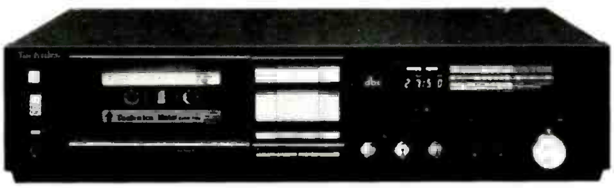
TIME TO SAVE . . . on Technics



SA-222 Quartz Synthesizer Digital Analog Receiver.

30 watts per channel, minimum continuous RMS into 8 ohms, both driven from 20 Hz to 20 kHz, with no more than 0.04% total harmonic distortion. Quartz synthesizer digital tuning assures exact drift free reception, with three ways to tune stations: pushbutton presets for 14 stations (7 AM and 7 FM), up/down manual tuning, and auto scan tuning. Digital frequency readout of tuned-in station plus analog display LED indicators for signal strength and quartz-lock. Pure-complementary OCL amplifier with electronic protection circuit. Phono S/N 75 dB (IHF '78) for excellent disk reproduction. Soft touch program selectors. Two tape monitoring switches. Subsonic filter. FM muting/mode selector. ***Adray's 'best deal' price:*** **$189.00**

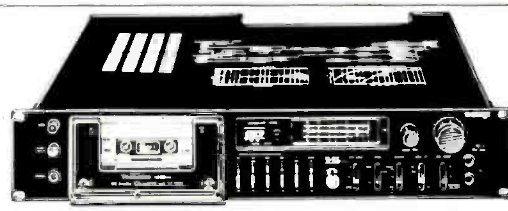
Manufacturer's suggested retail: $350.00



RS-M275X dbx/Dolby B-C, Direct Drive, 3-Motor Cassette Deck

with Microprocessor feather touch controls. Wide-range (-40 dB to +18 dB) 3-color FL meters with peak hold. Multi-function FL display: 4-digit real-time counter with memory repeat, and music select counter for up to 20 programs. Intro-Search samples each program for quick and easy program access. AX (amorphous) head improves high range response. Auto tape selector. Bias fine-adjust control. Auto input selector. Output level control. Auto input selector. Wow and Flutter: 0.03% WRMS. Frequency response: 20 Hz to 20 kHz (Metal). S/N (CrO_2): 92 dB (dbx in), 76 dB (Dolby C in, CCIR). ***Adray's 'best deal' price:*** **$398.00**

Manufacturer's suggested retail: $600.00



RS-M85 Mk2 Quartz Locked Professional Series Direct Drive Cassette Deck.

The system which produces incredible specifications and which was reported in a national magazine to have the best tape speed characteristics ever measured in a cassette deck. Frequency response: 30 to 17,000 Hz (metal). Wow and Flutter is reduced to a miniscule 0.035%. Signal to Noise ratio: 69 dB (Dolby in). Speed deviation: 0.3%. Slim 3⅞-inch high design. ***Adray's 'best deal' price:*** **$475.00**

Manufacturer's suggested retail: $700.00

RS-1500US "Isolated Loop" Open Reel Professional Series Tape Deck.

This is the two-track, 2 channel version of this series, which features the Isolated Loop system with direct drive capstan and reel motors with electronic tape tension sensing to insure extremely stable tape transport (0.018% wow and flutter WRMS at 15 ips). The direct drive capstan motor is quartz-phase locked. Frequency response: 30 Hz to 30,000 Hz ±3 dB. Signal to Noise ratio: 68 dB. Distortion at 0 VU: 0.8%. Full IC-logic transport controls. Removeable head assembly. ***This is a repeat of a previously sold-out Adray's 'reel deal':*** **$960.00**

Manufacturer's suggested retail: $1600.00

Where the pro's shop for their personal equipment!

At either location, or send check or money order for freight collect delivery.

Adray's

5575 WILSHIRE BOULEVARD
LOS ANGELES, CA 90036
(213) 936-5118

6609 VAN NUYS BOULEVARD
VAN NUYS, CA 91405
(213) 908-1500

For additional information circle #148

New Products

SOUNDCRAFT SERIES 400B CONSOLES

A new series of general purpose mixing consoles, the Series 400B is available in two formats and two sizes. Both formats are fully modular, include phantom power supply, and feature 4-band sweep-frequency EQ.

The Standard format, available with 16 or 24 inputs, features four auxiliary sends, 8-track monitoring, subgrouping, a set-up oscillator, and 100 mm faders.

The Monitor format, also available with 16 or 24 inputs, features eight discrete mixes for on-stage monitor-mixing; a master channel level control can be assigned via a pan control to a stereo mix bus for side-fills, or front-of-house mix.

The Soundcraft Series 400B, which will cost $5,500 for 16-input models, and $7,500 for 24-input versions, can be seen at the forthcoming AES Convention in Anaheim, together with a fully-automated Series 2400 console, and the new Series 1600 boards.

SOUNDCRAFT USA
20610 MANHATTAN PLACE
TORRANCE, CA 90501
(213) 328-2595

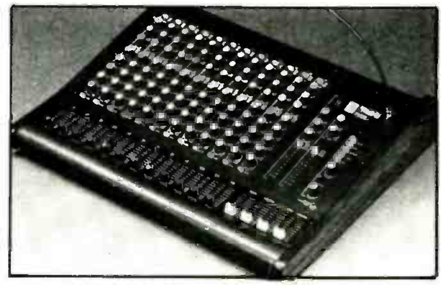
For additonal information circle #149

NEPTUNE ELECTRONICS 22 SERIES STEREO MIXERS

Available with 8, 12, and 16 input channels, the 22 Series stereo mixers feature mike- and line-level inputs; input pre-amp in/out jacks; mike/line switching; peak LED indicators; monitor, reverb (all consoles have built-in Accutronics type 9 tank), aux (with pre- or post-EQ/fader switching); 3-way EQ; pan; solo; and slide faders.

Master control features include extensive headphone monitoring and solo system; channels, the 22 Series balanced mike-switchable metering; panable aux return and line-level inputs; input pre-amp in/out jacks; mike/line switching; peak LED indicators; monitor, reverb (all consoles have built-in Accutronics type 9 tank), aux (with pre- or post-EQ/fader switching); 3-way EQ; pan; solo; and slide faders.

Master control features include extensive and panable reverb; slide fader master output controls for left/right, monitor master and mono. Rear panel connections offer both balanced and unbalanced line-level outputs on all main output functions (L/R, monitor and mono); high- and low-level aux return jacks; and master function inputs that allow two 22 Series consoles to be interconnected for more input channels.



In addition, the 22 Series can use its own internal power supply powered with NEI's optional XMP remote power supply.

NEPTUNE ELECTRONICS, INC.
934 NE 25TH AVENUE
PORTLAND, OR 97232
(503) 232-4445

For additonal information circle #151

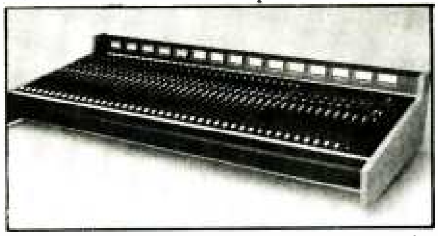
AUDIOARTS 8X SERIES MIXING CONSOLE

The 8X Series is intended for eight-track recording, and features three-band sweepable equalization, high-pass filter, phase reversal switching, phantom power, two effects sends, one cue send, and stereo monitor. Patch points,

For additional information circle #150

direct outs, group outs, and bus outs on all input channels are also provided. Pre-fader listen, post-fader listen and tape solo are standard.

All mike, line, bus and send outputs are electronically balanced to assure compatability with today's high performance multitrack tape recorders.



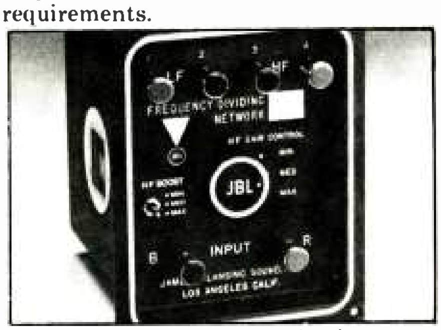
Prices range from $5,000 to $15,000, in mainframe configurations from 16 to 32 inputs. The 8X Series will be on display at the forthcoming AES Convention in Anaheim.

AUDIOARTS ENGINEERING
5 COLLINS ROAD
BETHANY, CT 06525
(203) 393-0887

For additional information circle #152

THREE NEW HIGH POWER FREQUENCY DIVIDING NETWORKS FROM JBL

Models 311A, 3115A and 3120A are each equipped with a three-position, high-frequency equalization boost switch which compensates for power response roll-off; the switch may also be used for tailoring of the high frequency response contour to individual program requirements.



High-frequency attenuation is accomplished with tapped autotransformers rather than conventional resistive losses. Ideally suited for use in high-powered sound reinforcement applications, each of the three new products handles 300 watts of continuous program power. Model 3110A has a crossover frequency of 800 Hz; 3115A 500 Hz; and 3120A 1.2 kHz.

JAMES B. LANSING, INC.
8500 BALBOA BLVD.
NORTHRIDGE, CA 91329
(213) 893-8411

For additonal information circle #153

BX-25E REVERB/DELAY FROM AKG

Replacing the BX-20E, the new BX-25E is based on the patented Tortional Transmission Line principle used in all AKG reverb units. The TTL-System has been improved by extending the overall length of the spring by about 25% over the BX-20E, yet at the same time reduc-

New Products

ing the overall size of the BX-25E dramatically by about one-third.

Independent decay-time adjustment (via remote control), high- and low-frequency equalization, external input/output level adjustments, and dry/reverb signal mixing are provided for each of the two electronically and acoustically separate channels. Decay time is adjusted silently through the use of motional feedback.

In addition to the shelving-type frequency equalization, AKG has incorporated a high-cut filter at the input and reverb drive amplifier. This enables the selection of a bright, more "aggressive" sound, or a more mellow natural reverberation sound.

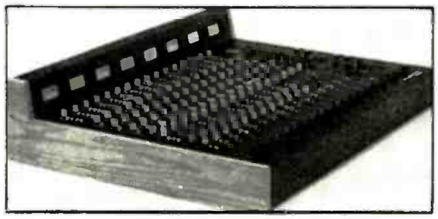
Also, the new M-250 Digital Delay module may be added to a BX-25E at any later time, or ordered within the unit. The delay module provides remote mix control between reverb signal and reverb plus individual reflections, and individually adjustable level for each of the discrete reflections in 2 dB steps from original level down to 20 dB below the original level. Initial delay for the reverb signal is switchable to 0, 30, and 60 milliseconds; two discrete reflections for each channel may be adjusted in 6 millisecond steps from 6 to 60 milliseconds. Bandwidth is quoted 12 kHz.

AKG ACOUSTICS, INC.
77 SELLECK STREET
STAMFORD, CT 06902
(203) 348-2121

For additional information circle #156

SPECK ELECTRONICS INTRODUCES SPECKMIX 16 CONSOLE

The new console features 16 complete input channels equipped with low-noise, transformerless mike inputs; eight mixing bus outputs; eight VU meters and eight-track panable assign. EQ is pro-



vided by six, 3-band equalizers. Facilities are provided for control room and studio playback, talkback and cue prompts. There is an independent stereo mixdown bus.

Frequency response on the Speckmix 16 is a quoted 23 Hz to 20 kHz (±1 dB); output level is +4 dBm with the maximum output level at +22 dBv; and noise -72 dB measured from mike input to bus output, and -80 dB measured from line input to program output.

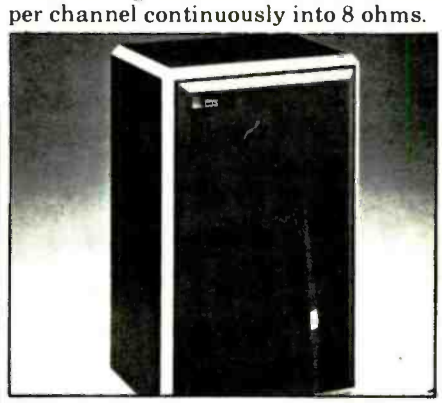
Suggested price is $3,975.00.

SPECK ELECTRONICS
12455 BRANFORD STREET, #2
ARLETA, CA 91331
(213) 897-4188

For additonal information circle #157

MEYER SOUND UNVEILS 833 STUDIO MONITOR

The new 833 Studio Reference Monitor consists of two vented enclosures — each housing a single proprietary 15-inch, low-frequency driver, passive crossover, and horn-loaded high-frequency driver — and an active stereo electronics unit containing subsonic filter, frequency and phase response correction circuitry, and Meyer's Speaker Sense™ driver protection circuitry. The new monitor requires a high-quality stereo power amplifier capable of delivering between 100 and 400 watts per channel continuously into 8 ohms.



Employing frequency-selective phase correction techniques, in combination with continuous monitoring and control of the amplifier output power, the Model 833 is said to offer all the advantages normally associated with bi-amplified systems, while utilizing a passive crossover, requiring only a single stereo amplifier. The electronics unit features an LED bar display of true amplifier power, and a user-setable peak limiter which acts on the signal at line level, and is designed to be set just below the power amplifier clipping point.

MEYER SOUND LABORATORIES, INC.
2194 EDISON AVENUE
SAN FRANCISCO, CA 94577
(415) 569-2866

For additonal information circle #159

NEW VERSIONS OF HARDY 990 OP-AMP

The 990 discrete op-amp is now available in two new versions: the 990-18V and 990-12V, for bi-polar 18 and 12 volts respectively. Other versions include the 990-24V and 990-15V, for bi-polar 24 and 15 volts respectively. Normally encapsulated in clear epoxy, the 990 is also available in black without labels, for DEM applications.

Basic specifications include: EIN -133.7 dBv (unstd, re: 0.775V, 20-20 kHz); slew rate 18 volts per microsecond with 150 ohm load; and +24 dBv output with 75 ohm load (990-24V).

THE JOHN HARDY CO.
P.O. BOX AA631
EVANSTON, IL 60204
(312) 864-8060

For additional information circle #160

MC-220 MINICUBE DIRECT BOX FROM AUDIO ENVELOPE

The MC-220 features active transformerless circuitry, which maintains ground isolation even when phantom powered; flexible powering from either one, 9-volt battery, or any console phantom power supply; instrument- and

amplifier-level inputs; fully isolated and buffered unity gain link output; balanced microphone-level input; switchable low-pass filter; ground lift switch; and totally recessed switches and jacks.

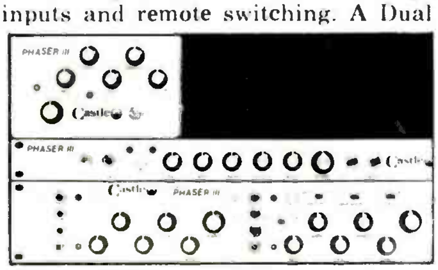
Suggested retail price is $75.00.

AUDIO ENVELOPE SYSTEMS
P.O. BOX 113
SCOTTSDALE, AZ 85252
(602) 834-3588

For additonal information circle #161

CASTLE INSTRUMENTS PHASER III EFFECTS UNIT

Features include five ultra-wide range parametric controls; switch-selectable 4/6/8 stages; optional control-voltage inputs and remote switching. A Dual



Rack mount version incorporates two complete phasers in a single package, with four crosslinking switches that selectably interconnect the LFOs, feedback paths, inputs, and outputs of the two units.

A built-in noise reduction system is said to pass all the subtleties of the most

New Products

delicate sounds, while effectively subduing any noise from the phase-shift stages. High-frequency pre-and de-emphasis techniques are augmented with full-band companding circuitry to achieve an overall dynamic range, the makers claim, that exceeds that of digital tape recording.

All models are available with transformerless balanced inputs and outputs.

CASTLE INSTRUMENTS
2 CARTERET COURT
MADISON, NJ 07940
(201) 377-8185

For additional information circle #164

PAIA ELECTRONICS HYPERFLANGE + CHORUS ANALOG DELAY

The Hyperflange + Chorus sweeps over a 72:1 delay range, from 0.35 to 25.6 milliseconds. This wide range is said to give extremely dramatic flanging effects, as well as lush chorusing sounds.

A unique "hypertriangular" sweep generator provides exponential or linear response for the smoothest possible sweep characteristics; companding noise reduction gives low noise operation. Other features include positive or negative flanging effects, clipping indicator, LFO sync input, control-voltage inputs for delay time and LFO frequency, and balance panpot.

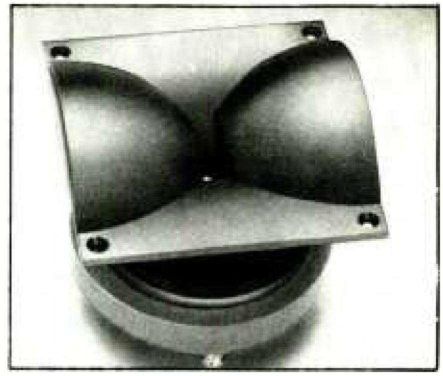
The Hyperflange + Chorus, with comprehensive assembly and applications manual, is available in kit form for $149.95.

PAIA ELECTRONICS
1020 W. WILSHIRE BLVD.
OKLAHOMA CITY, OK 73116
(405) 843-9626

For additional information circle #166

JBL 2370 BI-RADIAL CONSTANT COVERAGE HORN

Providing uniform on- and off-axis frequency response in the horizontal plane from 630 Hz to 16 kHz, the 2370 features a compound flare configuration for smooth response, low distortion, and even coverage. This exclusive computer-aided design is said to minimize the need for horn overlapping, virtually eliminating lobing and comb filter effects. In addition, exceptionally consistent horizontal dispersion eliminates the mid-range narrowing and high-frequency beaming problems typically associated with conventional sound reinforcement horns.



At the higher frequencies, the 2370's vertical mouth dimension is said to create a gradual narrowing of the vertical coverage pattern; as a result, there is acoustic equalization of response in the horizontal plane, and compensation for the falling power response characteristic of all compression drivers. Should constant vertical pattern control be required, two or more 2370s may be stacked to restore full Bi-Radial performance.

JBL's 2370 horn features an integral throat that will accept any one-inch diameter compression driver. Its flat front design allows for flush mounting on baffles; to facilitate installation, mounting tabs are provided for either enclosures or clusters.

JAMES B. LANSING, INC.
8500 BALBOA BLVD.
NORTHRIDGE, CA 91329
(213) 893-8411

For additonal information circle #167

MODEL 174 PITCH SHIFTER AND MODEL 175 DDL UNVEILED BY MXR

The new Model 174 Pitch Shift Doubler provides a number of effects, including "barber pole" flanging; realistic double tracking and unison effects in stereo; stereo chorus effects; 1.1-0.99 type effects; feedback suppression; and 12-string sounds. One-rack space high, the Pitch Shift Doubler has stereo outputs and a red/green LED signal present/overload indicator. It shifts pitch (up to ¼ step) up or down, and operates with minimal noise.

The MXR Model 175 Digital Time Delay provides flanging, doubling, chorusing, simple reverb, echo and slapback echo in an easy to use format. One rack space high, the Digital Time Delay has stereo output capability, and a red/green LED signal present/overload indicator. Other features include delay setting of 0.63 to 320 milliseconds, and a 4:1 sweep range.

MXR INNOVATIONS, INC.
740 DRIVING PARK AVE.
ROCHESTER, NY 14613
(716) 254-2910

For additonal information circle #169

NEW SKOTEL DIGITAL METRONOME FROM AUDIOTECHNIQUES

The Skotel DM-100 features crystal-controlled accuracy, and resolution to 1/100th of a frame. Both film and video (switchable internally NTSC or PAL/SECAM) frame rates are available, and are switchable on the front panel. This is helpful for productions done exclusively on videotape, or for scoring from a video cassette work print which has only timecode burned in and no film frame numbers.

In addition, the click rate may be changed on the thumb wheels while the metronome is running, without affecting the output.

A rear-panel interface connector includes remote start and stop func-

New Products

tions, as well as gating control signals and an output to drive a remote LED display. The tone of the output can be adjusted over a wide range via a rear-mounted screw-driver pot. The line output is balanced and transformer isolated, with enough signal level to overdrive console inputs.

The Skotel DM-100 Digital Metronome, which measures 8 by 6 by 1¼ inches, is priced at $775.

AUDIOTECHNIQUES
1619 BROADWAY
NEW YORK, NY 10019
(212) 586-5989

For additional information circle #172

TWO NEW ACTIVE EQUALIZERS FROM WHITE INSTRUMENTS

The Model 4100A two-channel, octave-band, L-C Active Equalizer is described as being perhaps the quietest 10-band graphic in the industry. White offers that recording engineers will find it to be a highly desirable program equalizer for use with the demanding 30 IPS, half-inch and digital formats.

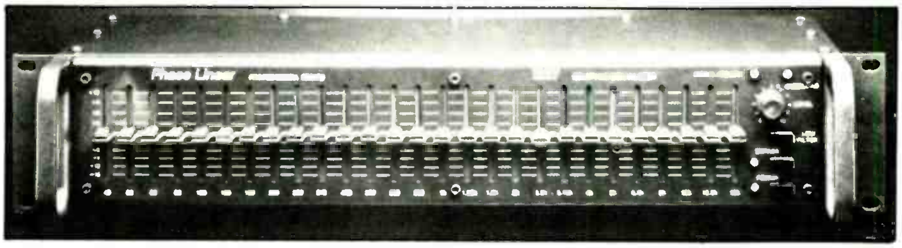
The Model 4400 is the company's new third-octave Monitor Equalizer, and features 28 filters from 31.5 Hz through 16 kHz; 10 dB adjustment range; 12 dB per octave high- and low-pass filters; tri-amp capability with three level trimmers; transformerless operation, or optional plug-in input and output transformers.

WHITE INSTRUMENTS, INC.
P.O. BOX 698
AUSTIN, TX 78767
(512) 892-0752

For additional information circle #174

PHASE LINEAR 27-BAND GRAPHIC EQUALIZER

"The E27 is unique in that it utilizes state variable filters to achieve amplitude change independent of bandwidth," says Peter Horsman, national sales manager for Phase Linear. "This design ensures one-third octave equalization throughout the adjustment range. It eliminates the tendency to broaden bandwidth at small adjustment settings, a characteristic that is typical of other equalizers."



Other features include: +12 dB, -15 dB control range; switchable 40 Hz high pass filter; 12 dB available gain; quoted signal-to-noise ratio of 111 dB below max output with sliders centered; passive bypass; and balanced input and output.

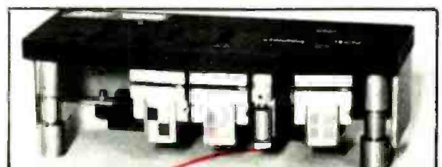
Suggested retail price of the E27 Graphic Equalizer is $549.

PHASE LINEAR
4136 NORTH UNITED PARKWAY
SCHILLER PARK, IL 60176
(312) 671 5680

For additional information circle #175

HALF-INCH HEAD ASSEMBLIES NOW AVAILABLE FROM JRF

The new retrofit half-inch, two-track head assemblies for MCI JH-110A tape machines are said to provide substantial improvement over standard ¼-inch, two-track performance specification. In addition to MCI, half-inch heads are also available for Ampex and Scully tape machines. JRF Company, known for its precision head relapping and assembly alignment services, offers direct replacement heads for most studio, mastering, and tape duplicating equipment.



JRF CO., INC.
17 BYRAM BAY ROAD
HOPATCONG, NJ 07843
(201) 398-7426

For additional information circle #176

For additional information circle #173

views

• RONNIE MONTROSE •

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The snare was miked with a Shure SM56 or an AKG C452, depending on the song and the way the snare was tuned on a given day. The kick was miked with a Beyer M88 or E-V RE-20. "The Beyer is mainly a vocal mike, but it sounds great on a bass drum," says Ronnie — and on the days when the PZMs weren't in the toms, Sennheiser MD 421s were used in the conventional positions. The high-hat was covered by an AKG C414 sometimes, and at other times a 452.

Carmassi's drum kit was set up facing the control room glass, about 10 feet away. "There was a U-87 directly in front of the kit, maybe a couple of feet from the glass," says Montrose. "Once in a while I put two PZMs right on the glass, right and left. I rolled all the bottom off, and used them strictly for 'sizzling' ambience. I also put U-47s — a really fat-sounding tube mike — way in back behind the drums but not in the corners, about 15 feet off the ground."

The bass — a Fender Precision with stock pickups — was recorded with a Countryman direct box. "We didn't use any amplification at all," says Montrose. "I've found that the best thing to do with the bass is to take it direct and add a little bit of soft compression with [UREI] LA-3A or an LA-4 at about 8:1, with the threshold set high. It just shaves the peaks, which allows you to bring the relative level of the bass up a little bit. After that you can do anything you want with it."

After the basics and some of the overdubs were completed, the band took a three-week break during which Montrose listened to "anything but the Gamma rough cuts." He felt it necessary to remove himself emotionally from the project for a while so he could regain his objectivity. When he went back to finish the record, he says he dumped several of the guitar parts.

"The song 'Stranger' had about 20 different versions before we finished it," he recalls. "Every day I'd come in to the studio and Mitchell would say, 'Oh, no — he's going to change it again!' We finally got it to where it felt right, but it took a long time."

For the guitar overdubs and the final mixing, Montrose worked closely with Jim Gaines, who engineered most of Steve Miller's hits. "We used two different setups. When I wanted room ambience, I used the big setup — a couple-hundred-watt head and eight 12-inch speakers — and I'd experiment with the miking of it," he explains. "But Jim recommended that we use a small Gallien-Krueger amp with one 12-inch speaker right there in the control room. I thought we'd have to have a gate on it, but that there were some null spots

views

• RONNIE MONTROSE •

beneath the monitor speakers that didn't pick up much leakage.

"Ken Scott and I had learned that we could record at extremely loud levels, getting feedback through the control-room speakers, if we rolled off the treble in the monitors. We got all the feedback we wanted without high-end screaming and that's the way nearly all the guitar solos on *Gamma 3* were recorded."

Montrose got involved on all levels while doing the keyboard overdubs with Froom. "That was rewarding," he enthuses. "It's hard to say to a musician, 'That was great, but would you try something else?' when you don't have a better idea of your own, but Mitchell never runs out of ideas!

"The great thing about working with synthesizers was that we could bring them right into the control room and get response immediately," he continues. "Everything was right there — the instrument, the EQ on the board, and the effects. If we knew there were going to be specific delays, we'd put them on, and where we were going to layer frequencies, we'd EQ as we recorded. The more we did on the way in, the less we had to reconstruct on the way out."

Davey Pattison's vocals were recorded with an AKG C452. Montrose monitored — but didn't record — through a Lexicon 224 Digital Reverberator. "When you're going through take after take with a dry vocal, it really starts to sound dead," he explains. He also worked very closely with Pattison on melody, phrasing and inflection: "Davey is basically a blues singer, and he enjoyed doing it this way because it gave him a chance to adhere to a melody rather than relying on blues phrasing," Montrose notes. "On a lot of things, the melody was written out.

"There's a texture, a timbre — a basic believability — that's recognizable in a

blues-rock singer, as opposed to a pop-rock singer. It's just a question of depth as far as I'm concerned," he says.

Engineer/Producer Relationship

On *Gamma 3*, Ken Kessie was the engineer for the basics, working on most tracks with assistant engineer Wayne Lewis, while Jim Gaines handled most of the overdubs and the mixing. Most of the recording was done in Automatt Studio "A," with some overdubs in "B."

"I was involved on all levels with Mitchell, working with the synthesizer and the board, but when it came to guitars I needed a different kind of objectivity. I got completely into my guitar playing, and if I started getting edgy Jim knew how to play it by feel, keeping with the mood; especially when I was trying to get something really 'fiery' down on the guitar. He gave me that objective ear, but he wasn't giving unnecessary opinions as much as he was just giving me support."

In monitoring and mixing, says Montrose, "I've learned that it's very important to use different speakers — but to *know* your speakers. Gary Lyons recommended ACD/John Meyer speakers, and I was completely blown away, and have insisted on using them ever since. I also have some very small Philips self-powered speakers. They hype me out — I know that — but I know exactly what I'm hearing from those speakers at all times. We mixed on the Meyers, the Philips and, once in a while, on Auratones."

"I'm proud of *Gamma 3*, says Montrose. "I removed myself and looked at the band — myself included — as players and participants in this ensemble. My job was to take what these five people had to offer, and direct as best I could the making of a successful album without compromising the essence of what they have. *That* to me is the job of a producer."

Montrose's managers are sufficiently impressed with his performance behind the board on *Gamma 3* that they are actively seeking outside production assignments for him. "A lot of unsigned bands — from around San Francisco and from elsewhere — have asked me to produce them, but I'm involved in so many types of projects that I have to pick and choose as time will allow.

"It's very satisfying to me to go into a studio with a group, start with the raw materials and finish with a great record. It's like having your own playground to work in, especially when there's a lot of talent involved." ■■■

news

— continued from page 15 . . .

HARRISON CHANGING INTERNATIONAL MARKETING STRUCTURE

As of August, direct factory representation is being provided through Harrison dealers in all export markets outside the United States and Canada.

"This change of marketing structure, which replaces our former method of export marketing through an exclusive export distributor, is consistent with our overall direct-marketing strategy, which has already been implemented in North America during the past year," said Claude Hill, vice-president of marketing at Harrison.

"Changes in the world market and economic situation have made it desireable to change and expand our export marketing organization. We are maintaining and strengthening our dealership arrangements with our existing export dealers. In addition, we are actively seeking out new representatives in areas where we are not now represented for our full range of broadcast, film-sound, and music recording consoles."

AUDIO KINETICS OPENS DEMO STUDIO

According to Steve Waldman, president of Audio Kinetics, "This new demonstration facility will enable us to provide editors, mixers, engineers and prospective owners with hands-on experience with the Q-Lock in a realistic environment. Even though the Q-Lock is currently being used at more than 150 major facilities worldwide, the whole concept of electronic sound editing and sweetening is still relatively new. By providing members of our industry with an opportunity to sit down and work the equipment, we'll allow them to discover that the Q-Lock doesn't change the essence of what they do, but will allow them to do it easier, faster and more accurately — in film and video — then they've ever been able to do it before."

. . . continued on page 185 —

SONY DIGITAL MULTITRACK DEBUTS AT RECORD PLANT

To determine sound quality and ease of operation, the PCM-3324 24-channel machine was used to record various types of music, ranging from acoustic instruments to jazz sessions and hard rock and roll. According to Michael Stone, Record Plant's chief engineer, "We've gone through all the functions, given it a thorough testing, and it sounds very good. I like the idea of getting away from the noise and distortions of analog recording, and the Sony PCM-3324 is the best multitrack on the market. The low-end is much better than analog — it's solid and clean.

"The editing is great as well. I edited with a razor blade as I would in analog, and every edit came out perfectly. There were no special digital problems at all —it's a huge step in the right direction."

"Electric rock 'n' roll is the hardest test for digital, and the clarity and punch were excellent," Stone continued.

Grammy Award-winning engineer/producer Bones Howe was on hand for the demonstration, and commented, "This is a machine that really fits into the standard recording environment. As soon as the digital Compact Disk is here, and the public can really hear what this technology means, the major industry moves to digital will begin to accelerate."

ROAD 80 RE-EQUIPPING WITH MITSUBISHI DIGITAL MACHINE

Tom Jung, New York-based engineer/producer and owner of Road 80, Inc., recently acquired a Mitsubishi X-80 two-channel digital audio recorder. Jung is said to have made the decision to purchase the unit after renting an X-80 from Mitsubishi for numerous recording projects over the past year.

The Mitsubishi X-80 (portable) and X-80A (console) machines are claimed to be the only digital recorders to offer the choice of either razor-blade, or automatic electronic editing. Jung's recorder is reported to be the first one in the US that is compatible with the standard sampling frequency of 48 kHz, proposed by the AES last year, and agreed to by Mitsubishi Electric early in 1982.

SOUNDSTREAM TO DEMO DIGITAL EDITING SYSTEM AT AES CONVENTION

Using a remote link with the Instant Access Editing System in Hollywood, Soundstream editors Jim Wolvington and Tom MacCluskey will be demonstrating the unit's editing capabilities, variable cross-fades, plus instant access to any location within the computer's storage.

Also being demonstrated at the Anaheim convention will be the company's new SMPTE synchronization capabilities, as well as the playing of selections from among their over 200 digital master recordings. Many AES participants will have the opportunity to observe firsthand Soundstream's digital editing system which, the company claims, is faster and more powerful than any other commercially available digital audio editing system in the world.

GEORGE DUKE PURCHASES MITSUBISHI X-80 DIGITAL RECORDER

Duke has used the Mitsubishi system for a number of recording projects over the past two years — most notably on his collaboration with bassist Stanley Clarke on *The Clarke-Duke Project*. The X-80 two-channel digital recorder was obtained from Mitsubishi Electric Sales America for over a year on a rental basis, and Duke recently decided to purchase one for his private studio. The system includes the firm's DDL-1 Digital Delay Unit. ■■■

views

THE AUDIO/VIDEO MARRIAGE

Combining State-of-the-Art Audio With In-House Video Facilities — A Case Example of Bullet Recording, Nashville

The story of Nashville's ill-fated Four Star Studio has circulated up and down the city's Music Row for years. The modern office tower was completed in 1974, but the grand studio that inspired it ran aground. First, there were financial problems. Then somebody was killed by a fall from the high atrium balcony. Murder charges were filed, and although the accused was eventually aquitted, the building's proposed sale got tied up in the legalities. The dream of Four Star's owner, Joe Johnson, to have a full-fledged audio/video studio — a bold idea for the early Seventies — began to fade.

The idea didn't fade completely, however. Seven years later, Randy Holland heard about the studio and decided to take the dream out of mothballs. This article relates how the supposed white elephant was transformed into what is rapidly earning a reputation as one of the country's most advanced audio/video studio complexes.

As Randy Holland recalls, "The Bullet idea goes back so far that it is hard to say when it really began. I came to Nashville in 1978 to attend the Music Business School at Belmont College, and a friend of mine helped me to get started as a freelance engineer. Things were going well, but I really wanted a place to work on my own projects. About that time I met Scott Hendricks, now Bullet's chief mixing engineer, and together we built the 'Lower Level' — which literally was the entire lower level (and then some) of my split-level house.

"As business picked up, I found it harder and harder to get in [to the studio] and work on my own projects. I was having to go to other studios, which I didn't like. When it began, I had a recording studio at my house; before it ended, I lived in a recording studio — a very frustrating paradox!

"Also, I was already aware that video was beginning to play an increasingly important part in the record business and, while discussing this with Scott, he mentioned that he knew of a partially finished audio/video facility, and although what is now our Studio B had been in business for some time as Richey House, and later as Island Recording, the big rooom had never been furnished."

Holland and Hendricks subsequently visited the studio, looked it over, and decided it was suited to what they had in mind for a state-of-the-art audio/video facility. With the pieces rapidly falling into place, it became time to "put up or shut up." Holland first contacted his father, an otolaryngologist (ear, nose and throat specialist) with a keen interest in psychoacoustics, and who had put the money up for the "Lower Level" project. He was emphatically interested. Within six weeks the paperwork was handled, and it was a case of all systems go.

As they proceeded, however, the team found that much of what had been built was of inferior construction. Holland knew early on that they would have to tear it down to the bare walls.

Before Holland committed to basing his operation in Nashville, some thought was given to either the East or West Coast, since he knew this would give the studio more exposure — at least in the short run. The move was ruled out, however, for several reasons.

Foremost, Holland considered, there would be an increasing demand for music video and related audio/video sessions. What was needed, therefore, was a facility that offered not only high-quality audio recording capabilities, but one that also could provide sufficient space and facilities for a full-blown video shoot. Nashville, he began to realize, was steadily becoming more of a universal recording center and, as such, the city would serve as an ideal location for the new studio.

Design Considerations

"From the beginning," he recalls, "we were determined to build a no-holds-barred showcase facility — not because we were trying to prove anything, but

views

• AUDIO/VIDEO MARRIAGE •

because we personally wanted a 'Rolls Royce' to work in, and knew that others would want to as well."

Holland was quick to realize that it would be much harder to compete in the marketplace if he built a run-of-the-mill studio with average equipment. For that reason, 48-track audio with intelligent automation, and one-inch video with on-line computer editing, were the non-negotiable guidelines from which to proceed.

While some of the original 1974 construction had its faults, Joe Johnson's basic concept for the facility was deemed to be fundamentally sound. The original architect, Jack Edwards, was contacted in Los Angeles. While on a visit to LA, Holland met with another person in Edward's firm, Sven Len, and told him what was needed — a room completely different from all the others in Nashville.

"First of all, I wanted a room that could accurately reproduce 40 Hz — none in town can without active equalization, to my knowledge," Holland stresses. "It had to be a large, live room with a high ceiling and low monitors with [Altec] 604-8G drivers."

However, the design brief that Holland specified was not without its challenges. "It was very difficult," he concedes, "because we were working with some unusual ideas." For example, although the audio facility and the video facility were both functionally sound, and independently complete on their own, they had to work *collectively* as one unit too. The interfacing requirements were extensive. "We had to run wire between the two rooms that *had* to be acoustically isolated from each other."

About 15 feet had to be added to the floor between the audio and video control rooms, extending it toward the studio.

Joe English video shoot in Studio "A" . . .

"The earlier plan had not included isolating that floor," Holland continues, "so we kinetically isolated it using half-inch-thick machine rubber in 16-inch square centers, sandwiched between concrete slabs, and decoupled from the walls. Adding the extra feet gave us the depth we needed in the pair of control rooms and, by turning the audio control room sideways to the main studio area, we created a side studio where overdubs can take place.

Acoustic Separation

"Due to the need for video and audio to be able to work separately on simultaneous projects, we had to achieve a great deal of acoustic isolation between the audio control room with its adjoining overdub room, and the video control room above, with the main room now functioning as a shooting studio."

Anticipating this problem, Holland investigated whether the floor to the studio was of structural importance to the building. Luckily, the building had been built only on its vertical pillars and, in fact, the studio floor wasn't even poured until after the building was up. tall, 180-linear-feet cycloramas, which he line of where the control room and isolation room would be built, straight down to the dirt. By doing that, the control room/isolation room area was completely decoupled from the rest of the complex. To further improve isolation, all the glass between the control rooms is ¾-inches on the control room side, and ½-inch on the studio side.

"The sound level drop between the control room/isolation room area and the rest of the world is such that if a producer is monitoring an overdub or mix at 106 dB, we could still carry on with a video shoot with absolutely *no* knowledge of his existence," Holland points out.

The floor extension also provided a balcony that has several uses. It not only serves as an excellent camera platform that allows an operator to pan all the way across the back of the studio, but also functions as a balcony for live audience seating, as well as a conductor's platform.

"I wanted the main studio to be very live," Holland continues. "The ceiling is at 26 feet and the decay time at 1 kHz is almost a full second. The rock and roll sessions we've done here have sounded absolutely sensational, as have large orchestral dates."

A video cyclorama, which provides different backgrounds during video shoots, also forms an integral part of Bullet's acoustic tuning capability. On one side of the studio there is a floor-to-ceiling pocket that stores three 18-foot tall, 180-linear-feet cycloramas, which pull out and circle 180 degrees around the studio. Three cyclorama curtains are available: a black velour used to produce a "cosmic" effect, and which completely blacks out everything behind the subject; a blue chromakey; and a light-weight, white shark's tooth weave scrim which, Holland says, "works great when trying to splash a lot of light on [video] tape. The problem with the white cyc [cyclorama] is that it takes very little light to get up to the 100 units threshold of maximum level for broadcast on tape. With use of the shark's tooth, we can pull the black velour out, and then the white scrim in front of it. When lit from the front, it comes out a very 'cool' white — actually a pale gray — which will take a lot more light before reaching maximum threshold. The blue chroma allows the added dimension of superimposition — the old trick where the subject is standing on a cloud, or out in space.

"The curtains can be drawn separately or together. If the room needs to be not quite so live or completely dead, it can be accomplished with the use of the curtains."

"When we got down to construction," Holland continues, "I was glad I had Scott [Hendricks] on my crew; he has a degree in architectural acoustics, and was able to keep a sharp eye on every detail, making sure everything was put together correctly. By this stage, I also had Bullet's present studio manager — and my right (and left) hand man — Piers Plaskitt on the team."

Wiring and Power

All of the wiring and interfacing at Bullet was carried out by in-house personnel. "One of the reasons the interfacing between audio and video areas was so taxing," Holland recalls, "was because there were no examples to follow. Sure, there are video studios with simple audio in them, and recording studios with video playback and/or interlock, but none that I'm aware of could come close to the sophistication in hardware and design we were after.

"There's nothing worse than having to tear down a wall to put in one more shielded pair, so Bullet staff made sure everything was taken into account, and that gaining access for adding wires could be done easily. The troughs are basically divided into three sections to minimize crosstalk: one is audio line/mike level; the second is audio cue/speaker; and the third, video. All three groups were kept six inches apart, and one can access them virtually anywhere along their path via specially designed hatches, which are hidden when not needed."

With all the heavy demands for power in such a complex facility — especially video lighting — a great deal of attention had to be paid to the electricity supply. So much power was required, in fact, that Nashville Electric Service had to tap into another main line several blocks away. Although the existing six-story building already had 2,600 amps of service, another 2,400 amps needed to be added for Bullet's new studio.

Two separate AC power systems are provided: one for "house" power (lights, office machines, etc.); and another for "technical" power (consoles, tape machines, etc.) The technical power is filtered, regulated and transformer-isolated, and grounded through three,

Reflections from Control Room Window

one-inch copper rods buried in rock salt. (The latter ensures that no "pops" or "hums" are induced into the system, and also lowers the studio's noise floor.)

Audio Equipment

When the time came to select the audio equipment for Bullet, Holland recalls, "some decisions were very easy — and some took a lot of research.

"First, the easy ones. There was no problem with the tape machines: Studers are simply the best. We have two A-800 24-track units, and they've proved a joy to work with. The A-800's ability to read SMPTE timecode in rewind and fast-forward was *essential*; we also liked the accessibility for maintenance.

"We also purchased A-80 half-inch machines, which have simply amazing performance characteristics. Recently, out of curiosity, we took a 24-track master recorded at 30 IPS with no noise-reduction, and mixed it to three different machines: quarter-inch Studer two-track running at 30 IPS; the half-inch at 30 IPS without Dolby; and a digital recording unit that shall remain anonymous. We played back all four tapes in sync. Although you could hear some noise on the quarter-inch machine, it was *very* hard to tell the difference between the 24-track and the other two mixdown machines. And no one in the room — including the digital rep — could reliably pinpoint a difference between the Studer half-inch and the digital.

"So, when people ask me about digital, I simply say, 'Digital isn't ready for us yet.' It's a promising new technology, but we can't justify it yet in terms of what it can offer our clients. There are little things that 'Digits' [Holland's term for digital design engineers] don't understand, like turning a two-inch tape over, and recording echo backwards. The fact is, my clients *do* things like that; they use the tape machine to manipulate sounds as only an analog machine can do — and so do I.

"Ordering the console was a big thrill for me, because we weren't sure we were going to work it into our budget. Once we did, we knew we really had it."

Holland decided on a Solid State Logic board, because he considered it to be the "most advanced console in the world. Think of everything you always wanted on a console, and SSL has already put it there — along with some other things you didn't think of, but which makes a lot of sense. I'm also a short-signal-path freak, and yes, the SSL has a very short signal path, believe it or not.

"The 'Total Recall System' [automation] function is *essential* on a console this sophisticated; its ability to store all the settings of EQ, echo, cue, noise gates, limiters and routing saves hours over the course of a long complex project."

SSL/Musicworks has been very supportive, "and has gone out of its way to help," Holland says, "even with matters not related to the console. Their engineer, Grey Ingram, was of invaluable assistance in the video area. His experience with audio/video shoots gave us important input, and resulted in some design modifications, such as an RGB-to-NTSC video switcher built into the console to show us program and preview from the video control during a shoot."

Since Bullet has two Studer multitracks and numerous VTRs, an Audio Kinetics Q-lock 310 synchronizer was chosen because of the flexibility it offers when handling three audio and/or video transports. When it came to outboard gear, Holland considers Bullet to be "fully-stocked." The studio boasts EMT and Lexicon digital reverb, DDLs, EMT plates, and even four live chambers. "Choosing video equipment took some hard-nosed compromises," he says, "since there is a great deal of good equipment on the market. The cameras are three Sony BVP 330As, all with diode guns. Of the seven one-inch Sony VTRs, five have time-base correctors and dynamic tracking, and two are portables. Switching is by a 12-input Crosspoint Latch 6124, reference color video monitors being Ikegami, and current editing capability a CMX 'Edge' computer system and a 3M switcher."

Bullet has total flexibility, Holland considers. "Everything needed to turn an idea into a finished product, from storyboard concepts and piano-vocal demos, to off- and on-line [CMX] computer editing, and 48-track automated mixdown. Having the video production room right above the audio control room enables the client to have all the creative people involved right on the spot. It brings back the possibility for real-time spontaneity; something that is too often lost when a product is assembled over a long period of time."

Towards the Future

Because Bullet has only been open for

. . . SSL console in audio control-room "A"

views

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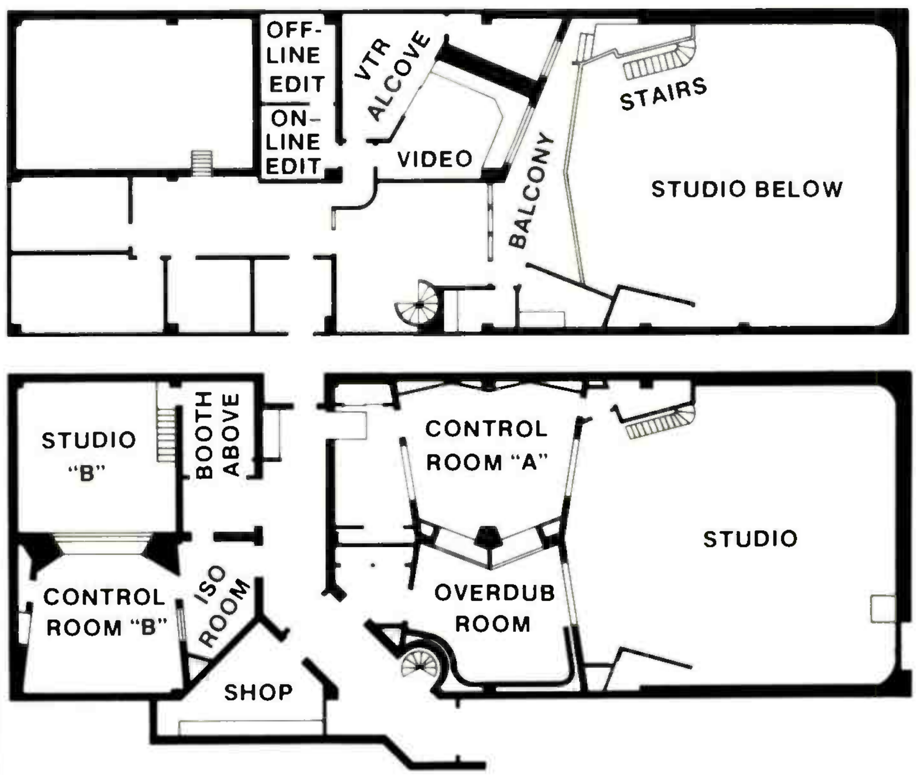
business since January 1982, long-range predictions about its success would be a touch premature. But there are two good omens, Holland says. First, in its record operations, around 40% of Bullet's clientele comes from New York and Los Angeles — about double the predicted business. And second, he says, clients have been "overwhelmed with the accuracy of what they are hearing in our room. To date, about 90% of product out of here has been mastered flat."

Later this year some significant new developments are expected at Bullet. The facility will be switching from a "hardware rental company," to an active company emphasizing diversified product and ideas, and relying on the talents of its in-house staff. Bullet's record division is currently setting up projects for both audio and video, and its advertising division, Holland offers, will be producing videos for training, teaching, and marketing purposes. Plant expansion plans currently call for the addition of a second video editing suite in the Fall.

Operationally, Bullet plans to handle some live FM broadcasts, and later in the year expects the video dimension to be added via either cable or network television linkups.

"Some people say Bullet was a $3 million shot in the dark," Randy Holland concludes, "but we're fully committed to the success of the audio/video marriage. With the consumer digital audio disk and stereo TV on the horizon, we expect to prosper along with that marriage." ■



news

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. . . continued on page 190 —

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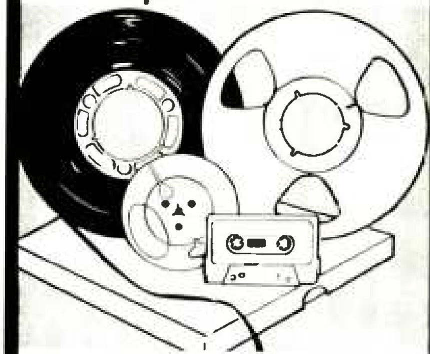

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news

— continued from page 185 . . .

Timilon's chairman of the board and chief executive officer, Glenn Epple, states that long-term leases are currently being negotiated with several major film and television producers, with a substantial percentage of the facility already committed. At present, 20 film and video soundstages along with six audio/dubbing studios are being engineered for the first phase of construction, which begins this Fall.

Also included in the initial construction, scheduled for completion within 18 months, are an office and support complex, satellite transmission and reception facility, special effects department, film processing lab, screening rooms, post-production rooms and editing suites, and complete living accommodations and recreation facilities.

A portion of the property has been set aside for a research and development park. Facilities will be available to evaluate the latest developments in digital audio and video recording, high-definition television, and advanced film technologies. And, the production center is expected to be the catalyst for many innovations in the near future.

The entire complex will be directed towards a very special clientele of producers, directors, artists, technicians, record labels and especially major film studios, offering them an environment in which they can work, live, and relax, without interruption. The facility will offer them the privacy, seclusion, security and atmosphere considered essential to a professional and creative process.

MCI/SONY SELECTED FOR SOVIET ON-LOCATION TAPING

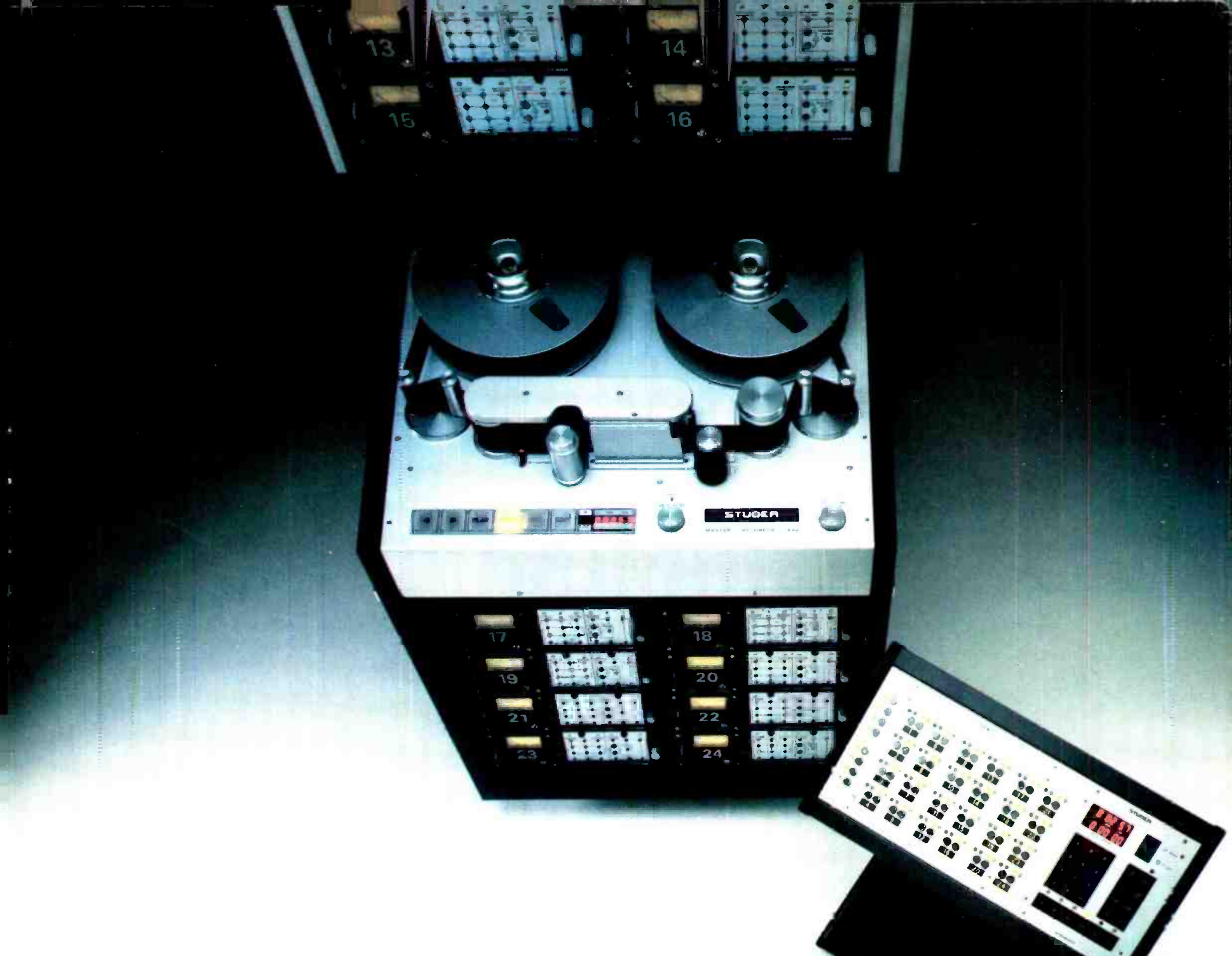
Melodiya, the Soviet state recording company, has acquired an MCI-equipped 32-foot remote van for the broadcast and recording of popular, classical and ethnic music.

"We came to Moscow for an exhibit and demonstration," said Lutz H. Meyer, MCI/Sony's vice president of marketing, "and the Soviet engineers did not even let the unit go home. They negotiated the sale on the spot."

The acoustically-treated van was designed and built in Great Britain by Clyde Electronics, Ltd. in conjunction with MCI/Sony. It is equipped with an MCI 24-track and multitrack console, in addition to two MCI stereo mixdown machines.

PEOPLE ON THE MOVE . . .

- **Hans Batschelet** has been appointed vice president for marketing at Studer Revox America, with primary responsibility for the marketing of Studer professional recording and broadcast audio products in the US.
- **Tom Behrens**, previously head of technical services at Valley People, Inc., is now in charge of national sales, according to Norman Baker, president. In the newly created position, Behrens will be responsible for sales of all Valley People products through their dealer network, and directly to OEM users in both the United States and Canada.
- **Michael Hurt** has been appointed as the Harrison Systems' factory export marketing representative, and will handle all matters related to export dealer relations and special engineering requirements. Also, **Dave Purple**, former sales manager at Harrison, has rejoined the company as sales and marketing managet for broadcast sales.
- At MXR, **Debra Alley** has been appointed marketing manager; **Peter Beverage**, director of sales, will be relocating to MXR's new Mid-Atlantic sales office in Berling, New Jersey; and **Michael Klickstein** has been appointed musical and professional products representative in California.
- **Charles W. Gushwa** has been appointed marketing manager for Crown International, with responsibility for professional and industrial divisions of the company.
- At Sound Technology, **Sonny Funke** has been appointed Pro-Audio and Broadcast Representative for the states of California and Arizona. Also, **Robert H. Millice** has been added to the company's marketing staff, with responsibility for sales management of the Western one-third of the US, as well as coordinating general sales and marketing activities on a National level.



Studer's Secret of Success

In years past, the Studer A80VU has carried widespread acceptance by the world's premier recording studios. And this success story is far from over: top studios continue to choose the A80VU MKII over other "all new" machines. The secret of this success lies in three basic rules:

1. If it can't be made better, don't change it.
2. If improvements can be made, make them – even if they don't show on the outside.
3. Use longer production runs to hold down final cost.

Now in its third generation, the Studer A80VU series once again offers state-of-the-art performance at a surprisingly modest price. The new A80VU MKIII system incorporates several significant improvements, including:

- **Transformerless Line Amps:** Low output impedance assures optimum performance even with long cable runs.
- **New Headblock:** Tight spacing of erase and record heads permits exceptionally accurate punch-in and punch-out.
- **Remote Unit:** Full channel remote with 20-address memory autolocator.

Instead of repackaging these changes in an "all new" machine, Studer kept the basic transport – a design with an unprecedented reputation for reliability. Also, because basic tooling costs have long since been amortized, the A80VU MKIII's price has been held down, thus offering a better price/performance ratio.

How much better? Call your nearest Studer representative for details. He'll help make our secret the key to your success.

STUDER REVOX

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