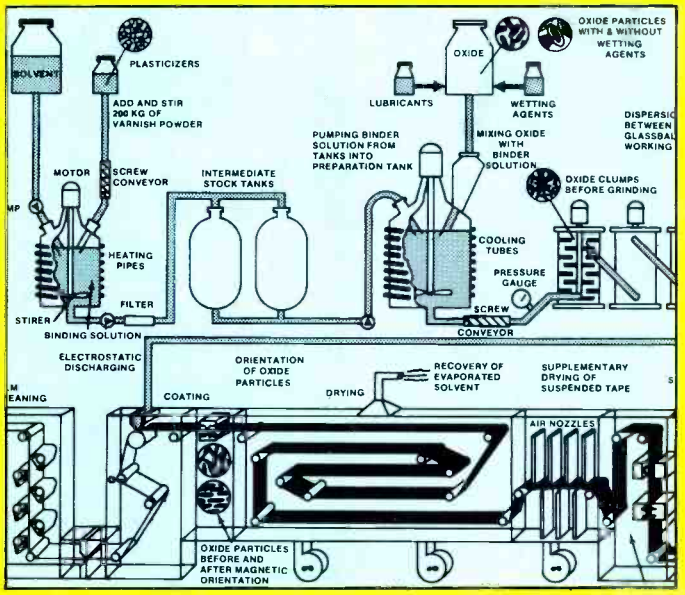
RECORDING ENGINEER/PRODUCER

$3.00
April 1983
Volume 14 — Number 2

"ON THE ROAD AGAIN" — Willie Nelson & Family Tour — page 48

Film Sound — page 64



Tape Manufacturing — page 43

PRODUCING AUDIO FOR • TAPE • RECORDS • FILM • LIVE PERFORMANCE • VIDEO & BROADCAST

ANGELA ...NOT JUST ANOTHER PRETTY FACE

When we created **ANGELA** we knew she had to have more than just good looks. We insisted she be the finest low cost 24 track console in the world. For her to achieve this, we had to maintain total integrity in all aspects of her design and construction. We refused to give her inferior components like PC board patch bays, ribbon connectors or an inexpensive chassis that flexes. Instead, we built her with the same superior design and manufacturing techniques found in our M2500 and M3000 Super Consoles . . . and is **ANGELA** built! She has Mil-Spec Patch Bays, a Hard-Buss Mother Board, Gold Pin Edge Connectors, all her amplifiers are 5532-4s and TLO 74s and she has a Perfectly Rigid Carbon Steel Chassis. Sound Appealing? To find out more about **ANGELA** or to arrange a personal demonstration, contact us at (213) 704-7528 for the name of your nearest **AMEK** dealer.

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For additional information circle #1

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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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April 1983
Volume 14 — Number 2

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THE CHOICE OF PROFESSIONALS.

When it comes to recording/remixing consoles, survey after international survey shows MCI ranking as the undisputed number one choice of the world's leading studios. Surprised? You shouldn't be. Feature for feature, Sony's best selling MCI JH-600 Series beats out every other comparably priced console on the market. And for performance and reliability, the JH-600 Series can stand up to any console at any price.

Let other manufacturers take credit for the "most expensive" console and make outlandish performance claims that have nothing to do with real world situations.

While they've been making the claims, we've been making the hits.

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SONY®

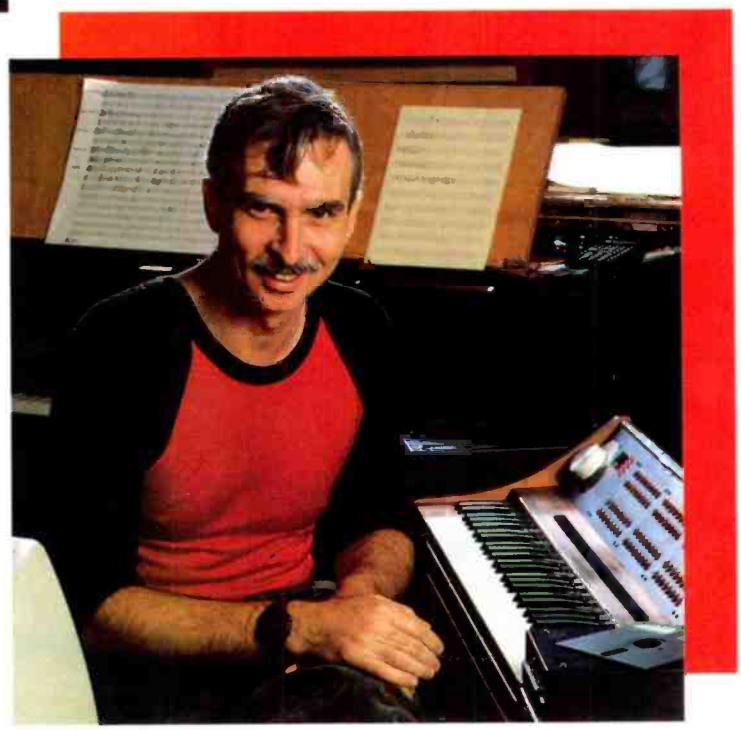
YES . . . THEY ALL
BUT LISTEN
SYNCLAVIER® II

TOM MOODY

Creative Director, Executive Producer, N.B.C. Television, Los Angeles

Well, it's hard to talk about any one aspect of the Synclavier II because everything about this instrument is so unbelievable; but if I had to boast about one area of its capability, it would be its speed and ease of use for composing new ideas. I find my scratch tracks suddenly become finished tracks, right out of the Synclavier II's 16 track digital sequencer to tape. This feature alone has increased my production output at least 300% in the last two years. Plus, the additional ideas you receive from the great number of Synclavier II owners who trade programs and innovative ways to use the system always keep the creative juices flowing.

When you combine those musical features with the fact that the instrument, through new software and options, keeps expanding and getting better, what I first thought was a major expense has paid for itself many times over.



PAT GLEESON

Synthesist/Producer, San Francisco

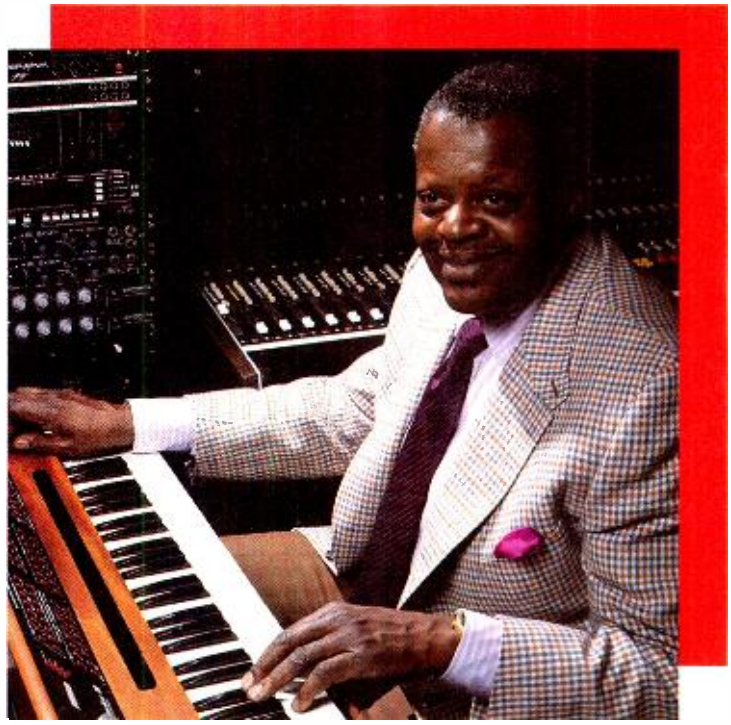
I used to own $85,000 worth of analog synthesizers, but I sold them all to buy one $40,000 Synclavier II which, by the way, paid for itself on my first Synclavier II project. The director of that particular film wanted to hear the music before committing it to tape, so how else could I do that except by using the Synclavier II's 16-track digital recorder.

Even more important is the attitude of the company. Over all of my years in the music business, N.E.D. is the first company that sincerely pays attention to its users. In fact, the great thing about the Synclavier II is that it's better today than when I purchased it, due to user input and software updates. I've heard about people who don't own one saying that in a year the Synclavier II will be half the price. The funny thing is, I've been hearing that for three years now.

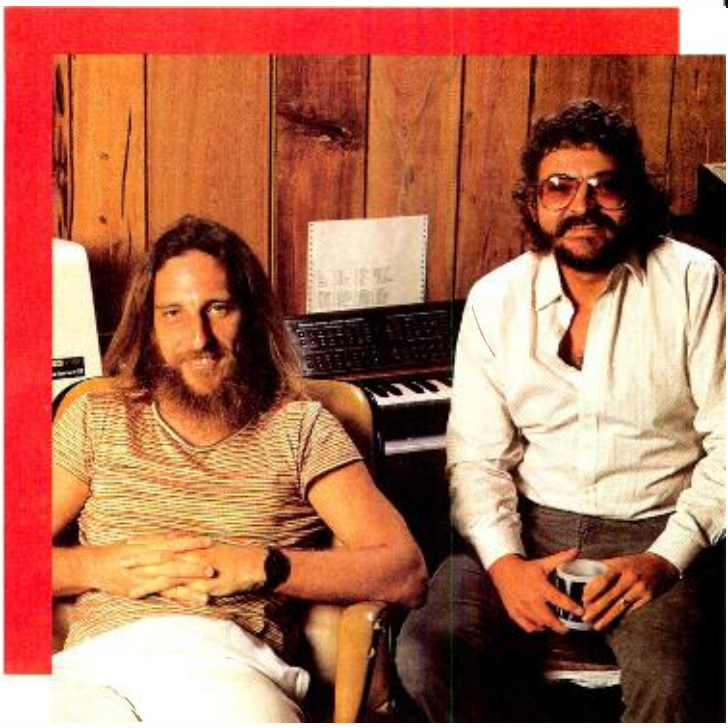
HESITATED AT FIRST, TO THESE OWNERS NOW!

OSCAR PETERSON

Toronto



A renaissance in music is happening right before our eyes and ears; and the Synclavier II is leading the way. This instrument saves many of my spontaneous compositions which before were lost. Then, to be able to automatically print out the music is just short of a godsend for me. What used to take two hours of transcribing music, now takes five minutes. Also, if there is a small feature which you think would improve the useability and musicality of the program, just inform the company and eventually you will have it. The printing feature gets better all the time. It sure is a pleasure to learn one instrument and grow with it. I have been more than satisfied with my investment and convinced it is one that will last as long as my first love, the piano.



ALBHY GALUTEN
KARL RICHARDSON

Karlbhy Productions, Miami (Producer/Engineer for Bee Gees, Barbra Streisand, Dionne Warwick and many others.)

I think we purchased the first or second Synclavier II built back in 1980 and without a doubt, it has been the best investment we've ever made for any computerized musical instrument. Recently we added the Sample-to-Disk™ option to the Synclavier II, which has opened up an immense amount of musical ideas while also saving valuable production time. For example, we were recording an artist and we wanted to change where the person sang and also correct a slight intonation problem. Instead of re-recording or splicing, we just sampled the performance to the Winchester of the Sample-to-Disk™. We then rolled back the tape, and played the phrase where we wanted to and corrected the intonation problem with the Synclavier II's pitch wheel. We also use it to precisely analyze the meter of any music. Of all the choices available to us, we're glad we choose the Synclavier II because we feel it's going to be the standard for every studio.

New England Digital Corporation
Box 546 M
White River Junction, Vermont 05001 U.S.A. 802/295-5800

Synclavier II is a registered trademark of New England Digital Corporation

For additional information circle #4

DO SOMETHING GOOD FOR YOURSELF

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

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

The 8X is an excellent choice for the small studio in need of upgrading performance or expanding format. For the large studio the 8X is an ideal system for your Studio B or 24 track mixdown room. Because it is compact the 3X is also ideally suited to video and remote recording applications.

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

AUDIOARTS® ENGINEERING
A DIVISION OF WHEATSTONE CORPORATION
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NEW ENGLAND DIGITAL THANKS ITS CUSTOMERS– PAST...AND POTENTIAL.

For the last two years, we've been working to build the type of instrument that would inspire satisfied owners to tell potential customers just what they're missing. Yes, we're very pleased with the response of our owners. In fact, at New England Digital, we like to think of them more as family than just customers. Although we've been saying that an investment in a Synclavier II is an investment that increases in value year after year, it's still nice to hear our customers express the same feelings. So, as we enter 1983, we would like to extend our appreciation to the nearly 200 Synclavier owners worldwide. The financial support and musical ideas of these visionaries have been essential to the success that the Synclavier II enjoys today. Ask any of them: you'll find we're a high technology company that places a high emphasis on people.

So, potential customers, be assured that New England Digital will not rest on its laurels. This year we plan to revolutionize the already revolutionary Synclavier II with exciting enhancements for Music Printing, and incredible new options: Digital Guitar, Stereo, and more.

So quit hesitating . . . talk to a Synclavier II owner or, better yet, visit New England Digital or one of our local distributors for a personal demonstration.

Brad

Bradley J. Naples
Director of Marketing and Sales

Synclavier II Instruction Manual
A complete and descriptive Instruction Manual is available for $85 (USA & Canada) and $100 US (elsewhere).

Synclavier II digital music systems start at $14,150.

Please write or call
New England Digital Corporation
Box 546M
White River Junction, Vermont 05001 U.S.A.
802/295-5800

New England Digital's only authorized distributors

New York: Digital Sound Inc 212-977-4510
Los Angeles: D.J.C.S. 213-274-8512
London: Turnkey 440-9221
Brussels: Trans European Music 569-1823

 Synclavier II s a registered trademark of New England Digital Corporation
For additional information circle #6

new england digital

views letters news

FAIRLIGHT CMI SYNTHESIZER

from: **Frank Serafine**
Serafine FX Studio
Los Angeles, CA

In reply to the letter from David Bross of Fairlight Instruments USA, printed in the February 1983 issue of *R-e/p*.

When discussing the Fairlight and Emu systems, I clearly stated that their functionality was being compared on a consumer level. That I dared to make this comparison certainly seems to have upset the Fairlight company.

Surely, one of the primary concerns of an informed consumer in the open marketplace is getting a quality product for an affordable and reasonable price. I feel justified in passing on my experience with the Fairlight, since I was given full use of the CMI for nine months while creating sound effects design and synthesis for the motion picture, *Tron*.

Thus I am quite aware of the "total functionality" of the Fairlight, and realize its position as the early forerunner of digital sampling machines. Although the CMI has made a major contribution in the area of electronic sound synthesis, the rapid growth and interfacing of this new technology is currently bringing about an increasingly flexible and competitive marketplace.

I look forward to keeping *R-e/p* readers informed on the innovations and applications of this future product.

OP-AMP CIRCUIT DESIGN

from: **Walter G. Jung**
Forest Hill, MA

I have been reading the latest article by Ethan Winer ["A Practical Approach to Circuit Design," February 1983 issue], and find it to be a generally useful and practical piece. I could not agree with him more that writings which illustrate by the use of practical examples are much more useful than purely theoretical ones.

R-e/p readers might find it of further use to know that there are (and have been) quite a few books and articles on the subjects Winer is addressing. For example, my *IC Op Amp Cookbook* is now in its second edition, as is the companion work, *Audio IC Op Amp Applications*. These books are available from either the publisher (Howard W. Sams), or at local libraries maintaining technical sections. The second book goes into quite some detail on IC op-amp distortion in audio circuits, and includes an extensive bibliography on the subject. This is, of course, in addition to the many working circuits, in keeping with the "Cookbook" format.

Keep up the good work in getting useful information into the hands of *R-e/p* readers. But let's also help them further by pointing out already existing and widely accepted work on the subject, where available.

INFRASTRUCTURE RIVALRY

from: **Richard McGrew, president**
A&R Record Manufacturing
Dallas, Texas

To the industry: I am writing to clarify and rectify an untrue and unfortunate rumor concerning my business, A&R Record Manufacturing Corp., which many of you may have heard in the past several weeks.

A letter was circulated in March by Disc Mastering, Inc., of Nashville, TN, which stated that A&R has recently closed.

This statement is not true.

A&R is open for business, and is in sound financial condition.

If you have heard this rumor, please disregard it. Actions have been taken by A&R Record Manufacturing Corp. to be certain that this rumor ceases.

We are ready, as always, to provide the quality services to you which we have provided in the past. I look forward to serving you in the future.

TUBE DIRECT BOX

from: **James Demeter, president**
Innovative Audio Systems
Santa Monica, CA

We would like to clear up some potential confusion in the article, "The Cutting Edge of Country," published in the December 1982 issue of *R-e/p*, regarding the Red Box mentioned by Bradley Hartman. We sold the Red Box, now called the Tube Direct Box, to David Cowert of Enactron, back in May of 1981. Since then our clients include Yamaha Research and Development Studios, David Lindley, Ry Cooder,

The Needle Knows

Even when your ears can't tell the difference, your VU meters can.

Which is why we test every reel of our 2" Grand Master® 456 Studio Mastering Tape end-to-end and edge-to-edge. To make certain you get a rock-solid readout with virtually no tape-induced level variations from one reel of 456 to another or within a single reel.

No other brand of tape undergoes such rigorous testing. And as a result no other brand offers you the reliable consistency of Ampex Tape.

A consistency that lets you forget the tape and concentrate on the job.

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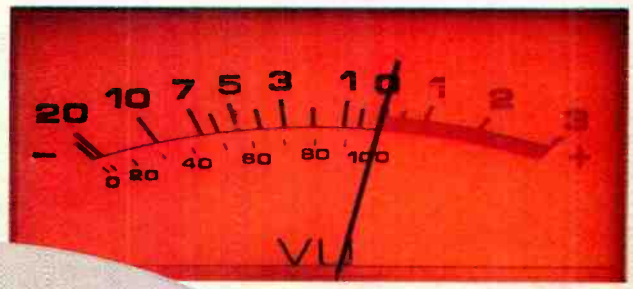


AMPEX

Ampex Corporation • One of The Signal Companies

4 out of 5 Professionals Master on Ampex Tape.*

*1981-1982 Billboard Magazine Brand Usage Survey

For additional information circle #8

David Landee, and Edward Van Halen.

Bradley Hartman was pleased with the sound of our direct box because of its extremely high input impedance — greater than 20 megohms. The Tube Direct Box does not alter the tone of the input like normal passive direct boxes; instead it samples the signal without changing it. The headroom is 40V peak-to-peak, with a signal-to-noise ratio greater than 85 dB. And, at 40V peak-to-peak, it has greater headroom than virtually all other active direct boxes. The Tube Direct Box uses a Jensen output transformer, polypropylene capacitors, and metal film resistors, all of which are housed in a virtually indestructible box.

The Tube Direct Box is manufactured and can be obtained on a special-order basis direct from Innovative Audio Systems, 2210 Wilshire Blvd., Suite 742, Santa Monica, CA 90403. (213) 559-2375, extension 742.

news

3M TO DEVELOP ACADEMIC CENTER FOR MAGNETIC RECORDING RESEARCH

The $12 million facility at the University of California, San Diego, will be the first such academic center for basic research in the recording field in the US. UCSD will contribute $1 million, land

. . . NEWS continues on page 16 —

THE SOUND REVOLUTION . . . DEMYSTIFYING THE MYTHOLOGY

A Guest Editorial
by Peter K. Burkowitz, Polygram GmbH, West Germany

One thing is for sure: the first movement of the sound revolution has just been played. Nothing since Edison's and Berliner's great inventions has as much revolutionized what can be heard from a record as has the most recent achievement, the Compact Disc. From the lowest sub sounds of a giant organ bass to the highest audible formants, this new medium enables the very reproduction of anything that happened in front of a microphone, or was created on the keys of a synthesizer.

When it comes to musical dynamic range, no sound can be so faint as to outperform the signal-to-noise ratio of this prime digital audio home system.

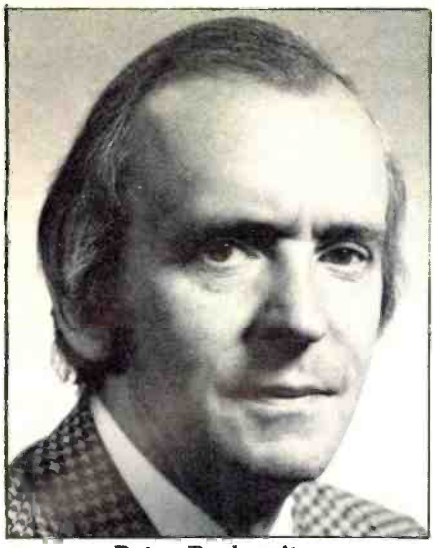
I said: the *first* movement has been played. Does this mean that other movements must follow?

Well, listening to numerous tunes during the development years, and reading the first bunch of reviews now, it becomes more and more obvious that, in fact, there may come further movements. It even appears that they not only "may" come, but "must."

Locating the next "must" is relatively easy. Almost any development follows the old rule that one step which strengthens a formerly weak link will render weak another link.

Of all other links the weakest one now, in terms of electro-acoustical facts and dogma, appears to be the one that is out of reach of electronic investors and equipment designers — the performance in the studio, and the ways in which performances are picked up and converted into a registered signal equivalent.

To many artists, "golden ears," producers, and certainly all hardware people, this statement at first may sound a little surprising. Through the decades haven't they done their utmost to explore every facet and edge of all thinkable methods and facilities? Eagerly steamed-up by legions of inventors and designers, haven't they squeezed out the last drop of perfection from multimiking, multichannelling, phasing and flanging, doubling, mixing-down, as long as something was still left on tape? Haven't they delivered wonders of spliced-up faulty playing and miracles of balance which never existed in the hall?



— Peter Burkowitz —

And haven't millions of listeners and squadrons of reviewers for decades often digested the results with enthusiastic acclaim?

Why should all of this suddenly be questioned? Only because there now is a carrier-means which no longer attributes any, not even the faintest, audible deterioration?

Let's be fair — any recording which sounded good in the past does sound equally good in the future. Recorded sound does *not* change because of the existence of the new carrier-means. The

phenomenon which we observe is rather one of an illusory and philosophical nature.

The illusory point is that, again, audiophiles and purists more frequently attend demo meetings, auditorium presentations, and again more dedicatedly prick up their ears for the faintest sensations that a freaky hi-fi channel can magnify.

I well remember the same complaints about blurred perspectives, multimiking interference, spectral de-focussing through pan-potting, etc, etc. This was when full-frequency-coverage recording and reproduction became possible, and when dogmatic one-mike-recording was given up in favor of more presence, more flexible balancing, and more effective accentuating, which the artists and producers — and many golden ears — so desparately wanted, and so vividly welcomed.

In fact, I bet that trying to reverse the trend in order to better please the ears of some purists would be bound to become a pretty nice and nasty battle — for what?

The other point, the philosophical one, focusses on a very basic confusion which for decades has accompanied our sound business, and probably will do so forever. It is the old confusion about what a sound recording really shall be. Should it be the ultimate in static, naturalistic "photography," comparable with a total in holographic still-frame? Or, should it better compare with a motion picture; i.e. with an artful, dynamic transition of one situation, the performance in nature, into a completely different situation which exists in the home of an average listener?

I think the vast majority of artists, producers, balance engineers, and consumers long ago has given the answer in favor of an inspiring and tasteful transition that should first and foremost convey the artistic message (whatever this may be) and only in the second place, if any, also should carry a geometric idea about the performing set-up and site.

The "old boys" in the business will remember the old (and new?) rule which governs the sensory charm of sound: The fewer microphones, the prettier the sound; The more microphones the more flexible the sound processing. Now, through the last two decades flexible sound processing has apparently been the pipe-dream of the whole recording industry.

Of course, there are borderline cases where, in fact, electro-acoustical dogma leads to ridiculous sound patterns, or where electro-acoustical libertinism results in exaggerated presence, felty strings, no depth, strange ambience, spectral de-focussing, blurred perspectives, and truly all such imperfections that just do not render a record less attractive to the market place, however much they displease the connoisseur. Well, turning it around, is there any rule or experience which would make us suspect that what pleases the connoisseur will *not* please the folks?

In so many areas of life we well know of the "golden mean." Certainly this also holds true for the recording business. And the awareness which recently has been directed towards the phenomena and philosophies of sound may well be worth the attention of all who can do something about it, be they artists, producers, engineers, mixers, marketing people, etc. And . . . it would be almost a miracle if pop people would give it a view too. ■■■

news

— continued from page 14 . . .

for the center, and four new faculty members whose speciality is magnetic recording research. The additional funding will be provided over a five-year period by 3M, IBM, Control Data Corp., Eastman Kodak, Data Electronics, Pfizer, and Verbatim Corp.

Magnetic recording — both audio/video and computer technology — is a $20 billion-a-year industry whose growth was significantly stimulated by 3M products. The company pioneered magnetic tape development in 1947, and produced the world's first videotape a decade later.

. . . More NEWS on page 123 —

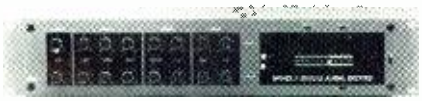
INNOVATIVE PRODUCTS FROM APHEX, FOR SUPERIOR SOUND REPRODUCTION

The remarkable Aphex Aural Exciter is a unique proprietary audio processing device that makes use of highly advanced psychoacoustic principals to effectively restore and enhance audio presence, brightness and intelligibility. The patented psychoacoustic process creates the perception of an increase in mid and high frequency energy, with *no* actual increase in power or level.

The Aural Exciter can produce dramatically improved clarity, dimension and character in any sound system or application. It can also reduce distortion in P.A. and sound reinforcement applications by providing increased penetration and audibility at reduced power levels. The device can be added to virtually any new or existing system with no danger of overloading other components or triggering compressors or limiters.

The Aural Exciter is a single-ended process, requiring no decoder. Once encoded, copies made from a processed tape sound every bit as good as the original.

The Aphex Aural Exciter is available in three models, each is specially designed for a specific application.



APHEX II-S $2,950.00

The Studio Aural Exciter is engineered for the sophisticated recording and production studio, as well as advanced sound reinforcement applications. In the studio, the Aural Exciter effectively restores the presence and clarity which the recording process removes, reviving that bright, unmistakable "live" quality. It can also make certain segments "stand out" without actually being louder. Used typically in stereo mixdown situations, this latest version of the Aural Exciter features increased flexibility so it's ideal for virtually all types of program material, from the hardest rock and roll, to the subtlest movie dialogue and sound effects.

The Aural Exciter is also well suited to stage and concert use. It can make any P.A. system sound much cleaner, brighter and intelligible without adding any level or feedback to the house or monitor system. It is particularly effective in filling acoustic spaces to eliminate dead spots. The device cleans up sound in overly reverberant halls and makes speaker location much less critical.

APHEX II-B $2,950.00

The Broadcast Aural Exciter has all the remarkable features and capabilities of the Studio unit, plus complete R.F. shielding and safety bypass relays in the event of power failure. Designed specifically for on-air use, this unit provides AM stations with the clarity and brightness of FM, while restoring to FM the naturalness and openess normally lost due to processing.

The most impressive aspect of the Aphex Broadcast Aural Exciter is the fact that the lower the quality of the playback system, the better the comparative benefit derived. The sound of your broadcast will satisfy the most demanding audiophile, and at the same time grab the attention of the rush-hour commuter.

APHEX AURAL EXCITER TYPE B $495.00

The Aural Exciter Type B is engineered for less demanding situations. It utilizes the same psychoacoustic principles to make Aural Excitement available to small clubs, studios, halls, restaurants, musicians, tape duplicators and sound contractors operating on a more modest budget. Retaining the most important features of its bigger brothers, the Aural Exciter Type B is a small, lightweight package with extensive capabilities limited only by the user's imagination.

SPECIFICATIONS APHEX II

FREQ. RESPONSE:	15HZ-50K HZ +0-.2 dB
THD, IMD:	.05% at max I/O
NOISE:	110dB below max output
CROSSTALK:	Better than -80dB
MAX I/O (with standard Jensen output xfrmr):	Selectable +21dB, +24Bm, + a user definable position
INPUT IMPEDANCE:	Selectable 600ohm or bridging, 40K Bal, 60K unbal
OUTPUT IMPEDANCE:	50ohm
METER:	2 color, 2 channel VTF with LED legends for drive, limit, peak and meter status indication
SIZE:	3-1/2" x 19" x 9"
WEIGHT:	19 lbs.
POWER REQUIRED:	100-240 VAC 50-60 Hz

*Single-ended transformerless and balanced transformerless outputs optional.

SPECIFICATIONS - TYPE B

FREQ. RESPONSE:	10 HZ-100K HZ ± .05 dB
THD:	Less than .01%
NOISE:	-90 dBV
OPERATING LEVEL:	Selectable -10 or 0 dBm
MAX I/O:	+20 dBm
INPUT IMPEDANCE:	47K ohm unbalanced
OUTPUT IMPEDANCE:	150 ohm unbalanced
METER:	Tri-colored LED for drive level
SIZE:	1-3/4" x 19" x 6"
WEIGHT:	4.5 lbs.
POWER REQUIREMENT:	100-130 VAC 50-60Hz (export version available)

COMPELLOR™ COMPRESSOR/LEVELER/PEAK LIMITER $995.00

The Compellor™ is a unique, revolutionary audio processing tool that combines the functions of a fast compressor with slow gain riding and an overall peak limiter. It provides complete flexibility in dynamics control when used as a broadcast pre-processor, as well as in the recording studio or live p.a. situation. The resulting sound is smooth and dense with an increase in perceived loudness and brightness.

The variable slope compressor operates over a 30 dB range with attack and release times controlled by program dynamics, eliminating "pumping" and the choked sound associated with deep compression.

Audio leveling over a 20 dB range maintains the audio in the "knee" of the compressor providing a uniquely dynamic compression which is rich in transient quality and openness, with an absolute ceiling maintained by the peak limiter.

The balance between compression and leveling actions is continuously variable; adapting the Compellor™ and its effects to an enormous variety of material.

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EQF-2 $449.00

The EQF-2 combines a 3-band sweep equalizer with a sweep Hi and Lo pass filter section. The EQ has switchable peak/shelf on the Hi and Lo sections, and reciprocal 12 dB of cut and boost on all sections. The filters are second order Butterworth and can be switched separately from the EQ section.

SPECIFICATIONS

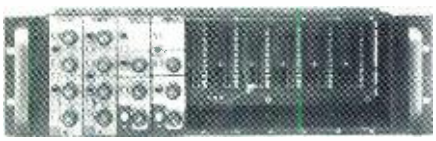
FREQ. RESPONSE:	± 1dB 20 Hz – 20 kHz all sections in
THD & IMD:	Below 0.1% at max. I/O
NOISE:	–123 dB below max. I/O
FILTERS:	Hi pass 20-500 Hz Lo pass 1-20 kHz
EQ LOW:	25 – 500 Hz
MID:	250 – 5kHz
HI:	1 – 20kHz
MAX. I/O:	+20dBm with optional Jensen xfrmr
SIZE:	1-1/2" x 5-1/4" x 6" (industry standard)
WEIGHT:	2 lbs.

CX-1 $449.00

The CX-1 is a very versatile module combining a "soft knee" compressor/limiter with a switchable expander/gate. The CX-1 uses the proprietary Aphex VCA chip to provide an extremely clean overall sound. The expander is adjustable from 0 to 100 dB of expansion (gating) and is the *only* noise gate on the market that can be guaranteed not to click or pop. The unit features a multi-functional LED display that indicates input, output, compression or expansion levels.

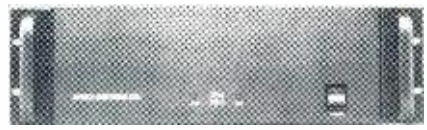
SPECIFICATIONS

BANDWIDTH:	± 1dB 20 – 20 KHz all sections
THD, IMD:	Less than 0.2% at max I/O
NOISE:	–85 dBm
MAX I/O:	+20 dBm (+30 dBm with optional Jensen xfrmr)
SIZE:	1-1/2" x 5-1/4" x 6" (industry standard)
WEIGHT:	2 lbs.



R-1 $195.00

The R-1 holds 10 Aphex modules and provides barrier strip access to all inputs and outputs. Power and ground are bussed.

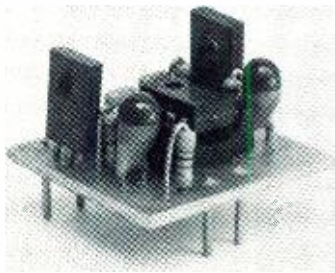


PS-1 $275.00

The PS-1 is a ± 16V @ 3.4A regulated supply with OVP that will power two R–1 racks.

4B-1 $349.00

Self-powered, the 4B-1 is for the mobile engineer. It holds 4 Aphex modules and has a built-in patch board on the rear with 1/4" and T-T size jacks.



2521 – OPERATIONAL MODULE

$35.00 (singles)

The 2521 Operational Module is a high speed, high output, short circuit proof buffer that takes on the characteristics of the IC that is plugged into it. It is current limited and can put out a full watt of power into a 62 ohm load.

The 2521 output transistors have a 3 amp rating for superior reliability. The unit is also 100% field repairable, so there's never a need to discard a complete module because of a defective 10¢ resistor. The 2521 can be continually and easily updated to meet changing needs.

FEATURES

100% Field-repairable
100% short circuit proof
Greatly improved overload characteristics
Built-in power decoupling
Socketed IC eliminates obsolescence
Extremely low noise current

SPECIFICATIONS

BANDWIDTH:	4MHz
THD (at clipping –1 dB):	0.02%
IMD:	0.02%
GAIN:	50,000 Min.
SLEW RATE:	>10 v/μ Sec.
OUTPUT NOISE:	–113 dBm
MAXIMUM INPUT:	30 Volts P-P
MAXIMUM POWER OUTPUT:	1 Watt (+30 dBm)
MAXIMUM VOLTS OUTPUT:	Supply –4 volts P-P
MAX. SUPPLY VOLTAGE:	±18 volts (with LF 351)

**High voltage, high output versions are available. Consult the factory for details.*

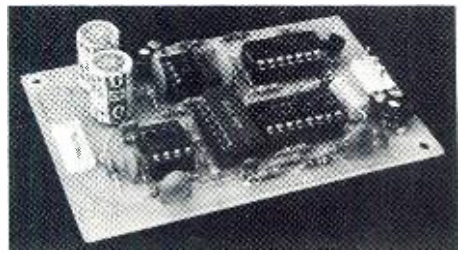
VCA PRODUCTS

1537A VCA IC $10.50 (100's)

The 1537A is the *only* monolithic Class A voltage-controlled attenuator on the market today. Its patented design features extremely low distortion, low noise, high stability and wide dynamic range. It can provide more than 100dB of attenuation at +20 dBm. Its high slew rate gives low T.I.M. and makes it useable from DC to 50 MHz.

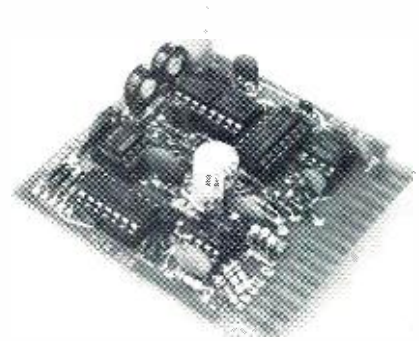
SPECIFICATIONS

BANDWIDTH:	DC to 50 MHz
THD:	0.004% TYP
IMD:	0.03% TYP
NOISE:	– 90 dBv worst case
MAX. ATTENUATION:	>100 dB, DC – 200 kHz



VCA 500A $89.00 (singles)

The new VCA 500 A utilizes a 1537A VCA IC to significantly improve the performance and overall sound quality of the MCI JH-500 series console. Conversion takes only a few minutes per channel with plug-in convenience.



VCA 505 $89.00 (singles)

The VCA 505 is an expanded version of the highly-acclaimed 1537A Voltage Controlled Attenuator. It utilizes a 15-pin card edge mount package for easy installation, has multiple bufferred control inputs for maximum versatility, and requires no additional circuity.

SIZE:	**2.75" high x 2.85" deep x .72" wide**

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EXPOSING AUDIO MYTHOLOGY

Laying to Rest . . . or at least exposing the false premises upon which they are based . . . some of the Pro-Audio Industry's more obvious "Old Wives Tales"

by John Roberts

This issue sees the start of a new column by John Roberts, president of Phoenix Audio Laboratory, in which he explores some of the more controversial aspects of audio mythology. It should be stressed, however, that the intention of this column is to establish a forum for discussion between *R-e/p* readers, rather than provide definitive answers one way or the other. If you agree or disagree with any of the author's comments, please feel free to write us about it — *Editor.*

As an introduction to this new column on Audio Mythology, it might be worthwhile considering the classifications into which audio professionals fall. At one end of the spectrum we have the highly objective, technically orientated "Meter Readers" — who quite often don't believe that anything can be explained unless it manifests itself as a measurement on a piece of test equipment — and, at the other, the more subjective "Golden Ears" who, foresaking technical measurements of an audio observation, rationalize their observations in more descriptive, human terms.

About the only thing that Meter Readers and Golden Ears have in common is a mutual disrespect for each other. Since we all are working toward the same goal — high-quality audio — any division within our ranks is counter productive.

I expect that to a great extent we are divided by terminology. The typical Golden Ears — if there is such a thing —uses a vocabulary to describe audio characteristics that can seem imprecise, or certainly unscientific, to the Meter Reader. This phenomenon will not be new to any pro-audio engineer who has been asked to turn up the "balls," or put more "edge" on a track. The engineer will translate that into an EQ setting, and move on. However, it helps that the engineer gets immediate negative feedback if he or she guessed wrong, a luxury not available to those of us trying to understand the Golden Ears' terminology.

This lack of a common vocabulary hinders communication, which truly is a shame. The engineer who ignores all GE commentary is a little like the banker who turned down every loan: he didn't make any *bad* loans, but he still went out of business because he didn't make any *good* ones!

While I don't have a glossary of GE terms with corresponding MR definitions, I have prepared a list of these terms. With a little help from you I will try to come up with some usable definitions, or at least educated guesses. If you can offer any further information on any one of the listed terms, please jot it down and send it to me care of *R-e/p*. I intend to compile the more popular definitions in a later issue. Perhaps we can come up with a "Rosetta Stone" to allow communication between those who make the music, and those who listen to it.

"Golden Ear" Terminology

1. Grain; grainy sound; grainy texture.
2. Soundstage (soundspace).
3. Imaging.
4. Transparency.
5. Front-to-back depth.
6. Brittle, metallic sound.
7. Coloration (distortion?).
8. Hard, edgy.
9. Airiness, air.
10. Sweet, sweet sound.
11. Tonal purity (lack of distortion?).
12. Delicacy.
13. Hard, hardening and coarsening of texture.
14. Thicken texture.
15. Definition.
16. Weight.
17. Congestion.
18. White sound.
19. Cool sound.
20. Harmonic richness (distortion?).
21. Dynamic contours.
22. Strain (IM distortion?).
23. Tube-like roundness (harmonic distortion?).
24. Character.
25. Softness
26. Fast sound.
27. Musicality.
28. Blurring.
29. Focus.
30. Nasal (mid-range resonances?).
31. Richness.
32. Glare
33. Eveness.

Please feel free to add to the list if you don't see your favorite terms, or to offer your own interpretation of what some of these highly subjective terms are meant to convey to a listener.

Audio Mythology

During the rather difficult task of recreating a musical event, usually in a vastly different sound space at a different volume level, there always will be trade-offs or compromises. In this month's column, and in future articles, I would like to address some of the more controversial engineering decisions, and hopefully establish a constructive dialog between those of differing opinion. While I would hope to be able to offer simultaneously two different viewpoints, this first column will be blessed with only one (mine).

Absolute Polarity:

I have chosen Absolute Polarity as our first Audio Myth, because it is an interesting example of a phenomenon that only a few claim to hear, and the vast majority who don't hear the effect doubt the opinion of those that do.

The premise of Absolute Polarity, simply stated, is: "The sound waveform created by the reproducer should be of the same polarity as the original sound." For example, if the initial attack of a kick drum is a pressure wavefront, then the speaker should also create a pressure wavefront. This runs counter to popular wisdom, which suggests that speakers should be in phase with each other to avoid cancellations, but that absolute polarity is *not* audible. This popular wisdom is so pervasive that, to this day, no standard exists to define the polarity of a signal through the entire record/playback chain.

While some segments of the chain have been standardized, this has been done more for convenience in multi-microphone or multiple-speaker applications, than for any concern about polarity. But even now one of the major speaker manufacturers still makes drivers that "Zig" while everyone elses "Zag"!

The conventional wisdom that Absolute Polarity is not audible (or any phase shift for that matter) was based on a popular experiment in which a square wave was passed through an all-pass network (to provide a known phase shift), and then compared to the original during listening tests. While the 'scope trace looked like anything but the original square wave, the perceived sound was unchanged. This result, added to the fact you never did hear *any* difference when you reversed the leads on your trusty HiFi, makes it easy to disbelieve in Absolute Polarity.

Well, contrary to substantual evidence, Absolute Polarity *can* be audible. If we substitute a simple two-tone signal (say 200 and 400 Hz) for that square wave, and pass them through that same phase shift network, we will hear an audible change in timbre with phase shift. Since a polarity inversion is equivalent to a 180-degree phase shift (at least to this signal), it follows that a polarity inversion causes an *audible change in timbre.*

The major distinction between the two experiments is the asymmetrical nature of the two-tone signal versus the symmetrical square wave. If we accept such an explanation, then it follows that speech and some musical instruments will be affected by polarity inversions, since they also have asymmetri-

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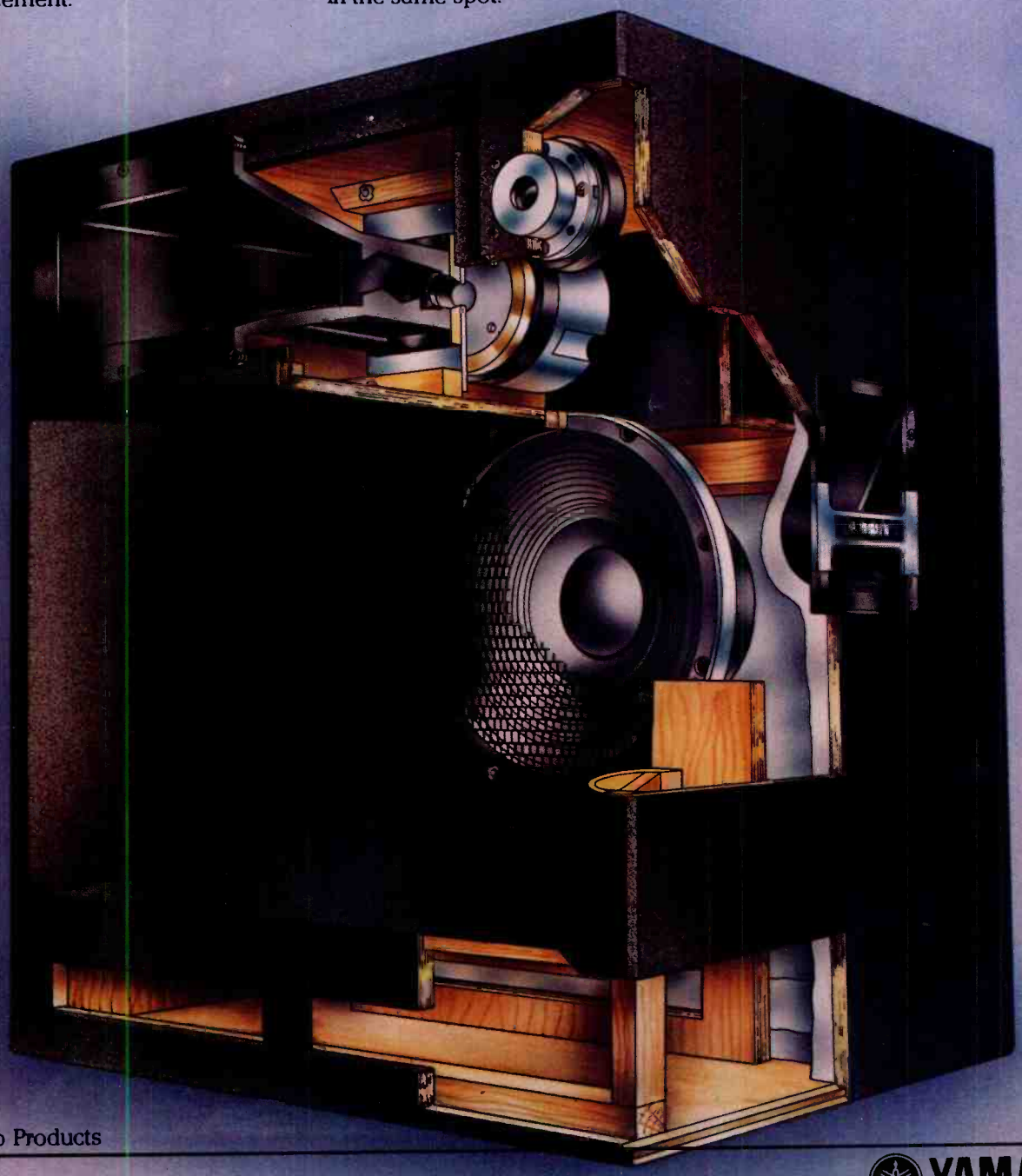

Combo Products

THE WAY IT WILL BE.

YAMAHA

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cal waveforms.

This effect, which is quite audible in the simple two-tone test, becomes weaker as the signal increases in complexity. Besides being weak, the effect is further reduced by reverberation (natural or otherwise), multiple-microphone techniques, and non-minimum phase circuitry. You will remember that conventional wisdom held that phase shift isn't audible, so why worry about it in circuit designs? What we can end up with is a signal whose overtones are phase shifted and, as a result, will not be in the correct phase relationship, no matter which way you connect the speaker wires.

I apologize for giving such a wishy-washy answer on our first example of Audio Mythology, but I have to give Absolute Polarity a definite "maybe." Research shows that the phenomenon does exist, but whether it can make it through a typical record/playback chain is another story.

Now there may be a few souls out there with good enough systems, and good enough ears, and last but not least with a good enough recording that "something," however subtle, might be audible. I would be interested in hearing from those *R-e/p* readers who *do* hear something, including details of what your system is composed of, and on what cut of what record you hear the effect. If I receive enough responses, I will tabulate the results and try to get some feedback from the recording engineer who was most successful at achieving the effect.

In the meanwhile, for those of us (me included) who can't hear the difference, I suggest we don't lose any sleep over the Absolute Polarity of our home systems, since the software will be random for some time to come. However, it would be nice to check out your studio system and recording chain for archival purposes. Just because the effect of absolute polarity is not particularly audible today, doesn't mean that it won't be 10 or 20 years from now, as speakers and other equipment in the chain improve.

References

For a discussion about the audibility of phase and polarity see reference #1; for information on the proposed tape standards see references #2 thru #5. Reference #1 contains a good bibliography of phase audibility research.

1. S.P. Lipshitz, M. Pocock, and J. Vanderkooy, *On the Audibility of Midrange Phase Distortion in Audio Systems*; Journal of the Audio Engineering Society, Volume 30, pp. 580-595 (Sept '82).

2. P. Butt, *A Fuss About Plus*; *R-e/p*, Volume 10, pp. 66-71 (Dec '79).

3. P. Butt, *A Proposed Method for Uniform Determination of Polarity Response of Magnetic Reproducers*, presented to 66th AES; Journal of the Audio Engineering Society (abstracts), Volume 28, p. 546 (Jul/Aug '80), preprint #1651.

4. D.S. Stodolsky, *The Standardization of Monaural Phase*; IEEE Transactions of Audio Electroacoustics, Volume AU-18, pp 288-299, (Sept '70).

5. J. Vanderkooy and S.P. Lipshitz, *Polarity and Phase Standards for Analog Tape Recorders*, presented to 69th AES (abstracts), Volume 29, pp 546 (Jul/Aug '81), preprint #1795.

NEW BUSINESS

As mentioned before, I would like to establish a constructive dialog regarding some of the more controversial topics present today. The following list of proposed topics is to be used as a starting point for this series. If you have an opinion or information, pro or con, regarding one of the listed topics (or unlisted one for that matter), please share it with us. We have nothing to lose but our ignorance . . .

Proposed Topics For Discussion

1. Digital audio sounds *good* because . . .

Digital audio sounds *bad* because . . .

2. Transformers *degrade* sound quality.

Transformers *improve* sound quality.

Transformers *don't affect* sound quality.

3. Gold-plated phono jacks, sockets, and switches cause audible improvements in sound quality. Or, No they don't!

4. The gauge (and type) of speaker

le. Or, No it isn't!
r slew rate should be 1,000
rosecond.
w rate should be 100 volts
ond.
Amplifier slew rate should be 10 volts per microsecond.

6. Negative feedback *degrades* sound quality.
Negative feedback *improves* sound quality.

7. Capacitors cause audible degradation. Or, No they don't!

8. Passive RIAA phono EQ is more accurate and/or sounds better. Or, no it doesn't!

9. 12 dB per octave crossovers sound better.
18 dB per octave crossovers sound better.
X dB per octave crossovers sound better (you supply the number "X").

10. Tube circuits sound *better*, more accurate, different than transistor circuits.
Tube circuits sound *worse*.

11. Discrete circuitry is superior to op-amps. Or, No it isn't!

12. Moving-coil cartridges sound better/different than moving-magnet designs. Or, No they don't!

13. Half-speed mastering sounds better because . . .
Half-speed mastering doesn't make a difference.

14. dbx sounds better.
Dolby sounds better.
No noise reduction sounds better.

15. You can improve the sound of your equipment by placing a weighted object on top of it. (Honestly, I didn't make this one up!)

16. You can improve the sound of your audio by running line-level signals through unscreened twin-lead antenna cable. (Nor this one!)

17. Unused speakers in the same room as driven ones will degrade the sound quality. (Like every control room I've ever been in!)

18. MOSFET power amplifiers are better than bipolar amps. Or, No they aren't!

19. +4 dBm systems are superior to -10 dBv systems. Or, No they aren't!

20. Dubbing tapes backwards improves the transient response of the copy.

■■■

STUDIO FINANCES

THE "PURCHASE/RENTAL" DECISION

How Changes in Recent Tax Legislation can affect a Studio's Tax Liability for Capital Acquisitions

by George Nadel Rivin, CPA

Recent tax legislation will have a significant impact on any studio operator purchasing new equipment in 1983. Because improvements in recording equipment are so rapid in today's technological environment, the anticipated period a studio operator expects to keep the equipment in service serves as a key input in the purchase/lease/rent decision-making process. Among the considerations to be discussed in this article are the planning phase; investment tax credit opportunities; depreciation rules; and sale-leaseback opportunities.

Every professional has some degree of ingrown desire to maintain a so called "state-of-the-art" facility. Thus, most recording, production, and audio-video studios exercise a form of periodic review of their present inventory, as well as their future equipment needs. This occurs as either a formal process, or through a series of informal discussions among those working on both sides of the glass. And, the decisions arrived at are the product of both *perceived* and *real* need, as well as the financial ability to acquire.

The discussion that follows relates both to *essential* (must have) acquisitions, and will center around fundamental descriptions of the new law and regu-

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In conjunction with Television Projects in London, Alan Bunting, the Audio Manager for BBC Scotland, specified the Series 2400 for their new dubbing suite – making this the second 2400 the BBC have purchased for their Glasgow studios so far.

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lations (without interjecting a discussion of the economic constraints affecting how the equipment is to be acquired), as well as to the admittedly more complex subjective discussion which applies to the acquisition of *enhancement* equipment, where the method of acquisition may very well affect the financial stability of the total operation. In this regard we will be talking about the analysis of a term and concept familiar to accountants and financial managers, but very possibly not to studio managers: *OPPORTUNITY COSTS.*

Planning the Capital Acquisition

The extent of equipment and facilities a studio should have on hand to meet the needs of both users and potential users, as well as to withstand competitive market pressures, is probably best arrived at through an economic analysis of the *oportunity costs* of *not* having another room, or of *not* owning a certain piece of equipment. Among the many questions that should be answered before deciding to acquire a capital asset are:

- Can the studio attract increased bookings?
- Can premium rates be obtained?

— the author —

George Nadel Rivin, BBA, MBA, CPA is Manager of Broadcast Services for Miller, Arase, Kaplan, Sheets & Co., Certified Public Accountants, based in North Hollywood, California. Prior accounting experience includes Senior Accountant positions with Deloitte, Haskins & Sessls, and Ernst & Whinney; Chief Financial Officer, Shepard Broadcasting Corporation; and Manager of Internal Audit, CBS. Rivin is a 1974 graduate of the University of Iowa (BBA, Accounting), and a 1982 graduate of Michigan State (MBA, Management and Marketing).

TABLE 1: PURCHASE/RENTAL ANALYSIS BASED ON CASH FLOW

Assumptions: Equipment Cost: $10,000 (purchased outright — all cash).
Equipment Rental: $100 per day — 15 days per year.
Average Cost of Capital: 12% for three years.
Studio is in 30% incremental bracket (Total Federal and State).

CASH FLOW IF RENTED

YR 1	$1,500 × 0.893*	= $1,340
YR 2	1,500 × 0.797	= 1,196
YR 3	1,500 × 0.712	= 1,068
YR 4	1,500 × 0.636	= 954
YR 5	1,500 × 0.567	= 851

Total: $5,409

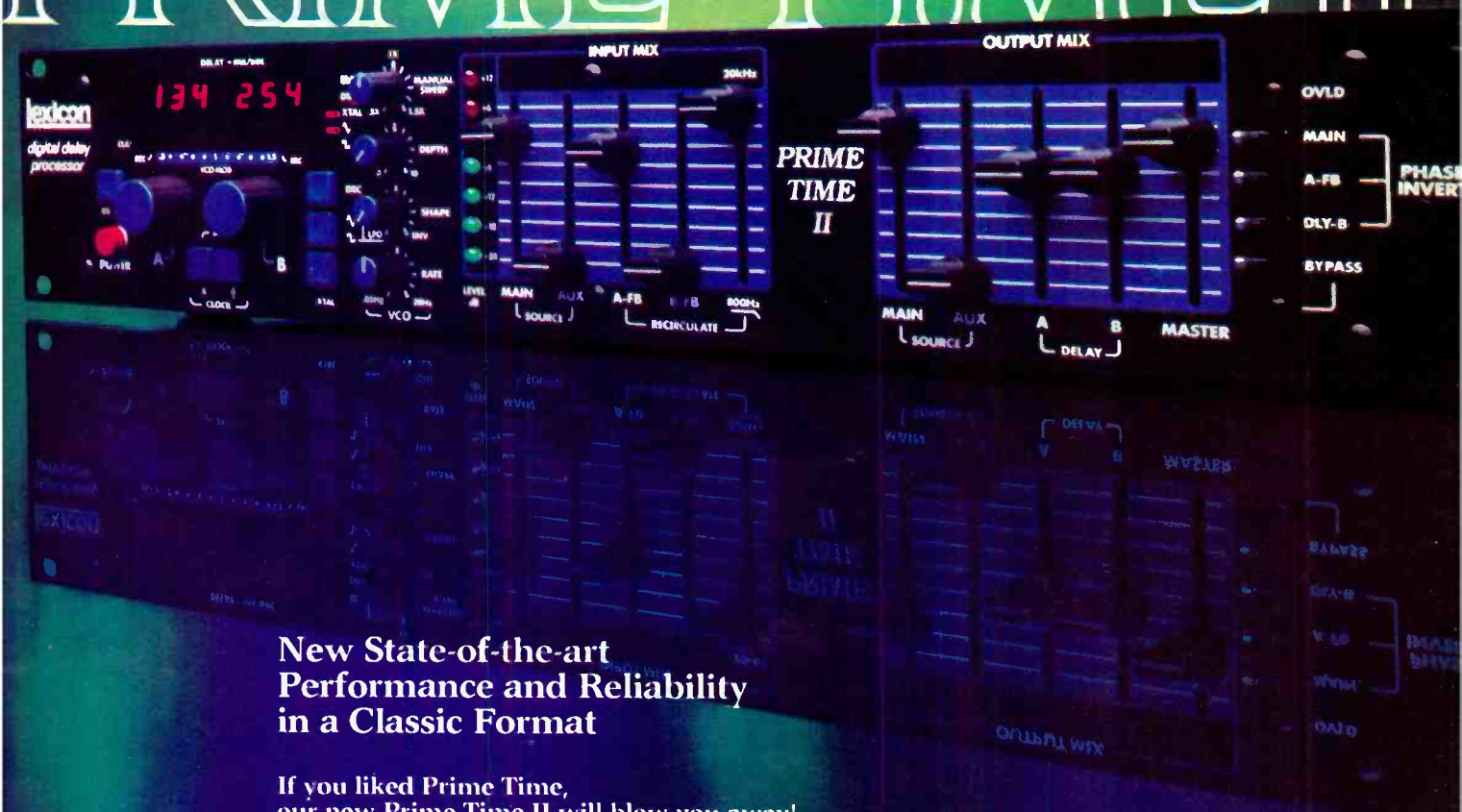
CASH FLOW IF PURCHASED

Purchase Price Net of Investment Tax Credit		= $9,000
YR 1 Depreciation —	$9,500 × 0.15** × 0.893 × 0.30	= (382)
	9,500 × 0.22 × 0.797 × 0.30	= (500)
	9,500 × 0.21 × 0.712 × 0.30	= (426)
	9,500 × 0.21 × 0.636 × 0.30	= (380)
	9,500 × 0.21 × 0.567 × 0.30	= (339)

Total: $6,973

*This column of figures represents the Discounted Preset Value Factor based on a 12% cost of Capital.
** These figures are taken from Accelerated Cost Recovery System (ACRS) tables.

PRIME TIME II



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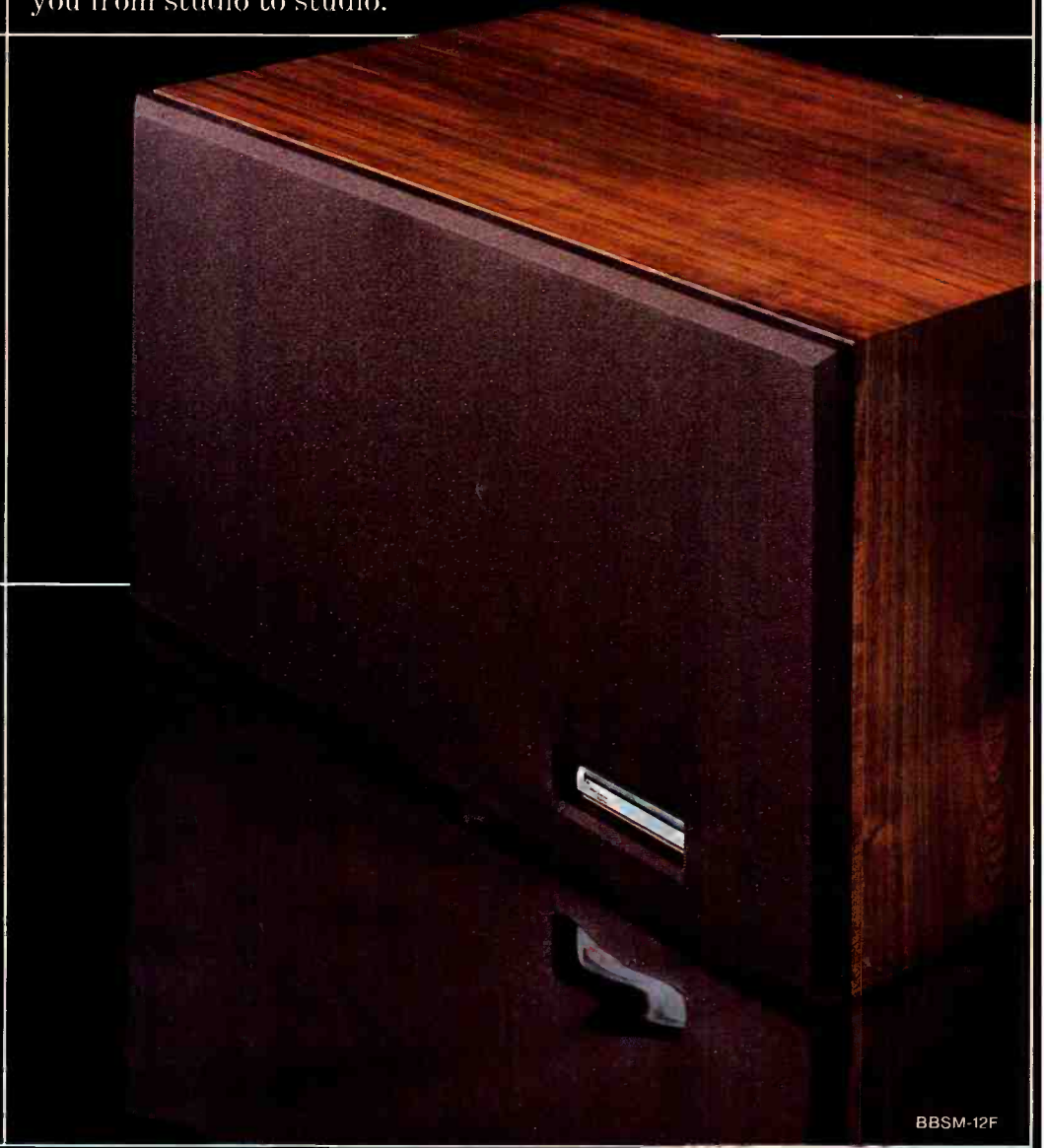
BBSM-6F

BBSM6



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• Can the increased demand for studio time be accommodated without excessive demand that might inconvenience regular users?
• Can the particular piece of gear be rented on short notice at reasonable rates?

Any review of purchasing enhancement gear, versus either the cost of equipment rentals, or the lost revenues resulting from not having the equipment needed by potential users, must consider the debt service required to fund the acquisition. Sometimes these are "hidden borrowings" that occur when no monies are borrowed directly for the purpose of financing the equipment purchase, but are rather *borrowed* from working capital. Interest on these "loans from working capital" should be computed based on the studio's average cost of capital, just as if the funds came from any financial institution loan. Even where no actual borrowing takes place, the opportunity cost of foregone interest income must be considered. An example of a purchase versus rental analysis would take the form given in Table 1.

Assuming no lost revenues, the total cash flow if rented figure of $5,409 is *less* than $6,973, the cash flow if purchased, so the decision is to rent. Adding the assumption that if the equipment must be rented, the probabilities of lost revenues would be $1,000 per year — 10%, $700 per year — 25%, $500 per year — 40%, and $300 per year — 25%, lost revenues would be calculated in the following manner:

$1,000 × 0.1 = $100
700 × 0.25 = 175
500 × 0.4 = 200
300 × 0.25 = 75

Annual Lost Revenues: $550

Which translates to:

YR 1 $550 × 0.893 = $ 491
YR 2 550 × 0.797 = 438
YR 3 550 × 0.712 = 392
YR 4 550 × 0.636 = 350
YR 5 550 × 0.567 = 312

Total Lost Revenues: $1,983

TEFRA-ACRS-ERTA

The information presented in this article is per the Internal Revenue Service Code, and incorporates those changes brought about by the Tax Equity and Fiscal Responsibility Act of 1982 (TEFRA). This massive piece of legislation, which became law on September 3, 1982, requires basis adjustments for equipment on which the investment tax credit is taken. In addition, it modifies both the Accelerated Cost Recovery System (ACRS) depreciation provisions, and the safe harbor leasing regulations set forth in the Economic Recovery Tax Act of 1981 (ERTA). Since some sections of the '81 Act were scheduled to be phased-in through 1986, it is crucial that the modifications of the '82 Act be adhered to when applying ERTA-originated provisions.

Rental costs from prior analysis: $5,409
Opportunity cost of lost revenues: 1,983

Total Cash Flow if Rented: $7,392

Now, taking into account the opportunity cost of lost rentals, the real purchase price of $6,973 is *less* than the total cash flow if rented, $7,392, so the decision is to purchase.

However, this calculation does not

TABLE 2: CASH FLOW WITH INTEREST INCOME FOREGONE.

YR 1	$9,000 -	$1,500 cumulative rentals expense =	$7,500 × 0.10 × 0.893 =	$ 670
YR 2	9,000 -	3,000 cumulative rentals expense =	6,000 × 0.10 × 0.797 =	487
YR 3	9,000 -	4,500 cumulative rentals expense =	4,500 × 0.10 × 0.712 =	320
YR 4	9,000 -	6,000 cumulative rentals expense =	3,000 × 0.10 × 0.636 =	191
YR 5	9,000 -	7,500 cumulative rentals expense =	1,500 × 0.10 × 0.567 =	85

Total Interest Foregone: $1,744

Real Purchase price from prior analysis: $6,973
Opportunity cost of interest foregone: $1,744
Total Cash Flow if purchased: $8,717

TABLE 3: TAX BENEFITS FOR EQUIPMENT PURCHASE

	Purchase Price	− Investment Tax Credit	× Basis Adjustment Factor	= Basis	× ACRS Depreciation	= Depreciation Deduction
Year 1	$10,000	$1,000	0.5	$9,500	0.15	$1,425
Year 2				9,500	0.22	2,090
Year 3				9,500	0.21	1,995
Year 4				9,500	0.21	1,995
Year 5				9,500	0.21	1,995
	Total Benefits	**$1,000**				$9,500

consider the opportunity cost of interest income foregone on the $9,000 net of investment credit purchase price. Assuming that an average rate of 10% could have been earned if the funds were free for investment, the analysis given in Table 2 would apply.

Now, taking into account both the opportunity cost of lost rentals, *and* the opportunity cost of interest income foregone, the $7,392 rental cost is *less* than $8,717, so the decision is to rent.

As the above examples indicate, probability analysis, the practice of using discounted present values, and the consideration of opportunity costs, are vital to making a purchase/rent decision. Probabilities must be established based on expert opinions, so as not to distort the opportunity cost of lost revenues. Present values must be applied, since conclusions can differ when the effect of discounting is overlooked.

Due to the dynamic nature of technological advancements within the recording industry, studio management may not always choose the most cost-effective alternative. If a circuit or transport improvement is expected to become available in a new, improved multitrack machine within 12 to 18 months, for example, absorbing additional rental costs during the interim period may be advisable from a long-term perspective. Also, once a studio decides on a timetable for converting, for example, to digital equipment, it will likely increase its dependence on rental houses.

The tax benefits achieved through capital acquisitions often make the difference as to whether or not the rent/purchase analysis recommends a purchase. Without the tax advantages resulting from the investment tax credit and depreciation deductions, most analyses would favor renting. However, as we will now discuss, the government's policy to encourage investment in capital assets can turn an even proposition into a "must" purchase.

Investment Tax Credit

Taxpayers can take a credit for qualified investments in eligible business property. In addition to operational studio equipment and furniture and fixtures, air conditioning and heating systems would qualify for the investment tax credit. Such credit is based on the properties' recovery period used in determining depreciation. A credit equal to 10% of the purchase price is available on all equipment with a life of five years or more. For three-year property, a 6% credit may be taken. Prior to 1983 the investment credit was allowed to offset up to 90% of tax liability over $25,000. TEFRA reduced the maximum amount of tax liability subject to offset to 85%.

Under the 1982 Act, the depreciable basis for all assets placed into service after December 31, 1982, must be reduced by an amount equal to half of the investment credit. Therefore, a $20,000 mixing console purchased during 1983 would be assigned a basis of

. . . concluded on page 124 —

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Getting Started in the Recording World . . .

GABE VELTRI

. . . Establishing a Rapport in the Studio

Interviewed by Robert Carr

About seven years ago, Gabe Veltri boarded a plane from Pittsburgh to Los Angeles, to pursue a career in the recording industry. Following the traditional studio route, he took odd jobs sweeping offices, cleaning up, and acting as gofer for the local recording studios. Before long, Veltri had worked his way into producer Richard Perry's Studio 55 in Hollywood, by setting up microphones for some of the best engineers in the business — Bill Schnee, Val Garay, James Guthrie, and Elliot Scheiner. After three-and-a-half years, his apprenticeship paid off. Veltri, as an independent, started doing all of Richard Perry's sessions, and has since racked up a credible track record of artist clients, such as Alice Cooper, Diana Ross, the Pointer Sisters, and A Taste of Honey. His current projects include international recording artist Julio Iglesais (reportedly the largest selling record artist in the world), The Crusaders, and June Pointer's solo album, as well as Jeffrey Osborne's single "I Just Want to Be Your Friend" for the movie *The Toy*, and Chuck Berry's medley for the movie *Class Reunion*.

R-e/p (Robert Carr): You seem to have struck gold by associating yourself with Richard Perry's Studio 55. Is it useful for an engineer new to the business to work in a popular studio?

Gabe Veltri: Studio 55 is a good place to gain experience, because I had the opportunity to work as a second engineer for three or four years around a lot of great acts, producers, and engineers — people like Jimmy Iovine, David Foster, Stevie Wonder and, of course, Richard Perry. Probably the number one engineer I've worked with is Bill Schnee. If you looked up the word "engineer" in the dictionary, I think you'd see his picture there!

R-e/p (Robert Carr): what about Bill impressed you so much?

Gabe Veltri: He's just totally professional, courteous, tactful. It's really a matter of attitude, which I think is one of the most important aspects of being an engineer. An engineer is responsible for maintaining a particular vibe in the studio. Whatever the vibe is in the room, *that's* what goes on tape.

I'll give you some examples. I can't see spending an inordinate amount of time with a musician trying to get a sound —at least not to the point of wearing him out. By the time the session rolls around, he's exhausted, and pissed off at you.

You have to determine when it's necessary [to spend time on a sound], and when it isn't. For instance, say you find a background noise in one of the instuments. You ask yourself, "Is this going to be innocuous as far as the end product is concerned? Will it be detrimental to the final mix?" If you spend too much time getting rid of the noise that is innocuous, you're destroying the

The face is young, but the credentials show fifteen years of experience in the industry. In seven years with A&R Recording and eight years as an independent engineer and producer, Elliot Scheiner has worked with the finest: Jimmy Buffet, Donald Fagen, Roberta Flack, Foghat, Billy Joel, Olivia Newton John, Ricki Lee Jones, Phoebe Snow and Steely Dan. With two Grammys as proof of his engineering skills, he now spends about a third of his time producing.

ON METHOD

"All of my recordings have basically been very, very clean. I like everything that's on tape to be heard, without strain to one's ears. My method is to clean up everything and make sure that everything that was intended to be heard is heard. I guess that's carried over to production. I don't really want to be categorized as...'Oh yeah, his stuff is real clean, it always sounds good.' I want to be able to make really good records of all types."

ON COMING UP

"I still feel the best way to learn about the industry is being in the industry. The recording schools teach basic fundamentals and that's OK. But it doesn't really apply. You have to go in there and experience it and get in trouble and work it out yourself. That's sort of how I grew up in the industry. I learned everything I know from Phil Ramone. But basically I started at the bottom and it was really the only way to go. It's a long process now days, but you learn a lot."

ON DIGITAL

"Well the first time I recorded in a studio with it, we were doing an overdub on a piano track and it was this wonderful grand piano, that sounded unbelievable in the room. We recorded it and I played it back for the first time digitally and it was like having my head under the cover of the piano. It's so real. It will have to get a lot more inexpensive to replace analog totally, but I definitely think that it's the future."

ON BAD EXPERIENCES

"There was a moment not too long ago when I got into the studio, producing and engineering, and I was really happy with what everybody was playing. The room sounded amazing that day. And when it came up to the first play back I was thrilled. We reeled back the tape and it starts to roll and it sounded terrible. There was no top end on the tape, the bottom end was ill-defined and I was embarrassed. We had a serious tape problem."

ON TAPE

"One of the maintenance engineers suggested that I try 226. The first playback just astounded me, I was amazed. The top end, the bottom end, everything sounded exactly the way I was listening to it when it went through the console. And I became a 226 freak after that. I can't be bought, so if I say I like 3M 226 it's because I believe in it. I really feel strongly about the tape and what it's done for me."

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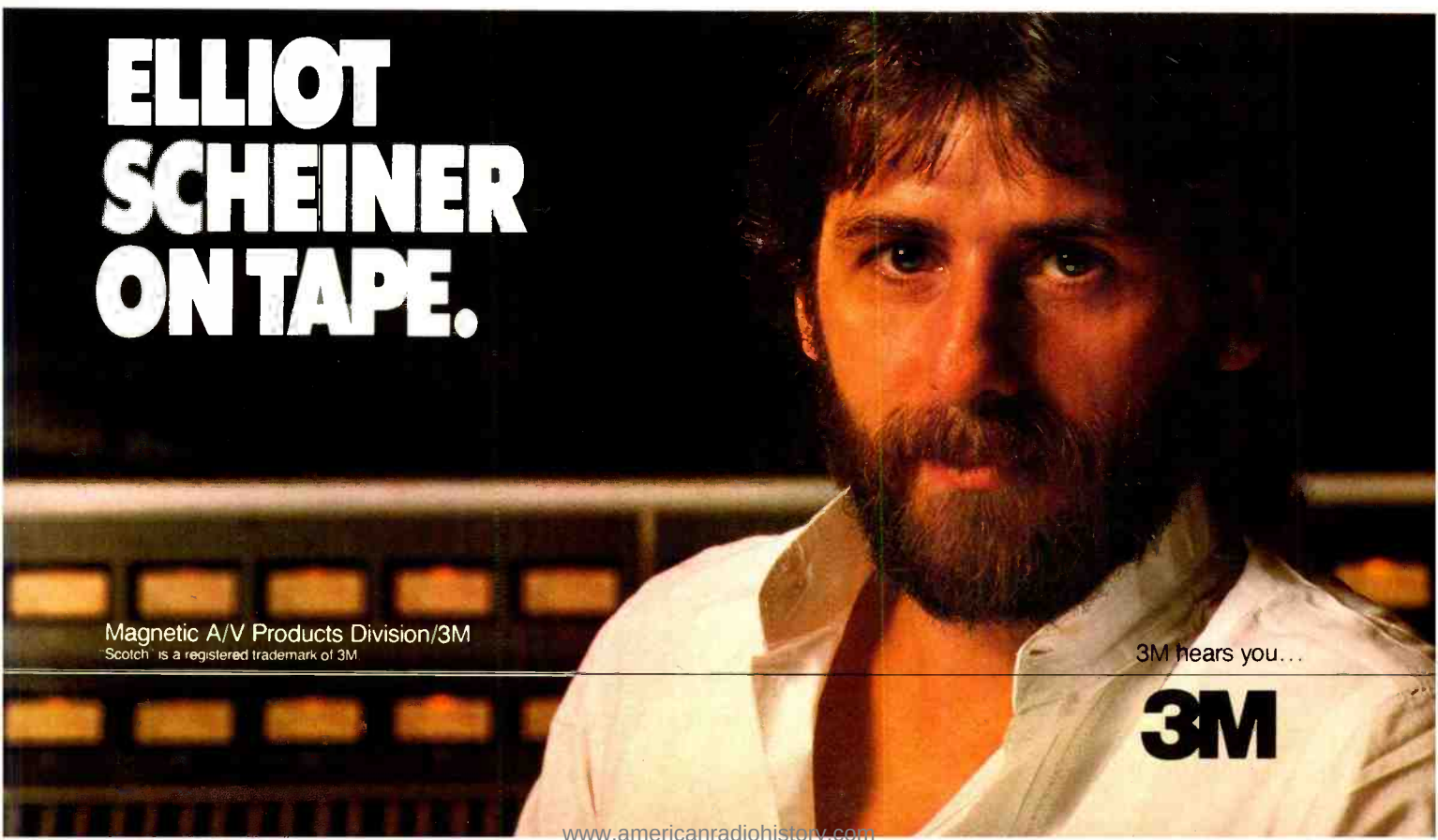

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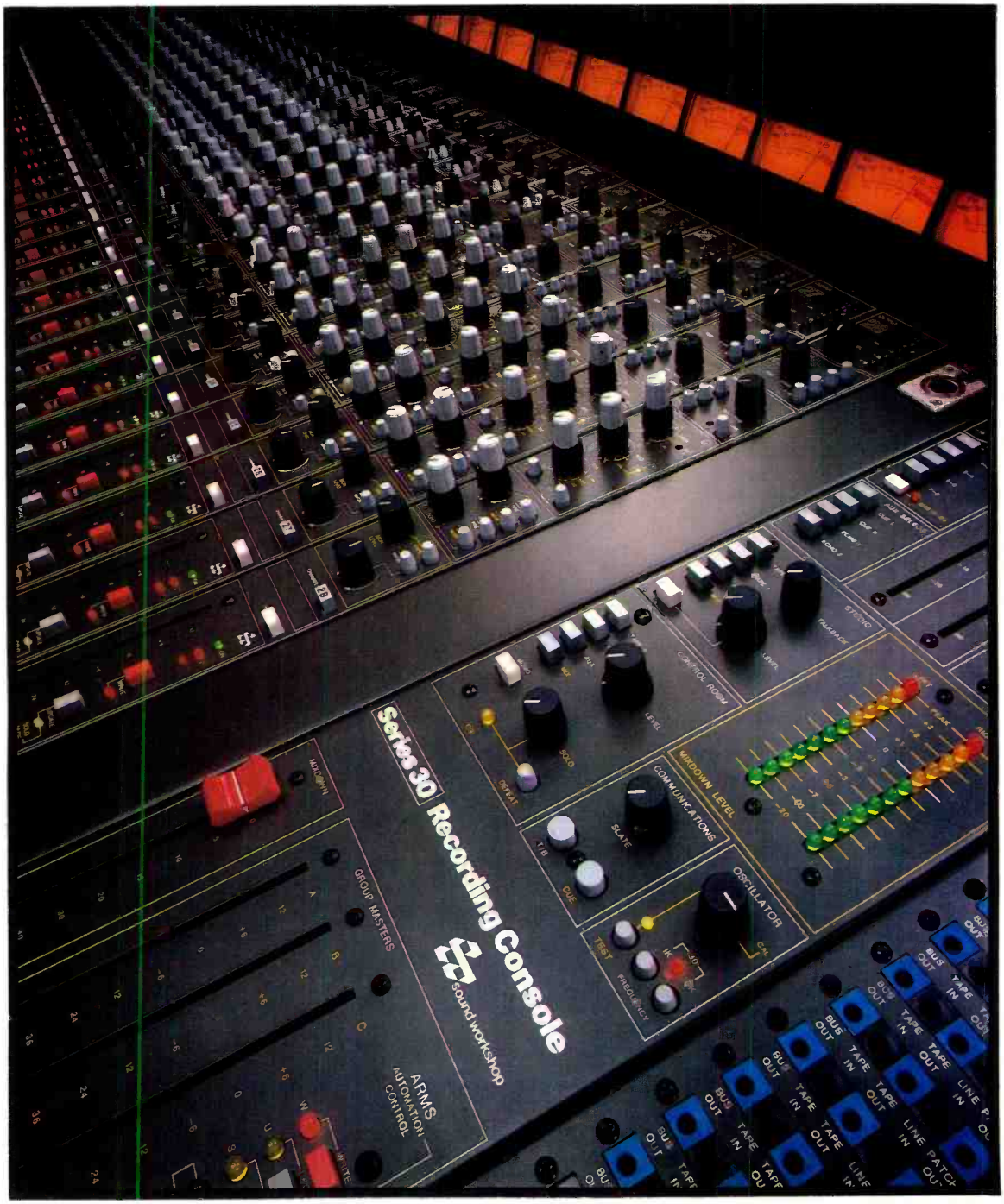

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vibe, which is infinitely more important to the success of the session.

Or with a producer, if *you* have an idea, it's best to get *them* to think of your idea, if you know what I mean.

R-e/p (Robert Carr): Just subliminally lead them around to your way of thinking?

Gabe Veltri: Right. You're dealing with people's egos. You don't want to say, "Hey, dig this. This is the way it should be." Probably the main part of the engineer's job is to allow the other person's creative energies to be used to their optimum. Be tactful.

R-e/p: It's almost a short course in psychology then. Work with the musicians to bring out the best . . .

GV: No, not necessarily bring anything out of them. That's the producer's job. But to allow that creative aspect to be brought out. There's a delicate balance that must be maintained. You're not staying out of the way; you're actively involved. Yet you must be cautious of other people's feelings. Your ego should never be a major part of the situation. You have to put your ego aside, but at the same time maintain total confidence in yourself. That's what engineering is all about — a delicate balance all the way around.

R-e/p: Having worked with so many talented engineers as a second, did you develop a technique for learning or absorbing ideas from the engineers you worked with?

GV: Obviously it was a good opportunity to watch somebody like James Guthrie or Elliot Scheiner mix, or Bill Schnee work with people and equipment, or Tom Dowd edit. You can't help but absorb what's happening. But I think the most important aspect is learning to hear all the subtleties. How you hear affects what you do. I've had guys ask me about a drum sound during a session — what kind of mikes I'm using, and how I'm setting them up. They would come back the next day and tell me, "I don't know how you do it. I couldn't get anything happening with those mikes and your setup." That's because it's just how you hear. It's a matter of balancing the different mikes on a drum kit; getting the sound you want to hear that's in your head, and compensating when you put the drums in with the rest of the tracks.

I did some work with [engineer] James Guthrie right after he finished Pink Floyd's *The Wall*. By that time I had worked with a lot of American engineers, and was familiar with the American style of recording. But aside from the fact that the English sound has less middle, I had only heard about English engineers going for sound just for sound's sake — getting into more abstractions. For example, breaking a picture tube in the parking lot with a sledge hammer and adding that sound to the album. Or one really horrifying sound, he informed me, was originally a wire recording of Stutka divebombers. He played me the tape copies of the wire recordings, and those things were horrifying. They [Pink Floyd] mixed all that into the music. It goes back to the approach of having pianos that didn't sound like pianos; guitars that didn't sound like guitars.

R-e/p: You mentioned that the "English sound" doesn't have as much middle as American recordings.

GV: They just take out a lot of the middle. They're taking away the fundamentals from the overtones, which has the effect of enhancing the overtones. A great example of the English sound is something like [Supertramp's] *Breakfast in America*, where they have a crispy upper-range and a nice, low bottom that makes the recording very listenable.

In fact, I went to see Supertramp in concert. They were really loud, but my ears weren't ringing when I left the show, mainly because they took out those middle frequencies — the telephone-type frequencies. It's just a stylistic approach. That's how they get a clear, crisp sound. But there's a trade-off. You have less punch when you do that.

R-e/p: After working with a variety of engineers and producers on many different projects, or parts of the projects —sometimes coming in midway through, or just for mixing some tracks — do you find any difficulty in tuning into the direction of the project?

GV: No, I like doing that a lot. If you get one project that lasts three months, that's all you really hear. But by moving around, I get to do a project that's R&B, the next day a rock and roll session, do some dance music, or whatever. I like the exposure to such a variety of music.

One kind of date feeds another: you bring something from one type of session to the next. In general, music today tends to be a bit *too* segregated. Everything is categorized. When you do all R&B dates, you just don't think of some of the things that would be obvious [when] doing rock and roll. A simple example: with one clique you say "Stevie," and everybody thinks of Stevie Nicks. To another group of people, the same word means Stevie Wonder. That may be the result of radio programming that concentrates on a specific category, rather than a broad spectrum of musical styles. Personally, I like a more homogenous approach to music. Doing a lot of sessions nurtures a synthesis of styles, rather than maintaining the great polarity of music.

R-e/p: Does that translate into a cross-pollination of miking techniques?

GV: In the beginning, I had to stick to particular stylistic techniques, but now I can get away with being more creative. Used to be that when you did an R&B date, you'd have to mike everything

close. The drums would be really dead; the toms would go "tap-tap-tap-tap-tap" — a tight, closed-in sound. Whereas on a rock date, you have room mikes giving you a bigger, boomier, "Led Zeppelin" drum sound. Now I can cross the techniques back and forth.

One area today where there's a definite synthesis of styles is in the dance clubs, where they feature the urban-contemporary type of music. There's New Wave music with a drum machine playing an R&B groove. [At which point Veltri plays a couple of cassettes with electronic drum playing an R&B rhythm, synthesized bass and other music parts, and vocals.] This is some work that I'm doing with [producer] John Barnes.

R-e/p: Do you find that producers and record-company people are preferring to use synthesizers, rather than bringing in traditional musicians like drummers, guitarists, horn or string players?

GV: It's just the Eighties style that produces a different feel that's reflective of the video games and their sound effects. The drum machine gives you a very precise track, and the electronic essence comes from the synthesizers.

R-e/p: Have you developed a procedure to prepare yourself for a jazz date, for example, as opposed to an R&B date, or a rock and roll session.?

GV: My head has to be different. No sound is an island. Some engineers try to make the music conform to their sound. I try to capture the sound of the artist's music.

R-e/p: Do you do anything to set yourself up for a new session, like going to the pre-production meetings and rehearsals with the group?

GV: Fifty push-ups [laughter]. It depends on the project. I don't have that opportunity all the time, although I like to do it whenever possible. I might just listen to the previous recordings of whoever the artist is. Any kind of input I can get. If it's a new studio, I'll have my second call and get a list of the microphones that are available. Then I'll spend some time figuring out how to use those mikes with the instrumentation and the sounds we want. I'll write down a list of the microphones I'll probably use for the date, so the second can set all that up for me before the session.

Generally, when I'm working with Richard [Perry], he'll give me a rough demo tape of tunes, and then tell me his concept of how he hears it as far as the types of sounds he wants. Then I'll plan on either getting a big fat sound for the snare, for instance, like those New York, Simon-and-Garfunkle records, or a trashy snare drum that you might hear on a Rolling Stones cut.

R-e/p: Suppose the drummer comes in with a metal snare drum, and you needed to get a big, fat warm sound. What would you do?

GV: Well, usually a session guy will come in with a couple of different snares, so you can work with the best one. That's where the physician's bedside manner comes in. You have to learn to talk the musician into giving you the sound you want.

I did a session about a week ago with the Crusaders. They were doing sort of a "groove" tune, and had Tony Williams on drums. He played it really well, even though you may not think of Tony Williams as a groove player. But he came into the session with a bass drum that had a front head on it, which is not usually how contemporary groove records are made. You generally have the front head off, put some padding in there, and so on. I knew that was the sound the producer had in mind.

While they were running down the tune, I tried everything I could think of to get that sound. And all the while I knew — I don't want to say it was preconceived or anything — but I had a feeling that the head *had* to come off. Well, I can't go up and start telling this guy how his drums should be; he's played drums all his life. So it was just the subtlety of gradually seeing whether he was opposed to this little point, then this point, then this one, until we finally got to where we could get the sound everybody wanted.

R-e/p: Did you get the head off?

GV: I got it off, and he wasn't too mad at me. I don't think he was mad at me at all. But it was a step-by-step process, rather than saying, "The only way we're going to get this sound is to take that damn head off." What's that going to do for the session? You have to show the musicians you're willing to work with them to solve the problem. Have

ENHANCING THE SOUND OF DRUM MACHINES WITH NATURAL AMBIENCE AND COLOR

The digital drum machine, capable of delivering realistic and easily programmable rhythm tracks, is becoming more and more popular on sessions these days. One main advantage, aside from the fact that the time is perfect, is that the person programming the machine — particularly if he or she is not a drummer —probably is going to come up with rhythms that a live drummer wouldn't naturally think of playing, or physically couldn't play on the track.

The main disadvantage of using drum machines, however, has been insufficient storage capacity in the memory chips that hold the digitized drum tones. [At press time, there were rumors that Oberheim was releasing a version of its drum computer with more storage memory —*Ed.*] To compensate, manufacturers of drum machines are forced to shorten the duration of the sound, thus cutting off any ambience and overtones that are generally associated with a live drum sound. Gabe Veltri has been experimenting with the Oberheim DMX, and offers the following suggestions for recording.

"Although the Oberheim DMX probably has the best kick-drum sound," Veltri offers, "all the drums have a characteristic dead tone, as a result of the storage capacity of the [memory] chip. The sound cuts off fairly quickly —the bass note, for instance, can't produce a 'ba-boomm,' and have the '-oomm' ring out. But even if it did, on a fast passage it may run into the next note anyway, like the first hit on 'boommboomm.'

"To get a more natural sound for a bass drum, I used a 24- or 26-inch Rototom from the studio. I tuned it to the frequency of a bass drum, and angled it perpendicular to the floor. Then I placed an Auratone [monitor speaker] right next to the head facing the Rototom, and about ½- or ¼-inch away. I miked the back of the Rototom as though it were a bass drum — the microphone approximately a foot or so in front of the speaker, but behind the Rototom. The air pressure from the drum machine being played through the Auratone hit the head like a beater. I got the initial attack from the digital signal through the speaker, and the 'ring' from the vibrations of the drum head. It sounds great . . . very realistic.

— *continued overleaf . . .*

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And you have to *listen*. Listen to what they say; listen to the sound of their equipment. You can't necessarily come in with a pre-conceived sound in your head. Listen to the sound they have, and keep in mind the sound the producer and the rest of the band wants. It's really a total synthesis of pulling all these ideas together, and coming up with something that pleases most of the people. And don't destroy the vibe. The vibe is on tape.

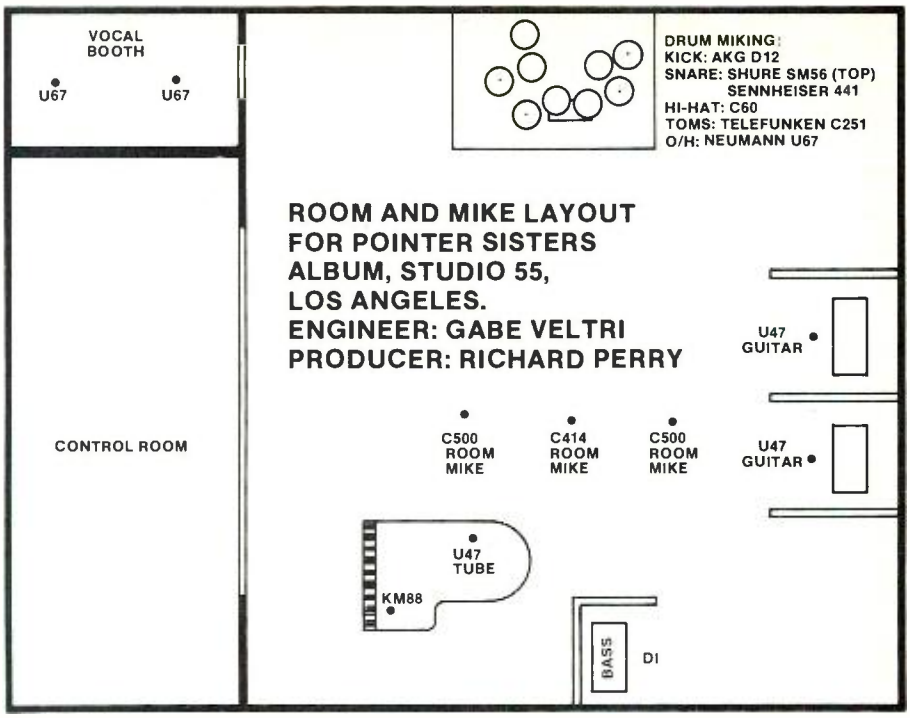
R-e/p: While we're on the subject of drums, what kind of mikes do you generally set up on the kit?

GV: I like the [AKG] D-12 on the kick drum, or sometimes a Sennheiser 421. I'm tending toward the 421, because the drummers are using harder and harder beaters made out of wood — maybe rock, I don't know [laughter]. They produce more "snap," which the Sennheiser gives you anyway on a kick drum with a little duller sound. But you can't get the bottom from a 421 that you can with a D-12. It's not a "boominess" bottom, but what's needed to compensate for the plastic head, where you sometimes have to EQ out a certain amount of mid-bass, like around 200 to 500 Hz. That's pretty much the case with plastic heads in general. So I use the D-12 if I need more bottom; the 421 if I need more attack.

For the snare I'll use a dynamic like a Shure SM-56, or maybe a [Neumann] KM-84 on top. I'll generally put a mike, probably a Sennheiser 441, underneath with the phase reversed to pick up the snares, although I do not always use it. Sometimes you'll have a big drum that just doesn't give you much of the snare "sizzle" when you mike it only from the top. In those cases, I may add in a little of the bottom mike just for color — a thicker, snappier sound. It depends a lot on the sound of the snare drum with the sound of the kit.

R-e/p: How far away is the mike you use underneath the snare?

GV: I try not to get too close. As long as you use a fairly directional mike, you can expand the distance and aim it at the snares. At the same time, the narrow pattern rejects the squeeking of the bass-drum pedal, high-hat pedal, and any other noises. The Sennheiser 441 is perfect.



Great microphones for toms are Telefunken 251s or, if I can get them, Neumann KM-56s. I should really get a collection of my own microphones together, but the ones I'd want would be mostly tube mikes, which are hard to find anymore. The prices that people want for them are outrageous. You can't even go to these out-of-the-way places anymore, because there are these guys who do nothing but put ads in recording magazines, and call through all the studio listing searching for old mikes in good condition. Now there's no place that doesn't know what a 251 is worth!

My final choice for the toms is a set of KM-84s. I really like 251s, but trying to get the spread a little better, I'm switching to more directional mikes lately.

I like to use an old AKG C-60 on the high-hat, and the overheads are Neumann U-67s or KM-84s.

R-e/p: The bass sound on the Pointer Sisters record is very clean. What's your technique for mixing the bass and drums?

GV: Every bass player has a different sound on bass — that's where you have to go into the date knowing pretty much what you want. I strive to get that foundation down really solid. I'll use EQ, limiting, or whatever it takes. Before I get my final sound, I have to see how the bass and kick fit together. I get the sounds while they're running down the tune.

R-e/p: Is that pretty much how you did "Slowhand" and "He's So Shy" on the Pointer's album?

GV: Yeah, on "Slowhand." "He's So Shy" was recorded quite a while ago. That was a funny session. We had to rent a drum kit. The regular tracking date required a synthesizer-like texture, but it didn't come out that way with the musicians. So we ended up keeping only the drums, and overdubbed everything else, which was mostly synthesizers, but also had strings, cowbell, and backgrounds.

There was a lot of synthesizer tracks, just to get the textures through doubling and tripling of parts — bring in a [Sequential Circuits] Prophet one day, a [Yamaha] CS-80 the next day to double that, and so on. In those days we were using only 24 tracks. We ended up filling up all the blank spaces in the tape with extra parts. One track, for example, may have had eight bars of backgrounds, four bars of strings, 12 bars of synthesizers — just crammed on this one track. About the only effect I really used on the whole tune was tape-slap for the staccato signature line, just to give it some bounce.

R-e/p: How do you record the Pointer Sisters' vocals?

GV: I can't really mike them individu-

AMBIENCE AND COLOR . . . continued —

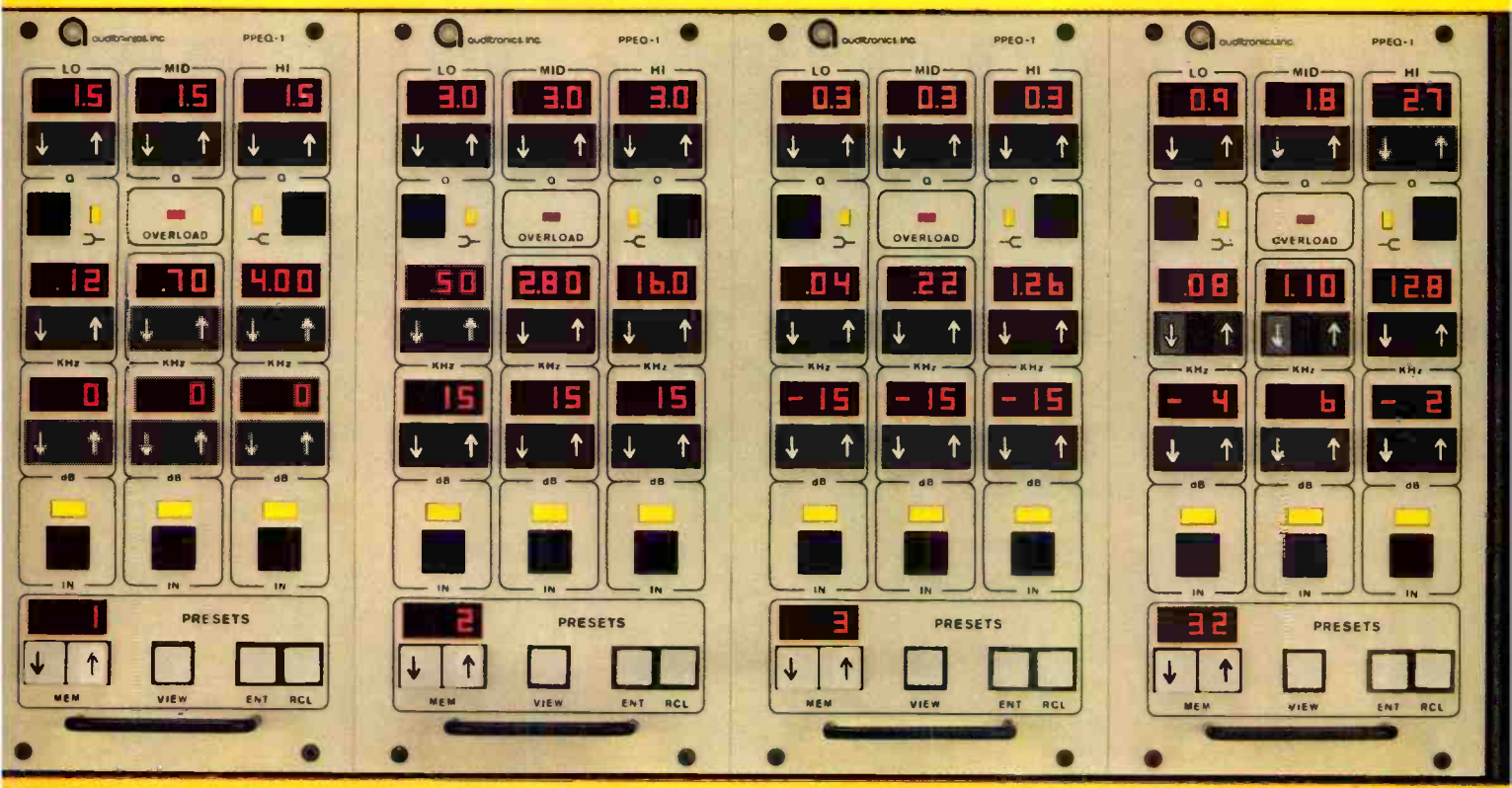
"The sounds need something like that, because the bass-drum waves are so long and such a low frequency, like 40 Hz, that the short time allowed by the machine to make the sound isn't enough to really even get the wave going.

"The toms are actually the most lacking. You want them to go 'oommm,' and instead you get 'ump-ump-ump.' I'm going to try the same approach with the rest of the kit. For one project, we brought a drummer in to do just the fills.

"The snare tends to need some life, too. I set the snare in a live room with hard, cement walls, and placed a Sony C-500 microphone in there to pick up the ambience [from a Yamaha 'bookshelf' speaker replaying the snare track]. It really sounds pretty good now, but it's still just a little bit too metronomic.

"Usually when I record any computer sounds, I try to do something with them. The short staccato sound is a little too cold. I've recently tried hooking up Syndrums to the output of the DMX, so those two sounds are mixed together. The possibilities are virtually endless." ■■■

EQUALIZATION
A NEW CONCEPT



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ally for a couple of different reasons. They sing really loud, which means I have to have some space between them and the microphone. It's not a lot of space, because they're in the iso booth, which is only about 4 foot by 7 foot. They also like to move around when they sing, so I can't regiment them to one place. By placing them back from the mike a little bit, they can move their heads around.

I generally mike whoever the lead singer is for the song with a Neumann U-67. A single Telefunken 251 about a foot-and-a-half to two feet away works good for the backgrounds, which is usually the three of them circled around the microphone singing as an overdub. I'll generally have to get the microphone a little above their mouths, because when someone is singing that loud, you tend to get pops in the mike.

Other times they'll do all the parts simultaneously. It's not unusual, when we're cutting the basic tracks, that what's supposed to be the guide track turns out to be the final vocal, because the performance they give right there is the magical one.

"Attitude is one of the most important aspects of being an engineer. An engineer is reponsible for maintaining a particular vibe in the studio. Whatever the vibe is in the room, that's what goes on tape."

R-e/p: Do you ever put any effects on the voices?

GV: We'll double the lead vocal sometimes, and occasionally put a flanging effect or slap echo on during the mix. But they are incredible singers; they blend really well. We'll play them a demo of a tune, and they'll write down the words, go into the studio and, without saying a word about parts, immediately sing all the harmonies.

R-e/p: The Pointer Sisters currently have a couple of extended-play singles on the charts. Do you do anything differently on the date if you know the record will be a long single version?

GV: The trouble is that a lot of times you just don't know that up front. You start out with a four-minute tune, and when it starts to go big, you have to remix it for the dance version. You start doing crazy edits to make it longer — maybe re-do a section a different way. Not by bringing in the band and recutting it, but simply by re-editing it. If you want to make the chorus twice as long, you get the 24-track tape out, mix it down and record it to the two-track one way. Then run the 24-track back, remix it again to the two-track, and splice the two choruses together to make it twice as long. Then maybe you'll go back to the middle of the tune and run it down again without the vocals in it. In other words, I mix all sorts of little sections, cut them up, and rearrange them again.

It's a lot of fun the first time, because I never know if it's going to work. And I've developed some tricks for editing, so it's pretty easy. But if I have to re-do it *again*, it becomes drudgery trying to remember, "How did I do this? How did I do that? Where did I get this piece from?"

R-e/p: In some ways, while doing such editing you're almost taking the role of an arranger. Do you have a musical background.?

GV: Yeah, I have a fairly solid background. I was a musician, played both guitar and piano, and worked in a music store when I was going to high school. I learned quite a little bit about every instrument they had.

R-e/p: Do you find yourself drawing on that experience now?

GV: Subconsciously, I think I do. I know where the sound comes from, what notes they're playing. I know the effect a certain kind of head will have on the sound of a drum, certain kind of strings on the sound of a guitar, when a reed's going mushy. I'm able to know what the problem is with an instrument when I hear it, and can offer suggestions on how to fix it quickly.

R-e/p: You mentioned earlier that you had developed some editing tricks. Any

CROSS-COMPRESSED GUITAR SOUND EFFECT USING A LESLIE SPEAKER CABINET

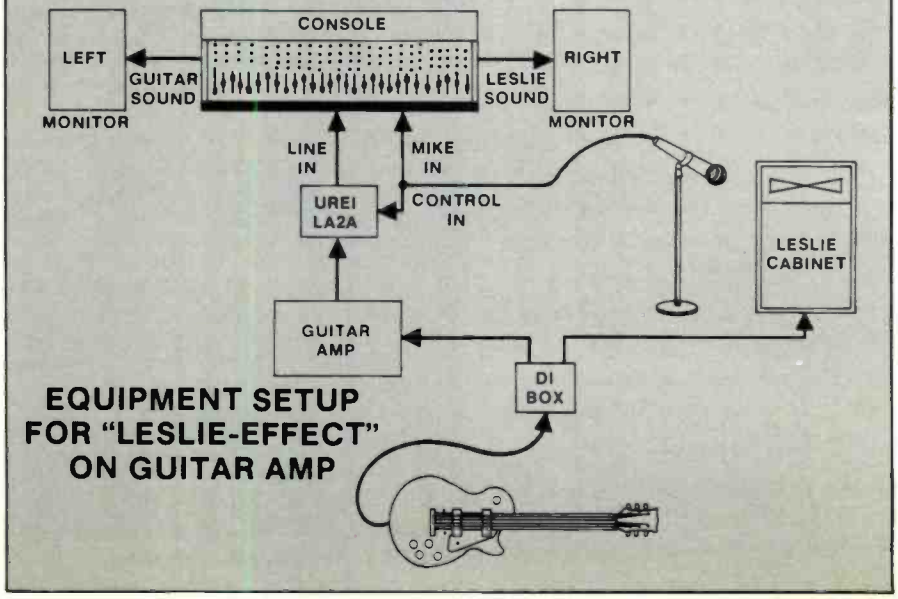
The Pointer Sisters' song "Slowhand" features a liquid sound flowing back and forth underneath the vocals, which sometimes appears to be a guitar and at other times like an organ. Gabe Veltri explains how he achieved that effect.

"While we were recording the tune," he recalls, "Richard [Perry] kept saying, 'What do you think we need on this? Do we need a guitar?' We had to add something to play with the voices, but not distract from them. It had to add a sensual sort of groove, to coincide with the lyrics of the song and the whole sound. We couldn't have anything jump out and scream: 'Here I am!' We originally called Andrew Gold to play guitar with his volume pedal, but he got sick that day. Instead we had Paul Jackson come in, and I tried to think of something that would work with his style.

"What I eventually did was to split the guitar signal, run one into a guitar amp, and the second signal directly through a Leslie cabinet. I miked the horns on top with a U67 about a foot away from the cabinet. That horn signal controlled a compressor [UREI LA-2A], which didn't compress the Leslie sound, but affected the guitar amp signal.

"Every time the live horn of the Leslie came around toward the mike, the increase in level activated the compressor, which in turn compressed the output of the guitar amp. The amp was panned to one side [of the mix], and the Leslie was on the other. That's where that liquid effect came from. You'd hear the whirling Leslie sound on one side, and then the guitar-amped sound changing level on the other.

"I don't remember the exact settings, but it was enough to compress the amp to a fairly low level. If the settings are too fast, the sound gets jerky." ■■■



EQUIPMENT SETUP FOR "LESLIE-EFFECT" ON GUITAR AMP

that you'd like to share with R-e/p *readers?*

GV: Like edits that my friends call "Hail, Mary" edits, because they didn't know how I did them, and they look impossible. The trick is that I have two or three marks on the front edge of the two-track tape machine about a foot to a foot-and-a-half apart, and equal to the length of tape for one beat of music, let's say. I make sort of a scale or ruler on a piece of tape laid out on the top on the machine. If there is no definite sound, like a bass-drum beat, where I need to cut, I just pull the recording tape out from the reel, measure it against the "ruler" for one beat, and cut off a copy. The thing was, nobody saw the marks I had on the front of the machine, and they couldn't understand how I knew where to cut!

Since we're talking about edits, on "Fire," the Bruce Springsteen song that the Pointer Sisters recorded, there's a line in the song that goes, "Drivin' in my car, turn on the radio . . ." Somebody had the bright idea of putting in the call letters from some radio station, I think, in Chicago. So Anita [Pointer] came in, we overdubbed the letters, cut that into the master, and made a dub of it for the radio station. Well, other radio stations heard it and wanted it, too. I ended up making tapes for about 200 radio stations. Anita was standing in the iso booth singing the different call letters to

200 stations!

R-e/p: How did you become involved with the music sessions for the movie, The Toy.

GV: I was working on a new album project for Taste of Honey with Trevor Lawrence, when he told me that the next day we'd be starting very early. We were doing a date for *The Toy*, a movie with Jackie Gleason and Richard Pryor. We had to do the whole thing in one day, because they had gotten the opening tune at the last minute. This was the last possible day they were supposed to have it — like 11AM the next morning, so they could put it into the prints to send to the theaters.

R-e/p: Since basically you had no time to prepare for The Toy *project, did yu arrange it as you went along?*

GV: They [Lawrence and Frank Musker, who wrote the song] had the basic idea of what they wanted to do. We brought in an Oberheim DMX drum machine, because they already had a drum pattern worked out. Then we added a real bass line, a couple of guitars, a solo, the vocal, and backgrounds. It was mainly a "head arrangement" that got worked out as we went along. We just kept overdubbing until we got to the point where we felt it was done.

Trevor Lawrence was producing, and also played sax on it. Jeffrey Osborne was the singer. Poor Jeffrey ended up waiting all day to put his vocal track down, which he finally did in a couple of takes. We started at 9 AM, and by the time we were ready to mix it was about 4 AM the next moring. I had to take two hours off to sleep before I tried to mix.

R-e/p: Your ears must have been shot after that long at the board.

GV: Well, that always happens when you do a session like that. You can hear the parts even if the tape isn't on the machine! Just working on one tune over and over for 12 or 14 hours, how are you supposed to know what's there, or how loud one track is in relation to another? I mean, I heard the song while I was driving home in my car [laughter]. ■■■

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MANUFACTURING MAGNETIC TAPE

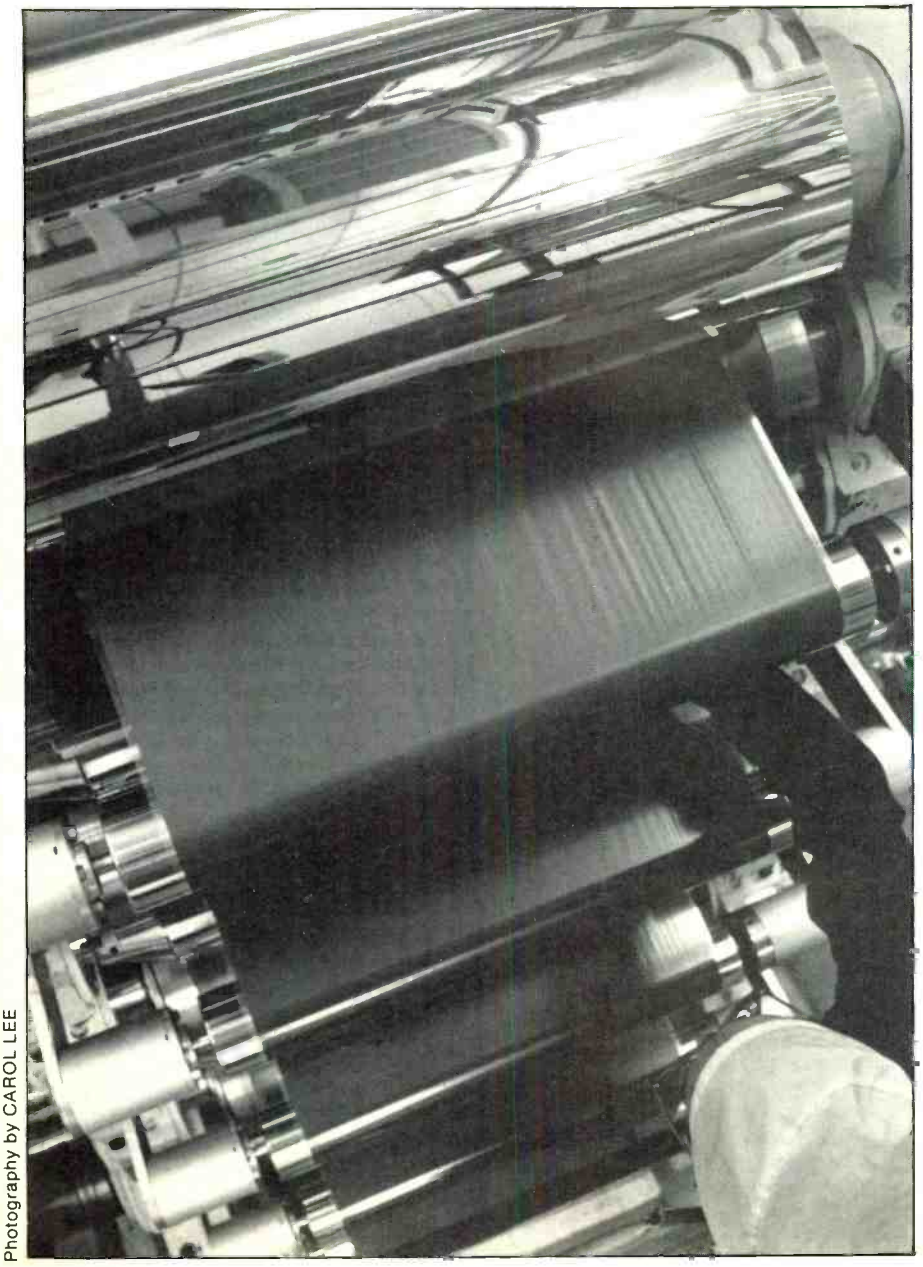
A Look Behind the Scenes at BASF's New England Ferric and Chrome Formulation Tape Plant

by Paul D. Lehrman

Just as everyone who takes a picture uses film, everyone who records speech or music makes use of magnetic tape. But there's a difference: professional photographers have to be aware of the various subtleties of film sensitivity, emulsion formulations, grain size, and developing techniques, while few audio or video professionals are that familiar with the technical parameters of their storage medium. Fewer still understand what goes into the making of recording tape. Most of us know what we like — or dislike — about one particular tape formulation or another, but few of us have ever bothered to find out what makes a good tape, or why magnetic tapes are different from each other.

And we may consider the converse is true as well. Do the companies that manufacture the material on which we record know, or care, about us — the engineers and producers who are going to use their product — and are they sensitive to our needs? Fortunately, the answer is yes.

To prove that this is in fact the case, *R-e/p* paid a visit to the magnetics factory of BASF, in Bedford, Massachusetts. BASF is a German-based chemical giant, founded in 1865. Total annual sales are over $16 billion. It employs over 100,000 workers in 63 countries to make some 5,700 products, ranging from plastics, to pharmaceuticals, to fertilizers, to synthetic fibers. About 10% of the company's output is magnetics: audio and video tape; and computer tape and disks. The Bedford site, located in that strip of high-tech companies that surrounds Boston known as Route 128, is where production of all of BASF's American magnetics materials takes place.



Photography by CAROL LEE

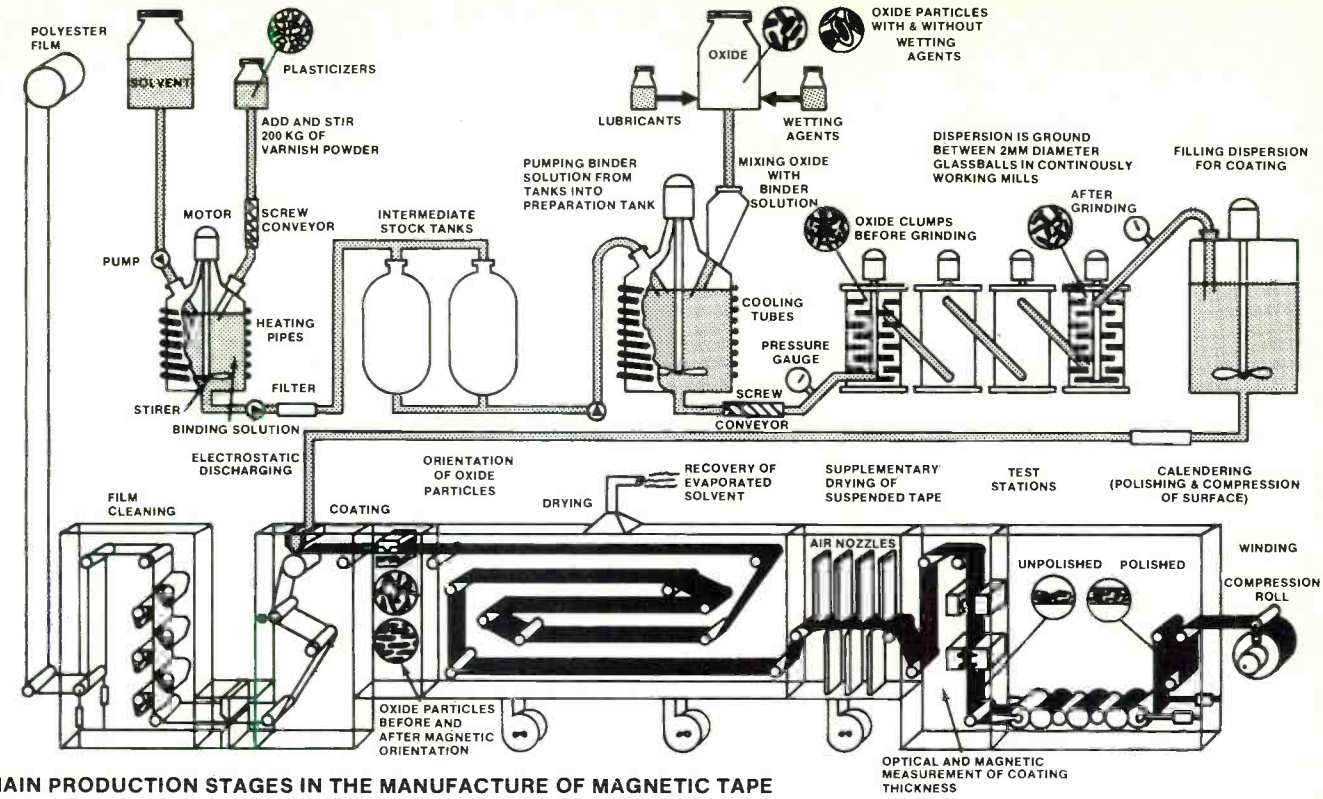
In the audio world, the company is best known for bulk cassette duplicating tape, ferric and chrome, and it commands about a 30% share of that market. Among its customers are record labels, including CBS, RCA, Warner Bros, and A&M, plus a host of independent duplicating companies. BASF also sells bulk tape and cassette shells to Mobile Fidelity Sound Labs for that company's real-time-duplicated "audiophile" releases. With five different cassette types in its consumer line, BASF has a respectable share of the retail market as well, and recently has started selling finished cassettes in non-standard lengths to the pro-audio world.

To complement its efforts, the company has introduced a line of precision calibrated test cassettes, as well as chrome-based bin-loop duplicator master tape in quarter- and half-inch formats. Reel-to-reel tape is an important part of the company's output too, although its current product line includes only the quarter-inch format, both in standard ferric and the new, chrome-based "Extra Efficiency" tapes, neither of which currently is made in the US. Plans are afoot to eventually expand the line into multitrack widths.

Key to a Good Product

When a company like BASF makes tape, how does it know that it is making the best product possible? The answer breaks down into two areas: control and research. The control aspects extend to the 53 raw materials that go into the finished product, to the automated manufacturing processes, and to the end result.

Because BASF is involved in so many aspects of chemical production, it can afford, like few other companies, to indulge in pure research, without assurance of immediate financial return. Such research, which with audio tape dates back to the Thirties, has led to the company's championing of chromium-dioxide tapes, when other firms were going with cobalt-doped, ferric "chrome equivalent" coatings. Today the com-



MAIN PRODUCTION STAGES IN THE MANUFACTURE OF MAGNETIC TAPE

pany is constantly looking at new ways of formulating all types of storage media, whose commercial applications may yet be several generations down the road.

The first step in controlling the quality of an audio tape is the preparation of the magnetic particles that will go into the coating. The particles, which are in powder form, must be small, uniform, and have no side branches or "dendrites." The smaller the particles are, the better the high-frequency response. Chrome particles are smaller than ferric, but that is of no advantage in tape that's going to be used at speeds higher than 7½ IPS. (However, they do exhibit 10 dB more headroom at 12 kHz at 3¾ IPS, according to Bob Piselli, national sales manager for professional products, which is why they are being used in the new high-speed loop-bin tapes designed to run at duplicating speeds of 32 and 64:1.)

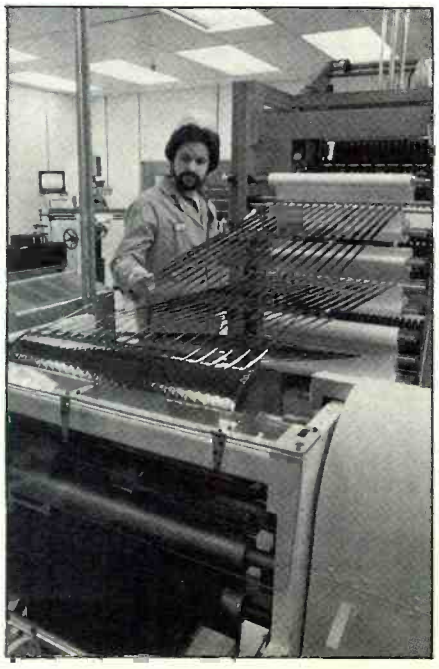
Making the particles uniform ensures that print-through is reduced to a minimum. When particles are of different sizes, the larger ones with stronger magnetic fields will "write" on the weaker particles, both within the coating itself and, more significantly, on adjacent tape layers.

Unlike the majority of tape manufacturers, BASF grows it ferric crystals in an alkaline bath. According to Bob Donadio, director of research and development at the Bedford plant, this alkaline technique gives better control over crystal growth, and results in smaller and more consistent particles with higher magnetic flux (as well as more benign by-products) than the more commonly used acid-bath technique. "It's not as popular," Donadio says, "because it's more costly."

Side branches, because of their randomness, and because they exhibit what's known as "magnetic drag" when they pass over a head, cause noise. "Once the particles are produced, there's no filtering," Donadio adds, "so they have to be made right, or you're in trouble."

"The quality of the process is only as strong as the weakest link in the chain," he explains. "For instance, we buy our pigments [magnetic powders] on the open market from whoever makes the best. We're not hung up on our own product, which is made in Germany. We and DuPont are the only ones making chrome, so that makes the choice easier, but the ferric particles may come from Pfizer, Hercules, or any of a number of places. Each company seems to make one pigment that's very good for one application. We work closely with our suppliers in open relationships, even though we compete with each other. Everyone knows everyone else's strengths and weaknesses, so there's no great rivalry or secrecy."

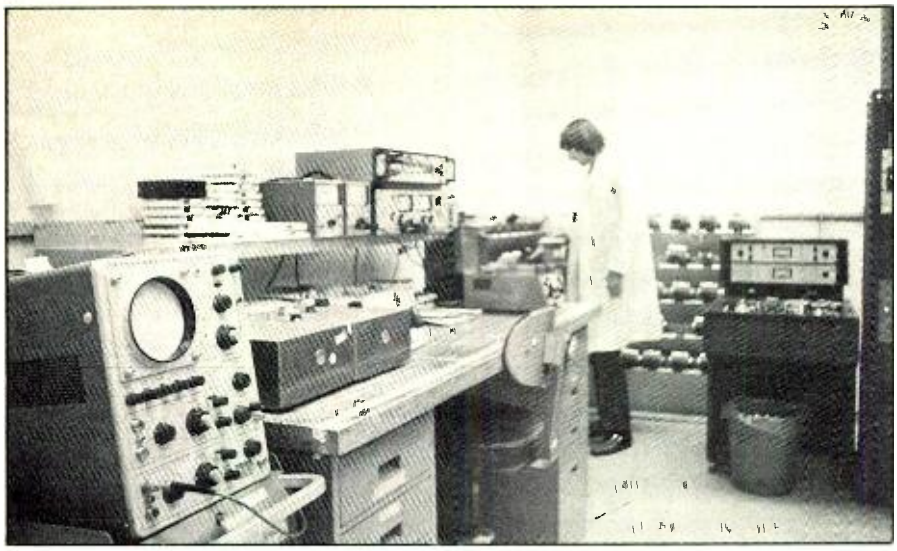
Tape slitting process to close tolerances



Next is the formulation of the binder. This is made from polyurethane and PVC (poly-vinyl chloride). Polyurethane has a crystalline molecular structure, while vinyl is more amorphous. "We combine them for durability, low-temperature flexibility, and adhesion," says Donadio. "We look at the molecular size and weight and the flow characteristics when the materials are placed in solution. The size and weight of the molecules determine the mechanical properties, while the flow characteristics affect how well the magnetic particles disperse themselves in the solution."

Blending of Materials

All of the raw materials are tested as they come in, according to Donadio, and there are differences between lots. He can blend lots together, or mix and match the various components so that they complement each other. Besides magnetic powders and binders, he has to think about the addition of conductive agents to discharge static electricity in the final tape, stabilizers to prevent breakdown under ultraviolet radiation (which can happen, for example, when a tape is left on a car's dash-



Many chemical and mechanical tests are performed by the quality-control laboratory

board on a sunny day), and even fungicides, to discourage the tendency of certain materials to attract mildew.

"All of the work is done very scientifically with computers, but sometimes the best computer is still up here," says Donadio, pointing to his forehead. "There's a little bit of magic left in the process, almost like brewmastery."

The next step is selection of the plastic base material. It has to have specific tensile properties: elongation and "creep" characteristics, and at a certain point it has to break. About a 5% stretch on polyester film should be recoverable — that is, within the material's plastic "memory" — at low load levels. If that figure drops, the tape will become permanently damaged when, for instance, a tape machine comes out of rewind quickly.

A new concern has arisen in recent years as tape coatings have gotten thiner. Pure polyester has a high coefficient of friction; a tape backed with such a material wouldn't move in any tape transport. As a result, small particles of inorganic materials like calcium carbonate (chalk) or calcium silicate traditionally are added to the base to act as lubricants. It seems, however, that the size of the "bumps" that these particles create has reached a significant proportion of the thickness of the coating — a particle of calcium carbonate may be 150 microinches in diameter, while the coating may be only 400 microinches —and this causes unevenness in the coating, which results in dropouts. The solution, which has been used for a while in videotape manufacture, is to eliminate the bumps by not using inorganic particles at all, but instead to chemically treat the surfaces of the polyester with an ultrathin layer of a clear, smooth, proprietary material.

"Luckily for us," says Donadio, "we make other types of media besides audio and video tape. So if we come up with a bad batch of something, we can use it for computer tape or another less-critical application. But that rarely happens."

The binder, to prevent flaking and rub-off, has to stick very well to the backing. Getting that to happen requires a thorough understanding of the chemistry of all the materials, as well as a certain amount of "cut-and-try" work. To make things even more complicated, binders become much less adhesive when the pigments have been mixed in. Donadio looks for proper adhesion at the manufacturing stage, and then subjects the tapes to environmental testing and laboratory "aging," checking to make sure that no loss in adhesion occurs.

Environmental testing is a crucial factor, both at the research level and with the finished product. A special room referred to as a Climatic Controlled Chamber houses a number of audio and video decks that run tape through several play and wind cycles at extremes of temperature and humidity — typically 110°F at 85% relative humidity. "The decks usually give up before the tapes do," laughs Piselli.

Ensuring High Quality

Quality control along the assembly line is maintained in a number of ways. Probably most significant is that the tape is made in lots of billions of feet —the plant's production of 180,000 cassette pancakes per week, put end to end, easily would reach to the moon — so that all of the ingredients are mixed in huge quantities (some 20 tons of raw materials pass through the Bedford plant every day), thus ensuring homogeneity and consistency. "A manufacturer who makes only a few feet of tape a day has problems with that," notes Donadio.

Another important check is made of the thickness of the coating as it is being applied to the backing: too thick a coating means that the signal level will be too high, resulting in print-through; while too thin a coating weakens the

— *continued overleaf . . .*

signal strength. On BASF's coating machines, sensors are mounted that continually monitor the coating thickness and, at the slightest sign of drift from the norm, adjust the mechanism accordingly.

After the coating is applied, the tape passes through a strong magnetic field which aligns all the oxide particles. For maximum output, all of the particles have to point in the same direction.

Still another checkpoint is where the 28-inch-wide rolls of finished tape are cut into smaller ribbons. "The knife blades have a charted life," explains Piselli. "We replace them at regular intervals. Whenever a set is changed, the width and quality of the cuts are checked with an electro-optical device: a microscope hooked up to a computer."

After cutting, samples of the tape are drawn from production and sent to the quality-control laboratory, where they are checked for both physical and magnetic characteristics. An in-house designed testing deck is set up to automatically take readings from the pancakes of sensitivity, distortion, noise, non-uniformity, and both high- and low-frequency output level.

Tapes are also checked through the electro-optical system for edge roughness, and are run several times through tape decks and then checked for oxide rub-off. There is also a tester that runs finished cassettes backwards and forwards, at slow and fast speeds, a total of 10 times to make sure they will hold up under normal use.

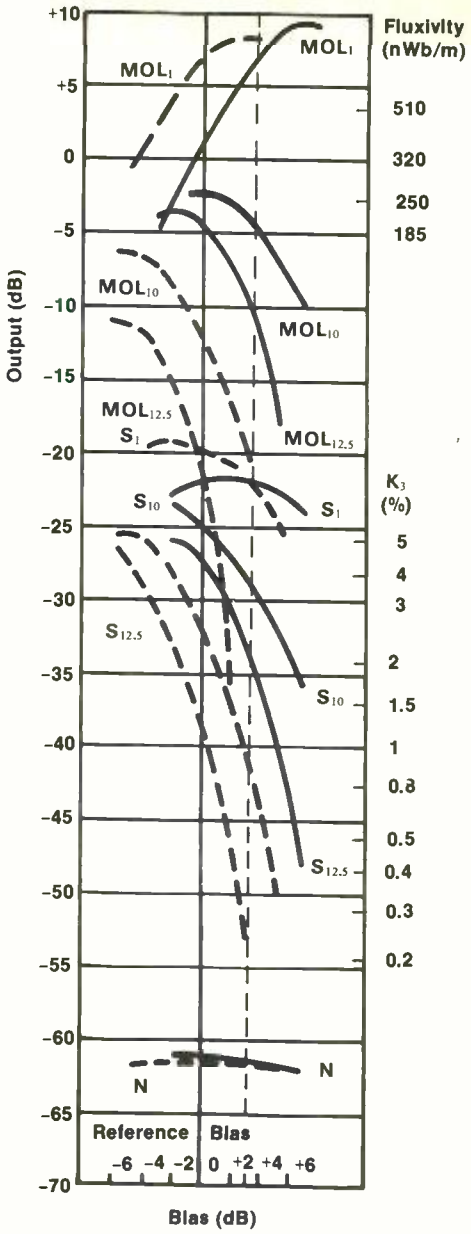
Limits of Ferric Formulations

While BASF is always researching new ways to produce standard tapes, it is also looking into the future of audio technology.

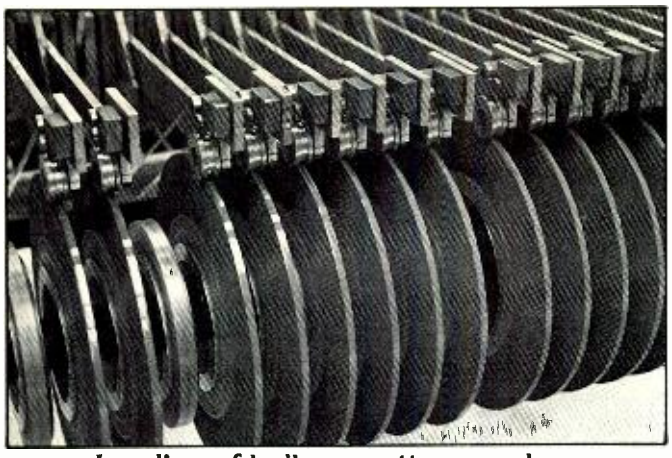
"We're at the limitations of ferric

CHROME-DIOXIDE TAPE FORMULATION

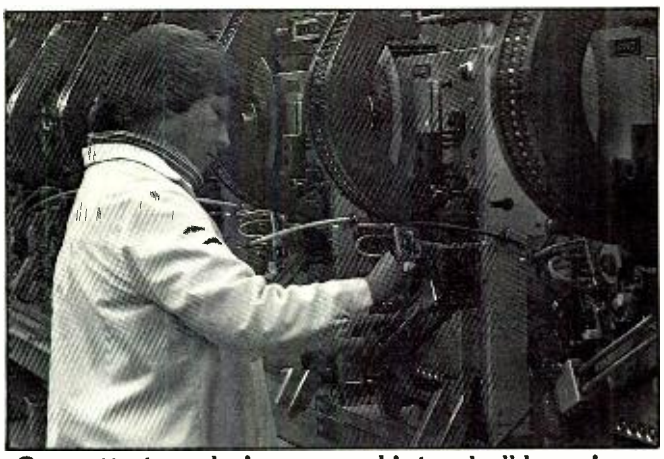
BASF has developed a new chromium-dioxide formulation, known as LPR35Cr, for use on reel-to-reel machines, and also for the preparation of loop-bin masters recorded at 3¾ IPS. The preliminary technical data shown here relates to the recording performance specification at a tape speed of 3¾ IPS, using a head gap of 7 micrometers, a playback gap of 3 micrometers, a 2.2 micrometer track width, 50 + 3180 microsecond replay equalization, and a reference fluxivity of 320 nWb/m. As can be seen from the graphs, performance at this slow tape speed is on a par with that available from conventional ferric formulations running at 7½ IPS. The diagram shows the main tape parameters as a function of bias current, while the values below were measured at a reference bias ratio of 0 dB for the reference C264Z tape (dashed lines), and +2.5 dB for LPR35Cr:



TEST PARAMETER (QUARTER-INCH TAPE)	C264Z	LPR35Cr
MOL1, Maximum Output Level (3% Third Harmonic distortion at 1 kHz)	+7.2 dB	+5.2 dB
MOL10, Maximum Output Level at 10 kHz (saturation)	-12.6 dB	-2.8 dB
MOL12.5, Maximum Output Level at 12.5 kHz (saturation)	-24.0 dB	-8.5 dB
S10, Sensitivity at 10 kHz	-12.5 dB	-6.5 dB
S12.5, Sensitivity at 12.5 kHz	-20.0 dB	-10.2 dB
S/N10, Signal-to-Bias-Noise Ratio at 10 kHz	48.9 dB	58.4 dB



Loading of bulk cassette pancakes



Cassette tape being wound into shell housings

tapes now," Piselli says. "But we can continue to approach the ultimate state of the art, economically, with other materials." Digital recording, for example, places very different requirements on a tape. Says Donadio, "There you're not looking to pass a wide bandwidth of frequencies; you're only looking at one. And the tape is either saturated, or there's no signal. The limiting factor is the available resolution on the tape, which ultimately boils down to particle size and strength." Therefore, according to Piselli, "Digital tapes will probably be chrome. They have better packing density, and the particles are more uniform."

Another technique currently being investigated is the use of exotic metal formulations, applied to tape in microthin coatings by an evaporation or "sputtering" process. When applied this way, a tape using such metals, which could be nickel, cobalt, or even gold, would have a high coercivity, and could deal with extremely short wavelengths. The technology to do this exists today, but ways to make it commercially viable have yet to be developed.

Bob Piselli, who has experience both in recording studios and record companies, sees BASF's research and manufacturing progress as important to the entire industry. "We want to develop the state-of-the-art to its maximum potential," he says, "so that the tape medium can reproduce the same music as the human ear can hear. We meet with our customers on an ongoing basis, and ask them what improvements they would like to see. We get comments from record companies and retailers, most of which are very positive, and they all want more of our new formulas. Cassettes, in particular, have improved vastly in the last year or so. What we are trying to do is to make as good a product as we can, within certain cost guidelines."

Which, of course, is just what everyone in the recording business wants, too. ■■■

Photo by GARY WHITELOCK

CONCERT SOUND

"On The Road Again"

THE WILLIE NELSON AND FAMILY TOUR

Sound System Electronics Design by Innovative Audio at the San Diego Sports Arena

by David Scheirman

It is perhaps safe to say that the Willie Nelson organization recently has been spending more production dollars for sound and lighting than most other country-oriented acts. Nelson's house mix engineer, Mike Garvey, of Dallas, Texas, explains his part in the decision-making process when it came down to choosing a sound system for the tour: "It's not as if I have a blank check or anything, but I was given the authority to go with the best. We are using Innovative Audio to provide us with Gamble consoles and electronics and stage monitors; the main house speaker system is from Showco.

"I could not find everything I wanted from one company, so we ended up with two. It is an unfortunate situation to split it up like that, but I have a commitment to Willie to give him the best I can find, and that is how it has worked out."

Showco House Speaker System

Garvey feels that Willie Nelson's touring requirements are a bit unusual, when compared to "typical" concert tours — if such a thing exists. "We aren't like Van Halen or somebody like that, where you play the same kind of venue in every city you do," he offers. "We play an extremely wide variety of places. We might be in a little-bitty club with 150 people one night, and be looking at 50,000 folks outdoors the next day. I needed a speaker system that could handle *anything*.

"A lot of the available systems are either good for short-throw, wide-dispersion situations, or else they are set up to be a long-throw system and end up being very 'beamy.' I need something very versatile. And, doing a show which is primarily vocals, I need to get those mids and highs all the way to the back of the hall — really get that articulation *out* there. The Showco pyramid system is one I am really confident in, and have had a lot of experience with." (See: *R-e/p* December 1981 issue; "Showco Doing The Audio For The Rolling Stones Tour" — *Ed.*)

Garvey, a former Showco employee, has a particular preference for the Showco method of stacking the mid-range horns inside the cabinets, and also is pleased with the low-frequency projection. "The pyramid box is actually a *vented* horn-loaded box . . . it has a 10-inch port. You get the best of both horn-loading and reflex action," he claims.

Innovative Audio's Front End

The choice of Innovative Audio came about largely due to the company's reputation as a reliable supplier of consoles manufactured by Jim Gamble Associates. Rapidly becoming the favorite of many experienced concert sound engineers, Gamble custom-built consoles are quite popular due to their many utilitarian features, and excellent sonic quality.

But why did Garvey go out of his way to hire a separate sound company just to gain access to a Gamble board? "There are a whole bunch of reasons!" he recalls. "It is really clean . . . it seems to be a real *bright* sounding board. To my ears, the console seems to have a different sound to it . . . a 'faster,' cleaner sound; it sounds better than any other console I have ever used. And, there is the built-in spectrum analyzer, onboard graphics, and so on (Figure 1).

"The graphic is the cleanest I have ever used . . . if I go even 3 dB up or down on a given frequency, I figure that something is wrong somewhere else. And the console is really reliable. In two years, I have only had a couple of failures . . . and one of them wasn't the board's fault; some guy threw a beer on the board!"

Innovative Audio's David Morgan echoes Garvey's comments about the Gamble console's reliability: "This board you are looking at [Figure 2] has probably done about 1,500 shows. We have eight of these available to us, and very rarely have *any* down-time."

Morgan says that the consoles were designed with easy field servicing in mind. "The modular panels are quickly and easily removed. And you can find many of the circuit board components at any Radio Shack. But don't get me wrong, there's some expensive stuff in there. The capacitors are Sprague-type solid tantalum; the resistors are Allen-Bradley 1% tolerance. It all adds up to a -130 dBv equivalent input noise signal ratio."

Innovative's consoles have been modified somewhat. "We work with a gentleman named Dirk Schubert," Morgan explains. "He, in conjunction with our company, has taken the Gamble board and made it only the beginning of what

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Figure 1: Gamble HC40-24 house console, with built-in 16-ladder spectrum analyzer for measuring system response.

we feel is the cleanest concert audio system available.

"As far as I know, Innovative can field the only fully transformerless, DC-coupled PA system in the world. There are *no* audio transformers anywhere in the signal chain, except for those in the microphones themselves. We do it all with BIFET active-balanced circuits, and the circuits are the proprietary designs of Innovative Audio, Dirk Schubert, and Jim Gamble."

Morgan claims that this circuitry enables the system to produce a guaranteed maximum of 5% phase distortion throughout the entire chain, measurable at any point in the system from the mike input to the crossover output. (Typically, an audio frequency amplification system will have a measurable phase distortion of between 5% and 40% at some point in the signal path).

"The combination of the Gamble consoles and our own crossovers [18 dB Thompson-Bessel minimum-delay, equal-phase-shift filters] give us an extremely transparent system," Morgan continues. "We use more chips, more integrated circuit amplifiers, than any other company would use transformers. Each Gamble console has over 1,600 ICs, and then we pass it through all active-balanced circuitry, down the snake, and into our amp racks, each of which has our own active-balanced line amplifier built into it. This way, each amplifier channel 'sees' the same signal level, and we eliminate any damping errors, rather than trying to cut the same-length speaker wire or whatever, like other people struggle with."

Figure 2: The original prototype Gamble SC-32-16 board, a veteran of perhaps 1,500 road shows.

House Mix

For the Willie Nelson tour, Mike Garvey's house mix was taken from the Gamble HC40-24 40-input console and sent through the Showco equipment rack, which contained two Showco four-way electronic crossovers, and dbx Model 165 compressor-limiters (Figure 3). The stereo mix utilized only two of the console's 16 main outputs, while two more were used to feed a stereo cassette tape deck for recording show mixes. Garvey used one of the eight effects busses to drive a MICMIX Master-Room rack-mount spring reverb unit.

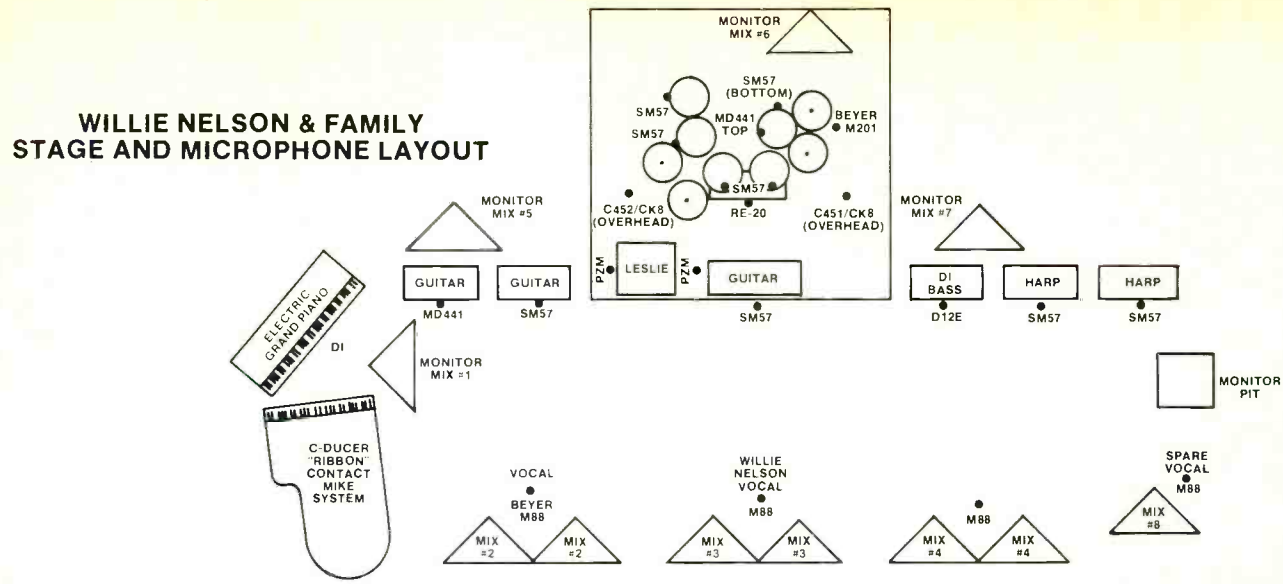
"This is a real bare-bones, straight-ahead approach to mixing," Garvey offers. "That reverb unit is my only effects device for signal processing; I don't use any pitch shifters or delays, or stuff like that. That new Master-Room, Model 515 I think it is, has a really good plate sound to it, and it allows me to get Willie's voice sounding just like the record on things like 'You Were Always On My Mind.'

"But I am not into cluttering up the sound of Willie and the band with a bunch of garbage. As far as we are concerned, if people come to the show and can hear the words at the back of the arena, and leave knowing they saw Willie and had a good time, then we have done our job."

When this writer pointed out the obvious lack of outboard equalization in Garvey's house rack, he stressed that the Gamble console's onboard third-octave graphics were completely adequate. "I just plain do not need anything else on it," he says. "Used to be that only cheap music-store boards had a built-in graphic on them, but this one is really *hot*. It is extremely accurate, and is right there near my master faders where I need it."

According to the console's designer and builder, Jim Gamble, these two graphics are normaled to the stereo main outputs. "Every fourth slider is

Figure 3: House electronics rack with Showco crossovers, dbx compressors, and MICMIX Master-Room reverb.



nel. We developed this amplifier in conjunction with Cerwin-Vega, because we needed a device with an ultra-high slew rate, and this one came out with a rate of 80 to 90 volts per microsecond; we use it in both the house and monitor systems.

"We use a single-point ground running through the monitor board; the whole system is grounded there. All of our amp racks have no U-ground, no Edison ground. That way, if there is a ground loop, it is a loop which someone else has introduced which is foreign to our system."

Stage microphone lines are fed into subsnake lines which terminate at the monitor board; the house console is then fed out directly from the monitor board (Figure 6). Since the monitor board patchbay points are half-normaled, changing a patch at either the house or monitor console does not affect assignment at the other end. Quick-disconnect strip connectors are used throughout the cabling system. According to David Morgan, this cabling scheme, combined with the single-point grounding concept, makes for a very quiet, noise-free system. "You run the console wide open with lots of gain, and all you hear is a little bit of pure white noise from the electronics . . . absolutely no hum," he claims.

Monitor Speakers

A total of 11 floor slants were placed on stage for Nelson and the band (Figure 7). Six of the slants were located in the front line, spread directly across the downstage edge. Each box contained a single 15-inch JBL K130 (or E130), and a 2441 HF driver on a 2311 horn with a 2308 slant plate. Additionally, each unit had one JBL 2402 "bullets," or high-frequency radiator. Nelson's pair of floor slants, however, was equipped with 2482 drivers. "It is a sound he likes," says Jennings. "He prefers the sharper midrange sort of sound, so we always give him '82s".

The monitor cabinets are bi-amplified, with the tweeters crossed over passively with an 8-ohm L-pad. The JBL 5234A electronic crossovers are backed by networks built by Innovative, utilizing LC/RC passive circuits to maintain accuracy and protection. The speakers are covered with a utility-gray indoor-outdoor carpet, which rendered them practically immune to nicks and scratches. Protective face-plate covers are secured with Sessions latches, which had wheels on them; once the covers were in place, the monitor speakers were ready to roll off to the truck loading area.

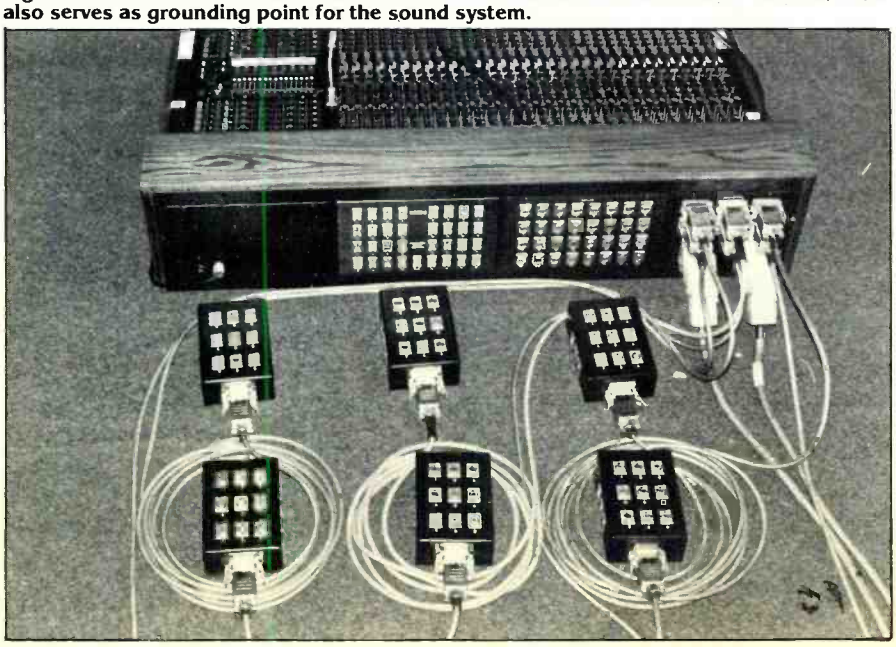
Showco/Innovative Cooperation

With two "competitive" sound companies on the road with the same act, I wondered whether there might be any friction between them. "Absolutely not," says Showco's representative Little Joe, who took care of hanging and checking out the house speaker system. "This really is like a family. We're here not because the company said to be here, but because we all really want to be doing this gig."

House engineer Mike Garvey echoes his sentiments: "This is almost a personality tour, you could say. A guy's attitude is everything out here. It has sort of worked out that what we have here is a group of guys who enjoy each other's company, and all happen to be very good at what they do. It works out to be a very smooth situation."

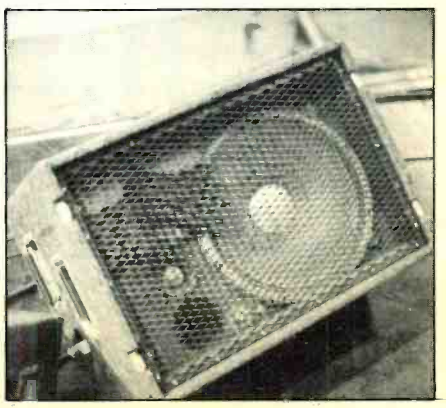
Things did flow rather well on the tour. While this was a large arena show

Figure 6: All subsnakes and main snake cables are terminated at the monitor console, which also serves as grounding point for the sound system.



GARY WHITELOCK

Figure 7: Single 15-inch floor monitor slant.



GARY WHITELOCK

Figure 8: A flown speaker system designed by Innovative Audio for the Doobie Brothers' Farewell Concert at the LA Forum, September 1982.

that I observed at the San Diego Sports Arena, the truck doors were not even opened until nearly 2 PM (although the rigging points already had been hung). After the lighting system had been raised, the monitor system was placed on stage and unpacked. The house mix position had been set up while the PA chain-motor hoists were rigged. After the house speakers had been flown, the stage gear was placed, and the house and monitor systems checked out with the pink-noise generators built into the Gamble consoles. No sound check was deemed necessary, and most of the crew had plenty of time for a leisurely meal and some video games on the crew bus before the show.

The sound of the concert was exceptionally smooth; very few surprises. Nelson's sharp-edged voice carried well all the way to back of the San Diego Sports Arena. The C'Ducer pickups made the acoustic piano stand out quite well, even though this particular arena is not noted for having exceptionally good acoustics for live music presentations. The Gamble consoles, the Innovative Audio monitors and the Showco pyramid speaker system comprised an excellent vehicle for presenting Willie Nelson's voice and music to a capacity crowd in an acoustically-poor setting.

Innovative Audio House Speaker System

Although for the Nelson tour Innovative weren't supplying speaker arrays, the company can handle full sound-system contracts. David Morgan explains the company's speaker system concept: "Essentially, Innovative Audio evolved out of Tychobrahe, which used to be one of the world's largest sound companies. When Tychobrahe was bought out, we took the old M-500 columns, which were one of the most common PA speakers of the early Seventies and overhauled them.

"The box used to consist of a three-way direct-radiator cabinet which was loaded with two JBL speakers, two 2482 drivers on a 2309 horn, and two JBL bullets. We made the cabinet a sealed enclosure which does not go below 100 Hz, and replaced the woofers with new JBL 2220s. We added two more tweeters per box, and changed over to 2441 drivers, which made a big difference in the cabinet's high-end response. Its cross-over points are 1 kHz and 10 kHz. The system gets its low bass from our 4×18 subwoofer bins; they each contain four Cerwin-Vega 18-inch speakers (Figure 8)."

The subwoofer box is 54 inches wide, by 53 inches high, by 30 inches deep, and is a 37.5 cubic foot enclosure. Morgan also commented that the company has plans to do an A-B comparison of the Cerwin-Vega 18-inch speaker with the new JBL 2240, with a possible change in mind should test results point in that direction.

The Innovative Audio house speaker system gives every appearance of being well-suited for both hanging and stacking; the rectangular cabinets are deep enough to support a fairly high vertical stack if the need arises. Half-sized three-way cabinets exist for hanging from the bottom of the flying speaker assemblies; full-sized sub-woofer boxes are placed on the outer corners of the stage area.

* * *

The Innovative Audio speaker system is typically representative of today's modular speaker systems; however, the company's system electronics can be considered to be unique. The transformerless signal chain is one more step in the direction of cleaner, more transparent audio-frequency amplification. And Mike Garvey, of the Willie Nelson organization, is not alone in liking what he hears. This system also is in use by Toto, Jefferson Starship, and a growing list of other touring acts. ■■■

All Photography: STEVE SATORI/S.U. Photo Center

THE AUDIO ARCHIVE AND LABORATORY AT THE UNIVERSITY OF SYRACUSE, NEW YORK

by David L. Moore

Ever wondered what happens to old records? Not the "Golden Oldies" of the Fifties, or the hits of the Sixties and Early Seventies that are treated almost reverently by some of today's AOR stations, but *old* records. Pre-World War II records. Funny how every now and then — like the Ragtime revival of a few years back — some of this music and its performers are brought back into circulation and exposure for a short period of time, and then quickly forgotten once the time has past. The fact of the matter is that quite a lot of old records, and the performances and artists that they made famous, simply disappear. For some reason, records always have been regarded in much the same way as the "paperbacks" of the publishing world: a best-seller one day, on the charts the next, and then out of print and forgotten in a few years. Hardly the respect given to those hardcover tomes which are catagorized, criticized, analyzed, Dewey-Decimalized, and then stacked away in libraries for the future regard of history.

There have been many reasons for this situation with recorded material — not the least of which is the commercial nature of the marketplace in which most records (or paperbacks) must survive. Other reasons include the problems of dealing with all of the various forms that records have taken over the years, and the fact that until the recent Sony/Philips development of Compact Disc technology, all of these storage forms have had serious draw-backs; they simply haven't been sufficiently robust to withstand the rigors of time.

Slowly, particularly in the past decade, recognition has been made of the fact that recorded history may well include phonograph and tape records made of early events and performances. This isn't at all a new idea, just one that seems to have been sidestepped in the interest of the vast commercial possibilities of record sales. Using the phonograph record to preserve the sounds of historic figures and events was an important part of Edison's vision during early development of the medium, and many "historical" recordings were made at that time. But again, owing to imperfections in the medium, and a general lack of concern for the value of older records, much of this material has been lost.

Compounding these problems, there were three (perhaps four) completely different and incompatible record formats used during the development of the phonograph as we know it today. Many consumers bought one kind of system, established a collection of records that could be played on it, and then new technological advancements came about which made the system obsolete. Consumer choices created a cycle of changes in hardware and software that left many consumers with equipment for which no new records could be found, or records for which no new playback equipment was being developed. All of these aspects, plus the fleeting popularity of most forms of commercial music, led to the loss of much of the sound recordings from the past nine decades.

Today we are poised on the verge of yet another complete change in delivery systems for pre-recorded music and sound. The digital Compact Disc unquestionably is the most perfect medium yet developed for the storage of recorded sound — far beyond the imaginings of recording's early pioneers. But, regardless of its container, or the magic of technology that brings the sound back to life, there is a cultural perspective to the recorded material that ultimately will determine if this sort of history is to be preserved, or discarded like those yellow newspapers found in Grandmother's attic.

The University of Syracuse, home of the internationally renowned Newhouse School of Communications, has long had an interest in the preservation of old recordings. In the Fall of 1982 the University and other interested individuals strengthened their commitment to historical sound recordings when they dedicated the Diane and Arthur B. Belfer Audio Laboratory and Archive. Describing itself as being "dedicated to the restoration and preservation of audio recordings, research in sound, and the collection and dissemination of recorded materials," the Audio Laboratory and Archive is equipped with some of the world's most up-to-date audio equipment, and staffed by a small, dedicated group which believes sincerely that the best of yesterday's, today's, and tomorrow's recorded history should be preserved for the enjoyment of all, and the enlightenment of those who may come after us.

An examination of such an archive and re-recording laboratory offers a new perspective on what an engineer or pro-

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ducer may be doing in the control room tomorrow. It also raises some interesting questions; some of them old questions just seen in a different light.

The Audio Archive Collection

The Archive collection was founded in 1963 with the acquisition of the Joseph and Max Bell Collection. Comprising a part of the Syracuse University Library system, the collection has grown to more than 250,000 phono disks of various types, 3,000 cylinder records, and 2,500 tape recordings. Aside from popular and classical music recordings from many eras, the archive material ranges from recordings of political figures, poets, philosophers, and famous actors, and also includes many transcriptions of early radio broadcasts, and unreleased disks from major recording studios. Although much of the material is from the pre-war era, newer recordings constantly are sought to expand the scope of the archive. Just dealing with the organization and preservation of such a massive amount of material is quite an undertaking. And one that is made even more difficult by the fragile nature of some of the early recording media.

Since part of the facility's goal is to make these resources available for research and to interested individuals (much like books in a library), the idea of re-recording selected material had to be explored in detail. Irreplaceable original materials cannot be played over and over again. As a result, one of the audio laboratory's purposes is to develop methods by which cylinders, disks, and tapes can be re-recorded, and then only these tapes subjected to the inevitable degradation caused by repeated playback.

Types of Re-Recordings

Like many aspects of the audio industry, there is considerable divergence of opinion among the experts as to what exactly should be done when an archival record is re-recorded. One school of thought, held closely by some dedicated hobbiests and a few people in the academic community, is that you should take advantage of every technological device available today in an effort to "recover" more from the record than those who heard it *in its period* may have experienced, and that the manipulation of the sound by electronic methods can best be done subjectively in order to achieve "a pleasing reproduction."

Control room equipment includes MCI/Sony console and tape machines in an LEDE acoustic design.

Another school, however, believes that you should only play back such recordings on the *original* equipment for which they were made, since the historical perspective of the period should not be jaded by a "better" reproduction. And others believe that objective technical standards should be developed and strictly adhered to, so that any re-recording of an archival record will be accompanied by a recognized methodology.

Audio archivists have had their fair share of controversy, and through annual meetings of their organization — the decade-old International Association of Sound Archivists — they have tried to establish some standards which can be universally recognized and adhered to (sound familar?). William Storm, assistant curator of the Archive and Laboratory at Syracuse, has long been a moving force in trying to have some sort of standards recognized. His perspective is analytical and objective; after years as a working musician, then a studio owner and chief engineer, Storm feels strongly that methodology, not mythology, will be the answer to ensuring that these historical materials are preserved in their proper sonic context. In a paper presented at the 1979 IASA Conference in Salzburg, Austria, he suggests that three basic guidelines be recognized, and that any re-recording be technically documented for those who eventually will hear it.

The first suggestion, not surprisingly, is that re-recordists adhere to what the audio engineering community would call "good technical practices": if tape is to be used for the re-recording medium, professional quality recorder/reproducers must be used, and routine maintenance and alignment carried out. The tape machines should be aligned to recognized record and reproduce curves, and these characteristics verified. A neutral monitoring environment should be established, and a standard for monitor speaker types and placement determined.

The last comment regarding monitoring speakers is particularly interesting, because for decades many modern audio recordists and producers have been suggesting the same thing — only to get buried under a wall of subjectivity.

The second guideline is that two different types of re-recording be recognized: the first being faithful to the best available reproduction off the record in its own era (called Type I re-recording); and the second being an attempt to recover a sound as close to the "live" sound of the original performer, as best as can be done with today's technology (Type II re-recording).

The last, and perhaps the most important suggestion made by Storm in his paper was that any re-recordings be approached objectively and scientifi-

The studio/re-recording area features variable acoustics to provide a controlled environment for replaying early acoustic phonographs, as well as recording live artists.

cally, almost as an anthropologist approaches his work. "Hobbiest" type restorations, no matter how sincerely motivated, cannot be recognized as having archival value unless, Storm offers, they are correctly documented and based on a sound scientific analysis of what should and could be done to an older recording in the interest of recovering the "live" sound.

Audio Recording Environment

The Diane and Arthur Belfer Archive and Laboratory has been equipped with these primary objectives in mind. The control rooms and studio are Chips Davis' LEDE designs, and MCI/Sony recording equipment has been installed. The main control room (a second one will be on-line later this year) is equipped with an MCI/Sony JH-618 console fitted with 16 input/output modules, half of them with MCI's Vari-Q parametric-type equalizers. A Crown BDP-2 Audio Microcomputer can be used for spectral and other types of analysis of material to be re-recorded. MCI/Sony JH-110B Series tape machines are used for transferring material to tape.

In addition, there are two pieces of specialized equipment available for archival recording and restoration. Made by Packburn, the first is a Transient Noise Suppressor, which takes information derived from both sides of the grooves of a laterally-recorded mono 78 (via a stereo pickup and pre-amp). If a scratch, noise, or other sonic discontinuity is found on one side ofthe groove, but not the other, then only the signal from the clean side is reproduced.

The other Packburn unit is a Continuous Noise Suppressor, a modified compressor/limiter that includes equalization curves for most of the old 78s. When the transition was made from acoustical recording and reproduction to electrical recording and acoustical reproduction in the mid-Twenties, it became much easier for the various record companies to change recording characteristics in the interests of a "hotter sounding" product. This resulted in the development of many different standards, some of which were completely undocumented. The Packburn unit's equalization makes it easier to deal with the more than 20 "standard" characteristics that were in use at that time, and outboard equalizers and parametric EQ fitted to the MCI console make it possible to "construct" almost any type of curve.

Old Mackintosh tube-type C-8 "Record Compensators" also are provided to give discrete control of rolloff and turnover frequencies, again in an attempt to match the wide variety of recording characteristics used prior to universal adoption of the RIAA characteristic. Monitor speakers are UREI Time Aligned Model 813s, driven by Mackintosh 2105 power amps. The studio also has a complement of calibrated AKG microphones, plus some Neumann and Crown PZM models.

Various size disks form part of the archive collection.

Re-Recording Techniques

Many different production techniques are used to make both Type I and Type II re-recordings. When the Archive Laborotory undertakes a Type I re-recording, first a careful analysis is made to determine the type of reproducer and horn to be used, taking into consideration the types of hardware in use at the time the record was first released. Often the record grooves are examined in order to determine the proper type of stylus to be used, and sometimes a stylus is made or modified on the spot. Most Type I re-recordings are then made by miking a carefully maintained acoustical reproducer of the same era with a calibrated microphone at a specified distance from the horn mouth. This procedure is carried out in the Laboratory's acoustically isolated studio. Since the acoustics of Chips Davis' designs are variable, sometimes the acoustical environment is altered to more closely approximate what might have been found in the parlors and living rooms of the day.

Walter Welch, one of the last living contemporaries of Edison, Berliner, and other recording pioneers, and director of the Archive and Laboratory, has developed a method by which two acoustical phonographs can be played back in synchronism, driven by synchronous motors, and reproducing the same disk. When the sync is sufficiently tight to avoid phase cancellation errors, the resulting higher output makes for better re-recordings, and reduces the impact of surface noise.

Type II re-recordings are made in the Laboratory's control room with electrical reproducing equipment. Often a wax impression is made of the record grooves, which then is sectioned and examined on an optical comparator in order to determine the proper stylus shape. The Syracuse Laboratory often has ordered a "custom" stylus from Expert Pickups, located in Surrey, England, specifying geometry, compliance and so forth, and then fitted it to a magnetic pickup "body" for use in re-recording of just one particular disk.

Again the record is analyzed carefully before re-recording is begun. For example, if spectral analysis of a given record from the electrical era shows many high-frequency peaks above 5 to 7 kHz, it can be safely assumed that enhancing this information will not bring back a "truer to life" sound, because we know that the microphones of that era had no usable response in this HF area (not to mention the rest of the signal chain). Rather than just reaching for the EQ control, many times experiments are made to change the mass/compliance relationship of the pickup arm and stylus assembly.

Both Type I and Type II re-recordings involve a great degree of empirical work, but the staff of the Syracuse Laboratory is quick to point out that experimentation based on the scientific method is more reliable than subjective evaluation.

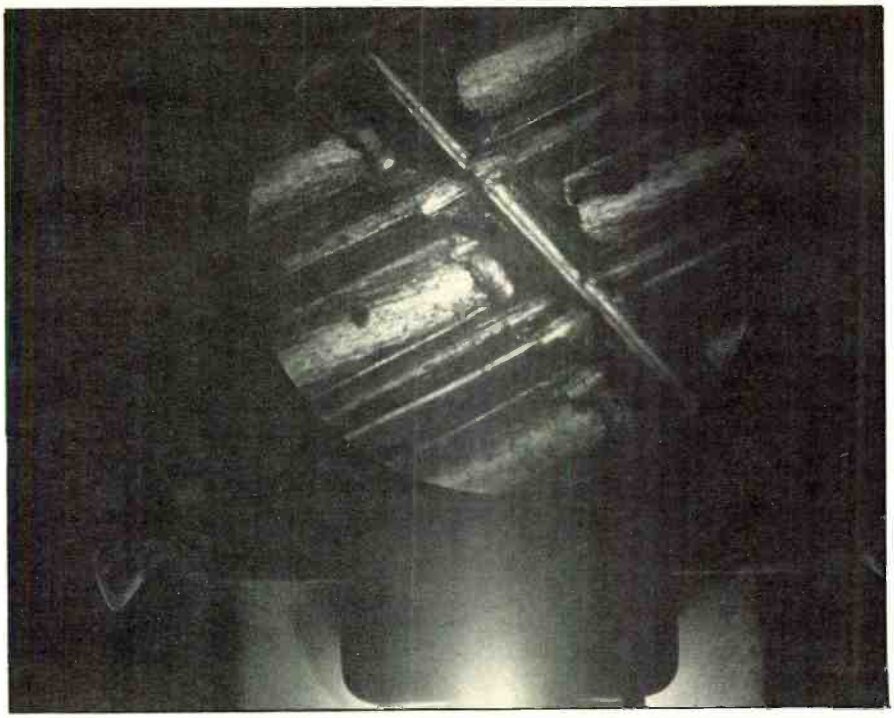
Restoration and Storage of Archival Recordings

An examination of some old recordings emphasizes the importance of the

Fungus, wear, and physical damage can cause serious preservation problems for older cylinder records.

re-recording work done at Syracuse and other laboratories. Many of the older recording media were extremely fragile, and prone to environmental damage. Just as time was running out for many of those old nitrate-based black and white film masterpieces — until, that is, archivists and film historians began to recognize their importance and transfer them to modern film stock — time also is running out for many irreplaceable recordings. And not only the recordings already under the care of audio archivists, but also those yet-undiscovered collections that may be deteriorating even faster because of where they have been stored over the years.

Early cylinders were coated with wax that was cut during the recording process; later they were molded from celluloid compounds. Time, and storage under conditions of high heat and/or humidity, have wrecked havoc with many of these recordings. Many historic radio broadcasts were laid down on "Transcriptions," which during the Thirties consisted of aluminum disks coated with soft acetate; later the base material was changed to glass. It is not unusual to encounter disks for which environmental conditions and age have caused the recorded media to dry up and spontaneously peel or strip off the disk, even without playing. Sometimes very difficult decisions have to be made as to whether or not to re-record a given disk with today's technique, simply to avoid further degradation. (But what if it is already recognized that this irreplaceable disk may be destroyed or significantly damaged by just this one playing?)



An optical comparator magnification of a scratch across the grooves on an aluminum disk. (Photo by Bill Storm.)

Other important archival recordings were made by embossing (not cutting) the surface of a soft aluminum blank. Such a method poses two major problems. Firstly, they cannot be played with any type of modern stylus, because a diamond, sapphire, or steel "needle"

will simply re-cut the soft material. Cactus thorns are the answer for re-recording purposes, but have you ever seen a "standard" thorn? (Maybe from a "standard" cactus?) If these aluminum disks have been played even once or twice with a hard stylus, the material on them may be lost forever. And secondly, the soft aluminum is very prone to oxidation, in which case the recorded material just turns to dust, and blows away.

For many archival re-recordings it must be accepted that the attempt to make the material time-stable and accessible to all may involve putting priceless pieces of recorded history at risk. The option, however, is to do nothing and let time take its toll. The staff of the Diane and Arthur B. Belfer Archive and Laboratory are constantly examining new ways of re-recording, and of late have been exploring methods involving contactless scanning of cylinder grooves by the use of lasers. Digital technology also is being explored as a method to "recover," or "reconstruct," the live sound of some of these historic artists and their performances. As the stage is set for the next massive hardware/software delivery system change, and adaptation of Compact Disc technology assures preservation of today's recorded history, it is interesting to note that both laser scanning and digital technology, keys to the future of sound recording, also may be the keys to unlock the past. ■■■

Described as the most comprehensive exhibit of magnetic recording technology in the world, the recently opened Ampex Museum of Magnetic Recording, located at its corporate headquarters in Redwood City, California, includes equipment that covers the technology spectrum from a rare wire recorder, to audio and video tape recorders from the present day.

The museum, developed over a two-year period under the direction of curator Peter Hammar, is said to represent an investment of over $1 million.

"Our concept from the beginning was to tell the complete industry story," Hammar says. "We have made the museum as comprehensive as possible, from the earliest wire recorders at the turn of the century, to today's state-of-the-art video recorders. Although Ampex is sponsoring this museum, we have included all of the pioneers in the industry."

Working as a consultant to Ampex, Hammar obtained assistance from 3M Company, BASF, AEG-Telefunken, Agfa-Gevaert, Studer, Sony, as well as CBS, ABC and NBC in gathering information and locating equipment for the museum. He also worked closely on the project with the late Harold W. Lindsay, who designed America's first professional audio recorder, the Ampex Model 200. (Sadly, Lindsay died in April 1982, shortly before the museum was opened for a preview showing.)

Extensive research in the US and Europe turned up rare pieces of equipment for the Ampex Museum, including:

- A 1911 Telegraphone Model C wire recorder, developed as an automatic telephone answering device and one of the oldest magnetic recorders on public display in the US.
- A 1936 AEG Magnetophon FT-2, the oldest tape recorder on exhibit in the US, and the prototype for the modern audio tape recorder.
- An Ampex Model 200 tape machine, first marketed in the US in 1948, and the machine that proved the practicality of magnetic recording in the US.
- An Ampex VRX-1000, America's first successful videotape recorder, which revolutionized television broadcasting when it was introduced in 1956.
- An RCA TR-22, america's first all-transistor professional VTR, introduced in 1961.
- A Sony EV-210 helical scan VTR dating from 1964, and one of the earliest recorders in the US market to utilize 1-inch videotape.
- An Ampex VR-3000, the first portable VTR to produce a broadcast-quality color picture when it was introduced in 1967.

Also on display is an array of magnetic recording media — wire, steel band and tape — produced from 1898 to 1965. The display will change periodically.

The museum is arranged in a series of 28 stations, each with its own video monitor. The traditional photos and text that accompany each piece of historic hardware eventually will be augmented by information on the video screen. Visitors will be able to call up on the screens computer-generated graphic information according to individual interests.

"Some visitors will want to read the story of how the Beatles became early users in England of multitrack recording, when they produced their *Sergeant Pepper* album on an Ampex AG-440 [four-track]," Hammar explains. "Others will be more interested in the specifications of recording equipment from that era. The museum goer makes the choice."

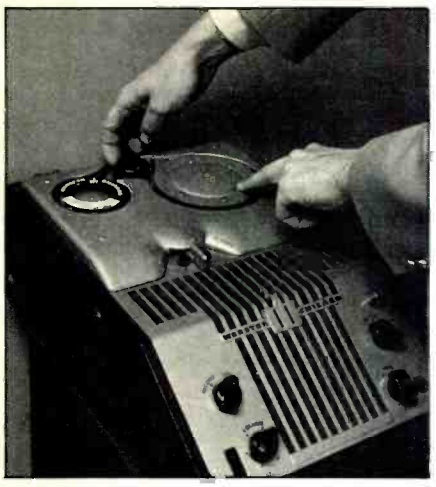
A series of short videotaped mini-documentaries will be added to the video programming later this year. "A push of a button will bring to the screen in color and stereo sound the stories behind the development of audio, data and video recording," Hammar says.

"I don't know of any other museum that offers such a coherent approach to the subject of magnetic recording technology," he considers. "Three of the most important technologies of this century are disk recording, magnetic recording, and radio. Both disk recording and radio are well covered in museums around the world. Magnetic recording has largely been ignored." The Ampex Museum helps correct this imbalance, he offers, with possibly the most comprehensive display of magnetic recording technology anywhere in the world.

A Dream into Reality

Hammar's interest in recording technology started during childhood, he recalls, when his father worked for the ABC (Television) in

Museum curator Peter Hammar with a mono Ampex Model 200 dating from 1948



Post-war attempts at recorder technology included wire recording, including this Electronic memory Module 80 produced by the Webster-Chicago Corp. in 1947.

New York City, as well as choral directors Fred Waring and Robert Shaw. At Earlham College in Indiana, the younger Hammar began work as an announcer/engineer at radio station WECI. After graduating, he took an announcer's job with WIAA-FM, the National Public Radio station in Interlochen, Michigan.

Hammar and his wife moved to West Germany in 1972, where they spent two years teaching English as a second language. It was while living in Germany that Hammer started collecting first-hand data on the development of magnetic recording technology.

"A lot of my high-school and corporate students were in broadcasting or related industries," he says. "In talking with people I discovered the story of the German Magnetophon [the audio tape recorder developed by Telefunken in 1935], learning a great deal about how the technology was developed in Germany before and during the war."

By 1980, with the assistance of Jay A. Clark, director of public relations for Ampex, Hammar had proposed a museum to tell the story of the European, American and Japanese development of magnetic recording. The project was approved by Ampex in May 1980.

Hammer started the search for machines by running advertisements in newspapers, trade publications, and technical journals in the US and Europe. "The response was just amazing; people both here and abroad were incredibly helpful," he recalls. "I expected some people to be disinterested, or to resent a manufacturer's involvement in the project, but nothing like that happened."

European companies such as BASF, Agfa-Gevaert, Studer and AEG-Telefunken provided assistance on the project — introducing him to their own corporate pioneers, opening their archives for research, and helping him track down equipment. Hammar also received valuable assistance from Heinz Thiele, a German pioneer in magnetic recording technology, whose support gave Hammar's work added credibility in the European technical community.

While some of the equipment now in the museum was purchased, the majority was obtained through trades, donations, and long-term loans. Locating some of it involved an element of luck, Hammar says, such as the discovery of the Telegraphone Model C telephone answering device developed in 1910, which showed up during the search of an Ampex warehouse. Company veterans don't know where it came from, but notes found inside indicate that the machine was acquired by Alexander M. Poniatoff, the founder of Ampex, and then set aside.

Ampex pioneer Harold W. Lindsay contributed equipment from his personal collection, and the Ampex Model 200 audio recorder on display came from John T. Mullin. It was Mullin who in 1946 first demonstrated a Magnetophon to an American audience in San Francisco; Lindsay was present for the demonstration and went away with the idea for the Model 200.

The VRX-1000 videotape recorder on display at the Museum is the first VTR to be made by Ampex, delivered to an American network, CBS in Hollywood. CBS returned the machine to Ampex in 1978 after 20 years of continuous service.

Few of the machines acquired by Hammar required extensive refurbishing, and all of them are in working order.

"A museum like this one has never really been done before in this industry," Hammar concludes. "We hope the museum shows that you can take a technical theme, draw on the educational aspects and the entertainment aspects, and make it interesting to all segments of the public." ■■■

THE EVOLUTION AND UTILIZATION OF 70MM SIX-TRACK FILM SOUND

Stereophonic Reproduction in Movie Theaters

by Larry Blake

With the sole exception of the various quad formats introduced in the early Seventies, traditional recording engineers always have had only two loudspeaker channels at their disposal. Stereophonic sound for motion pictures, on the other hand, has never used less than three tracks, a separate additional channel being necessary to avoid distracting placement for those seated away from the "sweet spot" in the middle of the theater. Similarly, as movie screens increased in width during the early Fifties, two additional speakers were found to be helpful in filling the gaps between the center speaker, and those located on the extreme left and right. The five channels behind the screen always have been supported by speakers placed through the auditorium, fed by a single "surround" track, giving a total of six available channels on a 70mm print.

This article reviews the history of the 70mm six-track system, outlining the unique potential offered by five speaker channels behind the screen, as opposed to the three utilized by standard stereo film soundtrack formats. Two films that made special use of the 70mm Dolby Stereo format, *Apocalypse Now* and *Altered States*, will be examined in detail, including the utilization of subwoofers and surround speakers.

Early research in stereophonic sound concluded that three loudspeaker channels were needed to provide an acceptable result for movie theater sound. In his famous paper, "Basic Principles of Stereophonic Sound," William B. Snow noted that such a three-channel system, while not giving listeners a copy of "reality," would give them an acceptable *illusion*. His research was conducted using various combinations of two or three microphones and loudspeaker channels. Snow stated that he used three channels in his tests for two reasons: firstly, that the fundamentals were easier to demonstrate; and secondly, that unless a total width of more than 50 feet was desired, three channels would suffice. The addition of more channels was largely a matter of economics, he noted, and offered that a screen width of more than 50 feet would probably require more than three loudspeaker channels.

The need for five channels was explored in the book, *Wide-Screen Cinema and Stereophonic sound*, written by the late Michael S. Wysotsky, who served as technical director of Mosfilm Studios in Russia. In it he charted the "percentage dependence of stereophonic quality, Qst, on the number of main channels used for sound transmission." Thus, a system with a Qst of 100% would be indistinguishable from the "natural" sound. The stereophonic quality of two channels was charted to be from 60 to 65%, three from 80 to 85%, and five, from 85 to 90%. Wysotsky went on to note that with five speakers behind the screen the point of diminishing returns is reached, and the addition of more channels would be "imperceptible."

For the first 50 years of motion pictures, movie screens rarely were more than 25 feet wide and, with the exception of Disney's *Fantasia*, only one speaker was used.

For the "purist" recording engineer, the concept of stereophonic recording and reproduction implies the use of coincident or spaced-pair microphone techniques, and two loudspeakers orientated in an equilateral triangle array. And, if the signal source is mono, then a realistic stereo soundfield can be created by panning the instrument, vocal, or effect between the pair of loudspeaker channels. Undoubtedly, the realism and illusion of depth and spatial separation created by "classical" Blumlein and coincident miking techniques can be attributed to the intricate phase relationships between the signals arriving at the listening position from the left and right aspects of the soundfield.

Where the phase relationship cannot be maintained upon playback, however, or the required equilateral-triangle relationship between the pair of stereo loudspeaker channels and the listener would severely restrict the number of people that could avail themselves of the "ideal" listening position, additional microphone and loudspeaker channels can help to enhance the stereophonic effect.

Such is the case with movie theaters, where the phase integrity of magnetic or optical soundtracks cannot be expected to maintain the necessary phase relationships between the left and right channels of a stereo pair, and for which the screen width —and hence loudspeaker separation — preclude more than a small proportion of the audience occupying an ideal listening position — *Editor.*

Cinerama

Probably the first experiments in stereophonic recording and reproduction using more than three channels were those for the Cinerama sound system. Initial work began in the late Forties, when the first test films for the three-camera system were made. Hazard Reeves, the entrepreneur behind Cinerama, and the founder of Reeves Sound Studios in New York, together with his chief engineer, Wentworth Fling, carried out tests to determine the number of channels necessary to support the visual "illusion" created by the 165-degree sweep of the huge Cinerama screen.

Early in their research the benefits of a five-channel system became obvious, and initial design of a six-track recorder was undertaken, employing two staggered three-track heads. This approach was abandoned because of obvious alignment and editing problems, and a seven-track head stack built. The seven-track recorder was used both in the recording of effects in the field, and in re-recording in the "studio," which actually was a converted tennis court on Oyster Bay, Long Island. Thus, if the mixing of a scene required playback from five reels, then six units would be wheeled in, the last one being used for the master recording.

The other two available channels were distributed in the auditorium, usually as left- and right-rear. On some scenes the "umbrella" configuration

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was used, with one track going to speakers on both the left and right walls, and the other to a speaker in the center of the rear wall. Such a set-up necessitated repatching during a performance (*sic*), and primarily was used for scenes requiring movement through the auditorium, an effect which has come to be known as a "fly-by."

"Stereophonic" was truly the operative word for Cinerama sound, as most scenes were recorded on location with seven microphones. The quality was further enhanced by the high speed of the 35mm magnetic film — 146.25 feet per minute, or a little over 29 IPS, compared to the 90 FPM/18 IPS of the standard 35mm system. In theaters this track was reproduced "double-system" on an interlocked reproducer.

Although the influence of Cinerama on Hollywood film studios has been well documented, it bears re-stating here. Reeves and Fred Waller, the inventor of Cinerama, had tried for years to interest major Hollywood studios in their system, but to no avail. Only after the success of their first film, *This Is Cinerama*, which eventually played on Broadway for two years — grossing $4.5 million at one theater with admission prices around $2 — did Hollywood begin to take notice. The morning after the premiere on September 30, 1952, for the first time ever a movie review appeared on the New York Times' first page, which may give some indication of the effect seven-track sound and 75-foot-wide screen made on the audience, which had seen only mono films (most of them black and white) since *The Jazz Singer* opened 25 years earlier.

Westrex re-recording console built for the six-track mix of "Oklahoma!" and equipped with 96 inputs divided into 16, six-track groups.

Todd-AO Format

Hollywood's — as opposed to Long Island's — initial attempts at stereo sound, first in interlock with a 35mm magnetic film reproducer, and later on composite prints in CinemaScope's four-track magnetic format with stripes on the sides of the sprocket holes, employed three channels behind the screen (Table 1, Line #1). It was not until late 1955, with the introduction of the Todd-AO process for the film *Oklahoma!*, that Hollywood took advantage of the five-channels-behind-the-screen format originated by Cinerama.

The sound system for Todd-AO was designed by Fred Hynes, and has come to be known as the Todd-AO discrete six-track format (Table 1, line #6). Approximately 60 films were released with 70mm discrete six-track prints, including blow-ups from the 35mm Technirama and anamorphic processes. True 70mm films were photographed with 65mm cameras, the extra space being used to accommodate wide magnetic stripes outside the perforations.

Dialog for Todd-AO films was recorded in mono, and placed to the actor during an elaborate panning premix. Mixers were careful to avoid "tennis matches" when dealing with close-ups on opposite ends of the screen. The degree and amount of "swinging" of

TABLE 1: CHANNEL ASSIGNMENTS FOR STEREOPHONIC MOTION PICTURE FORMATS

			Speaker Channels					
	Gauge	Format	1	2	3	4	5	Subwoofers
1	35mm	Four-Track Magnetic	Left	—	Center	—	Right	
2	35mm	Dolby Four-Track Stereo Optical	Left	Note #1	Center	Note #1	Right	Note #1
3	70mm	Dolby Six-Track "Split Surround"	Left	Note #2	Center	Note #2	Right	Note #3
4	70mm	Dolby Six-Track "Baby Boom"	Left	Note#4	Center	Note #4	Right	Note #3
5	70mm	Six-Track Spread (Note #5)	Left	Left/ Center Mix	Center	Right/ Center Mix	Right	
6	70mm	Discrete Six-Track (Note #5)	Left	Left-Center	Center	Right-Center	Right	

Note 1: The Dolby CN-160 optical bass-extension card extracts certain low-frequency information below 100 Hz from a stereo optical print, and sends this signal to a power amplifier/subwoofer combination. Satisfactory results can be obtained using existing speakers #2 and #4 in a 70mm movie theater.
Note 2: Bass-extension information below 200 Hz is recorded on tracks #2 and #4, as with standard Dolby 70mm releases. High-frequency information above 500 Hz for the left-rear and right-rear channels is recorded on tracks #2 and #4, respectively. Information below 500 Hz for both rear channels is taken from track #6. The high-frequency material above 500 Hz on tracks #2 and #4 is combined and recorded on track #6, allowing compatible playback in any Dolby 70mm installation. All six tracks of a "split-surround" print are Dolby encoded.
Note 3: Dolby's CP-200 Cinema Processor has a separate subwoofer output for information on tracks #2 and #4 below 100 Hz.
Note 4: These loudspeaker channels receive bass-extension information below 200 Hz.
Note 5: Though they are rarely used these days, both discrete six-track and six-track spreads can employ Dolby-encoded 70mm prints.

RE-RECORDING STAGE A AT WARNER HOLLYWOOD STUDIOS

In the past six years, the re-recording stage at the Samuel Goldwyn Sound facility of Warner Hollywood Studios has been the venue for some of the best-known Dolby Stereo film soundtrack mixes, including *Star Wars, The Last Waltz, The Empire Strikes Back, The Black Stallion, Raiders of the Lost Ark,* and over a dozen others. In addition, Warner Hollywood is one of the few facilities in the world that both records 70mm prints, and "shoots" stereo optical negatives.

In 1980 Don Rogers, technical director of Warner Hollywood, and John Bonner, the chief engineer, selected Harrison Systems to build a new board for Stage D, plus a sister board for a new re-recording studio, Stage A, to be built in 1982. Harrison had installed its first two PP-1 film post-production consoles in 1979, at Walt Disney Productions in Burbank, and Fantasy Films in Berkeley, California.

This was Bonner's first opportunity to supervise the building of a dubbing stage, and he was able to incorporate almost 30 years of ideas and experience. Acoustical design was by Jeff Cooper, who in recent years has designed three dubbing rooms in the San Francisco Bay Area — the old Zoetrope studio in San Francisco, Fantasy Films in Berkeley, and at Sprocket Systems (Lucasfilm) in San Rafael.

"First, I felt that the acoustics were absolutely and completely vital," Bonner says. "We wanted to hear *everything* on the track, and not have it obscured by reverberation or indistinct echoes. I think Jeff Cooper did an excellent job once we discussed what our objectives were. Earlier we had talked with another consultant, who was going to make the stage sound extremely dead; it would have been a disaster. We have some dead rooms that we use for looping, and nothing *ever* sounds right in there. And if you made it sound right in there, it would sound terrible in the real world."

Cooper was able to include in the acoustical design of Stage A concepts that he had implemented in a similar manner at his Zoetrope, Fantasy, and Lucasfilm installations. "Our fundamental philosophy," he says, "is to try to incorporate all the electrical and mechanical engineering, air conditioning, lighting, sound-proofing and reverberation control into the architecture, to form one cohesive unit.

"The reverberation versus frequency curve is designed to be smooth between 250 Hz and 4,000 Hz, plus or minus 1/10th of a second. The curve is designed to gently slope upwards below 250 Hz at a maximum rate of 150% per octave, to provide warmth and sonority in the lower regions of the spectrum. We achieved this by designing a diffuser that did three things at once: it absorbed selected bass; it absorbed mids; and it reflected highs.

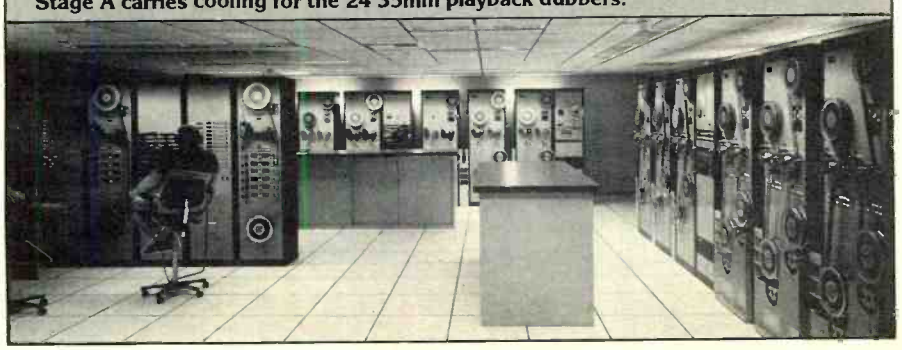
"The primary acoustic design goal was a reverberation curve that was rich enough to translate well into full theater sound, yet 'tight' [i.e., short] enough to achieve excellent articulation for mixing dialog, and for critical positioning of sound sources in the stereo-surround panoramic field."

Although there is no codified standard for the distance between the re-recording console and the screen, Bonner felt that, up to a point, the closer the better. At Warner, the center of console subtends an angle of approximately 54 degrees. "The old dubbing room at Fox was so long and narrow that you were less critical of crazy mistakes that you didn't find out about until you heard them in a theater," he says. It would sound great from the console, but from the first three rows you'd just be horrified. In a normal room the console would sit back closer to the rear wall, and I felt that it would be better to get it out in close where the audience is."

Harrison PP-1 Console

The console installed in Stage A, which opened last September, basically is a sister to the one installed on Stage D: 60 input modules, each with A/B input

In addition to carrying cabling, the machine-room floor at Warner Hollywood dubbing Stage A carries cooling for the 24 35mm playback dubbers.



dialog has decreased, and is now done very little, if at all. Sound effects also were primarily in mono, and many weeks would be devoted to panning them across the five channels for such films as *Spartacus,* and *The Alamo.*

Music was an area in which having five channels really helped, adding, in the opinion of mixers who have worked in the format, at least 50% in clarity. Which is one reason why musicals such as *My Fair Lady* and *The Sound of Music* sound so glorious in 70mm prints — assuming such prints can be found.

While the majority of those who have worked with the "discrete six" format swear by it, many studios did not have six-track recorders or boards, and would record a four-track master. Later, six-track "spreads" for their 70mm films would be made by combining the center channel with the left and right tracks to create information for the left-center (speaker #2) and right-center (speaker #4) channels. The resultant tracks are recorded 6 dB down in relation to tracks 1, 3, and 5. (Table 1, line #5.) A monitoring matrix would simulate the effects of this spread during the mix.

This four-to-six transfer process was considered a time- and money-saving expedience, and grew with the practice of photographing in 35mm and blowing up the negative to 70mm release prints, thus avoiding having to deal with large and bulky 65mm cameras on location, not to mention saving about $200,000 in film-processing and printing costs. Thus, by 1971, production of true 70mm discrete six-track films became a thing of the past.

Dolby 70mm

Dolby Laboratories first became involved with 70mm prints in 1976, with the release of *Logan's Run* in a few Dolby-encoded discrete six-track prints. *Logan's Run* also was the first 70mm film to make use of equalized monitors in the dubbing stage, which Dolby had been using for their 35mm optical releases since 1972, and their 35mm four-track magnetic prints since 1974. Equalized, wide-range monitoring is not considered feasible unless noise reduction is used on the print, be it magnetic or optical, because of the high-frequency hiss that would be revealed.

It was during the mixing of *A Star is Born* that Dolby engineers first came in contact with the six-track spread. The film had been mixed four-track and Dolby film consultant Steve Katz felt that by lowering full-range information going to channels #2 and #4 to the point where they could hardly be heard, the resultant mix sounded better. Ioan Allen, Dolby marketing VP, explains this phenomenon by noting that the full-range spread "gives you what appears to be a disembodied wall of quazi-mono sound, because there are random phase conditions occurring."

A few months later the Dolby engineers were at Samuel Goldwyn Studios in Hollywood for final mixing on *Star*

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Wars, and Katz came up with the idea of restricting the spread to speakers #2 and #4 to just low-frequency information below 200 Hz. The spread was used throughout the whole film, as was previously the case with standard full-frequency spreads. On later releases the low-frequency fill was added only as needed, to remove the chance that dialog, unwanted music or sound effects would be spread.

The unparalleled success of *Star Wars* and, six months later, of *Close Encounters of the Third Kind*, established in the public's collective ear the bass extension capabilities of the Dolby 70mm format, not to mention making "Baby Boom" the *de facto* standard for Dolby 70mm releases (Table 1, line #4).

The reasons for using channels #2 and #4 for bass extension go far beyond the avoidance of phase problems claimed to be created by spreading full-range information. Among them:

- The Altec-Lansing A-4 "Voice of the Theater" speakers used in most 70mm movie houses are notably deficient in bass response below 100 Hz — they were not designed for it — and can sound even worse when the speaker wings have been removed, or if there is no room for them in the installation. By doubling up on the number of speakers, there is an effective lowering of the bass cut-off point. The bass-extension capabilities of the Dolby 70mm format allows added punch and warmth on sound effects and music.
- Since, in theaters, 200 Hz lowpass filters are used for channels #2 and #4, no noise reduction or equalization is required, thereby saving theaters about $3,000 in equipment costs. The Dolby CP-100 Cinema Processor in use in 1977 could house only three channels of noise reduction, and required a certain amount of jerry-rigging to convert it to discrete six-track playback. (The CP-200 unit, introduced in 1980, can be modified quickly to play any existing or proposed motion-picture sound format.)
- Another practical consideration is that, as with previous six-track spreads, a master mix can be monitored and recorded four-track, thus enabling the 70mm six-track and the 35mm two-track (Lt-Rt) stereo optical printing masters to be made easily.

Practical and economic considerations aside, Dolby's Ioan Allen feels that few movie screens today are wide enough to warrant a discrete six-track mix. (As a general rule of thumb, it is often considered that three speakers will produce acceptable stereo reproduction if located up to 20 feet apart. Since the majority of 70mm screens are approximately 40 feet wide, each speaker is no more than 10 feet from the other.)

Surround Channels

Film sound first emerged from behind the screen in the roadshow engagement of *Fantasia* in 1940, when special notchings on release prints could cause the sound to be moved into loudspeakers placed throughout the auditorium. It was not until the Cinerama format appeared on the scene that discrete information was recorded especially for the surround speakers.

To take into account the precedence effect, and thereby ensure phase coherence of signals that are both in surround speakers *and* behind the screen, surround information for four- and six-track discrete mixes always has been printed 1½ frames (60 milliseconds) from the sound to be fed to the screen speakers. This time difference usually was accomplished by making a separate pass on a reel, advancing the master recorder six sprocket holes, and then recording on the surround track a mix of front-channel information, such as music and background effects.

The dialog track is almost never placed in the surround channel, unless it is, for example, an echo send to help simulate a PA sound, although there have been a few notable exceptions to this practice. The most famous was the placing of "God's Voice" in the auditorium for *The Ten Commandments*, while in Stanley Kubrick's *2001: A Space Oydssey* the all-seeing, omnipresent HAL 9000 computer makes a similar entrance. It should be noted that both voices were also in the front speakers, just in case.

Creative use of surrounds was made for *Woodstock*, where Chip Monck's voice on the stage PA system ("The brown acid is not specifically too good"; "Wheat Germ, Holly has your bag.") and the crowd track were assigned to the surrounds, thereby neatly helping place the theater audience right in the middle of Max Yasgur's farm.

Apocalypse Now: A Milestone in Film Sound

The acceptance of the Dolby Stereo format has, in the era of digital reverberation and delay lines, brought new meaning to the use of surround speakers. The most extensive use of surround channels was for *Apocalypse Now*, the first film to be released in the Dolby 70mm "split-surround" format. The split format utilizes the free high-

WARNER HOLLYWOOD STAGE A

. . . *continued* —

switching for a total of 120 inputs. The board is divided into three sections: one each for the dialog, music, and effects mixers. A total of 24 outputs are available — eight for each section — which can be combined in any conceivable configuration. Both boards are automation-ready, and Bonner will be keeping a close eye on the initial Harrison cinema automation installation, which is scheduled for installation this spring at Disney. "Harrison's proposed hard disk package will be able to handle two rooms, so we will have Stage A and Stage D interconnected," Bonner says.

In addition to fader-level, muting, and VCA subgrouping, the Harrison automation package will control many of the tedious switching functions that have to be done in a darkened room during the course of a film mix: in/out selection of the individual equalizer bands, and highpass/lowpass filters; patch insert keys; and A/B input selection. In addition, there are three Harrison Auto Graph automated graphic equalizers per section, with three-channel models in the music section, two-channel models in the dialog section (for occasions when one track is split off into two faders), and standard single-channel graphics in the effects side. Also mounted flush into the console are the standard paraphernalia of film-sound mixing, including a Dolby DS-4 monitoring unit, and control units for a pair of Lexicon 224 digital reverbs.

"We wanted to take advantage of the flexibility of the Harrison consoles — we had that going in — so that it would be convenient to work without driving ourselves crazy with patchcords," Bonner offers. "We did this by bringing the machine outputs into two different points. On this board the multitrack [i.e., 3-, 4-, or 6-track 35mm] machines come down basically to the A-side of inputs — this is the final dubbing mode. The monaural tracks come into the B-side — this is the pre-mix mode. Now that we see this is a good way to go, we plan to go back and do it to the board in Satge D, when we can get our hands on it. We had planned to do it when we installed that board, but couldn't because of a lack of trunks and jack row space.

"There are something like 2,500 jack points in this building — I counted them because I labeled every one of them, and was getting tired of it! When we are operating, the only patches that you would normally see are to take the machine outputs into the Dolby noise reduction units, and back again. The rest of the hardware works so smoothly that very few patches have to be used. With 2,500 patch points, we might have needed 1,000 patch cords!"

The machine room contains 24 Magna-Tech 35mm playback dubbers and two 35mm recorders, one of which can record up to six tracks, and the other up to four. Each dubber is brought out to two patch points, with some going to three, enabling simple patching to any console section. In addition, the machine room is pre-wired to accept two 16-track multitrack machines. Bonner feels that the 16-track format will be ideal for recording separate four-track mixes for dialog, music and effects, thus facilitating the necessary separation of effects and music elements for foreign stereo and mono release, as well as increasing the flexibility of the initial master mix.

— *continued overleaf* . . .

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frequency information space about 500 Hz on tracks #2 and #4 of Dolby 70mm prints, in conjunction with information below 500 Hz on the standard surround information on track #6. This provides, in essense, a quadraphonic effect from a 70mm print, while retaining compatible playback with standard Dolby 70mm equipment (Table 1, line #3). It has since been used on *The Jazz Singer*, and *Pink Floyd The Wall*.

The circumstances surrounding the preparation and mixing of the soundtrack for *Apocalypse Now* are considered by many to be unique. While complete coverage of the recording and post-production stages easily could occupy an article by itself, suffice it to say that all of the variables that determine the quality sound mixers can bring to the soundtrack — time, money, and equipment — for once added up in favor of the filmmakers. By the time that the eight-month mix began in December 1978, the film's sound crew already had been at work for a year and a half recording and editing the soundtrack. The long dubbing schedule enabled the mixers to take full advantage of the MCI JH-500 automated mixing console at Zoetrope Studios in San Francisco.

Not to be overlooked is the role that the film's story itself played: director Francis Coppola did not want the audience to watch a film *about* Vietnam, he want them to *experience* it. To this end the soundtrack would play a large role in the overall effect of the film. Which was part of the reason why the film was mixed in the Dolby Stereo split-surround format, although the initial idea for having the stereo surround information came from Coppola's hearing a Tomita synthesizer album in quad over a rough cut of the helicopter attack.

Richard Beggs, music mixer for *Apocalpyse Now*, was responsible for mixing the almost totally electronic score from 24-track tapes to the six-track format used for the film: left, center, and right front; left rear, right rear, and boom.

"The information for the surrounds was almost never discretely assigned to the rear," Beggs says. "It was pretty much in the front, with a reverbered surround.

"The luxury of *Apocalypse* was that we mixed the music against the effects and dialog. In other words, we put the effects and dialog pre-mixes up, monitored them, and mixed the 24 [multitrack music recordings] directly to six against those other elements, rather than just showing up at the mix with a six-track or a three-track, which is what normally happens.

"We were able to pull down the solo or musical figure that fought with the dialog, and keep, for instance, the bass line going strongly to really support the scene dramatically. You could really manipulate and weave the orchestra in and out of the action and the dialog, because we had access to all the elements of the music up to the very last minute."

Elektra Records sent Beggs a four-track dupe of Bruce Botnick's recording of "The End" by The Doors, which was featured in the opening montage of the movie. "There were several points where I put a lot of echo on Morrison's voice [in the surrounds] so he receded in space and into the back of the theater at the same time. It gave a disembodied sound."

Dale Strumpell, re-recording mixer on *Apocalypse Now* with Walter Murch, Richard Beggs and Mark Berger, notes one of the more subtle uses of the split-surround format, which occurs after the film's opening montage: "Martin Sheen is staring up at the fan. You're in his mind, and the sound [of the fan] is coming from both the back and the front of the [theater]. As he gets off the bed and

WARNER HOLLYWOOD STAGE A

. . . *continued* —

Bonner made tests "comparing a 24-track to our 35mm six-track, which is our least effective recording medium. Our six-track had more 'punch.' But then we got our hands on an Otari [MTR-90] 16-track, and felt that it was as good as our three-track recorder, which is our best medium."

Monitoring System

Five JBL 4675 theater loudspeaker systems are mounted in a wall behind the 38-foot projection screen. The 4675 speakers originally were implemented by Mark Engebretson and John Eargle to complement Advanced Technology Design bi-amplifiers. Each 4508 bass reflex cabinet contains two 2225J 15-inch woofers, and a 2441 compression driver mounted on a 2360 Bi-Radial horn. The system is bi-amplified, and uses the TH-X active crossover designed by Tomlinson Holman of Lucasfilm for its new re-recording stage.

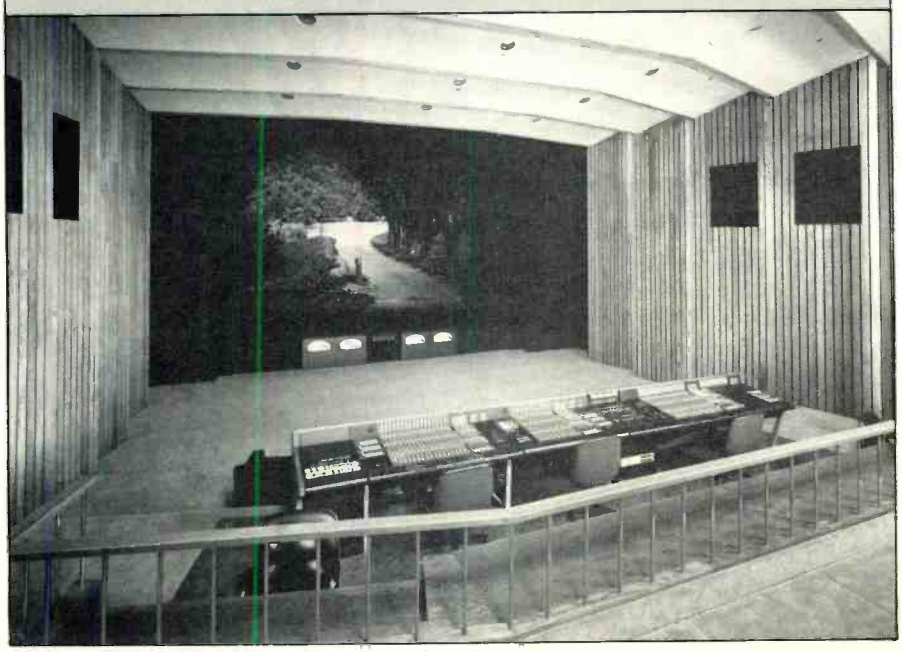
The major benefit of the new monitoring system, compared to the A-4 Altec-Lansing Voice of the Theater, the JBL 4676, and other horn-loaded reflex speakers, is that the response is smoother, and therefore requires little equalization to meet the wide-range monitoring curve used for Dolby Stereo mixes. This "X" curve, which is codified in ISO Bulletin 2969, is flat to 2 kHz, rolling off 3 dB per octave at that point.

"The biggest problem with the Voice of the Theater speaker, to my mind," says Bonner, "is the region where it goes from a horn-loaded speaker, which is quite efficient, into the bass-reflex mode of radiating the low frequencies. The change in efficiency between the two types gives us a bump in there that can create a 10 dB difference in one octave, which can be very tough to get rid of.

"In the long term, you appreciate the definition and lack of distortion of the JBL system. I feel very strongly that the mixers should hear everything possible. Otherwise, trying to mix in a room where some of the sound is obscured to you is like trying to paint without having enough light."

The installation at Stage A at Warner Hollywood calls for the speakers to be mounted in a plywood wall, creating an extended speaker baffle. When Bonner first listened to the JBL 4675 on Stage C at Warner Hollywood, he recalls, "it was sitting on the floor, and we saw the deficiencies. We found that we achieved smoother bass response by moving it back so that it was parallel to the baffles on the A-4." ■■■

Harrison PP-1 console forms the nerve-center of Stage A; acoustic design © Jeff Cooper.



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goes to the window, the sound fades from the back speakers to the front — left, center, and right — and then, as the hand comes in and moves the venetian blinds, it moves from the left and right speakers to only the center channel. So you have this whole thing where the sound is narrowing . . . narrowing . . .narrowing, as the dolly shot is moving in closer."

Coppola wanted to make sure that the care taken during the mix would be heard in theaters, and to this end Tom Scott was hired to scout and supervise the installations for the 70mm roadshow engagements. Scott comments on how he used test films in checking theaters: "Walter Murch [*Apocalypse Now* sound designer] took certain scenes that had things you had to listen for, to see if things were balanced properly. And since you can never tell anything [by hearing it] once, we repeated a one-minute segment four or five times. You had to hear it at least three times before you could say 'Oh, yes, that's breaking up,' or 'That's not quite loud enough.' One was a big music cue to check the stereo balance; another was just before the tiger scene in the jungle, when you can really hear the stereo surrounds very clearly; and another was when the Playboy Bunnies had landed and the helicopter comes over your head very, very loud in the stereo surrounds. We didn't have picture with it. After we were pretty sure that things were balanced, having played this segment, we played reel #6 from beginning to end."

Scott also had a channel placement film to make sure that everything was going where it should. "This is the left track . . ." Such a test reel was especially critical, because tracks #2 and #4 contained low-frequency "boom" information below 200 Hz and, above 500 Hz, high-frequency information for the left and right surrounds.

Altered States: A Sound Challenge

One of Steve Katz' first jobs as a sound consultant, after he left Dolby Labs, was to create special sound effects for *Altered States*. Certain sounds were panned during the final mix at The Burbank Studios, Katz recalls, with his personal computer. "Utilizing my Commodore PET, I built a computer-controlled panpot which was basically four VCAs. The reason I used the computer is that I could program whatever I wanted. In addition, the comptuer was totally repeatable and totally linear, meaning that it never made a mistake. If you set it for one panning speed it would do it forever. Also, I could gradually increase the speed, and also make *it* [the rate of increase] totally repeatable.

"I had a 'zig-zag' program that [panned the sound] from the left to the surround, and to the right and to the surrounds, etc. This program was used at the beginning of the 'down the throat' sequency at the end of the film to pan voices by [musique concrete composer] Pierre Henry.

Zoetrope Studios' re-recording stage built for the mix of "Apocalypes Now". Acoustic design © by Jeff Cooper.

"I could control all parameters, and could get a peak or dip in the middle [channel]. On the program I used for *Altered States* I went to a triangulation, so that the sound went from the left to the right to the surround, skipping the center. If it went through the center, when the sound went through the matrix on the 35mm stereo optical prints, I would have ended up with *double* the center track."

In the sequency toward the end of *Altered States*, where Blair Brown goes into the whirlpool, Katz panned the three horn tracks around the theater, with one clockwise rotation starting at the slowest speed, and then speeding up. Another went counter-clockwise, starting faster than the first, and still speeding up. "You had the effect of the thing circling you, which is dramatically what the sequence called for," says Katz.

Bass Extension

Earthquakes have provided an aural inspiration for film sound mixers since the industry's beginning. Probably the first film to use what might be referred to as "mixing" was *In Old San Fran-*

cisco, in which intrepid Warner Bros. engineers rolled an oil drum on concrete to create the sound of the earthquake. (This was a few months before Al Jolson spoke in *The Jazz Singer*, it should be noted.)

Ten years later MGM was looking for a way to enhance the presentation of its version of the 1906 San Francisco earthquake for the film, *San Francisco*. In certain theaters extra speakers and amplifiers were installed, and manually cued by the projectionist at the appropriate moments.

We now jump to 1974, and Universal Studios' *Earthquake*, for which the company introduced Sensurround, which included the installation of Cerwin Vega subwoofers in theaters. Later versions of the Sensurround system changed over the years, and was used on four other films released by Universal.

Nowadays, the majority of Dolby Stereo mixes, both for 70mm and 35mm stereo optical-only release, are recorded on a single four-track master. Later on, in what is basically a controlled transfer, the left and center channel information and the center and right channel information below 200 Hz placed into tracks #2 and #4 when desired.

Bill Varney, re-recording mixer at Warner Hollywood studios, has encountered a lot of low-frequency information over the years, since his first contact with the Dolby 70mm "baby boom" format in 1978 for *Grease*. Since then he has mixed *Star Trek: The Motion Picture, The Empire Strikes Back, Raiders of the Lost Ark, The Jazz Singer, Poltergeist, The Thing* and *The Entity* for 70mm Dolby release.

Re-recording of "Oklahoma!" at MGM Studios, the first use of the Todd-AO format, and due for re-release this year in new 70mm, six-track prints.

Here he talks about the benefits of recording the boom channels during the final mix. "If you know that you are going to have a 70mm six-track release," Varney says, "then I would prefer to do a six-track mix to begin with, and then take that element to create the four-track stereo optical [mix]. It's just a better way to do it; I think that's one of the reasons both *Empire* and *Raiders* worked so well. We knew in advance that our major release was going to be 70mm six-track, and we shot for that.

"When we make a six-track master, and don't spread a four-track, we almost never extract information [for the boom channels] from the sides or the center. What we do is take one or two elements from which, say, an explosion is made, and use it as our motivating low-frequency material. The reason for that is that you then have a clean, solid low-end. If you do extract information from the left, center, or right channels, then you might not be dealing with just the explosion, but also with the fire that has already started, the fire trucks on the way, etc. These other elements end up in the boom channel, and it becomes a mish-mash of sound.

"I can remember only a couple of cases where we have had any use for boom channels in terms of music. We've got to be careful of it because what piles up quicker than anything else in a theater is low-frequency information. If you over-use boom channels, then the product itself actually degrades because the intelligibility of dialog, or whatever other elements are playing at the time, can actually be hurt by it."

Apocalypse Now made great use of the low-frequency extension capabilities of the Dolby 70mm format, and indeed is one of the few Dolby films to have been mixed with subwoofers. The same Meyer Sound Laboratories 650-E subwoofers used at the Zoetrope mixing studio in San Francisco were installed by Tom Scott in six of the 16 initial 70mm roadshow engagements for

Apocalypse.

To help make sure that the information being fed to tracks #2 and #4 would play acceptably well in standard installations that did not have subwoofers, a switch was installed at the mixing console to turn them off. "There were times when there was just enough with them, and not quite enough without them," re-recording engineer Dale Strumpell explains. "The difference was switching them on and off. Which meant that the Meyer subwoofers were taking you just on the other side of some threshold of perception. You wanted the mix to be just on the other side of that threshold, even without the Meyers for the theaters that didn't have them. So we could turn them off and say, 'Okay, we had better bring the boom channel up 2 dB, and we'll be over the threshold even without any Meyers on.' And then with the Meyers back on again, it would still sound fine, and you knew that you had it covered when you didn't have them.

"This is similar to what happens when you compare a mix in mono and stereo. Sometimes you are operating on the low-end of the scale when it is *just* enough [in stereo]. And you say, 'Quick, go to mono and make sure that we can still hear that.' In changing from stereo to mono you are most concerned about dialog intelligibility. In both instances you have an enhanced version, and a regular version."

In conjunction with the 70mm release of *Altered States*, Warner Bros. offered to theaters a bass-enhancement package under the tradename of Megasound, each of which contained Cerwin-Vega L-36PE speakers. For the premiere engagement in Los Angeles during December 1980 at the Village Theater, Westwood, Steve Katz was given an unprecedented *carte blanche* to install the finest sounding system he could. The three-month run of *States* at the Village with Katz' system has become something of a legend among LA film-sound afficionados.

"The installation was similar to the one I did in the Paramount Theater in Hollywood for *Close Encounters of the Third Kind*," Katz remembers. "First we took the screen down and bolted the speakers together with two-by-fours. We put ¾-inch plywood in the front between all the cabinets, creating a solid wall, which picks you up a *ton* of bottom.

"We installed eight Cerwin-Vega Baby Earthquakes as subwoofers, and 21 Bose 901 systems as surrounds. I think the total amplification to drive the entire system was about 3.5 kilowatts.

"What we ended up with, at the loudest point in the show, was 110 dB at 50 cycles, and at 30 cycles we were down 25 dB, which was exactly what I wanted. If we had gone any lower, the sound would have been unpleasant."

Discrete Six-Track Mixes Today

Since 1971 only a few films released in 70mm have employed the original Todd-AO discrete six-track format, including *Logan's Run, Annie, Tron, Heaven's Gate, 1941*, and The Rolling Stone's *Let's Spend the Night Together*, with all but the last two employing Dolby-encoded prints.

One of the most adventurous of the recent discrete six-track mixes was for Walt Disney Productions' *Tron*, which was mixed at Lion's Gate Films in West Los Angeles by Mike Minkler, who had mixed many 70mm baby-boom films previously, including *Altered States*, and *Wolfen*.

"*Tron* was the first discrete film I had ever mixed," Minkler offers, "and people who had worked with six-track discrete had always told me what a difference it makes. And until you mix on it, and see the imagery that you can get, you will never appreciate its potential. You can really pin-point *exactly* where things are, and it almost becomes three-dimensional. Instead of things jumping from center to left, for example, you can really get precise.

"The pin-pointing at Lion's Gate dubbing stage is not as noticeable as it is in a big theater. You can get it here, but everything is a lot louder and tighter. You lose that in a big room, but you pick up more imagery. You can play games and be subtle with movements. Yet it's very noticeable, although it is not jumping huge amounts of space."

Only six theaters played *Tron* in Dolby discrete-six prints, primarily because of the added cost in installing extra noise reduction and equalization cards. As might be expected, a first-run house like Mann's (formerly Grauman's) Chinese Theater in Hollywood received a discrete print. On opening weekend, three speaker channels were blown, including the center, and the dialog was patched into track #4.

Regarding problems that mixers run into even at a showcase engagement in Los Angeles, Minkler considers, "What's absurd is that [The Chinese Theater] is the world's most famous theater — next to the Radio City Music Hall [New York] — and here you have just a lousy sound system. Their equipment is not up to snuff, and needs to be rebuilt."

Although Bill Varney has never used the discrete six-track format, he has probably mixed more 70mm Dolby films than anyone else in the United States. He believes that the discrete format offers enormous potential.

"Mixers are in a strange position now," Varney says, "because we really don't know which of the 70mm stereo formats should become the standard. My gut feeling is that the traditional Dolby 'boom' stereo is not the way to go. If you get into a big theater, the hole between speakers #1 and #3, and between #3 and #5 is just too big. You can sense it on pans on vehicles and things like that, if you listen very carefully.

"It's nice to have those extra low-

frequency channels available to you for low-frequency effects, but my guess is that soon we will see more use of the discrete six-track system. The Dolby system will allow it anyway today. And there's no reason why we then couldn't *still* use speakers #2 and #4 for low-frequency information.

"It will let us spread the dialog a little wider on the screen. I still don't think that we should get into a lot of moving. Now I only go out to speakers #1 and #5 for an off-stage line, or something that I want to shock with. But movement of normal dialog drives audiences *insane*. What I would like to do is diverge out slightly to #2 and #4.

"On the last two or three 70mm spreads that I've done, we have found that the image we create as we make the four-track stereo[mix] while monitoring through the matrix — which ultimately becomes the two-track for stereo optical release — is much too wide on discrete channels. We can't just literally transfer the four-track to a six-track, because the width is just *too* great.

"What happens is those sounds that sit where speakers #2 and #4 would be wind up in the six-track as speakers #1 and #5, because the matrix is not pulling them in. Now, when I make the six-track spread I tend to take the sides and bring them in. It would be easier if I could spread that movement to #2 and #4 during the mix, but obviously it's impossible to satisfy the requirements of both formats at the same time."

* * *

A 70mm six-track print offers the best sound experience available this side of a Pink Floyd concert. However, there is one important *caveat emptor*: too often, even in Los Angeles and New York, the presentation, both picture and sound, is abysmal. Anyone who really cares about Good Sound should take it upon himself or herself to *complain* when something isn't up to snuff. But don't try to explain to the assistant manager that dialog shouldn't normally come from the surrounds. Just demand your hard-earned money back in the interests of your wallet, and write the district manager of the theater chain about the problem. Clearly, there are numerous creative tools available to re-recording engineers that enable the quality of film sound tracks to be vastly improved. They deserve your support. ■■■

THE POTENTIAL OF DISCRETE SIX-TRACK FILM SOUND

A Personal Assessment by Larry Blake

It has been said that films are the end result of thousands of choices — and that the more options available, the more opportunity exists for a filmmaker to be creative. Fast film and lenses, small silent cameras, punch-in recording, and automated mixing all have shortened the list of what filmmakers previously were unable to do.

Dolby Laboratories' chief contribution to the art of film sound — and in this writer's opinion it has been a substantial one — was to make stereo release a practical matter by making it available on a standard optical track, and hence with no cost premium per print. Undoubtedly, producers of a large percentage of the over 300 films released in Dolby Stereo would not have even considered a stereo mix if the only stereo format had been the 35mm four-track magnetic system, whose prints cost twice as much as those with a standard optical track. Thus the creative experience afforded by a stereo mix has been available to more filmmakers and filmgoers than ever before.

Big-budget films that would have received a stereo release anyway also have benefitted, because their stereo mix is now heard in hundreds of theaters, and not just in the Top 30 markets. As of March 1983, over 3,000 theaters in the United State and 1,000 outside have installed Dolby Cinema Processors. In other words, the Dolby Stereo optical system made the extra cost and trouble of a stereo mix very easy to justify, and it is bewildering to examine the lengths that some anti-Dolby people in Hollywood go to ignore these facts.

Much anti-Dolby sentiment can be traced to what often are considered compromises that Dolby Stereo imposes on re-recording mixers: the 35mm stereo optical format, because it uses a matrix instead of recording separate information for all four channels; and the 70mm "baby boom" format, because it uses two of the tracks only for low-frequency information. This writer's opinion on the first question was hinted at earlier: that the idea of getting stereo to as many theaters as possible in mono-compatible optical prints is a prudent one. Furthermore, the two-track stereo optical format (compared to a four-track optical format) appears to have many practical advantages in the real world, regardless of the implied technical constraints.

The Dolby 70mm format has produced some stunning mixes that have used the bass extension capability to good advantage. Dolby's reasons for using the "baby boom" format are well known, and the idea of extracting low-frequency information and placing it into extra speakers *works*. (In fact, one of Dolby's least-known products, the CN-160 optical bass-extension card, does just this for 35mm stereo optical prints, feeding selected information below 100 Hz to speakers #2 and #4 or, preferably, to a separate amplifier/subwoofer combination. The results can fool professional ears in supporting Dolby's claim that the card does "much to bridge the gap between 35mm optical and 70mm magnetic release prints.")

Nevertheless, as a result of the Dolby 70mm format's *de facto* standardization, the concept that all five channels can be used for *discrete* information has fallen by the wayside. It can safely be stated that, with the notable exception of *Tron* and *Let's Spend the Night Together*, no film mixed in the discrete format has applied the tools and techniques learned in the late Seventies. Almost all 70mm mixes instead have been used to refine the "baby boom" format, while no one has explored the even greater potential of the discrete system.

What If . . ?

Discrete six-track would prove particularly useful as a tactile expressive tool, aiding both in power and in delicacy, providing mixers with *two* stereo pairs — one wide, the other tight — to match to the picture. Mixing in discrete six-track would be the equivalent of adding both wide-angle and telephoto sound "lenses."

At times during the truck chase in *Raiders of the Lost Ark*, for example, the mixers were able to lower the music in the center channel, and hence enable the music to keep sawing away in the left and right speakers while certain key truck effects had the center to themselves.

The discrete-six format would be very conducive to this kind of creative positioning, enabling mixers to use, say, speakers #1 and #5 for music, #2 and #4 for effects, and the center channel #3 for dialog. Or, maybe channels #2 and #4 with a 50/50 mix of music and effects, with effects having the center channel all to themselves. The force of scenes like the truck chase in *Raiders* would be greater, not to mention an increase in the clarity of dialog, music and effects.

On the delicacy side would be instances when speakers #2 and #4 could be used for sounds that might be head-turning if they came from the far-left and -right (speakers #1 and #5). For example, a knowledgeable director and adventuresome production mixer would be able to add a tangible feel of reality, by recording selected scenes in stereo. The MS stereo technique undoubtedly would be the most practical method, because it would enable stable center-channel dialog positioning — as if the scene had been recorded with just a standard hypercardiod microphone — while the stereo feed would go to speakers #2 and #4. This procedure would avoid the problems encountered in previous stereo dialog recordings, although it still would help if the boom operator was careful to keep the person being miked on-axis as usual, and not play the scene for stereo effect. It also would help if there would be little cutting between actors on opposite ends of the frame.

Another benefit of the MS stereo system would be that the "S" (figure-of-eight), side-facing mike could be brought in to flavor the "M" (cardioid), forward-facing mike, or easily abandoned if the stereo effect becomes distracting.

As with the discrete six-track mixes of yes-

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STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

☐ **ROCK MOUNTAIN STUDIO** (Atlanta, Georgia) has added a remote truck. Housed in an International Harvester 26-foot truck, "Big Ray" is equipped with a modified Bi-Amp 24×8 console, an Ampex 2-track, dbx limiters, Orban parametric EQ, and a DeltaLab Acousticomputer. The unit carries a full complement of Sennheiser, ElectroVoice and Shure microphones. *2528 Chamblee-Tucker Road, Atlanta, GA 30341. (404) 455-6600.*

☐ **CRITERIA RECORDING STUDIOS** (Miami, Florida) announces the appointment of **Richard Lee** the the position of general manager. According to studio president **Mack Emerman,** Lee will help diversify Criteria's activities to include the field of audio/video/film post-production. *1755 NE 149th Street, Miami, FL 33181. (305) 947-5611.*

☐ **SPECTRUM RECORDING STUDIOS, INC.** (Deerfield Beach, Florida) has upgraded to 24-track with an MCI JH-114 with Autolocator III and additional I/O modules for their Sound Workshop Series 30 console. Other recent equipment additions include an MCI JH-110B 2-track, Electro Voice Sentry 100 monitors, Nakamichi cassette deck, Sennheiser microphones, and a totally revamped cue system with Yamaha amplification and Fostex headphones. Spectrum is managed by **Ray Lyon** with **Michael Grosso** serving as chief engineer. *999 South Federal Highway, Deerfield Beach, FL 33441. (305) 428-0119.*

South Central:

☐ **THE SHOCK HOUSE** (Nashville) has added new gear to their 24-track studio, including a Fender-Rhodes piano, JBL 4430 speakers, a Lawson echo plate, and a Lexicon delay unit. *920 19th Avenue South, Nashville, TN. (615) 320-7161.*

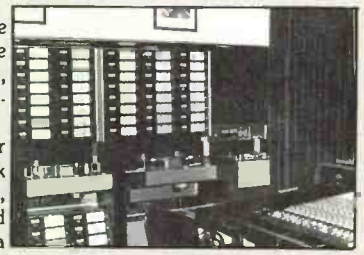
☐ **EAGLE, INC.** (Nashville) has purchased an Ampex AG-440B half-inch 4-track to facilitate its jingle and post-production work. The tape deck also is available for outside rental. Other additions include an Eventide 949 Harmonizer, Audio Arts parametric EQ, and an MXR 15-band EQ for room tuning. New mikes include a Neumann U-87 and KM-84. **Lynn Fuston** is the studio manager. *50 Music Square West, Suite 407, Nashville, TN 37203. (615) 320-7161.*

DARCI SOUND

☐ **DARCI SOUND RECORDING STUDIO** (Beaumont, Texas) has upgraded from 8- to 24-tracks with the addition of an Ampex MM100. Installation was handled by **Craig Jackson**. 2736 North 11th Street, Beaumont, TX 77703. (713) 898-4556.

☐ **SOUND EMPORIUM RECORDING STUDIOS** (Nashville) has been purchased by **Jim Williamson, Roy Clark,** and **CAC Investments.** Under the new agreement, Sound Emporium now consists solely of Studios A and B at the Belmont Boulevard location. All other interests have been deleted. **Jim Williamson,** president of the facility, stressed that the new owners will seek a closer working relationship with their clients in respect to budget considerations and special rate packaging. *3102 Belmont Boulevard, Nashville, TN. (615) 383-1982.*

☐ **DISC MASTERING, INC.** (Nashville) is the new name for Randy's Roost. Located in the RCA-Nashville Building West, the facility features Studer tape machines, a Neumann VMS 70 lathe with SX-74 head, and a Neumann SP 75 console with Neve 2087 custom equalizers. **Randy Kling,** the owner/operator of the studio, also announced the appointment of **Lois Walker** to the engineering staff. *30 Music Square West, Nashville, TN.*



CASTLE RECORDING

☐ **CASTLE RECORDING STUDIO, INC.** (Franklin, Tennessee) has aquired a second Studer A80 VU 24-track recorder and a Studer TLS 2000 tape lock system, thus providing 48-track capability. Other equipment includes a Harrison 3232C 48-input console with Autoset automation, Studer A80 and B67 half-inch and quarter-inch mastering decks, Lexicon 224 digital reverb, and Belle Klipsch monitors. Owned by Belgian native **Joseph Nuyens,** The Castle is tucked away on a secluded 35 acre hillside tract between Nashville and Franklin, Tennessee. The studio utilizes the natural stone and wood surfaces of the old home with drapes allowing selective use of live and dead acoustic areas. Other features are a full-length Bosendorfer grand piano and up-stairs living quarters for clients. **Martyn Smith** is general manager, while the studio manger and chief engineer is **Neil Wilburn**. *Old Hillsboro Road, Route 7, Franklin, TN 37064. (615) 794-0189.*

Midwest:

☐ **STUDIO A** (Dearborn Heights, Michigan) recently purchased a 16-voice New England Digital Synclavier II digital synthesizer. *5629 North Beech Daly, Dearborn Heights, MI 48127. (313) 561-7489.*

☐ **STERLING SOUND PRODUCTIONS RECORDING STUDIO** (Sterling Heights, Michigan) announces the opening of their new four-room, updated facility offering 4-, 8-, and 16-track recording. The equipment line up includes an Otari MX5050 Mark III/8, a Tascam 85-16B with six memory auto-cue, a TEAC M-15 console, JBL 4311 and Auratone 5C monitors, and Crown amplification. An Effectron ADM 256, LT Sound CLX2 compressor, ECC micro-plate reverb, and a Soundcraftsman AE2000 Real-Time Scan-Analyzer/EQ round-out the outboard collection. The studio is owned by **Rick** and **Gordon Carver**. *33018 Breckenridge Drive, Sterling Heights, MI 48077. (313) 977-7829.*

☐ **A SQUARE STUDIOS** (Ann Arbor, Michigan) has recently installed a 56-input Neotek Series IIC console, in addition to acquiring a new collection of outboard gear and a number of tube microphones. The 8.1 update has also been made to the studio's Lexicon 224X reverb. *3691 Morgan Road, Ann Arbor, MI 48104. (313) 434-2141.*

FAITH TAPE MINISTRY

☐ **AUDIO VILLAGE RECORDING STUDIO** (Bloomington, Indiana) has added Orban and Valley People parametrics, Valley LZ mike pre-amps, a new monitor amp by Crown, an Eventide Harmonizer and aStudio Technologies Ecoplate reverb system. *1000 West 17th Street, P.O. Box 291, Bloomington, IN 47402. (812) 332-7675.*

Mountain:

☐ **FAITH TAPE MINISTRY STUDIO** (Lakewood, Colorado) has upgraded its facility from 16- to 24-track with the addition of a Soundcraft SCM-762 24-track with Auto-locator, 24 tracks of dbx noise reduction, and a Soundcraft 2400 console. In addition to this gear, Barath Acoustics of Denver also supplied two Tannoy SRM-10 monitors, a Delta Lab DL-1 delay and a DL-5 effects unit, an Audi-Ence RFS-1 large plate reverb, Aphex Type B Aural Exciter, and an Orban 516-EC dynamic sibilance controller. *2166 Tabor Place, Lakewood, CO 80215. (303) 238-2341.*

☐ **BREEZEWAY RECORDING STUDIOS** (Waukesha, Wisconsin), formerly of Menomonee Falls, Wisconsin, has opened a new studio in Waukesha, Wisconsin. The new 8- and 16-track facility will feature a Sound Workshop Series 30 22/16 console feeding an MCI JH-24 16-track deck with full function Autolocator III, and an Otari MX-7800 8-track. Mix down is to an Otari 5050B Mark III-2. Among the outboard equipment are a Lexicon Prime Time with Memory Extension, a DeltaLab DL-2, Studio Technologies Ecoplate II reverb system, UREI and dbx compressor/limiters. Monitors are by JBL and Auratone. Mikes include Shure, Neumann, AKG, Crown PZM, Electro-Voice,

and Sennheiser models. The instrument list boasts a Hammond B-3 organ with a Leslie 147, a Fender Rhodes, Young Chang baby grand piano, and Prophet 5 synthesizer. Final design on the facility was by **George Zraick** of Studio supply, Chicago, **Jerry Milam** of Milam Audio in Pekin, Illinois, and **Fred Breitberg**, an engineer at Red Label Recording in Winnetka, Illinois. It features a fairly live music room with spruce walls and an oak floor combined with hidden trapping. The control room is of a European design with the console facing 90 degrees away from the window, instead of looking straight out at the musicians, and allowing for monitors to be placed on an axis with the listener, rather than above the window. Equipment was supplied by Milam Audio, Studio Supply, and Full Compass Systems. *363 West Main Street, Waukesha, WI 53186. (414) 547-5757.*

□ **STAR TRAX RECORDING STUDIOS** (Las Vegas, Nevada) has upgraded to 16-track with the purchase of a new Soundcraft SCM-762-24 recording deck with full auto-location and all the options. The announcement was made by studio owner **Larry Read**. *6156 Carl Avenue, Las Vegas, NV 89108. (702) 648-1510.*

Southern California:

□ **CUSTOM DUPLICATION** (Los Angeles) has added an Otari DP7000 high-speed duplication system with five slave system. *Los Angeles, CA.*

□ **HITSVILLE U.S.A.** (Los Angeles), a division of Motown Records, has purchased six Ampex ATR-124 multitrack recorders to replace older decks and to expand the studio's mastering capabilities. "We chose the Ampex ATR-124 because it is an ideal match for us in sound and electronic design," said **Guy Costa**, vice-president and managing director of the studios. "The system offers advantages for post-production work because the ATR-124 can handle 16-inch reels, which is ideal for one-hour, 30 IPS recording. We can now fully integrate our audio, video, and film capabilities to give our customers a total synchronization package." The facility has also installed three Neve consoles. *Los Angeles, CA.*

□ **ENACTRON STUDIOS** (North Hollywood) has taken delivery of an MCI JH-24 24-track recorder, a JBL 5234 Frequency Divider, and a White 4001 equalizer. In addition, **Howard Dresden** has been named maintenance engineer. *5102 Vineland Avenue, North Hollywood, CA 91601. (213) 761-0511.*

□ **AMBASSADOR COLLEGE** (Pasadena) has upgraded their radio production facility with the addition of fourteen Ampex ATR-800 full-and half-track recorders. The new decks, purchased from Coast Recording Supply, feature continuously variable shuttle, three-speed and variable speed operation, built in cue amplifier, single search-to-cue, and electronic tape timer. *Pasadena, CA.*

□ **THE JVC CUTTING CENTER** (Hollywood) announces the addition of **Joe Gastwirt** to the staff as a mastering engineer. Gastwirt was formerly with Frankford-Wayne in New York. *6363 Sunset Boulevard, Hollywood, CA 90028. (213) 467-1166.*

Northern California:

□ **PATCHWORK PRODUCTIONS, INC.** (San Rafael) is offering time and instructions on their new Alpha Syntauri Studio Pro Composers Workstation. The facility recently moved from their San Quintin, California address. *2111 E. Francisco Blvd., San Rafael, CA 94901. (415) 459-2331.*

□ **HYDE STREET STUDIOS** (San Francisco) has acquired an MCI JH-24 24-track recorder for its Studio C. The 3M 16-track formerly in Studio C is now in Studio A. Hyde Street has also added a new Otari 5050-B II half-track recorder. Engineer **John Cuniberti** has been promoted to the post of studio manager. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

□ **CUSTOM RECORDING/STUDIO C** (Stockton) has added engineer/synthesist **Ralph Stover** to the staff to handle the increased demand for the studio's New England Digital Synclavier digital synthesizer. *2220 Broadridge Way, Stockton, CA 95209. (209) 477-5130.*

□ **THE AUTOMATT** (San Francisco) has completed upgrading its Studios A and C. Two Studer 24-track tape machines have been added, with the result that Studio A is now a Trident/Studer/Meyer studio. Other new gear includes two Lexicon PCM 42 digital delays and two Studer 2-track recorders with half-inch capabilities. The announcement was made by studio manager **Michelle Zarin**. *827 Folsom Street, San Francisco, CA 94107. (415) 777-2930.*

STARLIGHT SOUND

□ **STARLIGHT SOUND STUDIO** (Richmond, California), owned and operated by **Peter Brown**, ahs added a fully automated Harrison 4032 console, a Lexicon 224X digital reverb, an Ampex ATR-102 half- and quarter-inch 2-track, and a Yamaha C7 grand piano. New arrivals can also be found in the extensive microphone collection. **Norman Kerner** is the studio manager and staff engineer. *617 South 13th Street, Richmond, CA 94804. (415) 236-2281.*

□ **WINE TREE VILLAGE RECORDING STUDIO** (Claremont) has re-opened after installation of a Harrison Series 4 28×24 console complete with Sound Workshop ARMS. automation. The console is interfaced with an Otari MTR-90 Series II 24-track recorder with Autolocator, a Lexicon 224X digital reverb with the newest 8.1 updates, and an Otari MTR-10 master recorder. A selection of vintage tube mikes is available, including the Neumann U47 and M49. A Yamaha Grand piano is in the music room, and acoustical design was handled by **Brian Cornfield** of Everything Audio. *Claremont, CA.*

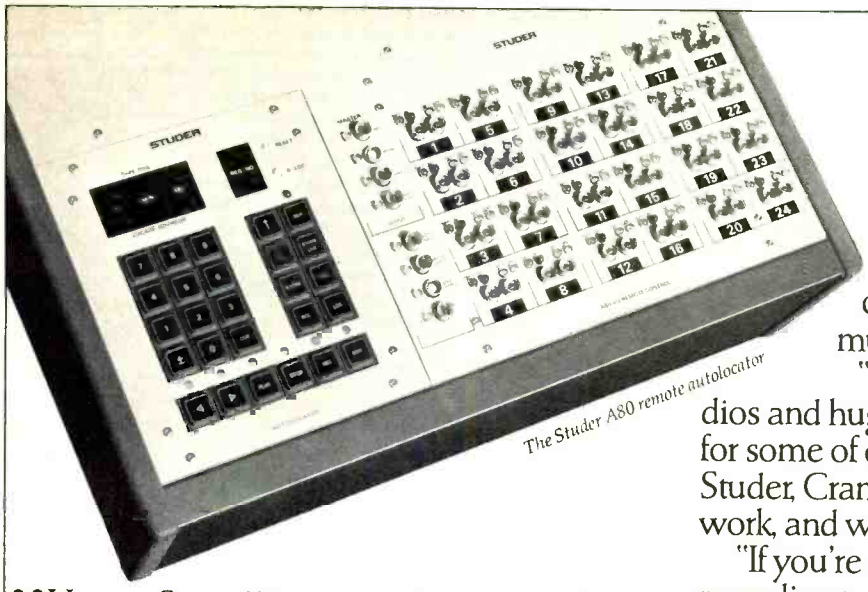
Canada:

□ **COMFORT SOUND RECORDING STUDIO** (Toronto) has taken delivery of a new Ampex MM1200 as well as a new Auditronics console. The new equipment will allow Comfort to offer 24-track recording along with 16-, 8-, 4-, and 2-track. *2033 Dufferin Street, Toronto, Ontario, Canada, M6E 3R3. (416) 654-7411.*

□ **MUSHROOM STUDIOS** (Vancouver) has acquired a Fairlight CMI Music Computer through Digital Audio, Inc., the sound creation and musical production company of Vancouver composer/producer **Jean Piche**. The unit offers sounds, effects, and composition capabilities, and the services include Piche's programming and arranging expertise as well as a complete orchestral sound library. The system is coupled with Mushroom's 40-track recording studio. *1234 West Sixth Avenue, Vancouver, Canada, V6H 1A5. (604) 734-1217.*

Australia:

□ **AAV STUDIOS** (Melbourne) has taken delivery of two Sound Workshop Series 40 recording consoles. According to AAV's manager of audio, **Roger Savage**, a new 32-input VCA grouped unit is being used for live concert recording and television simulcasting, and has been customized for mobile use with centrally positioned monitoring and a separately located patch bay. The console and two 24-track recorders have been installed in an Isuzu truck under the direction of **Peter Colby**, head of engineering and development at AAV. The second new console is for use in Studio 3, the province of voice-over production manager, **Brian Lawrence**. The mixing panel is equipped with parametric EQ and high resolution metering, including a spectrum analyzer display. Supporting facilities include a Lexicon 1200 Time Compressor. Rank Electronics Pty. Limited of New South Wales, Australia, supplied the gear to AAV, which operates six recording studios and a major video production facility. *Melbourne, Australia.*



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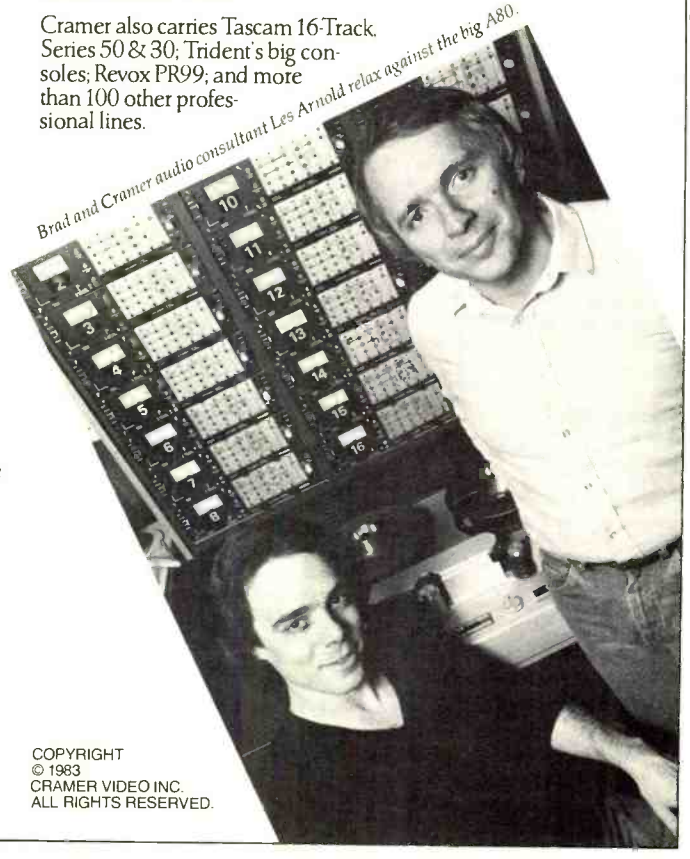
Brad and Cramer audio consultant Les Arnold relax against the big A80.

For additional information circle #46

STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

AUDIO/VIDEO UPDATE

Eastern Activity:

☐ **SCENE THREE VIDEO** (Nashville) announced the acquistion of an Ampex Digital Optics special effects device for use in video tape editing. Scene Three offers one-inch post-production facilities with both ADO and 2-channel squeeze zoom. In addition to these facilities, a new Studer 24-track audio recorder is interfaced with the CMX 340X editing system for multitrack post-production sweetening. **Ray Charles** was recently in the studio to complete his first music videos to promote his upcoming Columbia album. The first, "3/4 Time," is set in a honky tonk cafe named "Ray's 3/4 Cafe," and utilized the production house's Mathews Tulip Crane. The second, also placed in the cafe, added visuals to the song "Wish You Were Here Tonight." Both concepts were developed by **John Goodhue** for Columbia Records. *1813 8th Avenue south, Nashville, TN 37203. (615) 385-2820.*

☐ **CRAWFORD COMMUNICATIONS** (Atlanta, Georgia) has purchased $1.2 million of audio and video equipment from Ampex for its new post-production facility in Atlanta. The equipment line-up includes eight one-inch helical scan videotape recorders, Ampex Digital Optics, an ACE Touch Screen editing system, and an AVC-31 production switcher. Audio is handled by an ATR-124, two ATR-800s, two ATR-104s, and two ATR-102s. The facility also offers off-line editing as well as the on-line bays. **Jess Crawford** is president of the firm. *Atlanta, GA.*

☐ **NATIONAL VIDEO CENTER/RECORDING STUDIOS** (New York City) provided post-production facilities for MCA recording artists **Catholic Girls** and their video promo of "Boys Can Cry," currently airing on MTV. National's video editor, **Ron Harris**, employed the Datatron Vanguard Editing System and Grass Valley mark II DVE for special video effects. **Walt Schoenknecht** produced and directed for Midnight Media Group, Inc. Catholic Girls' manager **Stewart Turner** served as executive producer. *New York City.*

☐ **SHEFFIELD RECORDINGS, LTD.** (Phoenix, Maryland) has installed a new 3M 24-track recorder in its audio/video truck. The 26-foot GMC mobile also houses an Auditronics console, Thomson cameras, and Ampex one-inch video recorders. *13816 Sunny Brook Road, Phoenix, MD 21131. (301) 628-7260.*

☐ **SCHARFF COMMUNICATIONS** (New York City) provided its video truck for the 24-track location recording of the fourth annual PBS "Gala of Stars," featuring **Beverly Sills, Placido Domingo, Cleo Laine** and special guest star, **Miss Piggy**. The program was recorded at Lincoln Center's New York State Theater and produced by WNET. The program was rehearsed and taped in one day because the theater is dark on Monday only. The Unitel Odyssey II video production truck was interfaced with Scharff's unit, with crews working 26 hours to complete the program on schedule. SCI used dual 24-track Ampex MM1200s with dual sets of Dolbys. The technical crew included audio consultant **Jay David Saks** and audio designer **Bill King**, who were assisted by **Mel Becker** and WNET's **Bill Flood** and **Roland Thomesen**. SCI's **Aaron Baron** was engineer in charge, while **Gary Rotta** was chief engineer. Bob **Aldridge** handled audio stage manager duties, assisted by **John Rutherford**. The firm also supplied special signal processing gear to National Video for post-production. *1600 Broadway, New York City, NY 10019. (212) 582-7360.*

ONOMATOPEIA

☐ **ONOMATOPEIA** (New York City) has opened its fourth studio, this latest designed specifically for video and audio/video producers. "Video producers need a facility geared towards creative production, not just recording," says studio president **Matt Kaplowitz**. "Audio-for-video demands that the engineer know how to cut music to timings, how to select the appropriate sound effect and to be able to customize sound when it is required." The studio features a 16×8 Allen & Heath/Brenell console, an MCI/Sony JH-110 series 8-track recorder, a complete array of noise reduction and signal processing gear, and an automated BTX 4600 system to synchronize up to four tape machines via SMPTE timecode. Among the shows posted in the new facility is "Brain Games," an experimental game show pilot being aired on HBO. *37 West 57th Street, New York City, NY 10019. (212) 688-3167.*

Central Activity:

☐ **PRODUCERS COLOR SERVICE/VIDEO COMMUNICATIONS DIVISION** (Southfield, Michigan) has installed an Aphex Aural Exciter Type B for their audio post-production studio. "The Aphex adds clarity to voice-overs and on-camera sync tracks that sometimes get lost in tape generations," explains mixer **Dave Collie**. "It's a tremendous help when going for hot music mixes." Producers Color Audio Post utilizes a modified Solid State Logic computerized console to mix from an Ampex MM1200 24-track directly to Sony BVH 1100A one-inch or Ampex AVR2 two-inch video. *24242 Northwestern Highway, Southfield, MI 48075. (313) 352-5353.*

☐ **OMEGA AUDIO** (Dallas, Texas) provided its remote facility to interface with the videotaping of The Marlboro Music contest at the Dallas Palace in Dallas, Texas. The show featured ten regional country acts who were all video taped and recorded on 24-track. Omega provided audio mixing to picture in post-production. The winner of the contest competition was the Brooks Brothers Band. Producer of the project was **Paul Becker** with Nocturne Productions, San Francisco. Engineering was by **Paul Christensen, Bob Singleton, Ken Paul,** and **Marvin Hlavenka**. *8036 Aviation Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

Western Activity:

☐ **THE SOUND SERVICE** (San Francisco) has purchased a new Neotek Series II 28-channel console to upgrade their audio-for-video post-production facility. *860 Second Street, San Francisco, CA 94107. (415) 433-3674.*

☐ **THE POST GROUP** (Los Angeles) recently completed a number of music videos for Millaney, Grant, Mallet, and Mulcahey including **Supertramp**'s "It's Raining Again," directed by **Russell Mulcahey** and produced by **Jackie Adams**. The Post Group has also completed The Pointer Sisters' "I'm So Excited," produced by **Jackie Byford** and **Fiona Fitzherbert** and directed by **Kenny Ortega**; and "Only the Lonely" by The Motels directed by Mulcahey and produced by Byford. Editor for the pieces was Doug Dowdle of The Post Group. The firm also posted "air Supply in Hawaii" for Danny O'Donovan Enterprises. The show, currently airing on HBO, was directed by **Mike Mansfield. Steve Purcell** edited for The Post Group. *6335 Homewood Avenue, Los Angeles, CA 90028. (213) 462-2300.*

☐ **NOVA SOUND RESEARCH** (Los Angeles) worked in conjunction with Red Dog Productions in video taping the bands Jack Mack and the Heart Attack and Farenheit. The tapings took place at Skaggs Telecommunications Service in Salt Lake City and were lensed as a live show to allow spontaneity and control. Nova Sound provided a full sound system including monitors and two mixers for the live portion of the show. George Massenburg Labs limiters and equalizers were used. After a live taping with an audience, a second taping was done to playback for pick-ups and extra coverage. Audio overdubs recorded after editing. Audio producer was **George Massenburg** with **Ron Pendragon** acting as recording engineer. **Ed Maloney, Steve Callahan** and **Ray Eldred** all of Nova sound acted as live engineers. *2323 Corinth Street, West Los Angeles, CA 90064. (213) 659-5292.*

☐ **ALCON VIDEO PRODUCTIONS** (San Francisco) and its MusicArts division have just completed post-production of ex-Dobbie Brother Pat Simmons' new video. The cut, "So Wrong," is taken from his solo album *Arcade* on Electra/Asylum. **Bruce Gowers** directed for Gowers, Fields, & Flattery, and was in San Francisco for the one-inch editing with ALCON's chief editor **Jayne Paul**. *950 Battery Street, San Francisco, CA 94111. (415) 397-0490.*

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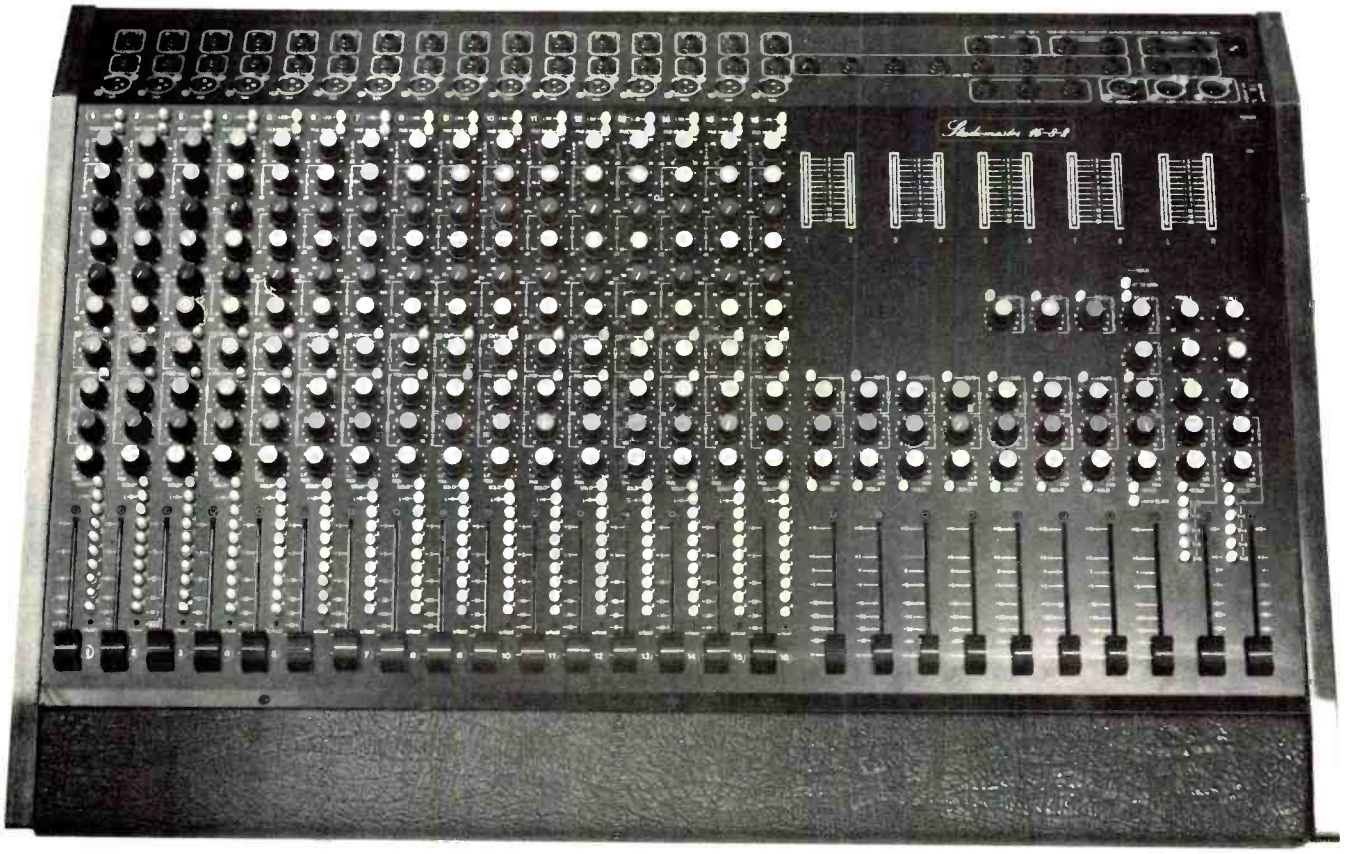

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RECORDING DRUMS AND PERCUSSION

by Robert Carr

Drums are unique instruments: they provide the driving force that supports musical ideas, and usually constitute the backbone against which all the other instruments play. The drumbeat is literally the pulse of the band; without it most music can easily end up sounding dull and lifeless. Yet at the same time percussion sets exhibit an enormous capacity for producing a wide range of expression, both from the drummer's playing ability, and also the engineer's technical expertise and creativity.

"Over the years I've found that drums are one of *the* most sensitive instruments I've ever tried to record," says veteran engineer Fred Catero. "They are very susceptible to dampness in the air, to temperature, and to where they are placed in the room. And in the hands of every individual, the same set sound *totally* different."

Just about everyone can play a couple of chords on the piano, for instance, and the sound will remain relatively consistent. But, from an acoustical point of view at least, drum sounds appear to vary more than any other instrument encountered in the studio. According to many studio musicians, the components of contemporary drum sets have been improved substantially with the introduction of plastic "all-weather" heads, and the evolution of quality-control methods employed by several leading manufacturers.

Brooklyn-born **Carmine Appice** had his first international success as drummer with Vanilla Fudge ("You Keep Me Hanging On"). In the Seventies Appice and basist Tim Bogert left the band to work with ex-Mitch-Ryder guitarist Jim McCarty in Cactus, and two years later with British rocker Jeff Beck to form Beck, Bogert and Appice. After five years of touring and recording with Rod Stewart (co-authoring "Do You Think I'm Sexy," and "Young Turks"), Appice set his sights on developing his solo career. He also maintains an active schedule of drum clinics, and writing educational books about drum technique. His current recordings are *DNA*, with guitarist Rick Derringer, and another Vanilla Fudge album.

However, such variations in drum sound shouldn't prevent an engineer from using "traditional" methods to maintain consistency at the instrument level, especially those that have proven themselves effective in many recording situations. For example: Paulinho da Costa, a Los Angeles studio percussionist, obtains his specially prepared calfskin conga heads from his native Brazil, and treats them with a fish-based oil to preserve their studio quality, regardless of weather conditions. "The heads on my main set, which are tuned quite high for a crisp sound, have lasted almost eight years," da Costa points out. "I concentrate on technique rather than power, and play softer with greater precision. Usually the drums hold their tone with no problem, even if the studio is a little wet or warm."

But the primary point to bear in mind is that once the sound leaves the instrument, and even how the sound is produced via the player's style, necessitates a serious study. Essential to developing successful techniques for drum miking — or any instrument for that matter — is a thorough knowledge of the theory and practical mechanics of its operation. (For in-depth information on the physics of drums, and the actual production of sound, the reader is referred to an excellent article by Robert Hodas, "Drum Tuning for the Studio Engineer," published in the April 1980 issue of *R-e/p.*)

Sound versus Music

Sadly, having observed some young engineers in the studio, as well as a few seasoned professionals, a brief overview of recording techniques in general, and how they specifically apply to drum miking, would be appropriate. The recording industry deals with music and sound, and unfortunately it appears, from this writer's experience at least, that there's a substantial percentage in the audio community who know very little about either.

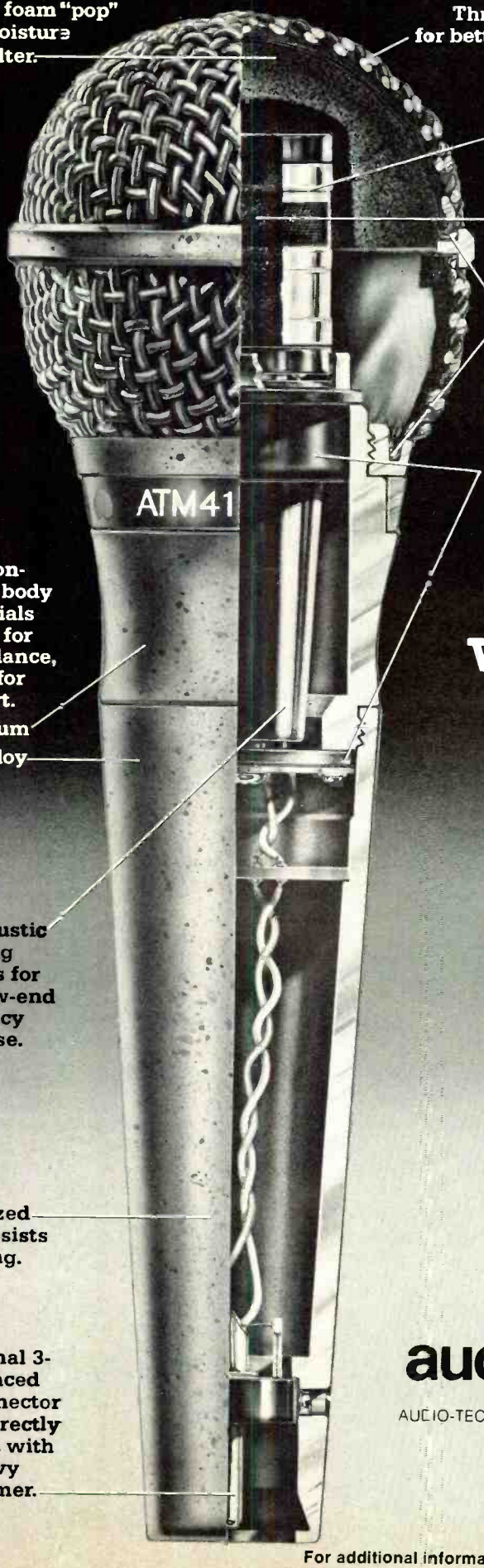
A studio/recording engineer has chosen to be a technically oriented artist who works in the medium of sound and, in essence, is no different than an engineer employed in any other field. Would you drive your car at top speed on a track built by an engineer who never really studied anything about the properties of the road's construction materials? Or fly in an airplane built by an engineer who didn't bother to understand stress factors, or aeronautical principles? Probably not. So why should anyone assume that an audio engineer doesn't need to have a thorough knowledge of the physics of sound, the parameters and components of music, or the means by which all the musical instruments generate sound?

An excellent analogy relating to this subject was offered by Fred Catero, and involves a violin maker choosing a piece of wood from which to build a new instrument. As a result of his study and experience, the craftsman is able to look at a piece of wood and know whether or not it will produce a quality violin. An unqualified maker needs to spend six months building the violin, only to discover that the wood was no good in the first place. The second case is a waste of time, talent and, ultimately, money for anyone thinking they are hiring a professional.

"At least half of the engineers today bumble their way through a session hit or miss," says Catero. "They don't bother to find out what happens when a sound wave is set in motion . . . when it hits a wall or other object . . . when it goes from one medium to another. No wonder they don't know what to do with screens [gobos], or what the pattern of a mike is supposed to do. There are strict laws of physics that sound adheres to. Understanding the laws of sound lets you become a better manipulator of it."

Audio engineering is not simply a matter of placing this mike in that position, with so much EQ, and panned to the left or right. An engineer acts much

Joe Porcaro acts as session drummer for the bulk of television soundtrack dates in Los Angeles, including *Fame, House on the Prairie, Falcon's Crest, Dallas, Fall Guy,* and *Fantasy Island.* While his son, Jeff, was collecting several Grammies for his musical work with Toto, Pocaro senior provided the driving force behind the orchestra for the live performances at the recent Grammy Awards Show. He's also an author of drum texts, an instructor and, in conjunction with Ralph Humphrey, a director of curriculum at the Percussion Institute of Technology, in Hollywood.



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like the hub of a wheel whose spokes are the producer, the musicians, the instruments, and a myriad of sophisticated recording equipment. An effective engineer should be able to translate a producer's request into an aural reality; make the drummer comfortable with the intrusion of microphones and cables around the kit; know how to get the most out of the electronic and signal-processing gear; and accomplish all of this in the shortest amount of time.

Contrary to popular belief, a set of drums is a musical instrument. Most drums have pitch, although the idea of tuning a kit is alien to some drummers. A properly tuned set of percussion instruments acts like a piano: hitting one drum sets another in motion in a sympathetic and pleasing fashion, whereas a set of drums tuned totally at random does not reinforce itself harmonically. No matter what kind of miking is tried, a badly tuned kit probably will not produce a big, full sound.

Even if the individual drums in the kit are in tune with themselves, the kit may not be in tune with anything else in the ensemble, which can spoil the tonality of the other tracks. Drum sounds, like the coloration of any instrument, change noticeably when added to a complete mix, as opposed to what is heard when the drums play by themselves. So important is this facet of the recording process that Grammy-winning engineer Al Schmitt strives to capture his final sounds in the track. "Too often, and I talk to a lot of drummers about this, engineers spend three, four, five hours or more finding the right settings," he says. "By that time everybody is tired, or not into the session anymore."

Schmitt spends only about 20 minutes working with the kit: "I try to get a sound as quickly as I can, so we can make a take, play it back, and let the drummer have the chance to hear it immediately. Then they make their minor adjustments. People like Jeff Porcaro or Steve Gadd know what their drums *should* sound like. If I have to make minor adjustments, I do that as we go along."

Toto's song "Rosanna," for which Schmitt served as recording engineer, was tracked on the second take. The basic tracks took about an hour, the drums, bass, congas, rhythm guitar and acoustic piano being laid down at the same time. Gobos were placed around only the bass and guitar amps, and the grand closed to the small opening. The conga player was located right next to the trap set, which was set up on a foot-and-a-half-high riser. (The riser eliminated the effect in the drum mikes of the bass leakage that normally runs along the floor, Schmitt recalls, and provided better eye contact between the drummer and the rest of the band.)

"While they ran down the tune a few times," he says, "I was getting the overall sound. By the time they were ready to

CREATING A LARGER DRUM SOUND

Duane Baron's Miking Techniques

Duane Baron came to the United States from Canada about six years ago. Over the course of the last five years he has worked his way up through staff positions at Pasha Music, Hollywood, and is currently an independent engineer. Baron recently completed an album for Carmine Appice and Rick Derringer, titled *DNA*. Here Baron explains the approach he and Appice followed while recording drum tracks for the *DNA* project.

"Carmine has Shure SM-57s mounted inside the toms and snare," he recalls. "Every microphone is [custom constructed to be] half the size of a regular 57. The miking system, marketed by May EA [Electro Acoustical] of Huntington Beach, California, was designed by Randy May, and installed at the Slingerland factory. Carmine originally planned to use the kit on stage but, when we tried it in the studio, it worked great. The set-up provides us with almost total isolation, plus we get the resonance off the bottom head, as well as the sound from the head on top.

"As long as the mikes are positioned correctly, there's no problem with transients, or too much air pressure on the head. There's no worry about phase cancellation either. Each microphone is positioned on a little bar [inside the drum body] that can be twisted around, or up and down, to adjust the angle.

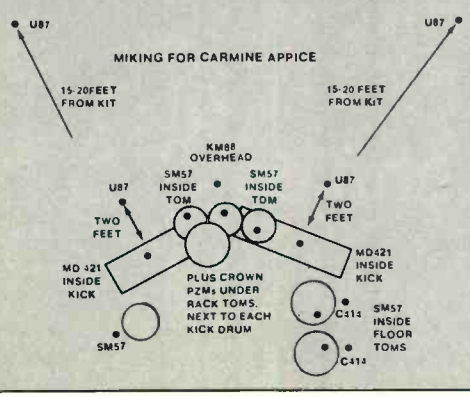
"The biggest advantage is that I still have the option to use other mikes outside the kit, and run them through effects [DDL, digital reverb] without losing the clean tom sound. Or I can 'effect' the inside mikes, and keep the outside signals dry. Then I have my normal miked kit."

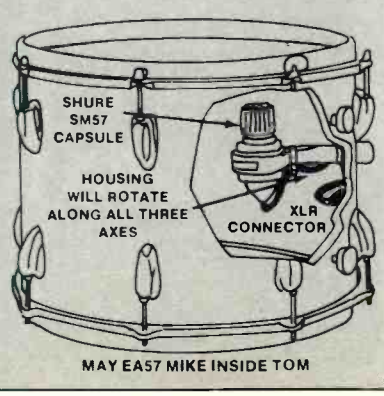
Overheads: "Generally, I use just one mono mike overhead, but I set up about three room mikes around the kit, and sometimes five — all Neumann U-87s. They pick up the cymbals."

Track Assignments: "I combine the room mikes and the overhead to two tracks [stereo left and right]. But you have to watch the phase relationship between the overhead mike and the distant mikes. When needed, I delay the overhead signal just enough to balance out all the mikes. In other words, the time setting I dial into the delay line is approximately equal to the length of time it takes for the cymbal sound to hit the room mikes, minus the time for the cymbals' waves to reach the overhead. If the room mikes are close enough, you won't run into a phase problem. But it depends on the room size and positions of the room mikes.

"So, from left to right on the board: kick inside, kick outside, toms left and right, snare, and room left and right — a total of seven tracks. There's no separate mike for high-hat, because I get plenty from the room and snare mikes. I'll use a separate track only if I want to put an effect on it. I usually get to hear the tracks before we record anyway, so I just work out the track assignments accordingly.

"We might also put glass, wood, or carpet around behind the drums for different tonal textures. It can be half way up the wall, or all the way. No one sound is greater than any other; they're just different, depending on the particular song." ■■■





Paulihno da Costa came to America in 1972 from his native Brazil, and has been working here ever since. After spending three years as percussionist for Sergio Mendez' Group, he quickly worked his way into the LA studio scene, playing a spectrum of percussion instruments. Some of his recent sessions have been with Quincy Jones, Lionel Richie, Michael Jackson, Earth Wind & Fire, George Benson, Pointer Sisters, Herbie Hancock, and Christopher Cross.

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CAPTURING NATURAL DRUM SOUNDS

Al Schmitt's Miking for Toto and Other Bands

Apex Recording Studio in New York City was where Schmitt gained his first experience with the music business, while recording many of the early Atlantic Records dates during the Fifties. In 1958 he moved to LA, and landed a job at Radio Recorders for about a year and a half, before settling at RCA. Schmitt went independent around 1968, and has since won five Grammys, the latest in 1983 for his work with Toto. His list of credits includes sessions with George Benson, Steely Dan, Randy Crawford, Bill Evans, and Henry Mancini.

"The most important aspect of recording any instrument is to get into the room and listen to the musicians play," he offers. "Then go back into the control room, and try to duplicate that as quickly as possible. I don't have a drum sound that I get all the time — I strive to capture the sound of the particular drummer. People like [session drummers] Steve Gadd, Jeff Porcaro [of Toto], or John Robinson [Pointer Sisters and Quincy Jones] have been playing for a long time; they've got their kits tuned the way they want them. My job is to get them that sound on tape."

Kick Drum: "I put my hand down in front of the drum when the guy is playing, and feel where the air dissipates. When I don't feel the air anymore, that's where I try to put the mike, usually a Telefunken [tube] 47. The positioning is pretty much on-axis, but that might have to be adjusted slightly for the drum and player."

Snare: "I use an AKG C452, or sometimes a Sony C-500, on the snare that comes up under the high-hat, and backed off a little from the drum. I try to aim it across the drum, to the sweet spot where he's hitting."

Toms: "I set up AKG C414s above the toms. Whether I mike each tom separately depends on the drummer, and the way he tunes his drums. If he has three toms, I might have three 414s — one for each drum. But for four toms, I may cut back to only two microphones: one on the two floor toms, and one for the two small toms on top."

High-Hat: "An AKG C452 on the high-hat faces straight down. The distance can vary from three or four inches, to a foot above."

Overheads: "I'll try to get the overheads [AKG C414s] as high as I can, which may be as much as five feet above the cymbals, or sometimes only three feet. When working in a really dead room I might have to come in a little more. Generally, I'll pull out a bit for a harder player.

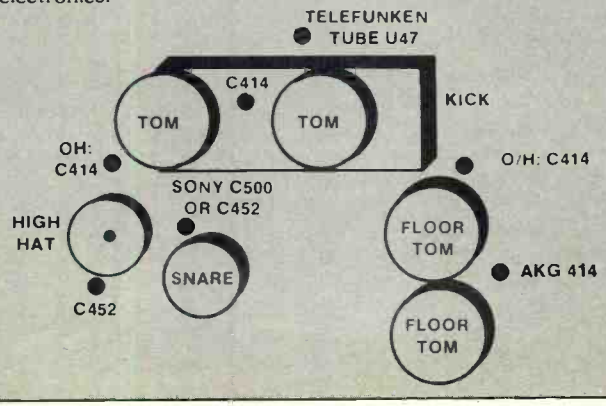
"The exact mike distances from any of the drums depend on how much 'air' I have in the room, and the drummer's style. But usually, the set up is not really tight. I may add a 10-dB pad on the snare or high-hat if I'm using a C452.

"I try for a complete, overall drum sound. The only time I worry about leakage — like the cymbals leaking into the tom mikes — is if the composite sound isn't good. Then I'll try to figure out where I need more separation, a different mike, whatever. Otherwise, I go by the overall sound in the monitors."

Track Assign: "The drums are usually recorded on four tracks: kick on one track; snare on another; and then the overheads, toms, and high-hat on two other tracks for a stereo field. I may separate the high-hat, but that's rare. I try to get my mix and my blend during the recording. I set up my drum sounds as if *I* was playing [the kit]. The bass and the snare are placed in the middle, and the rest of the kit is spread out from the high-hat on the left, to the floor toms on the right."

Kick/Bass Guitar Mix: "When I'm mixing, the first thing I'll do is get my bass sound, and then throw up the kick to get them both to work together. The snare and the overheads are next. From there I go to the Rhodes, the rhythm guitars, or whatever."

Equalization: "I don't use too many effects on bass drum to separate it from the bass guitar. I may use a tiny amount of equalization, but not very much. If I have to over-equalize a track — more than 4 dB up or down — either the microphone is in the wrong place, or the wrong mike is up there. It's so important to develop the proper mike techniques, and not depend on the electronics." ■■■



go, I was ready to go."

Fred Catero utilized a similar open set-up during Santana's *Ze-Bop* sessions, but still had plenty of isolation to overdub congas and timbales. "I can't impress upon you enough how much better an engineer would be if he studied acoustics," he confides. "We had no screens for these sessions, and everything was cut live. The congas were only three feet from the kit. The band was amazed that we had enough separation to re-do the congas and timbales.

"*That's* why you pay $1,200 for a mike — not to get the frequency response, but to get the pick-up pattern. If you know the pattern, what it's supposed to do, and how sound travels, you'll know how to mike and get good isolation."

Drum Setups

An appropriate drum sound starts with how and where the drums are set up. An experienced engineer working in a familiar room most likely knows the idiosyncracies of that environment. When working in a new studio, however, the second engineer can be the key to unlocking the room's secrets. "The seconds work there all the time, and they've probably found the optimum location for drums," says Catero. "Assuming it's a professional situation, I set up where he recommends. And when the band comes in I ask, 'Relative to the drums, where do you guys usually like to be when you rehearse or play onstage?' That tells me how to arrange the musicians for the session."

After weeks of rehearsing, the musicians in a self-contained band have conditioned their ears to hear all the parts in a particular way. Often the engineer inadvertently repositions the band members in a configuration to which they are unaccustomed. Hearing the keyboard player on the opposite side, or the guitarist's part coming from in front rather than off to the left, can be confusing for a drummer. The musicians already are disoriented, because their instruments sound deader than normal in the acoustically isolated atmosphere of the recording studio, and they then have to spend valuable time relearning their positioning.

"Drums are usually the only instrument that is really critical in terms of where they must go in the room," Catero continues. "Setting up the musicians based on the drummer's location always makes everybody feel comfortable, and saves time."

Professional studio musicians, on the other hand, should be sufficiently flexible and experienced to work with any kind of room arrangement. Yet to properly fulfill their assigned tasks, there are two considerations upon which pro musicians insist: eye contact with other players; and a good cue-system mix.

"I play a lot with the bass and the drums," says percussionist Paulinho da Costa. "I really need to listen to those instruments in my 'phones, and be able to have eye contact with the players so

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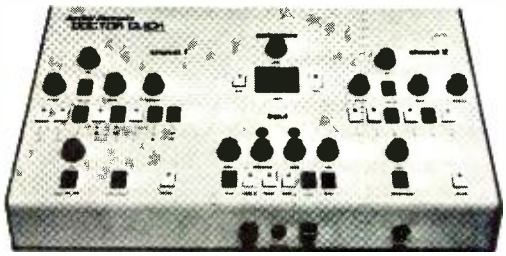

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WORKING WITH NO SET MIKING RULES

Fred Catero's Wide Range of Recording Experience

During the early fifties, at the age of 17, Fred Catero apprenticed in a small recording studio, and within a year was managing a studio in New York. From there he moved to Rockhill Recording for eight years, and then to Columbia Records as tape editor, mixer, and mastering engineer. In 1969, he and CBS staff producer David Rubinson relocated to California, and went to work for the Fillmore company. Within three years Catero established his own sound company, and is now president of Catero Records in San Francisco. (His company motto: "If it's commercial, it's probably on the charts; if it's good, it's probably on Catero Records.") To date he has received four engineering awards and 14 Gold Records with artists such as Santana, Janis Joplin, Chicago, Blood Sweat & Tears, Simon & Garfunkel, Herbie Hancock, Dave Brubeck, Bob Dylan, Count Basie, and many more.

"For recording drums," he advises, "there are no set rules, such as a given mike for snare, kick, etc. There are *too many* variables — the player's style, the kind of beater, whether the front head's on the drum, and so on. I have to see the drums, and be with them. Basically I try to use as few mikes as possible, because the more you use, the more acoustic phasing you're going to have."

Kick: "I would use a decent dynamic mike on the kick, like a Shure SM-56 or SM-57 — the standard workhorses. You can drop them, hit them with a stick, and they keep working. Position the microphone a little off-axis, just about even with the outside edge of the shell.

COUNTRYMAN EM-101 MIKE

"If the drummer refuses to remove the front head of the bass drum, you might have to mike it under his snare where the beater is. Although this gives you a good 'pop,' the snare may leak into that mike. You may avoid the leakage with a [Valley People] Kepex that gates the kick drum, and opens the microphone only when the beater hits. But it has to be set in such a way that the snare isn't too loud to open the gate as well.

. . . *continued overleaf* —

we can lock together. The drummer and I have to complement each other. If I can't hear him, I may overplay and get in his way. After that, a little piano track lets me know where the changes are, and the vocal part helps me tune into the emotion of the song and create a good part."

Trap drummers generally prefer a slightly different cue mix from everybody else on the session. The drums in front of him are so loud that he doesn't need those in his 'phones. Instead, a lot of bass, and perhaps piano or guitar, is a better reference.

Rock virtuoso Carmine Appice asks for a separate headphone mix with the drums turned down. "I'm used to the headphone cue, but play better live without them," he says. "Cans really effect your dynamics and the ability to get the individual volumes between all the limbs, because you're not hearing the cymbals or the high-hat that well. When you're not hearing something, your natural instinct is to hit it harder."

During his time with Rod Stewart's band, Appice found that replacing headphones with monitor 'speakers allowed him to hear the natural dynamics of his kit, rather than the electronic dynamics from the cue system. A pair of JBL 4311s were located right next to the toms at ear level, and about two feet away from each ear. "They were kicking ass," remembers Appice. "No drums in the feed — just all the other instruments.

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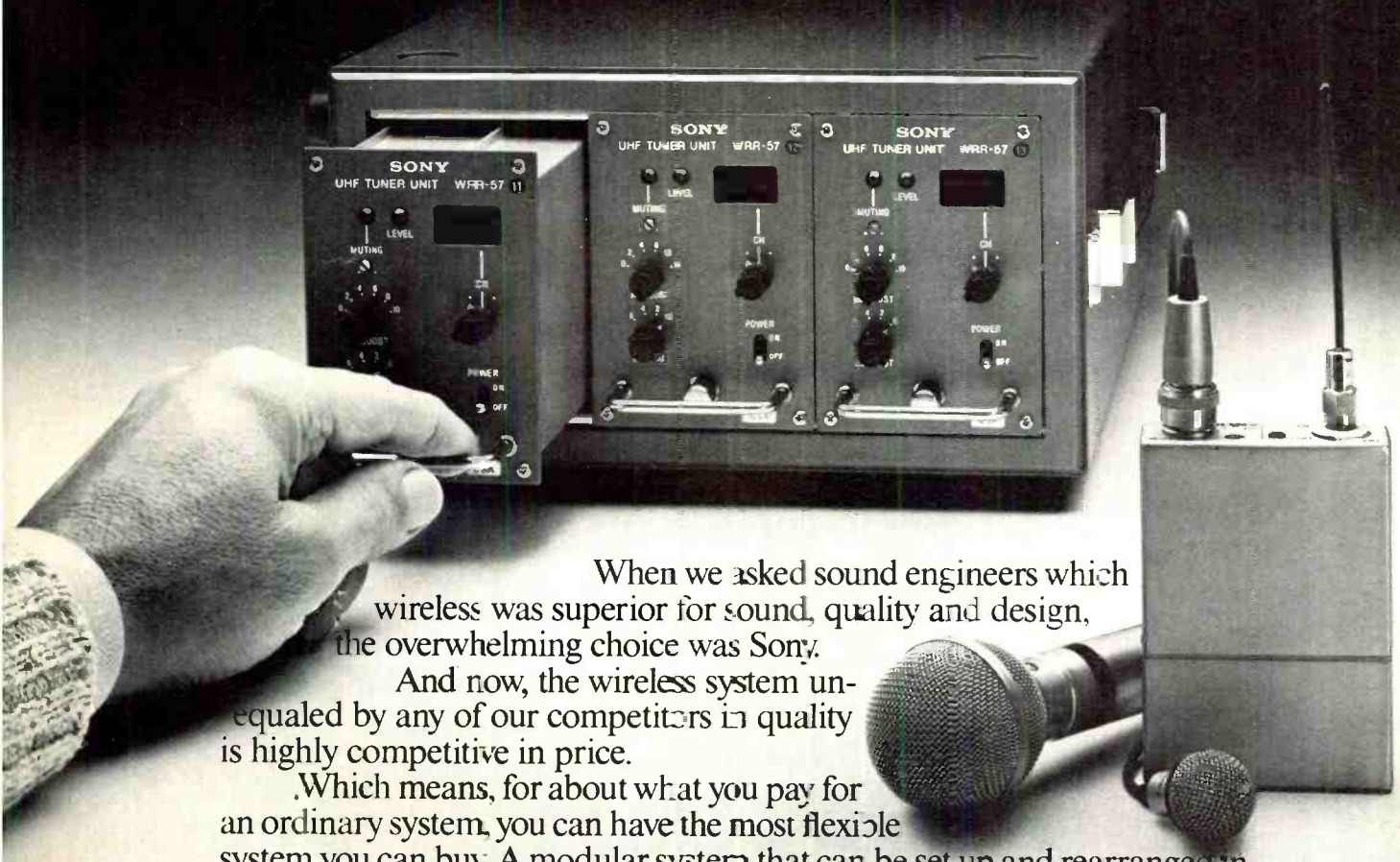

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By reversing the phase on the speakers, the signals cancelled each other out [at the kit], and didn't go into the microphones. I like the speaker setup because I can hear the actual tonality and dynamics of the drums."

Because Appice plays so hard, occasionally he has trouble with the drums going out of tune. Sometimes it's skin stretch, which just involves retightening the lugs. More often, the power of his blows knock the lugs loose, and the rims relax their grip on the heads. To prevent this happening, he's found that screwing a nut on to each lug prevents the rims from loosening up, and thus holds the heads in place and in tune.

Miking Techniques

Actual mike techniques from use on drum kits have been extracted from the main body of this article, and are included as sidebars for easy access to information. They consist of basic mike setups, accompanied by the respective engineer's explanations and recording philosophy. Their suggestions are meant only as a starting point, however, and not as absolutely infallible answers. But, for engineers with plenty of time to experiment, or who have the desire to find just the right sound for a particular tune, a library of reference recordings may be the way to go.

"For three days, we had the drums set up just about everywhere," says engineer Duane Baron, recalling his work with Carmine Appice for the album, *DNA*. "We cut tracks not only in the studio, but in the hallway . . . in the lounge . . . the rec room — everywhere we could think of. Sometimes we used one mike, sometimes two or three, sometimes 15. We mixed all the recordings down to a two-track reference tape, with a listing of notes and diagrams about what we did, where, and how. We were able to choose a sound from the two-track catalog to match the character of the tune we were working on."

Sticks and Grips

Once the session gets underway, it's not unusual for someone on the session to discover that the drum sound has changed from what it was in the beginning. In every instance, however, you do not always have to change mikes, positions, or even drums to find the magic tones. According to studio drummer Joe Porcaro, the musician may simply be able to change his sticks, or his grip on the sticks, and virtually provide the producer with a new set of drum sounds.

Drum sticks are manufactured with plastic and wood tips, the latter producing a "drier" sound than plastic, which brings out more of the undertones and overtones of the cymbal for a bigger spread. When a producer needs a drier cymbal sound, ask the player to change to a wood-tip stick, or vice versa for a "ringier" sound.

Altering the grip of the stick also changes the color of the kit. Impact of the stick against a drum or cymbal transmits vibrations through to the player's hand. By holding on to the stick in a loose manner, the vibrations travel more freely, and the cymbal, for example, rings more. By tighening up the grip, the vibrations are damped by the palm of the hand, and the ring is reduced, accentuating the attack for a staccato sound.

Similarly, the manner in which the drums are struck also changes their entire tonal complexion. There are two extremes for playing a drum. Using the snare as a point of reference: the player can obtain a sharp staccato effect by playing "into" the drum, which entails tightening the grip and holding the stick down after the impact — not on the drum head, but simple not letting the stick bounce up too far from the head. Joe Porcaro refers to this as "jamming the sound back into the drum."

"If you bounce the stick off the drum," he continues, "actually pull the stick up after impact in one continuous motion, then you're drawing the sound up for more of a legato [smooth] effect. That not only allows the drum to ring, but the looser grip lets some of the sound travel down the stick. That approach also works with cymbals and tympani.

"Too many young drummers buy heavy, 'pingy' cymbals that give them a dark, heavy tone. I would prefer to see them get a lighter sounding cymbal, and just approach the playing of it differently. The lighter cymbal affords greater versatility, because it can be used for a ride and a crash, depending on how you hit and hold the stick."

If such playing techniques don't provide enough depth to the drum sound, then perhaps a product called RIMS may be the answer. RIMS, standing for Resonance Isolation Mounting System, and manufactured by Gauger Percus-

FRED CATERO — continued . . .

"For a couple of hundred dollars, Countryman has a new mike [EM-101] that we've tested against a Neumann U-87, and found really no audible difference. It has a headroom of something like 150 dB, so you won't overload it, regardless of how close you get to the sweet spot. I don't say to do it, but the mike is so tiny — about ¼ by ½ inch — that you *could* tape it where the beater hits the head of the drum, or even on the beater itself. It would be much more convenient than the big, conventional mikes that get in the way."

Snare: "I like to use as small a mike as possible, like the Countryman, and put it about eight inches away from where the stick is going to hit — just to [make sure] the drummer doesn't hit the microphone. If I used a condenser, it would have to have a hard case, and a pretty direct pattern — rather than a wide cardioid — to pick up just the snare. Otherwise, you'll get a lot of high-hat leakage. I like the black pencil condenser mikes with the tilting heads [AKG C452s with CK Series capsules]. I bring those in toward the snare on top, and tilt the head down a little bit. I don't like miking the snare underneath, or top and bottom, although I have done it."

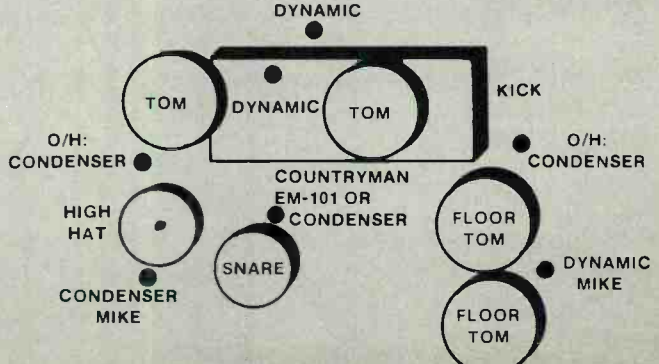
Toms: "Dynamic microphones are ideal for toms, because they're not sensitive to super-high transients that distort analog tape. And they don't have as much of a low-end frequency response as a condenser, which can go down to 20 or 30 Hz. A Shure SM-56, for example, goes down to around 40 Hz or so, and then rolls off. That's a definite advantage when you're miking drums. A floor tom produces information down to 20 Hz, and a condenser positioned too close to the drum will let that low energy through. Unless you're making an audiophile record, that information just uses up electrical energy that won't end up on the record anyway.

"Generally, for rack, floor and Roto toms I'll put up a dynamic between every two drums, about 1½ feet away. To get a decent balance, I'd rather move the mike either left or right between the two drums, instead of resorting to EQ.

"Whenever possible, I prefer to mike further away than most people do. But I find myself having to do what the producer and the musicians want, which could mean putting individual mikes on each drum, three inches from the sweet-spot. Most microphones [positioned] that close are over-driven; I try to back the mike up at least 8 inches."

Cymbals and Miking Underneath: "Each drum situation dictates a different technique. Although I've had to use under-the-drum mike positioning, I don't like it, because the mikes are facing the cymbals, and picking up all that leakage. You get much better separation by pointing the microphones downwards, with the back of the mike toward the cymbals.

. . . *continued overleaf* —



FRED CATERO — continued . . .

Dynamics sound bad on cymbals anyway, because they don't have the transient response of condensers to give you clarity."

High-Hat: "Definitely a separate condenser microphone for the high-hat facing away from the snare as much as possible for more isolation."

Overheads: "When I mike cymbals, I do it with condensers — Neumann U-87s, Schoeps, and AKGs — and keep them about a foot above to avoid the low-frequency vibrations of the cymbals. The microphones are about three feet apart and tilted slightly outward towards the cymbals in both directions, rather than toward the drummer. I get better separation that way."

Old Fashioned: "For an old-fashioned drum date using a regular kit, I cut back to four mikes: two overhead, one on the snare, and another on the kick. The snare is just a touch-up mike to support and round-out the pickup from the overheads. The kick mike, of course, is the main source for the bass drum, because the overheads are at right angles to the air pressure wave set up by the kick drum."

Track Assigns: "The kick goes on one track, the snare on a second, and two tracks for the main parts of a regular drum kit, which are the overheads, plus the tom mikes spread from left to right.

"If I'm doing a 24-track date, and there's just drums, guitar, piano, and bass, I afford myself an extra track for the high-hat. Otherwise, I put it with the left and right drums. The separate track allows me to put an effect on the high-hat, or control its level independently of the overhead signals, to get a better balance. I'm not an advocate of everything on a separate track, just because you want a better sound. You should go for the sound as you lay it down."

Effects on Kick Track: "I may find that Kepexing the kick, or running that track through a digital delay, helps separate it from the bass guitar. Or maybe putting the effect on the electric bass does the same thing better."

Drums to Build Mix: "When building the mix, I start with the drums as my central reference, usually because they have the most tracks. First is the bass drum, add in the snare, then the left and right stereo mix. When that's okay, I solo the kick and bass guitar, and try to get them to sound a bit different, so I can hear them as separate instruments.

"Instead of waiting until the mix, always check the acoustical phase of each of the microphones *before* recording, and go for the sound that is closest to what you want. If you get locked into two mikes on one channel that are out of phase, there's *no* way to change them. But even when you go to mix, reverse the phase one more time to see if it helps the drum sound. Don't be afraid to experiment." ■■■

arranges them in a cross formation, with the microphones forming an "X." "I place them between the drums," he says, "about a foot away, and aim them at the sweet spot where the drummer's playing."

Catero's first choice for close-miking congas is two dynamic models, usually with a mike devoted to each drum. Otherwise he'll go for one condenser mike overhead, and physically move it to the left or right to achieve an even acoustic balance between the two of them. The higher drum produces a reduced level on the console metering than the low one, he says. "The lead conga is generally about -3 dB on the meter, while the bass conga reads approximately +1 dB — look for that kind of relationship. When you put them in the mix, you should be able to hear both equally. Often, the lower-pitched one tends to get lost. The most important aspect I can stress here is to mix by ear, rather than by looking at the meters —nobody looks at the meters when they get the record home. Make sure it sounds good."

Tympani

Kevin Clarke has discovered that the best way to record tympani is certainly not close miked, or with a large-capsule microphone, "but rather with a small-capsuled AKG C452. The sound of the drum is so *huge*. The 452 seems to tighten it up substantially, and creates

sion, Inc., of Minnetonka, Minnesota, is a metal band that extends partially around the shell. In essence, the device suspends the drum, rather than grounding it with conventional mounting hardware. "The result is a 50% to 60% increase in resonance," says inventor Gary Gauger. "A drummer can use much smaller drums, and get just as big a sound as though he had a huge kit."

Timbales

"Miking about two feet above the timbales gives them a chance to breath," notes Fred Catero. He uses two good condenser microphones for the normal head sound. To pick up the rim playing, and the cowbell, which normally is located between the two timbales, Catero places a third mike — a dynamic — horizontally at rim level.

Al Schmitt also chooses two condenser mikes for timbales: "I might use Neumann U-67s, [AKG] C451s or 452s. With timbales, it's the way the guy plays, and what he's playing; sometimes he'll be playing the cymbal, and sometimes the side of he drum. The placement is *most* important. Listen to the drums in the room to get the sound, and then match it with the mikes. There are times [when] it's necessary to use 10-dB pads for the 452s, but try to avoid them as much as possible."

Congas

Schmitt also tends to go with two AKG C451s or C452s on congas. He

a very punchy, real sound."

If the player has three tympani, Clarke group-mikes the two higher-pitched one with a single C452, and adds another C452 looking at the very deep kettle. Both microphones are mounted about four or five feet above the drums. "Depending on how far away they are," he offers, "I'll choose either 10- or 20-dB pads. But if the capsule isn't being worked hard enough by the sound, too much padding destroys some of the excitement of the sound."

Less Is More . . .

Regardless of what kind of drums are being recorded, musicians don't always play all of their percussion instruments on every tune. A trap drummer may only need the snare, kick, high-hat, and one cymbal. Why leave the tom mikes live? "If I have a mike on a drum he's not using, I turn it off," says Fred Catero. "If he's only playing one of two floor toms, the second one rings sympathetically and muddies the track."

A microphone that is placed farther from the drums may be adding particular harmonics that are essential to the overall sound of an adjacent drum, or even the whole kit. "But you can get away with it for close miking," Catero continues. "And you get better signal-to-noise, better isolation, and less phase problems, while still maintaining the proper tone. This applies to guitar or bass, where they may be using two amps, and so on. I find a little communication always gets you better tracks."

■■■

JINGLE AND RECORD DATES — *A Conversation with Engineer Kevin Clarke*

Kevin Clarke comes from a family of musicians, and has done a certain amount of session work around LA himself. About seven years ago, John Baylor, a successful Los Angeles singer and arranger, asked Clarke if he would like to work at Baylor's home studio as an engineer. Clarke was put in charge of the facility, which expanded to 16 tracks 18 months later. He and Baylor are now partners in Tape Recorders, Inc, a 24-track room in Hollywood. Clarke is an independent engineer, but does about 50% of his projects in his own studio. Recent work includes jingles and commercials for Budweiser, Western Airlines, Continental Airlines, Lewis, the 1984 Olympics, as well as record dates for Quincy Jones (some tracks for *The Wiz*), Manhatten Transfer, Seawind, and drum sounds for the Oberheim DMX Drum Machine.

"Because Tape Recorders hosts many commercial dates," Clarke offers, "the studio has a house set of Pearl wood and fiberglass drums [three or four toms, Ludwig chrome snare] for visiting drummers. Although the microphone selection changes from one session to another, knowing the kit and where to position it saves a lot of time. I divide the players into two basic categories —'bangers' and 'tappers.' My choice of microphone varies according to where a given drummer fits in between these two extremes." Most of Clarke's comments refer to his jingle work, although he does contrast his approach toward that medium with his extensive experience gained on record dates.

"Unless we're doing a jingle that really features drums," he says, "we'll keep the set in a dead corner [of the studio] for added isolation. In the interest of time, recording is easier with the drums partially goboed from the rest of the room. The baffles are about four feet high — just enough to cut the immediate sound projection of the drums from moving too quickly across the studio."

Bass Drum: "In terms of what people want to hear, recording kick is really the least creative aspect of drum recording. It's basically putting a good dynamic microphone where you'll get just enough wind passing into it, and a proximity effect that creates a larger-than-life sound. I adjust bass drum mikes less often than anything else on the kit.

"The front head is usually off for a punchy, compressed bass drum sound. Mike positioning depends on how hard the guy is hitting [the kick]. For a real straight ahead R&B tune, the mike goes right in front of the beater, dead center of the drum. But the wind passing by the head of the mike can result in a loss of punch, like a 'P-pop' from a vocalist, but less obvious. And by going in too far, you lose the impact. My average placement would be [to have] the capsule four or five inches in from the outside of the drum, or about one foot from the beater on the inside.

"By moving the microphone a little off-center, but still looking at the beater, I can retain punch. Toward the edge of the shell I get a more passive sound. Except for an SM-7, the Shure dynamics don't seem to have a big enough capsule. I generally go with a Sennheiser MD-421."

Snare: "I've listened to close-miked top heads of snare drums for so long I'm sick of it. It's not *real*. And using only one mike on top tends to lose a little of the snare's presence. My favorite method is with two condensers — one on top and one on the bottom — and both heavily padded. I reverse the phase of the bottom mike, and create a nice depth on the drum. The bottom snare mike can also be a savior when the kit is set up tightly, with the high-hat close to the snare. The bottom head gives you a lot of the attack or 'speak' of the drum.

"I'll usually get further away with a 'banger.' But, in any case, close-miking either head tends to choke the sound. The expansion of the sound comes with moving the mike to about eight or nine inches above the drum, and as far back as possible. But if I hear the drummer is really sloshing the high-hat, the mike comes back in a little tighter to keep that leakage to a minimum.

"Generally on a jingle date I'm talking about a distance of four or five inches. With a banger, the mike gets back around the rim, looking more across the drum. If I can get some distance, I'll go with a heavily padded condenser, like an AKG C452 with a 20-dB pad. But I've worked with guys who hit the drums so hard, and have their top head so padded, that the only way to get any attack at all is with a dynamic, an SM-57, or a Sennheiser 441.

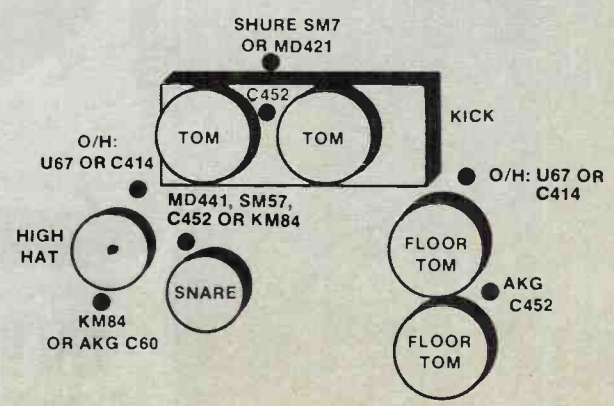
"For a little less of a banger, I may switch to a Neumann KM-84. The Neumanns are a little flatter sounding —not quite as 'pointed' as the 452 — and oftentimes [give] a prettier sound for a lighter touch drummer.

"The angle of the mike makes a tremendous difference. If the mike is looking straight down on the drums, tonality is lost. Being a little off-axis seems to keep any immediate reflections or air away from the mike diaphragm. A banger can move a [drum] head so violently that the air coming from the head may create a 'puffiness' within the capsule, which becomes really obvious when close miking.

"An average angle between the microphone axis and the drum head is about 45 degrees. But yesterday I recorded Jeff Porcaro, and ended up with the mike almost parallel to the head, looking back off the drum. So, there's no absolutes.

"On the bottom, I usually go for an angle complementary with the top head. If the top mike is at about a 45-degree angle, I place my bottom mike at an opposite 45 degrees, looking up the drum. When I have a good basic top sound that is giving me depth, some attack and tonality, and all I'm looking to do is pick up a little presence on the snares themselves, then I'll come pretty close to the drum with the bottom mike — maybe three, four or five inches away. The sound pressure level may be pretty tremendous, so I'm a little bit cautious with a condenser.

"If I want to make that sound deeper and open up the drum sound, I get that mike down just four or five inches off the floor, looking back up at the snare. With the right kind of placement I can get pretty good separation, without a lot of bass drum leakage into



it. The mike in this position is still padded as much as is available [on the microphone].

Toms: I almost always use dynamic mikes on a jingle session, because time is the most important consideration. I won't use a condenser for two reasons: firstly, everything is moving so fast that there's a tendency for the mikes to get whacked or dropped — neither is good for a delicate condenser; and secondly, over a TV speaker the added dimension and air that a condenser offers is almost inaudible. If the player is a banger, the dynamics capture the toms well.

"My ultimate set up for recording toms is [to use] C452s, because they are so small and the swivel capsules let you get into tight set-ups. For example, some ride cymbals sit very low over the rack tom. If the mike is too far above the head, all you've got is another overhead cymbal mike. I'm able to tuck the 452 into that tiny area, and get a successful sound.

"If there's a problem with the build-up, leakage, or phase distortion that's being introduced by close-miking, then I'll Kepex the more 'ringy' of the toms —usually every other one — instead of all of them. When I have the luxury of time and a patient drummer, I always try to get the mike as far away from the drum as I can, especially [to add] more space over on the deeper floor toms. The larger the drum, the more distance you really need to hear the tonality, and capture the sound of the drum. You can get in a little tighter on a smaller drum."

High-Hat: "My choice for high-hat is forever changing, and is based on the drummer's technique. Assuming he's a right-handed drummer, I always try to find out whether his left arm comes out very far when he's playing a two-handed high-hat part. I usually opt for hearing the side of the 'hat where the two cymbals rock and hit each other. I basically try to get a sweet combination between where he's attacking, and the side sound, but not to let any wind get back into the capsule when the mike is down on the side of the high-hat.

"I like to get the high-hat mike coming in along the left side of the drummer, and looking out away from the rest of the kit. Essentially, the XLR connector would be pointing up just past his left ear. If you visualize the high hat as an LP, the mike is on top, and back about three or four inches above the edge looking at the first cut, with the pattern capturing some of the stick hitting the cymbal.

"But sometimes I'm forced to mike the outside of the high-hat facing back towards the kit, which is not the optimum in isolation. My over-the-counter favorite is a Neumann KM-84 with a pad; I never use a condenser on a drum kit without a pad. I own an AKG C-60 — an old Norelco tube mike — which is superior for that application, but I won't use a tube mike for an average date."

Overheads: "I don't elaborately mike unless I have to. On a 'boom-chick, boom-chick' jingle session, I'll usually put a mike above the one crash cymbal he'll hit at the end of the tune.

"For anything more complex, I'll go for as much of a stereo-pair as possible, and concentrate my overhead pickup to the cymbals by coming in a little bit tigher than most people. My optimum on this new Manhatten Transfer album has been to put out my pair of Neumann U-67 tube mikes, which naturally capture the 'smile' and air that the cymbals put out.

"If it's rock and roll with crash, bang all the time, I'll get higher [up to four feet], and use a pair of padded AKG C414s, because they're so naturally bright. They require less EQ, and I'm *always* looking for less EQ. In radical situations I've used four overheads: two mikes for cymbals, and another two up maybe 15 to 20 feet in a big room."

Tom Spread: "If I don't know what kind of mix we'll be going for later on, I'll record the toms spread radically far left and right. I can always bring those overheads in closer to tighten up the mix. The main consideration is not to record anything dead center, to avoid the center channel build-up when the [AM and TV] broadcast engineers combine the stereo images and send it over the air. Any signals that are dead center, like bass drum, middle-rack tom, or vocal, suffer an incredible boost. For safety sake, I spread the toms out a little further. It pays to think ahead." ■■■

BUDGET RECORDING FORMAT

ACHIEVING BEST RESULTS FROM EIGHT-TRACK/HALF-INCH

Recording, Overdubbing, and Mixing Hints & Tips for This Flexible, Cost-Effective Multitrack Format

by Bruce Black

Half-inch/eight-track: the words often call to mind hastily converted garages where cars have been banned, the door nailed shut, and carpet hung from the walls. Memories may return of hum, buzzes, hiss, and tapes that sounded all *too* homemade. While, at first appearances anyhow, there might seem to be plenty of things to support half-inch/eight-track's undeserved tarnished image, the reality of the situation is that, in the right hands, it is a tape format that can produce a quality very competitive with the "big boys," and which can serve as a powerful, cost-effective medium. Which can be that is very good news indeed, at a time when record companies and advertising agencies face reduced recording budgets. Such clients cannot afford to let the final quality drop in the slightest, yet must maintain this quality when there is less and less money available to pay for it. Eight-track can provide an answer to such requirements.

Eight-track half-inch occupies a unique position in the analog recording world, with its creative capabilities and technical limitations. At one end of the spectrum there is two-track, which requires that all the stereo mix be established at the same time as all the instruments are recorded. At the other end of the format spectrum is 24-track, where usually the mixing decisions are made at a time later than when the instruments originally were recorded. Eight-track takes some of the advantages and limitations of each format, and creates a medium that requires unique capabilities and a broad range of special abilities on the part of the engineer.

Some of the limitations encountered when working with eight-track at times can be somewhat frustrating. The most obvious drawback is that you can only cram so much information on to eight tracks, and still maintain a reasonable control over the sound. Although you don't have the options offered by 16 or 24 tracks, the lack of multitrack flexibility can be made up with inventiveness, and sly craftiness on the part of the engineer. Necessity is the mother of invention when using half-inch/eight-track, and being in competition with studios that have more available tracks requires that you exercise this cliche to its maximum potential.

As many of you may have discovered, it can be disheartening when a client who was about to book time calls up to tell you that they've opted to go with that 24-track studio that's charging only $8 more an hour than your eight-track facility. (Don't ask *me* how they do it!) Your advantage, as an eight-track operation, is that you can keep studio rates low and survive, while the bigger facilities can't pay the engineer *and* the equipment mortgage, and always keep their price competitive. Also, since you must always be finding ways to make the available equipment do more and better things, you also are developing yourself into a more creative, inventive, valuable engineer. In the spirit of the "stay hungry" philosophy, you can't be complacent and slide through — your clients demand a lot more of you.

— the author —

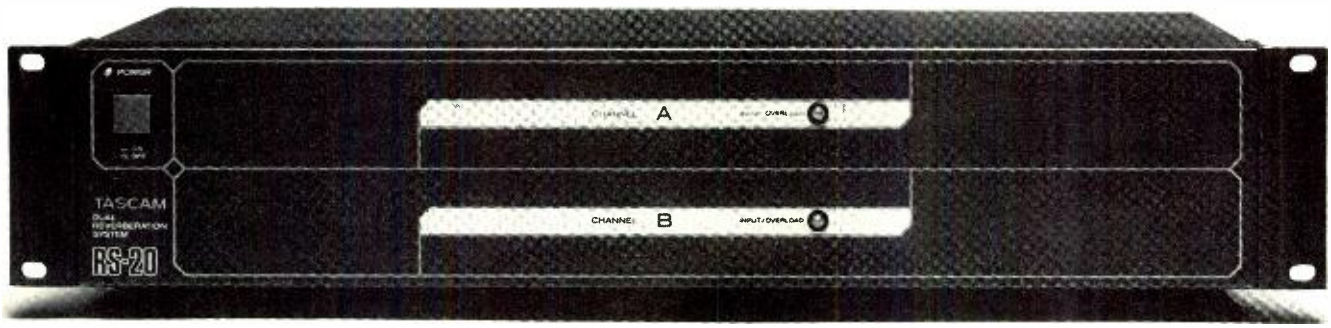
Bruce Black began working in audio as an adjunct to his musical endeavors, building a small sound system in 1970 for his band. Being one of very few people in his musical community with a sound system, he worked regularly mixing sound for a coffee house, clubs, and numberous local bands over the next few years. Later he apprenticed at several Hollywood 16-track studios, began mixing tour sound, and started doing four-track recording on the side with borrowed equipment. Currently working for Jensen Transformers, Black also operates an eight-track/half-inch remote recording vehicle, which he has named The Truck. Recent projects include a national TV spot, a film soundtrack for Swedish television, and several independent albums.

Quality Compromise

Another limitation often cited for eight-track is the available equipment quality. Recording eight pieces of discrete information on a half-inch wide piece of tape has its attendant problems of dynamic range, frequency response, crosstalk and, most noticeably, tape hiss. Over the past few years, however, improvements in half-inch/eight-track technology has significantly reduced such problems. With the explosive growth of smaller studios, both personal-use and commercial, there has been a similar growth in the amount of hardware appearing on the market. As a result, much more research and development is being devoted to producing less expensive equipment, and hence bringing more advanced technology, more features, more flexibility, and more quality to the eight-track facility.

While a certain amount of limitations have been reduced through major improvements in the hardware, there still remains room for improvement in the areas of tape hiss, and dynamic range. However, by adding noise reduction an eight-track facility can become very competitive in sound quality with more expensive studios. In terms of ease of operation and widespread availability, dbx noise reduction seems to be particularly suited to half-inch/eight-track recording.

But dbx does have its, admittedly minor, limitations that with eare and attention can be easily overcome. Sometimes a recording encoded on one dbx unit will not decode properly on another unit, resulting in the signal level "pumping" in time with a kick or snare drum, or some other percussive or bass-



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Many reverbs come with level controls, but not the RS-20—it's tweak free. After all, your mixer has echo send and return controls, so why pay twice for the same thing. Just set the rear panel sensitivity switch, and a pair of bi-color LEDs on the front panel help you set the mixer's send level. That's it.

Limiters ahead of the RS-20 drive amps then prevent overdrive or "twangy" spring sounds caused by high energy transients inherent in plucked guitar strings, etc., so the RS-20 remains "squeak free."

This exciting new unit incorporates a proprietary design with three different sized springs on each channel. Here's why: Most conventional single-spring reverbs have poor high frequency response, and those that don't usually compensate by "shunting" high frequencies directly through their reverb with a capacitor, a short-cut that cheats you of high frequency delay. The RS-20's multiple springs extend the frequency range at least an octave above conventional units (without a shunt). Also, three springs let us scale the decay time to the frequency for superbly natural sounding reverb.

You've got to hear it—sound that rivals high quality plates and digital systems, for a whole lot less. Even at twice the price, nothing beats the RS-20 Dual Reverb.

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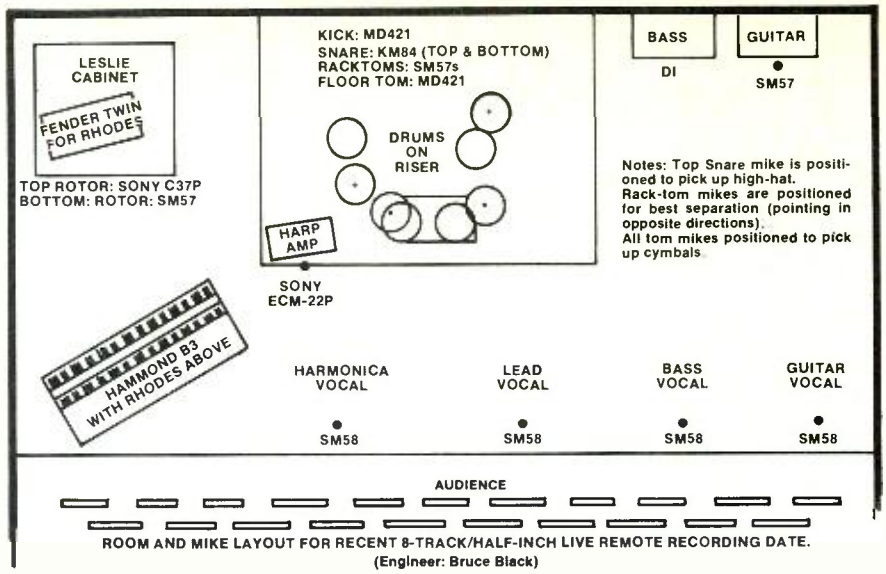
heavy instrument. While the problem is rare, it can be a very disturbing thing to have happen; the best way to avoid it is to encode and decode on the *same* dbx unit.

Session Planning

One of the most important requirements in seeing a recording project go smoothly and successfully from start to finish is advance planning. This requires particular attention from eight-track users, because of the frequent need to make mixing commitments while building up tracks. There also are a number of prospective clients who don't understand the implications and requirements of eight-track recording, and others who assume that you as the engineer will take care of everything. Eight-track facilities have the greatest exposure to clients with little or no prior experience in the studio, and therefore must develop ways of extracting information (short of bamboo under the fingernails) necessary for good advance planning.

Another common occurrence is to apply "24-track thinking" to eight-track capability, where a client wants each mike on a separate track and somehow doesn't realize that you very quickly will run out of tracks. You can't assume that a prospective client knows very much until such knowlege is actually demonstrated, and you'll sometimes find yourself gently educating them in things audio.

In planning out an eight-track recording project, it's important to find out how many instruments and of what type are to be recorded; how many are to be recorded at one time; how many are to be recorded in stereo; how many vocals there will be; and what kind of special effects are required. You need to decide if there will be any ping-ponging or track bouncing; the sequence of recording the various instruments; what demands will be placed on your outboard equipment; and how the tracks will be structured. While you are doing all this planning, you should keep an eye out for alternative ways of accomplishing the same goal, and for potential pitfalls that may occur during the project. Invariably there are going to be changes along the way, so it helps if you have scouted out the path, and know of ways to skirt unexpected problems and changes.



ROOM AND MIKE LAYOUT FOR RECENT 8-TRACK/HALF-INCH LIVE REMOTE RECORDING DATE.
(Engineer: Bruce Black)
TRACK ASSIGNMENTS: TRACK #1 — BASS, #2 — DRUMS LEFT, #3 — DRUMS RIGHT, #4 — HARMONICA, #5 — KEYBOARDS, #6 — GUITARS, #7 — BACKINGVOCALS, #8 — LEAD VOCAL.

The availability of outboard equipment also affects structuring of the tracks. If you have a delay line, or a spare tape machine, that double-tracked vocal can be done in the mix; one track saved. If you want a stereo string synthesizer, try using an Orban Stereo Synthesizer, or a delay line; one track saved. Want a stereo piano? Try using that active crossover that hasn't seen daylight in months. Feed the piano track to it, and pan the outputs left and right in the mix (play around with the crossover frequency until it sounds right); one track saved. This area of recording tricks is one special area in which to apply your sly craftiness to save tracks, and create new "phantom" ones.

Another important consideration during the advance planning stage is your track structure. As you go over the list of instruments to be recorded, keep an eye out for instruments that can go on the edge tracks one and eight. A phenomenon known as "fringing" causes the high-frequency response of edge tracks to noticeably fluctuate or "crackle." It's best to record low-frequency instruments, or certain types of percussion on these outside tracks, and avoid high-frequency instruments such as strings, synthesizers, bells, triangles, or anything with a sustaining top-end content. Because of the highs and sustain of cymbals, it's best not to print drum sets on the edge tracks.

Whether or not you'll be bouncing any tracks together will affect both your track structure and sequencing of recorded instruments. Ping-ponging is covered in detail a little later in this article, but it affects the planning stage because track bouncing requires a large percentage of the available tracks, and must be planned out in advance so that the transfers can be made early in the game. Remember that in a fit of artistic passion, a client will not be very interested in hearing "I'm sorry, but we're out of tracks." Advance planning will help minimize this problem, and enables you to foresee what possible change may occur, and what pitfalls may suddenly raise their gaping maw.

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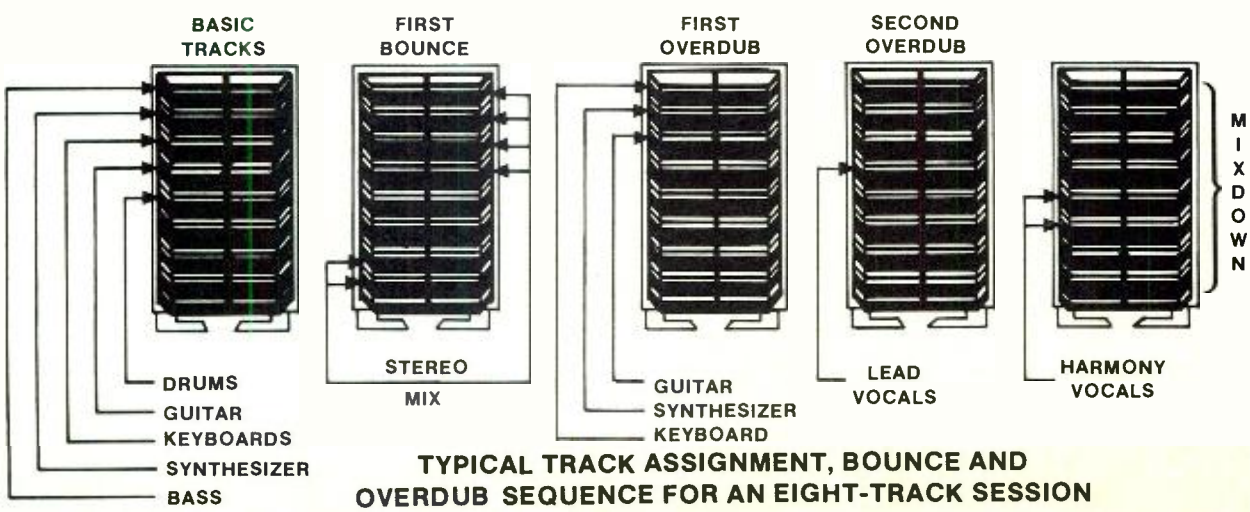

TYPICAL TRACK ASSIGNMENT, BOUNCE AND OVERDUB SEQUENCE FOR AN EIGHT-TRACK SESSION

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

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Track Bouncing

Because of the constraints of having only eight available tracks, you may have to rely on combining several tracks on to another one, to allow mixing decisions to be made *after* the performances have been recorded, but without tying up more than one track in the long run. This intermediate stage allows the engineer to concentrate on capturing a good performance, without also having to worry about the mix during the tracking stage. But the decision of whether or not you're able to bounce tracks can be taken away from you inadvertently early in the project if you don't structure the tracks correctly. For example, if you don't leave an open track that is separated from the tracks to be combined by at least one other track, ping-ponging won't be possible. Because of the narrow track width, magnetic feedback or "howlround" occurs when you try to record material from one track on to an adjacent track. Once again, this illustrates the need to plan ahead and structure the recording process so that the desired results can be achieved.

During track bouncing the tracks to be combined are played back from the sync or record head, and the resultant mix recorded on to another track, using the same record head to ensure that all the tracks remain synchronized with one another. As a result, you are at the mercy of the sync/record head's replay frequency response. Depending upon the particular eight-track machine being used, you may find that the high-end content of the tracks being combined is reduced during the bouncing stage. This is because the record head gap and other characteristics have been optimized for recording, not playback. Also, when you add high-frequency equalization to the mixed tracks being bounced, the tape noise or hiss is increased, which will be added to the inherent tape noise from the tracks being combined. It is at this point that noise reduction begins to make a lot of sense.

Another thing to remember during track bouncing is that once two tracks have been combined, you cannot pan each one of them to different places in the stereo specrum, nor can you change the mix between them. Of course, you *can* redo the bounce if you haven't tampered with any of the combined tracks, but once they have been erased or recorded over, you're commited to the mixing decision made during the bounce.

There is another method you might like to try for getting more out of eight-track. I call it "Cram Track" (cute, huh?), and it involves filling up every little open space on the tape with different instruments; it also requires the ability to count small musical intervals, and dexterous fingerwork on the record button. The ability to count small musical interals is necessary so you can find the little open spaces between notes where you can punch in a new part. A quick dexterous finger also is required to insert the punch-in in such a way that the multitrack has time to go through the process of putting itself in record mode, and yet none of the note before or after the section is lost. Sometimes this technique requires that you work with the client to make a few changes in the musical arrangement, allowing a punch-in to be made without erasing any of the existing recorded material. If this method is used frequently on a given tune, you'll be very busy during mixdown, possibly requiring a second set of hands. Also, usually there is not enough time to make EQ changes during mixdown, so it's important to make sure the sound of the instrument is right when it was recorded.

Cram Track and bouncing each lend themselves to a particular type of

AN EIGHT-TRACK/HALF-INCH CASE EXAMPLE

The Truck Mobile Recording Facility

Owned and Operated by Bruce Black

In equipping The Truck, Bruce Black chose equipment that offered, he says, a good compromise between cost and quality of performance. For his multitrack he uses an Otari MX-5050 MkIII-8, which he describes as "a marvelous, friendly machine," linked to a Biamp 1642 mixing console. For monitoring purposes Black uses a Biamp 883 console and a pair of Electro-Voice Sentry 100-As, or a pair of Auratones. The power amplifier is a Technics SE-9060. Mixdown is currently done to a Tascam 25-2 two-track.

Available microphones include an assortment of Sennheiser, Neumann, Sony, Electro-Voice, and Shure models. Outboard equipment comprises four dbx Model 180 noise reduction units for the eight-track, two Symetrix 501 compressor/limiters and a CL-100 compressor/limiter, two Biamp EQ-210 graphic equalizers, a Shure SR-107 graphic equalizer, an AKG BX-5E spring reverb, a Lexicon PCM-42 digital delay line, an Orban Stereo Synthesizer, an Echoplex, and two stations of Clearcom intercom.

Due to the sensitivity of spring and plate reverb devices to mechanical noise pick-up, Black hopes to purchase a good quality digital reverb unit sometime in the near future. "You should be able to pound on one of those and get nary a 'sproing' or rumble," he says. "In a remote situation, it's important to avoid picking up outside noises, or things or people banging against the truck."

Acoustic treatment of the 1½-ton stepvan began with a wall being added between the driver's compartment and the cargo area. The wall was framed with 2-by-4s, then covered on the driver's side with half-inch plywood. Spaces between the studs were filled with standard fiberglass, then the wall covered with Owens-Corning Painted Linear Fiberglas Cloth. This material also was used to cover the walls and ceiling to four feet back from the front wall, forming a fairly absorptive alcove in which the monitor speakers and console could be placed. The remaining walls were covered with Owens-Corning #735 E&A fiberglas insulation —which essentially is the same as the Painted linear Fiberglas Cloth without the gauze and paint covering — and then covered with decorative cloth panels. The open spaces behind the ceiling of tongue and groove knotty pine were filled with standard fiberglass.

Indirect lighting is used to provide a soft, uniform illumination, and a gentler atmosphere in which to work inside the vehicle. A three-lamp track light fixture is used to spotlight the console and equipment rack. All AC wiring is soldered to avoid any connection coming apart due to the movements and vibrations encountered in a remote truck application.

The microphone snake is kept on a Daniel Woodhead cable reel, and connection to The Truck is quick and easy via an AMP-G Series multipin connector mounted behind an external access door. AC power is also connected through this door, with two receptacles that feed the circuit breaker box from two, 150-foot three-conductor cables that can be used either in series or in parallel for reduced voltage drop. The vehicle's body can be connected to the ground pin of the AC power source, to provide improved RF shielding if required.

For further details about The Truck, contact Bruce Black at (213) 247-8358. ■■■

desired effect. Track bouncing works best and is easiest for combining tracks that are doing the same musical line, or are in harmony to each other, while Cram Track is best suited to squeezing in a separate musical line that is not in harmony with another track, or similarly related to it.

A note on punches: being able to do consistently good punches is a valuable skill, and requires that the owner of an eight-track be aware of certain things. No matter what the track format, a punch-in will always leave a small hole or blank in the recording at the point at which you hit the record button. The physics of the recording process require that the signal being applied to the record head is not just switched on suddenly, but that it be "ramped up" to avoid permanently magnetizing the head. This ramping-up process only takes a few milliseconds, so it's not too noticeable until you encounter the holes or blanks it makes when punching-in. Also, if you are piecing together a performance by punching in parts, you'll have a very difficult time if the instrument has a long decay time. If the decay holds over to the next note, and a punch is attempted, you will hear that blank spot loud and clear. But if you can place the punch *exactly* before the attack of the following note, the blank won't be as noticeable.

Another risk while making a punch is that you can inadvertently erase a portion of a recorded track you wanted to keep. When punching *out* — dropping out of record mode — you have to remember that the erase head sees the tape *before* the record head, which means that you'll also have a blank spot when you attempt to punch out of a track. The duration of this gap is a function of tape speed, and the distance between the erase and record heads. You need to anticipate when you do the punch-out so that you've stopped recording before the material you want to keep reaches the erase head, which is a point in time *before* you'd actually hear that material.

It goes without saying that the risk of removing a piece of track you want to keep requires that punches be approached carefully, and that the risks involved be determined beforehand. Such skills require experience to develop, so practice in situations where you can make mistakes, yet not get into too much trouble.

Outboard Effects

In addition to extending the capabilities of eight-track at the tape machine level, there are ways of enhancing the format by using outboard equipment. As well as using delay for double tracking, flanging, chorus, and repeat echo, analog, digital or tape-based delay can be used to create a concert hall-like ambience. The original sound is panned to one side of the stereo mix. Then, using an auxilary bus, a feed sent to the delay line, and its return panned to the opposite side; the delay time and feedback or regeneration is adjusted for the desired effect. A subtle improvement can be made by adding a little reverb to the delayed return.

A stereo synthesizer can be used to give tracks such as strings or horn parts a stereo feel. Try experimenting with a little delay on one of the returns from the stereo synthesizer, or add some reverb to one or the other returns, and try panning the reverb to various positions in the stereo soundfield.

Equipment Selection

Picking your multitrack equipment involves a number of considerations, such as price, availability, services offered by the seller, quality of equipment, financing availability, and features. Generally speaking, you want the *best* eight-track you can afford. In making your final equipment selection, there are two areas of equipment selection where the compromise between the various considerations should go in favor of quality (ie, spending more money). These two areas are microphones, and reverb or echo units, and they will have a more pronounced effect on the studio's overall sound quality than any other area of equipment purchase.

One of the least expensive microphones that will serve you well, and provide good quality service even as

your other equipment is updated and improved, is the Shure SM-57. This mike will set you back a little over $100, and will remain useful both in durability and sound quality for a very long time. When you get up into the $1,000 per mike category, it's easy to tie up a lot of hard-earned capital on just microphones, and many an eight-track operation has sacrificed some of the nicer things in life to buy a really fine mike.

Probably the one thing that eight-track users struggle with more than anything else is the client's (or their own) thirst for more open tracks. Compared to 16- or 24-track situations, eight tracks can sometimes seem pretty meager. Just remember that the inventiveness of an eight-track user will result in an increasing number of ways to extract more and more from the equipment at hand. The one thing that makes eight-track engineers stand out from other engineers is that they can't postpone all their mixing decisions until the final mix, yet still must be able to handle all the requirements of a multi-track situation. Drum mixes, vocal mixes, and anything else that requires recording more than one signal on to a single track must be well blended, because a bad mix will be haunting you through the entire remaining process. No one likes the unexpected hassle and expense of having to re-record a sour track, and putting a client through this can cost you their business.

Eight-track operators therefore have to develop the ability to mix some things early in the recording process, and in a way they're certain will work well in the final stereo mix. This requires the ability to "visualize" the final mix at a point where some of the instruments haven't even been recorded, and comes only from experience. You finally learn what to do, or what not to do, from each glorious success, or each ignominious failure, and is the process through which a quality engineer develops.

Quality of Engineering

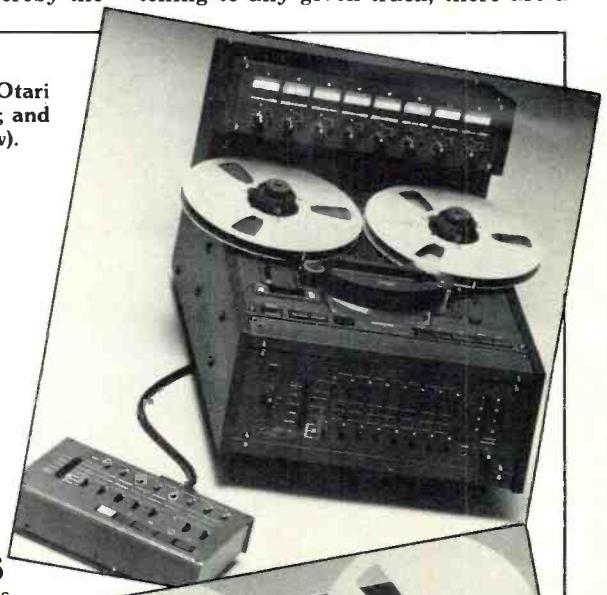
Recording engineers have to make many subjective judgements and decisions throughout the course of a session. The more one can recognize subtle sound qualities, and the more one learns about how sounds interact with each other, the better will be the creative decisions made during tracking and mixdown.

Having a "good ear" for sound is not something that's acquired like a piece of equipment, where once you've got it you don't worry about it anymore. Instead, it is a continuing process whereby the ability to recognize subtler qualities in sound is constantly being improved, through closely listening and studying sound.

There some exercises that can help to improve this ability. Listen to records, both ones that you think sound good, and those you think sound bad. Try to figure out what techniques were used in the recording process that make it sound good or bad. On records that sound good, examine the relative levels between the various instruments; in particular, the relative levels in the drum mix; and how the vocals fit in with the rest of the track. Listen to live music, and apply the same process.

This type of dissection may require you to put aside you personal opinion of the style of music for the moment. But if your intent is to hone your listening skills and creative judgement to a fine edge, that's part of the educational process you are putting yourself through. It's a process that will pay off in the long run, in terms of the quality level you can maintain as an engineer, and provides the added ability to make good subjective judgements on both technical and artistic levels.

This type of scrutinizing also can be employed during a mixdown. When listening to any given track, there are a



Tascam Model 58 (left); Otari MX5050 Mark III/8 (right); and Fostex Model A8LR (below).

OTARI AND TASCAM EIGHT-TRACK/HALF-INCH TRANSPORTS

Both Otari and Tascam produce high-quality eight-track on half-inch tape machines for the increasing demo, personal-use, and production studio markets.

The **Otari MX-5050-MKIII-8** is configured as a compact, tabletop console machine. A remote control unit, Model CB-110, provides total control of transport, monitoring, and record-mode functions, while the CB-116 autolocator provides single-button store and search of up to six cue locations, plus full "looping" capabilities. Nominal input and output levels to the MX-5050 can be set for +4 or -10 dB interface levels.

The **Tascam Model 58** features a fluorescent tape counter that displays positive and negative real-time tape position; during varispeed operation the counter displays percentage of the 15 IPS operating speed. All audio and transport adjustment trimmers are accessible from the front, even when the machine is rack mounted. A Zero Search button causes the tape to move forward or in reverse to the zero counter location; a single-point Search-to-Cue function also is included. Also available from Tascam: the "basic," budget-priced Model 38.

• In addition, Fostex recently announced an upgraded version of the Model **A-8** eight-track on quarter-inch tape machine, known as the Model **A-8LR**, which enables all eight tracks to be recorded simultaneously (unlike the A-8, which only allows up to four tracks to be recorded at a time). Both the A-8 and new A-8LR feature built-in Dolby C-Series noise reduction.

number of questions that can be asked. Can everything be heard? Is this instrument's level what it should be? Do the tonalities sound the way they should, and do they work well with the other elements in the mix? Do the tracks combine well, leaving the mix clean and easy to listen to? Is there something to remember the next time track bouncing is done? As the ability to recognize subtle qualities in sound improves, it helps develop a more refined creative judgement, and the answers to these questions become more in-depth, more apparent, and more easily found.

Improved Client Rapport

In addition to being well trained, experienced, more flexible, and highly inventive as an engineer, an eight-track operator often can be of more value to the client by providing better service. An eight-track studio usually sees a large share of artists that are fairly new to the recording process. Since such artists have a hard time getting used to certain aspects of playing in a studio —such as using headphones and performing alone to a track during overdubs, or being watched by those on the other side of the glass — you can provide a great service by keeping some helpful suggestions up your sleeve.

A number of bands and artists who use 16- or 24-track recording during their first studio adventure come away with a much poorer quality tape than they had expected. It takes a certain amount of experience to know how to manipulate that many tracks, and the flexibility and freedom available requires a good idea of what the end product will sound like, and how to get there. Multitrack recording is a tool for creating an artistic end product, and the broad flexibility of 16- and 24-track recording easily can become a large playground in which artistic goals quickly become lost. As such, eight-track represents an excellent format for artists new to recording. It provides a less intimidating atmosphere in which to become familiar with studio working, and its inherent limitations provide a better way to learn about recording, without the risk of going bananas with all those available tracks.

* * *

Half-inch/eight-track has come a long way since its early days. Currently it is enjoying rapid improvement due to high technology being applied to the hardware, and steady development due to its growing artistic and technical acceptability. Such considerations, combined with eight-track's excellent cost effectiveness in times when economics have become very important, make half-inch/eight-track a very powerful creative tool for an increasing number of people. In the hands of a capable engineer, it can provide recordings of superb quality at a fraction of what it might cost on 16- or 24-track. ■■■

New Products

MICROPHONE INNOVATIONS FROM NEUMANN, AKG, AND SHURE

NEUMANN UNVEILS TLM170i CONDENSER MIKE

The TLM170i condenser microphone is described as the first transformerless mcrophone of the fet 80 series. The direct, balanced signal output was achieved through the use of a completely new kind of electronic circuit, while maintaining a high degree of interference freedom and low current consumption. It has been possible, Neumann says, to reduce significantly the self-noise level of the microphone compared to similar types (14 dBA).

The TLM170i is able to handle sound pressure levels up to 140 dB with minimal distortion, which represents a dynamic range of 126 dB.

Five directional characteristics may be selected: omni; wide cardioid; cardioid; hypercardioid; and figure-of-eight. A future option will provide remote control function of the directional characteristics.

Excessive output levels, caused by high sound pressure levels, may be reduced by a 10 dB attenuation slide switch, while another switch rolls off frequencies below 100 Hz to eliminate low-frequency interference.

The microphone may be operated from the usual 48V phantom powering circuits, but will perform identically when operated from a 24V phantom source as well, without the need for switchover. The TLM170i is equipped with a tiltable, elastically suspended mounting bracket, which isolates the microphone effectively against mechanical noise interference.

GOTHAM AUDIO CORP.
741 WASHINGTON STREET
NEW YORK, NY 10014
(212) 741-7411

For additional information circle #58

ADVANCED AUTOMATIC MICROPHONE SYSTEM FROM SHURE

The new Shure Automatic Microphone System (AMS) is designed for use wherever speech-related multiple microphone setups are required, including conference rooms, churches, convention centers, and hearing rooms. In addition to sound reinforcement functions, the AMS is said to be suitable for recording and broadcasting applications.

The Shure AMS is described as a true "integrated system" consisting of complementary mixers and microphones that work together to solve the problems associated with multiple microphone installations. Each component incorporates several design innovations that directly address the needs of multi-mike users and installers.

The system consists of five basic component models: AMS22 microphone (low profile); AMS26 microphone (probe type); AMS4000 mixer (four-channel); AMS8000 mixer (eight-channel); and AMS 880 video switcher interface. Multiple AMS mixers can be patched together to effectively control more than 200 microphones.

Stand-out features include:

- *Simple setup for optimum performance.* Because the AMS is a complementary system of microphones and intelligent circuitry, there are no "threshold" or "sensitivity" adjustments to be made as with conventional automatic mixers. As a result, the AMS sets up quickly, and its operation is easy and uncomplicated.
- *Acceptance angle sensitivity.* Each AMS microphone is activated only by sound that originates within a 120-degree window of acceptance. Sound sources outside this window will not make the microphone turn on, regardless of their loudness.
- *Automatic, continuous sensitivity adjustment.* Microphones designed to work with the AMS are capable of individually adjusting their sensitivity in relation to the amount of background noise. As background noise increases, each microphone becomes less sensitive, and vice versa.
- *Reliable, consistent, quiet performance.* AMS's advanced circuit design is said to provide smooth gating action, eliminating clicks, pops, noise "pumping," and missed syllables.
- *Advanced logic capability.* AMS can

AKG ACOUSTICS ANNOUNCES "THE AKG TUBE"

To extend its present microphone line to include "tube enthusiasts," and to fulfill their specific sound requirements, AKG has introduced "The AKG TUBE." According to the company, AKG presently is the only manufacturer of professional "tube design" condenser microphones.

The unique feature of the new microphone is its tube-type pre-amplifier, using the 6072 (the same tube used within the AKG C12 condenser mike). Around this pre-amp the popular CK12 capsule and state-of-the-art componentry have been assembled to form a new condenser microphone. Directional characteristics from omni to figure-of-eight can be remotely controlled from the mike power supply.

For decades the electronic tube has been a necessity in circuit design — and not always a convenient one. The special power requirements, susceptibility of electrical noise and hum induction, non-linear transfer characteristics, and mechanical delicacy have been parameters, and sometimes limitations, with which the user has learned to live.

The solid-state technology of the last decade has overcome many of the shortcomings of the "tube age," and offered design engineers new possibilities for circuit layout of microphone preamplifiers: miniaturization, convenient powering, lowest possible self-noise, high overload capabilities, together with ultra-low distortion characteristics.

But, AKG reasons, there is one thing about microphones that should not be forgotten: the sound. And among all known microphone designs, the tube always has been regarded as one of the most pleasant-sounding alternatives. According to many recording engineers, this distinctive sound never has been equalled by any solid-state design.

Production of "The AKG TUBE" will be on a handmade and limited edition basis. Suggested US pro-user net price is expected to be $1,700.

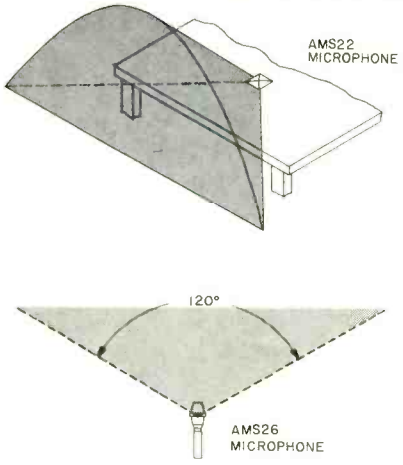
AKG ACOUSTICS
77 SELLECK STREET
STAMFORD, CT 06902
(203) 348-2121

For additional information circle #59

provide many additional functions, including channel muting, relay switching, video switching, microprocessor control interface, and other complex electronic operations.

The AMS26 is a probe-type, unidirectional condenser microphone with a front pop-filter grille and a dark brown finish that blends attractively with all wood finishes. The AMS22 is a surface-mounted, low-profile, unidirectional condenser microphone with an unobtrusive brown and black exterior.

To provide maximum flexibility, AMS mixers have a full complement of controls, indicators, inputs, and outputs, including: individual channel volume controls; master volume control; direct output jacks; channel LED indicators; hold time switch; off-attenuation control; normal and overload LEDs; aux inputs, outputs, and volume control; headphone output; and link jacks. In addition, the mixers feature logic terminals that make a wide variety of special functions possible, including chairman-controlled muting, loudspeaker muting, a "filibuster" mode, individual cough buttons, remote channel-on indicator, remote microprocessor control, and more.



SHURE BROTHERS, INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 866-2573

For additional information circle #60

MICMIX ANNOUNCE XL-404 PLATE SYNTHESIZER

The new Master-Room XL-404 reverberation system, called the "Plate Synthesizer," duplicates the reverberation qualities and properties of a plate-type reverberator. According to the manufacturer, unique technology is applied in this system (patents pending) to achieve the instantaneous, high diffusion of a plate.

The XL-404 is a fully self-contained, 5¼-inch rack-mount unit that provides adjustment of decay time on each channel from 1 to 4 seconds without damping. This important feature, MICMIX says, does not alter the overall frequency response as is the case with damping.

Other features include a four-band equalization section that allows the user to tailor the sound of the reverb from a warm-sounding plate, to one with an abundance of high-frequency content. A Mix control is provided that combines the direct and reverberated signals, along with a switch to select between stereo or mono operation. In stereo mode, the XL-404 operates in true stereo (as opposed to mono in/stereo out like most plates); in mono mode the echo density is effectively doubled due to the summing of both reverb channels.

According to the manufacturer, the XL-404 incorporates low-noise circuitry along with a toroidal transformer to achieve a superior signal to noise ratio when compared to the typical plate. Triple-shielding is provided to prevent acoustic or mechanical interference, allowing the unit to withstand high sound-pressure levels.

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

For additional information circle #61

New Products

GARFIELD ANNOUNCES "DOCTOR CLICK" UNIVERSAL SYNCHRONIZATION DEVICE

Described as a universal synchronization tool, the Doctor Click will read click tracks and built click tracks, live drum tracks, electronic drum tracks, and all of the sync codes used by Roland, Oberheim, and Linn. From any one of these drive sources, the new unit will provide the appropriate interface for practically every sequencer, drum machine and synthesizer made.

Since the Doctor Click will synchronize to click tracks, it is an invaluable tool to the film scoring business, Garfield claims. And, since it will sync to a live drummer, it allows the recording artist to cut his original tracks with a real drummer's feel, and then sync the computerized overdubs to *his* timing, instead of vice versa.

According to the manufacturer, these two features, coupled with the device's ability to read the sync codes used by Roland, Oberheim and Linn, and provide envelope modulation the rate of which is locked to the rhythm of the track, make it *the* synchronizing tool. In addition, the Doctor Click's metronome provides both beats per minute for musicians, and frames per beat for film makers, utilizing a 0.001% crystal.

GARFIELD ELECTRONICS
P.O. BOX 1941
BURBANK, CA 91507
(213) 840-8939

For additional information circle #63

TIMES ONE UNVEILS NEW AMPLIFIER SERIES

By incorporating a class A amplifier with a class AB amplifier using radio-frequency techniques, Times One says it has engineered three professional power amplifiers well-known for their clear open sound, wide stage imaging, and stability under load.

Features include 130 volts per microsecond slew rate; 0.01 to 0.05% THD 1 watt to full power, 20 Hz to 20 kHz; Damping Factor of 1,000, from 60 Hz to 10 kHz; 6 degrees of Phase Delay 20 Hz to 20 kHz; DC offset speaker protection; temperature status indicators; and dual mono construction.

System design of the Times One amplifiers is said to ensure an accurate picture of the actual sound. Studio engineers have commented that listening fatigue does not occur when using the Times One amplifier for monitoring. The apparent effect of "lifting the blanket off the loudspeaker," offers a dramatic improvement in clarity and presence.

List prices range from $1,600 for the 600-watt, dual mono Model RF600, to $499 for the 50-watt model RF100.

TIMES ONE
8000 MADISON PIKE
MADISON, AL 35758
(205) 772-9626

For additional information circle #64

LEXICON INTRODUCES PRIME TIME II

The new Model 95 Prime Time II retains most of the human engineering factors and control format of the original Model 93 Prime Time. Like Prime Time, the new digital delay processor features two independently adjustable delay outputs, complete input and output mixing, and a wide array of control and interface capabilities.

The Model 95 offers long delay capabilities: 1.92 seconds standard, and 7.7 seconds with memory option. Comple-

menting the unit's long delays are Lexicon's new and exclusive DRC (Dynamic Recirculation Control) and metronome features, which allow Prime Time II to be used as a short term digital audio recorder to create dramatic new sound-on-sound layering effects.

DRC automatically controls the level of recirculating delays to provide, according to Lexicon, clean articulated phrasing with long decaying echoes at the end of phrases. The metronome feature allows creative use of captured delay loops, which may be synchronized by the metronome clock, to drive external automatic drum and synthesizer equipment.

Prime Time II utilizes PCM encoding technology that provides ultra low distortion of 0.04% throughout its full power bandwidth, 90 dB dynamic range, and 20 Hz to 16 kHz audio bandwidth.

LEXICON, INC.
60 TURNER STREET
WALTHAM, MA 02154
(617) 891-6790

For additional information circle #65

PC400 SERIES PATTERN CONTROL HORNS FROM COMMUNITY LIGHT & SOUND

The new compact, constant directivity horn series, which includes the PC494 (90×40), PC464 (60×40), PC442 (40×20), and PC4124 (120×40), fit any 2-inch throat compression driver; adapters also are available for 1.4-inch, 1-inch, and screw-on drivers.

Frequency range is quoted as usable from 800 Hz up; pattern control is from approximately 800 Hz up (horizontal) and 1.2 kHz up (vertical).

Features include patented pattern control (constant directivity) technology, including "Dispersion Control Vane," which maintains rated coverage pattern over a wide frequency range in a very compact horn; unrestricted throat (no pinching in throat area), which eliminates undesirable throat reflections, and reduces throat distortion; and one-piece, balsa-reinforced fiberglass construction, which virtually eliminates undesirable resonances, and maintains strength yet allows very light-weight, weather-proof horn. Also, dimensions are less than half those of large constant directivity horns, yet PC400 Series horns are said to maintain pattern control throughout critical speech articulation range and driver loading to 800 Hz.

COMMUNITY LIGHT & SOUND
333 EAST FIFTH STREET
CHESTER, PA 19013
(215) 876-3400

For additional information circle #66

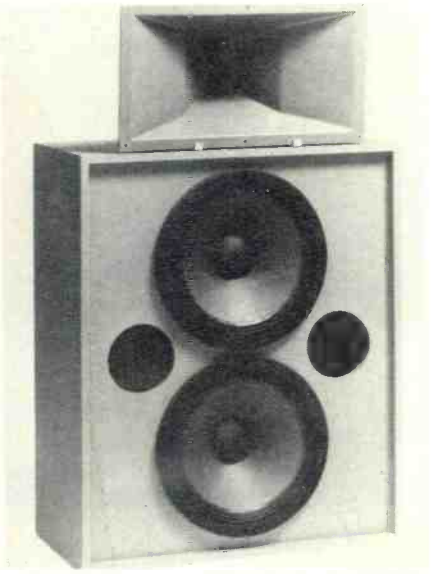
ALTEC UNVEILS NEW VOICE OF THE THEATRE SYSTEM

The new Model A6A matches a compact, optimally tuned, dual-woofer enclosure with Altec's newly developed mid-size Mantaray II Constant Directivity Horns. The Thiele/Small-tuned cabinet of the A6A was designed in conjunction with its high-efficiency 16-inch woofers for solid, powerful bass and the realism in voice reproduction that made *Voice of the Theatre* famous, Altec claims.

Through extensive comparative measurements and listening tests between the A6A and the popular A4 system, Altec engineers designed the new system to produce essentially the same body, punch and volume levels in the

New Products

theatre as the A4. And the A6A actually provides slightly better bass response from a cabinet less than one-tenth the cubic volume of the A4, Altec says. Overall, the A6A is less than 20 inches deep behind the screen, and costs about half the price of the A4 system.



Another major difference separating the A6A from other Voice of the Theatre systems involves sensitivity. The A6A requires more electrical power to operate, with 100 watts being the practical minimum for satisfactory sound levels in the theatre. This is a characteristic of all similar tuned-box/woofer systems.

ALTEC LANSING
1515 SO. MANCHESTER AVE.
ANAHEIM, CA 92803
(714) 774-2900

For additional information circle #70

TELEX ADDS REEL MASTER TO 300 SYSTEM

The new 10½-inch reel master transport option allows 300 system owners the opportunity to increase system speed, and gain the convenience of larger reel capacity with no modification to the rest of the units. The model

6300 master transport uses either standard 7- or 10½-inch NAB reels, and proper reel tension and torque for each size is selected easily with a convenient switch.

The system runs at an 8-to-1 speed ratio, which doubles the reel-to-cassette production available when duplicating from 7½ IPS masters. A tape speed switch selects either 30 or 60 IPS running speeds, with equalization set automatically.

All heads are pre-aligned and mounted on a heavy gauge plug-in headblock assembly, which also contains the tape guides and an optical infrared sensor for end of tape stop or stop-rewind-recue operation.

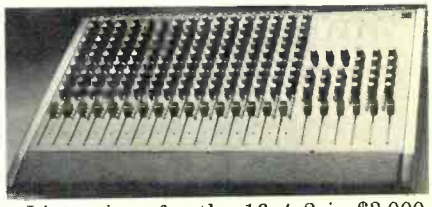
The 6300 starts at $2,800, and is available with quarter-track/four-channel and half-track/two-channel head configurations.

TELEX COMMUNICATIONS, INC.
9600 ALDRICH AVE. SO.
MINNEAPOLIS, MN 55420
(612) 884-4051

For additional information circle #71

TWO COMPACT MIXERS FROM STUDIOMASTER

According to company president Denny Handa, the two new 16-channel mixers — 16×4×2 and 16×8×2 — "represent simply the best value for the money. We've taken the features of our 8×4, and combined them with the real recording features of our 16×4. Plus we've added features like EQ defeat, 100mm faders and three auxilliaries. The result is a real crossover mixer. We see the majority of sales of the 16×8×2 into recording and video applications due to the surge of 8-track recording."



List prices for the 16×4×2 is $3,000, and $4,000 for the 16×8×2.

STUDIOMASTER INC.
1316 E. LANCASTER
P.O. BOX 2344
FORT WORTH, TX 76113

For additional information circle #72

AUDIO KINETICS ANNOUNCES Q.LINK INTERFACE

Q.LINK, a computer interface for use with the Q.LOCK 3.10C timecode synchronizer, includes the following features:

- External Q.LOCK control of Q.LOCK (dual Q.LOCK system controls five machines).
- Audio console automation/video editor control of Q.LOCK.
- External Computer control of Q.LOCK.
- Control assignable to Q.LOCK control unit while external line connected.
- Plug in option to existing Q.LOCK 3.10C synchronizers.
- New alphanumeric information and warning messages.
- Switch selectable RS232/422 standard at variable baud rates.

Interconnecting two Q.LOCK systems permits control of up to five machines from a single Control Unit; a

Cascade mode accesses the second Q.LOCK computer. Therefore, cue point memories, offsets and machine selections may all be accessed from a single control unit.

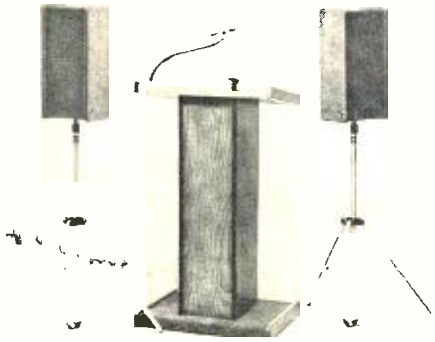
In addition, mixing console computer automation can be programmed to control three machines via Q.LOCK as effectively as it would normally control one, and selected video editors may be interfaced to any of the slave machines in the Q.LOCK library, be they audio, video or film.

AUDIO KINETICS, INC.
4721 LAUREL CANYON BLVD.
NORTH HOLLYWOOD, CA 91607
(213) 980-5717

For additional information circle #73

PORTABLE SOUND SYSTEM FROM VJ ELECTRONICS

The new mini-COLUMN is a completely portable sound system including pre-amp mixing, amplifier and loudspeakers in a compact enclosure. Especially designed for use with multi-image presentations or PA sound reinforcement, the mini-COLUMN can be used with audiences in excess of 500 people. The unit is only 19 inches tall, and weighs just 17 pounds. A built-in recessed tripod adapter on the bottom makes the system easy to set-up.



Music power rated at 50 watts, the system can produce a claimed 107 dB SPL at 1 watt at 1 meter. Features include low-impedance mike input, aux input, tone control, and external speaker output. Available as a standard AC-only unit, (model C-50P), or AC-DC operation with built-in rechargeable battery (model C-50PB), suggested list price for model C-50P is $389.

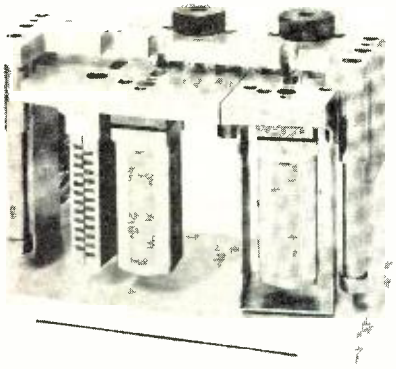
VJ ELECTRONICS, INC.
913 WEST 223RD STREET
TORRANCE, CA 90502
(213) 533-5980

For additional information circle #74

NEW PRECISION ALIGNMENT ASSEMBLY INTRODUCED BY JRF

The new Promix II alignment assembly is designed to adjust azimuth, zenith, wrap and track placement (height). High-frequency and peak adjustments are said to be made smooth, simple and repeatable.

The Promix II is specifically designed to reduce tape machine alignment time, as well as simplify magnetic head maintenance. The complete package includes a special new assembly cover with hinged top for easy access. For a limited time, JRF is offering free relapping and mounting with purchase of the Promix II.



Currently, the Promix II is available to fit most MCI JH-Series multitracks.

JRF COMPANY
101 LANDING ROAD
LANDING, NJ 07850
(201) 398-7426

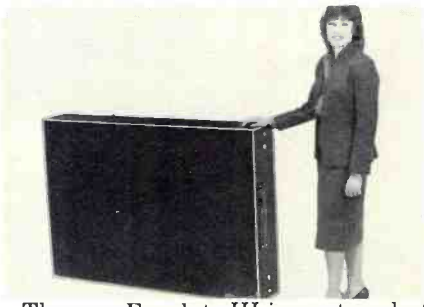
For additional information circle #75

STUDIO TECHNOLOGIES ECOPLATE III

Aimed at the cost-conscious studio owner, but retaining most of the features of the Ecoplate and Ecoplate II, reverb time is variable from 0.5 to 5 seconds, with a quoted signal-to-noise ratio of 65 dB, and a reverb frequency response of 80 Hz to 20 kHz.

New Products

Input is -10 or +4 dBm, 10k ohms unbalanced, standard phone jack, with stereo outputs +4 dBm (+24 dBm maximum), 50 ohms unbalanced, standard phone. In addition, both high and low frequency EQ is offered.



The new Ecoplate III is pre-tuned at the factory, and features a new shock resistant plate suspension system that is said to assure smooth, bright decay at all times, and eliminates tuning problems.

Quoted retail price of the Ecoplate III is $1,695.

STUDIO TECHNOLOGIES, INC.
6666 NORTH LINCOLN AVE.
LINCOLNWOOD, IL 60645
(312) 676-9400

For additional information circle #79

NEW SERIES 200 MIXING CONSOLES FROM SOUNDCRAFT

The Series 200 is available in three frame sizes: 8, 16 and 24 inputs, with the 8-input model being 19-inch rack mountable. Equipped with four group and two stereo outputs, the new Series features balance mike and line inputs, four-band equalizer, and four auxillary sends — two post- and two pre-fader. 48-volt phantom power, 1 kHz slate oscillator, input clipping indicators and 2-track replay are all standard features. Input, group output, remix and auxillary sends, are easily monitored via PFL switching in headphone outputs and metering.

The 200 series suggested retail price is $1,995, $2,350 and $4,500 for 8-, 16- and 24-input frame sizes.

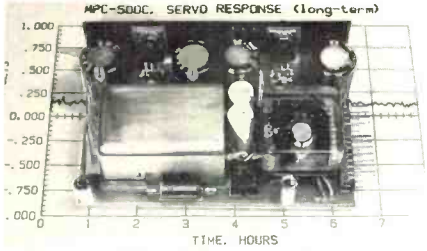
SOUNDCRAFT, INC.
1517 20TH STREET
SANTA MONICA, CA 90404
(213) 453-4591

For additional information circle #80

HARDY MPC-500C MIKE PRE-AMP CARD

Designed to directly replace the stock pre-amp card in MCI Series 500C consoles, the new MPC-500C has several features that provide improved performance compared to the stock card:

- 990 discrete op-amp which, according to Hardy, offers faster slew rate, higher output current, and lower distortion than the stock op-amp.



- Jensen JE 16-B microphone input transformer, providing one-third the distortion, the ability to handle signal levels 10 to 15 dB higher before saturation, flatter response, less overshoot, and more linear input impedance than the stock transformer.
- On-card power supply regulation, providing reduced crosstalk, increased stability, and eliminating the need for the "swinging transistors" used on the stock MCI card, which would compromise the sound and stability of the 990.
- Elimination of all coupling and gain-pot capacitors through the use of a special servo circuit, said to provide a significant improvement in sound quality.

The MPC-500C is priced at $195 in single quantities.

THE JOHN HARDY COMPANY
P.O. BOX AA631
EVANSTON, IL 60204
(312) 864-8060

For additional information circle #83

KLIPSCH INTRODUCES LONG-THROW HF HORN

The new MMTM high-frequency, long-throw horn is said to provide highly controlled dispersion, making it ideal for applications requiring high sound pressure levels at long distances, and those applications requiring precise control of the coverge patterns. Nominal beamwidth is 45 degrees horizontal and 20 degrees vertical when used as a single unit. The beamwidth changes to 45 degrees horizontal and 12 degrees vertical when used as a stacked pair.

Usable frequency range of the MMTM is 6.0 to 16.0 kHz, ±5 dB, and on-axis sensitivity 108 dB SPL with one watt input (124 dB SPL with 40 watts input) as measured in far field and referenced

New Products

to one meter.

The crossover employs a passive 6 kHz highpass filter, with 18 dB per octave slope. Four Klipsch high frequency drivers are used in each MMTM: there are two horns, each having two drivers.

Suggested retail price of the new MMTM is $408.

KLIPSCH & ASSOCIATES
P.O. BOX 688
HOPE, AK 71801
(501) 777-6751

For additional information circle #86

SCAMP S27 PROCESSOR MODULE FROM AUDIO + DESIGN

Developed from technology used in the Transdynamic Processor, the new module splits the audio signal into four bands with crossover points at 220 Hz, 1.6 kHz, and 4.5 kHz (these points being chosen as optimum after much testing and research by ADR's R&D department). The four signals are then routed out for processing or, if used as a crossover, to drive amplifiers.

If the band-split signals are processed by external processors (for example, F300 Expander-Gates for noise reduction without breathing effects, or S01 compressor-limiter for bandsplit compression/limiting), they are then returned to the S27 and re-combined.

According to ADR, the S27 makes use of 12 dB/octave, fully phase compensated filters; there is *no* phase shift.

Applications include sound reinforcement, for driving amplifiers; as a selective limiting system in disk mastering and tape duplicating; and as an overall production aid in the studio, both as a multi-band compressor and as a single-ended noise reduction system. It can also be used as a four-band summing amplifier.

Quoted retail price of the Scamp S27 Crossover/Four-band Processor module is $340.

AUDIO + DESIGN RECORDING
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #87

ORBAN MODEL 536A TWO-CHANNEL DE-ESSER

The new unit features the same proven circuitry as the single-channel 526A; a lower price results from better packaging economy, and the elimination of the mike-level input found in the 526A.

The Model 536A provides constant de-essing with input levels that vary as much as 15 dB. De-essing is adjusted with a single Threshold control per channel, which makes the unit easy to set up and operate in time pressure situations. Dual LEDs provide accurate indication of de-essing action. A click-free in/out switch allows de-essing to be introduced at any time during the program without audible side effects. Active balanced inputs and outputs are standard, with a transformer output option.

The new Model 536A will sell for $539.

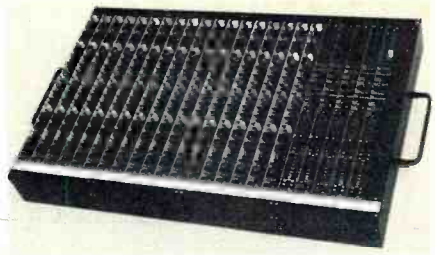
ORBAN ASSOCIATES, INC.
645 BRYANT ST.
SAN FRANCISCO, CA 94107
(415) 957-1067

For additional information circle #88

INTERFACE ANNOUNCES MODEL 310 STAGE MONITOR MIXER

The new Series 310, which replaces the Model 104L, is of modular construction for ease of maintenance, and is built in frame sections holding six modules that can be readily expandable from 12 to 42 inputs. The mixer provides eight output mixes, plus a side-fill pair with send and panpots.

As with all stage monitor mixers, each input can send independently to each output, and each send on the 310



also has an in/out button and a pre/-post equalizer button. New features include transformerless input, four equalizers (two tuneable, and with wide/narrow switch), high- and low-frequency cutoffs, five LED level indicators on each input for instantaneous identification of changes, solo to operator's monitor, master solo to check outputs, and return solo to permit listening to signal after external processors, plus 10-segment LED level indicators on masters and operator's monitor.

An 18-in/10-out Series 310 mixer will list for about $7,000.

INTERFACE ELECTRONICS
6710 ALDER
HOUSTON, TX 77081
(713) 660-0100

For additional information circle #89

MXR ANNOUNCES TWO NEW DIGITAL TIME DELAYS

The new Model 175-1 features adjustable delay times of up to 1 second in six ranges. An optional Repeat-Hold capability allows the user to capture entire musical phrases and replay them indef-

initely, eliminating troublesome tape loops. Also available: the Model 175, which has a delay range of 0.31 to 328 milliseconds.

Both units share a number of common features, including a sweep frequency range of 0.1 to 10 Hz, variable over a 4:1 range; stereo outputs; switchable high-Z line- and instrument-level inputs; built-in by-pass capability; Regen, Mix and Invert delay controls; and green/red signal present/overload indication. Both models are standard rack mount width, and one rack unit high.

Model 175-1, with a 1-second delay range, has a suggested retail price of $650; Model 175, $500.

MXR INNOVATIONS, INC.
740 DRIVING PARK AVENUE
ROCHESTER, NY 14613
(716) 254-2910

For additional information circle #92

MODEL 174 PITCH-SHIFT DOUBLER FROM MXR

One rack space high, the new Model 174 allows continuous pitch shift over a

±3% (¼-step) range. The Pitch-Shift Doubler is described as not just another chorus/doubler device; it uses a small, varying delay time (0.6 to 5 milliseconds) with a constant pitch change to produce the effect.

The new unit detunes the signal to create a true unison sound. The delay component of the signal processing optimizes this effect in the upper-end harmonics, making the unit ideal for vocals, MXR claims. The overall effect is virtually identical to unison strings on a 12-string guitar, detuning oscillators, or two vocalists singing a tight unison line.

Other features include stereo outputs, green/red LED signal present/overload indicator, Mix and Regen control, buffered high-Z inputs and outputs.

Suggested retail price of $450.

MXR INNOVATIONS, INC.
740 DRIVING PARK AVENUE
ROCHESTER, NY 14613
(716) 254-2910

For additional information circle #93

MELKUIST MASTERMIX LOW-COST, DISK-BASED AUTOMATION SYSTEM

At an expected price of $13,000, Mastermix offers 64K of on-board random-access memory, and utilizes mini 5¼-inch floppy disks as the storage medium for concole fader levels and channel mutes.

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used in mixing consoles: not only the processor-based grouping systems of Harrison (Series 4) and MCI, but also the Melkuist, Valley People Fadex, and Harrison Series 2 and 3 consoles.

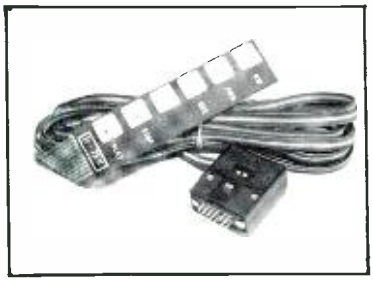
For under $20,000 Melkuist can supply VCA fader packages with Mastermix automation to fit all consoles having a separate fader section (for example, Neve and Trident).

STUDIO MAINTENANCE, INC.
12458 MAGNOLIA BLVD.
NORTH HOLLYWOOD, CA 91607
(213) 877-3311

For additional information circle #94

NEW RS-1 REMOTE CONTROL STRIP

for Tascam, Otari & Fostex Machines



20 foot ribbon cable interface connector allows record, play, fast forward, rewind, stop, and pause functions *without* the need for a bulky remote box.

The RS-1 Remote Control Strip, which measures 5⅛ by 1⅛ inches, has an adhesive backing for mounting on any flat surface, such as a blank mixer module.

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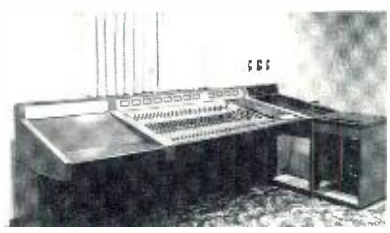

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STOLEN EQUIPMENT

• The following microphones have been stolen from Bullet Recording, Nashville, Tennessee: Neumann U87, #37843; Neumann KM84, #42173; Neumann KM84, #42174; two Shure SM57CUs; AKG C414EB, #7747; AKG C452EB, #50794; Neumann U47FET, #4147; Neumann KM84, #42175; AKG C414EB, #6303; AKG C414EB, #8657; Sennheiser MD421 and MD441, #019797; Neumann U87, #31650; Neumann U89, #588, and AKG C414EB, #8508.

Any information should be relayed to Piers Plaskitt at (615) 327-4621.

• On March 17, 1983, Artisan Sound Recorders, Hollywood, California, was illegally entered and the following equipment stolen: an Ampex MM-1200 24-track, #4381188; Lexicon 224 digital reverb, #2210; five AKG C414EBs, #s 1706, 1178 1213, 8549, and 4271; two AKG C452s, #s 13577, and 2708; Neumann U47, #2558; Neumann U67, #3120; two Neumann U87s, #s 27195, and 27196; Neumann U89, #C437; Neumann SM69, #3499; Neumann SM69 power supply, #B251; two Sennheiser MD421s, #s 04268, and 07371; Sennheiser MD441, #005692; and a Shure SM7, #1256.

Call William Rogers at Kendun Recorders, (213) 843-5900.

HY JAMES RELOCATES TO NEW PREMISES

The company has moved from its Ann Arbor, Michigan, location to 24166 Haggerty Road, Farmington Hills, MI; the new telephone number is (313) 471-0027. Also, the company's Ann Arbor (994-0934) and WATS lines remain in force.

Recently Hy James was named a dealer for the Otari MTR-90 24-track recorder, only the 10th in the US. To make room to demonstrate this and other products from over 100 manufacturers represented by the company, three functional demo rooms and an IEC Listening Room are being included in the new 6,000 square-feet facility. A full service equipment maintainence station for in- and out-of-warranty repair also is being included.

SOUNDCRAFT WEST COAST SALES OFFICE MOVES

Soundcraft Electronics, based in London, England, has moved its US sales office to a new facility in Santa Monica, California. The move is promp-

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news

ted by dramatic growth in sales, necessitating a tripling of support staff.

The new company address is Soundcraft Electronics, 1517 20th Street, Santa Monica, CA 90404. (213) 453-4591.

JBL TO MARKET UREI PRODUCTS IN US

The company will begin marketing UREI brand name products in the United States, effective July 1. The URC Corporation, of which UREI is a division, recently was acquired by JBL parent, Harman International.

JBL and UREI will share a combined sales organization, including common representatives, regional managers, and national sales manager. No major changes are anticipated in the UREI dealer distribution, according to JBL VP of sales and marketing, Ron Means.

TANNOY ESTABLISHES NORTH AMERICAN DIVISION

The recently established division, to be headed up by sales and marketing director, David Bissett-Powell, and which will stock and service the full line of Tannoy loudspeakers and related products, will operate from the following address: Tannoy Crown Acoustic, Ltd., 97 Victoria Street North, Kitchener, Ontario, Canada N2H 5C1. (519) 745-1158.

AMS APPOINTS U.S. DISTRIBUTORS

Advanced Music Systems, of Burnley, England, has appointed the following company to handle US distribution and service of its range of digital effects and reverberation units: Harris Sound Systems, 7138 Santa Monica Boulevard, Hollywood, CA 90046. (213) 469-3500.

GRP GLENN MILLER RELEASE MASTERED ON 3M/JVC SYSTEMS

In The Digital Mood, by the Glenn Miller Orchestra, is one of the first digitally-mastered releases by GRP Records' co-presidents Dave Grusin and Larry Rosen. "The distinctive, crisp melodies of Glenn Miller never lose their freshness," Rosen says, "and now with the intervention of modern technology, they can be heard on record as never before."

The album was recorded by Rosen at A&R Studios, New York, using a 3M 32-track Digital Audio System, and then digitally mastered in GRP's Review Room with a JVC DAS Series 90 Digital Processor and Editing System.

The Miller release also will be available on Compact Disc, becoming one of the first CDs released in the US.

STUDIO FINANCES

— continued from page 30 . . .

$19,000 ($20,000 less half of the $2,000 investment tax credit). Alternatively, an election can be made not to reduce basis, but to reduce the investment credit instead.

ACRS Depreciation

Under the accelerated cost recovery system a taxpayer may elect to expense up to $5,000 in lieu of capitalizing. Per ERTA, up to $5,000 of capital acquisitions may be expensed in 1983. This increases to $7,500 a year in 1984 and 1985, with $10,000 available in subsequent years.

The ACRS concept of faster depreciation for all fixed assets based on shorter (IRS audit proof) lives with no salvage value was introduced as a part of ERTA. A 15-year life for buildings, and a five-year life for virtually all non-automotive equipment, was prescribed by IRS regulations. The percentage to be taken as depreciation in early years was scheduled to become more favorable in 1985, and 1986. However, the 1982 Act eliminated future increases in depreciation rates.

Virtually all studio equipment qualifies for five-year life treatment. An amount equal to 15% of the item's basis is eligible for depreciation during year #1, and 22% may be depreciated during year #2; 21% of the basis may be deducted during years 3 through 5.

The example given in Table 3 illustrates the tax benefits received from the purchase of a $10,000 item of studio equipment.

"Safe Harbor" Leasing

TEFRA also reduced the scope of "safe harbor" leasing provisions established by the '81 Act. Safe harbor leasing facilitates the transfer of ACRS tax benefits and investment tax credits from companies that do not need them, to companies that can benefit from them. As of December 31, 1983, safe harbor leasing laws will be repealed; until then the benefits originally provided by ERTA are greatly curtailed. While the benefits to the lessor are reduced, and the amortization of the financing of the lease is accelerated, safe harbor leasing is still a viable alternative for 1983 tax treatment.

Given the rapidly changing fortunes of the sound recording and reinforcement industries, whether for records, tape, broadcast, or video, any operator that wants to keep one jump ahead of the opposition (or simply stay in business, for that matter) has to remain abreast of current tax and accounting procedures. If there are any general areas of financial developments and techniques that *R-e/p* readers would like me to cover in greater detail, please write to this author care of the magazine. I will attempt to devote subsequent articles to answering some of your questions. ■■■

Studer Re-States the Art

With the new A810, Studer makes a quantum leap forward in audio recorder technology. Quite simply, it re-states the art of analog audio recording.

By combining traditional Swiss craftsmanship with the latest microprocessor control systems, Studer has engineered an ATR with unprecedented capabilities. All transport functions are totally microprocessor controlled, and all *four* tape speeds (3.75 to 30 ips) are front-panel selectable. The digital readout gives real time indication (+ or − in hrs, min, and sec) at all speeds, including vari-speed. A zero locate and one autolocate position are always at hand.

That's only the beginning. The A810 also provides three "soft keys" which may be user programmed for a variety of operating features. It's your choice. Three more locate positions. Start locate. Pause. Fader start. Tape dump. Remote ready. Time code enable. You can program your A810 for one specialized application, then re-program it later for another use.

There's more. Electronic alignment of audio parameters (bias, level, EQ) is accomplished via digital pad networks. (Trimpots have been eliminated.) After programming alignments into the A810's memory, you simply push a button to re-align when switching tape formulations.

The A810 also introduces a new generation of audio electronics, with your choice of either transformerless or transformer-balanced in/out cards. Both offer advanced phase compensation circuits for unprecedented phase linearity. The new transport control servo system responds quickly, runs cool, and offers four spooling speeds.

Everything so far is standard. As an option, the A810 offers time-coincident SMPTE code on a center track between stereo audio channels. Separate time code heads ensure audio/code crosstalk rejection of better than 90 dB, while an internal digital delay automatically compensates for the time offset at all speeds. Code and audio always come out together, just like on your 4-track. Except you only pay for ¼" tape.

If you'd like computer control of all these functions, simply order the optional serial interface. It's compatible with RS232, RS422, and RS422-modified busses.

More features, standard and optional, are available. We suggest you contact your Studer representative for details. Granted, we've packed a lot into one small package, but ultimately you'll find that the Studer A810 is the most versatile, most practical, most *useable* ATR you can buy.

The Swiss wouldn't have it any other way.

STUDER REVOX

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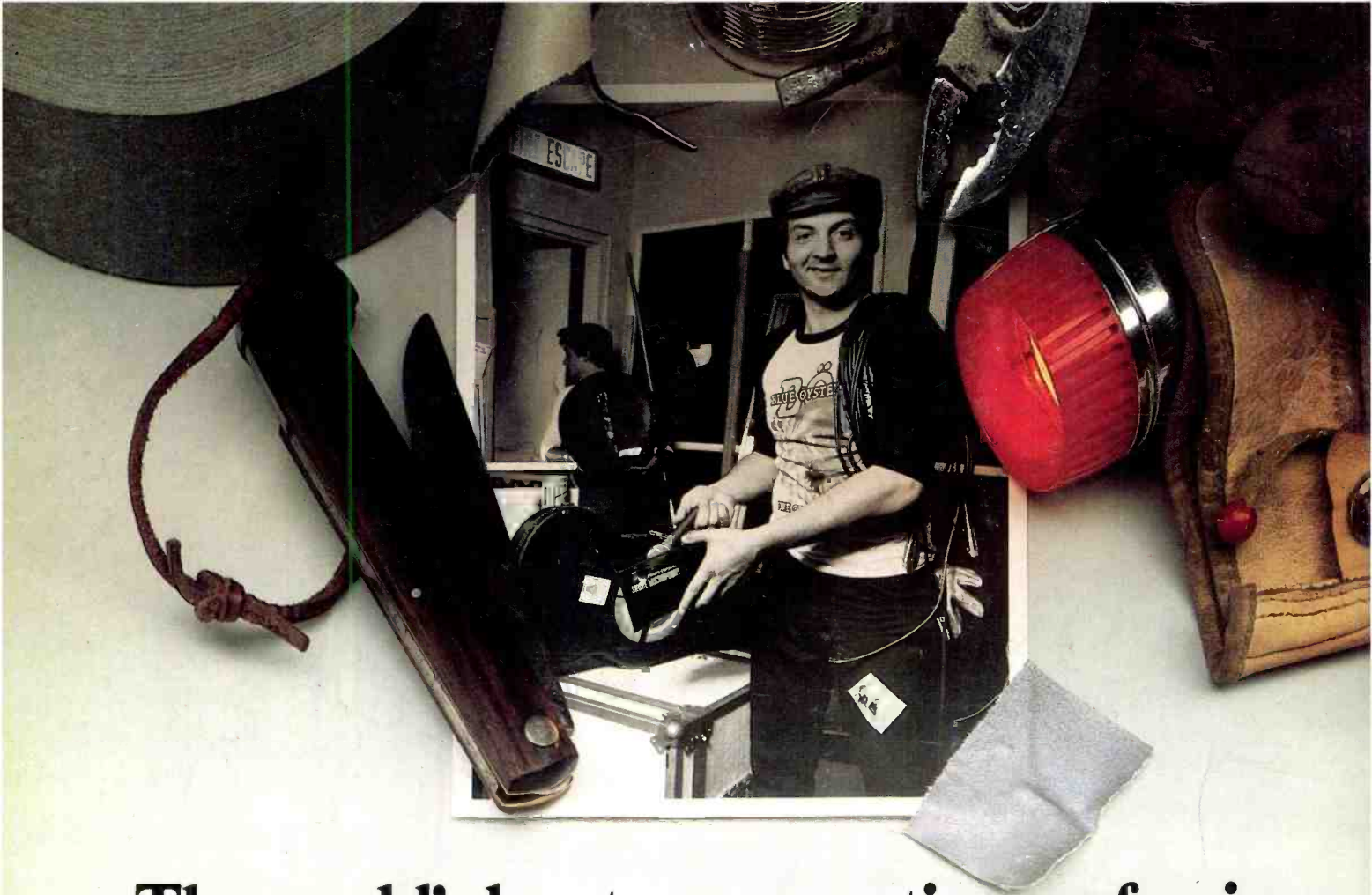


The world's least conservative profession has maintained one rigid tradition. The SM58.

In an industry that discards electronic products like ice cream wrappers, the SM58 and its close cousin, the SM57, have remained the overwhelming choice of rock, pop, R & B, gospel and jazz vocalists for the last 16 years.

Why?

Simply because there is no sound quite like the SM58 sound. Its punch in live vocal situations, coupled with a distinctive upper midrange presence peak and fixed low-frequency rolloff, give it the trademark quality no other manufacturer can imitate, although others have tried.

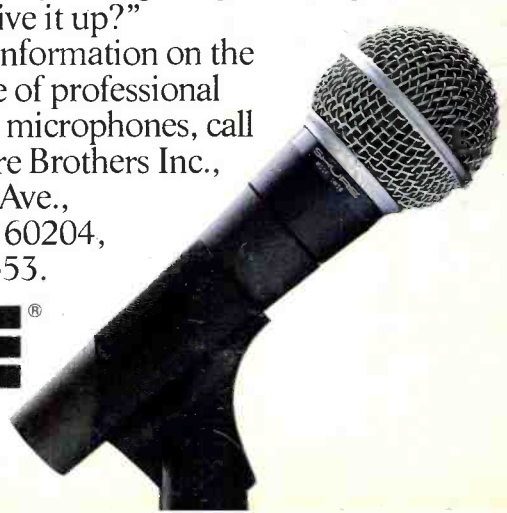
And to protect that sonic perfection, the SM58 is extraordinarily tough. Even six-foot drops on hardwood floors won't faze it. Ask any roadie who has used–and abused–one.

Performers the world over favor the weight and balance of the SM58, especially in hand-held situations. Even the finish is totally professional–a non-glare grey that looks as great on stage as it does on camera.

The crispness of the closely related SM57 enhances musical instruments the way the SM58 handles vocals. Beautifully.

Musicians are tough to please, but with the world-standard SM58 and SM57, they'll tell you, "when you've got a good thing going, why give it up?"

For more information on the complete line of professional performance microphones, call or write Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, (312) 866-2553.



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