Interview with Frank Zappa

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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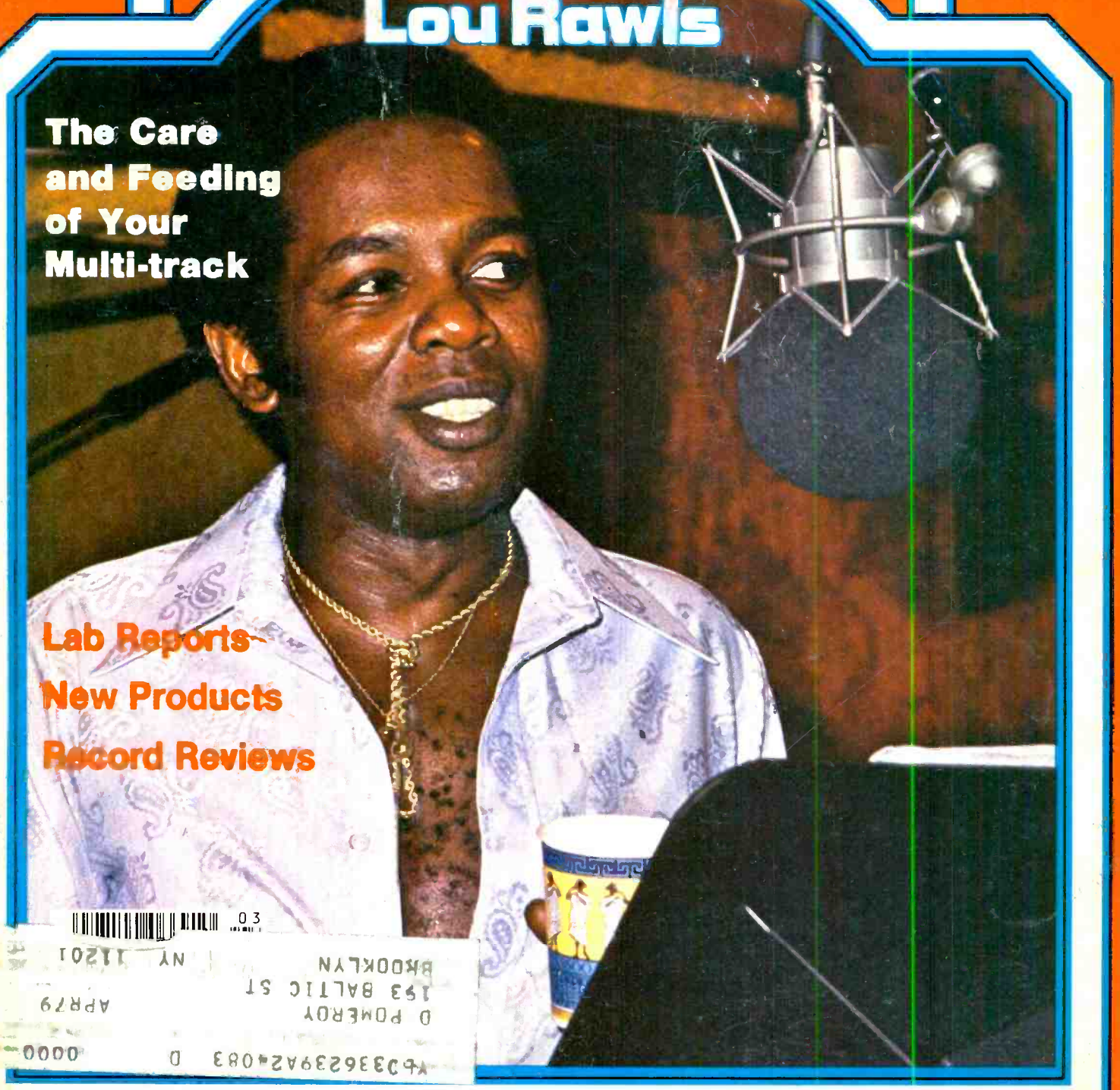
VOL. 3 NO. 6
MARCH 1978

A Session with Lou Rawls



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The MXR Digital Delay gives the audio engineer a tool for creative application that is unparalleled in versatility, precision and ease of operation. The MXR Digital Delay is designed for a wide variety of applications including; amplified musical instruments, vocals, PA and recording mixes. The basic unit delays a sound between 0.08 milliseconds and 320 milliseconds, fully variable while retaining the dynamic range of the program source. The delay range is expandable to 1280 milliseconds in increments of 320 milliseconds by means of up to three additional plug-in memory boards. These boards are available from MXR and may be installed by the user.

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MXR's Digital Delay can lead the way to new possibilities in creative sound at a price considerably lower than any comparable delay.

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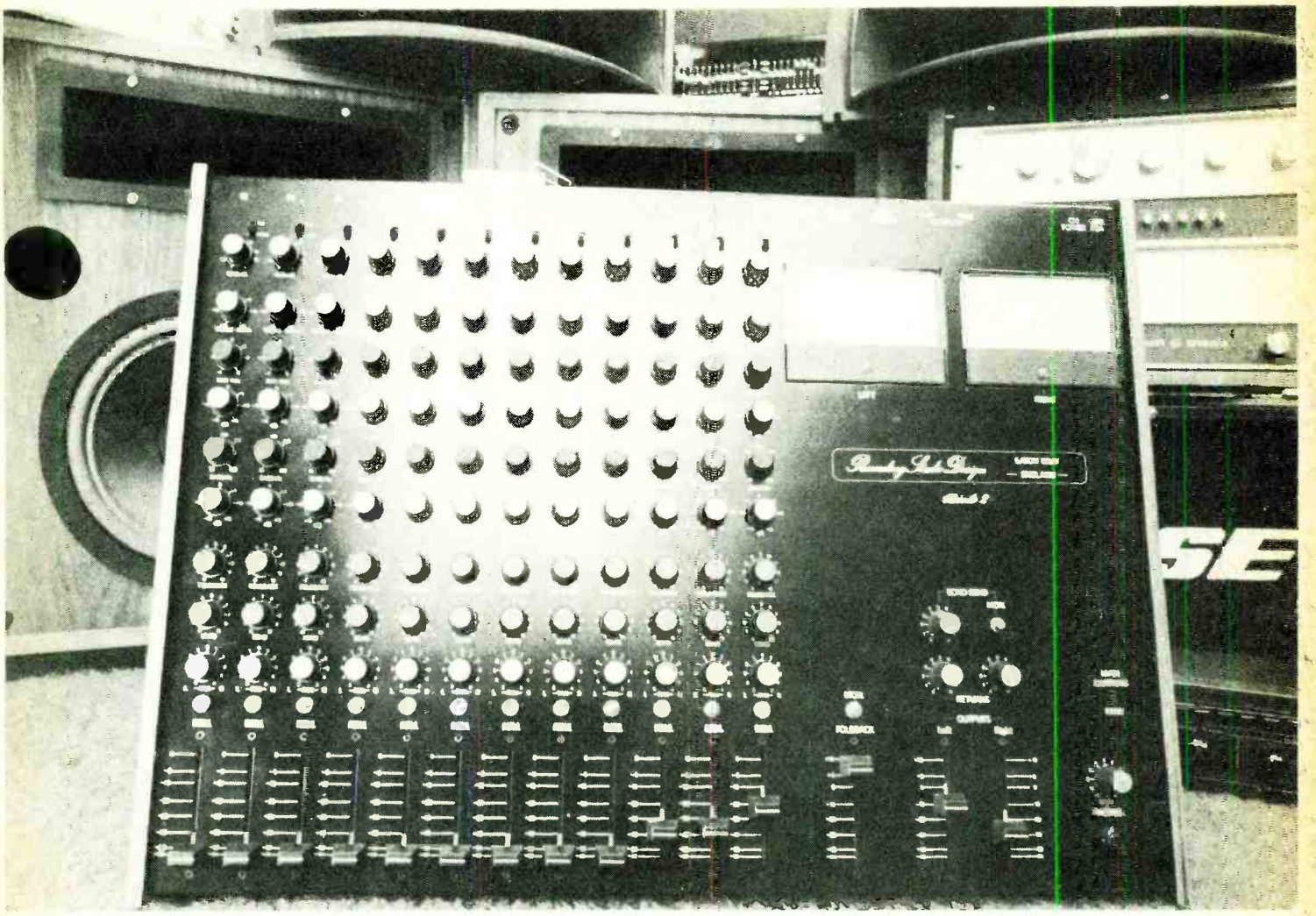
CIRCLE 97 ON READER SERVICE CARD



the pros stick together...

and that includes the **Recording Studio Design** 12 x 2 foldback mixing console. Fresh on the American market after two years of trendsetting success in Europe, **RSD** has specifications which easily meet and surpass its competition, as well as two very important, unique extras. Incorporated into the panel is 3 band equalization with continuously variable mid and bass controls allowing a parametric equalization effect not found on other n… the **RSD** is equipped with a multicore connector coupling allowing the hook… with stack-able winding spindle) to its 12 channel stage box . . . a truly p… ailable only through the highly selected dealers you see listed below.

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CIRCLE 85 ON READER…

You're Invited to Participate in the Marriage of Music and

You can pay for someone else's studio or you can invest in your own.

Our new Studio 8000 gives you that choice. And you won't have to sell your soul to get it.*

The eight tracks give you room to spread out your music, and your own studio gives you the option of turning on the equipment whenever you turn on to a good idea... 24 hours a day.

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And for absolute quality stereo mastering, plug in the 25-2. DBX is built in, and so are speed, simplicity and accurate final editing capability.

So why go on paying for time in a studio that someone else owns? Especially when the total dollars involved wouldn't buy much more than a new car at today's prices.

See the Studio 8000 at your authorized TASCAM dealer.

TASCAM SERIES BY TEAC®

A new generation of recording instruments for a new generation of recording artists.

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MARCH 1978
VOL. 3 NO. 6

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!

A Session With The Tierra del Fuego All-Girl Choir
Miking The Concorde "Live!"
Rapid Recording Techniques, or How To Record Running Water in the Arctic

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THE STAPLES

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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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LETTERS TO THE EDITOR

Inexpensive Sources

This letter is in response to Mr. Don Chin's Letter To The Editor in the December 1977 issue (see "The Number's No Object, page 4).

There are many less expensive places than Gotham Audio to buy high quality electronic parts in small or large quantities. I would like to share with you and your readers some of the places with which I have had excellent success.

For resistors, I buy from Digi-Key. The price in quantities of 100 per value comes down to 1.7 cents for 5% deposited carbon (not carbon rod) resistors. They also have many other items such as ICs, capacitors, switches relays, diodes, transistors, etc., etc. Their address is Digi-Key Corporation, Box 677, Thief River Falls, Mn. 56701, telephone number 218-681-6674.

A good source for inexpensive capacitors, ICs, IC sockets and trimpots is James Electronics, 1021 Howard Ave., San Carlos, Ca. 94070, telephone number 415-592-8097.

I have used a Triad transformer for our microphone transformers with very good results. The model number is T-1X. It is essentially flat in response from 50-16,000 cps, it is cheap (less than ten dollars) and has a choice of three impedances (50, 250, and 600). Their address is Triad Utrad, 305 North Briant St., Huntington, In. 46750, telephone number 219-358-7100.

We have a 16-track recording studio here in Nashville and have undertaken many electronics projects for the studio, so I know how difficult it is to find reliable electronics parts sources, and I hope I have helped someone out there with this information.

—Tom Behrens
Chief Engineer
Music Square Productions
Nashville, Tn.

A Thank-You For Objective Journalism

Thank you for publishing the very useful "Hands-On Report" on the Soundcraft Series II mixing console (see Hands-On, December 1977, page 76).

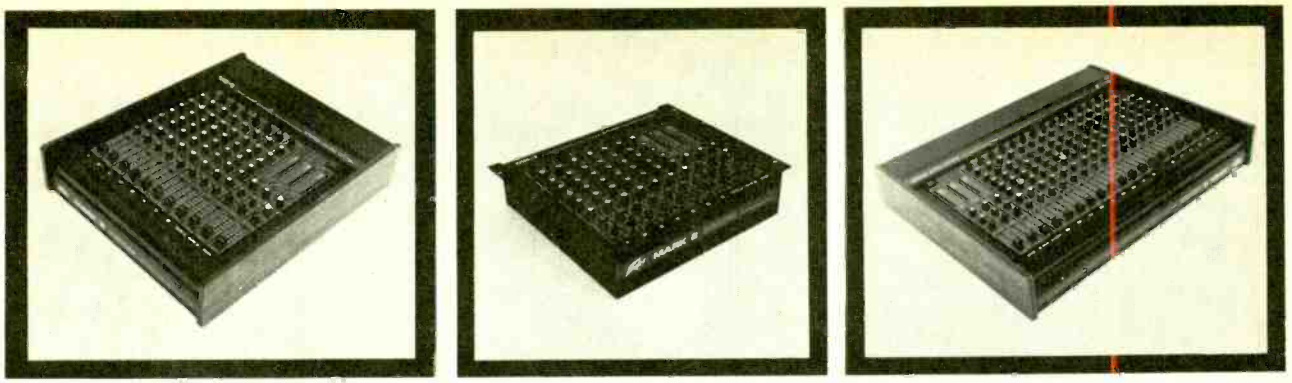
We appreciate the objective criticism of Brian Roth and Jim Ford which concurs with our own retrospective evaluation of Series II. Future development of Series II will eliminate these minor shortcomings. We will be taking a closer look at the problem of R.F. although I am glad to say that, having sold several hundred Series II consoles, we have only received very occasional complaints about this.

The general description at the beginning of the article left out one EQ function, (Section F4): Treble shelving (boost or cut) control.

The unit in question was a typical production unit and had received no closer attention by us before it was delivered than any other production unit receives. We would like to point out that Series II is available in the following formats: 12, 16, 24 or 32 input, four or eight group.

We are glad to see this sort of objective journalism and we hope that the report will assist your readership in making an equally objective choice of console—hopefully ours!

—Philip Dudderidge
Managing Director
Soundcraft Electronics Limited
London, England



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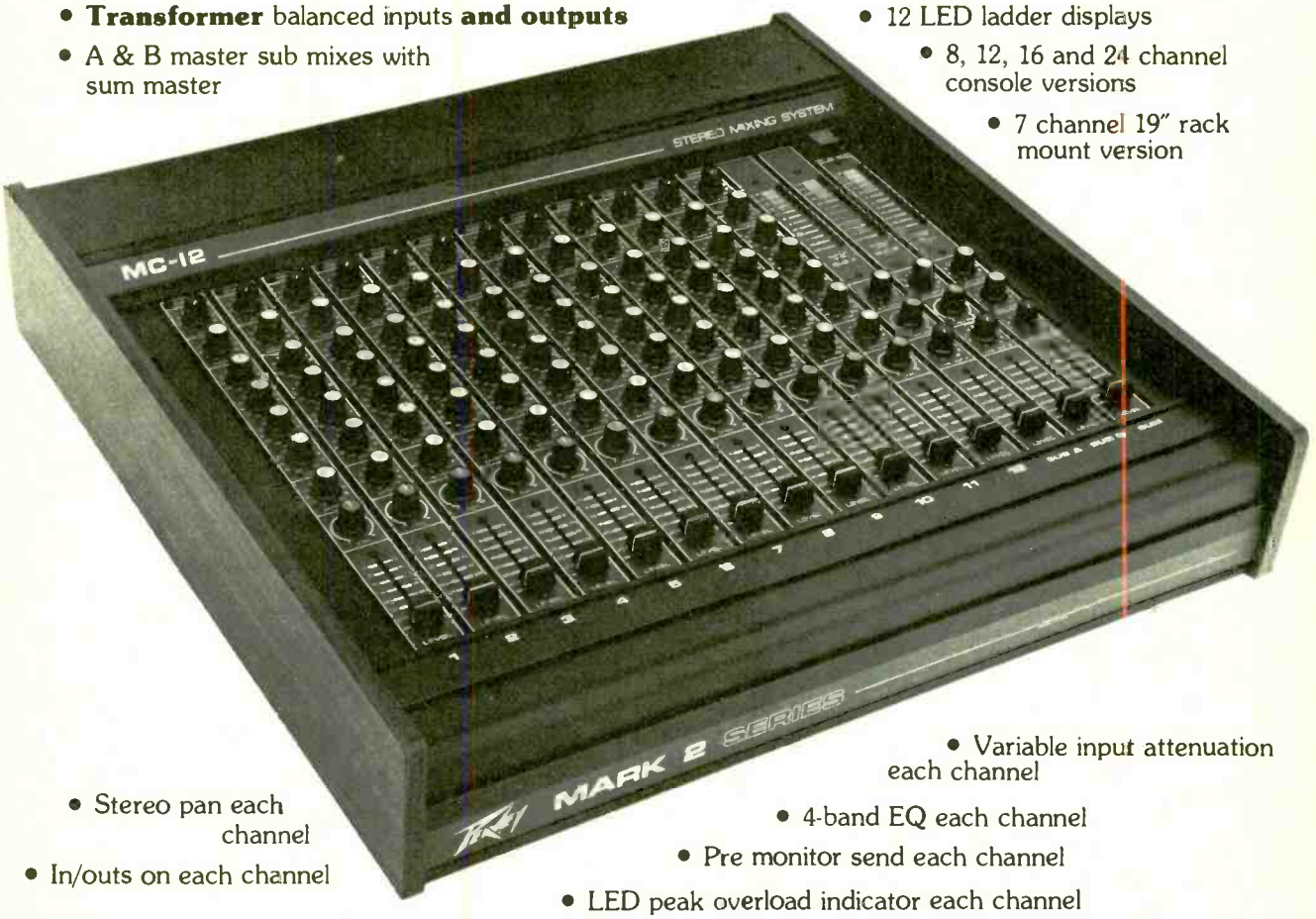
- **Transformer** balanced inputs **and outputs**
- A & B master sub mixes with sum master
- 12 LED ladder displays
- 8, 12, 16 and 24 channel console versions
- 7 channel 19" rack mount version



- Variable input attenuation each channel
- Stereo pan each channel
- 4-band EQ each channel
- In/outs on each channel
- Pre monitor send each channel
- LED peak overload indicator each channel
- +20 dBM levels at all outputs

Years of research, testing and "on the road" experience have gone into the development of an entirely new line of professional state-of-the-art mixers from Peavey...The Mark 2 Series.

This is just an indication of the many features we've designed into this new series. It would be impossible to fully describe or make you really appreciate the operational flexibility of the Mark 2 Series in this advertisement. We won't even try. Write us and we will send you complete information about the most dynamic sound value on today's market...The Mark 2 Series Mixers.

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Scully Source

Could you please send me some information on how I might obtain some information on Scully recorders (i.e. prices, brochures, etc.). Also, I'd like to take this opportunity to congratulate you on your fine magazine.

—Johnny Petro, Jr.
Jackson, Ms.

For the information that you require, write directly to Scully/Metrotech, 475 Ellis Street, Mountain View, California 94043 or you can call them at 415-968-8389.

A Pressing Problem

I am an aspiring record producer in the Pacific Northwest and I recently did an album for a Seattle-based band. I now find myself faced with an interesting problem.

Upon completion, the master tape was taken to the nearest mastering lab (which was in Vancouver, B.C.) which turned out what I felt was an excellent master lacquer. The next day it was sent to Los Angeles to the pressing plant.

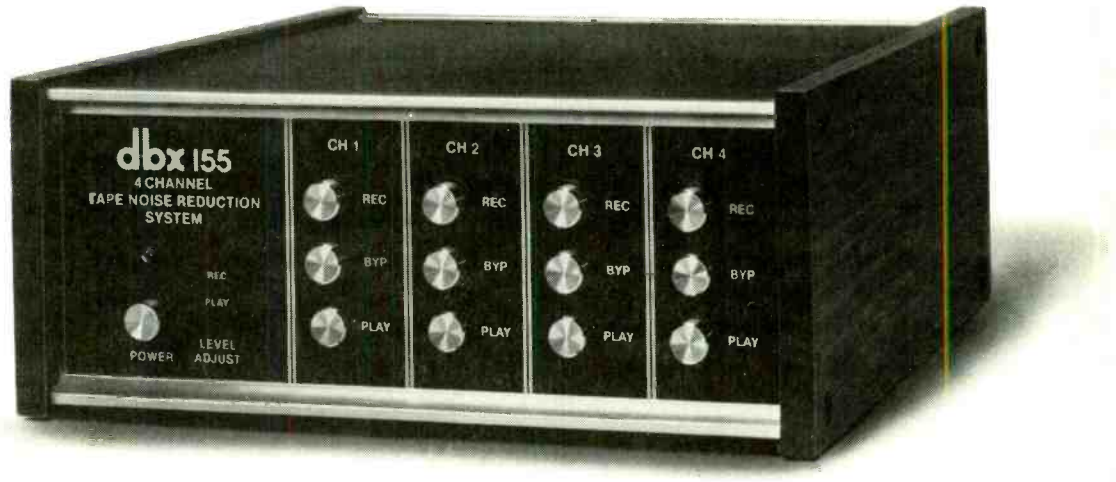
I just received the test pressing and the high end is *gone* (I would guess anything above 8 kHz). I talked to the engineer who edited the master tape and he told me that the "mother" should be made within 24 hours after cutting the master lacquer and that high end loss is the result if this is not done. He also said that he assumed the lab in Vancouver (also a pressing plant) was going to make the "mother" right there. I suppose now that he also assumed that I was flying the next day to the pressing plant in California that was contracted to do the pressings!

At any rate, it was about four days before the master lacquer arrived in Los Angeles and hence an inferior product. Now, because of the time delay and the added expense, I fear that the whole project may be scratched. I would greatly appreciate more information about these unforseen occurrences so that I might better protect myself and my recordings in the future.

—Paul Speer
Seattle, Wa.

Knowledge is power—or in your case, at least a great help. Before you make any further arrangements, read the article by David Moyssiadis (one of the most respected mastering engineers in this country) in the December 1977 issue of MR *entitled "A Producer's/Consumer's Guide To Better Record*

HUSH UP YOUR 4-TRACK.



Introducing dbx professional four-track tape noise reduction for under $500.

You've just settled on a TEAC, Tascam, Otari or Dokorder four-track tape deck for that studio you always wanted to have. You've chosen the mikes, the carpenter is almost finished (or maybe you even built it yourself). Your console's ready to be wired into place.

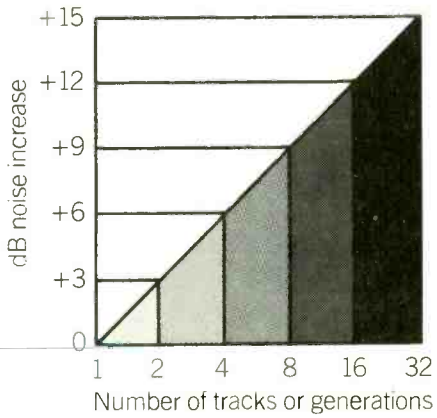
But...haven't you forgotten just one important item?

Noise reduction.

Because every time you (or your group) want to bounce a track, you're adding at least three dB of tape noise (see chart). So the great artistic result you plan to end up with, might end up sounding like a rainstorm.

Fear not. Help is at hand. It's the new dbx 155 four-channel tape noise reduction system. You can add it for far less money than you ever imagined possible. Here, on one compact chassis, is a complete dbx noise reduction system. But the best part is, it will give your tape deck an extra 10 dB of headroom, and reduce tape noise by 30 dB. That means no audible noise whatsoever will be added to your tracks. And, because dbx tape noise reduction operates by linear compression/expansion, you won't have to get involved with tedious level calibration, either.

Additive Noise Chart



All you need do is press the playback buttons to hear noise-free, full dynamic range reproduction of your music.

The new dbx 155 also has user-changeable modular circuit boards, so in the unlikely event that one processor fails, the other channels remain operational. You can even keep a spare on hand.

Visit your dbx professional dealer now, for a demonstration of our new 155 tape noise reduction system. Discover how you can put an end to tape hiss, without putting an end to your bankroll.

dbx

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CIRCLE 53 ON READER SERVICE CARD

Pressings" (page 56). When told of your problem Dave had some interesting information for us to pass on to you. While it is nice to make the mother within twenty-four hours after cutting the lacquer, it is by no means necessary to do so. As for your loss of the high end, Dave told us that the plating plant must have polished the stamper to make it quieter. Dave covered these points and much more about this entire process in his article. While an incident of this sort is unfortunate, look on it as a learning experience—you'll be that much better informed next time around.

Saving Those Old Reels

We're a six-piece band currently on the road in the mid west, who, utilizing our A-3340S TEAC and an A77 ReVox to tape our performances have acquired many overused reels of tape.

Recently, while playing an engagement in Canada, we met another TEAC owner who told us of high-quality Ampex tape available on styrofoam "pancakes" at a low cost.

Could you tell us if this tape is available in the U.S. and help us locate outlets for it? By utilizing old metal reels, the savings to us would be substantial.

—Jonesin'
Des Moines, Ia.

Get ready to load up those empty reels! George Armes of the magnetic tape division of Ampex Corporation of California informed us that tape "pancakes" are available at any Ampex sales office nationwide. After learning of your home base in Iowa, he advised us to send you to the Ampex outlet at 2201 Lunt Ave., Elk Grove Village, Illinois 60007. Direct your request to Bob Pellino and you should soon be on your way to more thrifty recording.

Plaudits For Reader Service

By chance today I came upon a copy of *Modern Recording* in my corner stationery store. Never having seen the magazine before and having an amateur interest in recording, I promptly purchased it. Without a doubt this has to be the most informative, interesting recording magazine that I've come across.

As a "layman," as it were, I found a few places in the articles beyond my comprehension due to the technical jargon, but I'm sure continued exposure to *MR* will remedy that! I have been interested in home recording for about two years and have an Akai 4000 DS. I have been shopping around for ideas for new open-reel machines as well as other pieces of equipment I might purchase and with your free Reader Service information plan, the decision should be made much easier.

I'm sure your publication is well-received by both amateur and seasoned recordists. May I extend my congratulations on a fine magazine.

—Bill Nagle
Surrey, British Columbia, Canada

The Magnificent Microphone

When are we going to see an article on microphones? I'm convinced they are the most important item in a studio's inventory!

—Ed Perrone
Gloversville, N.Y.

You must have missed our May 1977 issue for it contained quite a bit of information on microphones. There was

fact:
you can choose your microphone to enhance your individuality.

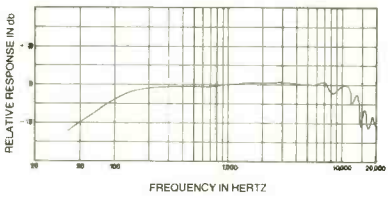
Shure makes microphones for every imaginable use.
Like musical instruments, each different type of Shure microphone has a distinctive "sound," or physical characteristic that optimizes it for particular applications, voices, or effects.
Take, for example, the Shure SM58 and SM59 microphones:

SM59

Mellow, smooth, silent...

The SM59 is a relatively new, dynamic cardioid microphone. Yet it is already widely accepted as a standard for distinguished studio productions. In fact, you'll often see it on TV . . . especially on musical shows where perfection of sound quality is a major consideration. This revolutionary cardioid microphone has an exceptionally flat frequency response and neutral sound that reproduces exactly what it hears. It's designed to give good bass response when miking at a distance. Remarkably rugged — it's built to shrug off rough handling. And, it is superb in rejecting mechanical stand noise such as floor and desk vibrations because of a unique, patented built-in shock mount. It also features a special hum-bucking coil for superior noise reduction!

Some like it essentially flat...

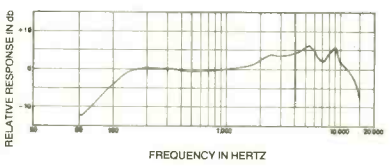


SM58

Crisp, bright "abuse proof"

Probably the most widely used on-stage, hand-held cardioid dynamic microphone. The SM58 dynamic microphone is preferred for its punch in live vocal applications . . . especially where close-up miking is important. It is THE world-standard professional stage microphone with the distinctive Shure upper mid-range presence peak for an intelligible, lively sound. World-renowned for its ability to withstand the kind of abuse that would destroy many other microphones. Designed to minimize the boominess you'd expect from close miking. Rugged, efficient spherical windscreen eliminates pops. Lightweight (15 ounces!) hand-sized. The first choice among rock, pop, R & B, country, gospel, and jazz vocalists.

...some like a "presence" peak.



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Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

CIRCLE 52 ON READER SERVICE CARD

John M. Woram's article, "John Woram Presents Mike Graphones" (page 31), as well as a Hands-On Report by Jim Ford and Brian Roth on common sound reinforcement microphones (page 62). If these two pieces don't answer all your questions, you might refer to the first Modern Recording Buyer's Guide *in which Larry Zide's article on microphones raised some interesting points (see page 44 of the guide for that).*

A "Live" Lady

I have just finished reading the October issue of *Modern Recording*. In reference to the Letter to The Editor entitled, "Areas of Interest," (page 6), I have been mixing sound for a Long Island (New York) based band for one year. Being a woman in the field has offered me many challenges. My background is, however, actually Speech Pathology and Audiology.

Your magazine has been most helpful to me because of the clarity of its format. I need some advice on reference materials for "live" sound reinforcement and resources geared for those without electrical engineering training. Thank you.

—Sonya Diamond
Long Beach, N.Y.

Certain principles remain important throughout the spectrum of sound reinforcement applications so some very basic readings regarding sound reproduction and acoustics, etc. are very important to someone like you. We're assuming you are a steady reader of MR *and are familiar with Jim Ford & Brian Roth's definitive three-part "PA Primer." This is an excellent resource piece for a person in your situation. If you missed the issues in which it appeared, you'll be happy to learn that it's reprinted, in its entirety, in the 1978 Buyer's Guide. You can also learn a lot about the basics from John M. Woram's* The Recording Studio Handbook, *all presented in simple, clear language.*

Other books that can provide some solid background for your include Modern Sound Reproduction *by Harry F. Olson, Van Nostrand Reinholt, Co., New York, N.Y.;* The Acoustical Foundations of Music *by John Backus, W.W. Norton and Co., New York, N.Y.;* The Manual of Sound Recording *by John Aldred, Fountain Press, London, England; and* Microphones: Design and Application *by Lou Burroughs, Sagamore Publishing Co., Plainview, N.Y.*

Also, don't neglect to keep up with the "live" sound reinforcement stories

Para-Power

(Parametric Equalizers by SAE)

SAE has long been involved in the field of tone equalization. From our pioneering efforts in variable turn over tone controls to our more recent advancements in graphic equalizers, we have continually searched for and developed more flexible and responsive tone networks. From these efforts comes a new powerful tool in tone equalization — the Parametric Equalizer. Now you have the power of precise control.

Our 2800 Dual Four-Band and 1800 Dual Two-Band Parametrics offer you controls that not only cut and boost, but also vary the bandwidth and tune the center frequency of any segment of the audio range.

With this unique flexibility, any problem can be overcome precisely, and any effect created precisely.

With either of these equalizers, you have the power to correct any listening environment or overcome any listening problems that you are faced with. Whether you need a third octave notch filter, tailored bandwidth to resurrect a vocalist, or a tailored cut to bury an overbearing bass, the control flexibility of Parametric Equalizers can fill these needs and many more. And of course, as with all SAE products, they offer the highest in sonic performance and quality of construction.

For Complete Information Write:

SAE

Scientific Audio Electronics, Inc.

P.O. Box 60271 Terminal Annex, Los Angeles, CA 90060

CIRCLE 88 ON READER SERVICE CARD

that are featured almost monthly in MR. *You can pick up quite a few pointers and "tricks of the trade" from them.*

A Spark Of Interest

Since I began reading your magazine, I have been taken with the idea of becoming a professional studio recording engineer. I would appreciate it if you could give me some information that can help me develop greater knowledge of this field. I would also appreciate it if you could give me the names of some books which deal with this field.

—Russell Valenti
South Ozone Park, N.Y.

We're happy to have sparked your interest in this really fascinating and fast-moving field. Some readings that will be helpful to you are The Recording Studio Handbook *by John Woram, Sagamore Publishing Company, Plainview, N.Y. 11803, 1976;* Modern Recording Techniques, *by Robert E. Runstein, Howard W. Sams & Co., Inc., Indianapolis, In. 96206, 1974;* The Audio Cyclopedia *by Howard M. Tremaine, Howard W. Sams & Co., Inc., Indianapolis, In. 96206.* Modern Recording Techniques *is the book used by the Recording Institute of America in their classes. You might even be interested in enrolling in one of their comprehensive courses. You can write the RIA at 15 Columbus Circle, New York, New York 10023.*

Partial Success

I've tried without success for some time to locate sources for the following parts which I'm sure many of your other readers would also like to find. Perhaps you will be able to help me find them: rubber/nylon recessed washers for rack mounts (to protect the surface of the equipment), large rubber feet (same as used on Anvil cases and Mesa/Boogie amps), and speaker mounting clamps (for fast replacement).

Thanks for any and all help.

—T. Young
N E Sound Services
Thomaston, Ct.

We spoke first with Wayne Thompson of Anvil Cases and found out that those "large rubber feet" are made exclusively for Anvil and are not available to the public. On your other two items, we had greater success. Dave Jet of Martin Audio (320 W. 46th St., New York City, N.Y. 10036) told us that he carries 200-300 lines of equipment and parts and that the pieces you need are located in his sixth floor small parts department and are available immediately. Call him at 212-549-5900 and he can fix you right up.

More On Marshall

I am interested in the Marshall Time Modulator ("first voltage controlled, time sweepable analogue delay line") reviewed by Norman Eisenberg in the April 1977 Product Scene (page 25). Can you send me more information—price, manufacturer's address, etc.?

Thank you in advance.

—Antonio Russek
Condesa, Mexico

You can write directly to Marshall Electronics, Box 177, Joppa, Maryland 21085 for information.

Introducing the Series 1600.
A new philosophy in console design.

Sound Workshop has created a high-performance, automation-ready audio mixing console available in several mainframe sizes, all fully expandable to a maximum configuration of 36 x 32. Ranging in price from $10,000 to over $50,000, the Series 1600 is the new alternative for intelligent studio planning.

The Series 1600 Audio Mixing Console represents a new philosophy of console design. A philosophy that directs itself not only to the performance and function of the console itself, but also to an intelligent studio design plan . . . one that fulfills the needs and wants of producers and engineers today, while considering the economic factors necessary to keep up with the state of the art tomorrow.

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Bringing the technology within everyone's reach.

TALK BACK

"Talkback" questions are answered by professional engineers, many of whose names you have probably seen listed on the credits of major pop albums. Their techniques are their own and might very well differ from another's. Thus, an answer in "Talkback" is certainly not necessarily the last word.

We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel that we are skirting any issues, fire a letter off to the editor right away. "Talkback" is the Modern Recording *reader's technical forum.*

Some Questions Answered

I have been a subscriber to your magazine for the past year and have had many questions answered by the straightforward articles you have. As in any good learning experience, the more I learn, the more questions I have.

First, I have seen many references to "fold back" and "slap back." What do these terms mean and how did they get these names?

My second question is about using reverb. I use a small mixer that has a reverb send for each channel but only one main reverb send and return. I've noticed that some higher-grade mixers have the same single reverb send/return. Does this situation cause problems, such as the piano reverb signal ending up with the drums and so on? In "live" sound reinforcement situations I can see where this might not matter, but in recording it would.

Is the single reverb send/return feature found on truly professional boards? If not, is each signal requiring reverb sent to a separate reverb unit? Also, does one type of reverb (plate, spring) work better for certain instruments or applications than others?

—Charles C. Ferrill
Pueblo, Colo.

Fold back is a term that seems to have come from England and means the same as "monitor" when speaking in studio and sound reinforcement circles. It simply means that with a monitor, or fold back system, you have the capability of sampling, in real time, your program material. Studio fold back will almost always be through headsets and not over the so-called studio monitors. These speakers are used for listening to taped playback and are not usually part of the fold back system.

Slap back, or "tape slap" as it is sometimes referred to, is a term that originated in the recording studio and is really a form of taped echo. It is achieved by using an additional tape machine other than the main recording unit and creating a loop so that a signal may be injected back into the track of its origin. With this process, echo may be added in real time as the original tracks are being laid down without the tremendous costs of digital delays, etc.

Most mixers that are designed for general purpose use (mostly sound reinforcement and some recording), fit the description you gave for your own personal unit.

For cost reasons, as well as packaging for portable mixer systems, it is necessary to offer, in most cases, only one internal reverb unit. However, where the studio is the primary target, the requirements are quite different as far as features are concerned. When absolute separate tracking is necessary in recording, those tracks requiring reverb should be run through separate chambers or springs, as the case may be, to prevent the delayed signals from mixing.

When two or more tracks with reverb are to eventually be mixed down together, then obviously there would be no need to record originally through separate reverb systems. There are studio consoles (very expensive) presently available with at least four separate systems for reverb; but as you stated in your letter, this would not be necessary for sound reinforcement.

Most reverb systems will work satisfactorily for many different instruments but some studios use very expensive chamber systems and digital delays for vocals.

—Hollis T. Calvert
Director,Sales Promotion/Education
Peavey Electronics Corp.
Meridian, Ms.

Keeping Your Heads Straight

What's a reasonably simple method of aligning the zenith on a tape deck (say, a TEAC 3340, for example)?

—Bob Bierstein
San Jose, Ca.

An old Ampex manual refers to the "standard grease pencil method" as the easiest way to adjust zenith. The tools are simple (a grease pencil, an appropriate tool for adjusting the zenith screws on your particular machine, and a roll of tape that you do not plan to use again), and you can attain as much precision as your time and patience allow. First comes the basic method, then the details and pitfalls.

With a grease pencil, mark the head area across which the tape passes, thread up a reel of tape, and hit the start button. After about twenty seconds, stop the machine, remove the tape from the head, and carefully observe what has

become of the grease pencil mark. If the mark is rubbed away evenly from top to bottom, the head is already straight, and you can proceed with the rest of the normal alignment procedure. If the remaining mark is thicker at the bottom of the head than at the top, then the head is tilted forward at the top; if the mark is thicker at the top, then the head is tilted back at the top.

Locate the two fore and aft screws which determine zenith. Depending on the head height and mounting configuration, alternately loosening and tightening these screws will swing the head forward and back. For instance, if your mark was thicker at the bottom (and therefore the top of the head is tilted forward), you will need to turn the back screw clockwise and the front screw counterclockwise. You may need to adjust only one screw if the alignment is already close, but be careful of overtightening, and be careful to maintain the correct head height. Make successive adjustments in small increments until the wear is even.

Several observations will help confirm that the adjustment is correct: the tape is not twisted but lies flat and planar between supporting members; the tape always stands perpendicular to the top plate of the deck; the guides and front edges of the heads are perpendicular to the top plate; and the tape doesn't skew across the head when you start the machine. Be sure to get all the greasy gunk off the heads and guides before doing anything else to the machine. The procedure described will seem tedious perhaps, but there's no substitute for having your heads straight before you start recording.

—Jonathan Thayer
Maintenance Engineer/Recordist
A&R Recording Studios
New York City, N.Y.

The Final Judge

We use a tape recorder in our act and record ourselves playing five or six different instruments and then augment that with "live" instruments and vocals on stage. All instruments are performed by my wife and myself, and the basic tracks are layed out on our TEAC 80-8 eight-track, then mixed through a TEAC Model 5 board with various ancillary equipment to a TEAC 3340 four-track. The final is mixed and played back on stage on the 3340 in three channel sound. (Bass and drums on center channel, keyboards, brass, strings and guitars,

etc., on the other two channels mixed in stereo. The fourth channel contains programmed information for a computerized slide and light show.) I have explained all this to you so you may visualize my situation.

In mixing down from my 80-8 to the final three channel master sets on the 3340, how do I establish a relative volume balance for each song (without a great deal of trial and error)? I find that by watching my VU meters, two different songs recorded at a 0 dB VU may have disturbing volumes in performance depending on the information in each song.

Thanks sincerely for a great mag. I have grown tremendously from reading it.

—Jim Robson
Family Waye
Mesa, Ar.

When all else fails, rely on the old reliables: *your ears!* The way to avoid the overall level difference between songs on your final mixes is simply to listen for this consideration during the mix and correct according to what you hear instead of what you see on the meters. The meters are merely tools or indicators of what is happening at various stages of the recording process. To rely on them exclusively would be a mistake.

One song which you average up to zero-VU might have louder parts than the next one because the meter is only showing you a part, or average, of the total signal at any given instant. The meter is intentionally dampened, in terms of ballistics, so that it may respond in an averaging manner. Some signals are of such intensity and speed that the meter can't rise fast enough to the correct position and, by the time it gets moving, the signal has passed into history. For example, you and I perceive the sound of castanets to be sharp and loud, rising above almost all other instruments in an acoustic orchestra. However, this signal of extremely high energy barely gets the meter going. In fact, it occurs so fast that you would be hard-pressed to find a peak-reading meter that would show it accurately. This tells us that we can't compare loudness solely by watching the levels reflected on the meters. Each song is a blend of different types and levels of information. Some of this may add to the volume or impact without even being registered on the meter.

Since your ear is the final judge which has determined that there is a discrepancy, let your ear judge the relative levels during the mix also and your problem will be eliminated. The general purpose of meters it to establish levels leading to the most efficient recording of signal onto tape. However, by relying entirely on meters, the musical concept that you wish to interpret and transfer might get lost or camouflaged in the process by the time you have the final product. During a pre-final mix to the 3340 adjust the overall print level in relative accordance with the previous piece. The key is to use your eyes *and* ears.

—Roy Kamin
Consumer Relations Manager
TEAC Corp. of America
Montebello, Ca.

Intercepting That Interference

We have a Nova IV bass synthesizer wired into the guts of our Hammond B-3 organ via a harness connection to

THE TAPE THAT'S TOO GOOD FOR MOST EQUIPMENT.

Maxell tapes are not cheap.

In fact, a single reel of our most expensive tape costs more than many inexpensive tape recorders.

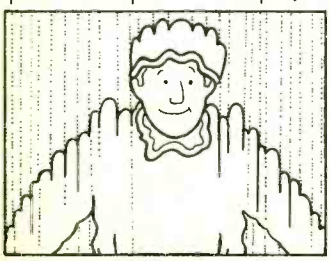
Our tape is expensive because it's designed specifically to get the most out of good high fidelity components.

So it makes no sense to invest in Maxell unless you have equipment that can put it to good use.

THE REASON OUR TAPE SOUNDS SO GOOD IS BECAUSE IT'S MADE SO CAREFULLY.

Every batch of magnetic oxide we use gets run through an electron microscope. Because if every particle isn't perfect, the sound you hear won't be either.

And since even a little speck of dust can put a dropout in tape, no one gets into our manufacturing area until he's been washed, dressed in a special dust-free uniform and vacuumed.



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WE CLEAN OFF THE CRUD OTHER TAPES LEAVE BEHIND.

After all the work we put into our tape, we're not about to let it go to waste on a dirty tape recorder head. So we put special non-abrasive head cleaner on all our cassettes and reel-to-reel tapes. Which is something no other tape company bothers to do.

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OUR TAPE COMES WITH A BETTER GUARANTEE THAN YOUR TAPE RECORDER.

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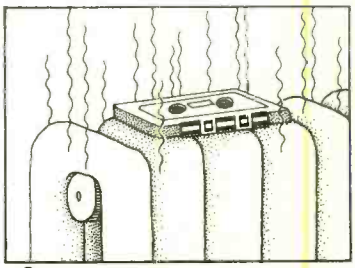
So our guarantee is simplicity itself: anytime you ever have a problem with any Maxell cassette, 8-track or reel-to-reel tape, you can send it back and get a new one.



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the generator. The unit, keyed by the pedals or by the first two octaves of the lower manual, is virtually part of the organ as was the original Hammond bass, only it sounds infinitely better and it has a separate signal output to drive a bass amplifier or feed a mixing console input channel.

Since the Leslie bass sound is not enough for our band, we connect the extra Nova output into one of our Yamaha PM-1000 inputs and the sound is phenomenal. Unfortunately, by doing this, a loud, nasty buzz/hum emanates from the Leslie (*not* the system speakers) in about half the places we play; consequently, we defeat all grounds in our system, solving the Leslie noise and leaving a more or less tolerable residual buzz in the system.

I accidently discovered recently (after months of frustrating equipment tests) that certain reostat-controlled lights in the room we are playing in (and apparently some of the other rooms we perform in) are the cause of the noise. I remember the Talkback mentioning RFI filters ("Stop That Buzz!" August 1977 issue, page 15). After many calls to electronics outlet stores, theatrical suppliers, etc., nobody knows where to get them, who makes them or how to use them.

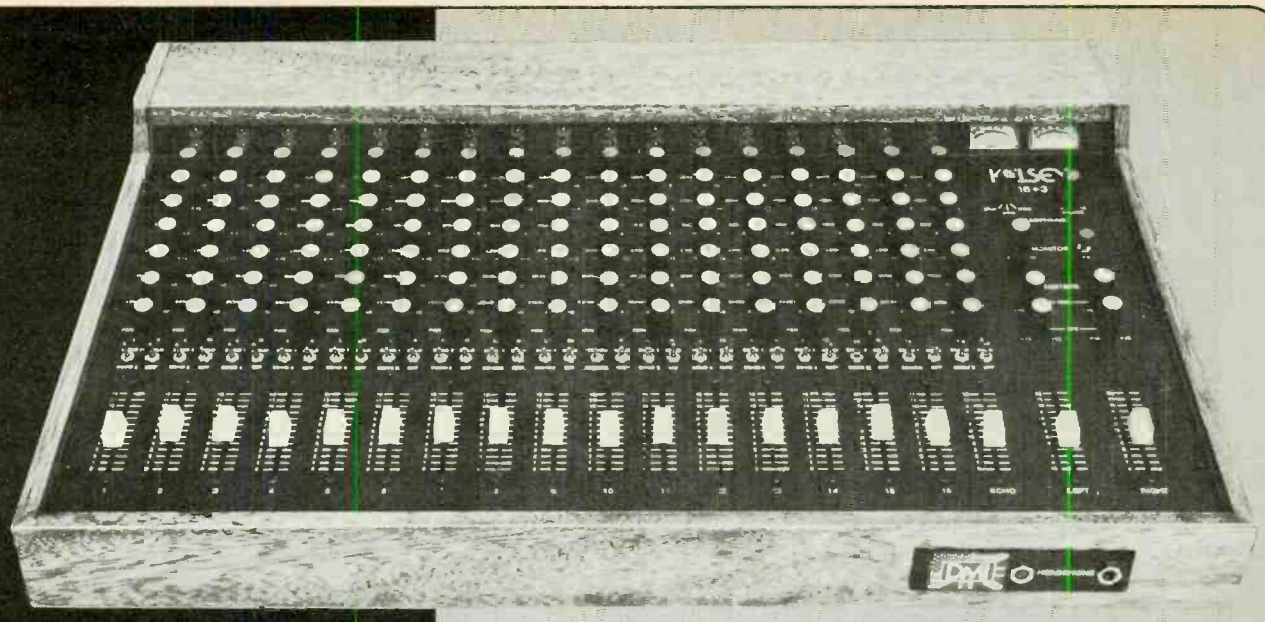
Should we place an RFI filter between the B-3 and the socket since only the Leslie makes noise, or should we filter all RFI coming into every AC line in our system?

To de-ground $30,000. worth of sound equipment is an unhappy situation, but we are reluctant to disengage the B-3's Nova from the system's console and we cannot tell the club owners to shut off their lights. Help! We need more info on RFI filters or any other ideas you might have.

—Jim Woodard
Ft. Myers, Fl.

First of all let me point out that the noise your system is picking up is one of the most insidious forms of RF interference around and usually comes from inexpensive solid state dimmers. The best cure for this RFI would be to treat it at the source, and some of the higher priced dimmers do this with moderate effectiveness.

Obviously you can't apply this solution without going broke very quickly, so a possible remedy (and the word "possible" implies no guarantee) would be to isolate the feed to the Leslie via a one-to-one audio transformer. A second transformer between the Nova and your



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SPECIFICATIONS:

INPUT

IMPEDANCES: Hi Z = 50 K unbalanced; Low Z = 200 Ohm transformer balanced. **MAX. INPUT LEVELS:** Hi Z = +20dBm; Low Z = +8dBm. **GAIN:** Hi Z = 0-46dB, continuously variable; Low Z = 12-58dB, continuously variable. **EQ:** High ±15dB at 10K, shelving; Middle ±9dB at 2K, peaking; Low ±15dB at 100 Hz, shelving. **MONITOR:** Pre-EQ, unaffected by off switch. **ECHO:** Post-EQ, Post fader. **LEDS:** Green lit from −10 to +21; Red lit from +15 to +21; 6dB headroom left when Red lit. **EQUIVALENT INPUT NOISE:** −110dBm from Hi Z input; −122dBm from Low Z input. **T.H.D.:** @ 1kHz, any level up to clipping typically less than 0.1 percent.

OUTPUT

IMPEDANCE: Nominal 600 Ohm unbalanced. **MAX. OUTPUT LEVEL:** 8.8V RMS @ 10K Ohm (+21dBV). **GAIN:** Mike in to line out +60dB. **EQ:** Hi ±15dB @ 3.5 kHz; Low ±15dB @ 35 Hz. **V.U. METERS:** "0 VU" = +4dBm at output of buss amp., switchable from stereo mix to monitor mix. **FREQUENCY RESPONSE:** Mike in to line out — ±1dB, 30 Hz - 20kHz. **SIGNAL TO NOISE:** Mike in to any output — typically 70dB. **T.H.D.:** Any output 1kHz any level up to clipping typically less than 0.1 percent. **POWER REQUIREMENTS:** ±15V DC @ ½ Amp.

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Yamaha might be needed as well.

Since the noise is broadband, filtering the audio lines wouldn't do much, but a line filter in the AC feed to the Hammond and/or the Leslie might help.

If possible try to power your equipment off a separate AC feed and avoid running audio lines close to, or parallel with the AC lines which are acting like antennae in radiating the buzz.

If you detect a certain hesitancy on my part to offer a positive cure for your problem it's because there is no real cure for this type of noise. That's why you'll probably never find a solid state dimmer anywhere near a professional recording studio.

The transformers mentioned above can be any 600 ohm 1:1 audio transformer, such as the UTC A-20, the Triad A-57J, etc. AC line filters by Cornell-Dubilier, Corcom, etc. are available from Newark Electronics, 1360 S.W. 65th Ave., Ft. Lauderdale, Fl. 33313 or any major electronics warehouse.

—Gordon L. Clark
Manager, Technical Services
Audiotechniques, Inc.
New York, N.Y.

[You might also wish to refer to the December 1977 Talkback entitled "Dimming The Buzz" (page 19) in which Wayne Pommer of Holden, Hamilton and Roberts Recording Studios discusses other manifestations of RFI and some possible cures. —Ed.]

Reasonable Requirements

In the December issue of *Modern Recording*, I saw a Letter To The Editor concerning a distributor in New York City (Gotham Audio) who sells small quantities of metal film resistors direct to the consumer (see "The Number's No Object," page 4). I have a problem similar to that of the author of that letter.

I have the schematic for a high-quality power amp. Designed by J. Lohstroh and M. Otala (AES reprint H-6(R)—"An Audio Power Amplifier for Ultimate Quality Requirements," IEEE Transcription on Audio and Electroacoustics, Vol. AU-21, No. 6, December 1973, pages 545-551), it requires polystyrene and polyester capacitors. I am having a very hard time finding these components in small numbers.

Although I don't know for sure, I believe these capacitors have good phase characteristics in the audio range. Could you please tell me where I can buy these components and the probable reason for their inclusion in the design?

—Charlie Engren
Olympia, Wa.

[Desirous, as we always are, to get all our reader's questions answered, we first contacted Russ Hamm at Gotham Audio once again. Russ supplied us with the name of the Sprague Electric Company in Seattle, Washington that manufactures the capacitors in question. Call them at 206-632-7761 and the problem of availability should be solved. Jackie Harvey of the Audio Engineering Society then supplied us with the full name and address of Professor Matti Otala, designer of the power amp and Director of the Technical Research Center of Finland Electronics Laboratory. The following is his response to Mr. Engren's second question.]

There are several reasons which influence the choice of different component types in a specific circuit. In the ampli-

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fier you mention, polystyrene capacitors were used as compensation capacitors because 1) the compensation must be stable in order to prevent oscillations and 2) the capacitance is not allowed to change with applied voltage, as this would create distortion in the sound. The nearest competitor to polystyrene would have been ceramic capacitor, which usually has a relatively large temperature coefficient, and whose capacitance is often strongly voltage-dependent. In general, this kind of capacitors should not be used in the signal path.

Polystyrene capacitors were used as bypass capacitors because of their low internal resistance and inductance at frequencies of 100 kHz to 2 MHz. This is the frequency range where the power supply line inductance starts to create oscillatory problems, when the amplifier handles rapidly changing signals or is driven into saturation. Note also, that the inclusion of the 1-ohm resistors in series with some of the capacitors is mandatory in order to create appropriate damping in the resulting resonance circuits.

Metal film resistors were used in the amplifier front end to minimize noise and to eliminate the voltage dependence found in some carbon composition resistors. This voltage dependence would again have created distortion.

When designing an amplifier which must have distortion values below 0.1% *before* the feedback is applied to correct the small residual imperfections, one must be very careful in the selection of components, and strict adherence to the specified component types (and sometimes to the specified makes) is advisable.

—Professor Matti Otala
Director
Technical Research Center of Finland
Electronics Laboratory
Oulu, Finland

A New Way To Use dbx?

I recently came up with an idea about dbx and its use. I thought that I'd "let it out of the bag," so to speak, so that if it is indeed a good idea, it can be put to use by others. The normal way to use dbx is like this: (see Figure 1). Why not do it like this? (See Figure 2.) My idea is that if you compress the signal *before* the EQ stage, you will reduce the peaks the later stages of the board have to handle, as well as the level of the peaks at the input to the deck. (See Figure 3.) Mathematically it works out equally, but the later stages only have to handle *half* the level as before.

Also, any reverb or external EQ or flangers, etc., will be receiving a dbx-compressed signal. The dbx only works half as hard cutting distortion all-around. And because the signal is flatter, you can use more gain on the preamp stages without noise or additional distortion.

—Kevin Reinholdt
Dallas, Tx.

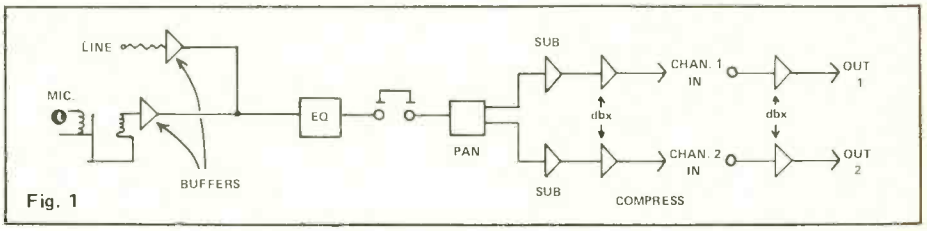
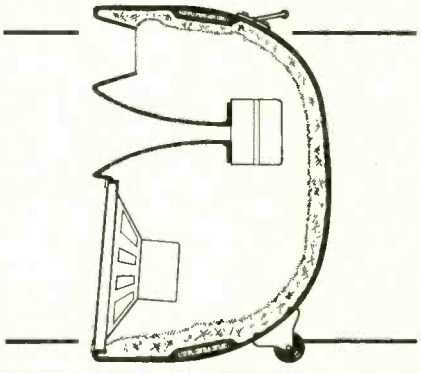


Fig. 1

At first
glance our cabinets
look a little different.
Which is understandable. They are.

It's quite plain that these are not your usual assortment of little flakeboard boxes with holes. These are functioning enclosures whose dimensions, determined by the laws of physics, have been precisely stated in hand laminated fiberglass.

All of our enclosures are of a solid round-backed design which frees them from the standing waves and out of phase diaphragmatic panels which plague "the boxes". Our design not only cuts down on unwanted resonances but also creates unparalleled durability and the strength to survive extensive touring and triumph over playful roadies. The GGM, our dual driver ported bass enclosure, fills a variety of functions. As the bottom end of a PA stack it makes a dynamite low frequency cabinet. Pre-amped and powered it becomes a bass guitar cabinet the likes of which you've never heard. The PBL,



our full-range super portable cabinet, incorporates bass, port and HF horn in one rugged enclosure. Perfect for club PA applications, this unparalleled (literally!) cabinet is perfect for keyboards and other instruments and also happens to be a fantastic side fill stage monitor. And our NC12? Only the loudest, cleanest monitor around. And everything you need to hear is aimed right at your face. Write or call us for more information. It's time you got the best.

	HF DRIVER	LF DRIVER
NC 12	1" - 1 3/8"	12"
PBL	1" - 1 3/8" - 2" - 2.8"	15"
GGM	n.a.	TWO 15"

Here's Looking at You, Kid!

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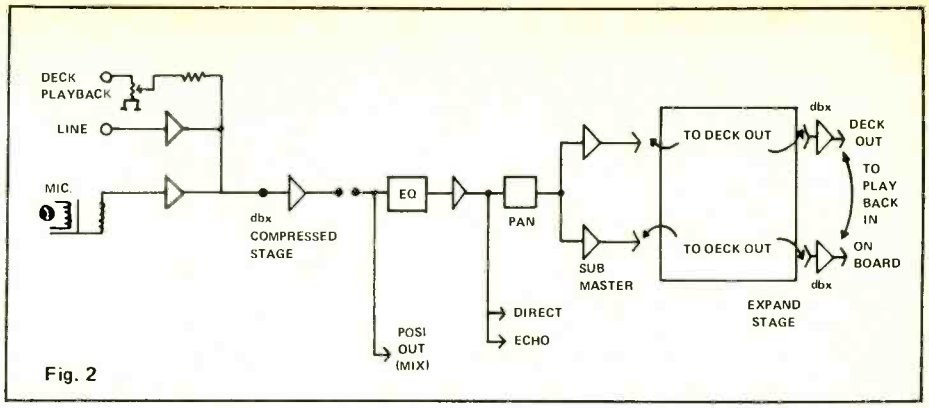

Fig. 2

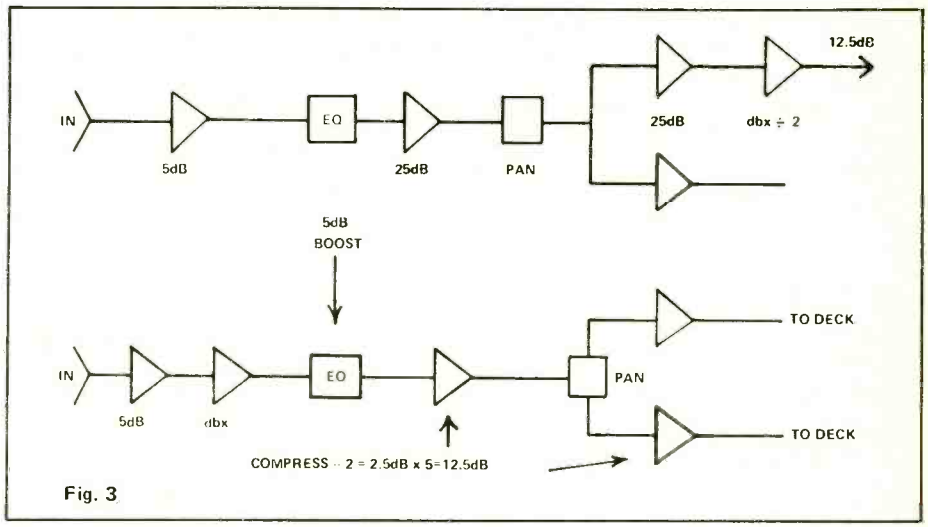

Fig. 3

Your idea is good in theory or at first glance; however, there are a few things that will not allow it to work properly. If a dbx encoder is placed at the output of the microphone pre-amplifier of a console, normally echo, cue, and equalization would follow.

The first problem with this is that non-linear processes such as equalization and compression should not be placed between a dbx encoder as it will cause decoding errors.

The other problem that arises is that when using a mixer, normally several microphones will be mixed together and combined to an output bus (channel). This means that a number of dbx encoded signals would then be combined together at the output channel. Two or more encoded signals cannot be combined and decoded by a single decoder. The compander process depends on a mirror image operation; the compressed information contains certain control information that tells the decoder precisely how to return the encoded signal to normal. When two or more encoded signals are combined together and decoded by a single decoder, the decoder does not know which information to look at and gets confused.The result would be some undesirable side effects.

If you were to do what you proposed and use no equalization and take each individual input directly to a channel on your tape recorder, you would have a separate track for each microphone which would require more tracks than you would normally use. These tracks could then be individually decoded and mixed down and give you the advantages that you sought in the original question. However, the noise in the microphone pre-amplifier, if any, and noise in the microphone, if any, would still remain as dbx would not remove any noise prior to the encoding process. When re-mixing the encoded tracks, whatever noise existed in your console, if any, would still remain.

I hope this answers your question. If you have further questions regarding this, please send another inquiry to Talkback and I will be happy to answer it at that time.

—Larry Blakely
Director of Marketing
dbx, Incorporated
Newton, Ma.

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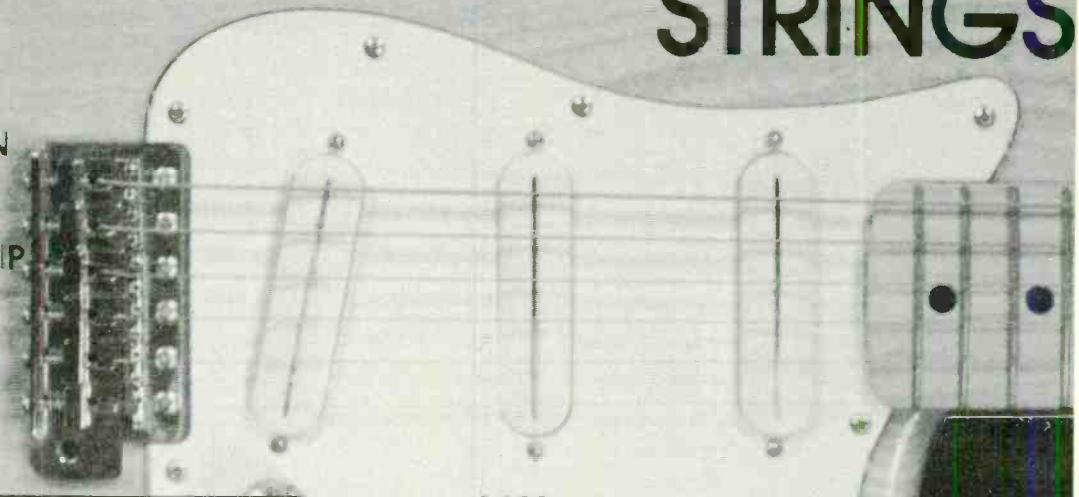
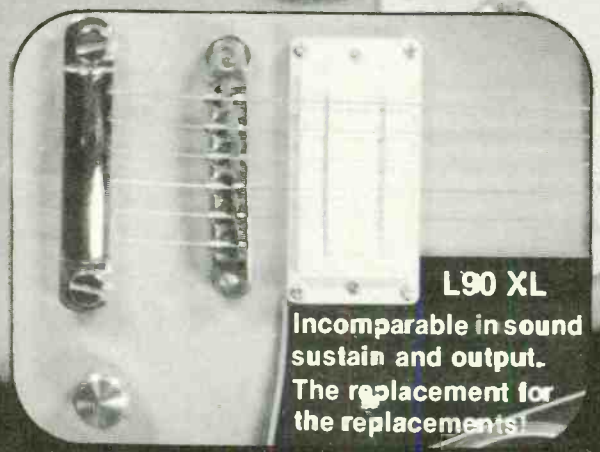


L220



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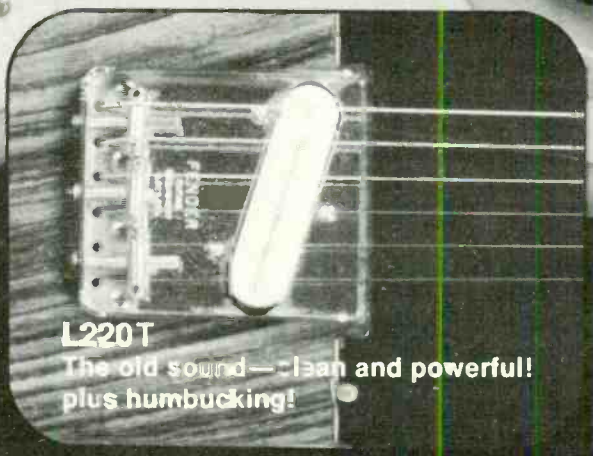

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PRODUCT SCENE

By Norman Eisenberg

MIC PREAMP/LIMITER

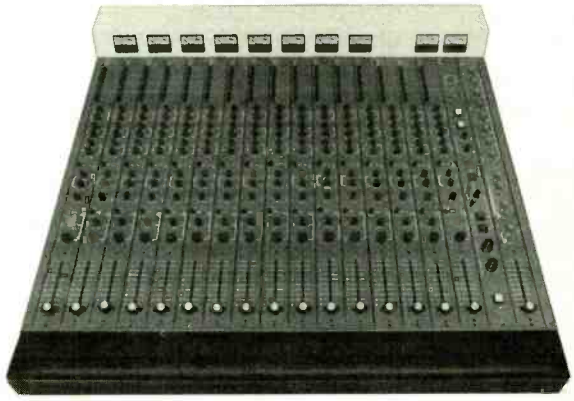
From RTS Systems of N. Hollywood, Ca. comes word of its model 1400 adapter, described as an in-line microphone preamplifier/limiter with many audio applications. The 1400's gain is continuously variable so that it may be used to boost low-level mics to drive low-gain mixers, or boost mic signals to line level, or to buffer medium level lines. The built-in limiter can be used to prevent overdrive of individual mic or line inputs while maintaining high average levels. According to RTS, the model 1400

has a fast rise time with absolutely no overshoot, even with 20 dB of limiting applied. Circuitry is said to be stable enough to withstand highly capacitive loads with no oscillation or ringing. These and other characteristics, says RTS, make the model 1400 useful in fixed or in portable sound reinforcement, location recording, remote broadcasting, broadcast production, as well as in industrial and scientific test labs. Low output Z handles mic cables of 1000 feet or longer with no significant loss of highs and while improving the S/N ratio. A battery-operated device, the 1400 runs on a pair of 9V alkaline cells that form a bipolar supply with operating life of about 100 hours.

CIRCLE 9 ON READER SERVICE CARD

MIXING CONSOLE

A 16-input, 16/8-output, stereo out mixing console, the model SP800C, has been announced by Speck Electronics of N. Hollywood, Ca. Expansion to 24-track operation is possible by the addition of the Speck 01 or 02 options. Designed primarily to operate with MCI, 3M, Stevens, Ampex, and other professional multi-track recorders, the SP800C also is said to work well with semi-pro tape machines. The input module features a slide fader, 6-knob/3- band parametric EQ, 8/16 track-assignment buttons, post echo send, monitor send control, two cue sends, solo button (which allows stereo panning), mic/line switch, program/sync switch, and an attenuator switch of −10 dB or −20 dB. The output section is said to contain "everything you need to do a professional and efficient recording session, from the stereo master fader to the 8 submaster level controls." Included are control room and studio level controls; cue 1 and cue 2 level controls, each of which can be soloed; slate and talk buttons with level control; two cue prompts; two cue returns; two 2-track playback controls; two echo returns and a self-contained microphone.



CIRCLE 10 ON READER SERVICE CARD

DBX ADD-ON FOR NAGRA

A custom-designed add-on noise reduction system for Nagra IV-S portable stereo recorders has been made available by dbx, Inc. The new model dbx 193 weighs five pounds, requires no power supply of its own, and provides two channels of simultaneous noise reduction plus a two-channel audio amplifier that, says dbx, can be used to drive small field monitor speakers.

Four separate signal-processing circuits—two for record and two for playback—permit the processed signal to be monitored during recording. The device bolts directly to existing tapped holes in the Nagra case using hardware supplied. Interconnections are made via two 7-pin Tuchel DIN connectors; cables are supplied with the dbx unit. When integrated with the Nagra, the dbx 193 is said to provide an additional 10 dB headroom, and better than 30 dB noise reduction. No level matching is required. Suggested retail price is $850.

CIRCLE 13 ON READER SERVICE CARD

REVERB TIMER

A device for measuring a room's reverb time (decay time)—as an aid in prescribing corrective acoustic treatment—is the new model 232 Reverb Timer announced by Acoustilog Inc. of New York City. Claimed to be a state-of-the-art measurement system, the model 232 is designed to compute room decay time within each of seven octave-wide bands. Using digital readout, the device allows the operator to make accurate measurements while freeing him from the time-consuming calculations required. The model 232 uses two sets of octave-band filters, one each in the send and receive lines. Its 10-millisecond resolution capability is said to facilitate quick and accurate measurements of small studio control rooms with decay times of less than 300 ms. A free-standing mic (user-supplied) prevents interference effects near the system case from causing misleading readings. The model 232 requires 1¾ inches of rack space.

CIRCLE 11 ON READER SERVICE CARD

PROTECTIVE ENCLOSURES

Amanita Sound Inc. of Easthampton, Maine is offering a new line of protective loudspeaker and standard 19-inch equipment enclosures designed to withstand "the ultimate" in heavy road use. Made of lightweight, rugged, low-density polyethelene with low resonance characteristics, the units are available in five colors. Handles and clasps are recessed, and stacking ribs facilitate handling and transit. Since there is no case to store, Amanita points out that appreciable space on the job may be saved.

CIRCLE 14 ON READER SERVICE CARD

TUNABLE CROSSOVER

Furman Sound announced its second professional product, the model TX-2, described as a low-cost, studio-quality electronic tunable crossover and bandpass filter. Suggested applications include use as a variable frequency splitter for PA and concert reinforcement loudspeaker systems, studio monitor systems, component musical instrument amp systems and as a bandpass filter. The device is tunable from 20 Hz to 20 kHz and is switchable for both stereo bi-amp and for mono tri-amp setups. Front-panel controls provide up to 8-dB gain for driving power amps. Circuitry employs active Butterworth filters with 12 dB per octave rolloffs, claimed to provide smoother frequency handling and minimal phase displacement. High and low-pass outputs of each crossover point track each other automatically as frequency is changed. The TX-2 measures 19 inches wide by 1¾ inches high and 8 inches deep. Suggested price is $250. Furman Sound products are distributed by Rothchild Musical Instruments of Englewood, N.J.

CIRCLE 12 ON READER SERVICE CARD

NEW TAPE TRANSPORT CONSOLES

Ruslang has introduced two new tape transport consoles. The RL 600 is designed to accept the new style of tape decks 19 inches up to 21 inches which have their electronics integrated with the transport. The RL 700 accommodates transports 19 inches up to 24½ inches such as the Ampex 300 series. (Ruslang's model RL 500, introduced last year, handles 19 inch by 15¾ inch decks.) All three models incorporate such features as front-panel access in both horizontal and vertical positions, plus a rear shelf for power supplies.

CIRCLE 17 ON READER SERVICE CARD

TWO NEW ACCESSORY LINES

Two competing tape firms have announced new accessories. From TDK there's word of a head demagnetizer and a head-cleaning kit. The former device comes in cassette format, containing its own circuitry and is self-powered via a 1.5-volt cell. To use it, you place it into the deck, push the PLAY button and a few seconds later an LED lights up to indicate that the heads have been demagnetized. Retail price is about $20. The cleaning kit contains mirror, brushes, pads and liquid all housed in a cassette box. It costs about $6.

The Memorex products include a tape recorder care kit for use with cassette, 8-track and open-reel equipment. Included are a pad and wand, inspection mirror and cleaning brush. Memorex also has announced three disc-recording care kits. One is a tracking record cleaner; another is a stylus care kit which features a built-in magnifier on a VTF gauge; the third is a "deluxe" record cleaner featuring a brush to apply a cleaning fluid to a disc. Prices for these items are: tape care kit, $9; tracking record cleaner, $8; stylus care kit, $8; deluxe record cleaner, $15.

CIRCLE 15 ON READER SERVICE CARD

TIME DELAY UNITS

From Altec Lansing there's word of three new digital time delay systems, the models 1640, 1660 and 1661. The 1640 is described as a low-cost model that makes audio time-delay available in applications that were previously restricted from using this technique due to its cost. The other two models are designed for sophisticated sound-reinforcement applications.

The 1640, says an A-L spokesman, features shift-register type digital circuitry and provides a maximum delay time of 120 ms, with six outputs at fixed 20-ms delay intervals. The 1660/1661 features random access memory (RAM) type circuitry and accepts up to six memory modules for a maximum delay time of 510 ms, with five output modules. Full details, including illustrated specs, are available from the manufacturer.

CIRCLE 18 ON READER SERVICE CARD

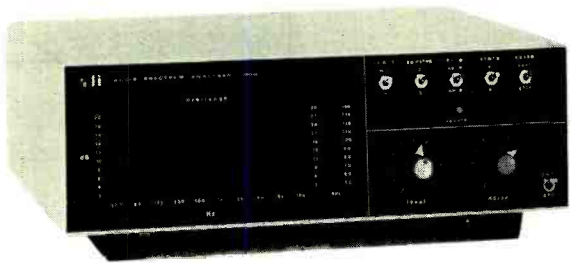
NEW POWER AMPS

Two new stereo power amplifiers have been announced by QSC Audio Products. The model 4.1 is rated to deliver 75 watts RMS per channel into 8-ohm loads; the model 5.0 amplifier is rated to produce 120 watts per channel into 4-ohm loads. Both power levels are given with reference to a 1% clipping level. Typical distortion figures are 0.25% THD and 0.25% IM at power outputs of 70 watts and 114 watts per side, respectively. Each amplifier may be rack-mounted, and each has current-limiting short-circuit protection, thermal cut-out and fused collectors on the output transistors to protect the load. Both amps also have calibrated gain controls scaled in both dB and Vrms. Either model can be ordered with or without internal electronic crossover that allows it to be used in a bi-amp setup. List prices are $308 for the model 4.1, and $368 for the model 5.0. The internal electronic crossover module costs an additional $50.

CIRCLE 16 ON READER SERVICE CARD

ADI OFFERS ANALYZER AND EQUALIZER

Two new products have been announced by Audio Developments International (ADI) Corp. of Palo Alto, Calif. One is the model 1000 Audio Spectrum Analyzer, said to be designed for the most critical measurements of real-time acoustic phenomena. Suggested applications include fast loudspeaker and room frequency response testing; equalizing acoustic environments; tape recorder tests and calibration; program material frequency analysis; before and after "freeze" comparisons; real-time analysis, collection, and storage of data. Among its features are two digital CMOS RAM memories, a



"truly random" pink-noise generator with gating capability and a precision condenser microphone.

ADI's other new item is the model 1500 automatic graphic equalizer, which is claimed to be capable of achieving "the most critical equalization . . . without peripheral equipment whatsoever. . . ." Ten bands on two channels are controlled by sliders with ranges of ±12 dB. The 1500's automatic display system uses LEDs in red and green color combinations to indicate different levels of high, low, optimum and flat.

CIRCLE 19 ON READER SERVICE CARD

SOME RECENT SPEAKERS . . .

Some recent speakers that I have auditioned lately pose the question: Can these models be termed "monitor speakers?" I leave that open to debate since—as I have pointed out in the past—that concept seems as much in the realm of semantics as acoustics. A speaker system that is reasonably smooth and relatively "uncolored" across the audio range, and which can handle power levels sufficient to cover *your* listening room or studio with clean sound, would qualify for monitoring use. Above all, it should reveal the program material, not conceal it or color it to make you believe the program material has more (or less) in it than it actually has.

The Jensen LS-6, list price $290, is a three-way floor-standing system rated for input power of 10 to 90 watts. Bass below 50 Hz is somewhat distorted but there's still clean output down to about 25 Hz. Upper tones are well dispersed with a rolloff to inaudibility at about 15 kHz.

Bigger, and costlier, is the $695 Altec model 19 which appears to be an updated and refined version of the old Voice of the Theatre system—big horn tweeter and bass-reflex loaded woofer. It pumps out a very big sound with relatively little input power (rating is 10 to 65 watts). Here the low end remains quite clean down to 40 Hz, and continues with some doubling to the 20-Hz mark. Middles and highs are adequately dispersed; a dip is evident at about 15 kHz but rsponse goes on to beyond audibility. At almost 40 inches high, the model 19 is very much a floor-standing system.

One of the cleanest-sounding imports I have heard is the British-made Celestion 66 which is sometimes used as a monitor at the BBC and elsewhere abroad. A floor-standing "vertical" system, the model 66 uses tweeter, midrange, and woofer plus an auxiliary or passive bass radiator. Rated input power is 10 to 70 watts, but the upper figure is typically conservative British—I ran it up to nearly 175 watts (with occasional peaks nearing 400 watts) and experienced no problems. The low end rolls off below 50 Hz but there is still plenty of fundamental bass down to 25 Hz. Middles and highs are exceptionally smooth and well distributed to beyond audibility, with a rolloff starting at about 14.5 kHz. Price is about $500.

For the "small" studio, consider the new Bose 601 which, at $558 per stereo pair, offers more than usual versatility in placement thanks to the unusual arrangement of cross-firing tweeters. At 25 inches high, a model 601 can be floor-based or perched on a stand. It seems to defy placement problems that would faze conventional speakers and within its rated power range of 15 to 50 watts it produces a surprising amount of very clean and transparent sound with a low end that makes it down to 20 Hz (albeit with a rolloff and increase in doubling from about 50 Hz), and a smooth and well-dispersed upper range that carries out to beyond audibility. The 601, unlike Bose's 901, uses no equalizer.

MUSICAL NEWSICALS

MIXING CONSOLES

Over the past few years, one of the best values in mixers has been the English-made Kelsey modular console. Dallas Music Industries, the American distributor for Kelsey, has available the new series of non-modular Kelsey mixing consoles which are in 8, 12 and 16 channel versions. Each input channel has a transformer-balanced low impedance mic input with 3-pin XLR connector, and a high-impedance, unbalanced input via a ¼-inch phone jack. Preamp gain is continuously variable from 12 to 58 dB in the mic input and 0 to 46 dB for the high-Z input. A green LED indicator indicates proper signal level at the preamp, and a red LED indicator indicates the 6 dB headroom point to warn of approaching distortion. Equalization comprises ±15 dB high- and low-frequency shelving equalizers and a ±9 dB midrange peaking/dipping equalizer. A pre-EQ monitor send and a post-fader echo send with a two-bus selector switch are provided along with panpot, channel on/off/headphone solo switch and a 60-mm sealed fader. An extra input channel is designated as the echo return, having two high impedance inputs. The output section has the left and right master faders, bass and treble EQ for the main outputs, monitor send master, two VU meters and a selector switch and level control for the 4-watt per channel headphone amp. Prices for the 8, 12 and 16 channel versions are $1350, $1700 and $2100 respectively, including fitted SMF Tour Series road cases.

CIRCLE 3 ON READER SERVICE CARD

POWER AMPLIFIERS

Audio Marketing, Ltd. here in the U.S. will be distributing the product line of HH Electronics of Cambridge, England. The top of the HH power amplifier line is the S-500-D, a solid-state, two-channel amplifier rated at 210 watts RMS per channel into 8 ohms (or a maximum output power of 500 watts RMS per channel into 2.5 ohms) at the clipping point. The unit may also be operated in a bridged mono mode for 640 watts RMS into a single 8-ohm load or 900 watts RMS into a minimum impedance of 5 ohms at clipping. Physically, this HH model is a very interesting and compact design. The extra-heavy gauge aluminum, rack-mount chassis is only 3½ inches high and contains a massive power supply and two self-contained, plug-in circuit modules which may be replaced in a matter of a few minutes in the unlikely event of a failure. Most high-power amplifiers available today require external fans to get sufficient air circulation for proper heat dissipation. Each of the two HH circuit modules incorporates its own Force Cooled Dissipation system which includes a computer-type rotary fan and special aluminum foil coolers as integral parts of the module. In the event of a fan failure, air-flow restriction or prolonged electrical fault conditions, the amplifier will be shut down temporarily by an automatically-resetting thermal sensor. Also of note is the use of a 1 kilowatt rated toroidal power transformer to minimize hum induction into the amplifier circuitry.

CIRCLE 4 ON READER SERVICE CARD

SOUND REINFORCEMENT

From England comes news of Zoot Horn amplifiers, distributed in the Americas by Concert Systems of Canada. The Zoot Horn line comprises three speaker systems and the E500 amplifier to power them. The E500 is a 200-watt RMS amplifier with tube output stages and sophisticated control features. The amp has two inputs with individual volume controls. These are followed by a 9-band graphic equalizer with a ±20 dB boost/cut range for extreme tonality changes; center frequencies are at octave intervals from 50 Hz to 12.8 kHz. In addition to the Graphic equalizer are a 12 dB/octave low filter with cutoff frequency variable from 20 Hz

up to 2 kHz, and a 12 dB/octave high filter with cutoff variable from 320 Hz to beyond audibility (flat response), and a Power Shaper Effect which boosts bass and treble by a fixed amount with the flick of a switch for even greater tonal variations. Other features of the E500 include a master volume control, two preamp outputs, Cannon speaker output connectors, and a 4/8/16-ohm speaker selector switch. On the speaker side, Zoot Horn has three models for various purposes. Model SD18 is a rear-loaded, folded horn enclosure for a single 18-inch driver. The CB15 is a compact, high-efficiency bass enclosure suitable for bass guitar, organ, or as the bass section of a small P.A. or keyboard amplification system. The cabinet is a rear-loaded horn accommodating a single 15-inch speaker. The Model TX 200 is a full-range enclosure with frequency response from 60 Hz to 15 kHz. The TS200 houses two 15-inch drivers in an infinite baffle enclosure for the bottom end plus a compression driver and horn for treble.

CIRCLE 2 ON READER SERVICE CARD

ACCESSORIES

Musitronics Corp. has a new piece in their line of Mu-tron sound modifiers, the Mu-tron Opti-Pot Volume-Wah Pedal ($130.). The Mu-tron Volume-Wah is an AC-powered pedal using the exclusive Opti-Pot photo-electric system to eliminate potentiometer noise and mechanical unreliability forever. Separate footswitches with LED indicators are provided for volume and wide-range wah-wah functions to allow volume-only, wah-only, or combined volume-wah operation.

CIRCLE 1 ON READER SERVICE CARD

The people at Electro-Harmonix have introduced a new analog delay device, the Memory Man Deluxe. This new unit generates a signal delay which is variable from 15 to 400 milliseconds for a wide range of effects from vocal doubling at 15 msec through echo and reverb effects to a slapback echo at 400 msec. Unlike many other delay devices, the signal bandwidth of the Memory Man doesn't change as the delay time is varied; frequency response for the delayed output is 10 Hz-3 kHz at any delay setting. Signal-to-noise ratio is 60 dB, and audible quality is improved further by the built-in noise gate. Controls are provided for input gain (with overload indicator), delay time, feedback (repeats) and blend. Memory Man Deluxe is AC-powered, and has separate outputs for direct and echo outputs.

CIRCLE 5 ON READER SERVICE CARD

DOD Electronics of Salt Lake City, Utah, has been on the scene for several years and their latest showings include additions to their steadily growing product line. The Overdrive Preamp ($49.95) is a solid-state distortion device featuring a level control to set the distorted output level and a gain control to vary the amount of distortion. The E601 Graphic Equalizer ($79.95) is a battery-powered 6-band equalizer with a ± 18 dB boost/cut range and center frequencies slightly more than one octave apart from 65 Hz to 3.3 kHz. Other products in the DOD line include two phasors, a compressor, an FET preamp, a 4-input passive mixer, an A/B switch box and a low-Z/high-Z transformer box.

CIRCLE 6 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

The people of Orange Musical Industries of England tell us that among their latest pieces of equipment is their new Orange Overdrive amplifier ($645), a 120-watt RMS tube amp based on their popular Graphic amplifier. The main difference between the Graphic and the Overdrive is the addition of a master volume control to allow overdrive distortion of the preamp section at low or moderate playing volumes. EQ on the Overdrive include bass and treble controls with a ±10 dB range plus a presence control. Front panel jacks are provided for high and low sensitivity inputs and effects send and return, while the back panel has output jacks for 4,8 and 16 ohm loads. Also noteworthy in the Orange line-up is the Model 103T, a general purpose 200-watt RMS solid state slave amplifier.

CIRCLE 7 ON READER SERVICE CARD

Norlin Music, Inc. of Illinois, has introduced a brand new line of sophisticated amplifiers known as the Lab Series, which drew on the expertise of engineers from Gibson and Moog as well as musicians, music dealers and recording studio personnel in their design. Lab Series amps use several innovative circuit developments including a new type of distortion circuit, a built-in compressor and an octave-band mid-

range control with selectable frequency. In addition there is a feature called Multifilter which "rearranges the harmonics to give them the characteristics of a highly resonant acoustic instrument's harmonics." The Lab Series amps were designed for physical ruggedness and reliability, using fingerlocked pine cabinets with steel shoulders, a shock-mounted electronic chassis and eight high-voltage output transistors for electronic reliability.

CIRCLE 8 ON READER SERVICE CARD

THE CARE AND FEEDING OF YOUR MULTI-TRACK RECORDER

By David Moyssiadis

If you're the owner of a small studio or are charged with the responsibility of doctoring studio equipment—but do not have a solid technical background—then this article is for you.

We will assume that holding a screwdriver and using it is not your idea of an exotic and perverted experience, but that the thought of a recorder being out of whack terrifies you. There's no need to feel uneasy even if you don't understand the service manual or don't have one. This article will give you the SOP for routine maintenance which will keep a healthy recorder in top condition for years.

Too many people buy a tape recorder, plug it in and start recording without the slightest regard to bias, EQ and standard levels, let alone azimuth and a whole bunch of other imponderables. There are many facets to recorder alignment that you may not know about but that are essential to making your recorder give you everything it is capable of. If you don't put your machine into proper alignment you may as well take half the dollars you spent on it and light cigars with them.

The procedures are not difficult if you have *any* mechanical or electrical ability at all. Although the theory of operation of a recorder is highly complicated and paradoxical (so much so that if you understood it you would be convinced that recorders can't possibly work!), the care of one is, in reality, easy, much easier than owning most other animals.

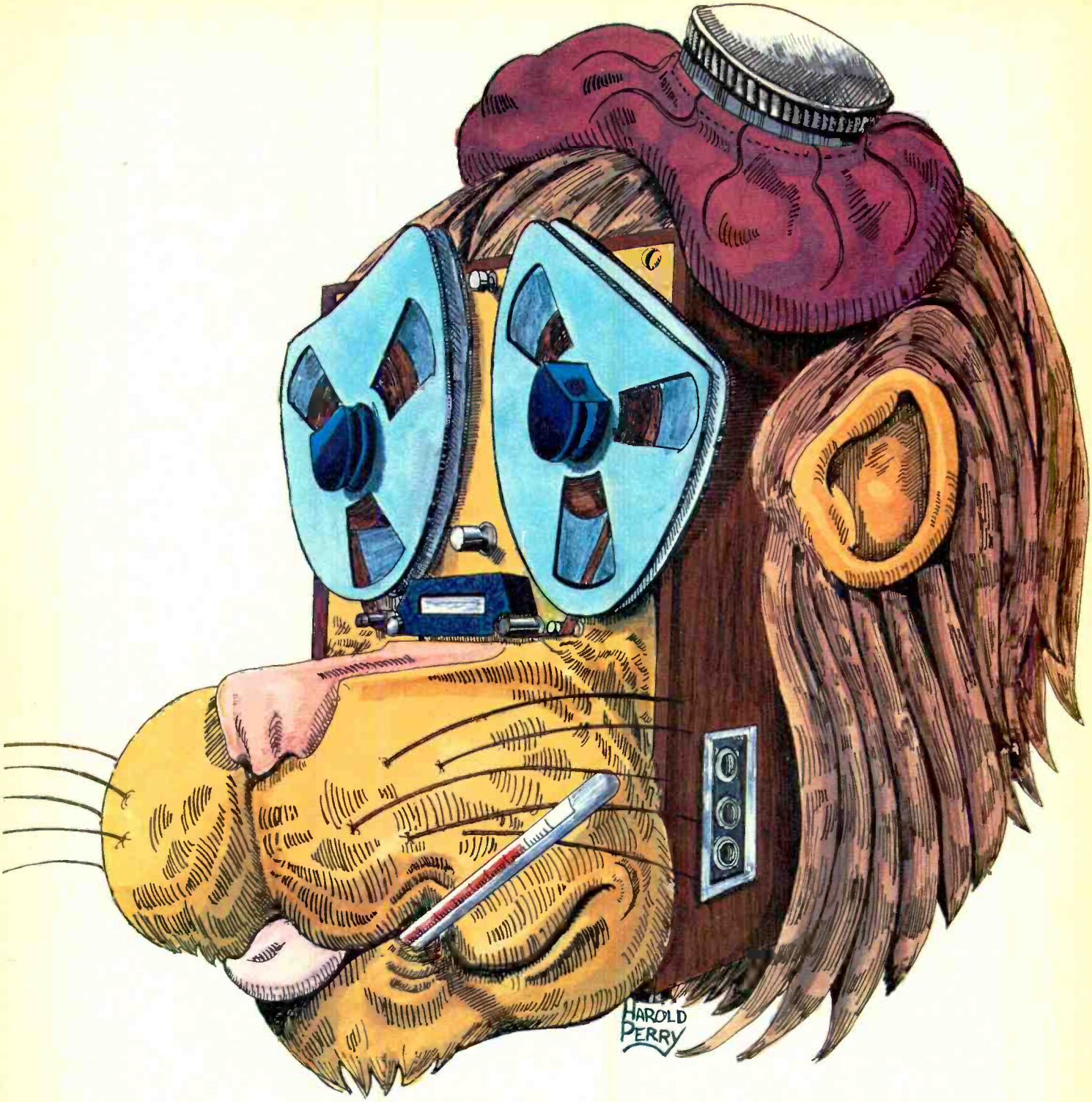
Tools of the Trade

Although for the most part we no longer wish to record music exactly as it is played, our machines still must do exactly what they are told. This means that they must magnetically read and write precisely what they are given. In order to do this they must be of high quality to begin with, and then they must be properly maintained. The tools you will need are as follows: test tape; reel of blank tape of the brand and formula you intend to use; audio oscillator; VTVM; tape head degausser (demagnetizer); and an assortment of screwdrivers and nutdrivers that fit your recorder.

The test tape is essential for you to find out where all the other recorders in the world are at levelwise; it is your standard. Many moons ago when life was simple there was only one standard level and few formulas. Today there are too many formulas and too many standards, and thus even standards aren't standard. Everyone wants to get into the act and have thereby confused the issue (and a few recording engineers, too). Anyway, pick your level and let's go.

"Standard" Levels

The original standard level was 185 nanowebers/meter, then came the first elevated level at 200 nw/m, which was about 0.9 dB higher than the standard, and now we have 250 nw/m which is about 3 dB higher than the original standard. There is also the European DIN standard of 340 nw/m which is about 4 dB up, but



they keep that stuff in Europe. These higher standards, in all fairness, do serve a purpose and that is to fill the need of the new high output tapes. You can use the 185 nw/m level and take advantage of about 6 dB more head room or you can record 6 dB hotter and get that much less noise, or split the difference which is what most people do.

Next is the oscillator. Any halfway decent audio oscillator will do as long as it is capable of going from 20 Hz to 20 kHz without too much distortion.

Since the VTVM is the ultimate check on the oscillator and the recorder it must be good, i.e., recently aligned and calibrated by the manufacturer or its representative.

Almost any head degausser will do; and the tools should be demagnetized. If you can pick up light pieces of iron or steel such as iron filings or razorblades or even nails and screws you don't want the tool near the recorder's heads.

First Things First

Now the first thing you must do is to clean and demag the heads. NEVER place an alignment tape on a

recorder before demagging the recorder's heads—even if you did the job just an hour ago. That tape is expensive—$30 or so, with the two-inch alignment tapes going for several hundred dollars—and if it gets near a magnetized head or a magnetized anything it's rendered useless. How do you demag? *Very slowly.* First get the unit about six feet from the recorder and then plug it in. Turn off the recorder, especially if it is a solid state job (unless it has a demag position, if so, use it). It is possible to burn up the input stage of the electronics on some transistorized units by virtue of the power generated from the demagnetizer into the head. When the electronics have discharged and died bring the pole piece of the demagger near the pole piece of the head, but do not touch the face (you may want to put some plastic tape over the degausser's face to prevent scratching the recorder's head). Now slowly move the demagnetizer up and down the face of the head and *very slowly* move on to the next head. At the last head VERY SLOWLY remove the degausser to a distance of about five or six feet from the recorder before unplugging it. If your unit has a switch on it you have to be even more careful. Do not turn the thing on or off within six feet of the heads. If you do so it is possible to magnetize the heads more strongly than the degausser is capable of demagnetizing. That's because the magnetic field is stronger on build-up or collapse than in a steady-on state.

The way it works is like this. As the program material on recorded tapes passes over the heads the heads gradually pick up a permanent magnetic charge. The same thing happens when you put the recorder into record—the record head gets magnetized. After some time the heads are magnetized enough to partially erase any tape passing over them. The high frequencies are the first to go. The demagnetizer sets up a strong alternating magnetic field which first magnetizes the head in one direction and then the other. As the degausser is pulled away this field gradually diminishes until it is practically zilch at the head—demagnetized.

Test Tape Tones

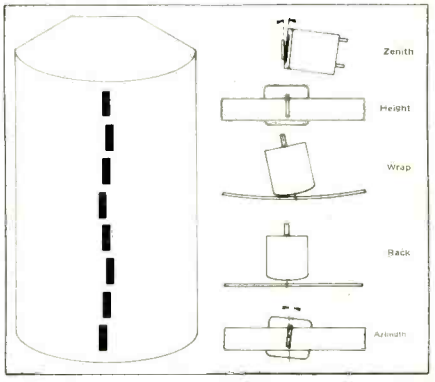
Now you're ready for the test tape. Make sure you have one that is less than a year old, and *not* one your buddy made on his machine. Turn the recorder back on and thread the tape; you will find a series of tones. Some tapes start with 700 Hz for a level set. Play this tone and set the recorder's VU meters to read 0 dB. Then check the output with the VTVM at the line out jacks. (All the equipment should be stabilized by turning everything on about one-half hour before.) After you get your playback level established the next tone should be an azimuth check at about 15 kHz. But first you may want to look at a lower frequency such as 7 or 8 kHz as a coarse adjustment, and then the highest tone as a final adjustment.

In any event there are about four methods used to adjust azimuth, one more accurate than the next. First is the coarse, crude, klutzy way, which is to adjust for maximum output on the VU meters. Don't do it; it's too coarse, unless you have a full-track mono recorder, in which case that is the only way. The next way is to mix both outputs into one meter on the mixing board and adjust for maximum level. This may not seem more accurate, but it is because the peak is sharper. More accurate than this is to mix the two channels *out of phase* and adjust for a null instead of a peak. This is the most accurate for the effort and does not require much knowledge. The fourth method utilizes an oscilloscope. However, if you know how to use an oscilloscope you probably aren't reading this anyway, so . . .

When you adjust azimuth be sure to adjust the *azimuth* and not the *zenith*. The two adjustments are easy to confuse on most machines. There are five different adjustments for a tape head. All you are interested in is the azimuth. If you mess up the others you're in trouble. Also, it is easy to align on a side peak. Don't fall into that pit. That's what the coarse adjustment is for. When you mix the two channels both parts of the signal should be equal or the peak or null will be hard to define. If you are adding in phase set each channel to −6 VU, when added together both should equal 0 VU; if you are mixing out of phase set each to 0 or even +3 VU, when added together adjust for null on the VU.

Adjustments

If you are adjusting a multi-track of more than two channels plan on a visit from the queasies. You'll never know if your azimuth is right on or not, although you can get it reasonably close. This is a result of a condition known as *gap scatter*. Manufacturing techniques have not been capable of getting all the individual gaps for each channel to line up perfectly, thus gap scatter. The figure shows an exaggerated rendition as the actual offset on an expensive professional recorder is only 5 microns or 5/1,000,000 of an inch (the head cap is 4 or 5 microns). But all you need is a 1/100,000 inch offset to mess up the peaking or nulling adjustment by 2°. The best you can do is to pick two tracks near the edges and adjust as described—e.g., on a 24-track recorder use tracks 1 & 24 or 2 & 23 and hope they are reasonably close with respect to their mutual



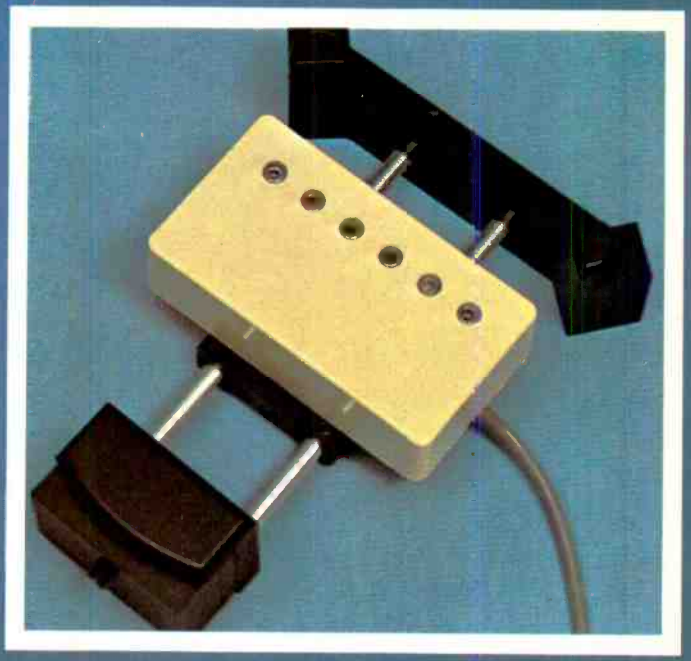
Gap scatter (enlarged to show detail) and the five basic head adjustments.

alignment. Never use more than two tracks. Then rub your rabbit's foot and count your worry beads.

From here on in all the adjustments are the same whether you have a mono recorder or the latest 24-track machine.

After the azimuth tone on the test tape will be a series of tones going down to about 30 Hz. All you do is watch and write down the VU meter readings for each tone. If all the tones hit 0 VU you don't have to do a thing. Rots o' ruk fella. Indubitably you will have to make adjustments. All machines whether purebred pedigrees or plain old mutts have the same parameters and the same adjustments. It's just that there are some machines which do not make their adjustments accessible or adjustable. Which simply means that the manufacturer chose not to make a certain parameter adjustable, either out of an overblown sense of pride or from enough wisdom to keep

From the company that changed the sound of playing electric...



The DiMarzio Acoustic

The Acoustic Model II is a humbucking, magnetic pickup for steel string acoustic guitar. A simple, sliding rail arrangement gives the Model II the unique ability to tune any acoustic guitar for harmonic balance — a player can emphasize overtones or fundamentals in any proportion he chooses.

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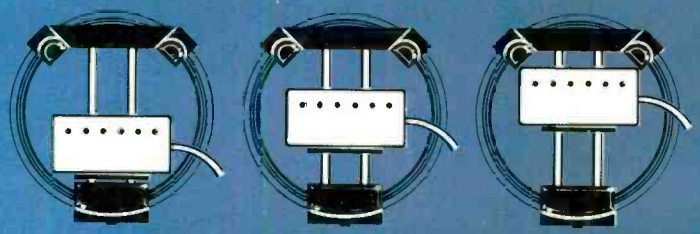

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CIRCLE 90 ON READER SERVICE CARD

someone from thoroughly screwing up the machine. Nonetheless, here is a list of the adjustable parameters of a tape recorder's electronics section by section:

PLAYBACK SECTION	RECORD SECTION
Playback (output) level	Record (input) level
High-frequency eq	Meter calibrate
Low-frequency eq	High-frequency eq
	Low-frequency eq
BIAS OSCILLATOR	**TRANSPORT**
Bias adjustment	Azimuth (head)
Bias calibrate	Zenith (head)
Erase current	Rack (head)
Bias frequency	Wrap (head)
	Height (head)

Not all machines have all of these adjustments. In fact, only a few expensive machines have all of them. The ones that your recorder doesn't have cannot be adjusted, of course, so you don't have to worry about them. As you run down the tones and note where the machine is off make the appropriate correction with the equalizers. To keep from going bananas make sure 10 kHz on the high end and 100 Hz on the low end are at 0 VU and let everything else fall where it wants to. The reason for this is that most recorder equalizers peak at these two frequencies and when these are on the mark that is usually the closest you can get the machine over all.

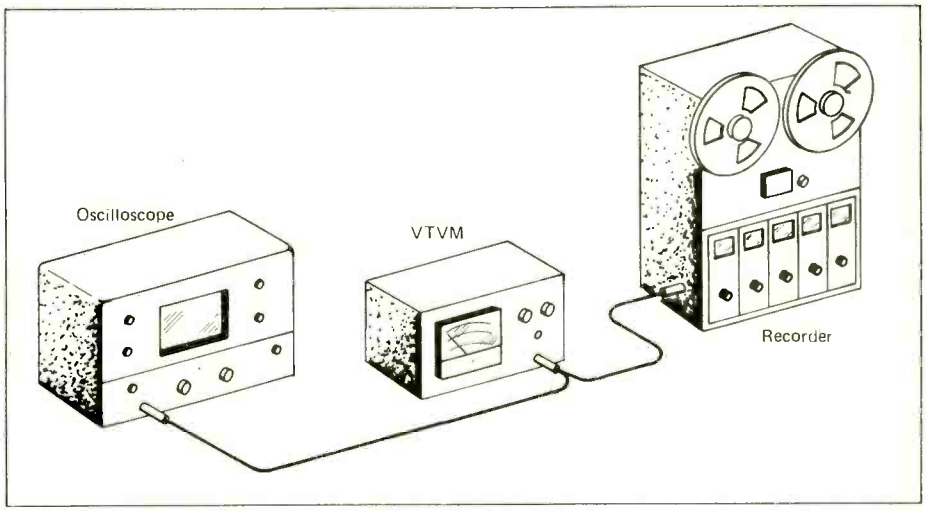
If the recorder is not of very good quality and you find that when 10 kHz is at 0 VU that 7.5 kHz and 15 kHz are both down about 2-3 dB then you can go up perhaps 1.5 dB at 10 kHz to achieve a flatter sounding curve. If on the other hand you are flat out to 12 kHz and you find that 15 kHz is way down (−4 dB or more) there either is dirt on the heads or there is a wear problem which requires relapping or a new head. (Dirt can get on the heads in the time it takes to play the test tape, so don't think because you cleaned the heads that you have a wear problem. Dust off the head with a cotton swab again to make sure.)

After all the PB adjustments are made go back to the level set tone at operating level and tweak it for 0 VU again, then remove the knobs and don't adjust them anymore since now this is your only point of reference. If you touch the output level after the alignment tape is off the machine you have to demag and go through the PB alignment procedure again. This is the *only* way you can tell what your machine is doing—by looking at playback *not* input.

Now thread some blank tape of the type you have decided to use from now on onto the recorder. Set the oscillator to 1 kHz and feed the signal into the line input. If your recorder does not have a bias adjust but has those inane switches that give you a choice of high or low bias settings try to get the highest reading on the PB and that's it. If you have real bias adjustments do this: get a reasonable level (always looking at PB) about mid-scale and then adjust for the maximum level you can get for each channel. Anything less than the maximum is no good, you will lose more than you gain.

Some smart guys who have "the answer" are running around telling people that their secret is to slightly underbias their recorders (usually about ¼ dB) and this gives them a better high end. This is true, it does give a better high end, but what these guys don't know is that it also gives them more noise and more distortion and a worse low end along with the better high end. If you need more high end that's what the equalizers are for. Conversely, if you overbias you get a better low end and less noise, but also a worse high end and more distortion. So the best way is to follow the advice of the people who design recorders, they know a little more than these other guys, and they say peak for maximum output.

Anyway, once you have done this adjust the record or input levels so that the output reads 0 VU. But make sure you are feeding "0" in. Connect the VTVM to the oscillator as shown in the figure to verify this. Then adjust your azimuth on the record head the same way you did the PB head. If your machine is equipped with sync you can take a shortcut. Be sure the recorder is demagnetized, put the machine in sync and play the test tape azimuth tone and adjust the record head directly. However, each time you adjust anything in the tape path the azimuth goes out on everything else. So you may have to go back and forth a few times to find the mid-ground.



The proper order for hooking up test devices to your tape machine.

This is a minor concern on ¼-inch machines but becomes a major problem on 1-inch and 2-inch decks. The problem increases at least tenfold. Once azimuth is adjusted continue on with the record mode adjustments, these are similar to the PB, so far as high- and low-frequency adjustments go. As you sweep the oscillator or change frequency keep checking the VTVM to be sure you are always feeding 0 VU into the recorder. Few oscillators are as flat as they should be. Also remember you are looking at PB, and whatever the PB did before the record should be doing this time. That means that if the test tape read +2 dB @ 15 kHz your record adjustment should read the same way. If the test tape read −1 dB @ 30 kHz so should your record. An exact match will rarely occur, but take note of the *difference*. The number of dB above or below is the key. If the record reading is 2 dB higher than the test tape reading then the record amplifier is at +2 at that point. If it is 1 dB below the test tape reading it is −1 dB at that point. You have to calculate each reading for your true record response. Now sweep the

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oscillator down the frequency range. You may note that from about 400 Hz on down your response has more curves than Raquel Welch, Charo and Dolly Parton put together. That's called "head bump"; unavoidable up to the present. It's another design problem that has to be overcome. Don't worry about it.

Maintenance Cont.

In addition to the electronic maintenance you also have to do mechanical maintenance on the transport. Your routine should go like this: every day, clean tape path and demag tape path; every week, check and tweak PB alignment; each month, check and adjust PB and record, lubricate motors and bearings; every six months, complete mechanical alignment. What is a mechanical alignment? A trip through hell in a wheelbarrow. If there is no way to adjust your mechanicals, do something nice for your mother-in-law to show your appreciation for small favors. But generally the following adjustments are to be made: supply reel holdback tension, take-up reel tension, pinch-roller tension, supply reel clockwise brake tension, supply reel CCW brake tension, take-up reel clockwise brake tension and take-up reel CCW brake tension. Turntable height. And on some machines edit brake tension for supply and take-up reels. You have to use the service manual for the specifics as each recording machine is different.

One last word. When cleaning your recorder you have to clean and demag everything in the tape path between the supply reel and the take-up reel. You can use toluole, xylene and alcohol on the heads and guides. Use only alcohol on the pinch roller as the others will cause it to deteriorate. Also toluole and xylene may attack the cement holding the guides and head laminations on some machines. So the safest thing to use is alcohol. But if you do use it don't use rubbing alcohol. That has too much water in it. Get the 97% isopropyl available in drug stores.

If you follow the above procedures your recorder should remain in top shape for many years barring accidents or defects.

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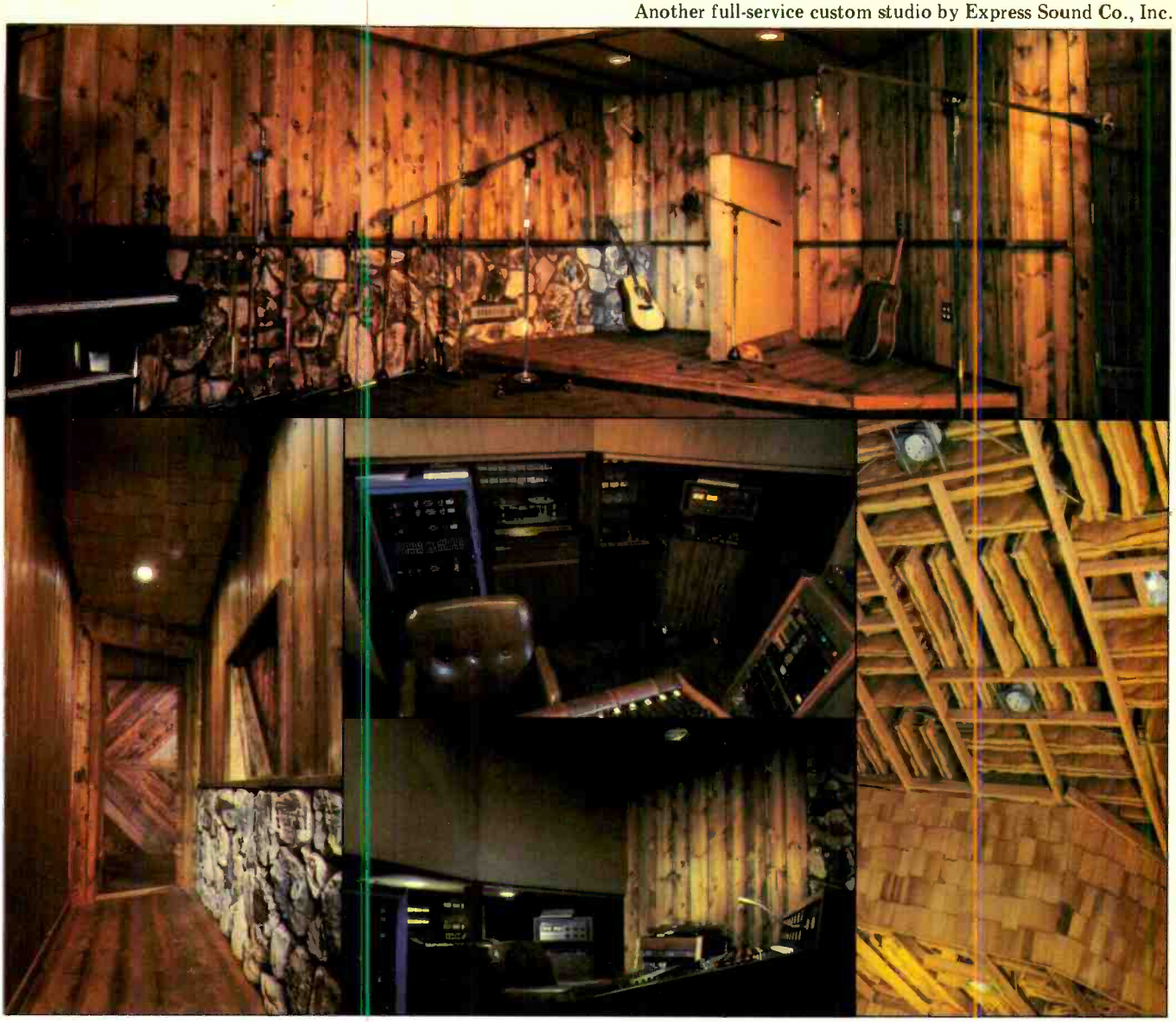

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A Session With Lou Rawls

Sigma Sound Studios' owner and chief engineer, Joe Tarsia, sits before the console in Studio 4 mixing tapes for the newest Lou Rawls album, *When You've Heard Lou, You've Heard It All*, the third album Lou has done in Philadelphia. Joe is describing the essence of the famous "Philly Sound" that has been the dominating force in Black music for almost ten years now, since the heyday of Motown in the sixties.

Joe explains, "The Philly Sound is a style of writing, playing and recording that, through a lot of trial and error, developed into a pattern different in many ways from the rest of the music industry. At the time we started Sigma Sound in 1968, there really wasn't a cohesive recording community in Philadelphia to draw from. It was almost like starting a language on a remote island without coming into contact very often with people who speak other languages.

"The roots of the Philly Sound, though, are basically rhythm and blues. Before us r&b was 'feel' music, *a la* James Brown. Gamble and Huff were the first ones to put [combine] class songs with r&b rhythms, and then combine that with lush string and horn arrangements."

Unique Sound

Kenny Gamble and Leon Huff, of course, are the songwriters, producers and founding members of Philadelphia International Records. They first gained recognition with their successful production of "Expressway To Your Heart" by the Soul Survivors in 1967, soon followed by their first gold record for the Intruders' "Cowboys To Girls" in 1968. They went on to work with Jerry Butler, Archie Bell and the Drells, the O'Jays, Billy Paul, MFSB, Harold Melvin and the Blue Notes and the Three Degrees, Today Gamble and Huff also produce Lou Rawls and Teddy Pendergrass.

And even though they were the first in Philadelphia to develop a unique Black sound, Gamble and Huff were not the first generation of hit-making Philadelphia producers. That began with the burgeoning popularity of Dick Clark's American Bandstand in 1956 and the creation of the Cameo/Parkway label. The Philadelphia stars of those early days were Frankie Avalon, Fabian, Bobby Rydell, the Orlons, Chubby Checker and Dee Dee Sharp (now Mrs. Gamble).

But the first Philadelphia generation was dominated largely by pop sounds that faded long before Gamble and Huff began churning out their soul hits. Joe Tarsia says of Gamble and Huff, "They don't write songs. They write records. If you listen to the material for Lou Rawls, the O'Jays and Teddy Pendergrass, you'll find it's all very different. When Kenny and Leon sit down, they write with a particular artist in mind. I don't think Lou Rawls' "You'll Never Find" would fit Teddy Pendergrass. It's the difference between a ready-made suit and

By Stan Soocher

Photo courtesy of Phila. Inter. Records

(Left to right) Producers Leon Huff and Kenny Gamble.

Probably the most identifiable aspect of Sigma Sound recordings is the use of echo delay for the "slap-back" effect which is especially emphasized in the snare drum sound. The chambers are EMTs and rather than sending the echo directly from the output module to the chambers, they run it through a tape or digital delay. Although the method may not be totally unique, the continual recurrence of the sound in the mix is special to the Philly Sound. A good example is "Me and Mrs. Jones" by Billy Paul, an extremely "wet" record.

As for equalization, Joe Tarsia says, "There are no standard EQs here because instruments and players vary so much from time to time. What works well one day may not work well the next. We also try to hold lead vocal equalization to a minimum. After all, a vocalist will vary according to whether it's morning, afternoon or night. And whether they've been working or haven't been working can really change the timbre of an artist's voice.

a tailor-made one."

Joe considers Gamble and Huff's involvement as songwriters and producers to be the key element to the Philly Sound. "Generally," he says, "the sound of our records is mostly determined by how it's produced and orchestrated, not necessarily the recording equipment."

Sigma's Selections

Nevertheless, careful attention has been paid to the selection of equipment and there are specific identifiable techniques particular to Sigma Sound. The console itself is "the world's standard stock model," a 24-track MCI with an Allison MLH outboard for computerized mixing which has been operable for the last three years. All tape machines are 3M, and 3M 206 tape at 15 ips with Dolby is used for most sessions. However, "Stevie Wonder was in last week and he records at 30 ips *with* Dolby."

Over at the newest addition to Sigma Sound, a 24-track room in New York, there is a specially designed board with electronically controlled faders. Automation is built directly into the modules, and instead of the usual plastic faders, an inset band is moved up and down producing a light-beam display that indicates gain setting. According to New York chief engineer Gerry Block, although it is easier to operate, the performance of the New York board is the same as the consoles in Philadelphia.

Block and the other Sigma engineers believe that the sound quality of a record is mainly influenced by the speakers the music is mixed on. Both Philadelphia and New York Sigma Sound Studios rely on Altec 604 "Big Reds," KLH 17s, and a 4-inch Panasonic Portable AM/FM speaker. Other indispensable studio accessories include Allison Dual Limiters and Kepex, an Eventide flanger, an Orban-Parasound Sibilance Controller, stereo parametric equalizer, stereo limiter, MXR stereo flanger and a Marshall Time Modulator.

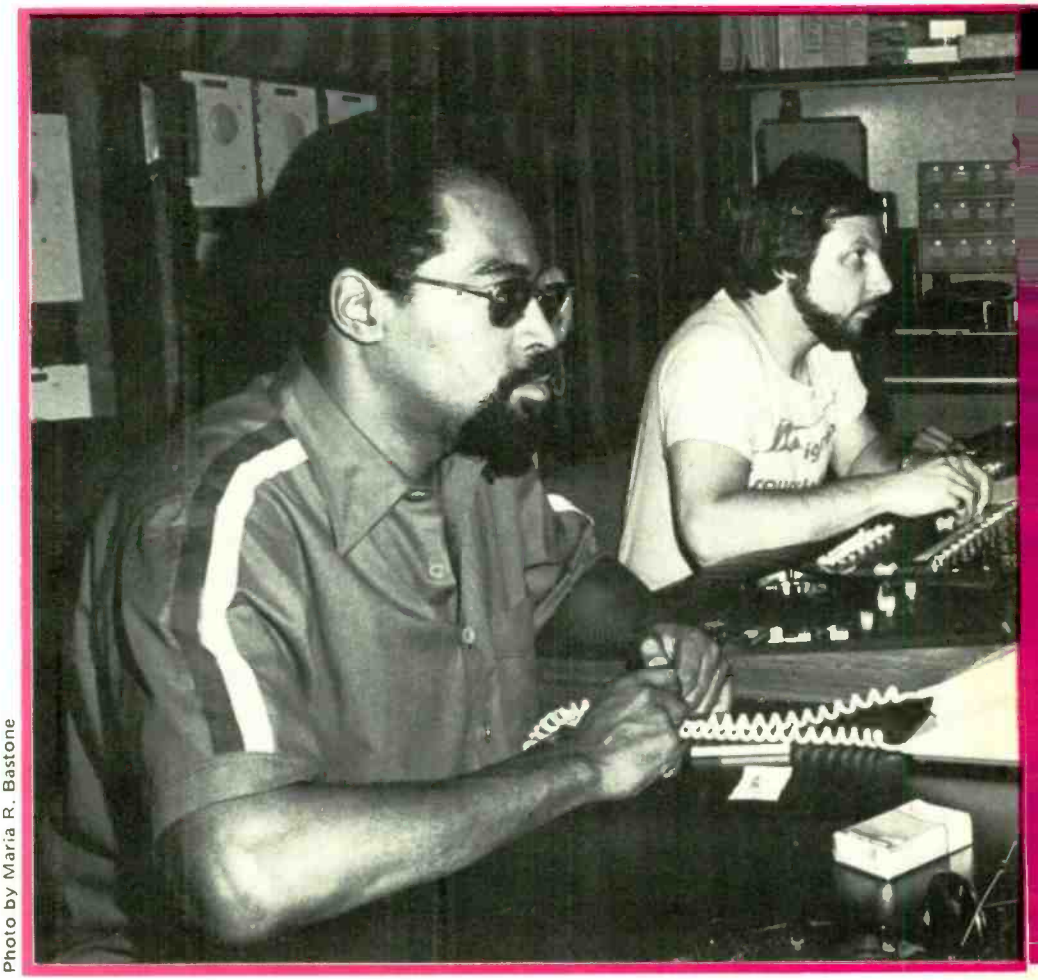
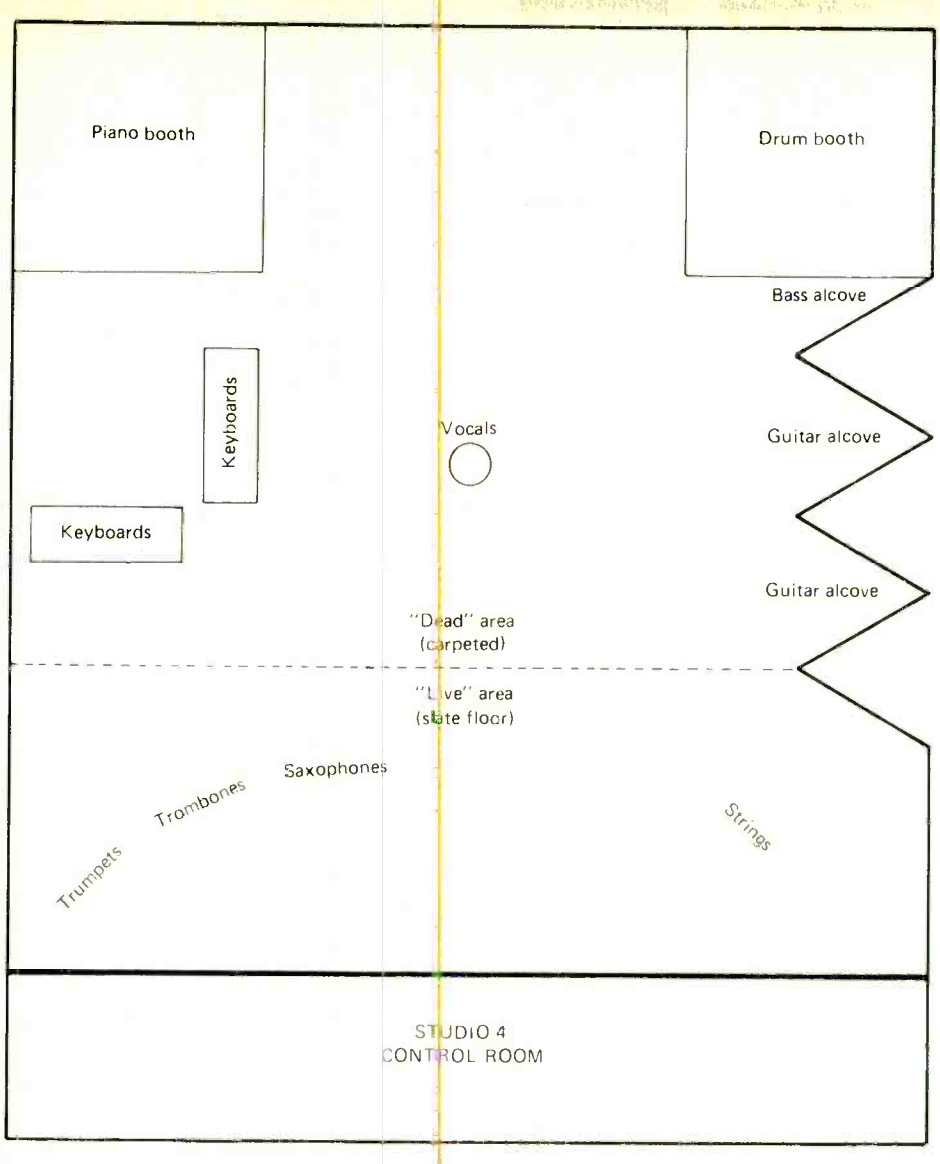


Photo by Maria R. Bastone

Producer Kenny Gamble (left) and engineer Jay Mark.

The specifications of Sigma Sound have attracted the likes of Elton John and David Bowie. Songwriter/producer Thom Bell, the third important member of the Gamble/Huff team, is also based in Philadelphia while maintaining offices in Seattle, Washington. He has produced, among others, the Stylistics and the Spinners.



show. A recording contract with Capitol followed; it lasted nearly ten years.

Lou's first hit records were "Tobacco Road" and "Love Is A Hurtin' Thing" in 1966, leading eventually to two Grammy Awards for albums and five Grammy nominations. In 1975, when he became associated with Gamble and Huff, Lou recorded his first platinum album, *All Things In Time*. Two weeks before the mixdown session with Joe Tarsia, we attended a vocal session with Lou in Studio 4 and he explained why he is working with Gamble and Huff.

"Gamble and Huff give me the freedom to select the material I want to do. In the past, some of the producers I've encountered have handed me a stack of music and told me, 'This is what we're going to do.' I've had to adapt myself to the material as opposed to having them adapt the material to me, which is, I think, a mistake for an artist. Gamble and Huff let me choose from a group of songs and then live with the material for a while. Then I get with an arranger —either Bobby Martin or Jack Faith—at the piano and we work out a key for the song. Afterwords, an arrangement which will not be all over the vocalist but will enhance the impact of the song is written around that.

"About recording methods, though, like a lot of other artists I'm not into the technicalities. I'm concentrating on taking the song and making it hap-

(Below) Sigma Sound's owner and chief engineer, Joe Tarsia, at the console.

Photo by Maria R. Bastone

We never like to punch-in on an existing vocal track on a different day because there is that much difference in the voice.

"As for recording Lou Rawls' voice, the sound of it is pretty much up to me. I've worked with Gamble and Huff for over ten years and we really don't have to say very much to each other. Lou, of course, gives you an awful lot to work with vocally. He has an extremely broad, very natural voice. You can put singers into two categories: those who have great styles and those who have great voices. Fortunately, Lou has both. When it gets down to engineering, you don't have to generate a sound at the board because Lou gives it to you at the microphone. I usually just cut Lou's voice basically flat with very little EQ."

On previous albums done in Philadelphia, Lou's vocals were recorded with a Neumann U-87. For the album, a U-47 is utilized because it captures the bottom end of Lou's voice with truer quality than the U-87.

Capsule History

Lou Rawls first came to record in Philadelphia two years ago after a distinguished career that encompassed the recording of close to thirty albums. Born in Chicago, he made his singing debut in a church choir at the age of seven and a half with the late Sam Cooke. In the fifties he moved to Los Angeles where he met Herb Alpert and Lou Adler, who were a struggling songwriting team at the time, and together they cut a forty-five released locally in L.A. When Dick Clark was on the West Coast producing a Bandstand showcase, he hired Lou for the

pen in the delivery. There are certain things I can and cannot do with my voice which the public will accept."

Group Participation

Although Gamble and Huff are the main songwriters and producers for the Philly Sound, a group of songwriters, arrangers and co-producers are assigned to each project. Because Sigma's Studio 4 is housed in the same building as Philadelphia International Records, Kenny and Leon will be in and out of the studio (there are also two "live" Sigma Studios located on 12th Street in Philadelphia).

During the recording of the rhythm tracks, Gamble and Huff stay inside the studio with the session players who appear on most of the Philly Sound records. They are Roland Chambers and Dennis Harris on guitars, Charles Collins on drums, Michael Foreman on bass, and Larry Washington on percussion.

Like Lou Rawls, Gamble and Huff are not technically-oriented. Their role in the studio is to help the musicians set a groove. They come in with a general direction for a song, but honing and refinement is done on the spot. Gamble and Huff look for creativity among the players so that the musicians can assume an active role in the production. Lou Rawls also attends the basic sessions to get the feel of each song. Kenny Gamble's most famous quote may be: "Music is the only natural science known to man." All technicalities, therefore, are left up to the engineer.

As Kenny Gamble explains, "Leon and I feel there is so much involved with each project that it's better to let everyone participate in some way in the creative process, and then make final decisions with the artists in mind. We leave the vocal performance up to Lou, for instance, because he will give us exactly what's necessary for each song to bring out the strength of the words and the music." Leon Huff, the keyboard player, is the quiet half of the team who, rarely speaking, moves among the musicians like some soul-injected hobbit.

The typical order for recording tracks is drums, bass, guitar, keyboards and percussion for the rhythm, then background vocals, lead vocals, strings and horns, depending upon the nature of the material. Occasionally, some percussion is added at the end, and the string and horn sweetening is specifically written around the lead vocal track.

Joe Tarsia says of the often admired string section, "The players make the Philly String Sound special. You'll notice no one is reading the newspaper during a rundown. Everybody digs in and plays to give it all they've got; that's the reason why the strings always come off biting."

Smooth Takes

For the first song Lou sings during the vocal session, "Lady Love," arranger Jack Faith is the co-producer. Jay Mark, who has worked with Lou before, is the engineer. Lou records reading from a lead sheet in the center of the studio. Four relatively smooth takes are run through in an hour. Take three is chosen as the best.

During the next song, "House Full of Flowers,"—which does not appear on the completed album—Kenny Gamble comes in to produce. Most of his attention is given to anticipating Lou's vocal delivery with little notice given to the board. Jay Mark continues as engineer, and only three takes are needed till everyone is satisfied.

After a fifteen minute break, vocal recording continues on "That Would Do It For Me" and "I Coulda, Woulda, Shoulda." Within two hours, the vocal session is over. After all the tracks are recorded, Kenny, Leon and Lou must give the okay before mixing begins.

During the mixdown session Joe Tarsia points out, "Most people mix the same way—starting from the left end of the board with the rhythm tracks to get a tight sound. Then you work your way across to the strings and horns, bounce them out, lay the vocals on top of that and go back to the beginning and readjust. It's really not that difficult once you are familiar with a board. At that point, it's really up to the talents of the artists, producers and musicians that you have brought into the studio."

MIC SELECTION

Vocals:	Neumann U-47
Piano:	Two Neumann U-87s in stereo pair configuration
	One U-87 overhead
All other keyboards:	Direct
Guitars:	Shure M57
Bass Guitar:	Direct
Drums:	
Kick—RCA BK5	
Snare Top—Altec 633 ("Salt Shaker")	
Snare Bottom—Shure M57	
Hi-hat—AKG 451	
Floor toms—Neumann U-86	
Toms—EV Re15s	
Overheads—Two AKG 451s	
Congas:	AKG 84
Strings:	
Six Violins	Four U-87s in cardioid position
Two Violas	One U-87 in figure-8 pattern
One Cello	U-47
Horns:	
Trumpets	One Beyer M-160
Trombones	RCA 77DX (individually miked)
Bass Trombone	U-47
Saxophones	U-87 in cardioid position (individually miked)

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ARTISTS' PROFILE

FRANK ZAPPA

By Joseph Laux

Frank Zappa has been involved with recording as an artist, engineer and a producer since the early sixties. Zappa is known as a recording and musical innovator with a somewhat unorthodox manner, and first came to the greater public's attention through the work of his group "The Mothers of Invention."

This interview was done in the studio while Zappa was recording his upcoming album, Leather, *and begins with a discussion of Paul Buff—creator of several recording products issued under the aegis of Allison Research—who was a major influence on Frank Zappa's early recording career. As you'll see, from that point on, it was definitely Zappa's trip.*

MR: Since we know you and Paul Buff worked together for a while in the old days, would you tell us about this relationship?

FZ: I met Paul when he had a studio in Cucamonga, California. The equipment was very basic, most of it was homemade. He had learned his electronic skills in the Marine Corps and he came out here to decide if he wanted to go into the record business. He taught himself how to play sax, piano, bass, guitar and drums. He also taught himself how to sing, and built the studio so he could record himself, or any of the local bands who were in town at that time. He only charged about $30 per hour. Two surfin' hits, "Pipeline" and "Why About," were recorded there and the master was released to either Dot or to another record company.

Most of my basic training in recording came from working with Paul. I started playing on some of the demos that he was cutting there at the studio and got interested in the equipment. When we would have dinner at the bowling alley we used to eat at, he would draw diagrams on the napkins of how the equalizer worked. He gave me a lot of basic information on how things work and even in 1963 he was working on developing some of the inventions which are on the market today, through Allison Research. I think not too long after 1963 he built the prototype of the Gain Brain, and the first Fuzz Tone that I ever saw, by overdriving a phonograph pre-amp. He built his own equalizers for his little console that we had there at Cucamonga. He also built a five-track tape recorder with staggered heads because they didn't have in-line heads. The record head was also the playback head and the machine had its own erase head and we used half-inch tape.

The other equipment in the studio consisted of about five or six Shure microphones, and a homemade condenser mic. We had two pianos, a Steinway upright and some type of Baby Grand. I can't remember the monitor system in the control room but I gathered there were a few Hi-Fi speakers from the boxes I saw. It was basic and good enough for rock and roll. We could overdub on that machine and could ping-pong on it. Allison, the girl who's picture is on the Kepex, used to be the waitress at Cucamonga Moltra [we'll assume the Moltra was/is a local restaurant]. Paul met her and fell in love and they moved into a trailer together. He produced a bunch of records where Allison sings. She is a very good singer and has a nice voice so she did a lot of recording out there at Cucamonga, but I don't know if she has recorded since then.

I think Paul is definitely one of the pioneers of the modern multi-track

recording industry. He's a genius.

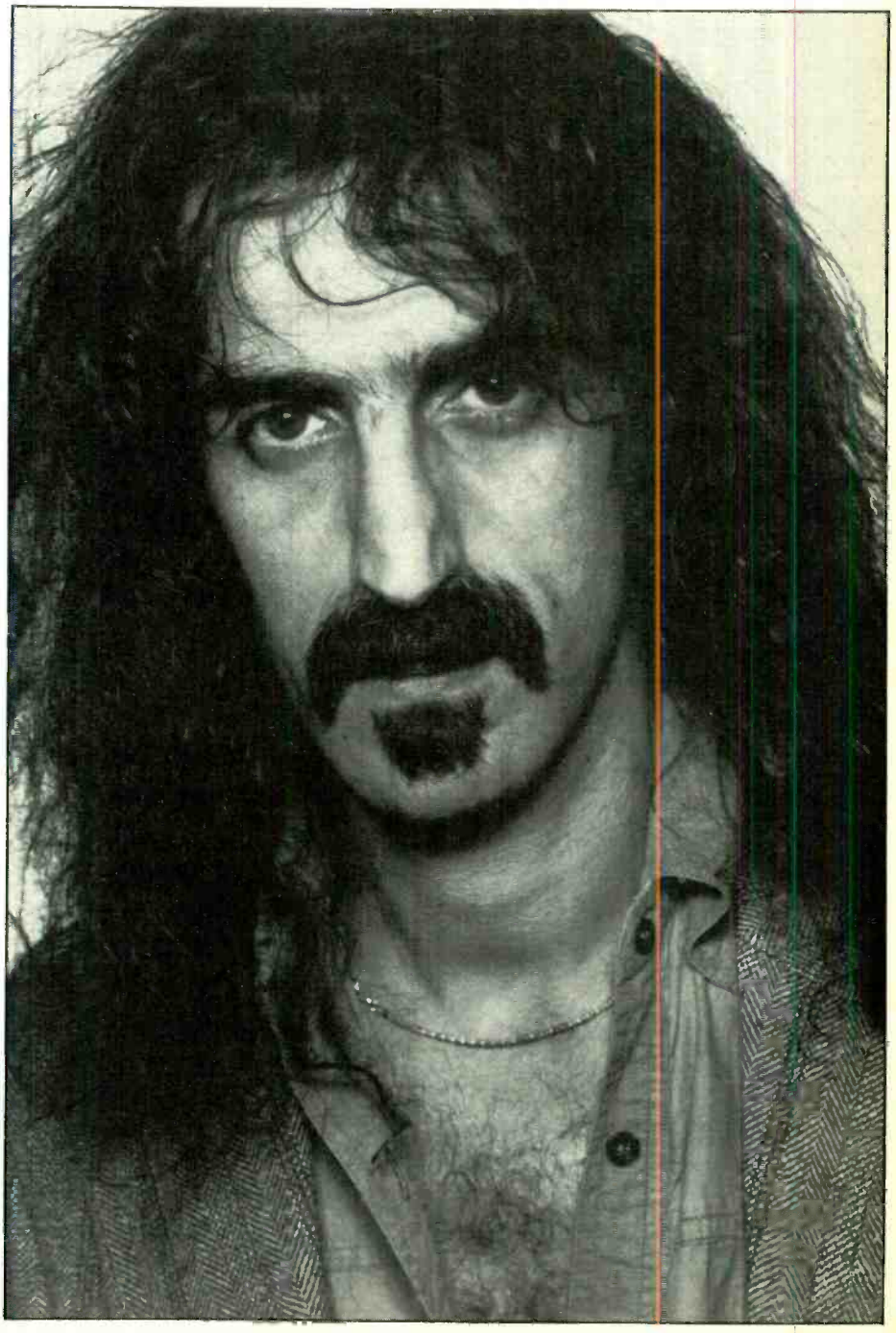
MR: Were there any other people who influenced you as a producer or engineer?

FZ: No. What happened was that I eventually bought the studio from Paul, so after he drew the diagrams on the napkins, everything else I learned was just from making mistakes on my own. I had that studio for about nine months, 11 or 12 hours per day to just record.

MR: Do you have any favorite studio at the moment?

FZ: I like Caribou, except for the fact it is 8000 feet up. If it was at sea level it would be a fantastic studio for my purposes, but so many funny things have happened to people and musical instruments when the air is so thin. The distance between the air molecules affects the sound of the instruments and the acoustics, and also creates other problems for monitoring.

Right now I think my favorite studio is Kendun, because of the way it's laid out and the equipment. I like the idea that you can cut right away so you can keep track of your progress. Unfortunately, the disc-cutting facilities over there are booked up so much. However, it is an ideal situation—you have disc-cutting facilities while you are working on a project. A lot of people when making



an album forget how it's going to be in the end, which is on a piece of plastic in somebody's home. So it can be very deceptive when you are working on a project two or three months only hearing it over the big monitors which are coming up from the tape with a dynamic range. So it's nice to keep tracking to make reference during the project.

It would be great to make a direct-to-disc recording. I think that would be a good way to record jazz or classical music. You get a good string quartet and bring them in and put them in direct-to-disc. It would be perfect because they usually know exactly what they are doing before they start up.

MR: Let's suppose you are the only engineer in a totally unfamiliar studio where they have every piece of hardware ever made; what would you start with?

FZ: First of all, if I'm going to a studio which is completely unfamiliar, they won't just turn it over to me without some kind of assistance because they have a financial investment to protect. If I was God they probably still wouldn't let me walk in and just run everything. From studio to studio the problems are what they have plugged into the patch bay, so just for that reason alone you need some kind of assistance. All the recording I do now is 24 track and all the tapes run at 30 ips. I have been recording everything with Dolby but I think that the new Telefunken system is much better. As far as microphones go, I have different favorites for different instruments. I use U-87s, Sony C-37s, AKG 414s. I like to use shotgun mics for the piano to get certain types of effects. For kick drum I like an RE20 or two U-47s. It depends on what style the drummer has, whether he uses heavy footwork, or is a studio-trained drummer that plays light and gets that "fat" sound. You have to suit the microphones not only to the instrument that you're recording, but also to the way in which the musician plays the instrument.

MR: When you are recording is the bass an amplified channel, or miked?

FZ: I usually take the bass direct—very solid master amplification on the bass. If you want to get an amplified sound you can always play it back to the cue system and the little amp [a Pignose Inds. amp] when you are mixing. It saves you an extra channel during the recording. That's the decision you have to make right there on the spot. If you lock yourself into the proportions of mics versus direct you may regret it later, so it is better to go on with direct, getting a little compression on it so that it sustains the right level all over and if you want to get more rock sound on it, just play through an amp. Play back to the Pignose, when you are mixing.

MR: Do you try to make a record as good as the one before or do you try something different everytime?

FZ: I am trying something different everytime, because everytime I go to the studio and spend that money for studio time it is not only the opportunity to make another album but it's an opportunity to experiment with all the new things that have come out that year.

MR: For this latest album, were there any tunes actually written in the studio?

FZ: Most recently, everything has been prepared before we go to the studio.

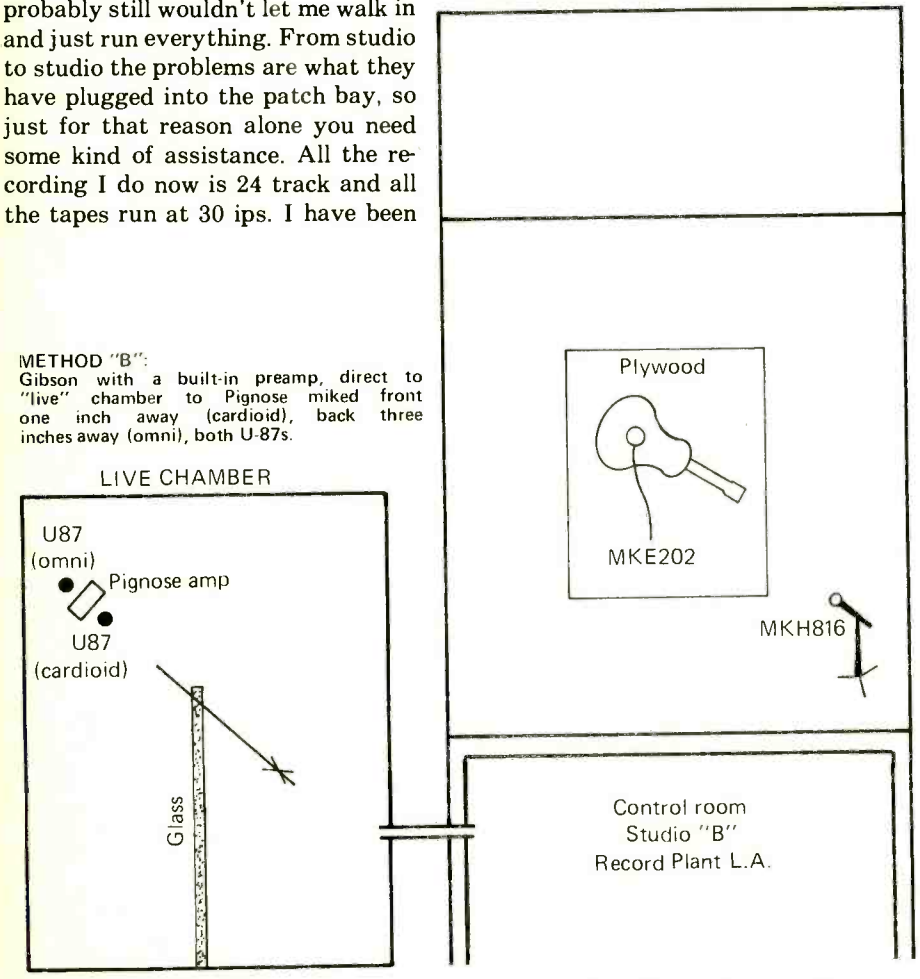
MR: After seeing the huge tape library in your house, I'd like to know if you record every concert on the road?

FZ: What we normally do is tour about six months out of the year and I take a four-track machine and just record about every show that we're doing. Even if it's the same song. So if we play 100 concerts in a year, that's 100 takes which is a better average than what you would get in the studio. So we come up with some very good things that way.

I also like recording different parts of the four-track in different kinds of environments. For example, we take a "live" track and get a lot of drive from a rock performance where the audience's enthusiasm is there. It's possible, if you are miking right, that you will never even know that it was a "live" track because you won't hear the audience. If you're taking care during recording you'll come up with a basic track that's got more power and spontaneity than you are going to get in the studio. So bring that back to the studio and do your vocals under studio conditions or add any sweeteners or electronic effects and you've made a high-breed kind of an album where the drive is there but the sound quality is better.

METHOD "A":
Acoustic guitar was miked with an MKE 202 on the body and an MKH 816 ten feet away pointed down the neck at strings.

METHOD "B":
Gibson with a built-in preamp, direct to "live" chamber to Pignose miked front one inch away (cardioid), back three inches away (omni), both U-87s.



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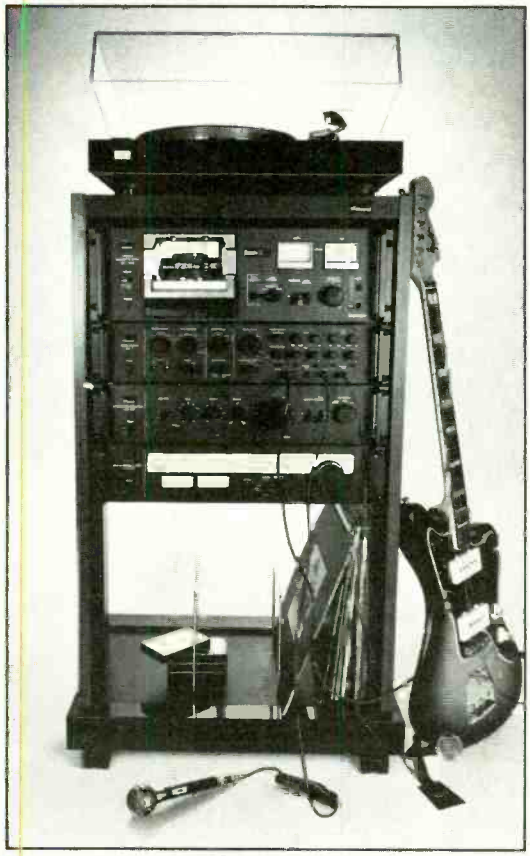

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CIRCLE 91 ON READER SERVICE CARD

MR: Which remote truck do you most frequently use?

FZ: I've used many different remote trucks, the most recent one was from Manor in London. In Los Angeles we've used the Wally Heider truck, which is very good, and we've also used the Record Plant truck. On the East Coast we've used FEDCO.

MR: Could you describe some of the problems that have occurred during "live" recording?

FZ: The first problem that we have is finding all the groundloops. You can be chasing those for hours. Another thing that you have to do if you're on stage, playing—it means you can't be in the truck—is to put a lot of confidence in the engineer. Hopefully, he can get the right sound. The basic problem for a remote truck is the monitoring system. Some guys may monitor through 4311s. It makes you scared, because the JBL 4311s are not the ideal speakers to mix through. You may come back with a tape without top end and too much bottom or vice versa.

MR: Are the things you were doing in the old days at the studio radically different from what you do now?

FZ: When I first started recording every place in town was four track. By the time my third album came around they were just getting into eight track. It was in New York about the fourth album they were using twelve track, and about the fifth and sixth album, sixteen track was coming into existence, and later twenty-four track. The things that you had to do in the studio in the old days versus the things you have to do today are vastly different. Some techniques don't change, and some change very radically. One thing that I'm really insistent about is making sure that there is a tune-up tone on the head of the first reel of the multitrack master. That way, for overdubs, you can always tune up by recording an 8440 cycle reference tone using the fix pitch instrument of your group. For instance, Fender Rhodes or Hammond organ, or in our case it would be marimba. You record that "A" on tape and everyday before you overdub, you tune your strobe tuner to that "A" and everybody stays in tune with it. I think it's very useful to take time to print that "A."

MR: Could you mention some of your favorite studio hardware?

FZ: My favorite limiter is the dbx stereo limiter. If you want to mash large portions of the track, you can use the dbxs for individual instruments. I like the UREI LA-3As and the Burwen Dynamic noise filter. I also like the Neumann stereo expander or the one by Neve. I have an outboard rack so we go on the road with: 1 dbx stereo limiter; 2 LA-3As; 1 Spectra Sonics limiter; 6 White graphics and a White third-octave spectrum analyzer; 2 Gain Brains; and 2 Kepexes. That's our basic outboard gear just for doing a concert.

For studio the basic set up would be: one 24-track; one 4-track delay; two 2-tracks; one Harmonizer; and four dbxs.

MR: What do you want to accomplish or avoid when making an album?

FZ: You learn the hard way. On most of the early albums I made I think the EQ was bad. I think there was hardly enough on top and there were a lot of things wrong, but they were being made with the equipment which was the best you could get at the time. By listening to those things now I can learn from those mistakes, hearing it back and finding out how to change it.

Another thing a person has to know if he wants to have control of his composition is not only to understand what's happening in the studio with the studio equipment but also to find out about disc cutting. You can know all about getting your tones right on the tape recorder and getting the azimuth right, making sure everything is at the right levels and eventually come up with this fantastic tape, but if you're not careful when you put it onto a disc, it is still going to sound horrible and you won't know what went wrong. So you have to go to the disc-cutting place where they are putting your music on record.

MR: Let's say you now have a great master disc, don't you still have to watch how they are going to press it?

FZ: You are absolutely right. You can never be 100%. Even if you sit there all the way through the pressings, personally play each one to see if it is perfect, place each one in the wrapper and box, make sure they didn't warp and then make sure they got to the store right. Then the biggest problem of all happens because you will never know what kind of machine it's going to be played on. So you can't be completely sure what's happening, but if you aren't as careful as you can be all the way, you are really in trouble.

MR: Who is your favorite mastering engineer?

FZ: I use John Golden at Kendun. The last album was done at Amego with Ernie Costa, because I couldn't get into Kendun, but I think Ernie did a fantastic job on it. Ernie also mastered two of the earlier albums: *Grand Wazoo* and *Waka Ju Waka*.

MR: How would you design an album in terms of level and time?

FZ: One of the things you have to think about when you are sequencing an album is what kind of song you're going to end each side with because the tone quality is going to suffer on that song. It's always better to go with something that is softer as the last song on each side, because if you want to save your hottest number until the last cut on an album, it's not going to sound as loud. It's hard to keep the level if the side goes over 18 minutes.

I'm trying to put the most powerful cuts on the outside of the groove so I have all the top and bottom left, but sometimes I can't do that.

MR: Did you use any special effects on your upcoming album, *Leather*?

FZ: We used a Harmonizer on the bass. The "live" portion of the album was done partly in London and partly in New York. So where we have Harmonizer on bass, it was split left and right, which gave us a real nice sound on the bass. It's a different sound and a difficult sound to get on disc, because of the phasing problem. The Harmonizer is used on the guitar quite a bit and I'm using the MXR flanger on a few of the guitar sounds.

There are also a number of overdub guitar solos on that album that we have done with just a Pignose amplifier. In fact, I have used a Pignose extensively on the last five albums. It's a very reliable sound source for the studio. It take very little time to set up and you know what the tone is going to be. We've put the Pignose in a "live" chamber to simulate "live" sound and to get a really large sound. Instead of just mixing it and sending the mic signal into the echo chamber, you can get a completely different effect by actually having the amplifier in the chamber, and sending the guitar through the echo send into the Pignose and picking it up with the

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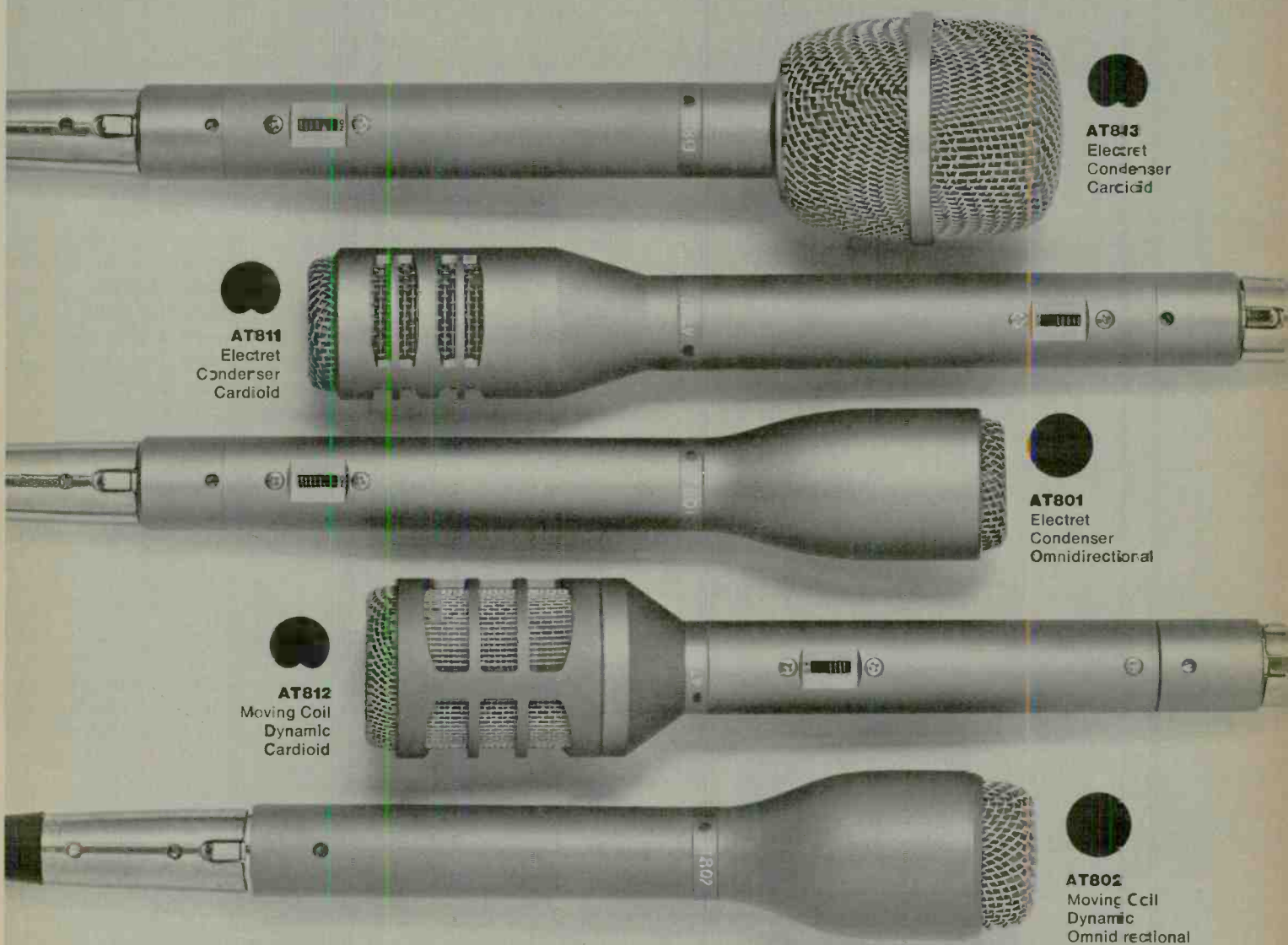

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microphone in the chamber room.

MR: Listening to your albums I get a feeling of strong positioning [in the stereo spectrum]. How do you avoid leakage from channel to channel?

FZ: Lots of times if they set the crossover on the disc cutter at too high frequencies, the result will almost be like mono. So you try to avoid using high frequencies as much as possible. This helps your stereo spread.

I have recorded drums two different ways. Most of the drums' actions are from tom-tom passes. If you record each tom-tom mic individually, instead of an artificial panning situation you'll get a certain type of movement. In some songs that movement can be a little bit too harsh, so I use a C-24 overhead and I'll get natural stereo spread.

One way that you can give the illusion of the stereo spread being wider than it actually is is to set it [the spread] up at the intro of a song. I'm going to try this, by the way. On my next album the drums have a certain type of stereo action throughout. You set them in the middle of the spectrum and if the guy [drummer] is going to cross, the tom-tom is still going to pan. As soon as the rest of the band comes in, you can do other things, like setting them a little bit further out from the mid. Then when everything comes in full blast, you bring in two pieces of information—like mono tracks—exactly left and right, and then for a period the whole stereo spectrum just spreads out.

Another way to make things sound wider than they are is by taking an instrument like the marimba, a scale-type source—physically a long instrument—and putting two C-37s looking down, as far as you can, to cover the instrument's keyboard. When the musician plays most of the information right underneath the mics, all he's got is one note on the top and one note on the bottom and suddenly you find out how big that instrument is. And it's a really great effect.

A very nice effect I've used that won't be coming out until the next album is to have vibes and marimba—a mono track of each—playing notes that are rolled, put the tracks just slightly in from left to right and double each of those tracks. There are some mono tracks of woodblocks playing the melody, so the woodblocks match up exactly on each stroke of the vibes and marimba and give us a rattling texture that's great, especially in earphones. It's fantastic.

MR: During mixing, for playback, do you use the reference tones exactly as they naturally exist, or do you change the top and the low ends?

FZ: One thing that I've been discovering since I've got the new Ampex tape recorder is how much difference there is in the quality of the playback—as far as playing around with the recording reference tones. For example, people usually set the tape up to the tones exactly right and then what they do is play with the equalizers in order to get some top end out of it. However, if you take your top end reference tones and set them 2 dB hotter it's fantastic. You let the machine work for you and you don't have to incorporate the additional noise of any outboard EQ in order to get that extra brilliance. Although, some of that eventually is going to be buffed away when they make the disc. You might even tweak the low end a half dB, and leave the middle where it is.

UNI-SYNC INC

LIVE MUSIC

A JOURNAL PUBLISHED BY UNI-SYNC, INC., THE PROFESSIONAL PRODUCTS DIVISION OF BSR.

Deciding on a Mixer

The variety of mixing systems on the market makes it difficult to judge which is best for you. Features blend together forming a mirage of switches, controls and pots; all looking alike. Therefore, we've come up with the *sound artist's guide to mixing buys*. It will give you independence when it comes to buying a mixer. The following is an abbreviated version of *the guide*, which you can send for free of charge. We hope it serves you well and would appreciate hearing from you.

The Sound Artist's Guide to Mixing Buys

What is a sound artist? We've come to realize the existence of a new category of performer. This is the person who creates, through the use of various tools a sound that appeals to the audience: therefore the sound artist. How well the information communicates is left to the talented ears of this individual, and the manner in which these tools are utilized. The sound artist, with today's technology, has become an instrumental part of the performance.

Mixer is the Basic Tool

The basic tool of the sound artist is the mixing board; with it he can create a myriad of sounds from his fingertips. It not only has to sound good, be reliable and versatile, as well as having excellent specifications, but has to have just the right touch. For the person mixing relies upon his hands as much as he does his ears.

DESIGNED ON THE ROAD

It is for this reason that Uni-Sync, in designing the Trouper Series is extremely innovative in the choice of front panel controls. Michael V. Ragsdale, president of Uni-Sync as well as chief designer of the Trouper Series, has built, serviced and most of all, operated sound systems prior to forming Uni-Sync. The Trouper Series was designed on the road, the true proving ground of sound reinforcement equipment, from a practical viewpoint. For example, the use of slide faders as opposed to rotary pots was an extremely important decision based on ease of operation as well as visual and tactile indication of position. Rotary pots are hard to read under the dimly lit conditions of sound reinforcement, whereas a slide pot gives instantaneous recognition.

Live or Recording

One of the most important decisions to make is where your mixer will see the most use. Is it for a "Home Studio" or to be run live. Live boards have different gain and level structures. In a studio, you are dealing with a controlled environment, but live sound is just that; live and wide open. To handle that kind of sound you need to have a mixer that has been specifically designed for that purpose. This is the Trouper Series, designed and built for the road or permanent installation for mixing live sound...it is a live music mixing system.

Inputs You Can't Grow Out Of

Next on our list of important decisions, is to determine how many and what type of inputs you need. Mixers come in various configurations, application determines the need. The Trouper I, for example, has on each channel: low Z balanced and high Z inputs, and an in/out jack. This allows for maximum flexibility.

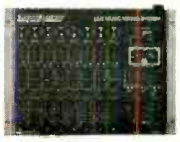
Now, how many inputs do you need? Most mixers come in fixed quantities; for example, six, twelve, or sixteen. Once you grow out of it, you have to buy a new board. Not so with the Trouper Series. The basic mixer is an eight input/output control module that is expandable through the addition of a ten input expander module, that simply plugs in. You never grow out of a Trouper.

Build Yourself a Custom Board

If you had the freedom or ability to build a mixing board perfectly suited for your needs, what would you put in it, how big would it be? The Trouper Series gives you this freedom at an affordable price. Our mixers are big boards in little packages, giving you the opportunity to custom design a system that is tailored for your specific needs. You build what you want, not what someone else thinks you need.

Dollars Per Input

An excellent way of determining the value of the mixing board being considered, is to divide the cost of the board by its total number of inputs. This gives you an objective analysis of the mixer, and by comparing and contrasting features per dollars, you can arrive at a decision. For instance, the mixer at $100 per input may have far greater features than the one at $85, and would be a more valuable purchase.



Mono or Stereo

The Mono/Stereo issue is one of the most controversial at hand today in the retail sales of mixers. Most installations and gigs are best handled in Mono. But many groups today, want the added flexibility of a Stereo board. We are presently introducing the Trouper I Stereo, which is probably the most flexible and versatile mixer on the market for its price. At $898 (suggested pro net), each channel features a house pan pot along with an echo pan pot enabling you to pan the echo to or away from the house signal. A little imagination can create some very interesting effects.

The choice for Mono or Stereo is based on budget and application. Practically speaking, Mono will satisfy most of your needs.

Send For Your Free Guide

That's the abbreviated version. If you'd like the complete guide fill out the attached coupon and send it in to us right away. You may want to get some of the other Trouper Series goodies like T-shirts or director's chairs for a comfortable place to mix from. We're looking forward to hearing from you.

Thanks,
Larry Jaffe
Marketing Manager

Dear Larry.

I'm interested in what you have to say. Why don't you send me a copy of SOUND ARTIST'S GUIDE TO MIXING BUYS. While you're at it send the goodies I've checked off.

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CIRCLE 51 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

4-4-4, 4-3-4 or 4-2-4

No, those aren't safe combinations! They are abbreviations for the various methods of broadcasting 4-channel sound over FM. To paraphrase the great American humorist, Mark Twain—"Reports of the death of quadraphonic sound are very much exaggerated." True, in the last year or so there have been singularly few 4-channel records released in *any* of the once popular formats (CD-4 discrete, SQ matrix or QS matrix) and, if you walk into an audio shop these days asking for a quadraphonic receiver or amplifier the salesman will probably look at you as if you are out of your mind. But there are things happening behind the scenes—important things which could have a profound effect upon the recording industry, the broadcast industry and all of our relationship to those industries. First, a bit of background information.

Way back in May, 1972 the Electronics Industry Association (EIA) formed a committee known as NQRC (National Quadraphonic Radio Committee) for the purpose of evaluating and field testing various proposed systems that would permit discrete broadcasting of four program channels over a single FM station. Five systems were evaluated and field tested and the report of the committee was submitted to the FCC in late 1975. All of the systems are somewhat similar in that they all require additional sub-carriers, over and above the single 38-kHz carrier now used in conjunction with stereophonic FM broadcasting. All of them provide for compatibility for stereo as well as mono listeners. In other words, the four channels of programming are mixed together in such a way that a mono listener hears all four, mixed together, over his or her mono FM set while a stereo listener, equipped with a present-day stereo tuner or receiver, hears the left-front plus left-back signals out of the left front speaker and the right-front plus right-back signals mixed together out of the right-front speaker in a stereo array. Such compatible systems are given the general name of 4-4-4 systems, because the four signals are added and subtracted to each other to create four new mixed signals, which are then broadcast in four-signal form (using one of the proposed multi-sub-carrier techniques) and then, at the receiving end, they are unscrambled into the four original signals for reproduction over the four loudspeakers of a quadraphonic setup.

One of the systems proposed also makes provision for an alternate, 4-3-4 approach. In that system (an alternate proposed by RCA), separation from any channel to any other turns out to be around 10 dB (better than simple "matrix" separation that doesn't use logic enhancement circuitry but not as good as "fully discrete" 4-4-4 systems). This alternate proposal has the advantage that presently transmitted SCA (background music, private subscriber services transmitted over a 67 kHz sub-carrier and not heard on home receivers) doesn't have to be moved to a new sub-carrier frequency. So much for the discrete and semi-discrete four-channel broadcast systems.

Now, as you may know, it is also possible to broadcast matrix-encoded discs over the air right now—with no FM rule changes at all. Since a matrix disc (be it SQ, QS or any other kind of matrix) is already "encoded" into a two-channel format all a station has to do is *play it* over the air, and if you, as a listener, are equipped with a proper decoder (for that particular matrix system) and the necessary extra stereo amplifier and speakers you can listen to the program in four-channel form. But, as we shall see in a minute, this is where a hitch comes in. If you have a super-sophisticated decoder (and some really good ones for SQ and QS have been developed) you will hear 4-channel sound that is virtually indistinguishable from its discrete counterpart. If, however, you use a minimal-type of matrix decoder, the quadraphonic reproduction will lack separation and proper imaging and will be less than terrific.

The Problem

Now, some time ago, CBS petitioned the FCC to issue *standards* for 4-2-4 broadcasting (you can now figure out how that abbreviation comes about) which would center around their SQ matrix system, exclud-

ing any other possible matrix approach and, for the moment, excluding any need for a discrete 4-4-4 broadcast system. Since the FCC really had no extensive information regarding the subjective merits of 4-4-4 versus 4-2-4 (from a listener's point of view) they put together a test program to gather just such information. Some thirty-two listeners from various backgrounds were subjected to many comparison tests (using the same material for each version of the tests) of 4-4-4, 4-3-4 and 4-2-4 matrix systems.

The results of these tests were made known in August of 1977. Indeed, discrete 4-channel (played from 4-channel tapes in this case, since the issue was not whether they could be broadcast but how they sounded compared to matrix systems) won out by a slim margin, but of all the matrix systems tested, the SQ system (by CBS) was highly favored. It should be pointed out that the matrix boys used their very best available and most sophisticated logic-decoders for these listening tests as they might have been expected to do. So, in the opinion of CBS, the results supported their argument that *no* discrete system (4-4-4) of FM broadcasting is needed at this time and that the status quo should be maintained, since SQ records can already be broadcast (and often are) over FM. What this means is that the FCC is being asked to endorse *one* particular matrix system, to the exclusion of all others. Interestingly, at the present time *any* matrix disc can be played over stereo FM, but CBS feels that if more than one system is used, consumers will stay away from the whole 4-channel hardware scene because of the confusion that this raises. That's an argument that I can't disagree with, on the surface. But what I do believe is that there is a much better solution which, at first, seems to exclude matrixing but which in fact does nothing of the kind.

We Need a 4-4-4 Standard!

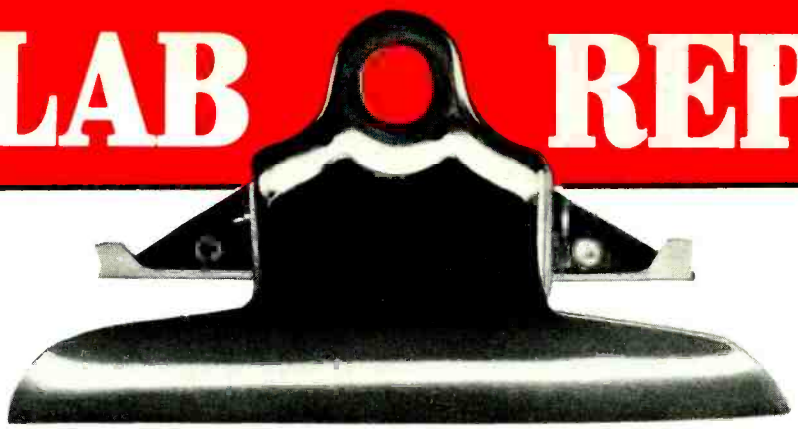
Here's my reasoning. If the FCC were to now approve a 4-4-4 system (and I leave it up to them whether that should be the QSI, RCA, Cooper-UMX or Zenith proposed systems), the way would be left open for further technological improvements both in discrete and matrix source material. In addition, and this is most important, the public would not be penalized. Here's an example of what I mean. Suppose that a station likes a certain SQ disc, and wants to broadcast it in four channel. All they would have to do is buy a *single* super-terrific SQ decoder (like the one used in the FCC listening panel tests) and *decode* the disc right at the studio. The recovered four channels of the program would then be broadcast in decoded, discrete form, using the newly approved 4-4-4 standards that I am asking the FCC to approve.

The owner of a four-channel receiver or tuner (or the owners of stereo tuners who have connected 4-channel FM decoders which are sure to appear on the market once a system is approved) would not have to buy *any* matrix decoder—minimal or logic-enhanced. Such listeners would hear a decoded version of the SQ disc directly off the air and, since the station needs to buy only one matrix decoder they could afford to buy the very best one available for that SQ disc. Furthermore, if next year you or I (or anyone else) invents a new matrix system that is clearly superior to SQ, QS or you-name-it, the station could, with the aid of an appropriate decoder or dematrixer, play discs made in accordance with that new matrix system as well. It would then be the public that would decide which matrix system (if any) is superior and the winner would find people buying records made in that format. Ultimately, a single matrix format would emerge—a desirable outcome which, if it had occurred a few years ago might have resulted in the further acceptance of 4-channel sound instead of in its premature "death."

On the other hand, stations could also broadcast discrete 4-channel tapes produced either in-house or passed along from one producer to another. For these, they would need no decoder at all and no encoder either, since, with a 4-4-4 system any and all 4-program sources, whatever their nature, could be broadcast directly. If CD-4 records of better quality are revived, they too could be first decoded using a CD-4 demodulator *at the station* and would be received in full 4-channel format by listeners equipped with a proper 4-channel tuner or receiver. Stereo listeners would still get a proper stereo fold-down and so would mono listeners, as previously explained. Thus, no one is forced to go out and buy anything if they don't want to, but all the options are left open for the future.

As far as the record industry is concerned, they would not be "locked-in" to any particular type of record, matrix or otherwise and would be free to explore and experiment with whatever new record formats appear in the future. If the FCC acquiesced to CBS's current petition, we would see only two kinds of records in the future—regular stereo records and SQ encoded records, since you can be certain that all record manufacturers would want to make sure that their releases can be legally broadcast over the air. It seems to me that placing such technological restrictions on the record industry is not properly the province or function of the FCC.

I hate to bring up an old sore point, but those of us who remember what went on when the FCC was considering the establishment of new stereo broadcast standards in the late 1950s and early 1960s remember only too well that, for reasons which I feel have never really been justified, we were given a stereo broadcast system which can be up to 20 dB noisier than monophonic FM and one which is particularly susceptible to multipath distortion. We could have had a much better system—one of the ones proposed at the time, but that, of course, is water over the dam. Let's hope that this time the FCC does not act too hastily. A proper decision in favor of 4-4-4 could prove a boon to the record industry—and that means all of it—the SQ folks, the QS folks, the CD-4 people and the as yet unheard from disc formats of the future.



NORMAN EISENBERG AND LEN FELDMAN

Technics RS-9900US Cassette Recorder

General Description: The Technics RS-9900 is a two-piece cassette recorder, the transport being housed in one unit and the amplifier in another. The two units may be stacked, placed side by side or rack-mounted. Interconnection is made via two multi-pin cables supplied.

Besides being designed in this unusual way, the 9900 offers some interesting features, highlights of which are a closed-loop, 3-motor, double-capstan transport with direct-drive motor; full three-head complement (erase, record and play); logic control of transport functions; a pitch control for varying tape speed in playback; a tape-time meter that shows remaining program time. The recorder also includes a complete Dolby-B system with calibration adjustments; bias and EQ adjustments; azimuth adjustment; input mixing; sound-on-sound; memory-controlled play and rewind. Recording and playback with an external timer is possible, as is the use of a remote-control device available as an optional accessory. The VU meters are peak-reading. The 9900 also identifies CrO_2 cassettes automatically and adjusts its circuitry accordingly when one is used and the power is turned on.

The main power switch is on the transport unit's front panel. The cassette door swings down to open and the cassette is inserted more or less horizontally. You can then view the cassette in the illuminated compartment both fore and aft—in addition to looking at it directly, you can see a reverse image of it in a mirror. The time and memory switches are on this panel, as are the pitch control and the tape-time meter. In addition to this novel indicator there's the usual tape index counter and reset button. Transport controls are "feather-touch" buttons for the usual functions. Fast-buttoning is possible except for going into the record mode which requires using the stop button first. The cassette eject control works only when power is on. In the event of no power, there's a manual button inside the cassette compartment.

The front panel of the electronics unit contains numerous controls and adjustments. Switches in the top row handle tape/source monitor; mic attenuator;

tape selector (CrO_2, normal, FeCr); the built-in oscillator (400 Hz and 8 kHz signals); the Dolby system; and the multiplex filter. Below these controls are several more, behind a removable transparent cover. This group includes screwdriver adjustments for left- and right-channel play calibration; left- and right-channel record calibration; bias; and left- and right-channel EQ. Associated with the bias pot is a button to select either variable or preset bias.

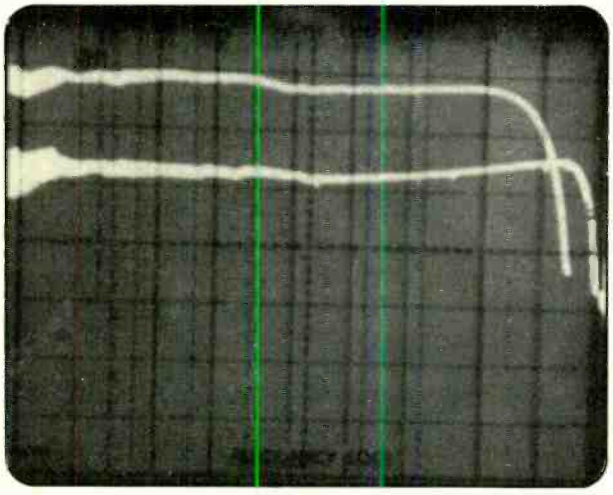


Fig. 1: Technics RS-9900US: Record/play response at reference record level (upper trace) and −20 dB, using Maxell UD-XL II tape.

Below this group are signal jacks, including a headphone output with its own level control, a stereo pair of mic jacks and stereo jacks for aux in and aux out signals.

The two VU meters occupy the upper right portion of the panel. They are peak-reading, calibrated from −40 to +5, with −8 marked in red and corresponding to the overload point of a standard VU scale (thus, the 0 dB mark on these meters would be +8 on a standard VU meter).

Below the meters are the level controls: one dual-concentric pair handles left- and right-channel mic/aux levels; another similar pair handles line input levels. Both pairs have rotatable rings that may be set to markings on a 0 to 10 scale for reference. The output level control is a single knob for both channels simultaneously; it is a stepped control with 2-dB increments from +8 to −26, and additional steps for −29, −34, −42 and "infinity."

The rear panel of the transport unit contains a grounding post, the AC power cord, the remote-control accessory socket and the multi-pin sockets for linking to the amplifier unit. The rear panel of the amp contains corresponding multi-pin sockets, a stereo pair of line-input jacks and two sets of line-output jacks. There also is a stereo pair of "throughout" jacks which are paralleled with the line-input jacks to permit taking input signals before they are recorded. In this way, it is possible to split a source and feed it simultaneously into two recorders, or to feed the input while recording into some other device for special purposes (an oscilloscope, for instance).

Topside of the electronics unit are additional screwdriver adjustments for left- and right-channel FM calibration, and for left- and right-channel oscillator calibration. Also presented on the top surface is a block diagram of the machine showing signal paths, plus three graphs depicting bias characteristics, record EQ adjustments, and peak meter response.

The front panels of both units are slotted at the ends for rack installation, and are fitted with handles. Finish is matte black; lettering is subdued and somewhat small, requiring good lighting for visibility.

Test Results: While our tests confirmed or exceeded many of the published specs on the 9900, readers will note some areas that seem to disagree (e.g., harmonic distortion and signal-to-noise). This apparent discrepancy is due to the calibration of the 9900's meters. Nominal reference level of 160 nanowebers/meter appears on these meters at −8 dB. Dolby calibration is at −5 dB. This is explained in the owner's manual in terms of the meters being wide-

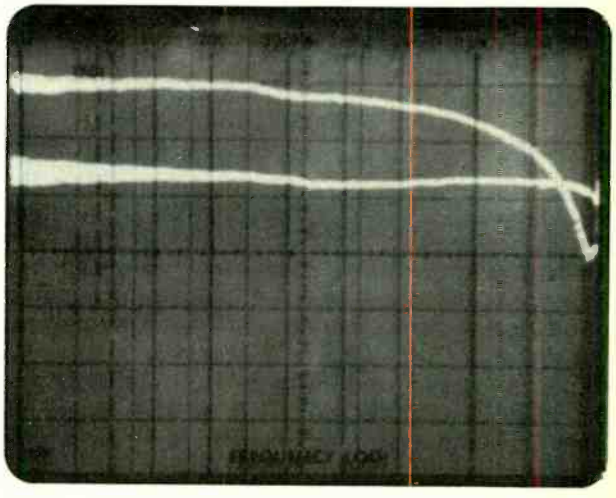


Fig. 2: Technics RS-9900US: Record/play response at reference record level (upper trace) and −20 dB, using Sony "DUAD" (FeCr) tape.

range peak-reading types. Thus, if you are accustomed to pushing record levels up to and beyond the usual 0-dB mark when recording music, you can run into trouble with this machine as far as distortion and tape-saturation are concerned. All of the measurements shown in our "Vital Statistics" section take this fact into account, and the frequency response displays (Figs. 1, 2 and 3) were taken at the *true* reference level and at points −20 dB below that level rather than at the "0 dB" mark on the meters.

There is a great deal of flexibility built into this recorder which means that one should take more than usual care if its ultimate performance is to be realized. For instance, azimuth alignment of the record head seems absolutely essential every time you insert a new cassette into the machine, and with each reversal of tape direction when you flip the cassette over. While the built-in high-frequency signal does a fairly good job of allowing you to align the record head each time, the kind of frequency response reported in our lab measurements would probably be achieved more certainly with the use of an external test signal (preferably one at around 12 to 15 kHz).

The available adjustments provided for bias and EQ make it possible to optimize performance with virtually any high-grade cassette tape, but with so many controls available, we would advise staying with the factory-set fixed positions unless you do have auxiliary test equipment with which to truly tweak the vernier bias and EQ adjustments. Happily, with the tapes we used (TDK SA, Maxell UD-XLI and Sony

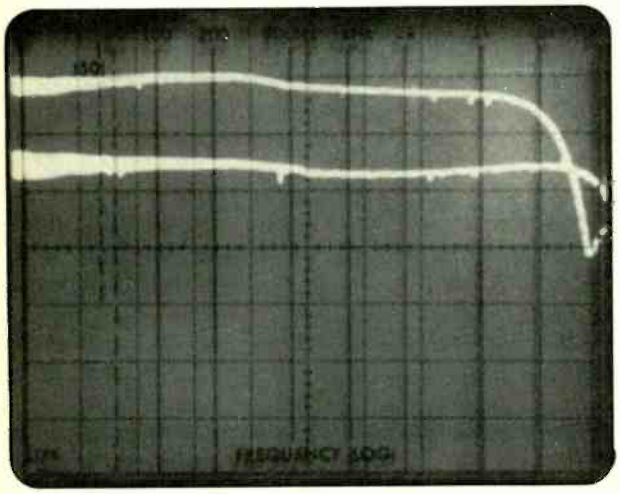


Fig. 3: Technics RS-9900US: Record/play response at reference record level (upper trace) and −20 dB, using TDK type SA tape.

Duad), the factory-set positions did provide recorded results that were almost precisely as good as what we were able to obtain by separate tweaking of the variable bias and EQ controls. For the preset controls, several tape brands and types are listed in the owner's manual.

The 9900's three-head configuration, combined with a flawlessly executed double-capstan, closed-loop three- motor drive system results in just about the smoothest cassette deck transport we ever have tested. The weighted wow-and-flutter figure of 0.04% is of course very low, but even the unweighted measurement proved to be a very low 0.05%. Low levels of modulation noise, as observed in the spectrum analysis (Fig. 4) further confirm the elegance of the 9900's transport system.

General Info: Transport unit dimensions are 19 inches wide; 7⅞ inches high; 14¾ inches deep. Weight is 33 pounds. Amplifier unit dimensions are: 19 inches wide; 6⅞ inches deep. Weight is 19⅞ pounds. Price,on both units as system, $1500. Owner's manual is complete and amply illustrated.

Individual Comment by L.F.: There seems to be no limit to what can be done with the cassette tape recording format if one is willing to spend the money for a superior cassette deck system such as this top-of-the line model from Technics by Panasonic. As explained in our Test Results section, though, the more flexibility built into a cassette system, the more care the user must exercise to realize its full performance potential.

The obvious question that comes up in evaluating such a unit is: "Have the people at Technics engaged in a program of 'overkill' as far as cassette recording is

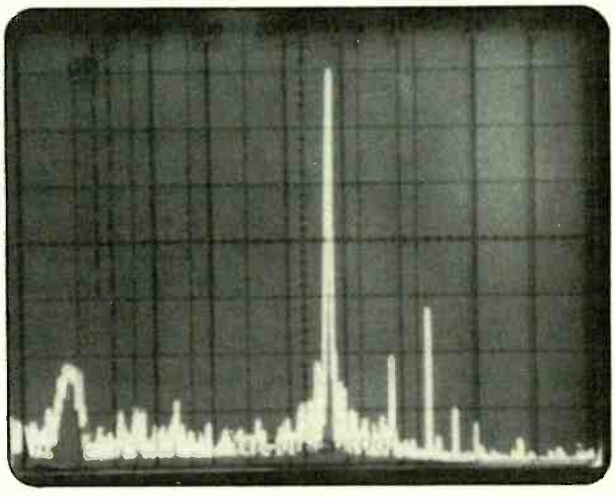
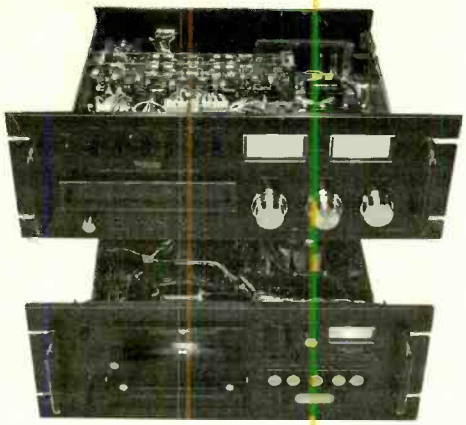


Fig. 4: Technics RS-9900US: Spectrum analysis of 1 kHz signal reproduced using the RS-9900US illustrates nature of actual THD components and low order of tape modulation noise.

concerned?" When one considers the fact that you can purchase an open-reel deck, such as Technics' own magnificent Model RS-1500US for the same suggested price as this two-piece rack-mountable cassette machine, one can only conclude that a purchaser of this latest cassette effort must have a pretty darn good reason for wanting to do so. For such a user, the 9900 probably will serve as a standard against which lesser cassette machines will be judged for quite some time to come.

Individual Comment by N.E.: My first encounter with this two-piece, rack-mountable, black-panel-with-handles-and-over-a-dozen-adjustments-



Technics RS9900US Stereo Cassette Deck: Internal view.

equipment which was, after all, "only a cassette machine" proved something of a techno-cultural shock. I attempted to get over it by engaging in a question-and-answer session with an imaginary companion which may help put this product in some perspective:

Q. Why do you suppose Technics brought out the 9900?

A. To show how far the cassette technology can be pushed and probably also to develop some clout and charisma in this and in related product areas.

Q. Just how good is the 9900?

A. Probably as good as the best in cassette recorders.

Q. Come on now, be more specific. Is it better than the Nakamichi 1000?

A. I don't know. I think the Nakamichi holds frequency response to closer tolerances, ±2 dB rather than ±3 dB. The Technics has very slightly lower wow and flutter. How can one give a clear-cut verdict? Let's say that from our test results it appears that the 9900 is in the same super class as the Nakamichi.

Q. I'm going to pin you down yet. If you had to choose between them, which would you pick for your own use?

A. Maybe I'd toss a coin since I can find no way to document the clear-cut superiority of one over the other. (I like the Tandberg TCD-330, too, for that matter.) With all these machines you get a feeling of rock-solid stability of tape motion, and unrestricted audio response that combine to make cassettes sound fantastically good.

Q. Don't hedge. Choose between the Nakamichi and the Technics.

A. If you insist—on final balance, it seems that the Nakamichi takes less installation space and is somewhat easier to use and still get the same results.

Q. Maybe so. But that two-piece design appeals to me. It looks . . . well, professional. It has macho.

A. Exactly. Which brings us right back to where we started.

TECHNICS MODEL RS9900US CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, standard tape*	± 3 dB, 25 Hz to 18 kHz	± 3 dB, 24 Hz to 18.5 kHz
CrO_2 tape**	± 3 dB, 25 Hz to 20 kHz	± 3 dB, 25 Hz to 20 kHz
FeCr tape***	N/A	± 3 dB, 23 Hz to 22 kHz
Harmonic distortion at −3 VU (1 kHz) normal tape/CrO_2	NA/NA	1.2%/1.4%
Harmonic distortion at 0 VU (1 kHz) normal tape/CrO_2	NA/NA	1.3%/2.3%
Harmonic distortion at +3 VU (1 kHz) normal tape/CrO_2	NA/NA	1.4%
Record level for max 3% THD, normal/CrO_2	NA/NA	+6 dB/+1 dB
Signal-to-noise ratio, std tape, Dolby off	57 dB	52.5 dB
std tape, Dolby on	67 dB	62.5 dB
CrO_2 tape, Dolby off	57 dB	52 dB
CrO_2 tape, Dolby on	67 dB	61 dB
Wow and flutter	0.04% WRMS	0.04% WRMS (0.05 unweighted)
Fast-wind time, C-60	70 seconds	66 seconds
Mic input sensitivity	0.25 mV	0.23 mV
Line input sensitivity	60 mV	68 mV
Line output level	0.42 V	1.0 V
Headphone output level	0 to 900 mV, 8 to 125 ohms	168 mV, 8 ohms
Power consumption	45 watts	38 watts

*Maxell UD XLI **TDK SA ***Sony Duad

CIRCLE 20 ON READER SERVICE CARD

Morley Model EDL Electrostatic Delay Line

General Description: The model EDL from Morley (a division of Tel-Ray Electronics) is said to be the only delay line in the world using an electrostatic memory system. It is intended for use with any amplified instrument or microphone, regardless of impedance. In conventional operation, the device is connected between microphone or instrument and an amplifier via its input and output jacks. Additionally, it may be used for generating "stereo or other effects" by splitting the signal using the regular output jack to one amplifier and a delay-out jack to another amplifier. Additional applications, including the use of a P.A. mixer or console that has an echo/send receive circuit, are described briefly in the instructions furnished with the unit.

Styled for being carried around, the Morley EDL is fitted into a black leather-like case with metal corner reinforcements, four small feet and a carrying handle. The front panel contains four signal jacks: input, direct out, delay out and output. These all are standard ¼-inch phone jacks. There also are four knob controls for input level, delay, repeat and delay level. At the rear there's a jack for plugging in an optional foot switch, the unit's AC power cord (which may be neatly wrapped around a metal holder) and the unit's power off/on switch.

Although scarcely any "inside information" was supplied with our test sample, our curiosity as to how the device worked led us to remove the back cover which revealed a rotating flywheel driven by a belt from a high-speed motor. Just as it is possible to magnetize a tape loop with an audio signal and then retrieve that signal later in its travel (the distance between record and play heads determining the delay time of the signal), so it is possible to apply an electrostatic charge that varies in accordance with an applied voltage to a revolving storage medium. This may be thought of as one plate of a capacitor that is in motion. The varying charge then can be "read" some time later on the moving electrostatic loop. Ingenious, original and effective—and it sidesteps the drawbacks of a tape loop (such as the tape wearing out or breaking or getting jammed).

Test Results: Since Morley provides no specifications for the unit, there wasn't much we could measure other than the range of time delay, and the distortion and response of the delayed output. We found that the amount of delay was adjustable from a bit over 10 milliseconds to about one-third of a second. The device also provides repeat-echo effects, from subtle "fading" ones to a complete runaway.

Measured distortion for the delayed output was fairly high—up to about 0.5 percent at mid-frequencies with moderate levels of input signal (under 0.5 volt or so). It also was noted that the frequency response of the delayed output began to roll off fairly soon, at above 2 kHz or so with moderate time delay dialed in, or even lower if you opt for the longest time delay. In an actual "live" performance, this probably will be of little concern since primary fidelity or good response will be heard from the direct sound. The delayed sound itself, of course, can be mixed with the direct sound or, alternatively, can be fed to the second channel of a stereo setup with the direct sound available from a separate output on the EDL.

The controls on the EDL were well-tapered, providing the kind of gradual control over delay and repeat that a performer will find easy to use. The input level control can handle a wide range of signal levels so that regardless of the kind of instrument pickup used, there should be no problem of overload.

General Info: Dimensions are 12 inches wide; 7⅞ inches high; 9 inches deep. Owner's instructions somewhat minimal but adequate for use. Price: $299.95.

Joint Comment by N.E. and L.F.: A most unique device, the Morley EDL does its intended job. It provides a bona-fide delay, albeit it does so in a man-

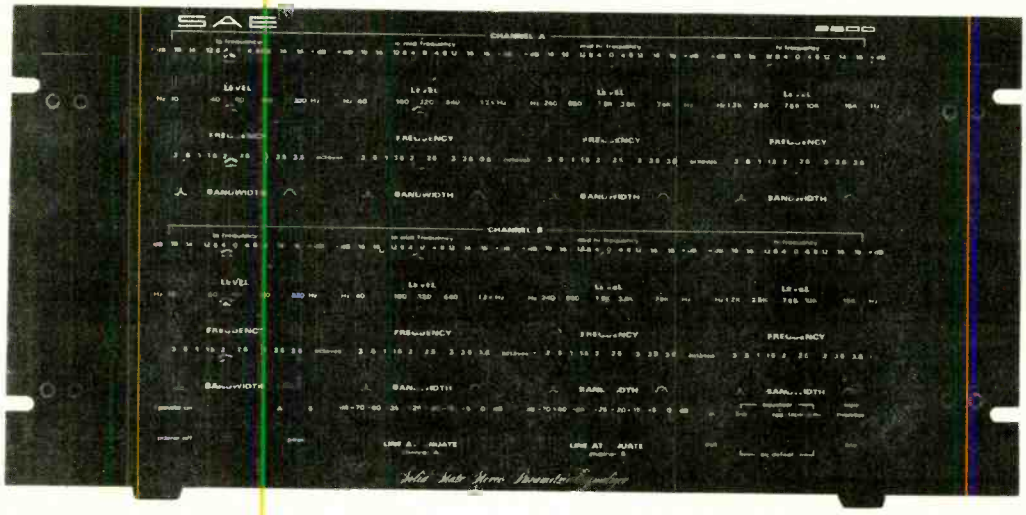
ner unlike any similar device we have yet encountered. The few measurement-tests we made on it indicate a competently-operating device, and its controls and signal jacks are adequate for the various applications possible with this product which also, by the way, could include introducing delay in a playback system.

MORLEY EDL ELECTROSTATIC DELAY LINE: Vital Statistics
(No manufacturer's specs are furnished)

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Delay time	Adjustable, c. 10 msec. to c. 330 msec.
Repeat echo	available
Distortion, delayed output	c. 0.5%, mid-frequencies, for input of c. 0.5-volt
Response, delayed output	Rolloff above 2 kHz (see text)

CIRCLE 21 ON READER SERVICE CARD

SAE Model 2800 Parametric Equalizer



General Description: SAE's model 2800 offers parametric equalization for four bands on each of two channels. The bands are designated as low, low-mid, high-mid and high (frequencies). Each band is adjustable by three sliders that move horizontally. One slider handles signal level (markings indicate from −16 to +16 dB). Another slider chooses the frequency. The indicated markings for the four bands, respectively, are: 10 Hz to 320 Hz; 40 Hz to 1.2 kHz; 240 Hz to 7.6 kHz; and 1.2 kHz to 15 kHz. The third slider selects the bandwidth characteristic, designated visually as ranging from a very narrow peak to a relatively broad one. The "parametric" function of varying center frequency and bandwidth is thus provided within these four frequency segments.

In addition to these 24 sliders (12 per channel), there are two more sliders for output level on each channel (called here "line attenuate"). To the left at the bottom is the unit's power off/on switch. Next to the power switch are three LEDs—one is a power-on indicator and the other two show overload for each channel. Grouped at the lower right are four more buttons that select and control the output format. With all four buttons left out, the model 2800 is bypassed and its outputs are not affected by EQ. Various combinations of pushing the buttons in permit selecting EQ or non/EQ signals at the main output and with or without affecting tape inputs and outputs.

All these controls and markings fill the front panel of the 2800 which is finished in black, with end-handles, and slotted for rack mounting. The rear contains stereo pairs of signal jacks (hi-fi pin types) for main in and out and for tape in and out. Also at the rear is the unit's AC power cord and a fuse-holder. The SAE 2800 is intended for normal connection into the tape recorder inputs and outputs of a typical audio component system, such as the tape-monitor circuit. The tape recorder itself then is patched into the model 2800. The device may also be connected via its main in and out jacks to a separate preamp and power amp.

The 2800 may be installed in a rack, or fitted into a cutout (horizontally or vertically), or simply placed upright on its four feet. An optional walnut-sided cabinet also is available.

Intended uses of the 2800 include program material and room equalization and speaker correction.

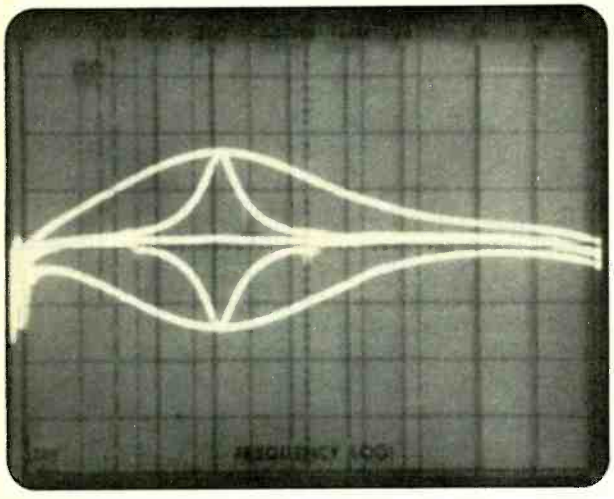


Fig. 1: SAE 2800: Please see text for explanation of this figure and of figures 2 & 3.

Test Results: Lab measurements confirmed the generally excellent performance of the SAE 2800, while a good measure of its versatility may be understood by looking at the various 'scope photos of frequency sweeps we obtained in the lab. The superimposed sweeps in Fig. 1 show the maximum boost and cut range of the 'mid-hi' band, with its center frequency adjusted to the low end of its range and its bandwidth adjusted first for maximum width (gentle sloping curves of boost and cut), and then for narrowest bandwidth (or highest Q). Of course, any degree of boost, cut or bandwidth in between is obtainable via the level and/or bandwidth controls for each band.

In Fig. 2 the same set of controls (same band) was used—except this time the center frequency was shifted to its upper extreme. The reader should note what a large section of the audible spectrum can be handled by this single band.

As shown by these 'scope pictures, there is a good deal of overlap possible between adjacent bands, and it is this characteristic that enables the parametric equalizer to do things that are not possible with a conventional graphic equalizer. An example of this is shown in Fig. 3. Here, the "lo-mid" band control was adjusted for a center frequency of about 250 Hz, and the level control was set for a substantial boost—a situation not unlike that which might be employed by a listener who likes the upper bass tones to predominate. Such extreme boosting, however, could very likely cause any room-resonances in the bass region to become all the more objectionable. So, to further illustrate the point, we set the low-frequency band controls for a center frequency of about 60 Hz, reduced the bandwidth of this section to minimum (about 0.3 octave, nominally), and set the level control for a deep cut. As shown in Fig. 3 (upper trace), the *overall* rise in response at the upper bass is still there, but we have managed to "null" a specific and narrow group of frequencies right out of the system at the objectional resonant frequency.

Had the "standing waves" or other resonances been at a somewhat higher frequency (say, 200 Hz or so), moving the frequency control on the "low" band up to that frequency would provide the needed null at 200 Hz, while the overall, gentle rise in the bass region—introduced through the use of the adjacent band's controls—is still maintained, as in the lower trace of Fig. 3. In our view, this kind of equalization—possible with a parametric device such as the 2800—is beyond the ability of any conventional graphic equalizer.

General Info: Dimensions are 19 inches wide; 8¾ inches high; 3½ inches deep. For handles, add an additional 1½ inches to overall depth. Weight is 20 pounds. Price is $550. Owner's manual is complete and amply illustrated.

Individual Comment by L.F.: To quote from an informative "white paper" written by the people at SAE, "By offering control over not only the cut and boost (as in conventional graphics) but center frequency and bandwidth, the parametric (equalizer) can actually be tailored to the specific needs of the listener whether he wants to modify an individual fundamental or harmonic or broad band boost or cut. The parametric's key is its ability to exactly control and modify the signal as the listener wishes."

Syntax notwithstanding, that just about says it all for this or any other parametric equalizer. In the several years during which I have been experimenting with a variety of equalizers (parametric or graphic) I have always felt that for most applications and acoustic environments I really needed only two or at most three frequency segments worth of control. In my listening room (and using my particular ears) I usually need to modify some portion of the upper bass region (usually upwards), an upper mid-range area and, in cases where I am using speaker systems with less than superb high-end response, an occasional assist for the treble region. On rare occasions I may take issue with a recording engineer's particular ideas of EQ (whoever said that all discs are recorded strictly in accordance with RIAA characteristics omitted what

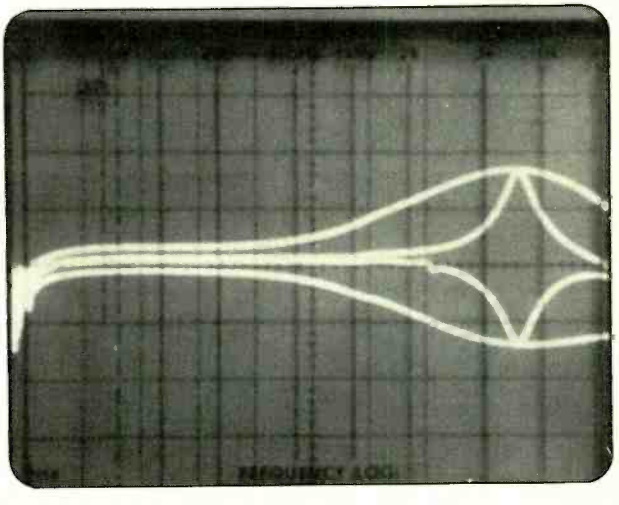


Fig. 2

happens at the studio console, for sure) and want to "fix up" what to me sounds like an unbalanced program. But even that "fixing up" requires modification of only one or two relatively narrow portions of the audio spectrum.

The problem, of course, has always been that no two situations require exactly the same center-frequency points of compensation, nor are the bandwidths requiring correction ever the same from one situation to the next. Graphic equalizer proponents solve this problem by giving the user as many individual band controls as possible. With third-octave equalization and with each control operating over a relatively narrow bandwidth (or possessing a high enough Q) it becomes possible to create an almost infinite variety of overall response curves. Multiple use of adjacent controls provide smooth, more gentle curves while use of specific narrow-band controls is effective in removing unwanted resonances in the listening room or those caused by other components in the system.

If you go along with the idea that only a few areas in the spectrum ever really need EQ in any given situation, then the parametric equalizer, though equipped in the case of the SAE 2800 with only four separate bands of control, can do as effective a job of overall tonal compensation as can those twenty-plus control jobs known as graphic equalizers. It's possible that they can even do a better job.

The important thing about any equalizer (as far as we are concerned) is that it not introduce added noise or distortion when inserted in the signal chain. These requirements are met more than adequately by the SAE 2800, as evidenced by the excellent measurements shown in our table of Vital Statistics.

As to the versatility of this particular unit, at least some of that versatility will be understood when you examine the various 'scope photos of frequency sweeps which we obtained during our lab tests of our sample 2800, as explained in the Test Results section. Try doing that with any conventional graphic equalizer you know of!

To summarize, different people use equalizers for different purposes and no one can argue the merits or demerits of a person's sonic taste. If you are faced with

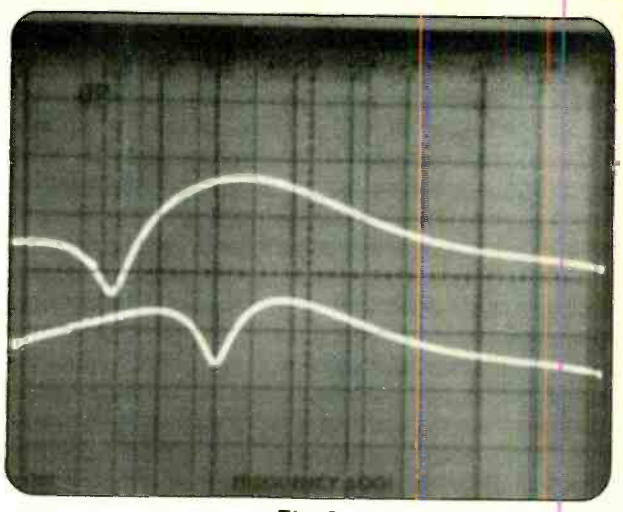


Fig. 3

EQ problems similar to those we have illustrated (or any others) chances are a parametric equalizer such as SAE's new 2800—even with only four bands of control—will be able to handle them. Considering the sophistication of the circuitry, accuracy of calibration and response, low distortion and excellent signal-to-noise ratio of this model, the serious audio enthusiast or recordist shouldn't find the price of the 2800 too hard to handle, either.

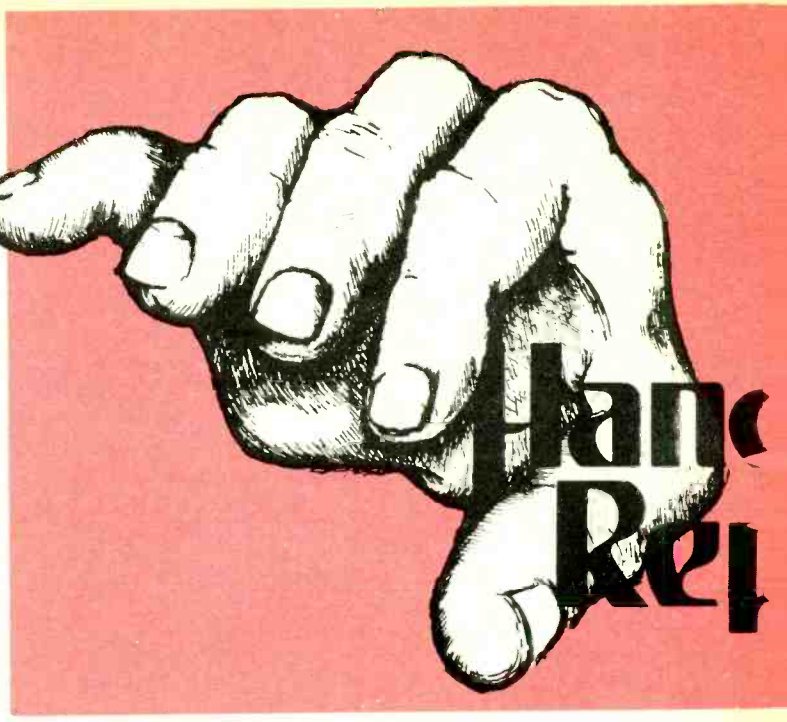
Individual Comment by N.E.: While the SAE 2800 probably can be used in some of the more "workhorse" type of applications associated with equalizers, it seems to me the kind of device that would be favored primarily by a very critical music listener who wishes to bring program material, playback system (and/or tape recorder) and listening room into some kind of meaningful togetherness, acoustically speaking. The four bandwidths and their frequency ranges seem to have been chosen on the basis of their relationship to musical tonal-spectrum structure, and to fundamentals and harmonic overtones, in terms of their interrelationships and their influence on instrument and ensemble balance. In this sense, the device stands as a "living example" of the close ties between technology and art, which makes me rather happy.

SAE MODEL 2800 PARAMETRIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response	± 0.25 dB, 20 Hz to 20 kHz	± 0.2 dB, 20 Hz to 20 kHz
THD (20 Hz to 20 kHz)	<0.2%	0.007% at 1 kHz 0.025% at 20 Hz 0.017% at 20 kHz
IM distortion	<0.02%	0.0035%
S/N ratio	100 dB	92 dB below 2.5 V out
Rated output	2.5 V RMS	(used as reference level)
Output at clipping	9 V (10 K ohm load; + 14 dBm)	9.5 V
Input impedance	100 K ohm	confirmed
Output source impedance	500 ohms	confirmed
Insertion loss (controls centered)	< 1.0 dB	0.7 dB
Power consumption	20 watts	18 watts

Aphex Aural Exciter

By Jim Ford and Brian Roth



We had heard all sorts of tales and rumors concerning a new audio signal-processing device called the Aphex Aural Exciter. It was reported to significantly improve the sound quality of practically any source by making the sound "brighter, clearer and punchier."

It was a natural reaction (we felt) to laugh at these original whisperings. After all, there isn't *any* audio processing that hasn't been done before. Or . . ., was there something new under the sun?

Finally, toward the end of last year, two trade journals (including *Modern Recording* in the October 1977 issue) gave some concrete information, although it seemed that the articles raised more questions than they answered. Our curiosity, needless to say, was working overtime now!

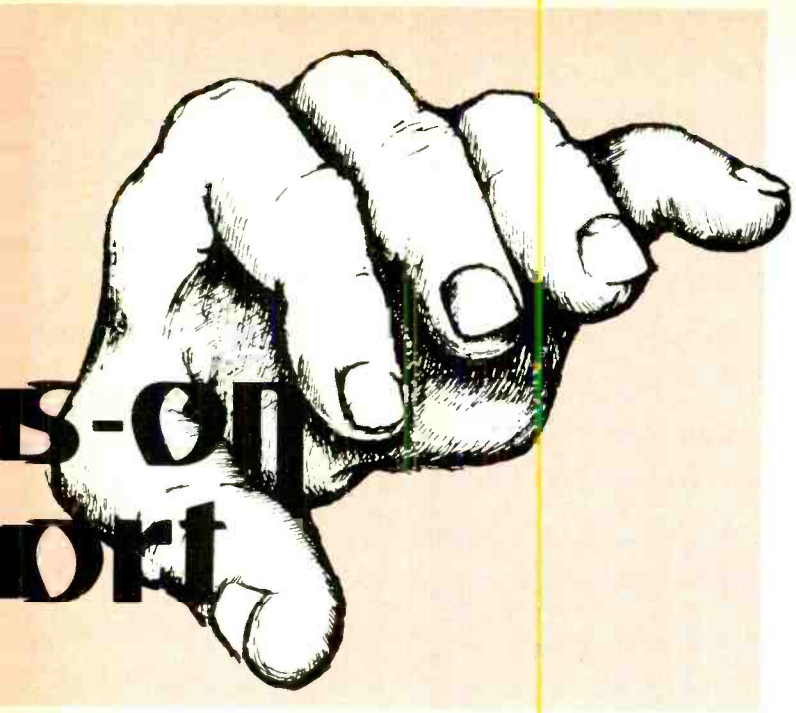
Then came the call from the Editor of *Modern Recording*. He informed us that the magazine had arranged for us to check out this mysterious device.

We contacted Marvin Caesar at Aphex Systems. When asked exactly what the Aphex did, Marvin described its effect as making the sound brighter, clearer, etc., etc.,—pretty much what we had already heard. Marvin went on to emphasize that the unit wasn't an equalizer, or some type of gimmick. He explained that the processing involved frequency response and phase manipulation as well as harmonic modification. Marvin pointed out that the unit is interfaced into a recording or sound reinforcement mixer via the echo send and receive circuits rather than being inserted directly into the signal path as a limiter or equalizer would be (although the Aphex does include front panel mixing controls to allow operation without a mixing console). Thus, the echo send pots are used to selectively "add Aphex" just as these pots can be used to vary reverb intensity.

Marvin said that Aphex is most effective on low distortion acoustical material—strings, piano, acoustic guitar, vocals, and so forth. The Aphex had the unfortunate side effect of making noisy program sources noisier and distorted signals dirtier. However, with quality signals, Aphex would restore the brilliance and clarity that the microphones, mixing equipment, recorders, loudspeakers and all the rest of the equipment in the chain had removed.

Well, we had to hear it to even think about believing it. Consequently, Marvin made arrangements for us to audition the unit with its inventor, Kurt Knoppel, present to demonstrate it.

It is not our policy to have the manufacturer of a piece of equipment under evaluation looking over our shoulder as we test, but in this case we decided that it was a satisfactory arrangement. Additionally, at that time, there was much more demand than Aphex units



available, so the one we were to check out was to be delivered to a studio in Denver by Kurt the day after our examination.

Friday, the day of Mr. Knoppel's arrival, was one of those hectic days. There were several phone calls to area studios to find open time (last minute sessions required some juggling around) plus the traffic jams on the way to the airport to pick up Mr. Knoppel and his marvelous Aphex.

We arrived (late) at the airport and found a genteel man quietly sitting on an Anvil case labeled "Aphex." After the usual niceties, we sped across town to the 16-track studio where the listening evaluations were scheduled to be held.

There, several people were also waiting. Randy, the owner of the studio was pleasant, but we could sense that he felt that the whole Aphex thing was some kind of joke. Nevertheless, he was curious to see what it would do, as were a couple of others who were working on projects in the studio at that time.

Interfacing the unit to the studio's MCI console was simple since the inputs and outputs utilized "cannon" type connectors, plus the fact that the Aphex had bridging (comparatively high impedance) inputs and transformer balanced low impedance outputs.

The front panel control layout seemed straightforward and the controls operated with a quality feel.

A master tape was put on the Ampex MM-1100 and we found some piano tracks. The moment of truth; advance the echo send pots that were driving the Aphex.

Immediately upon hearing the result, the skeptical studio owner scoffed, "Why, that's just treble boost!"

Kurt Knoppel objected, "Try duplicating that sound with your equalizer."

With a grin, Randy reached over to the console's EQ controls. Soon, his grin diminished. The MCI equalizer could not make the piano track sound the same as it had with Aphex processing.

Now Kurt was smiling. "I have a standing offer of $100 to anyone that can duplicate the sound of Aphex," he stated.

Randy muttered, "Let's try the ITI Parametric Equalizer," referring to his favorite outboard unit in the equipment rack next to the console. It was soon apparent that even the ITI, with all of its flexibility, couldn't come close.

Now, all of the spectators in the room began to become excited. Gary, a jingle writer and producer, loaded one of his masters onto the multi-track. He experimented with various tracks and making statements like: "Great," "Fantastic." We listened closely ourselves and found that we too were pleased by the results. Aphex did indeed improve the recorded tracks.

Kurt gave a number of suggestions concerning operation of the Aphex, and showed us via the device's meters that the Aphex "subcarrier" (as he calls the output signal from the device) was considerably below that of the unprocessed signal to which it was being added. The console VU meters confirmed this fact.

"Since the Aphex level is so far below that of the dry signal, there is no danger of excessive program levels when Aphex is added. It only takes a small amount," Kurt pointed out.

Finally, it was time for other business at the studio, and we prepared to depart. By this time, Randy had become quite interested in the Aphex and arranged to keep in touch with Kurt.

Before we left, Alan, who was completing an album project, made arrangements for Kurt and us to return later that evening so that the Aphex could be used on his mixdown.

That done, Kurt left the Aphex for us to check on the test bench while he checked into his hotel. We took the unit and returned to our shop.

The first test we did was frequency response and we found that Aphex exhibits a shelving-type response that reaches its maximum point at about 20 kHz and drops at a rate of around 4 dB per octave below that. Around 2 kHz, the roll-off steepened as the effect of the device's 500 Hz, 12 dB per octave filter came into play. This measured response shape correlated with what we had heard at the studio.

The next check was output noise, and it was no higher than −80 dBm (unweighted, 20 Hz - 20 kHz) on either channel. This confirmed the listening test results of low noise contribution from the Aphex.

By this time, Kurt had returned. Unfortunately, time was growing short due to the studio engagement, so we didn't have time to do any type of harmonic distortion analysis. However, we did note some slight asymmetry in the sine wave display on the oscilloscope

which would indicate the Aphex was generating some low-ordered harmonics. Also, when adjusted for normal operating level as indicated by the illumination of a front panel LED, the Aphex clipped the sine wave asymmetrically, but in a "soft" fashion like a vacuum-tube amplifier. This, plus the high frequency boost, should be the main reasons for the device's sound characteristics.

Before we left the shop, we examined the interior of the Aural Exciter. The construction method utilized a motherboard almost the width and depth of the unit, and small circuit boards plugged into sockets on the motherboard. Parts quality was adequate, but construction quality left something to be desired. Kurt quickly stated that this model was from the first limited production run and future models would be improved. Later conversations with Marvin Caesar revealed that he himself may have been the circuit card "stuffer" (Marvin's background is in real estate and accounting), and that that duty had since been given to someone experienced in soldering. So, it appears that this situation won't apply to current units.

Due to the fact that the unit is leased from Aphex Systems, we quizzed Kurt about getting repairs should they be necessary. He indicated that since almost all the circuitry is on the plug-in cards, it is a simple matter to contact the factory with a description of the problem and they would dispatch the necessary card without charge. This eased our minds since, due to the unit's patent status, there are no published schematics (and the fact that the actual Aphex "generator" is sealed in an epoxy cube).

After a stop for dinner (where Kurt discussed his accidental discovery of Aphex when he miswired a stereo amplifier kit), we returned to the studio for the mixdown session. We "left the driving" pretty much up to Alan so we could sit back and listen to the results. Occasionally, we reached over to the console to mute the Aphex returns to compare the program with and without processing. In all cases, the signal quality seemed markedly improved with Aphex.

Alan also experimented with the stereo placement aspects of the unit. Since Aphex is a dual channel, it is possible to spatially dislocate the "subcarrier" from the main signal. Thus, the user can pan the dry audio to one side and Aphex to the other. This created some interesting results—an apparent widening of the signal's image in the stereo field.

Finally, in the not-so-wee hours of the next morning, the exhausted group decided to call it a night (day?). All seemed happy with the results of the mixdown.

Kurt and his black box departed later that Saturday. A few days later, we listened back to the mix at our shop and compared it with another mix · done earlier without Aphex. The second master did seem to be brighter and more distinctive. In fact, there may have been a bit too much of the Aphex, but this didn't appear to be as bad as having, say, too much reverb or excessive flanging or other effects.

Conclusions: During the first experiments we did at the studio, we ran the two-track recorder so we would have a tape of various sound sources with and without Aphex in a direct "A-B" fashion. We have practically worn the oxide off the tape playing it for ourselves as well as for colleagues.

The immediate reaction of the majority of listeners has been: "Treble boost!" A nice treble, but a boost nonetheless. Naturally, these listeners weren't present to actually use the Aural Exciter. However, if some were to ask us what the Aphex did, we certainly would have to say that the high end boost is the most apparent effect.

The reason for the console and outboard parametric equalizers' inabilities in duplicating the sound is partially explained by two facts. First, the two other equalizers more than likely had totally different frequency response curves (when in the treble boost mode) than the Aphex. This obviously would create audibly different results. Second, the Aphex is interfaced as a send and receive processor and *not* as an in-line processor as is the normal practice for inserting equalizers and the like.

This last fact prompted us to try patching an equalizer into the send/return loop on a console to see what would happen. The effect is somewhat similar to the Aphex results. The available equalizers did not have the same response curve as the Aural Exciter so the results couldn't be expected to sound the same.

Of course, the Aphex also introduces phase shifts (more than likely a side-effect of the response curve) plus some low-ordered harmonic products. These should also affect the final sound. The ability to pan the Aphex signal to a different location than the dry signal is another advantage of this type processing.

We pondered the fact that the VU meters showed the Aphex "subcarrier" to be much weaker than the main signal. This can be attributed in part to the lack of any low-frequency energy in the Aphex's output. This will cause the meters to indicate a low reading although the actual high-frequency level could be fairly strong. This metering phenomenon was partially duplicated during our experiments with an equalizer in the send/return path.

We all should know that adding treble energy to a program will enhance its apparent quality. Try setting up a stereo system with a *moderate* amount of treble boost. Put on a record, and ask a friend to listen (don't let him see the tone control settings). After a few minutes, set the tone to flat. Nine times out of ten, he will prefer the treble boosted sound.

As we have emphasized, the Aphex does give some different results from a simple treble boost. We feel that it will be up to the user to determine whether or not the effect is satisfactory in his or her application. If you do like the sound, you will be in good company since it seems the entire West Coast recording community has flipped over it. Listen to the unit yourself and then decide.

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CIRCLE 93 ON READER SERVICE CARD

GROOVE VIEWS

Reviewed by:
SEDGWICK CLARK
NAT HENTOFF
GIL PODOLINSKY
RUSSELL SHAW

POPULAR

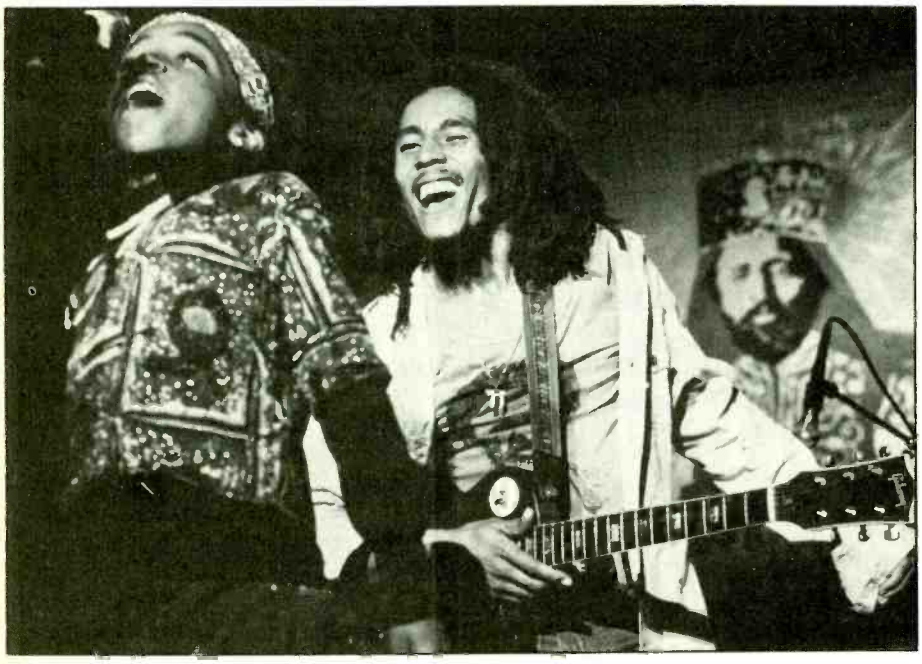
BOB MARLEY AND THE WAILERS: *Exodus.* [Bob Marley and the Wailers, producers; Karl Pitterson, Guy Bidmead, Terry Barham, engineers; no studio credited.] Island ILPS 9498.

Performance: **Honest**
Recording: **Flawed**

If it weren't for Island Records, the United States might never have been introduced to Reggae, a viable and creative musical form. The significance of this really hit home when I was informed on the Q.T. by several different record company executives that their respective labels planned to drop their reggae acts due to poor sales. I'm not surprised, for, in general, reggae gets little airplay. There isn't too much you can do to the music to make it more commercial either, although a few reggae "hard cores" have told me that they feel that *Exodus* shows Marley & Company in a somewhat commercial light. It may well be that this LP is the most commercial of all his albums thanks to occasional back up horns (compressed and placed so far down in the mix that they have no impact on the song) or slightly commercial lyrics like those on "Waiting In Vain." However, reggae is essentially a religious/political hybrid of music and there are a sufficient number of tunes to support this on the album.

The title track is one in particular. Lyrically it tells of the plight of Jah people (Marley's people) which is the never-ending theme of reggae music. Track assignment begins with a scratchy electric guitar right and a comping clavinet left, flanking a dry acoustic piano which fills in the bottom octaves in the center of the mix. They are joined by terribly flat and compressed horns right and arpeggioed clavinet left. The lead vocals are centered and backed by a pounding bass. The song eventually builds to a finish with a growling Hammond and the backing vocals left, drums right of center and the acoustic piano far right in the background.

Other than the annoying horns and the overall feeling that the entire LP went through a compressor/limiter, the quality of the material is as high as ever. G.P.



BOB MARLEY AND THE WAILERS:
A somewhat more commercial effort?

PHOEBE SNOW: *Never Letting Go.* [Phil Ramone, producer; Jim Boyer, engineer; recorded at A&R Recording, New York City.] Columbia JC 34875.

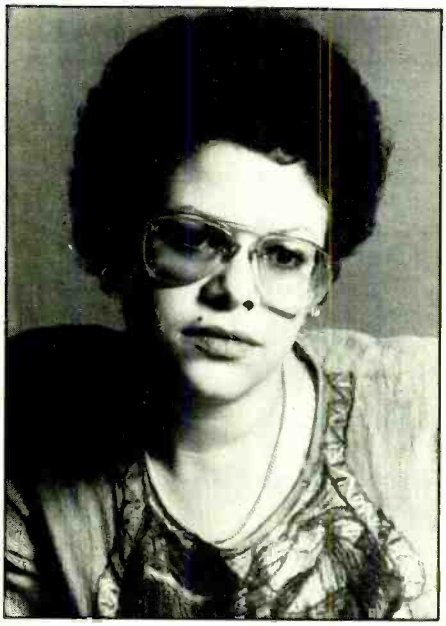
Performance: **Evocative and believable**
Recording: **Eloquently appropriate**

Phoebe Snow has a subtle tinge of irony in her voice which makes her frequent lyrical role as the all-American hardluck girl loser that much more credible. A superb songwriter, she has learned to exploit both the range and nuances of her tones.

The best new Snow composition here is "The Middle of The Night," in which the singer is a lonely soul who is destined to spend the wee hours loveless—at least until her crafty beau gives her a ring at some ghastly hour. Then...

sweet accomodation. "Electra" is another winner, a graphic account of the psychological traumas surrounding divorce.

As opposed to the usual folksy strum, accompaniment here rests with a whole slew of hired, jazz-oriented musicians. As always, the arrangments here are uniformly tight, ranging from the vibratto-inflected soul funk of "Love Makes A Woman" (a remake of a 1968 Barbara Acklin hit), to the quiet snappiness of Paul Simon's "Something So Right." All these players have achieved considerable recognition; try on for size the likes of Hugh McCracken, Ralph MacDonald, Grady Tate, Steve Gadd, Tony Levin,



PHOEBE SNOW: Simply superb

Bob James, and Phil Woods. Saxophonist Woods, who was rated tops on his instrument in a recent poll of jazz critics, plays a hauntingly lovely passage during the gentle "Never Letting Go."

The production is impeccable, ranging from the universally apt choice of musicians to the arrangements, which always seem to match the mood to a tee. What more could anyone ask? R.S.

GENTLE GIANT: *The Missing Piece.* [Gentle Giant, producer; Paul Northfield, engineer; recorded at Relight Studios, Hilvarenbeek, Holland.] Capitol ST 11696.

Performance: **Rejuvenated**
Recording: **New approach**

This album marks a distinct change in the way Gentle Giant records. Prior to

GENTLE GIANT: Closer to the vision

this album, the approach was the basic track by track method, with, for example, drummer John P. Weathers being asked to lay down a drum track to a song he's never heard and is not finished. The obvious problem was getting five people excited about the same piece of music when they were recording each instrument separately. Their last album, a "live" release, enabled them to both buy time and evaluate their next move. Having never heard themselves "live," except for bootlegs, they were quite pleased with the result. They stopped viewing recording and performing as two separate media. They also realized that "live," the material came closest to the way each piece was envisioned originally, and more satisfactorily than any of the studio LPs. So, they began playing the new material on their last tour *before* it was recorded, allowing the material to be refined to the fullest degree. The result is the tightest studio album they've ever done.

The spontaneity and energy that was present on the "live" album but was all -too-often lacking in the studio is very present on this consistently fine album.

While there is no denying that *The Missing Piece* is intentionally more commercial than anything previous, it should not be interpreted as a rejection of their high artistic standards. Side 2 is basic Gentle Giant, with the material on Side 1 being shorter and more singles -oriented. The subtleties and classical influence is as present as ever and no longer comes across as an abrupt change in the flow since both the piece and the performance are now woven together into a natural state. The recording is straight ahead, with no use of studio gimmicks, just rock 'n' roll, Gentle Giant style. With luck, this LP will not go by unnoticed, for Gentle Giant is truly an exemplary band. G.P.

STEELY DAN: *Aja.* [Gary Katz, producer; Roger Nichols, Elliot Scheiner, Bill Schnee, Al Schmitt, Lenise Bent, Ken Klinger, Linda Tyler, Ed Rack, Joe Bellamy, Ron Pongaliman, engineers; recorded at Village Recorders, West L.A.; Producer's Workshop, Hollywood; Warner Bros., North Hollywood Recording Studio; ABC Recording Studio, L.A.; Sound Labs, Hollywood; A&R Studios, N.Y.C.] ABC AA 1006.

Performance: **Genius at work**
Recording: **Superior**

For my money, Steely Dan is the best American rock/pop recording entity. This album shows a fantastic blend of musical and production intellect, for it succeeds where practically all others who've made this type of effort have failed. First, the lyrics are as pertinent, exact and meaningful as always, but then Fagen/Becker have spoiled us over the years to the point where enlightening lyrics are expected, and are therefore taken for granted. Actually, Steely Dan lyrics are a great study in short story writing.

They've made a style change, embracing a jazz feel with such understated

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STEELY DAN: Enlightening

naturalness that you'd swear they've sounded like this all their career. That's the high point of distinction. Whereas nine out of ten acts end up sounding like someone else when they experiment with style, The Dan always sound like The Dan. They have a unique, easily distinguishable and immediately identifiable sound, and as yet I've never heard anyone succeed in sounding like them.

Gary Katz has performed a miracle by getting new life out of session regulars. I've heard most of these guys play on just about every record that comes out of Los Angeles, but not like this! Katz should win a Grammy if for no other reason than that. The problem I have as a reviewer is deciding which of these seven gems I want to single out. When in doubt, start at the beginning.

"Black Cow," like all the songs presented here, is a complex meshing of instruments. The discretion used and the ability to adhere to it is amazing. Starting off with a clean, ungimmicked electric guitar left, bass and crisp snare center, high-hat and clavinet right, the intro quickly gives way to Fagen's vocal at center. The gorgeous female chorus add background harmonies in full stereo, backing off on the guitar and clavinet and into a Rhodes for a few arpeggios. Second chorus brings in a horn section, also placed center, holding notes to build supportive chords behind the vocals. J. Pankow of Chicago could learn from Tom Scott's fine arrangements on this LP.

The solo is taken by the Rhodes, slightly vibratoed, center mixed. A sax solo then takes over, alternating with the re-emerged vocal, almost improvising between the two and playing off of the full horn section. Fade the whole thing out and five minutes and seven seconds have elapsed too quickly. It's been said that the best jokes are the shortest lived, and some of that truth applies to Steely Dan, for no Dan tune ever evolves into an unwieldly jam. Each bone of framework is well placed with nothing either left over or to chance.

It matters not that we've been waiting a year for a new Steely Dan release. What does matter is that it's arrived and is again beyond everyone's expectations.

G.P.

JAZZ

ERIC GALE: *Ginseng Woman.* [Bob James, Joe Jorgensen, producers; Joe Jorgensen, Don Pulse, engineers; recorded at Mediasound and Columbia Recording Studios, New York.] Columbia PC 34421.

Performance: **Boring**
Recording: **Clean, but then so is blank tape**

In what I'm sure was a move to counter Warner Bros.' pop-jazz moves, Columbia signed Bob James to a producing

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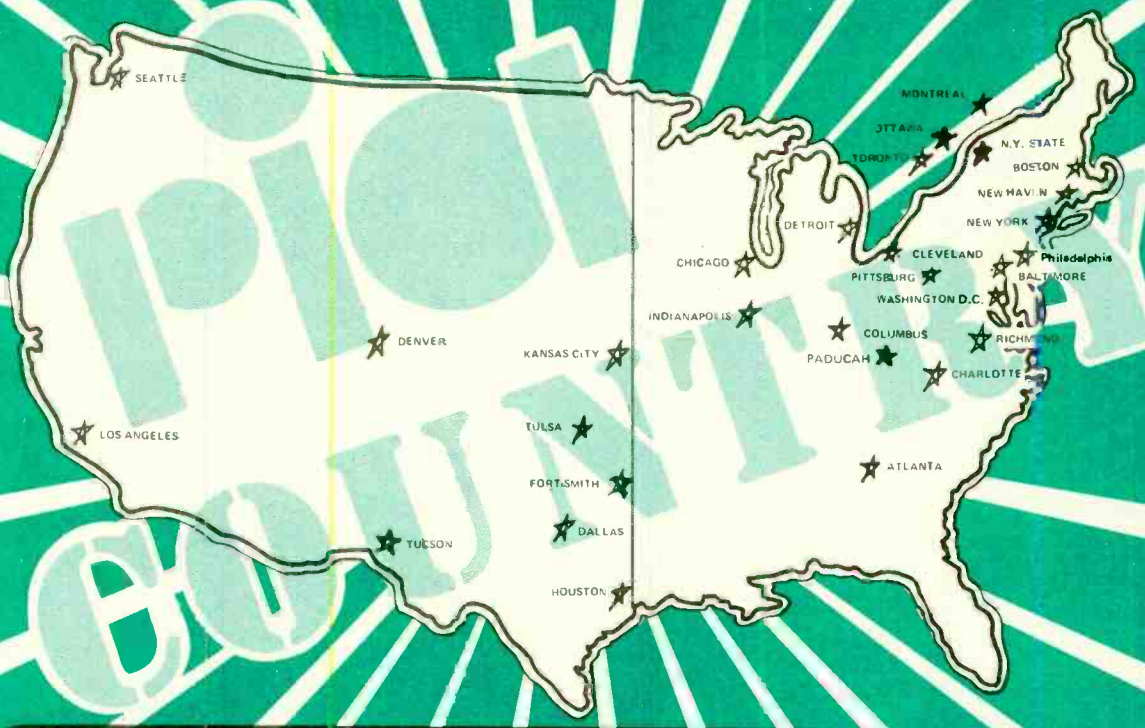

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two remarkable, if unsung, originals

By Nat Hentoff

Booker Little was only 19 when I first heard him with Max Roach. A quick and flexible trumpet player, clearly fond of Clifford Brown, Booker was moving away from bop and into freer improvisation with longer lines. He also had a devout belief that if your imagination is sufficiently resourceful, there can be no such thing as a "wrong" note. During the last two years of his life—1960-61—we worked together on dates I produced for Candid. He was a sideman on several; and on *Out Front* (which is about to be reissued on Janus), Booker was leader-composer-arranger. His writing was even more distinctive than his playing—each horn having a strong, independent line that sinuously intertwined with all the others to form a rich though subtle mosaic.

He was 23 at that point, and he had already accomplished so much that it was obvious Booker would be a seminal jazz force. But in October, 1961, Booker died of uremia, and since then he has been largely forgotten by all but musicians and a few lay aficionados. Until now. Not only will *Out Front* astonish recent listeners to jazz with Booker's lyrical power; but the very last session he ever made has also been resurrected, with good sound, by Bethlehem under the title, *Victory and Sorrow.*

All but one of the compositions are by Little and they reveal that he was growing until the very end. Both as player and arranger-composer, Booker was continually searching for new forms that would accomodate what can best be described as his penchant for thoughtful feeling. He certainly had the "cry" of jazz, his feelings were very intense, but he had many deeply reflective things he wanted to say with those feelings. And so, as in this album, Booker kept subtly stretching the language of jazz to etch moods and tell stories that challenged the minds as well as the emotions of his listeners. This is more than music to groove by, though you can do that too. It probes at you relentlessly, and makes you bring to it your own experiences and ambivalences, including some that may have long been hidden.

Little at least achieved some recognition during his awfully brief career, but Herbie Nichols—who died at 44 in 1963—was hardly known at all. A friend of Thelonious Monk, who encouraged him, Nichols was a pianist of rare melodic freshness, and his rhythms were always inextricably integrated with his harmonies and melody lines. That is, he did more than swing—although he certainly did that too.

Yet Herbie Nichols was able to record only three albums under his own name, and one of them, *Love, Gloom, Cash, Love* is fortunately now available again on Bethlehem with the new title, *Herbie Nichols.* Seven of the ten compositions are his, and as is the case of Booker Little, they establish the unmistakable presence of an authentic jazz original. On this competently recorded trio date, Nichols is attentively served by bassist George Duvivier and that long-time Mingus percussionist, Danny Richmond, who understood how vital to Nichols was a drummer who could draw a spectrum of colors from his instrument as well as keep time.

Herbie Nichols was also a great poet, but nobody paid much attention to those works either. Now, finally, at least his music is being really heard, though too late for Herbie.

BOOKER LITTLE: *Victory and Sorrow.* [Teddy Charles, producer, no information on engineer or recording studios.] Bethlehem BCP-6034.

HERBIE NICHOLS: *Herbie Nichols.* [Lee Kraft, producer, no information on engineer or recording studios.] Bethlehem BCP-6028.

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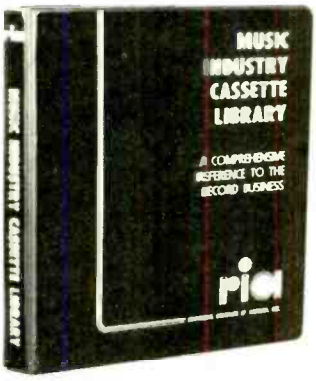

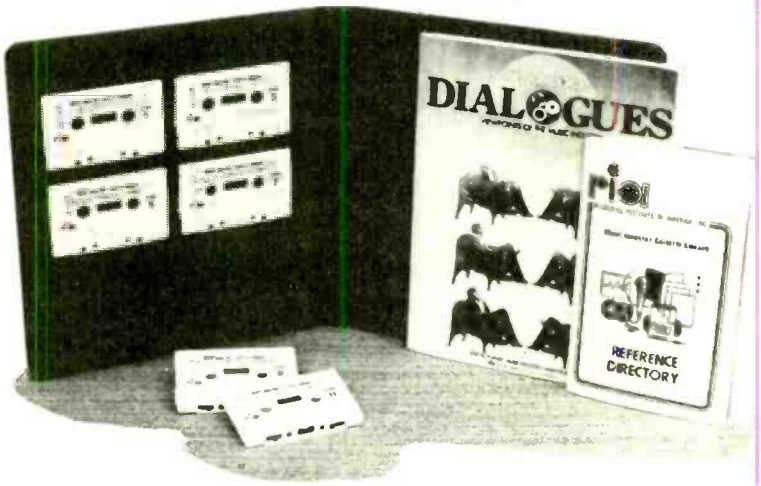

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contract and gave Eric Gale a solo album. To date the only notable project of James' has been co-producing Kenny Loggins, *Come Celebrate Me Home* with Phil Ramone, and he obviously hasn't learned much from the experience. Gale, on the other hand, has done nothing memorable to date, and doesn't begin here. The story is the same as when he was a member of Stuff and on his previous solo mistake: most session musicians make terrible solo artists. This LP is leaderless. To let you know Gale's the featured artist, Jorgensen pumps him up in the mix, but who wants to hear a guy compete with a horn, rhythm and vocal section?

Smokey Robinson once told me that in the early two-track days of Motown, there'd be up to thirty-two musicians and singers recording in the studio at once, with the engineer recording and mixing simultaneously. I can't recall one Motown record sounding overworked. Forty musicians and singers appear here and the overall feeling is that the listener is forced to turn the wrong way up a one-way street. This album is too crowded! Voices are only used to sing one line song titles in the chorus—original, eh? Everything less than an eighth note has been thrown out and everything that's left is rather expressionless. There is just no life in the music. The only thing that I learned from this album was the proper spelling of ginseng, and how often can I use that? G.P.

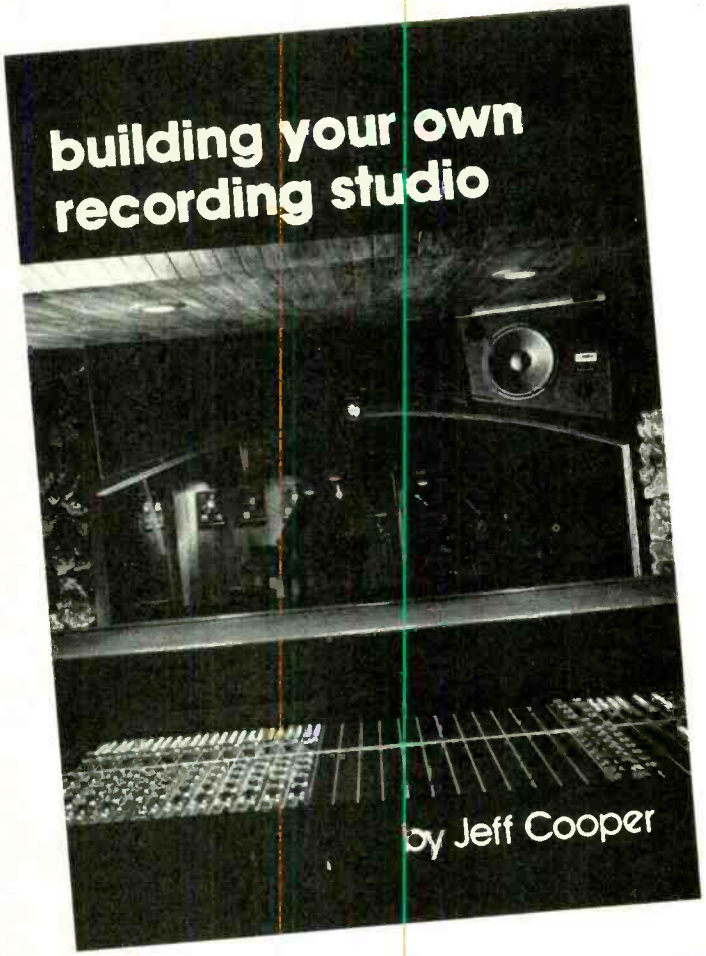
BILLY COBHAM: *Magic.* [Billy Cobham, producer; Dennis Mackay, engineer; recorded at Electric Lady Recording Studios, New York City, N.Y., and Manta Sound, Toronto, Canada.] Columbia JC 34939.

Performance: **Aimless**
Recording: **Functionally adept**

What is the purpose of a Billy Cobham album? While gymnastic, his drumming is imprecise; he trips over the basic changes while pulling off some difficult ones. His crew of backup help also contributes little to the proceedings, leaving the impression that their's is only a contract fulfilling exercise.

The material here ranges from dull to annoying. During "On A Magic Carpet Ride," (actually a moth eaten throw rug) we are taken through the usual series of Diatonic scale and Dorian-mode bartered solos on guitar, piano, etc. "AC DC" is more of the same

FINALLY!!



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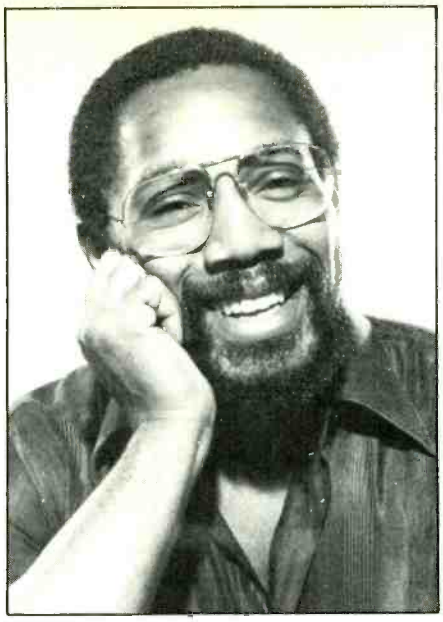

BILLY COBHAM: Bored the engineer?

pointless jamming, a kind of musical masturbation. Cobham is there throughout, authoring his cymbal splashes and snare crashes with vapid, predictable repetition. Once in a while he does his four bar drum solo, which is usually no more than an accelerated paradiddle.

Most of the tracks are like this. The only relief from this mechanical clockwork is the lively "Puffnstuff," a cute ditty tagged with a controversial chant. Cobham declares ". . . Gotta be free so that my mind can breathe the weed all around me." He goes on to deplore certain herbs, a choice which is strictly his own. Yet listening to this series of tiresome exercises, one wonders if he goes through life without any musical inspiration at all.

As expected, the drums are top-heavy in the mix; yet after emphasizing Cobham, what is an engineer to do but yawn and let the tape roll? R.S.

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DVORAK: *Slavonic Dances, Op. 46; Scherzo capriccioso, Op. 66.* Bavarian Radio Symphony Orchestra, Rafael Kubelik cond. [Dr. Rudolph Werner, production; Hans Weber, recording supervision; Heinz Wildhagen, recording engineer.] Deutsche Grammophon 2530 466.

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credits as above.] Deutsche Grammophon 2530 593.

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DVORAK: *Legends, Op. 59.* English Chamber Orchestra, Rafael Kubelik cond. [Same production team.] Deutsche Grammophon 2530 786.

DVORAK: *Stabat Mater, Op. 58.* Edith Mathis, Soprano; Anna Reynolds, Contralto; Wieslaw Ochman, Tenor; John Shirley-Quirk, Bass; Bavarian Radio Symphony Orchestra and Chorus, Rafael Kubelik cond. [Same production team.] Deutsche Grammophon 2707 099.

Performances: **See below**
Recordings: **Tight and vibrant**

The first two discs listed above were released in 1975, the second two in 1976 and the last three late in 1977. As I never got around to reviewing the earlier records, I decided to list them here since they are excellent in every way and anyone interested in the repertoire should know of their existence.

Kubelik's Dvorak recordings have always elicited a positive critical response from most corners. I had to demur when it came to the complete symphonies box (deleted, with only the Sixth through the Ninth now available singly) with the Berlin Philharmonic; the performances lacked a tight symphonic line, I thought, with only the "New World" managing to convince me. But these newer discs—except for the *Legends*, which seem rather aimless and lacking in purpose—are very successful. Like Bruno Walter, Kubelik's primary response to music is from the heart. This is not to say that these performances lack dynamism or power when called for, but that one will rarely encounter more poetic readings of these works.

The sound is superb on all of these discs—exceptionally open and clear, with plenty of visceral impact in climaxes. It's a lean Toscanini-like sonority, recorded in not-too-large a hall so that balances are absolutely precise—quite different than the label's recordings for Herbert von Karajan, which emphasize his taste for a rich, blended orchestral pallet (but even Karajan's recordings have recently been aiming for a closer, less resonant ambience; listen to his magnificent new Bruckner Seventh, for example).

Kubelik is not the last word on Dvorak, of course. The late Istvan Kertesz made many superb recordings on London of the Czech composer's music, the Szell/Cleveland performances of the *Slavonic Dances* on Odyssey are more attuned to Dvorak's *furiant* nature, and Vaclav Neumann and the Czech Philharmonic, on a three-record Telefunken set which I use to test cartridges and speakers for warmth of string sound, have recorded a glorious selection of Dvorak's smaller works. S.C.

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