JUNE 1977

VOL. 2 NO. 6
$1.50

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

Recording Direct to Disc

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The Peavey CS Series

Last year when Peavey introduced the CS-800 Stereo Power Amp, professional sound men and engineers acclaimed it as the most versatile high performance power amp available for under $1,500.00.

Now, there are two superbly engineered additions to the Peavey CS series, the CS-200 and CS-400. These new high performance amplifiers are built with the same meticulous quality control and engineering standards that go into the CS-800.

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CIRCLE 69 ON READER SERVICE CARD

Introducing, an incredible new advance in synthesizer technology.

Bye bye, Johnny one note. Polymoog is here—the first and only truly polyphonic synthesizer.

Polyphonic. It means you can play as many notes as you want... all at the same time... and make music that's full and rich and harmonic and expressive... all with the almost limitless electronic sound potential of a Moog synthesizer.

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CIRCLE 88 ON READER SERVICE CARD

JUNE 1977
VOL. 2 NO. 6

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
A Session with Patti LaBelle
Recording Crosby, Stills and Nash
How Acoustics Affect Recording

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THE STAPLES

Cover photo by Kirby Veach

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SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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Letters to the Editor

Unsolicited Response

We have been producing records for major labels for some time and have learned at least one thing. "As soon as you learn everything there is to know, your information is obsolete."

Recently, a studio engineer laid a couple of copies of *Modern Recording* on us, and we were astounded at all the things we didn't know! Those two copies are now marked from cover to cover in various colors of ink, carefully dogeared and filed with the master tapes for quick future reference.

It's a hell of a magazine!

—Jack Blanchard and Misty Morgan
Orlando, Fl.

The Demerits are Erased

One demerit and a minor slap on the wrists of Norman Eisenberg and Len Feldman for their Feb/Mar 1977 Lab Report.

First, if the Technics RS-1500US transport was "quite unlike anything we have yet encountered," I suggest they visit a few of the major recording studios and notice the transports of some of the 3M Mincom series professional machines produced in the last few years. The Technics, although scaled down for a "consumer" machine, is almost a direct copy of this excellent transport.

Secondly, Len Feldman's comments in the Sansui BA-5000 review concerning sound reinforcement speaker quality, though justified in some cases, should not lead to a description of sound reinforcement as "that less than hi-fi application," nor to the conclusion that a high-priced, high-power amplifier such as the BA-5000 would be out of place in such an application. Many of the post-concert headaches suffered by an audience are caused by *not* using amplifiers with as much power capability as the BA-5000, resulting in systems that cannot supply the severe peak power required to cleanly reinforce a "live" contemporary musical program.

As a sound mixer in a major theme park as well as a recording engineer in a professional studio, I can state that a properly constructed stage sound system can indeed be high fidelity and can provide "the kind of distortion-free SPL levels that studio personnel and recording artists (and may we add concert audiences) prefer."

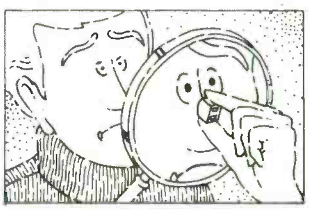
Our normal stage sound set-up is a bi-amped system using Electro-Voice Sentry IV low frequency cabinets (modified with EVM-12L drivers) and E-V HR-Series horns. Power is provided by Crown DC 300A's and BGW 750's giving a potential 150-watt RMS to each driver or cabinet. The system is set up, phased and equalized with a one-third octave band analyzer and filter set. With careful adjustment we can usually achieve a smooth response curve on the order of 50 Hz to 5 kHz ± 3 dB with a 6 dB/8 va [*we assume Mr. de Ganahl means 6 dB per octave*] roll-off above 5 kHz. We sometimes pull out our favorite Sheffield Disc and find the system is capable of providing clean average levels of 105-110 dBA at the center of the audience area, about fifty feet from the center of the stage. Needless to say, this level is for testing only, most of our show levels average 90-100 dBA depending on the artists. High fidelity sound reinforcement systems are available to anyone who is willing to take the time to properly design the system and to pay the price for the equip-

ARE YOU BLAMING YOUR TAPE RECORDER FOR PROBLEMS CAUSED BY YOUR TAPES?

Every day people all over the country go into hi fi dealers with complaints about their tape recorders.

When in reality what they should be complaining about is their tapes.

Because the fact is, a lot of the problems that plague tape recorders can be attributed to bad tape.



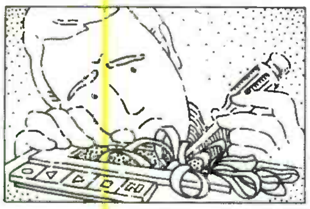
HEAD WEAR IS CAUSED BY YOUR RECORDER. OR IS IT?

If you have to clean your tape heads more than usual, for example, it could be your tape doesn't have a special nonabrasive head cleaner.

Maxell has one.

If your recorder jams, it can be any number of things. Maxell does something to prevent all of them.

We make our cassette shells of high impact polystyrene. And then so they won't crack



JAMMING IS CAUSED BY YOUR RECORDER. OR IS IT?

even after years of use, we finish them to tolerances as much as 60% higher than industry standards.

Inside, we use free rolling Delrin rollers so the tape doesn't stick.

And finally, we screw instead of weld everything together because screws make for stronger cassettes.

If your recorder frequently suffers lapses in sound, it could be the tape is of inferior quality. And nobody's bothered testing the tape for dropouts before it leaves the factory.

DROPOUTS ARE CAUSED BY YOUR RECORDER. OR ARE THEY?

Maxell tape is made of only the finest polyesters. And then every

POOR TRACKING IS CAUSED BY YOUR RECORDER. OR IS IT?

step of the way it's checked for even the slightest inconsistencies.

So if you're having problems with your recorder, try a Maxell cassette, 8-track or reel-to-reel tape.

You might find there's really nothing wrong with your tape recorder, just with your tape.

MAXELL. THE TAPE THAT'S TOO GOOD FOR MOST EQUIPMENT.

Maxell Corporation of America, 130 West Commercial Ave., Moonachie, New Jersey 07074.

CIRCLE 73 ON READER SERVICE CARD

ment needed to fulfill the design.

Enough of the minor criticisms for they have only provided an excuse to write this long-intended letter. As one of the first RIA graduates and a two-time graduate of Don Davis' excellent SYN-AUD-CON seminars, I have followed your magazine since its inception in October 1975 and have found it a most welcome addition, perfectly filling the gap between the highly technical journals and the layman's stereo magazines. MR is extremely interesting, informative, and well-written. Perhaps the fact that it has taken this long to find enough to complain about is in itself a comment on the overall excellence of your magazine. I look forward very much to receiving MR once a month.

—Frank A. de Ganahl
Stage Operator
Walt Disney World
Lake Buena Vista, Fl.
Recording Engineer-Technician
Bee Jay Recording Studios
Orlando, Fl.

We are delighted to respond to Mr. de Ganahl's letter.

We think the key paragraph is the last one. Firstly, we never said that there were no other open-reel machines that utilized the principles of the Technics RS-1500US, simply that we had not run across them before, and that is true.

As for sound reinforcement generally being a "less than hi-fi application" of audio, we were, of course, generalizing and the fact that Mr. de Ganahl had to spell out his "exception" sort of confirms the generalization. Our own experiences at concerts—outdoor and indoor—causes me to maintain that more sound mixers at "live" events ought to equip themselves with the training and background of such individuals as Mr. de Ganahl. If one would have to rank sound quality generally, I still think the order of things would be (1) the pro recording studio, (2) the home hi-fi system and (3) the auditorium sound reinforcement system.

—Norman Eisenberg and
Len Feldman
Audio Editorial Board
Modern Recording

Equipment Resale?

Recently, I have been reading a lot about sound reinforcement companies. An example is the article in the latest (Dec/Jan 1977) MR on Clair Brothers Audio in Lititz, Pa. It would seem that they are forever repairing their equipment and when repairing becomes uneconom-

The Precision Decision. We made it. Now it's your turn.

We believe that precision is the most important factor in turntable design and performance. Which is why we've built such a high degree of precision into our advanced new line of turntables. So you'll need a whole new set of reasons to choose the one that's right for you. And when it comes to value, all seven will play second to none.

Take our new QL-7 Quartz-Locked and JL-F50 Fully Automatic direct drive, shown above. They're both unusually close when it comes to some important specs, but what will surprise you most is that they're also both in the same price range.

For instance, the JL-F50 checks in with 0.03% wow and flutter (WRMS); 70dB signal-to-noise ratio (DIN B). And it offers a host of convenience features as well, with most controls up front so you can operate them without lifting the dust cover. Its fully automatic operation gentles your favorite records, and lets you repeat them from one to six times, or infinitely. A built-in strobe makes speed adjustments easy and accurate. And the JL-F50's looks are in keeping with its precision design.

The QL-7's looks are equally great. And in its electronic heart, it's a tiger. All business, with the incredible accuracy only a Quartz-Locked machine can boast. Truly for a perfectionist, the QL-7's wow and flutter measures only 0.025% (WRMS); S/N is more than 74dB (DIN B). Figures that no other QL turntable we've seen in its category can touch. It's totally manual, with strobe speed indicator, and priced less than any other QL machine on the market.

The way we see it, you're left with a superb decision: our JL-F50 at less than $250*, with all the convenience and performance most people could ever want, or our QL-7, the finest under $300* turntable available today for the discriminating audiophile.

Either JVC you choose, you'll have made the right decision.

JVC America Company, Division of US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, N.Y. 11378 (212) 476-8300. Canada: JVC Electronics of Canada, Ltd., Scarborough, Ont.

For your nearest JVC dealer, call toll-free (outside N.Y.) 800-221-7502

*Approximate retail value.

JVC

We build in what the others leave out.

CIRCLE 65 ON READER SERVICE CARD

ical, replacing their sound equipment. My question is whether the equipment they are replacing is available for purchase by the general public?

Also, is it possible to buy equipment from manufacturers, that did not pass their final inspection? Again, it may be uneconomical for a company to repair such equipment. Could you outline steps one would take in obtaining such equipment?

—Michael Gallo
Weehawken, N.J.

For information regarding the purchase of equipment that Clair Brothers may be replacing, they have asked us to refer you directly to them at Clair Brothers Audio, RD1, Lititz, Pennsylvania, 717-733-1211.

Unavailable Issue

In your last issue (Feb/Mar 1977), you have a very interesting article on monitors by Rob Lewis. In it, he made reference to an earlier article on sound absorbers and traps. Since I am presently battling this very problem, I would like to have a copy of the article or the magazine in which the article was printed. I am a new subscriber and have only a couple of issues.

I shall greatly appreciate your assistance in this matter.

—Henry L. Brooks
Los Gatos, Ca.

The article you are referring to is "Building your Own Recording Studio for Under $500" by Jeff Cooper. Unfortunately, the Dec/Jan 1976 issue of MR in which it is contained, is unavailable.

Compliments Galore

I'd like to compliment you on the fine work you have done in your publication. It's very entertaining as well as being very informative. Well done!!

Thank God for *Modern Recording* going monthly. Keep up the good work.

—Harry Yee
Irvington, N.J.

Talkback Chatter

I enjoy your magazine immensely. Could you expand your Talkback section into a monthly special or even a separate but larger section? So far nothing has been published that deals as an open forum. It's always been limited and you could never be sure of getting an answer. I know myself that I have many questions that I feel I should find an answer to. I can usually write directly to manufacturers but sometimes I wish I could place the question to people who have had experience with the problem "on the road," so to speak.

I am into P.A. systems and am currently stage manager for a band. I have been into playing and working for bands for 12 years and I am self-taught. But there are still a lot of questions.

—Robert DeMoss
Fayville, Ma.

Talkback is an important part of MR because it does provide the opportunity for readers to have their questions answered from professionals in the field. Since our first issue, Talkback has answered hundreds of such questions and we attempt to get all of them answered.

We feel that all the staples like Ambient Sound, Musical Newsicals, Lab Reports, the Product Scene, Groove Views, and our various feature articles, are equally as important as Talkback for providing our readers with the information they are looking for.

Truthful Testings

Maybe you can help me decide about a subscription to your magazine. First, it is informative and handy, but second, please tell me about your Test Reports. Are they "on-the-level" Test Reports or are they bullshit advertisements?

Please set me straight with a reply and then perhaps my friends and I can set MR straight with a few subscriptions for you.

—Robert J. Von Schleusingen
Fairfield, Ct.

Set your mind at ease. All of our Lab Reports are legitimate and are based on actual test findings. If you look through MR, you will find that many of the manufacturers of the pieces that we do test do not even advertise within the magazine. I hope that we have set you straight because we stand by every Lab Report that is reported in MR.

Writer's Fan Mail

After reading Ed Garnier's editorial in your Dec/Jan 1977 issue, I decided to write him a letter hoping he may possibly be able to answer some questions I have that are very pertinent to his essay. Please forward my letter to Mr. Garnier.

Keep up the good work on MR. It's just the magazine I was waiting for.

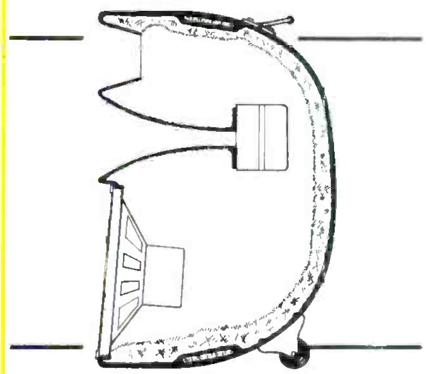
—Mark D. Kerm
No address

Your letter has been forwarded to Ed Garnier as you requested.

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 flake-board 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" - 1⅜"	12"
PBL	1" - 1⅜" - 2" - 2.8"	15"
GGM	n.a.	TWO 15"

Here's Looking at You, Kid!

Community

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CIRCLE 36 ON READER SERVICE CARD

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.*

A Clean Noiseless LP?

In doubling an instrument with a delay line (digital) in multi-track situations, would it be better to have the doubling on a track of the original recorder or save that track for something else and just use the delay during the mixdown to the master two- or four-channel tape?

I collect records and have found that it is less expensive to purchase a 45 rpm record of a song I want than to buy an album. When I get a clean, on-center 45 rpm stereo copy, the signal-to-noise ratio usually seems better than the album cut. My question is this. Since the 45 rpm verson is abbreviated from the longer album version, is the 45 version taken from a tape that was derived from the master? In other words, on the shorter 45 rpm versions, am I actually getting one or more *extra* generations of tape hiss and/or distortion?

On clean, noiseless albums when you listen carefully between cuts to hear where tape ends and begins, the sound just starts with absolutely no hiss. Is noise gating used or are the professional tape machines with noise reduction just that quiet?

—Linc Chamot
Brook Park, Ohio

The answer to the first part of your question really depends on several things. How much do you really need that extra track? Are you certain that you really want that doubling effect? And how complicated is the mixdown going to be?

If you have more than enough tracks, go ahead and use it for the delay signal. If you have the uneasy feeling that someone is going to walk in with an extra horn section and didn't tell you about it, keep the track open. There is another option and that is to mix the delay signal on with the real-time signal. But this is obviously hazardous because once you do it you're stuck with it and the effect is locked in. So should you have second thoughts about the mix proportions or delay time or maybe it wasn't such a good idea in the first place, you can't change it short of re-doing the whole track over. If the mixdown is the kind that would give an "octopus" problems, it may be better to put it on the multi-track and be done with it. However, if the mix is a piece of cake, save the delay for the mix session. You will have one less track of noise to mix and *all* your delay options are open, such as how much delay to real-time you want to use, how long a delay, etc. If the delay time you chose during the recording turned out to be less than ideal, you are stuck. Adding delay during mixing gives you a better perspective and you can adjust to the best advantage.

In answer to the second part, it is impossible to say whether the 45 rpm is a copy of the album cut or vice-versa for any given record. Each record company and record producer has their own policy and procedures. Sometimes a group records a single first and then subsequently makes an LP into which the single is inserted. Sometimes an LP is recorded and one song proves popular enough to produce a 45 rpm record. Not only that, but both could be "first generation" masters. Most contemporary music is recorded multi-track first and the master tapes are derived or mixed down from that. These are considered to be first generation although they are really second generation. So actually, you can have several master tapes of the same generation. These masters can be edited into any length—album length, single length, AM radio length, or disco length.

Confused? So are we. Not only that, but some producers make separate mixes suited for earphone listening, AM, FM, TV, singles, discos, albums, quad, etc. Fortunately their number is few. But whether you are getting first or second generation is a negligible consideration since a tape copy with noise reduction is difficult to distinguish from the original. What you are probably hearing are differences in the quality of vinyl and level between a 45 rpm and an LP. You may notice that an average 45 rpm is about 4 dB louder than an average LP, thus there is 4 dB less noise on a 45 rpm than on an LP given the same quality vinyl. However, there should be more distortion on the 45 rpm than on the LP since the "hotter" level presents a much more difficult groove to trace for the playback stylus in your record player.

A clean noiseless LP! Where? What? Who? I'll buy it!! Unless you've made a typographical error, your question implies that you have in your posession a quiet LP! I haven't heard one of them since before the alleged oil crisis. Actually, the reason records are quieter between songs is that there is no recording tape at those points. A master tape intended for LP master recording has paper leader tape between the selections, at the beginning and the end. There are four good reasons for this. 1. Leaders help you find different sections easily. 2. They preclude the possibility of print-through on the tape. What you hear on records is another problem peculiar to disc mastering and pressing. 3. There is of course no tape hiss. 4. The leaders make the use of automated disc mastering equipment possible. Thus, when the music starts, the hiss starts too. It's just that you can't hear it because the louder music masks it. Even so, professional tape machines with or without noise reduction are that quiet. Noise gating is not used to any great extent for that purpose. What you hear between songs is pure electronic noise levels plus whatever noise is inherent in the vinyl medium.

Incidently, the vinyl is the weakest link in the chain of noise accumulation. Usually the pressing is magnitudes noisier than the professional recording and mastering sequence. It may interest you to know that a master phonograph record without noise reduction is quieter than most professional tape recorders *with* good quality noise reduction systems. And present day recorders are quiet!

—Dave Moyssiadis
Frankford/Wayne Recording Labs
Philadelphia, Pa.

Distortion Problem

How about a report on the Sound Workshop No. 220 Vocal Doubler? We have one with a distortion problem if used with the full 40 ms delay. Is this common for an analog system?

—Dave Molter
Brighton, Pa.

The Sound Workshop 220 Doubler/Limiter utilizes analog circuitry to obtain its delay characteristics. At this point in time, digital systems cannot be manufactured at the price level of the 220 and the "bucket-brigade" IC is still the key to low cost delay systems.

You should not be experiencing any distortion problems even at the full 40 millisecond delay setting. Perhaps you own one of our early production units (below serial number 1200) that did not incorporate the limiter in the front end. We would be happy to modify your unit, free of charge, if you return it to us (1040 Northern Blvd., Roslyn, NY 11576).

We are currently experimenting with both analog and digital techniques for use in our next delay product.

—Michael Tapes
President
Sound Workshop Professional
Audio Products, Inc.
Roslyn, N.Y.

Rattling of the Drums

In our studio we have a fourteen-piece drum set. The bass drum is twenty-two inches in diameter and has both heads. We cut a 4½" hole 3½" off center in the front head, then placed an EV RE-20 completely inside. The mounted toms and floor toms, ranging in sizes from 8x12 to 16x18, are miked with Sennheiser 421's, EV RE-15 or 20's. These mics are also used on the added concert toms. The snare and hi-hat are miked with AKG 451-E's. The overhead mics are EV CS-15's. The result is a decent sounding set that is rich in tonal quality, with good contrast and separation between each piece. The only problem is the buzzing and rattling of the snares. Not only when the snare is played, and also when the other drums are played as well. Presently, we put a cloth between the bottom of the snare and the floating snare frame. Although this helps considerably, the sound of the snare is destroyed. It would be greatly appreciated to know how the professional audio engineer's overcome this problem.

—Gary Guthrie
Blue Goose Sound, Inc.
Fort Smith, Ark.

This is a tough question to answer briefly. I don't have any idea how much "buzzing and rattling" you consider to be too much. Although it's possible to eliminate altogether, the sympathetic rattling from the snares on our Ludwig Drum, the amount of tightening of the snares that this requires, might be unsatisfactory for the over-all sound desired.

No matter what kinds of music you listen to or record, we can probably agree that the difference in sound and style between Tower of Power and Led Zeppelin is considerable. The former has a very "tight," close-miked drum sound, while the latter is more like a "live" experience even on a studio recording.

You mention the type of overhead mics used, but not how far over the set you use them. Are they so close to the cymbals that they are essentially cymbal mics, or so high above that they actually add room ambience? What I'm getting at is how much of your drum sound do you typically derive from your overheads and how much from your close mikes? Even with a fairly noisy snare, much of the unwanted rattle can be minimized (if not eliminated) by closely miking the top head of the drum. It's frequently not until you mix in a good bit of your overheads that you are even aware of a noisy snare.

There are a lot of variables I still haven't touched on here. If you would like to provide more details, please write or call me, and perhaps I could be more specific.

One closing thought: Take a good listen to your drums *in the studio* before you even bring up the faders. A set that's poorly tuned, or a snare drum with loose or broken snares isn't going to magically result in the Ultimate Drum Sound by "fixing" it in the control room.

—John J. Bradley
Vice President
Ultra-Sonic Recording Studios, Inc.
Hempstead, N.Y.

Impedance Differences

I'm assembling a small studio centered around a TEAC A-3340S, a dbx 154 noise reduction device, a single-channel compressor/limiter, and a Shure SR101 8 in/1 out audio console and I have the following question concerning some of my considerations.

The dbx's 100 ohm outputs feed 100 K ohm inputs on the TEAC, whose 10 K ohm outputs in twin, feed 50 K ohm inputs on the dbx. Do the impedance differences of inputs and outputs affect the noise reducer's function? If so, how can it be corrected?

—Stephen Kayser
Stamford, Conn.

Input and output impedances are most often confused. A simple rule is this. Any output impedance (signal source) will drive an input that is the same impedance or greater. For example, a 100 ohm output will easily feed any input that is 100 ohm or more. There would only be a problem if an output were to drive an input with an impedance less than its own. For example, a 100 ohm output would not be happy driving a 50 ohm input. Again, any output is happy feeding any input that is the same impedance or greater.

—Larry Blakely
Director of Marketing
dbx, Incorporated
Waltham, Mass.

Absorbing Vocal Booth

First of all, let me express my gratitude to the staff of *Modern Recording* for a worthwhile and informative publication.

My question involves the construction of a vocal booth. My control room is well isolated from my rather small (12' by 6') studio which is acoustically very absorbent. My two spring reverb units do their best to simulate ambience in my vocal recordings, but I'd like to obtain a more realistic vocal sound. I have an unused plaster-walled cubicle adjacent to the studio which measures eight feet long and six feet wide with an eight foot ceiling. What acoustical treatment would be most conducive in utilizing this space for vocal recording? Also, considering the size of the proposed vocal booth, is there any way in which I could obtain some amount of delay in the booth aside from the means I have in my control room?

—Boyd T. Wheeler
Arabesque Productions
Long Beach, Ca.

Your symptoms are typical of many small studios. The response of a room can be thought of as decay time. A small room like yours in an untreated acoustic condition has innumerable peaks and dips in its decay or response characteristics. The studio room that you describe as acoustically absorbent is only absorbent at certain frequencies, probably only those commonly thought of as high or upper mid-range. Sound below this area is probably not affected by the absorbing devices you are using. This room character is probably the dominant factor in your vocal problems.

Part of the answer to this question can be found in the reader's statement of wanting to "obtain a more realistic vocal sound." Keeping this in mind as we look at the reader's further request for generating delay (reverberation) in a new vocal booth, we must make a decision to either make an echo chamber or a vocal booth.

Going back to the "realistic vocal sound," the choice would seem to be one of broad band vocal booth rather than an echo chamber. The size of the room is in itself a very limiting factor. It can be minimized to a small degree by making it as absorbent at as wide a range of frequencies as possible. Traps consist of free hanging sound board covered with Fiberglas. The remainder of the surfaces should be treated fifty-fifty with hard and soft surfaces.

—Glenn Phoenix, President
Westlake Audio, Inc.
Los Angeles, Ca.

How the Pros Do It

I really enjoy your magazine and the helpful answers in the Talkback column. My problem is how do I eliminate the click made on the tape when I punch-in to correct a wrong note?

I realize I am another amateur asking another dumb question. I have a TEAC 3340S, a TEAC Model 2 mixer, and AKG C451 mics. I really love this equipment and it works fine. Recently, I drove ninety miles to record on a large pipe organ and a technically difficult piece about five minutes long. It took me four

Home Cookin'!

See that guy at the board? Once upon a time he was an engineer at the busiest studio in town. The place had everything big money could buy. And it cranked out super-slick albums at an absolutely psychopathic rate. But because its hourly rate matched its image, it wasn't only the busiest studio in town, it was also the most expensive. Which was alright if you had a fortune to spend—which the band you see here didn't.

After years of being a staff engineer he decided he'd been sitting behind somebody else's board long enough, thanks. So with the money he'd saved, he invested in a complete TASCAM Series recording studio by Teac—80-8* eight track, 25-2* two track, mixing consoles—the works!

Two days later, he was making tracks like they'd never been made before—in his home! And at a fraction of the price charged by his former employer.

Well, the band cut a demo at his new studio and with it they got a record deal. And with the front money, they invested in their own Teac mini-studio. So with the band members taking turns at the board, they laid down the tracks for their album. And to make sure they got the most out of the tracks they made, they asked the old pro to do the final mix.

Could this story have happened without Teac recording equipment? Not on your life. But it's the sort of thing that's bound to happen whenever a second generation engineer and a second generation band team up with a new generation of recording instruments.

TASCAM SERIES BY TEAC

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

*Nationally advertised value. Model 80-8 tape recorder shown above, less than $3,000. Model 25-2 tape recorder also shown, less than $1900 (Rolling Consoles not included). Actual retail prices to be determined individually at the sole discretion of authorized Teac Tascam series dealers. Prices subject to dealer preparation charges where applicable.

TEAC Corporation of America, 7735 Telegraph Rd., Montebello, CA 90640 ©TEAC 1977 hours to get a perfect take with no flaws or mistakes. I thought how nice it would be to be able to correct one note or even do two pages over by punch-in. I recall in one of your "History of Recording" articles, that with modern equipment it is not necessary to play the piece perfectly. Yet I am stuck doing just that. How do the pros make a correction on a single note of a solo instrument or voice? Can it be done on a TEAC 3340S?

—James R. McNiel, M.D.
Fort Dodge, Iowa

An "oversimplification" of the punch-in procedure can be very misleading, as your letter verifies. The ability to punch-in depends upon factors involving (1) the features of the mastering tape recorder, (2) the type of instruments involved and the rhythmic structure of the music at the point of punch-in, and (3) the acoustical quality of the recording environment.

(1) *The Mastering Recorder.* Two machine features are absolutely necessary for undetectable punch-in/out. (a) Sufficient switching logic or manual switching capability within the recorder to monitor sync/playback up to the point of punch-in; then monitor machine input (console or mixer output) while recording the part(s) to be corrected; then simultaneous punch-out of record and switch back to sync reproduce monitoring. (b) The ability of the recorder to enter record mode directly from sync reproduce mode without introducing a "pop" or transient "click" onto the tape; then to return to sync reproduce or transport stop, again without introducing noise. A 3340S will not silently switch a single channel directly from sync to record. The only procedure for quiet, rolling punch-in on a 3340S is:

(1) Set channel(s) to be punched to RECORD MODE ON (ready)
(2) Monitor sync reproduce up to punch point
(3) At punch point, flip simul sync switch to NORMAL
(4) Place transport in RECORD
(5) Simultaneously switch appropriate channel output select from TAPE to SOURCE

Silent rolling punch-out is not possible. To punch out, you must enter PAUSE mode, thereby stopping the transport, then disengage RECORD.

This is obviously a fancy maneuver and pretty near impossible if you're playing "engineer" and "artist" at the same time. A machine like a Tascam 80-8 makes it a lot easier to do because it has automatic function switching logic, and it requires only that you select the channel to be punched into record when the time comes, and switch back to play when you're finished. All monitor switching is automatic.

(2) *Instrumentation and Rhythmic Structure.* Any instrument or group of instruments which produce a lingering resonance such as a harp or piano using a sustain pedal, require that the ring-out be permitted to diminish prior to the punch so that the instant of silence between the punch and pick-up is not noticeable. A punch in/out should occur at a natural rest or on a definitive beat in the music.

Punching at any other point requires that the level and intonation of the segment to be replaced and the new part be perfectly matched, and that the tape recorder switch instantaneously. This kind of punch is not generally recommended on semi-professional machines or outside the studio environment.

(3) *Acoustical Quality.* Acoustical quality is usually a factor only in location recording or where the reverberation factor of the room is evident in the track being recorded (as it is with a pipe organ). Punching is usually performed on dry (no echo) tracks, for the same reasons mentioned above. Therefore, punching a solo instrument or a voice in a "live" environment is not recommended.

One solution to the problem of dealing with solo instruments is good old-fashioned editing. Simply pick up a few bars *before* the part to be corrected. Stop a few bars after and splice the good piece in at a musically convenient place.

—Norman J. Cleary, President
Audio Innovators, Inc.
Pittsburgh, Pa.

Hotter Records than Normal

What is meant by the term "hotter" when referring to pressing records?

—Frank Waterson
St. Louis, Mo.

To answer this question correctly, I am going to have to give you a fast course in disc mastering.

First, let's clear up one error on your part. When you use the term "hotter," you are referring to the disc mastering portion of record manufacturing, not the pressing itself. The pressing is just the mere duplication of the master record.

At the mastering end, it is the job of the disc mastering engineer to take the final stereo mix, which usually ends up

in the form of a master two-track stereo tape, and transfer this to disc by "cutting" a stereo record called a "master lacquer." It is from this master lacquer that the metal plates are made, and from these plates the finished vinyl records will be "pressed."

When mastering a record, the engineer has to take many things into consideration. Two of the most important considerations are the length in time of each side of the record and the type of music (classical, hard rock, acid rock, etc.,). Let's take an LP. You can see that you have the same amount of space into which you can fit grooves whether you are cutting a fifteen-minute record or a twenty-five-minute record, except that you have to fit more grooves into the same space for the longer record. This means the grooves will have to be put closer together. The longer the record, the closer together the grooves are.

Now to the word "hotter." This pertains to the amount of audio level on a disc. When we transfer the music from the master tape to the master disc, grooves are cut onto this disc. The grooves are made to wiggle or "modulate" as to the sound on the tape. The louder the sound, the larger the wiggles. When the grooves of a record are spaced very far apart, as in the short LP, you have a large distance that these grooves can wiggle before they touch each other. When the grooves start to touch, you have reached the limit. If you go any further, the record will begin to skip. In the case of the long LP record, the grooves are already close together, so there is not much room for these grooves to wiggle—thus the level of this disc has to be kept much lower than the short LP. In other words, you at home have to turn up the volume control of your amplifier higher to get the same amount of sound.

When a disc is mastered, normally it will be cut at a level somewhat less than maximum. This is done as a safety factor to insure that the record will track properly even on a cheap turntable. It also produces a better frequency response with less distortion. Knowing this, when you hear a person say that a record is "hotter" than another, or a producer says that he wants his disc to be cut "hot," it means that he wants to disregard the safety factor and he wants the record to be cut at the maximum possible level.

I hope this answers your question.

—Stewart Jay Romain
Custom Disc Mastering Engineer
Columbia Records
New York, N.Y.

Stereo Lost in Radio

When music is multed to mono on the radio, how much is lost on the left side? How much is lost on the right side?

—M. Ryan
Boulder, Colo.

I think the question could be more accurately stated—"What happens to the overall sound when a 2-channel source is multed to mono?" The answer is the same for both questions.

When a stereo program is combined to mono the result will be different than the stereo version. The amount of

Waveform fidelity. If you don't understand it, you could be making a $350 mistake.

Today a good amp and tuner can easily cost you $350, $500, $700 or even more. But no matter how much a component costs, if it doesn't have waveform fidelity, the music that's put into it won't be the music that comes out. And that's an expensive mistake to make.

It's also a mistake Technics won't let you make. Because Technics' two new integrated amps, the SU-7600 and the SU-8600, as well as our two new tuners, the ST-7600 and the ST-8600, have superb waveform fidelity.

With both tuners, the waveform being broadcast will be the waveform you'll receive, with virtually no distortion or cross modulation. Because both have flat group delay filters in the IF sections. So the time delay is constant for all frequencies.

There's also a Phase Locked Loop IC in the MPX sections. That's why, for example, with the top of our line, the ST-8600, you'll get stereo separation of 45dB at 1kHz and 35dB at 10kHz. And a frequency response that's as flat as it is wide. 20Hz to 18kHz (+0.2dB −0.8dB).

And with an 8-ganged tuning capacitor (5 for FM and 3 for AM) and a Technics developed 4-pole MOS FET, broadcasts with the ST-8600 will sound more like master tapes than FM.

You'll also find waveform fidelity in both amps. Including our most powerful one, the SU-8600. *With 73 watts per channel, minimum RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.08% total harmonic distortion.*

And the SU-8600 will stay 73 watts regardless of the power-hungry transient bursts found in many musical waveforms. The reasons: Sixfold independent power supplies for the control voltage and power amplifiers. The results: Virtually no transient cross-talk distortion. And optimum waveform fidelity.

So before you make a $350 mistake, or an even more expensive one, listen to our new amps and tuners. Your Technics dealer has them. Along with Technics waveform fidelity.

Cabinetry is simulated wood.

CIRCLE 23 ON READER SERVICE CARD

Technics
by Panasonic

variation between the two and to the degree that they are audibly different depends on a combination of factors. When multing a stereo source to mono. . .

(1) Anything panned to the center on a stereo mix tends to be boosted. This can be a positive aspect giving a tune more "punch."

(2) Phasing problems of some sort usually appear. Sometimes they are very obvious, sometimes less so.

(3) A lack of definition between instruments may result in a masking effect. For example, if the hi-hat on the right channel is louder than the maracas on the left, combining them to mono tends to feature the louder of the two.

The way to deal with the problem is to make A/B comparisons between stereo and mono as you mix and try to achieve an acceptable balance between the two.

—Evan Williams
United Audio
Santa Ana, Ca.

Monitor Feedback

Since I'm relatively a novice at sound reinforcement, I've encountered a problem with our monitors that I'm sure could be cleared up by someone more experienced. Our system consists of four monitor cabinets, each using Electro-Voice components which are: a 12-inch cone, T25 mid-range driver, 800 cycle passive crossover and 8HD horn front. A Soundcraftsmen 2212 Graphic EQ (one-half of which is used for the house system) and BGW power complete the system. Vocal mics are EV RE-15's, while instrument mics are AKG D1000 and D 190. All this is run from a Tapco 6100 board.

Our problem is this. No matter how much we experiment with speaker placement, the monitors always feedback between 640—2560 cycles. Even with the 2560 slider on the Soundcraftsmen cut the full 12 dB, the ring is present if the master monitor control is pushed past 2½ on the Tapco. We've tried varying the mix on the individual level controls and have moved the speakers themselves into every conceivable position, but the ring is still there, sometimes making usable monitor levels nearly impossible. We aren't a particularly loud band, but our monitor levels should really be higher.

My question would be: Are the RE-15's too susceptible to feedback to be of value as vocal mics in this situation? I've noticed that some people favor SM 58's for vocals. Could switching to these mics help solve the feedback problem? Or are our monitor components and placement to blame? Or, worse yet, is everything wrong? Any suggestions you can make regarding substitutions or different approaches will be greatly appreciated.

Also, it occurs to me that I should mention a situation with the Soundcraftsmen that has bothered me from the day we started using it. The "balancing" lights are very rarely in the "plus" position when the group plays, tending to jump to the bottom "minus" light when anything is played. There is no distortion in the system, but this bothers me. Should it? Your opinion, please?

Let me close by saying that Modern Recording is doing a real service for musicians and the recording industry. The list of questions you've answered since I became a subscriber is a mammoth. Thanks for your time and effort, and keep up the excellent articles and product reports.

I hope I haven't asked too many questions. My apologies to your staff and those who already know the answers.

—Dave Molter
New Brighton, Pa.

SOUND REPUTATION.

Gallien and Krueger are busy building a professional reputation. They're doing it by building professional sound equipment like the Gallien-Krueger 1000-1S power amp. Take a look at the 1000-1S *guaranteed* specs and you'll begin to see what we're getting at.

Specs can tell you a lot about a particular piece of equipment, but they can't tell you everything.

For example, the 1000-1S is the only professional quality amp that is rugged enough to be *truly* portable. We can say it because we *build* it that way. In addition to assembly, we do our own metal fabrication, build our own transformers, *and* print our own circuit boards. The 1000-1S is built to be rugged and reliable, in the rack, or out.

And when it comes to sound quality, once you optimize slew rate and minimize distortion – and *all* amps have some, even the 1000-1S – it's up to you and *your* ears. So we're not going to spend time *talking* about our sound reproduction. We want you to decide about that for yourself, at your Gallien-Krueger dealer.

The Gallien-Krueger brand of quality doesn't exactly come cheap, but we think you'll like the sound of the 1000-1S price too.

Guaranteed Specifications

Power	200 watts	RMS per channel into 4 ohms with both channels driven from 20 to 20kHz
	125 watts	RMS per channel into 8 ohms with both channels driven from 20 to 20kHz
Harmonic Distortion	$<$.05%	at 200 watts RMS per channel both channels driven from 20Hz to 20kHz into 4 ohms
	$<$.025%	at 125 watts RMS per channel both channels driven from 20Hz to 20kHz into 8 ohms
Intermodulation Distortion	$<$.025%	from .2 watt to 200 watts into 4 ohms
	$<$.01%	from .1 watt to 125 watts into 8 ohms
Power Bandwidth	±.25dB	from 20Hz to 20kHz at 200 watts RMS into 4 ohms
Frequency Response	±1dB	from 5Hz to 200kHz at 1 watt
Phase Shift	±15 deg.	from 20Hz to 20kHz
Damping	$>$400	into 8 ohms at 1kHz
	$>$200	into 4 ohms at 1kHz
Voltage Gain	30dB	
Sensitivity	.9v RMS	for 200 watts into 4 ohms
Slew Rate	40V/μ sec	
Input Impedance	25k ohms	
D.C. Offset	$<$10MV	adjustable to zero
Hum and Noise	110dB	below 200 watts RMS

Gallien-Krueger
504 B Vandell Way
Campbell, CA 95008

GK GALLIEN-KRUEGER

You'll hear from us.

CIRCLE 94 ON READER SERVICE CARD

The only thing more difficult than solving a sound reinforcement feedback problem is trying to solve one without hearing it first-hand, but here are some comments.

Regarding your ringing problem, let us say that you did not advise if this problem exists in one location if it continues regardless of what room or outdoor area you are playing in. Obviously, the room acoustics and placement of the mics, P.A. and monitor speakers play a major role in the feedback and overall sound of the system.

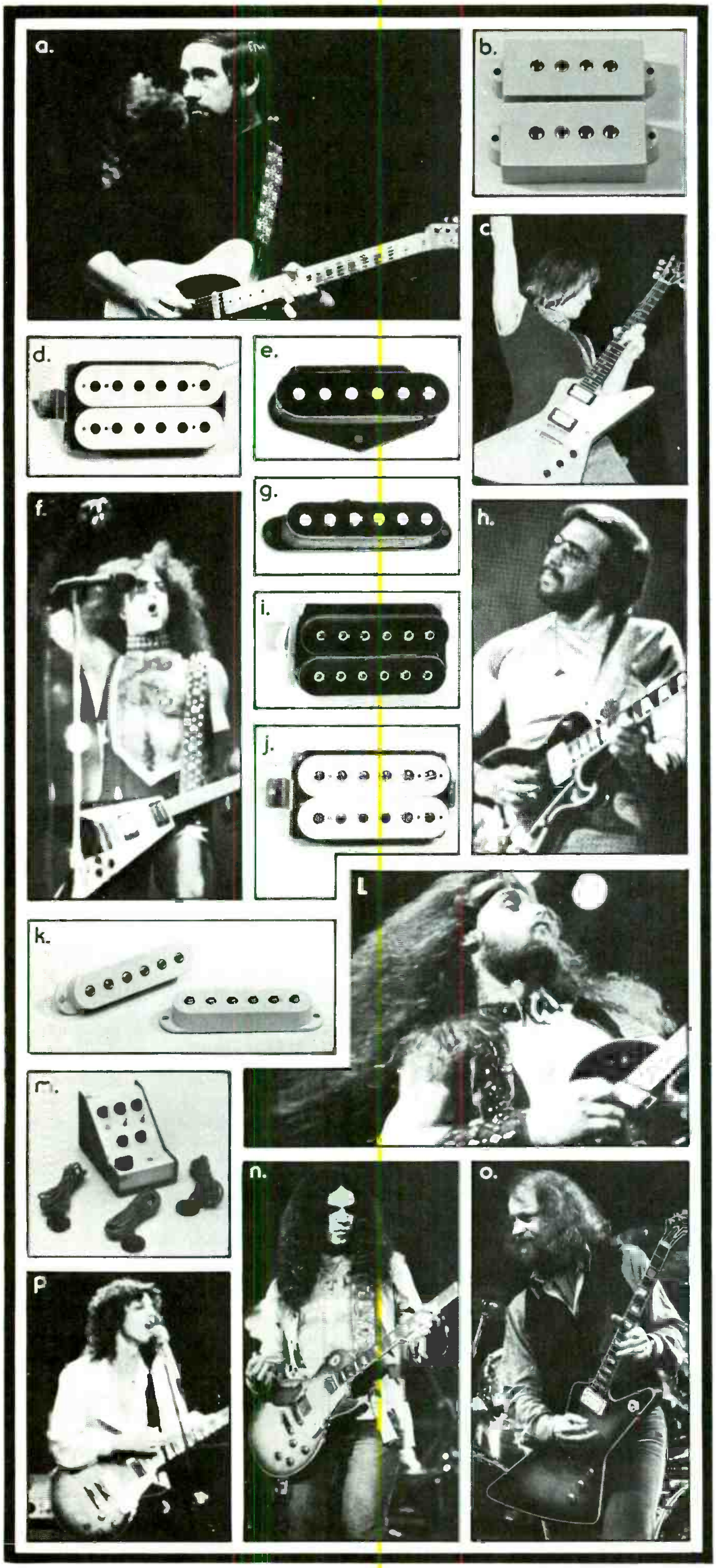
We would suggest that ideally all of your monitors be located in front of the vocal mics and above the heights of the microphones. This will serve as the first step to help you isolate your possible problem area. You can then experiment with alternate placement of the monitors.

In any event, if the ringing problem is so severe further cutting of frequency through equalization might eliminate the ringing, but would then result in removing too much program material along with the ringing. We urge you to contact Electro-Voice if our suggestions do not provide a definite improvement.

In answer to your question on the LED balancing circuit let us say the following. The LED balancing circuit on the RP2212 compares the input level of the equalizer to its output level. After you have set the individual octave controls, you should readjust the zero gain controls so that all four LED's are glowing equally. This should be done with music playing. All four LED's will continue to glow and to blink as music is played through the equalizer. If only the bottom lights glow you should adjust the zero gain controls until the top and bottom LED's are balanced. If only the top LED's glow, the same applies. It is possible you wouldn't hear distortion as this would depend on the next piece of gear in the chain *after* the equalizer. The whole idea behind the zero gain circuit is to eliminate the possibility of exceeding the input voltage capability of the equipment *after* the equalizer.

We hope the above helps, but if you have any more questions regarding the equalizer's operation feel free to give me a call collect at (714) 556-6191.

—Tom Thomas
Sales Manager
Soundcraftsmen
Santa Ana, Ca.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THE PRODUCT SCENE

By Norman Eisenberg

YAMAHA SHOWS ELECTRONIC CROSSOVER

Yamaha's model F-1030 frequency-dividing network is a professional electronic crossover, switchable for two-way or three-way operation. The device is offered for use in portable concert reinforcement systems, studio monitor systems and discotheques. Crossover frequencies are selectable at 250 Hz, 500 Hz, 800 Hz, 1 kHz, 1.2 kHz, 1.5 kHz, 2 kHz, 2.5 kHz, 5 kHz, 6 kHz, 7 kHz and 8 kHz. In addition, the crossover slope rates can be switched to 12 dB/octave or 18 dB/octave. Low-frequency speakers may be protected from sub-audio transients by the use of a switchable 40-Hz high-pass filter. Inputs include a pair of balanced XLR, and unbalanced phone, jacks. Each pair is wired in parallel for "chaining" to additional crossovers or other signal devices. Outputs also come to XLR and to phone jacks, and the XLR connectors have polarity-reversing to facilitate acoustic phase-matching. The F-1030 accepts signal levels up to +30 dB (24.5 volts). The output will produce up to +24 dBm (12.3 volts) signal level into a 600-ohm load. Three LEDs indicate +14 dB (3.88 volts) output; there is 10 dB of headroom left above the LED turn-on point. Rated response of the combined outputs is flat within +0.5, −1.5 dB from 20 Hz to 20 kHz. THD is given as less than 0.05% at +24 dB output from 30 Hz to 20 kHz; hum and noise are 76 dB below maximum output. The F-1030, in semi-gloss black and weighing 16.5 pounds, may be rack-mounted.

CIRCLE 20 ON READER SERVICE CARD

16-TRACK RECORDER

The model 90-16 is the newest of the second-generation TEAC Tascam series of professional equipment. A 16-track, 16-channel machine, it handles 1-inch wide tape at 15 ips, with optional dbx processing. A combination record-reproduce head allows full reproduce frequency response in the sync mode. There also is a separate monitor head, and an erase head. The transport system offers full logic for feather-touch operation. The direct-drive capstan is powered by an AC servo-controlled motor and is capable of variable speed operation of ± 30 percent. Output signal select buttons permit source calibration, usual record functions (recording, overdubbing or sync, playback) and monitor activation. Function select buttons automatically accomplish all monitoring combinations and the encode/decode switching of the optional dbx package, which is designed to increase dynamic range, S/N ratio and signal headroom.

CIRCLE 8 ON READER SERVICE CARD

IMPULSE NOISE REDUCTION SYSTEM

The SAE 5000 is an electronic device which, patched into a disc-playback system, eliminates pops and clicks caused by disc surface defects, scratches and static-charge buildup. It may be used for direct playback or when dubbing discs onto tape. The 5000 senses impulse noises (electronically known as "spikes") and deletes them from the musical program. Since a spike is usually no more than one-thousandth of a second in duration, it is possible to extrapolate forward a tiny segment from the musical information immediately preceding the signal-gap. In this way program continuity is maintained. The unit has a threshold control which varies the sensitivity of the spike-detection circuit. There's also a defeat button that allows the circuit to be bypassed. Since the model 5000 is designed for patching into the tape-monitor circuitry of audio components, it has suitable jacks and switching to replace that facility for the user. Weighing 6 lbs., and measuring 10¾ by 9¼ inches, the SAE 5000 is priced at $200.

CIRCLE 1 ON READER SERVICE CARD

ECONOMY AUDIO RACK SYSTEM

Designed to serve as a reasonably-priced rack but yet strong enough to take constant abuse is the new QSC Audio Rack, constructed of vinyl-covered wood with heavy steel corner reinforcements. Manufactured by Quilter Sound Co. of Costa Mesa, Ca., the 19-inch unit comes with thick, tapped steel rails and is packed with mounting hardware plus a cable-dressing kit. Options include a fan pack, casters, support brackets for heavy equipment and a soft cover. Dimensions are 19 by 22¾ inches inside, and 20½ by 25 by 16 inches outside. Weight is about 38 lbs. List price is $124.

CIRCLE 19 ON READER SERVICE CARD

PORTABLE OCTAVE BAND ANALYZER

Just made available is White Instruments' Model 150 Octave Band Analyzer, a device useful in sound system set-up, speaker placement, speaker check out, horn aiming, playback system and noise surveys. The Model 150—battery operated and hand-held—incorporates triple-tuned filters to meet ANSI 1.11 Class II specifications. Level in each of the ISO octave bands, centered from 31.5 Hz to 16 kHz, is displayed on an LED matrix. Display ranges are 14 dB and 28 dB for a resolution of 1 dB or 2 dB. Acoustic sensitivity ranges are calibrated from 34 dB/SPL to 110 dB/SPL. Either flat or "A-weighted" measurements may be made. A mic, detachable for remote measurements, is supplied with each unit. Also shipped with the model 150 is a model 151 pink-noise source, battery charger and carrying case. The model 150 will operate for about five hours between charges; the model 151 will run at either line or mic level for about 30 hours from nine-volt transistor batteries. List price is $1400.

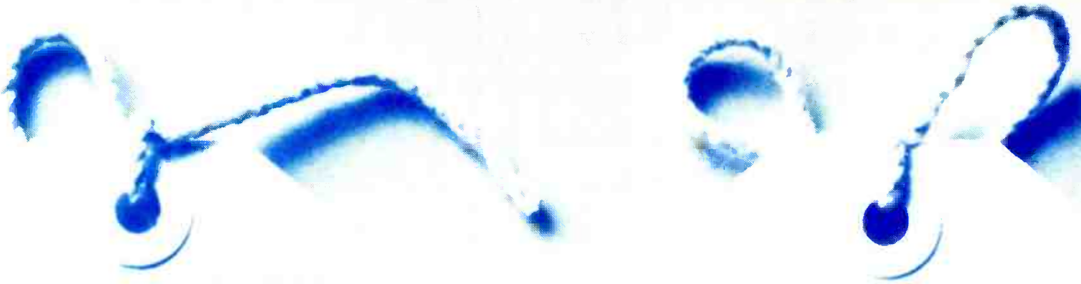
CIRCLE 14 ON READER SERVICE CARD

CELESTION OPENS U.S. BRANCH

Celestion, a British-based loudspeaker manufacturer, has set up a U.S. distributor, Celestion Industries, Inc., with headquarters in Holliston, Mass.

The line to be available in the U.S. will include six models priced from $159.50 to $499.50 and in various sizes and power-handling abilities. Celestion systems in general use high-quality drivers (made by the parent company in England) in various direct-radiating configurations, including some with passive radiators for bass enhancement. They enjoy high repute abroad. In Canada, the Celestion line will continue to be handled by Rocelco, Inc. of Montreal, Quebec.

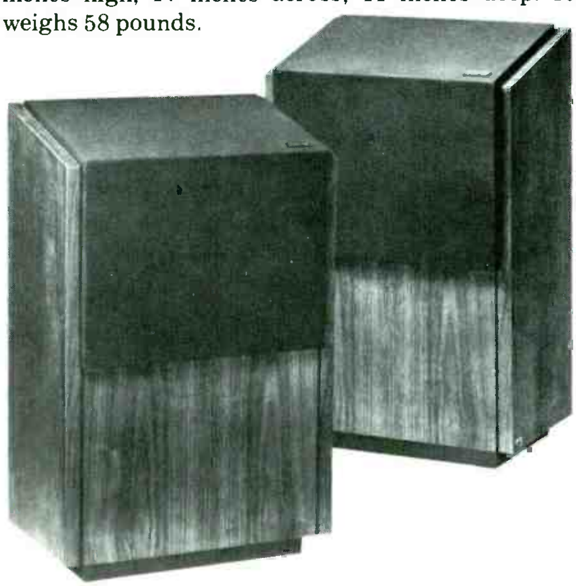
CIRCLE 7 ON READER SERVICE CARD



TWO NEW SPEAKERS FROM EPICURE

Epicure Products, Inc. has announced its model Twenty Plus, in which one set of woofer and tweeter radiates upward at a 15-degree angle, and a duplicate woofer and tweeter radiate outward from the front panel. The result, says Epicure, is "nearly hemispherical dispersion of 180 degrees from both firing planes." Rated response of the Twenty Plus is 35 Hz to 20 kHz, ±3 dB. Recommended power range is 20 to 100 watts RMS. Finished in walnut veneer, the Twenty Plus weighs 64 pounds. Dimensions are 29 inches high, 18½ inches across, 12 inches deep. The woofers are low-mass eight-inch units; the tweeters are the firm's new one-inch air-spring models featuring a tweeter-cap support of machine-hardened phenolic. Vis-a-vis the former version of this tweeter, the new unit is claimed to be capable of holding tolerances better than .005-inch, permitting each unit to be tuned to the desired resonance. The high-temperature voice-coil can operate at 200 degrees Centigrade versus 100 degrees for the earlier version. Price of the Twenty Plus is $275.

Another recent speaker system from Epicure is its EPI Division's Model 200, a more conventional looking system but one of relatively high efficiency. The Epicure EPI 200, priced at $225, uses an eight-inch woofer integrated with a 12-inch passive radiator. The 1-inch tweeter handles the middles and highs. Rated response of this system is 34 Hz to 15 kHz; recommended power range is 15 to 125 watts. Finished in walnut, the EPI-200 is 30½ inches high, 17 inches across, 11 inches deep. It weighs 58 pounds.



CIRCLE 6 ON READER SERVICE CARD

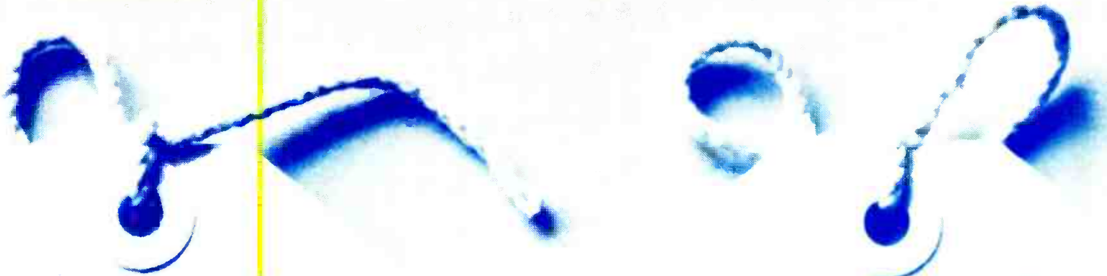
INFINITY EXPANDS PRODUCT LINE

Infinity Systems, Inc. of Canoga Park, Ca., has announced its entry into tone-arms and headphones. The arm, called the Black Widow, has an effective mass of only 3 grams. A cross-member on the lightweight, slender arm allows the pickup cartridge to be attached without the use of a head-

shell. This, and other design advances, have minimized mass—especially near the stylus tip—says Infinity. Antiskating provision, calibrated with respect to stylus force, is built into a lower extension from the arm pivot. The arm's cueing device is oil-damped. Price is $200.

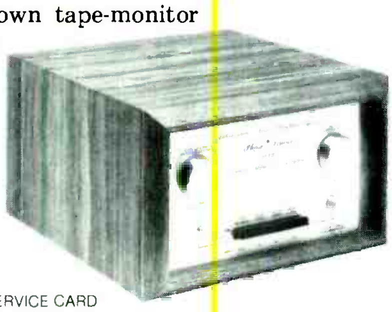
Infinity's headphones, the model ES-1, are an electrostatic type using an adaptor that contains power supply and matching transformers. Claimed response of the ES-1 headphone system is 20 Hz to 25 kHz within ±2 dB. Diaphragm material is a low-mass, conductive and lightweight Polyurethin which, claims Infinity, can follow signal changes better than polyester. Front-panel switching allows the headphones to remain connected whether they or the speakers are being used. The adaptor can accommodate an additional headset if desired. Sensitivity is rated for 98 dB/SPL at an input of 2 volts. At 100 dB output, harmonic distortion is less than 0.3%. Maximum output is 118 dB. The headset weighs 9 oz.; price of the system is $275.

CIRCLE 15 ON READER SERVICE CARD



NOISE AND DYNAMIC RANGE PROCESSOR

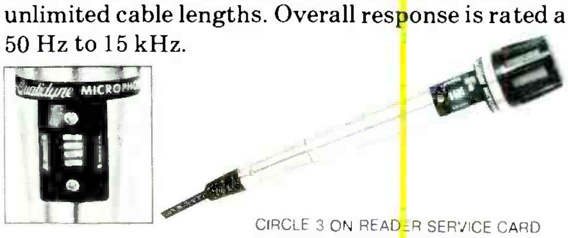
Phase Linear Corp. of Lynnwood, Wash. is offering the Autocorrelator portion of its previous model 4000 preamp in the form of a separate component, the Model 1000. In this device special circuitry is designed to improve music reproduction by reducing noise, dynamic range recovery and dynamic low filtering. Operating on any stereo source without the need of pre-encoding, the noise-reduction action is claimed to remove up to 10 dB of noise without affecting the music. For dynamic range the device employs three independent expander circuits tied together with logic. Expansion is selective and in small amounts, available up to 7.5 dB of increase. The low filter, designed to reduce rumble and hum, is adjustable. The model 1000 may be patched into a playback system via the tape-monitor path in which case its own tape-monitor facility may be substituted.



CIRCLE 9 ON READER SERVICE CARD

EQUALIZER MICROPHONE

Shure's model 516EQ unidirectional microphone is a dynamic type with built-in capabilities for equalization or feedback control. Four switches on the mic handle may be used to activate filters that attenuate response by about 6 dB at each switch frequency. In addition to response-shaping, the four switches provide up to sixteen different combinations of special effects, ranging from the elimination of nasal and sibilant sounds to emphasizing various instruments. By means of these controls, Shure says the 516EQ can be used to improve the quality of home tape recordings without the need for elaborate studio equipment. The mic's cardioid (unidirectional) pattern picks up sound from the front of the mic only, suppressing unwanted side and rear sounds. The unit's low impedance permits unlimited cable lengths. Overall response is rated as 50 Hz to 15 kHz.



CIRCLE 3 ON READER SERVICE CARD

SURROUND SOUND FROM TWO CHANNELS?

JVC has "reintroduced" the idea of genuine binaural recording in the form of a headset that includes, in addition to a good pair of listening headphones, a tiny electret condenser microphone imbedded in each earcup. When you use the device (model HM-200E) for recording, it "hears" "live" sounds with the same aural perspective that a human being would. You can wear the device when recording this way or you can drape it over a dummy head supplied in the package.

The results of the playback, wearing the headphones this time for listening, may startle you out of your seat (or make you sit down if you are standing), since virtually all of the "live" ambience, relative spacings and distances of sound sources, etc., are completely preserved and faithfully reproduced. Some listeners, with whom I have experimented, insist that the experience is more dramatic and convincing than anything they've heard in quadraphonic sound reproduced over four speakers.

Why this is so can be explained in psychoacoustic detail, but the effect is genuine, and it does work for an individual listener. The illusion of movement about the wearer is almost uncanny, and few can resist turning the head in the direction from which a particular sound originates, especially when you have deliberately recorded the sound to vary in distance from the microphones.

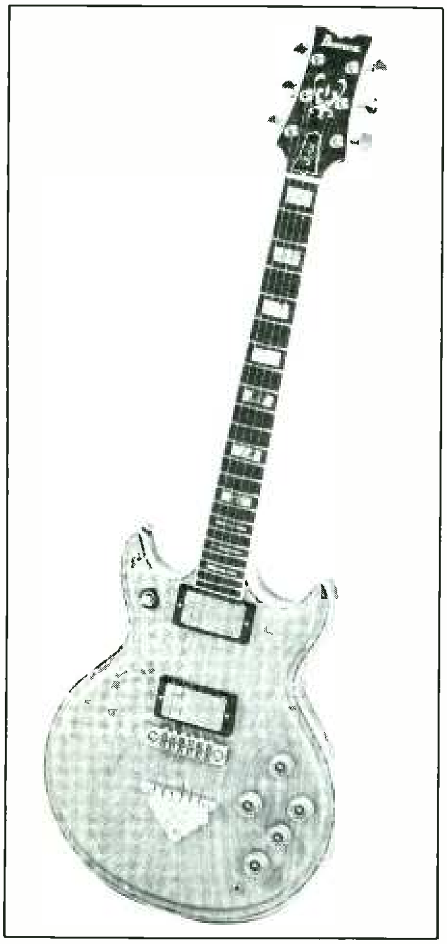
While demonstrating that true binaural sound can be more convincing (at least to an individual headphones wearer) than multi-channel sound reproduced over speakers, the JVC product also raises the implied question of just how far the latter technique can go in terms of material prepared for playback on speakers. For, even using the JVC binaural mics to record, if you play the tape over a speaker setup, all you get is traditional stereo—good sounding to be sure, but without that special icing-on-the-cake that is true ambience, or "live" room-effect illusions.

You can mull this one over and debate it far into the night, including what long-range effects it may have on the whole commercial recording field if these headphones become really popular. In the meantime, if you're interested in trying them, they are retailing at JVC dealers for about $80. The complete system includes the twenty one-ounce headset, penlight cells for the built-in mics, cables and connectors and the dummy head. Have fun.

MUSICAL NEWSICALS

This month's column completes the wrap-up of new products shown at the National Association of Music Merchants Western Market trade show in January, and brings us up to date on new products announced in the months since then.

INSTRUMENTS. . . The Ibanez Artist EQ is an interesting new electric guitar available from Elger Company (P.O. Box 469, Cornwells Heights, Pa. 19020) and Chesbro Music (327 Broadway, Idaho Falls, Idaho 83401). The guitar is a solid-body, double cutaway instrument with the unique feature of a built-in pre-amp and three-band equalizer. The



preamp section has input attenuation and preamp gain controls for precise control of presence, and has enough clean gain to overdrive any amplifier's input without adding any distortion of its own. The equalizer section has controls for bass, midrange and treble; each control has a ±12 dB range and has a detent in the "flat" position. Also provided is a switch which instantly bypasses the preamp and EQ circuitry. Power for the preamp comes either from a 9 volt battery mounted in the guitar, or from a regulated DC phantom power supply furnished with the guitar system. The phantom supply requires the use of a two-conductor shielded cable with stereo (tip-ring-sleeve) ¼-inch phone jacks, but normal single-conductor shielded guitar cords may be used in the battery-powered mode.

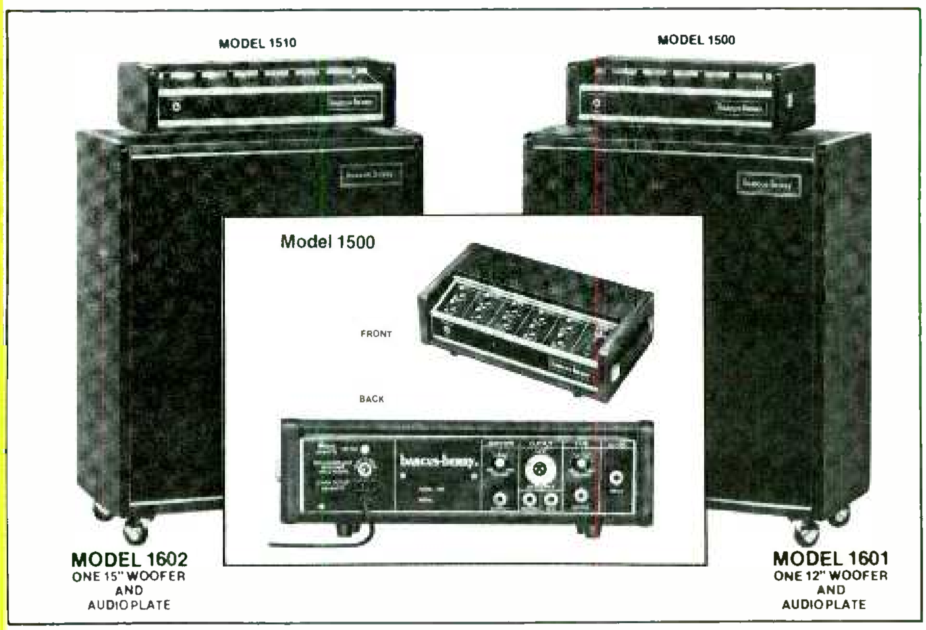
SOUND REINFORCEMENT. . . Sunn Musical Equipment Co. (Auburn Industrial Park, Tualitin, Ore.) has once again expanded their line of sound reinforcement products with the introduction of their low-cost Alpha Series mixing consoles. The Alpha Series currently com-

prises three high impedance microphone mixers with integral power amplifiers accomodating 4, 6 or 8 inputs. All three models feature volume, bass, treble and reverb controls on each input and utilize Sunn's Phase-Sync circuitry. Each model also has master volume and reverb controls and includes a 100-watt RMS power amplifier. The Alpha Four ($299.00) is a basic four-input mixer, while the Alpha Six ($499.00) is a six-input model which includes high- and low-gain inputs, Sunn's exclusive Auto-Patch feature and a three-band graphic equalizer in addition to the standard Alpha Series features. The Alpha Eight ($699.00) adds two more inputs and a five-band equalizer to the features of the Model Six.

Stage Amplifiers, now being distributed by Unicord, Inc. (25 Frost Street, Westbury, N.Y. 11590) offers a compact, six-input stereo mixer for $399. Each input of the Stage mixer features volume, bass and treble faders, stereo panpot, effects send and monitor send pots and a channel standby switch. The master section includes left and right master faders, two effects faders, two VU meters and low- and high-frequency compensation switches. Frequency response is given as 20 Hz—20 kHz ± 1 dB, and the signal-to-noise ratio is 65 dB. The standard version Stage mixer has high impedance inputs with ¼-inch phone jacks, but a professional version with low impedance inputs and XLR-type connectors is available at a somewhat higher cost.

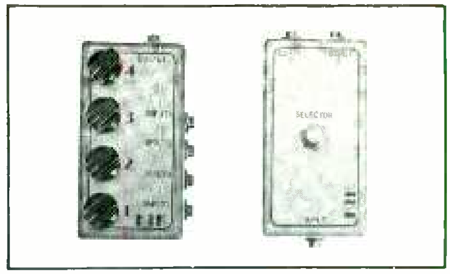
SYNTHESIZERS . . . Beckman Musical Instruments (P.O. Box 22289, East Los Angeles, Ca. 90022) recently announced the Roland System 100 Synthesizer package. System 100 uses a building-block approach and makes possible the assembly of a "complete synthesizer studio" for under $2000. The basic block is the Model 101 Synthesizer ($695.00). The Model 101 has a 37-note keyboard with provision for portamento, and provides full synthesizer functions with the following functional units included in its circuitry: VCO, VCF, VCA, ADSR, LFO, noise generator, high-pass filter, mixer, and a headphone amp. The model 102 Expander ($549.50) patches into the Model 101 to add the following circuit functions: sample & hold, a second ADSR, a second LFO, a second VCO, a ring modulator, a three-input mixer to combine VCO and ring modulator outputs with an external signal, a second VCF/high-pass filter combination and a second VCA. Also included are a level control for the low-level and high-level line outputs, a combining network to mix the outputs of the Model 101 and Model 102, and a headphone jack which carries the mixed signal. The Model 103 Mixer ($325.00) is a four-input stereo mixer for microphones or high- or low-level line inputs. Each input has a fader, a panpot and an echo/effects send pot with a switch to send either to the built-in reverb or to an external effects device. The master section includes a master fader, master balance control, reverb master, effects master, reverb/effects panpot, two VU meters, and amplifiers for headphones and small monitor speakers. The Model 104 Sequencer ($425.00) is a 12-step analog sequencer which has two sets of twelve knobs for presetting the voltage at each step of the sequence. The two registers can be accessed separately (parallel) or together (series) for maximum versatility. A voltage-controlled clock oscillator and various sequencer program controllers complete the Model 104. The System 100 package is completed with the Model 109 Monitor Speakers ($149.50 for the pair). These speakers are 16cm (6.3 inches) full-range, 8-ohm speakers designed to be driven by the amplifiers in the Model 103 Mixer.

AMPLIFIERS. . .Barcus-Berry, the pickup people, are now also amplifier manufacturers with the introduction of their modular sound systems (Barcus-Berry Sales Corp. 5782 E. Second Street, Long Beach, Ca. 90803). The systems are unusual in that the power amps are integrated with the speaker modules while the preamp is a separate unit. Three models of speaker cabinets are available; all three are two-way designs using Barcus-Berry's exclusive Audioplate treble speaker and include a 100-watt average power, solid-state amplifier in the top of the enclosure. Audioplate is a flat sound-radiating plate which operates on a molecular shock-excitation principle and has remarkable clarity in reproducing mid-range and treble frequencies. Model 1601 ($435) has a 12-inch speaker in a reflex enclosure in addition to the Audioplate, while the 1602 ($470) is a 15-inch reflex system and the 1603 ($495) is a 2x12-inch open-backed model. Three preamp/control units are also available, the most basic being the model 1500 ($325). The 1500 features an instrument input with both sensitivity and volume controls, a back-panel mixer input, a back-panel tape input with level control to allow mixing of "live" and taped music, conventional bass and treble controls plus midrange EQ with variable frequency, bass and treble expander circuits, high impedance (for connecting to power amp) and balanced low impedance (for connection to a mixer or tape recorder) outputs with separate level controls, record output, and monitor output with 1 watt output power into 8 ohms for headphone listening. To these "basic" features, the model 1510 ($435) adds reverb with adjustable decay time and provision for remote switching, echo send/return jacks, and accessory send/return jacks with level control and footswitch jack. The 1520 ($495) goes that one better by adding a phase shifter on top of all the other



features. The versatility of the control units and the choice of nine different control/speaker combinations makes the Barcus-Berry system a flexible one indeed.

ACCESSORIES. . .KIK, Inc. (6620 Whitsett Ave., North Hollywood, Ca. 91606) is a new company that currently offers two simple but useful accessory products. The KIK 4 to 1 Mixer ($35.00) is simply



a four-input passive mixer with a level control pot for each of the four inputs in a very compact cast-aluminum box. The KIK A/B Footswitch Box ($35.00) can be used to switch back and forth between two instruments, two amplifiers, two microphones, two effects devices, two speakers or two anythings, that are feeding or are being fed by the same piece of equipment.

Sigma Engineering (11320 Burbank Blvd., North Hollywood, Ca. 91601) is the Safety Products Division of the Norton Company, and they have come up with a product that should be of interest to rock musicians, namely a new type of ear plug. Sonic II Noise Filters attenuate high level impulse noises through a special acoustical passageway while allowing normal sounds to pass through. This means that musicians can still hear the music to help them stay in tune and in time while significantly reducing the possibility of permanent hearing damage. Additionally, Sonic II Noise Filters are open enough to allow ventilation of the ear canal, preventing the uncomfortable stuffed or plugged-up feeling that has kept many musicians from using other types of ear plugs.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NOISE REDUCTION
Part II

By George Klabin

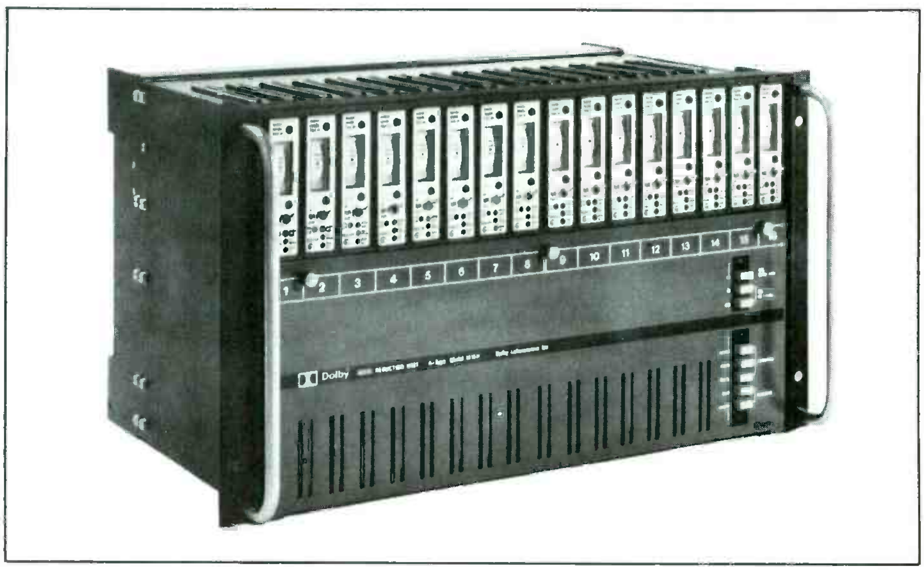
In Part One of this survey I examined what I called *After the Fact* noise reduction, those systems which reduce noise on recorded sources. Though they are useful and important, they do not really eliminate noise from tape recordings.

Sometime in the mid 1960s, it was decided that the best way to truly reduce noise on tape would involve a system which operated before *and* after the signals were put on tape. This approach would allow enough "processing" to be effective. Once noise is recorded on tape at normal signal levels, it is much harder to separate it from the signal and reduce it, without also affecting the signal. Ray Dolby devised a method whereby, through processing, he could finally place the noise farther apart from the signal in the final product. He was the first to devise a successful noise reduction unit which used compression and expansion in a reliable, effective manner, significantly reducing the levels of hum and hiss in relation to signal, without greatly affecting the audible signals.

Dolby Examined

Since Dolby was first to make and market a widely accepted compander noise reduction unit, let's begin by examining his approach. The Dolby "A" system, which is the professional version and the one giving the most noise reduction, breaks the audible bandwidth into four sections, compressing them upon recording, expanding them on playback. The bands are 80-Hz low pass; 80-Hz to 3000 Hz; 3000-Hz

George Klabin has been a recording engineer for twelve years. He has worked with the Dolby, dbx, Burwen Noise Filter, Phase Linear Auto-Correlator, and Kepex noise reduction systems for many years, and has engineered countless recordings with them.



Dolby A-type noise reduction system

high pass; and 9000-Hz high pass. Each band's compressor functions only when there is information in that band, eliminating needless compression. The same is true for the expander sections, which operate in a perfectly complimentary fashion. In addition, high-level signals above the selected "Dolby 0" level are unaffected by compression or expansion, because Dolby employs the well-known "masking effect," which, simply stated, means that high-level sounds tend to mask audible hiss and hum by their mere volume in comparison to the lower level of the noise. In the Dolby system, compression and expansion take place only below the "0" Dolby level, which is usually set at +4 dBm, down to about 36 dB below that point. The compression ratio starts to rise from 1:1 at "0" to a maximum of 2:1 at −30 dB, returning gradually to 1:1 at −36 dB. Total maximum compression is only 15 dB. The reciprocal ratios apply to the expander. Since Dolby does not affect the higher level sounds, distortion at those levels, where it would be greatest, is minimal. Dividing the audible spectrum into four sections allows each one to be designed for optimum operating parameters in the frequency area it serves. Thus Dolby employs faster attack times at high frequencies and slower at low frequencies. The overall net reduction in noise is from 10-12 dB maximum. Overall distortion is low, and transient response is very good.

The Dolby system sounds ideal. It has proven itself reliable in over ten years of operation, and is manufactured to high standards. Dolby still offers lifetime free service on all "A" units—of whatever vintage or model—through a free exchange plan in which a defective unit or module is immediately traded for a working one.

Dolby Drawbacks

As well as Dolby works, it does also have some limitations. The main one is that one gets a reduction in noise levels, but by no means elimination. Dolby's noise reduction processing works only in a critical 36 dB range, but below that range there is still a steady hiss level. If one is mixing a Dolby multi-track tape and decides to add a fair amount of limiting or a mid- to high-frequency boost to even one channel, that channel's hiss level will be brought back up and added to the cumulative sound of the mix. Since we are talking about a signal-to-noise ratio of about 77-79 dB with Dolby processed tapes, it is very possible to degrade the entire ratio 6-7 dB by limiting or equalizing one track on the mix. Such limiting and equalization is common in mixing, and, in fact, often occurs to more than one track. Therefore, much of Dolby's noise reduction can easily be defeated by such common practices.

Another problem about Dolby is that it is important to record at relatively high levels on tape to get the full benefit of its noise reduction. Generally one must record each track so that peak levels often reach 10-12 dB above the "0" level on the meter, and even at a conservative Ampex 185 nanoweber play level, recording at such levels can risk higher levels of tape saturation. After all, the point of three percent 3rd harmonic distortion (THD) on the best of today's tapes is only between +8 to +13 dBm. Therefore, with Dolby, it is important to pay close attention to the levels being recorded on each track, so that the optimum amounts give full noise reduction and minimal distortion. While this is by no means impossible, it is a slightly limiting factor in today's hectic and pressured multi-track recording scene.

The most serious drawback to using Dolby is its dependence on very critical matching of record and play levels. If one does not playback Dolby-encoded tapes at precisely the same levels they were recorded at, there will be a change in frequency response, which will vary more or less precisely with the difference between record and play levels at 1000 cycles reference. A mismatch in one direction causes excessive high frequency response, in the other, a loss of highs. Fortunately at louder levels, where the frequency response is most audibly critical, the change is almost exactly 1 dB for every 1 dB mismatch in level, due to the mild compression ratio at the higher levels below the "0" level. Nevertheless, any frequency response change is undesirable.

Dolby supplies a special "Dolby Tone" feature. This "warbling" tone is supposed to be recorded on each reel of tape at the user-selected "0" level, and also serves as an audible identification that Dolby processing was used. Upon playback of the Dolby tone, the gain of the reproduce tape amplifier is set to precisely indicate "0," thus assuring a perfect record-play level match. A great amount of time is required to align and check Dolby and recorder levels, usually forty-five minutes to one hour for a 24-track recorder. In the "heat" of multiple sessions with different formats, this process is often either overlooked or imperfectly performed. In my experience, more than half the Dolby multi-track tapes and some of the ¼-inch mixes I have received from other studios did not have any Dolby tones on them. In many cases were there no tones at all on the multi-track tapes. Thus, it was impossible in playback to be sure of duplicating the frequency response of the music as recorded on the tape.

Speaking of frequency response, one common criticism of Dolby that has been floating around the industry for years is that using it on one machine and playing back on another can cause odd minor "changes" in high-frequency response, which have been likened to "phase shifts." Here I must come to the defense of Dolby and state that the Dolby design does not seriously aggravate phase shift—those "changes" heard between tape recorders might well have been the slightly different azimuth angles of the heads between machines. In multi-track recording, far less attention is paid to azimuth adjustment, since in practice there are actually a range of angles of azimuth which are acceptable and produce "in-phase" recordings, but which may still produce very slight changes in the high frequency response of complex waveforms such as music.

One last minor drawback to the current Dolby "A" systems is that one can only listen to a decoded Dolby processed tape after the recording is completed and the tape rewound, otherwise you must spend twice the money and get two Dolby units, one for encoding and the second for simultaneous decoding. This is really only a potential problem in "live" recording, where it is often desirable to hear the decoded sound as it is recorded. Obviously, all the Dolby units allow one to hear the "input" without processing during recording, though the processed signal is feeding the tape. This is usually satisfactory for most users.

So, despite being developed eleven years ago, the Dolby system is still a fine example of a noise reduction design giving moderate but significant signal to noise improvement and causing the least changes to the sound because of its moderate approach.

Enter dbx

When Dolby first appeared there was really no need for more than a 77-79 dB signal to noise, since disc recording hardly offered that good a ratio itself. However, as time passed and technology produced improved recording processes, the need for even more noise reduction increased. Therefore, about seven years ago, dbx came out with a system which, in two areas, was a definite improvement over Dolby. The system offered a minimum of 30 dB improvement in signal to noise and an additional 10 dB increased headroom. It also had totally non-critical level matching since it worked by compressing the entire audible range on a 2:1 ratio and then expanding it at precisely the same ratio—restoring the original sound and effectively "eliminating" hiss. Of course, no noise reduction system currently available truly gets rid of hiss entirely, but for most practical purposes the tremendous reduction offered by dbx put the hiss levels out of range of audibility.

With dbx there is such a wide dynamic range available that it is possible to record at lower levels—below +4 dBm—and take advantage of lower tape distortion while still preserving the hiss-less quality. Level alignment is not critical either (except in two-track mixdowns where it is easy to check). During mixing, tracks may be limited and equalizations boosted to a great extent without adding audible hiss to the mix.

Perhaps the most important aspect of the entire dbx philosophy is that they wanted to penetrate the consumer and semi-professional market. In the early seventies a whole new group of recording enthusiasts were able to take advantage of fantastic technological breakthroughs which allowed formerly high-priced equipment to be made in low-priced consumer versions. In both the pro and amateur world of recording, hiss and hum were still a major enemy. But who could afford four tracks of Dolby at $700 each when their entire budget might be $3000? Dbx set out to design a system which would work well with any recorder—even casette decks with their limited signal-to-noise ratio—and yet cost far less than Dolby. They succeeded in manufacturing consumer units costing as little as $100 per encode or decode channel, and delivering far more noise reduction than Dolby. Dbx's units also cost far less than Dolby on the professional end because their semi-pro line was essentially the same as their "pro" line. Most important, there was now a way to attain as much as 90 dB of signal-to-noise ratio on a good tape recorder, and somewhat less on less expensive types, but still enough to effectively "eliminate" the hiss.

dbx Difficulties

Now all this praise does not mean that dbx has no problems. In designing low-priced systems, and ones which offer large amounts of signal to noise improvement, there are always compromises. Any system which compresses and expands across the entire audible range cannot deal with all frequencies in the best possible manner. First of all, any error in frequency response in either record or play on a tape track will be doubled by dbx and its 2:1 ratio. Thus tape decks should be held to + or − 1 dB tolerances from 50 to 15,000 cycles. This is not difficult for professional machines, but is difficult for consumer decks. Dbx took this into account though, as shall be explained later on. So, in any use, attention must be paid to proper and critical recorder frequency response. Fortunately, this is one area of professional recording which is carried out relatively diligently by most studios, since incorrect frequency response is undesirable with or without use of noise reduction.

Another problem with dbx involves severe subsonic frequencies (below audibility), such as building vibrations or subway noises, which can cause mistracking of the unit and result in hum or breathing sounds being added to the signal. Dbx employs a high-pass filter at the compressor input which reduces such subsonics, but in extreme cases they will still confuse the decoder section and cause it to track the level of the subsonics instead of the program. However, any recording environment with subsonics loud enough to cause this problem will cause plenty of other problems as well and result in recordings which lack low frequencies from the subsequent attempts to filter out the lows. The lows, though they may not be heard at their fundamental frequency, often set other objects in the recording studio in motion at frequencies which are very audible. In my recording experience I have had no problem with dbx and such low frequencies.

Modulation Noise

While we are discussing low frequencies, we come to a problem which plagues tape recording and which has to be dealt with effectively by noise reduction systems, or else they will fail to reduce all the noise. This area is called "modulation noise." There are two main types. The first is caused when a loud low-frequency source, such as a bass or the low end of a piano, is recorded alone—without other higher frequency sounds, which normally would mask the modulation effect. Surface irregularities of tape produce a constant form of noise called "asperity noise," and it is found across the entire audible frequency spectrum, but is most annoying at mid and high frequencies. Loud low frequencies, when recorded on tape, tend to "modulate" this asperity noise, causing it to increase and decrease in level with the strength of the low frequencies. This "breathing" of the noise is often more noticeable than steady hiss and normally is masked by mid- or high-end program material. If this is absent, as described, the modulation of the noise is apparent.

The other type of modulation noise is called "sideband modulation," which is noise occurring at frequencies very close to the original source. For example, with a 180-cycle tone, the sideband modulation would be the noise in the frequency range within a few cycles above and below the 180 Hz signal and would fall off in level rapidly as one moved away from that frequency in either direction. This type of noise also is caused by the tape's surface irregularities, which cause the asperity noise previously described. The graph (Fig. 1) shows both asperity noise at its fixed state with no signal applied to the tape, and then the sideband modulation caused by recording a 180 Hz tone at various levels. The audible effect of sideband modulation is amplitude change which results in a "muddying" of the original signal. A common audible example of this type of modulation noise is heard when a pure sine wave test tone is recorded and reproduced at the same time on a tape track, creating the well-known "burbling" effect.

Dbx reduces the effects of modulation noise by employing pre-emphasis to the signal before encoding, and de-

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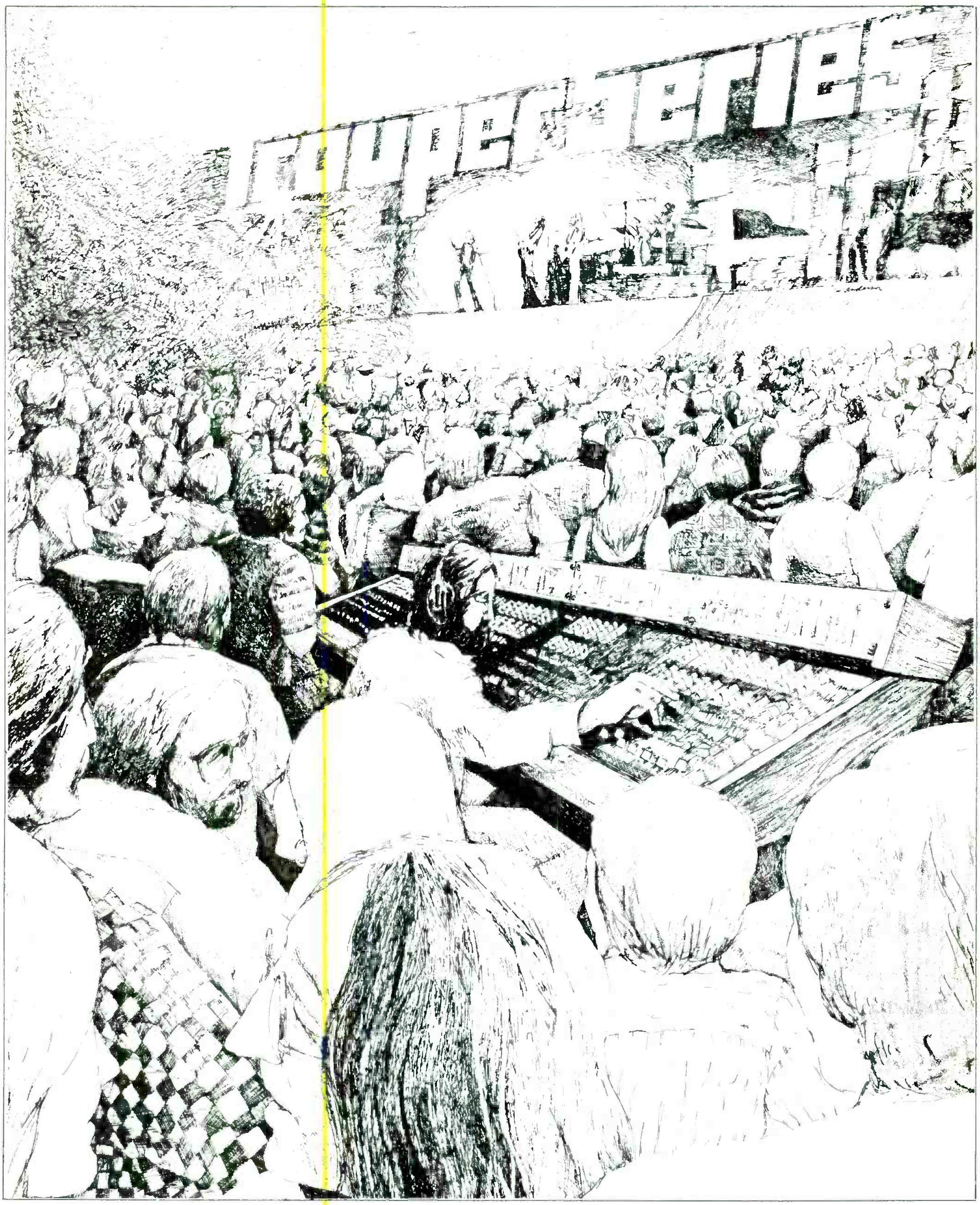

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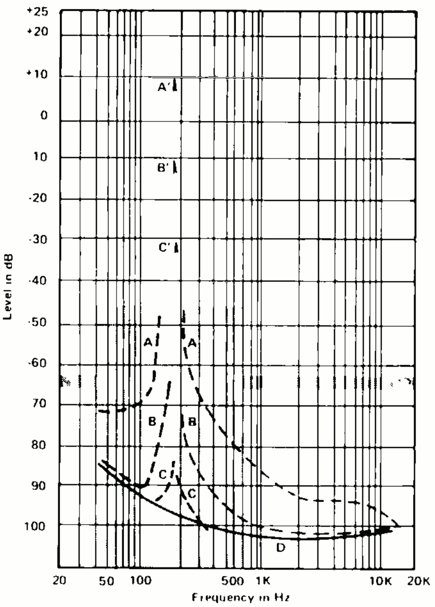

Fig. 1

Reprinted from dbx Model K-9-22 Instruction Manual, 1976

Asperity noise yields the curve shown by "D." The recorded signals (A, B and C) are all sine waves at 180 Hz. The noise sidebands created by these signals are illustrated by curves A, B and C. Observe that the higher the recorded signal level, the higher the noise sideband level. This level-dependent noise is known as *tape modulation noise.* The noise sidebands are masked partially by the recorded signal, but only for about two octaves on either side of the signal. This masking is depicted by the shaded box in the chart. The ear is less sensitive to lower frequencies, so the lower sidebands are masked sufficiently by the signal. Notice that the upper sideband of the +10 dB recorded signal (curve A) extends beyond the masked area, and at a level which would be audible in a program of 100 dB dynamic range.

emphasis on decoding. Frequencies above 400 cycles are boosted to a maximum of 12 dB at 1600 cycles and above upon recording, then rolled off in the opposite manner—but starting from 200 cycles and above—in playback. The net result is a reduction in modulation noise by almost 12 dB, while maintaining a flat record-play frequency response through the precisely complimentary operation of the two curves. However, the pre- and de-emphasis systems do degrade the signal-to-noise ratio, which, in this 2:1 system of dbx, could have been 120 dB with a tape recorder having a 60 dB basic signal-to-noise ratio. Still, the degradation produces a usable range of 90 dB, which is exactly what dbx wanted.

Dolby does not need any pre- or de-emphasis in its design because it breaks the bandwidth into four sections, such that the fundamental frequency in the low range will only affect its band (and possibly the next one up), while leaving the other two untouched. Since the upper two bands do not change their gain, they do not add much, if any, audible hiss.

No currently marketed noise reduction system can effectively deal with the sideband type of noise modulation since it occurs in its worst form—from a technical standpoint—at frequencies so close to the original program. But the audible effects are worse at mid and high frequencies, where each system has its way of dealing with it. The amplitude variations caused near the fundamental frequency are still not severe enough to cause much muddying, though this is a matter of subjective judgment.

Remember that modulation noise of both types occurs in tape recording, with or without noise reduction. Dbx suggests a method to reduce modulation noise in all tape recorders. Record a five cycle tone at a reasonably loud level, say around 0 dBm. Listen, at a loud monitor level, to the playback of the tape channel while recording. The tone is below audibility, but a raspy sort of noise should be heard seeming to fade up or down as the five cycle tone modulates it. By adjusting the bias level control on the tape amplifier, one should be able to find a setting where the level of the noise is greatly reduced or disappears almost completely, depending on the monitor level used. Make adjustments slowly to allow for the time delay between the record and play heads. Slight overbiasing between one-quarter to three-quarters of a dB above "peak" level at 1000 cycles should null out or reduce the modulation noise. A beneficial side effect of this slight overbiasing claimed by dbx is that it should coincide with the point of minimum 3rd harmonic distortion. Anyway, most professional recorder manufacturers, and many semi-pro brands too, recommend slight overbiasing of the tape channel when using high quality tape. Having followed this simple procedure, you will have improved the performance of your tape deck whether or not you employ noise reduction.

Level Changes

Another problem area for noise reduction systems is abrupt level changes caused by dropouts in the tape surface. These level changes can cause mistracking and gain errors because the level detection circuitry would read them as changes in gain. Dolby's design does not accentuate dropouts, but dbx's can, if the dropout is long enough. The dbx RMS level-sensing circuitry has two potential problem areas, due to the selection of the six millisecond interval of level sensing. This period was chosen by the designers as the optimum compromise after considering that the system had to work across the entire audible bandwidth and thus must process both lows and highs with the same time constant. When the rate of change of level is small, as in minor dropouts or even sideband modulation noise, the system level will not change and cause any error. With longer dropouts or those with severe drops in level, where the change lasts for several milliseconds, the detector may respond and introduce a level change in the decode channel. The most important factor to remember is that any dropouts which cause problems with dbx will cause problems without it as well. The best way to avoid dropouts is by using only the highest quality tape and keeping all surfaces which touch the tape clean.

The other problem area involving the time constant of the level detector is that the six millisecond time constant is not long enough to process frequencies below about 160 Hz without introducing some minor harmonic or intermodulation distortion into the audible signal. In theory, the dbx design takes this distortion into account and cancels it by the complementary design of the expander, but in practice, the phase shift of the tape recorder itself may interfere with the cancellation and produce some leftover distortion. Again, the amount of this distortion is so low that *very* few people can hear it—which is why dbx has been accepted by so many users.

While we are discussing the level detector circuit, I want to add that both Dolby and the new Telcom system which I shall shortly discuss employ peak level detection rather than RMS. Dolby also uses average level detection, but the peak detection section in both these systems can be

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confused into dynamic level errors due to phase shifts in the tape recording process which cause changes in the relationship of peak to average level in complex waveforms. The dbx RMS level detection circuit is always looking at all of the energy—not just the peaks—and thus the changes mentioned do not affect it.

Attack Behavior

One of the most interesting and widely discussed areas of deficiency in compander-type noise reduction systems is poor transient response arising from the attack time performance of the compressor and expander sections. Attack time in music is defined as the time necessary for a note to reach its full volume. Percussive instruments have short attack times and reach maximum volume quickly, while wind instruments have long attack times and reach maximum volume slowly. In audio, attack time is defined as the time required for the circuitry to return to a steady state after application of the signal to its input. This interval is measured in micro (millionths) or milli (thousandths) seconds. When designing any noise reduction system the attack time is very critical to maintaining proper program dynamics. The main enemy is "overshoot," in which the gain of the device to which the signal has been applied jumps far above the proper level for a certain period before settling back to the correct level. Recently I received a test report made by AES/Telefunken comparing attack behaviour of their new Telcom C4 noise reduction system against Dolby and dbx. According to these tests which used only tone bursts at 3 kHz and 15 kHz and various input levels, dbx showed a serious "overshoot" in the compressor section, and serious "undershoot" (the opposite of overshoot, when the signal remains too low in level for a period of time before rising to the proper level) in the expander. As shown, this deficiency was great enough to cause severe saturation of the tape, which would of course soften program attacks and change the dynamics of the music. The main problem with overshoot is that it can cause this saturation easily, since tape has limited overload capacity before it provides its own "limiting" effect. The test showed Dolby performing well at higher levels, where noise reduction was not working, but evincing erratic attack behaviour at levels where noise reduction was in effect. The time required for the signal to attain a proper level varied with different input levels and changes as slight as 1 dB. Telcom performed almost flawlessly with all the signals at both frequencies and at all input levels.

The evidence seems damaging, especially for dbx and Dolby, but the fact is that all the test showed is that the new Telcom system responds far better to impulse tone bursts than do Dolby and dbx. What one must remember is that the rise time of the 3 kHz signal used in much of the test is 83 millionths of a second, assuming zero crossing point switching of the tone bursts. This is far faster than the rise time of almost any sound that presently can be mechanically produced by mankind, with the exception of electronic music. The fastest attack time of a sharp, high-pitched cymbal crash is several milliseconds—far slower than the tone burst used. What is most important for dbx and Dolby is how they perform with more realistic types of program material. Dolby's specs state an attack time of 1 millisecond (on the mid and high ends) for a return to steady state, and maximum overshoot of 2 dB. This is faster than the musical transients' rise times and thus Dolby handles them well. The dbx, being a device which operates over the entire audible bandwidth, has a longer time constant for attack, being 6 milliseconds maximum for a return to a level not more than 3 dB above (or in expansion, 3 dB below) the correct steady state level. Don't forget that with dbx's RMS level detection, attack and release times are not fixed, but are a function of the rate of change in level of any signal. If the rate of change is fast, the attack time will be fast. If slow, it will be slow with a maximum of 6 milliseconds. However, the dbx cannot process tone bursts on the order of millionths of a second without serious overshoot. The expander section of dbx is designed to compliment the overshoot by introducing undershoot in precisely the same manner and amount. Thus, any overshoot is cancelled out by this design—with one exception. If the tape medium between the compressor and expander is saturated, then of course there will be a change in program dynamics. If the saturation is caused by a tone burst of very short duration, the distortion should not be audible anyway, though it will show up on the type of test made by Telefunken. Most importantly, dbx and Dolby will not cause any serious overshoot with normal types of program material. In the realm of electronic music, however, it is possible that they will introduce some problems if the rise times of the signals are very fast. Whether the resulting overshoot will saturate the tape and how the ear will respond to the short duration of it is another matter which is totally subjective.

To summarize my experience with dbx, I would say that the few problem areas as discussed are rarely if ever audible, and then only in such small amounts that many listeners cannot distinguish them as "changes" in sound. The sense of clarity and space that dbx gives is a unique experience and often hard to believe at first. As I stated in Part One of this article, total quiet on recordings has often been mistaken as a lack of highs by first-time listeners. But systems such as dbx are addictive. Once one gets used to hearing sounds with no hiss, it's hard to accept a normal recording and its noise.

Noise and the Newcomer

Recently I received two of the new C4D noise reduction cards from Telefunken to try for a test period. Telefunken has examined existing noise reduction systems and come up with one which attempts to combine the best of Dolby and dbx and to eliminate any of their deficiencies. The only version currently available in the U.S. is the plug-in C4D card which is designed to fit into any Dolby M series of 361 mainframe. These cards will start to be available for purchase around September of 1977. Drawing upon Dolby's idea, Telcom, as the system is called, has split the frequency range into four bands—30-215 Hz; 215-1450 Hz; 1450-4800 Hz; and 4800-20,000 Hz. Separate bands allow attack times to be optimized for each frequency range. Thus the first band has a 350 microsecond time, the second 50 microseconds, the third 18 microseconds and the fourth (highest) 5 microseconds. Preliminary literature on the system lists some differences in the exact band divisions, but does not give attack times. However, I received the information from Steve Temmer of Gotham Audio, which exclusively distributes the Telcom in the U.S.

The attack times mentioned are fast enough to handle any conceivable tran-

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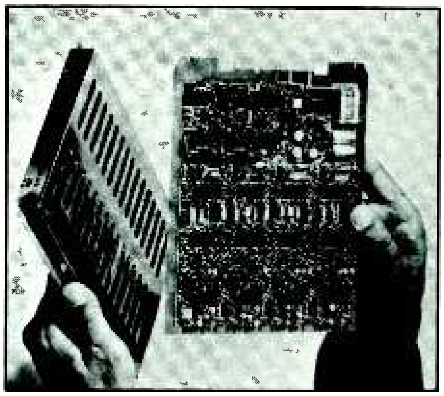
sient with minimal overshoot, which is why the unit performed so well even on tone-burst testing. The control circuit uses peak level sensing on all four bands, and since Telcom feels none is necessary, there is no compensation in the design for phase shift. The main difference is that all four bands compress in a ratio of 1.5 to 1 or a 33% slope and function at all levels, not (as does Dolby) cutting out operation at certain levels. Telcom explains the compression ratio as follows: "The dynamic range of the level after the first amplifier is compressed to two-thirds of the range at its output, or by thirty-three percent in the logarithmic scale. This method has the advantage of operating very precisely over its entire range due to the high degree of feedback in the control system. Furthermore the control range of the three amplifiers in each band is reduced to one-third of the input dynamic range, simplifying the design of such circuitry. The retrieval of the original dynamic range in the expander is obtained in a complementary way."

To reduce modulation noise, the level passes through a weighting network with an 18 dB per octave slope. Again, like Dolby, Telcom relies to some extent on the band-splitting to minimize the affects of asperity noise being modulated by low frequencies. From tests that we performed on the Telcom cards, we found that, subjectively speaking, the modulation noise in the mid-frequency range, seemed about equal using Telcom, dbx and Dolby, and slightly better in the high end with the Telcom system.

Another advantage of four bands in this system is that frequency response errors are not so critical, since they are not magnified to so great an extent as with dbx. The usable increase in signal-to-noise ratio claimed is about 30 dB, similar to dbx.

The picture seems rosy for the Telcom system, but it has not been in any large scale use yet, and I have reservations in some areas. The very fast attack times—on the order of millionths of a second—for the various frequency bands, can produce amplitude modulation from the rapid change in gain allowed by such a fast attack response. In large amounts this amplitude modulation could produce an audible click on some transients, though, to my knowledge, this has not been experienced by anyone using the Telcom. Scope photos do not show this sort of modulation unless very sophisticated equipment is used, but more importantly, whether the modulation occurs or not, is it at all audible? Only time will tell.

The fast attack times also can produce what might be termed "extra"



Telcom C4 system

frequencies, or aberrations of a sort. The audible effect of this might be a loss of clarity, but if these aberrations occur as quickly as one might suspect, they would also end too quickly for the ear to hear them. The one aspect of human hearing that we must keep in mind is that it takes time to hear things change, and if we are dealing with changes that are restored or start and finish in a matter of millionths of a second, we can't hear them, so, subjectively, they do not exist. With sophisticated test equipment we can discover all sorts of little "errors" or changes, but what really counts in this area of noise reduction is audibility. Noise reduction is designed with that question in mind.

It might be added that the Telcom system does not seem well-suited to consumer use with recorders having a high-noise level, because there will come a point at which the signal level drops down to the noise level, and causes mistracking between the expander (whose control circuit cannot distinguish between the noise and signal) and the compressor. This problem is inherent in any compander noise reduction system which operates at all levels, but is more likely to be troublesome in a system such as Telcom where the compression ratio is only 1.5 to 1. The Dolby totally avoids this problem by cutting out all compander action below −36 dB relative to the Dolby "0" level, and thus uses the steady hiss which exists below that point to mask any deficiencies in its operation.

The main disadvantage of the Telcom is that it is as yet unproven in long term use. Also, its price is relatively high, at $700 per encode or decode channel; it is necessary to use it with a Dolby mainframe; and unless two channels are used, the Telcom, like the Dolby, can only monitor the decoded signal after rewinding and playback. These drawbacks seem minor, though of course only time will tell of Telcom's reliability. Telefunken has a history of making superb quality products which are extremely reliable. The system seems to offer several improvements in areas of alignment ease, total noise reduction and minimal side effects. At present, Telcom has heavily invaded the satellite communications and long lines transmission field since their system is unique among the three we have examined in being able to withstand many stages of amplification with various gains—which would cause tracking errors in the other two systems. It is claimed that Telcom allows an improvement in transmission of two full bits from analog to digital back to analog signals. This is an impressive achievement and a money-saver for the transmission firms.

Other Uses

I'd like to mention a few other uses for noise reduction systems besides mastering on tape recorders. For example, noise reduction is valuable in recorders or devices used for delay and echo effects, as it reduces the noise they contribute. Noise reduction can be used on the reverb send channel, and again on the chamber output, to reduce chamber hiss. It will change the decay time of the chamber, but this can be compensated for by lengthening the decay time setting. Dbx is currently incorporated in the older 1745A model of the Eventide delay line to improve its signal-to-noise ratio, and some form of noise reduction is found on other less expensive delay lines which use less bits and must compensate for the higher noise level with this addition in their circuitry.

Most important of all, when using noise reduction, it is advisable to mix down the multitrack tape onto a noise-reduced ¼-inch tape to preserve all the wide dynamic range. Making copies from noise-reduced tapes is yet another advantage since there is no need to decode the signal. Merely copy the tapes in their encoded form being sure to use the unencoded test tones

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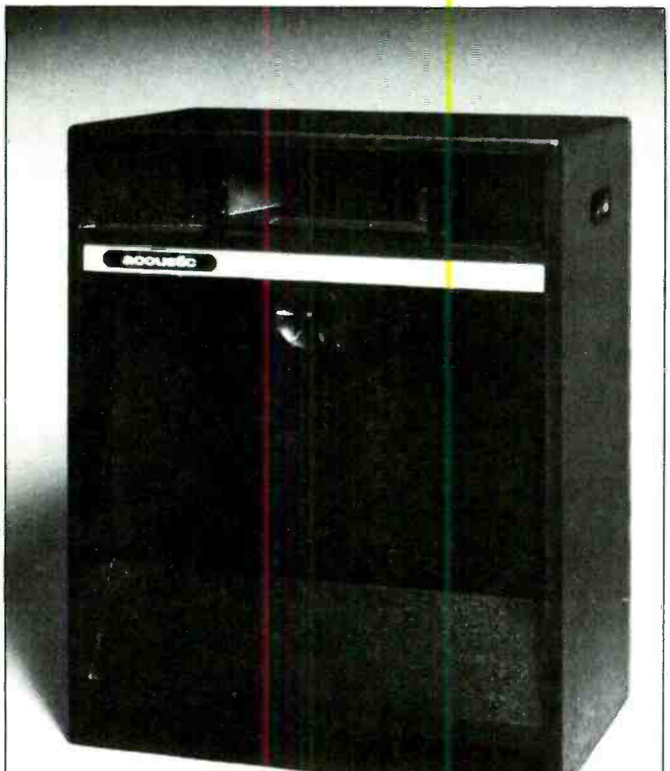
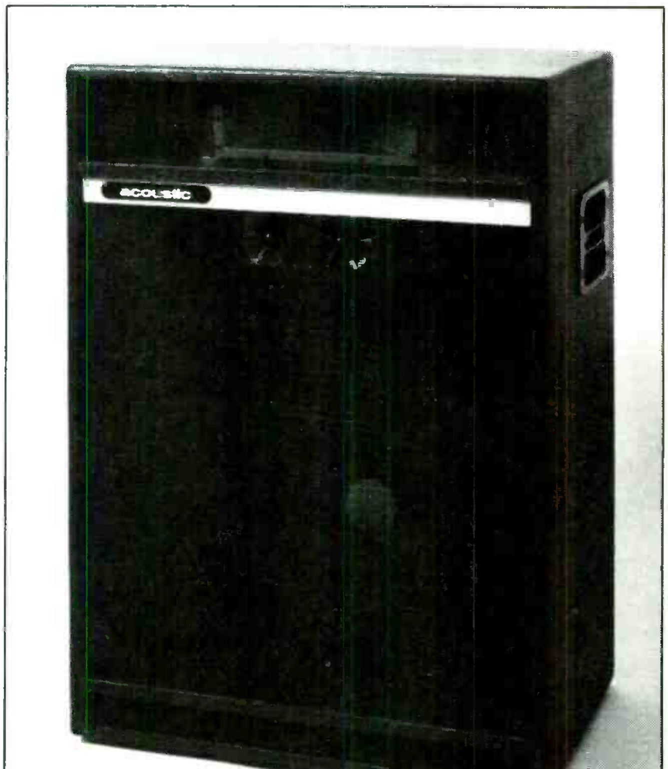


Acoustic Model 806



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on the original for proper level and frequency response adjustments. In this manner the copy will not be degraded in signal to noise to the extent that the additional noise from the second generation would become audible. With no noise reduction, a properly made second generation tape will have up to 6 dB degradation in signal to noise.

It is essential to record tones on all mixes, and is recommended on at least the first reel of each session's multitrack master. I suggest recording 1000 cycles for thirty seconds, followed by fifteen to twenty seconds each of 10 K, 15 K and 100 Hz to check low end response. These tones (unencoded of course) will "lock in" the frequency response of the particular recorder used on that day, facilitating perfect playback at any later date on any machine. One last hint: be sure to write the type of noise reduction used clearly on each box of tape, so that there can be no confusion at any time in the future.

The Question is . . .

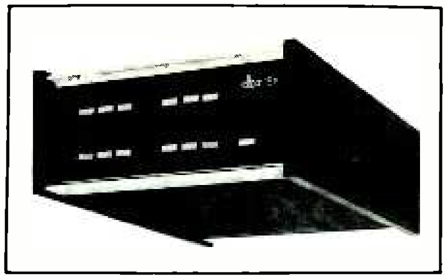
Well, now the question is, if you want to use noise reduction, which system should you buy? As stated in Part One last issue, any studio owner who wants to do large-scale business and work on tapes recorded at other studios must own or have available one or more noise reduction systems. The choice at present seems to be between the greatest compatibility with other systems, largest amount of usable noise reduction or absolute best technical performance regardless of price. Dolby has over 25,000 channels of "A" type in use throughout the world. Dbx, four years younger than Dolby, already has 18,000 pro and semi-pro channels in use, running an ever-closer second in that category. Telcom has very few channels in use at present. The least expensive channel of "A" Dolby runs about $600 when purchased in the M-series, with a minimum of 16 channels for that price per channel. In single units the model 361A costs around $750 each. The "B" series Dolby is designed for consumer use and is substantially less effective than the "A" because it processes only one band of frequencies, and it is not compatible with "A." The Telcom C4D cards must be used with a Dolby mainframe and cost about $700 each. A self-contained Telcom system will soon be available in Europe, but its cost is presently at least as high as the individual plug-in cards. Telcom has no consumer system at present.

Dbx's professional Model 216 runs around $600 per channel, self-contained for 16 channels, but as low as $187.50 per channel for the semi-pro 154 which gives almost identical performance except for slightly lower (but still high) overload point and unbalanced operation. The 154 is not a simultaneous encode-decode device, but the 157 series, at $300 per channel, is, as is the 216 pro series. The new dbx K-9 card is designed to plug into existing Dolby mainframes at only $250 per card. Dbx also has a true "consumer" line of 120 series models which offer the same maximum 30 dB signal to noise improvement and cost as little as $100 per channel. This dbx II series, as it is also called, is not compatible with the pro or semi-pro lines, because it is specifically designed for tape or cassette decks with poorer frequency response which could cause mistracking of the pro series. In the II series, dbx added a steeper rolloff on the low end, down 1 dB at 30 Hz, and most importantly, having the RMS detection circuitry insensitive to frequencies above 10 kHz, where the losses in such unsophisticated equipment may occur. The II series does process the entire audible spectrum, however. Some recorder manufacturers such as TEAC are installing dbx directly into their tape decks, just as the Dolby B series has been incorporated in many brands for years.

If someone asked me which noise reduction system to buy, I'd say buy all three, if you own a commercial studio. Since many people cannot afford that, the information in this article should at least make it easier for them to decide. Whatever system one ends up buying, it's important to remember that another link in the recording chain is being added, thus increasing chances of problems due to malfunction, misuse or misalignment.

Compromised Benefits

We can now see from examining all the noise reduction systems, that each one offers tangible benefits, but with some compromise—at least from a technical viewpoint. All compander type noise reduction systems must of necessity (with present technology) produce some aberrations of the signal waveform. The expander is designed to complement the compressor in such a manner as to cancel out most of the



dbx 154 four-channel system

changes in sound. Unfortunately, tape in its present form is not an ideal storage medium and introduces some errors itself. To review the aberrations of waveforms most common in noise reduction systems, we have:

1. Changes in level from the original, caused by overshoot or undershoot of compressor or expander sections, caused by too slow an attack time.
2. Possible amplitude modulation caused by too fast an attack time in the compressor or expander.
3. Distortion at low frequencies caused by too fast a time constant in the level detection circuit, and phase shift in the tape recording process.
4. Hiss or asperity noise "trails" from improper or less than ideal decay times in the expander sections.

Though there are other minor problems, these four areas are probably the worst "offenders," and are all caused by design considerations in the compander style of noise reduction. But without noise reduction, there is the problem of hiss as well as deficiencies in the tape recording process—such as phase shift and modulation noise. The choice between noise reduction or no noise reduction is very subjective. How audible are the problem areas listed above? How important are the audible benefits of noise reduction? Some people are sure they can hear any noise reduction system working. Others are sure they can't. Test instruments show all the tiniest changes that any component introduces. But from an audible standpoint, the changes introduced by noise reduction last for a matter of millionths of a second or, perhaps in rare cases, a few thousandths. Even pure musical instruments can create distortion. How many people have heard the intermodulation or harmonic distortion caused by two trumpeters playing the same note, perfectly in tune, yet strangely dissonant from the ugly "clashing" that can occur if one of them "overblows" and adds a great deal of 3rd harmonics to the note? In addition, we must ask, how much of the changes produced in the noise reduction process are masked by other sounds? It's all very personal.

Future Noise

Sometime around 1975 a company called Ambiphon in Bronx, N.Y., succeeded in manufacturing a prototype 4-track tape recorder using normal audio tape and analog circuitry which produced a usable dynamic range of 90 dB! In addition, the recorder had almost unmeasurable distortion over the entire range, and apparently had somehow reduced modulation noise as well. All this was achieved merely by a tremendous concentrated effort to optimize every aspect of the tape recording process. This involved specially designed distortion cancelling circuits, cross-field biasing at extremely high frequency and hand-selected low-noise components. Though the company folded, and no commercial models of any recorder were sold, it proves that the so-called analog tape recorder of the future has already been made.

Current manufacturing technology has produced lower cost, more reliable products which today perform functions totally unknown or out of reach to the layman several years ago. But these new inventions are still highly sophisticated and require increased understanding of their functions and more attention to proper alignment. The layman or amateur, beset by the current atmosphere of heavy advertising, promising everything short of "cooking your dinner for you" at "cheap" prices, does not have the knowledge that only experience can bring, and will often be pressured into buying "wonder toys." These "toys" can become "weapons" in the wrong hands, causing degradation of the signal rather than improvement.

Where are we going in the future of tape recording? What is the role of noise reduction in the next ten years? I feel there are two major developments presently being refined which will eventually affect us all. The first is digital recording. In this process, sound in its current analog form will be transformed or encoded into digital "bits" of information; these "bits" will then be recorded on special tape and decoded to analog for playback. This process holds the promise of practically eliminating distortion and giving usable dynamic range in excess of the 90 dB offered by current noise reduction systems. The idea would be to record digitally and mix to a digital master for cutting on a digitally encoded disc, which only when played on the disc player would be decoded to analog form, preserving the quality till the very last stage. At present, the digital recording field has just begun to develop. Present digital tape recorders for music use are extremely expensive and do not offer even 90 dB signal to noise in any but the most sophisticated and costly versions. Studios are reluctant to purchase such a totally new concept because it means eventually abandoning analog tape recorders—themselves costing upwards of $30,000 dollars for a 24-track recorder. However, I feel that within ten years digital recording of some sort will be in fairly widespread use in the industry.

The other development which is already available and beginning to be accepted, though very slowly at present, is a system which reduces noise on disc. At present, the best signal-to-noise ratio found on an LP vinyl pressing is 70-72 dB. This is far from the 100 dB necessary for full dynamic reproduction. Dbx or Telcom encoded tapes, and even some Dolby tapes, have a greater signal to noise than any vinyl disc. The newest answer to this problem comes from dbx with its II series disc encoder and decoder. Using it, a specially encoded disc is produced which, when played back through the decoder section, provides an over 90 dB signal-to-noise ratio with marvelously clean dynamics, even reducing some types of inherent disc noise as well. The dbx II disc encoder/decoder is already on the market. One company has released a series of dbx encoded records. At present few consumers own the system, and it will take much money and time to convince the major record companies to put out two versions of each LP—encoded and normal. However, quadraphonic records are pressed along with normal stereo releases, so it is not at all inconceivable that within two or three years, many record companies will offer some, if not all, of their catalog in encoded form. No matter which system is finally adopted, and at present there is really only the dbx for disc, the necessity for improving disc signal-to-noise ratio is well upon us. The quality of current disc pressing has sadly deteriorated in the past few years, and only adds to the noise problem.

Direct To Disc

The music came from a sliver of a building stuffed between a hardware store and a linen closet—a tiny one room "stereo salon." I paused in front of the entrance because what I was hearing was not the standard cacophony of muddled sounds and voices, but something so rich, so unbelievably crisp, so *right there*, that I walked in, almost in a daze. I could not believe it, but all that marvelous sound was coming from a pair of small bookshelf speakers. As the salesman stood idly by (a wry smile on his face, knowing of the sure sale), his customer was experiencing aural ecstacy. The experience was soon too much for the customer to bear, and he purchased the speakers, the turntable, the receiver, and a few headaches when the next month's bills arrived.

His commission for that hour earned, the salesman now turned his attention to me, a non-paying customer, seeking information only. I inquired as to the nature of the album and was politely told some technical mumbo jumbo that left my question unanswered but my determination greater. At that moment, some bulging wallets came through the door, and the salesman (casually picking up the album) sauntered over to their questions. I apologized to my extra thin wallet as I purchased the album for the outrageously high price of ten dollars. Being the era of the three dollar album, my action bordered on lunacy, but I knew the sound of my dreams and was hooked. It was only later that I learned that the album was a direct to disc recording.

Direct to disc recording has a two-part history. Many records recorded before 1945 had greater presence and dynamic range than later recordings. It was determined that with the advent of the tape recorder in the 40s, some of the presence and dynamic range was inexplicably eliminated.

In 1959, Doug Sax and Lincoln Mayorga, friends and musicians as well, tested the above hypothesis by recording a piano with the signal from the microphone being fed directly into the cutting lathe, which normally receives the signal from the master tape. What they heard when they played back the test lacquer was a phenomenally "live" sound, and direct to disc recording was born (again). The record has always been the weakest link in any audio system. Direct to disc recording is simply an attempt to elevate the recorded disc to a level of quality equal to that of today's sophisticated audio equipment.

During the past few years some incredible developments have been made in electronic taping equipment. But even with each improvement in today's multi-track taping process, the resulting master tape still seems to lose the elusive excitement that Sax and Mayorga sought. When the final track has been layed down on tape the resulting multi-track recording is played back through a mixing console. Here the special electronic effects, equalization and the mixing down process takes place. From the eventual master tape, the master lacquer disc is cut. Each time the original signal is altered by re-mixing or overdubbing, the resultant impulse on tape has lost some of the punch—some of the transient response that is everyone's original goal. Each time the signal is passed through the mixing board, or any electronic device, noise is added.

The basic "why" of direct to disc

ecording

By Jeff Weber

Photo by Kirby Veach

recording in the technical vein is that the process has greater headroom, permitting an increased dynamic range. The full instantaneous peak energy of most musical instruments, especially percussion instruments, is often higher than any standard meter can indicate, and it is these transient peaks which usually saturate tape and create distortion. Most recordings today avoid this problem by the use of some sort of peak limiting, resulting in a distortion-free record, but one with a limited dynamic range. Direct to disc recording utilizes few multi-track recording techniques, and this results in lower phase shift and the loss of two generations of tape electronics.

The basic "how" of direct to disc recording is not as mysterious as most first time listeners believe. Starting with the blood and guts differences, there are some techniques, inherent in multi-track recording, which simply are not possible with direct to disc recording. You cannot overdub, edit or bounce tracks. With today's multi-channel recording, the musicians that lay down the basic tracks may never see the horn players, string players, background vocalists or maybe even the lead vocalist. In direct to disc recording everyone's there, in the same room at the same time. The musicians go through it all as a unit.

Bud Wyatt, a design engineer with over seven direct discs to his credit, is currently in charge of Sheffield Lab's direct disc research and development facility. He explains, "Although you are prohibited from bouncing any tracks, you can use any kind of processing device you want; for instance, a digital delay can be used to delay the echo send. It would break between the echo output of the board and the echo chamber. You could also use phasing or flanging devices, as they're called. In other words, you can use any type of 'real time' audio processing. That covers limiters as well as equalizers. We do not use any limiters, but I think that other companies may use them. It's one of those kind of things—it's not what you do . . . it's how you do it.

Just like in a regular session, instruments can be sent directly to the board. You can have any combination of direct and miked instruments. Normal studio hardware and techniques are used here. You have what is called a direct box (for recording or mixing an electric instrument directly, without the need for a microphone) and it does all the impedance conversion and everything for "talking" to the console and it also lets you feed an amplifier out in the studio so they can hear their instruments. If you want to do a combination, you also can mic the amplifier, like on an electric guitar, electric piano, bass or any electrical-type instrument."

Wyatt stated that the console, the cable layouts and the miking techniques are often very similar to the normal recording session. As far as the microphones themselves are concerned Wyatt revealed that Sheffield removes the transformers from the mics. Why? "We've never met a transformer we couldn't hear. It's much harder not to use transformers," Wyatt continued, "as it requires much more attention to things like grounding and radio frequency interference."

In discussing the limitations of the console, Wyatt said, "The input limitations are a function of the inputs of the console you're working with. If you wanted to, and you had a hundred input console, you could use a hundred inputs, but basically, you are hardware limited. As in normal recording, the microphones lead into the studio consoles. From the studio mixing console, special lines take the stereo output of the mixing console and connect the mixing console with the lathe master console."

In theory, the lathe must be in close proximity to the studio, but this does not always occur. Wyatt explained that in the making of *The King James Version*, featuring Harry James and his big band, the Sheffield Lab people found ideal acoustics in a small chapel located one and a half blocks from the lathe. So, 650 feet of line was run from a mixing console located in the chapel's men's dressing room, across the street, through an alley, and across a parking lot to the lathe master console. Needless to say, research has begun on installing the lathes in trucks for remote disc mastering.

If the specific electronic and other techniques of direct to disc recording seem a bit sketchy, it is decidedly so. Each company, realizing it takes more than a lathe and a studio to produce a quality recording, spends months refining and evaluating their hardware as well as their plans. Each company believes that it is something special and is reluctant to let anyone peek in and discover the physical products of their labor.

The magic of direct to disc recording begins the moment the musicians step into the studio. They usually recognize one another as the yardstick of his/her instrument and the excitement and tension begin. The producer of the album enters the room and asks the performers to study the charts carefully, as this session is going to be slightly different from most of the others they previously have been involved in. Lincoln Mayorga, who, along with Doug Sax, originated modern direct to disc recording, perhaps states it best: "There is hardly an expert professional musician who would deny that the quality of performance in the first take or a performance in a "live" situation in front of an audience is more intense, more exciting and a better total effort than a tape recorded studio performance, the second, third or forth time around. Once a lacquer is started, there is no stopping for some seventeen to twenty minutes. During a normal recording session, one of the acknowledged difficulties is keeping all the musicians awake, but when recording direct to disc, each musician is literally on the edge of his seat, alive, knowing that any mistake will ruin the entire side."

Musicians and technicians alike discover that recording "live" for a seventeen to twenty minute span is brutal. The musicians begin to understand that unlike tape mastering, which is actually an individual effort and performance, direct mastering is a communal effort, each artist working together to form a cohesive and responsive unit. The tension and emotional pressure are tremendous.

The technicians realize that it is their mixing boards, their mics, their amplifiers that are being given one of the most stringent tests imaginable. To them, recording direct to disc is more of a religion, requiring months of laborious research and experimentation; they cannot afford to be thwarted by sonic failure or sub-par performance. The machines they build for the project are direct extensions of their own beliefs that the project is unique, the goal so meaningful. Every piece of equipment is the embodiment of their reputation, and like the work of each musician, it is always on the line, always subject to critical scrutiny. Then it is in the hands of the engineer, who must understand all of the equipment and trust it implicitly, as his task is to mix the entire side "live" in "real time."

One reason direct to disc recording is so difficult and so rarely done is due largely to the complex task it imposes on the disc cutting engineer. The direct to disc process prevents the use of a computer built into the cutting lathe and, instead, forces the cutting engineer to make the many delicate adjustments to the lathe during the recording that ordinarily would have been handled by the computer. He must manually adjust the groove spacing to get as much music as possible on a side, and must also compensate for loud passages which require a deeper groove cut.

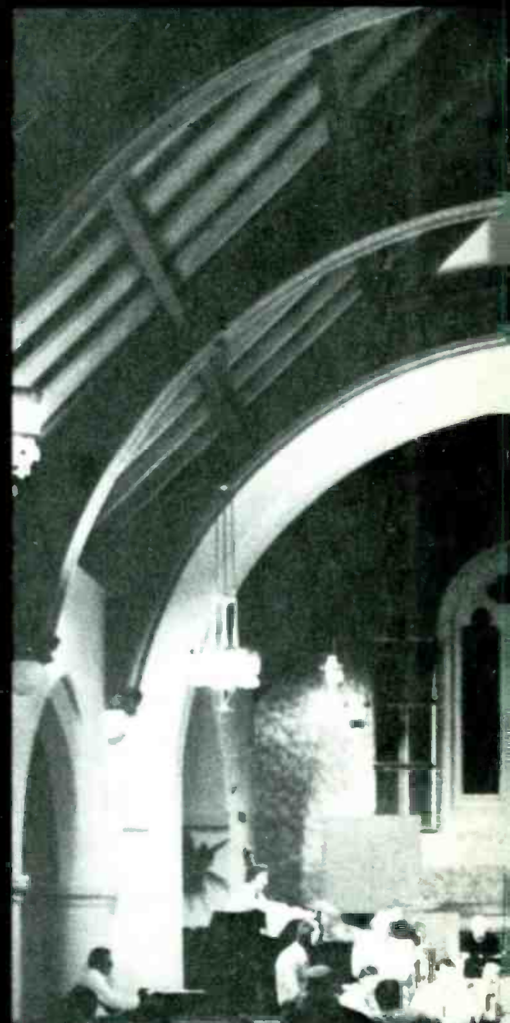
Mike Reese, a lathe operator with Sheffield Labs, the first and largest producers of direct discs, speaks of the tension: "Nothing really happens til the drop of the cutting head. You never quite know what is going to happen. For me, the first thirty seconds of the first pass is a time of shaking and I experience an incredible rush of adrenalin. After that, I am concentrating on so many things at once that I hardly notice the tension. The whole feeling is like holding your breath for fifteen minutes."

The tension of the session does not subside once the recording session is completed. It is now time for the mechanics of producing records to take over. From here on, if every manufacturing step is not taken with extreme caution, the finished product will be damaged in the same manner it would have been had a mistake occurred due to the musicians, the engineer, or the lathe operator. Once the lathe is finished cutting a groove in the master lacquer (a record is actually a single continuous groove), the lacquer is then electroplated with silver and then a nickel alloy, yielding a "mother." The mother, in turn, produces the stampers from which the final record is pressed. If the master lacquers (specially coated aluminum discs) are left standing too long before they are plated, the grooves tend to "bleed" together. Under ideal conditions, the lacquers should be in the nickel plating bath within thirty minutes after the vinyl skin has been cut by the lathe's cutting head. The chances of bleeding increase with every minute's delay. It is this bleeding that causes distortion and noise on the final stampers. Most direct disc lacquers are plated almost immediately.

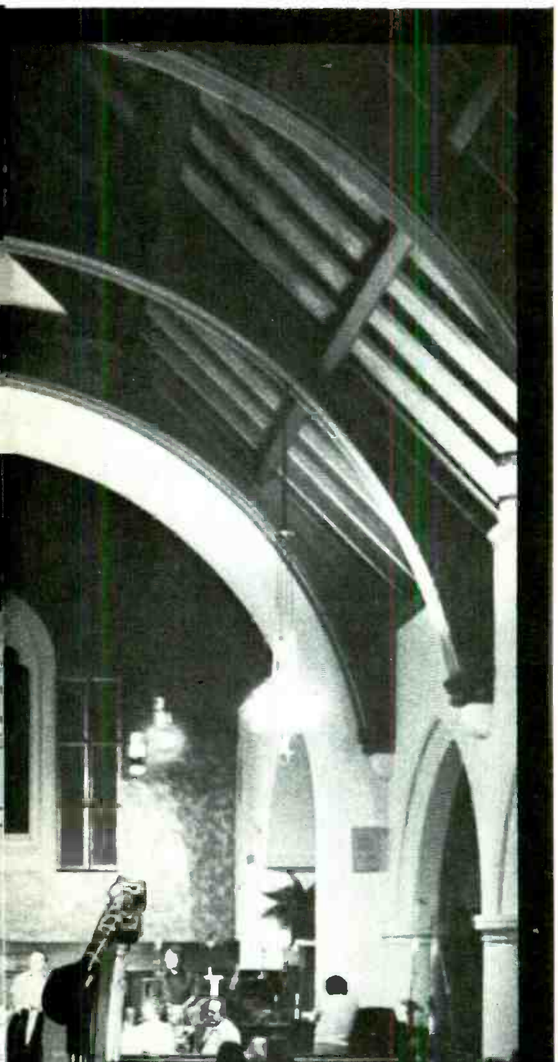
The lacquers are then inspected by hand. Most companies employ a plating technician to sit with a microscope and examine each groove of the mother. With a tiny needle the technician smooths out any plating ticks during quiet spots or on the fades. This process is called de-popping and is an arduous, time-consuming job, but no expense is spared in insuring the audible superiority of the direct to disc recording.

Every step of the normal recording and manufacturing process is tediously examined for possible improvements. When the stampers are ready to press the records, the question of vinyl becomes important. The quality of vinyl is directly related to the signal-to-noise ratio since better vinyl results in less surface noise. Conventional recordings may employ as much as sixty per cent reground vinyl—usually derived from defective or returned records. This reground vinyl causes the pops, clicks and hisses heard on most recordings. Every company making direct discs uses only one hundred per cent virgin vinyl.

Vinyl material in its natural state is



clear. Choosing a color involves argument. Some companies, such as Nautilus Recordings believe that "any color other than carbon black would result in a decrease in sound quality." Crystal Clear Records uses a special white vinyl compound and to date is the only direct disc company which utilizes the costly white vinyl. Their white pressing is thirty per cent thicker than the normal black pressing. Marty Ansoorian, of Award Record Manufacturers, responsible for the pressing of the white records for Crystal Clear, states, "Although not yet conclusive, the lack of traditional carbon black in vinyl may make for a quieter record."



More arguments are perpetuated when the decision has to be made as to the playback speed selected. Some companies use the standard 33⅓ speed while others opt for the faster 45 rpm. Michael Phillips of Crystal Clear states, "The use of 45 rpm as a disc speed enables the playback stylus to track high levels at frequency extremes with noticeably lower distortion, especially on the inner-most grooves." Other companies will not use the faster speed as it would offer the listener twenty-five per cent less music and reportedly could damage the finer cartridges designed to play records at the standard speed.

Even disc packaging has been scrutinized with a most critical eye towards disc protection. Nautilus Recordings has developed a lightweight package of polystyrene that purportedly protects the disc from dust, heat, warping, and the conduction of static electrical charges. The standard cardboard jacket and paper liner are eliminated. The new package has been designed to come into contact with the record only on the outer rim and the center label.

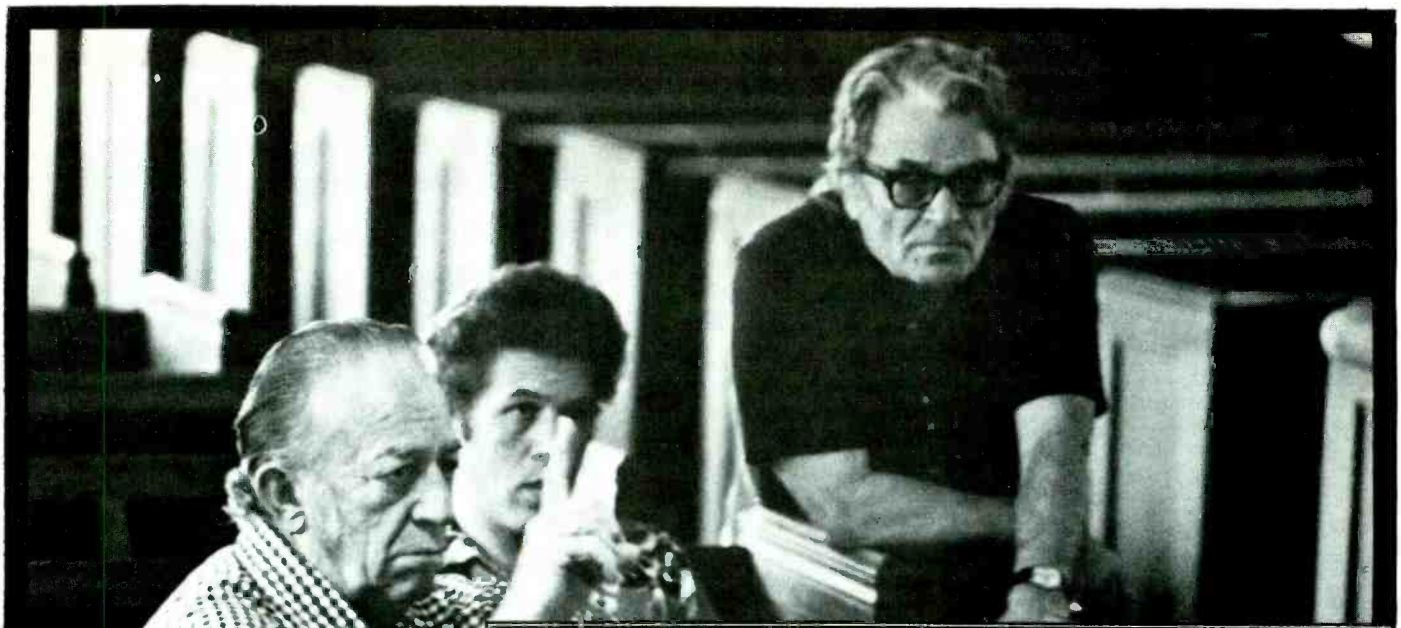
Up to this point it would appear that direct to disc recording is finally *the* answer. Not so. It does have one drawback. Not everyone will be able to enjoy these recordings. Because the tape medium is eliminated, the only record of the recording session is on the master lacquer that was cut in "real time." From these lacquers the mother masters and stampers are made. Only a limited number of pressings may be made from any master if the original quality is to be maintained, and once the stampers deteriorate, no further duplicates can be fabricated. At the very best, approximately 100,000 records can be produced. At the worst, production could be limited to only 50,000 pressings.

The final result then is an item of intentional quality, one that is truly a Limited Edition. The initial purchase price is understandably high, around twelve to fifteen dollars. Nautilus Recordings has gone one step further. Nautilus offers, in conjunction with their regular direct disc product, a hand signed, sequentially numbered edition of the first 1,000 pressings. The first 1,000 records were hand signed on the inner portion of the record by the artist. With a price of $25, each disc comes with a notarized certificate of authenticity. A purchaser of the specially signed and numbered edition will have the option of purchasing the same serial number of any future release.

When a direct disc becomes unavailable, its value soars. The first modern direct disc was released in 1969 by Sheffield Lab and sold for the then "outrageous" price of five dollars. Today, a copy in good condition can be sold for $200 and a copy still in the original shrink wrapping may get $350. The value of direct discs turns their limited availability into a small financial investment.

In a development unique to direct to disc recording, audio equipment manufacturers are beginning to play an increasingly important role in direct disc production. In what may be the first modern direct to disc commercial recording of a symphony orchestra, the Cleveland Symphony Orchestra, conducted by Lorin Maazel, has been featured in a project in which advertising, promotion and distribution to retail stereo outlets will be handled by Discwasher, Inc. of Missouri, the prime financial backer for the project.

Other developments find audio manufacturers financing direct disc by advertisements in the album itself. Audio Directions Grab Bag Inc. of Tennessee, released an album supported by six advertisers. In a double fold jacket, the inside cover is com-



(Above, left to right) Harry James, Lincoln Mayorga and Pee Wee Monte.

pletely taken up with advertising. The Ear Drum, a Los Angeles audio dealer with the largest selection of direct discs available anywhere, has the center section. TEAC, Micro-Seiki, Great American Sound, Dahlquist and Kenwood round out the rest of the inside cover, each featuring a model in the top of their line.

If the direct to disc record is so far superior, many may wonder why more musicians or major record companies refuse to become involved. The answer is simply that few musicians or record companies are willing to submit to the unforgiving circumstances of direct to disc recording. The limited pressing potential, particularly in view of the costs and risks involved does not attract major labels because of the inherent recording difficulties—and the fact that the album isn't likely to produce much profit.

Is it time to burn up all that nice tape you have collected over the years? Should you sell your tape deck and religiously convert to direct discs? Of course not. . . at least not until you have had the opportunity to compare a direct disc with a fine tape. The answer may then be obvious.

Available Direct Discs

What follows is basically a catalog of every direct disc recording currently known. Most of these records should be available through a local stereo dealer with an inventory of high quality audio products. If your dealer does not carry the album, the records may be available by mail. Write to: The Ear Drum, 5146 West Imperial Highway, Los Angeles, Ca. 90046.

SHEFFIELD LAB—Santa Barbara, Ca.

S-9	*The Missing Linc*—Lincoln Mayorga and Distinguished Colleagues, Volume 1. (No longer available)
S-10	*The Missing Linc*—Lincoln Mayorga and Distinguished Colleagues, Volume 2. (No longer available)
Lab 1	*Lincoln Mayorga and Distinguished Colleagues, Volume 3.*
Lab 2	*I've Got The Music In Me*—Thelma Houston and Pressure Cooker
Lab 3	*The King James Version*—Harry James and His Big Band
Lab 4	*Brahms, Handel & Chopin*—Lincoln Mayorga, pianist
Lab 5	*Discovered Again*—Dave Grusin with Ron Carter, Harvey Mason, Lee Ritenour and Larry Bunker

CRYSTAL CLEAR RECORDS—San Francisco, Ca.

CCS 5002	*Direct Disco*—Gino Dentie and the Family
CCS 5004	*San Francisco Ltd.*—Various artists
CCS 8001	*Virtuoso Guitar*—Laurindo Almedia

NIPPON PHONOGRAM CO. and **JAPANESE VICTOR CORP.**,—Japan

Eastwind	EW 10001	*The Three - Take 2*—Joe Sample, Ray Brown, Shelley Manne
Eastwind	EW 10002	*The Pentagon*—Cedar Walton
Eastwind	EW 10003	*Pavane Pour Une Infante Defunte*—L.A. Four
Phillips	PD 10001	*Ole*—Belmonte and His Afro Latin 7
Phillips	PD 10002	*Latin Roots*—Chico O'Farrill and the N.Y. All Stars

(The Japanese direct discs are currently sold only in Japan but plans are being made to import the records into the U.S. in the very near future.)

UMBRELLA RECORDS—Canada

(Available in the U.S. through Audio-Technica dealers.)

DD1	*Rough Trade Live!*
DD2	*Nexus*
DD3	*Violin and Piano Sonatas of Effram Zimbalist Sr. and Jr.*

AUDIO DIRECTIONS GRAB BAG INC.—Tennessee

Direct Disc

M&K SOUND INC.—Beverly Hills, Ca.

10014	*Blu*—Joe Marcenkiewicz
10015	*Blu, Jam Session*—Joe Marcenkiewicz

NAUTILUS RECORDINGS—Shell Beach, Ca.

The First In Line—Randy Sharope

SOUND 80 INC.—Minneapolis, Minn.

Natural Life (availability doubtful)

IT TAKES A VERY SPECIAL CASSETTE DECK TO GET SO MUCH BEAUTIFUL MUSIC OUT OF SOMETHING THIS LITTLE.

The recording tape in a cassette is only an eighth of an inch wide.

Crammed into that eighth of an inch may be as many as 64 tracks mixed down to two. A hundred musicians. Countless overdubbings. Not to mention the entire audible frequency range.

Any cassette deck can reproduce part of what's been put down on that eighth of an inch.

But the Pioneer 9191 was designed to reproduce all of it. Superlatively. Without dropouts, unacceptable tape hiss, or noticeable wow and flutter.

Take our tape transport system, for example.

Since the tape in a cassette moves at only 1-7/8 inches per second, even the most minuscule variation in tape speed will make a major variation in sound.

To guard against this, where most cassette decks give you one motor, the 9191 comes with two. The first is used only for fast forward and rewind, so the second can be designed to maintain a constant speed for play and record.

Of course, having a great tape transport system means nothing if you don't have great electronics to back it up.

We do.

The 9191 comes with an advanced three stage direct coupled amplifier that extends high frequency response and minimizes distortion.

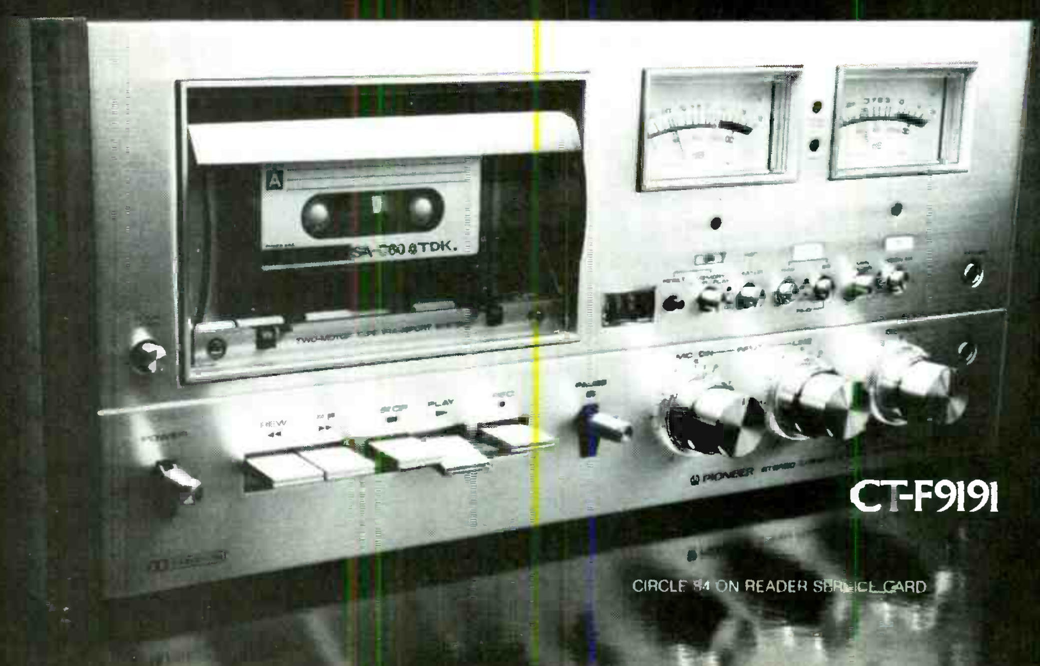
Our Dolby system can reduce tape hiss by as much as 10 decibels in high frequencies.

And our peak limiter lets you cram as much as possible onto a cassette without distortion.

There's also a memory for going back to a favorite spot on the tape automatically. Separate bias and equalization switches for getting the most out of different brands of tape. And electronic solenoid controls that let you go from play to rewind, or from rewind to fast forward, without hitting the stop button, and without jamming the tape.

Go slip a cassette into a Pioneer 9191 at your local Pioneer dealer.

You'll find it hard to believe such a little thing could come out sounding so big.



CT-F9191

PIONEER

High Fidelity Components
U.S. Pioneer Electronics Corp
75 Oxford Drive, Moonachie, New Jersey 07074

CIRCLE 84 ON READER SERVICE CARD

PROFILE

By Russell Shaw


Pete Lowry: Recording the blues.

Five years ago, Pete Lowry was a Biology instructor at the State University of New York in New Paltz. His hobby bore little resemblance to his main vocation; rather than go on field trips in search of new fauna, he'd spend his summers traversing the Southland, recording solo blues musicians. Financial rewards were not forthcoming, but the satisfaction of "vinylizing" a vanishing art form was.

In 1972, the axe of academe struck. Mr. Lowry, unequipped with a doctorate, was deemed unqualified to teach students the intricacies of biology. Yet rather than opt for a teaching career on a lower level, he took the dismissal as impetus to establish Trix Records. In the ensuing five years, the company has managed to garner a reputation in the blues world for its peerless dedication to the folk form, specifically exemplified by the fifteen releases issued under its aegis.

Lacking the high-force promotion team of the major labels, Lowry's Trix is, in fact, a one-man operation. Indeed, his business cards attest to this, reading: "Pete Lowry, President, Engineer, Shipping Clerk." The usual jack-of-all-trades trip. Concurrently, if the wearing of many hats is conducive to gaining a broad perspective on a given situation, than Lowry, 35, is the man to talk to about recording the blues, the state of the idiom, and the many trials and tribulations inherent in working with a folk form which, frankly, doesn't exactly cause stampedes to the record stores of America.

Pete was on one of his periodic field trips when Modern Recording *sat down to tape some of his ideas and observations on his individualistic existence. Part stoic (necessary for perpetuation of sanity), part court jester, and distinctly intense, his views are both informative and relevant.*

MR: Have you always been into the blues, or was there one particular event which alerted you to the idiom?

PL: I had gotten into the blues earlier than most people—around '58 or '59. My first love, however, was jazz; something that I picked up from my father. In high school, my father had a guy working for him in a packaging plant who used to play first trumpet with Harry James and Tommy Dorsey. He subscribed to *Downbeat*, the jazz magazine, and after he was finished reading each issue, I got his copies. I read a review of a Big Bill Broonzy record; the critic gave it five stars, so I went out and bought it. That was my first exposure to pre-war solo blues.

MR: What factors were involved in your transition from blues fan to blues researcher?

PL: Around '63, I started writing for off-the-wall blues magazines. I was in graduate school at the time, so this was a good outlet. My initial experiences were quite favorable. My first interview was with B.B. King. It was positive reinforcement; I got back stage at the predominantly black Apollo Theater in Harlem, and there was no uptight hostility on anyone's part. I'm lucky I started out with something like that.

MR: How did the recording aspect blossom out of the journalistic endeavors you involved yourself in?

PL: This grew out of research I started about 1969. A few years prior, while in Europe attending medical school, I met Bruce Bastin. Bruce is a scholar who at the time was contemplating writing a book on Georgia and Carolina blues. So, in 1969, we got together in Atlanta and interviewed a blues artist named Buddy Moss. This, in turn, provided impetus for the beginnings of Trix Records in 1970. I had done some recording and felt uncomfortable sitting on [holding onto] them, so I put out two 45s—one by Eddie Kirkland, "Eddie's Boogie Children"/"Lonesome Talking Blues" (Trix 4501) and one by Baby Tate, "See What You Done Done"/"Late in the Evening" (Trix 4502).

MR: You obviously didn't have a "hot" product on your hands. What were some of your early marketing techniques?

PL: I pressed a thousand of each, and gave a bunch to both of the artists. They sold some to their friends, and the copies I had left, I peddled by mail order the best I could. I issued some more forty-fives, but it was still informal. I was not incorporated, and I was still teaching. Therefore, most of my activity was undertaken during summer vacation. I'd travel a thousand miles a week. The 45s served a purpose; they showed people what I had, and that the material was out there.

MR: How did you find the artists you recorded?

PL: It wasn't a case of blind-driving through the boonies; most of the people I already had worked with led to somebody else. Plus, when I worked as a researcher with Bruce Bastin, I gained many contacts and leads which ultimately were useful in finding the whereabouts of various artists.

MR: You've done a good deal of recording in the field. Many anthologists, such as Alan Lomax, have tried, in their work, to present an authentic scenario—dogs and chickens in the background, etc. Yet your records seem to have a "cleaner" sound. How does your approach differ?

PL: On my records, you'll never hear dogs barking, chickens squawking, kids screaming. You'll hear the artist; that is the most important thing.

Someone like Lomax just turns on the tape recorder and lets it go, but I find that stuff a little annoying. I'll use the Holiday Inn "Recording Studio"—my motel room—or else find someplace in the country where it is quiet.

MR: Oftentimes, it is just a singer with a guitar. Do you like to record an album in one sitting, or do you prefer several sessions?

PL: Doing a whole album in one sitting is a bit boring. I like to try for as much variety as possible—in material as well as in sound. I like to record in at least three sittings.

MR: Why don't you take us through a description of your recording equipment?

PL: Before we talk about the various technical instruments I use, I'd like to say that I travel about with several guitars, many of which I use on sessions. Frequently, an artist will not own a guitar, or have one in really terrible condition. So, I have a Gibson SJ flattop, a Vintage 39 National, and I use one or two others from time to time.

MR: Tape recorders and mics?

PL: I started out with a Uher 4200. This is one of the best battery-operated portables made; newsmen carry those things around. I prefer to run my taping equipment at 7½ ips. Some people consider that a drawback but it was the industry standard for quite some time. I use Sony ECM 21 microphones. They have little amplifiers built into them. The frequency response is as flat as you can get, and I'd rather get stuff flat on a master tape, because you can mess with it later. Gradually, I've worked up into an inventory of six ECM 21s, and two Sony 64Ps—omni-directional mics which are better for vocals. I don't like to use 21s for vocals; they are very sensitive, overload easily and will pick up voice pop. The 64s, however, can damn near be swallowed by the singer but won't react in the same way. The 21s, through, are especially good for picking up acoustic guitar.

MR: What is your general attitude during a session; are you a libertarian or a disciplinarian?

PL: I tend to work with the artist. Someone like Eddie Kirkland, from Macon, Georgia, is prone to making mistakes but is professional enough to build them into what he's doing. Of course, if it is a real heiner, we'll do another take. Basically though, I let the artist do what he knows; making suggestions and trying to draw things out of him. I like to set things up ahead of time rather than pursue a "play-as-you-go" philosophy.

MR: How many tracks do you use in your studio sessions?

PL: Usually eight, sixteen only when absolutely necessary. For example, when we recorded an album with Robert Junior Lockwood [to be released] we used eight tracks—the drums on three tracks and one each for everybody else. I made sure everything was done flat. By flat, I mean keeping the equalization neutral, with no reverb or echo. You can add it later, but if you include those features on a multi-track master, you are stuck with it.

MR: Do you have a favorite recording studio?

PL: Well, I've been doing some stuff at Minot Studios in White Plains, New York. Eddie Kirkland's latest album [yet to be released] is being recorded there. The studio is used by Strata-East [a jazz label] and by a country-pop singer named Chip Taylor. They have all the modern equipment you'd want, plus plenty of good vibes.

MR: How do you cut your masters?

PL: I put my two-track masters on a Teac 3340; put them through a Sony MX 12 mixer, through a Harmon-Kardon preamp which has a small equalizer going into it, and then ultimately through a Revox A77. I roll the Revox at 15 ips—it is much easier to edit that way. My equalizaiton is kept pretty normal; sometimes though, I might have to cut out the really deep bass response in order to tone down a pounding foot.

MR: What brand of recording tape do you favor?

PL: With the Uher, Scotch 203; with the Revox Master, Scotch 206.

MR: Where are your recordings pressed?

PL: I use Peerless Plastic Molding Company in Plainfield, New Jersey. They do nothing else but this and perform a fine job. I've pressed approximately 15,000 albums, with a dozen or so defective returns. A small label like mine can't afford to put out shitty pressings.

MR: Once your records roll out of the plant, how do you get them out on the marketplace?

PL: Every small label will tell you that distribution is tough. They [distributors] are certainly more interested in peddling Motown than Trix, Flying Fish or Alligator. I've got about a dozen distributors spotted through the country, but I have some gaping holes, particularly south of a line running from Los Angeles, through Kansas City, to Washington D.C.

MR: Being white, do you sense any hostility among blacks for exploring "their" music?

PL: I've had no problem dealing with blacks in the South. I tend be laid back, which helps things. Plus, we've never gone into a situation totally cold; we always have had leads. Yet we have had some problems with whites. One time, we ran into an illiterate deputy sheriff in Newton County, Georgia. We were out in the boonies recording Roy Dunn; some white folks saw a car with out-of-state license plates hanging around a black man's house. They were paranoid, and eventually had both ends of the road staked out. We persuaded the deputy that we weren't doing anything illegal. Ironically, the county sheriff was either the son or the nephew of Howard Odom, who collected black music around Newton County about 1920.

MR: The past few years have seen the deaths of many prominent blues artists. Is the blues dying?

PL: Certain styles of blues have a very good likelihood of dying. For example, there are very few young acoustic solo artists around today. However, it will take longer for the modern electric styles to die out.

MR: Will the blues remain an eclectic music appreciated by few, or will it become a popular form?

PL: That is up to the record companies. We smaller labels have no money for promotion, certainly not for payola. The major companies, however, don't want to know about it. Artist and repertoire is controlled by the legal people; what is kept in catalogue is determined by the accountants. You can't get stuff placed in a black record store or played on a black radio station. The attitude among younger blacks seems to be that this is music of a bygone era, and that you can't do the Hustle or Bump to it. Yet when all the acoustic artists die off, they will wonder why funny looking white guys with long hair and beards were the only ones to care about this music.

* * *

For a complete Trix Records catalog, write Drawer AB, Rosendale, N.Y. 12472.

Ambient Sound

BY LEN FELDMAN

All Those Tone Control Switches and Knobs!

The high fidelity component industry seems headed in two directions at once (at least, as far as tone control features on amplifiers and receivers are concerned). On the one hand, some of the makers of so-called "esoteric" preamplifiers and amplifiers have abandoned bass and treble tone controls altogether and are now producing ultra-low distortion products which purport to approximate the long sought after "straight wire with gain." The presumption on the part of these manufacturers is that the sophisticated audiophile is going to seek out the very finest loudspeakers obtainable which will deliver "flat response" into a perfect acoustic environment that requires no tonal compensation. Total abandonment of tone controls makes an even more unlikely assumption: that *all* program sources (discs, tapes, FM programs) likely to be played over the tone-control-less system have been produced with optimum equalization in terms of the listener's preference.

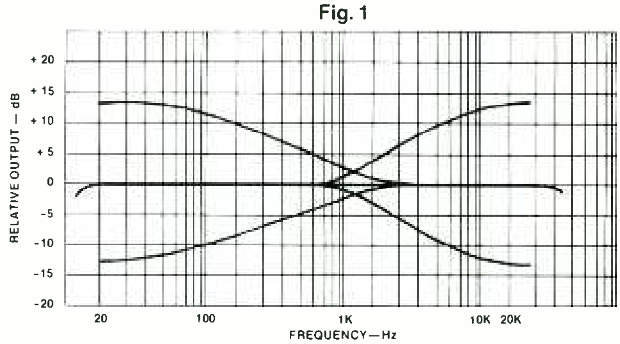
If you confront any of these manufacturers with the realities of audio life, they will quickly point out that if, indeed, your room acoustics are not what they should be or if you have been less discriminating in your program source selection than a purist should have been, you can always resort to the incorporation of one of the new multi-band equalizers to correct such sonic aberrations. But, as we pointed out in a previous column, making best use of a graphic equalizer is not that easy and often results in overall sound that is either distorted (because of excessive boosting of certain octave-bands) or more unbalanced than it was in the first place.

The second, less radical school of audio designers, confronted with the growing popularity of graphic equalizers, has sought to develop a series of switching and control alternatives which, while falling somewhat short of the flexibility of a multi-band graphic equalizer, affords a degree of tonal compensation which far surpasses that provided by the familiar bass and treble controls.

Variable Turnover Tone Controls

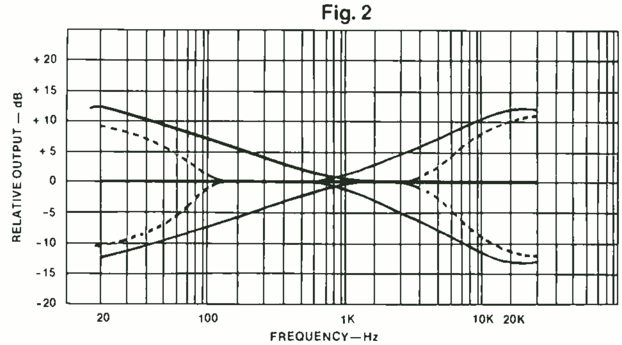
The first variation of the standard bass and treble control to appear on home hi-fi was the variable turnover bass and treble controls. In the standard bass and treble configuration, bass or treble boost or cut is usually pivoted around a frequency of 500 Hz to 1000 Hz. The total range of such controls is usually about ±10 or 12 dB, at 100 Hz and 10 kHz respectively,

Fig. 1



which, if plotted at extreme boost and cut settings of each control results in the familiar "bow-tie" graphic plots of Fig. 1. While providing a very audible "change" in tonal quality (and hence easily demonstrable to the audio neophyte), the usefulness of such tone controls as a means of re-establishing flat energy response in a listening room is highly limited. More often than not, what is really wanted is the ability to boost frequency extremes by moderate amounts to compensate for extreme low-end or high-end roll-off of woofers and tweeters. Attempting to achieve such boost of frequency extremes results in undesired boost (and attendant coloration) of upper-bass, lower-mid and lower-highs simultaneously.

Fig. 2

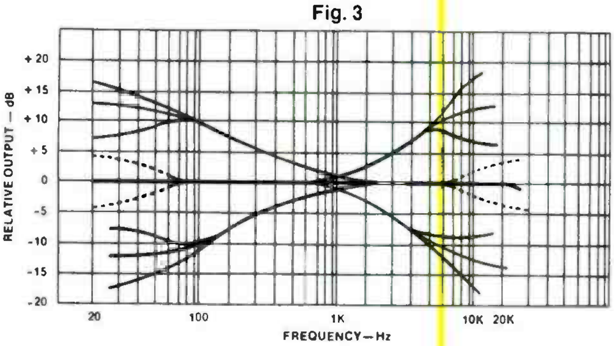


By offering alternative turnover points, a manufacturer increases the utility of the normal bass and treble controls by providing optional frequencies at which the tone control "begins to act." Typically, extra three-position switches are associated with the rotary bass and treble controls. One position for the bass switch might be labelled 500 Hz while another position might call for a turnover frequency of 125 Hz. As for the treble control, turnover frequencies of 1 kHz and 5 kHz might be provided. The third position might be used on each of these switches to defeat or by-pass the tone controls altogether—both to satisfy the purist who still prefers not to incorporate tonal compensation and for making instant listening comparisons between compensated and uncompensated sounds. Figure 2 illustrates the total range of control that might be provided with dual-turnover tone controls engineered for the frequencies discussed. The dotted lines represent available bass and treble range when switches are set to the 125 Hz and 5 kHz points, while solid curves are similar to those shown earlier in Fig. 1. Clearly, setting the switches to the 125 Hz and 5 kHz positions gives the user better control over response at the frequency extremes without affecting the "flat response" of the system over the balance of the audio spectrum.

Main and Sub Tone Controls

At least one well known hi-fi component manufacturer has solved the problem of increased tone control flexibility somewhat differently. Several of Pioneer's integrated amplifiers and receivers use "main" and "sub" tone controls for bass and treble adjustment, with knobs usually scaled to match these designations. The sub-controls affect frequency extremes but operate in addition to the main controls, which provide the usual center-frequency pivot points. In one sense, this arrangement actually provides even more control possibilities than do the variable turnover switch/control combinations just discussed, since it becomes

Fig. 3



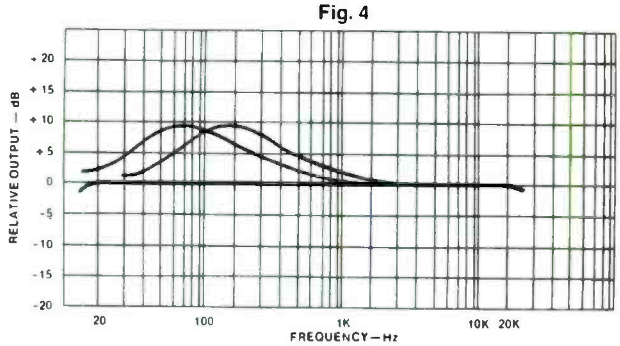
possible to raise or cut overall bass or treble response and then emphasize or attenuate frequency extremes over and above the adjustment "trend" imparted to half of the audio spectrum. Examples of the "piggy-back" range of adjustment afforded by the main/sub control arrangement are illustrated in Fig. 3.

Poor Man's Graphic Equalizer

When a manufacturer equips his receiver or amplifier with a mid-range (sometimes known as a "presence") control, he is just one step short of providing a minimal graphic equalizer built into his product, albeit a three-band equalizer. This control's alternate name derives from the fact that vocally sung frequencies contain predominant energy in the region from 500 Hz to about 3,000 Hz and if these frequencies are boosted while playing a combined orchestral/vocal program source, the apparent audible effect is to "bring the vocalist forward" in the resulting sound field—or to lend a sense of "presence" to the soloist. The mid-range control acts as a sort of wide-band notch or boost filter, with its center frequency located somewhere between 1 kHz and 2.5 kHz. One manufacturer has already elaborated upon this mid-range adjustment feature by providing a choice of two center frequencies in much the same way that switch positions are provided for multi-turnover bass and treble controls. The lower of these selectable mid-frequency boost positions tends to add "presence" to male voices while the upper frequency creates the same effect for female voices. The manufacturer makes no suggestion regarding what to do in the case of a male-female duet!

One manufacturer has solved the problem of lower-bass boost by adding a rotary switch to the front panel which operates independently of the ordinary tone controls and, in association with an adjacent rotary knob, applies fixed amounts of bass boost from about 125 Hz

Fig. 4



downward (without affecting upper bass or lower mid-range tones). The switch selects center frequencies of either 80 Hz or 125 Hz, as illustrated in the diagram of Fig. 4. The presumption here is that the feature will be used primarily for speaker system performance enhancement—and not for compensation of program source material.

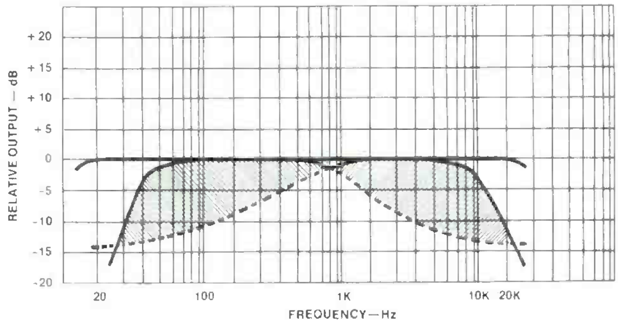
A "Tilt" Control

One high end company has introduced what is perhaps the most subtle (yet simplest to operate) tone control of all. Known as a "linear equalizer," this control featured on Luxman amplifiers and preamps consists of a single front panel knob which, in its several settings, actually "tilts" the entire response curve of the system by minute amounts. Turning it to the left of center positions tilts the playback response curve so that bass is somewhat attenuated (no more than a dB or two for each step) while simultaneously, treble is slightly boosted. Turning it to its right of center settings reverses the process. The chief purpose of this all-in-one secondary tone control is to take care of "disagreements" between the listener and the recording engineer whose ideas of proper equalization may not correspond exactly with one's own listening tastes.

Filters vs. Tone Controls

So-called high-cut and low-cut filters (often supplied on equipment with multiple choices of cut-off frequencies) might seem, at first, to be nothing more than fixed-position bass and treble cut controls. If properly designed, however, their purpose is quite different from that of ordinary bass and treble knobs. The purpose of a high-cut filter is to attenuate random high-frequency noise (hiss from tape, record surface noise, FM background noise) while imposing as little alteration to musical response as possible. This is accomplished by combining two ideas: shifting the turnover or cut-off point to a high enough frequency so that most musical energy is unaffected while higher noise or hiss frequencies are attenuated, and providing a steep cut-off rate or slope so that even though response remains flat to, say, 6 or 8 kHz, at 15 kHz or above, response may be down 12 or more dB. The same principle applies to low-cut filters, except that the amplitude attenuation now occurs at preselected bass frequencies. The diagrams of Fig. 5 clearly show the difference between the action of bass or treble cut controls (dotted curves) and that of properly designed low and high-cut filters. Either type of control delivers the same attenuation at, say, 30 Hz and 12 kHz, but achieving this degree of attenuation at those "noise" frequencies using cut filters preserves (or attenuates to a lesser degree) *musical* frequencies contained within the shaded areas shown in the diagram. Often, manufacturers will supply 6 dB/octave slope filters as opposed to the 12 dB/octave steeper slopes illustrated in the diagram of Fig. 5. While such slope rates are not as effective in removing unwanted noise, some audio people claim that the steeper 12 dB/octave slopes, particularly in high-cut filters, provide such extreme phase shifts when introduced into an audio chain that a secondary form of coloration is introduced into the music. Undesired phase shift is one of the reasons why

Fig. 5

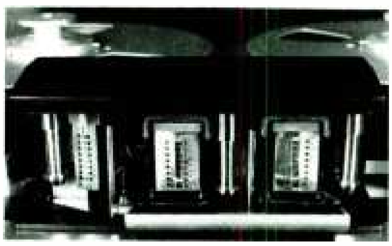


even some of the most complex tone control systems are equipped with a by-pass switch. It is also the reason why, as we said earlier, many audiophiles wouldn't be "caught dead" owning a component equipped with any form of tone control circuitry—however simple or complex. We certainly don't intend to take sides in the pro-or-anti tone control debate but hope that this brief description of the various types and permutations of controls available will help you to decide just how much tone correcting circuitry you need in your hi-fi system—or if you really need any at all.

OTARI MX-7308: Unquestionably your best buy in one-inch eight-track machines.

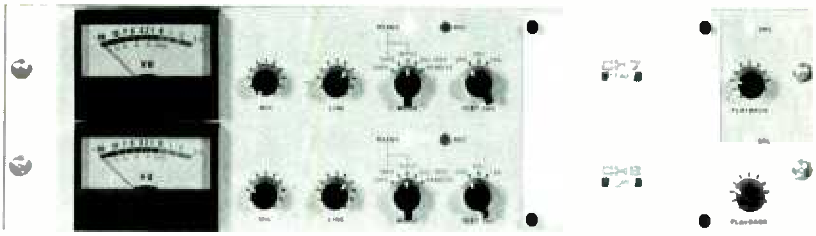
If you're tuned into the multichannel scene these days, you may have heard this news: Your best value by far in one-inch eight-track machines is the Otari MX-7308. Why? We believe it's because it has the same performance and features as the other eight-tracks, but costs 20% to 25% less: $8150 including console. That's enough savings to let you buy a two channel mix-down machine as well.

And when you stack it up against used machines (most of which sell for about the same price as a new factory-fresh Otari), you find the MX-7308 gives you several new features that just weren't around in the old days. Things like motion sensing, reliable FET equalization switching, reel tension servo, long life deep gap heads, and LED peak reading indicators, among others. Consider these features:



Long-life deep-gap Otari heads; plug-in convenience.

Heads: Superior quality extra long-life deep-gap heads brass block mounted Permalloy construction for even wear and quiet performance; plug-in for ease of removal and realignment. These heads really deliver the goods and they keep on delivering session after session.



Superior electronic performance; all front accessible adjustments.

Electronics: S/N greater than 65 dB; synchronous reproduce response to 15 kHz for excellent fidelity when ping ponging; large standard VU meters *plus* peak reading LED indicators; professional XLR connectors; balanced 600 ohm outputs at +4 or +8 dBm; all electronics adjustments front accessible; plug-in PC boards; built-in two-frequency test oscillator to set bias and record EQ; bias test points on rear panel.

Excellent start time and tape handling from integral reel servo.

Transport: 30 and 15 ips speeds (15 and 7½ ips on special order); smooth, gentle tape handling with fast start time (500 milliseconds at 15 ips); hinged access to transport adjustments; motion sensing to prevent tape damage; full edit mode to spill tape; head lifter defeat for cueing; rugged, heavy duty power supply; all steel console—built like a tank.

Want to know more? Call us for the name of an MX-7308-equipped studio near you so you can hear their story first hand.

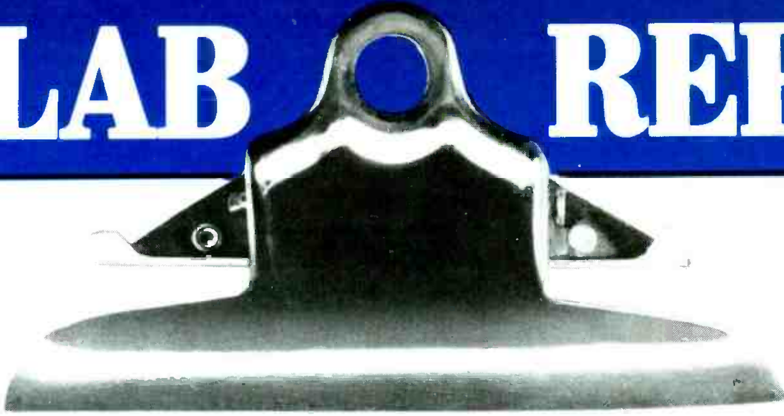
OTARI

Otari Corporation
981 Industrial Road
San Carlos, California 94070
(415) 593-1648 TWX: 910-376-4890

Manufactured by Otari Electric Co., Ltd., Tokyo, Japan

MODERN RECORDING

LAB REPORT



NORMAN EISENBERG AND LEN FELDMAN

Pioneer RT-2022/RT-2044 Tape Recorder

General Description: The last two numbers in the model designations of this recorder indicate the head configuration. RT-2022 is a two-track, two-channel version; RT-2044 is a four-track, four-channel version. Either way, the complete product consists of the tape transport model RTU-11 and the separate electronics embodied in the amplifier unit model TAU-11. For two-channel, two-track recording and playback, one TAU-11 amplifier is required. For the four-channel version, two amplifiers are used. Conversion of the transport from the "22" to the "44" configuration (or vice versa) may be accomplished by substituting one complete head assembly for the other, which is held in place on the transport by allen-type screws (allen wrenches are supplied with the extra head assembly). The necessary adjustments and interconnections for both "22" and "44" configurations all are built into the system. Relative positioning and placement of the units is fairly flexible. If one wants the completely vertical look, there are latches and mounting hooks to secure the sections together. There also are carrying handles and even a compartment on top of the TAU-11 for storing cables.

Transport and amplifier(s) are linked by multi-pin socketed cables and standard signal cables (all supplied). The RT-2022 setup provides for two-track, two-channel stereo record and play. The RT-2044 setup provides for four-track, four-channel record and play (requiring the use of two TAU-11 amplifiers), and for four-track two-channel record and play (requiring the use of one TAU-11 amplifier). From the TAU-11 (or TAU-11s as the case may be), the usual signal inputs and outputs to and from other equipment are made.

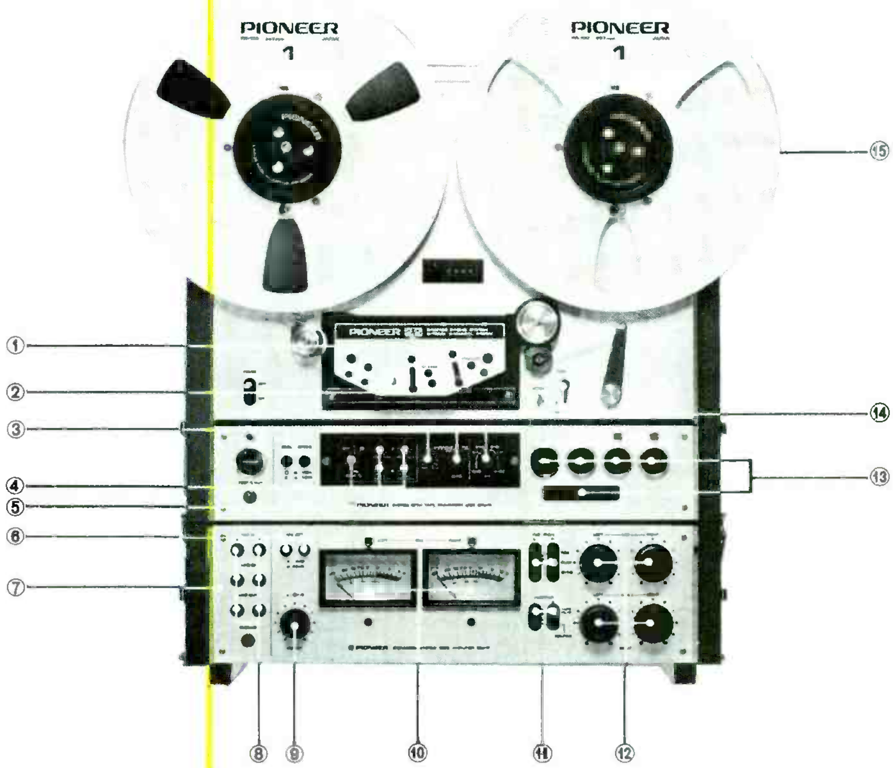
Tape speeds are 15 and 7½ ips. The transport handles reels up to the 10½-inch (NAB professional) diameter. Separate heads provide for erase, record and play.

The transport is laid out in standard fashion, with the supply and takeup hubs at the top, and a four-digit index counter and reset button below them. The head assembly is flanked by a guide-roller arm at the left and the capstan and pinch-roller at the right. Farther to the left is a pause button. Below the capstan is a cue-lock button and an associated cue lever. To their right is a tape tension arm which incorporates an automatic shutoff.

Transport action buttons are grouped at the lower right bottom section. These are fast-responding buttons that permit changing from one function to another without the need to first press the stop button. Functions include rewind, fast-forward, play, record and stop. To record, both the play and record buttons must be pressed simultaneously.

At the left corner of the transport are the power switch and pilot lamp; an output jack for a test signal from a built-in oscillator which can be "tapped" for adjusting other equipment; a reel-size adjustment switch; and the tape-seed selector.

Centered at the bottom of the transport are the following: a control to select either 1 kHz or 10 kHz as the test-oscillator signal (the 1 kHz frequency is used for bias adjustment; the 10 kHz signal for head and frequency response adjustments); a group of four controls for adjusting the level of the test-oscillator signal (use of 2 or 4 of these depends on whether the system is the "22" or the "44" version); the bias current adjustment control and an associated bias selector which chooses between "fix" or "variable" (in "fix" position, bias current is standard for Scotch 206 low-noise tape; in "variable" position, bias current may be altered to suit other kinds of tape); and finally an equalization selector. Details on using these controls are carefully



(1) 3 -Head Plug-In Head Unit (2) Scrape Filter (3) Bias/Equalizer Tape Selector (4) Built-in Test Oscillator (5) Playback Preamplifier (6) MIC Attenuator (7) Independent LINE/MIC Input Circuit (8) Jack for Inputs/Outputs (9) Output Level Control (10) Wide-Range Level Meters (11) Synchromonitor Mechanism (12) Input Level Controls with a Memory Marker (13) Feather Touch Operation Buttons (14) Cue Lever with Lock Button (15) 10½-inch reel

spelled out in the owner's manual, including an unusually long list of tapes by brand-name and number with recommended EQ and bias settings for each. Instructions for using the test oscillator signals to check frequency response, and also to make one's own test-tape, also are given.

At the rear of the transport unit are several important connections, more involved than the usual recorder's rear panel. To begin with there are three multi-pin sockets. One is for an optional remote-control accessory. The other two interconnect the transport with the TAU-11(s). In addition there are several pairs of signal jacks that must be interconnected with the TAU-11(s) for normal signal transfer back and forth, as well as for a sync feature, whose action may be engaged by a control on the TAU-11, and which—as is customary in this type of operation—permits the recording head to act as both recording and playback head so that new signals may be added to existing signals (on another channel) in perfect synchronization. The sync, output play and input jacks all are color-coded and correspondingly color-coded signal cables are supplied with the equipment. The rear also contains a power socket for the separate AC line cord, an unswitched AC convenience outlet (300 watts maximum) and a system grounding terminal. The transport has vinyl covered sidepanels and may be fitted when not in use with a front cover (supplied).

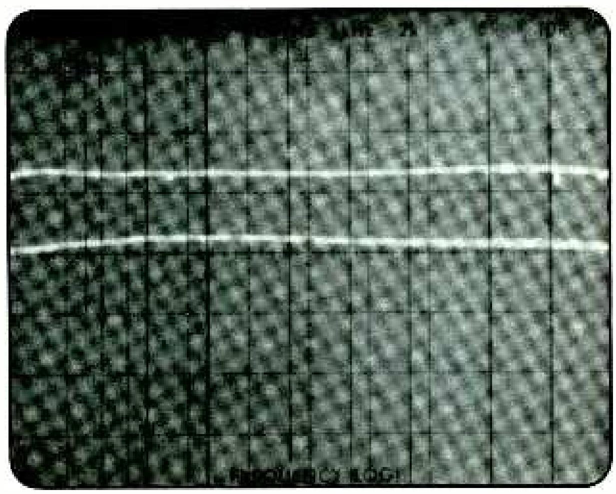
The TAU-11 amplifier has a very "busy" looking front panel which is dominated by a pair of VU meters calibrated from −40 to +6. They operate in both record and playback and each has an indicator lamp for the recording mode, as well as a zero-adjustment screw. Left of the meters are various signal jacks: stereo pairs for mic in, line in, line out; and a stereo headphone jack. In addition there are mic attenuator buttons (0 dB; 20 dB down); and dual-concentric output level controls which handle line out and headphones. The output level control has a click-stop at the optimum level setting for standard recording tape (Ampex), 450 mV at 0 dB.

Just to the right of the meters are monitor selector switches, separate on each channel; and recording mode switches, separate on each channel. The latter set up the system for recording, or playback, or the sync recording-monitor option. Farther to the right are the input level controls, separate on each channel and separate for mic and line. Input mixing of line and mic is of course feasible with this arrangement.

The rear of the TAU-11 contains the mating multi-pin socket for the cable from the transport, plus the corresponding color-coded signal jacks that link it to the transport for sync, input play and output. For line out play there are two sets of stereo jacks; similarly there are two more sets of stereo jacks for line input (recording) signals from external sources. When the front-panel line-in jacks are used, the rear-panel line-in jacks are disconnected. The amplifier's sides match those of the transport.

In addition to the recording functions described, the RT-2022 (or 24) can be used with an external timer, can make sound-on-sound recordings and echo-effect recordings. It also can make "run-in" recording (termed here "follow-up" recording) whereby you can go into the recording mode directly from playback without first stopping the transport. With either head configuration it is also possible to record monophonically.

Test Results: MR's tests of this unusually complex and versatile recorder produced results that easily confirmed, or were better than, the manufacturer's published specifications. At either speed, wow and flutter were very low; distoriton was well under the claimed figure; response extended higher than claimed; signal-to-noise was better than claimed. In our lab tests we used TDK Audua back-coated tape for all measurements. This tape is "hotter" than Scotch 206 and no doubt accounted for the consistently "better than spec" results obtained. It should be pointed out that Pioneer's frequency response claims were made with reference to a −10 dB recording level for the 15-ips speed, and with reference to a −20 dB level for the 7½-ips speed. When MR attempted to duplicate their measurements, results were so much superior to those claimed, that we jacked up the test levels and proceeded to take response curves at a 0 dB and a −10 dB level, respectively, for the 15-ips and 7½-ips speeds.



Pioneer RT-2044: Record/play response at 15 ips (upper trace) and 7½ ips (lower trace).

Even so, results were still far better than claimed by the manufacturer. Unweighted wow-and-flutter measurements made on our test sample were actually superior to the claimed weighted wow-and-flutter figures at both speeds.

Mechanical operation of the transport was flawless. In half-track mode (the "22" configuration), we found that we picked up a few dB of signal-to-noise capability, although distortion and frequency response remained essentially the same as in the "44" configuration. The accompanying 'scope photo, taken from the face of our spectrum analyzer, shows the r/p response from 20 Hz to 20 kHz for both the 15-ips speed (upper trace), and the 7½-ips speed, the latter measured at a −10 dB lower recording level. In both instances, even the first signs of high-frequency rolloff occur beyond the range of sweep, or well above 20 kHz. Even the response obtained by using the record head as a playback head during sync mode was better than expected—being down 3 dB at 10 kHz.

In all, the Pioneer RT-2022 or 2044 combines a high order of electronic and mechanical excellence, with a number of eminently sensible, useful features and options. The owner's manual is very thorough, painstakingly detailed and amply illustrated.

General Info: Transport (RTU-11) dimensions: 15½ inches high, 17 5/16 inches wide, 8 3/16 inches deep. Weight: 15 lbs., 4 oz. Amplifier (TAU-11) dimensions: 4 13/16 inches high; 17 5/16 inches wide; 8 3/16 inches deep. Weight: 11 lbs., 5 oz. Supplied with two 10½-inch reel adapters; head-cleaning kit; dummy plug; AC power cord; lock plug; multi-pin connector cables; pin-plug cables; phone-jack cables; felt cushions; cover. Prices: Model RT-2044, $1600; model RT-2022, $1250.

Joint Comment by L.F. and N.E.: It took Pioneer thirty-eight pages of owner's manual to describe the various features and extreme flexibility of this open-reel deck, and we can hardly cover it all in the space allotted here. Summarizing our mutual reactions to this machine we can say that there doesn't seem to be a thing that Pioneer has overlooked. And everything appears to be in tip-top form, electrically and mechanically.

Of course, this is not a recorder for the rank amateur or neophyte. Given its many adjustment capabilities, its variety of bias and EQ settings, not to mention its recording-mode versatility, you had better know what you are doing before tackling this deck. But if you are patient enough to wade through those thirty-eight pages of instructions, you probably will find there is little you could ask of a high-quality reel-to-reel recorder that the RT-2022 or 44 does not provide.

The transport, as supplied, was fitted with the "22" head. Accompanying it was a separate box containing the "44" head. Replacement of the former with the latter, and alignment of the new head assembly went off without a hitch, thanks to the built-in test signals and the use of the machine's VU meters to perform azimuth alignment and optimum bias adjustment.

The idea of providing a fixed bias switch position as well as a variable position (on the front panel, at that) is an excellent one since it permits the owner to work with a variety of tape types without ever having to upset the bias carefully adjusted for one's most often used "standard" tape.

Mechanically, the transport operated without a flaw. We did feel that the time it took to come to a full stop after operation at one of its fast-wind modes was a bit longer than we would have liked, but this did not prevent us from being able to "rock" the transport (from rewind to fast forward) with great precision for cueing and editing.

We both agree that the separate electronic module concept is a welcome professional touch that makes for flexibility in installation, and also permits easy changeover from the "22" to the "44" arrangement. We also liked the built-in oscillator with two test frequencies, the very versatile bias adjust, the ease of interchanging head assemblies and the high-quality sync option. As is true of all tape recorder testing, it is impossible to speak for the long-term reliability of the product, although the entire system appears to be carefully designed and well constructed. In all other respects, we feel the RT-2022 or RT-2044 would serve nicely in a small recording studio as well as in a "super-system" of a dedicated audiophile-recordist.

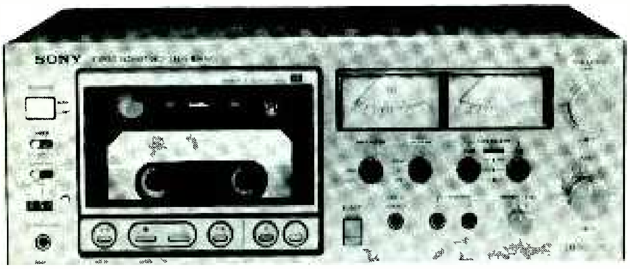
JT-2022T:
2-channel, 2-track head assembly unit.

JT-2044T:
4-channel, 4-track head assembly unit.

PIONEER RT-2044 TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Record/playback frequency response	
15 ips	±3 dB, 35 Hz to 36 kHz (at 0 VU)
7½ ips	±3 dB, 25 Hz to 23 kHz (at −10 VU)
Harmonic distortion: 15 ips	0.6% at 0 VU; 0.65% at +3 VU
7½ ips	0.65% at 0 VU; 0.7% at +3 VU
Recording level for max 3% THD	+12 dB at 15 ips; +13 dB at 7½ ips
Sync head p/b response	−3 dB at 10 kHz at 15 ips
Best S/N ratio, std tape	59 dB on RT-2044; 64 dB on RT-2022
Input sensitivity, mic	0.10 mV
line	38 mV
Output level, line	480 mV
headphone	68 mV/8 ohms
Bias frequency	125 kHz
Speed accuracy	±0.5%
Wow and flutter: 15 ips	0.025% WRMS; 0.04% unweighted
7½ ips	0.025% WRMS; 0.06% unweighted

Sony Model EL-5 Elcaset Recorder



General Description: The Elcaset is a new tape format of which the Sony model EL-5 is the first actual unit made available to MR for evaluation (and so far as we know the first of the format to be released by any manufacturer, at least in the U.S.).

As the name suggests, it is a variation of the cassette idea but with several important changes that make it a separate format, utterly incompatible with existing tape formats. The Elcaset itself is a plastic housing $5\frac{5}{16}$ inches by $4\frac{3}{16}$ inches and 5/8-inch in thickness. These are very close approximations in inches of the dimensions which of course are specified in metric terms, but they do indicate that the Elcaset is substantially larger than the conventional (Philips type) cassette. The tape housed in this packet is 1/4-inch wide and it runs at $3\frac{3}{4}$ ips speed (both quantities being, of course, double those of standard cassettes). The basic track configuration is quarter-track (or "four track") 2-channel stereo in both directions of tape movement. That is to say, the cassette may be removed from the machine and re-inserted for use of "side 2." The tape runs between two hubs within the Elcaset housing but it is under greater control by the deck transport since there is no guide-pin inside the packet as in conventional cassettes. The Elcaset is fitted with built-in reel locks that prevent the tape from loosening when the packet is out of the machine but which are released automatically when inserted into the recorder. Also found on the Elcaset housing are left and right protective lids that are opened automatically when the Elcaset is in place and the record or play controls are activated.

It appears there will be three kinds of tape for the Elcaset format—designated as Types I, II and III. Type I is "standard low noise." Type II is ferrichrome (FeCr). Type III has not yet been released or identified other than for a statement that it is "designed especially for the Elcaset format."

Two kinds of machines have been announced for handling the Elcaset, one lower-priced and less sophisticated; the other, costlier and embodying more advanced features (such as three separate heads). The EL-5 is the former model, a two-head recorder. A front-loader, it is designed for vertical installation, with all operating controls and features arranged on the front panel. To the left of the Elcaset compartment is the power off/on switch, a switch for use with an optional external timer, a memory-counter option, a three-digit tape counter and reset button and a stereo headphone output jack. Grouped beneath the Elcaset compartment are the transport buttons for rewind, stop, play, fast forward, record and pause. These are lit during use, respond to "feather-touch" pressure and operate through a logic-controlled solenoid system to permit mode and direction changes without the need to first press the stop button. The pause control may be used to prepare for recording, and it is possible to go from playback to recording directly (punch-in recording) by holding down the playback button while depressing the record button.

The two VU meters to the right of the Elcaset compartments are calibrated from −20 to +5. Below them are switches for mpx filter (for recording FM from the air), the Dolby system (including the FM/Dolby copy option) and bias and EQ. These have three positions each, designated by the tape types mentioned earlier.

Below these controls are the Elcaset eject button, a line-in jack (stereo standard phone-plug type), microphone-in jacks (standard) and a headphone level control. At the extreme right of the panel are the recording input controls: a dual-concentric pair for line, and a similar pair for mic. Each stereo channel is separately adjustable, and input mixing of the two sources is possible.

The rear of the EL-5 contains stereo pairs of phono jacks for line output and line input, the latter being pre-empted by the front-panel line input if both happen to be connected at the same time. In addition the rear contains an output level control that may be manipulated by finger or by screwdriver and which controls both output channels simultaneously. There also are left- and right-channel FM calibration adjustments for use when recording Dolby-encoded FM broadcasts; a socket for connecting an optional remote-control accessory; two AC convenience outlets, one switched; and the recorder's AC line cord.

Test Results: To evaluate the EL-5, *Modern Recording* subjected it to the usual tests, conducted critical tests twice (one each for the two kinds of tape recommended). In general, the Sony EL-5 met or exceeded published specifications. The only discrepancies noted were relatively minor—we measured line input sensitivity as 65 millivolts instead of the 95 mV claimed, and line output level as 700 mV instead of the 775 mV claimed.

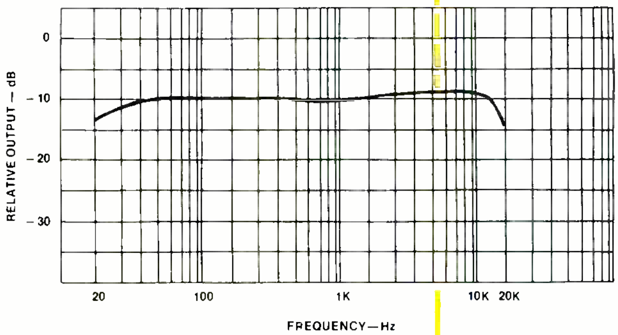
Far more important were the unit's responses, taken at a −10 dB recording level for both Type I and Type II tapes. These figures indicate the recorder's ample and linear frequency range as well as the correctness of the switch positions for the two kinds of tape. S/N, with either tape, remained better than 60 dB; distortion with either tape was lower than the 0.8% specified. With either tape, ample recording headroom is available without exceeding the 3% THD level; there's a +8 margin for Type I tape, and a +9 margin for Type II tape.

Mechanically, the EL-5 ran as claimed—smoothly, quietly, and responsively. All controls functioned as they were intended to, and it was felt that the unit presented no problems in terms of handling and operating it, even for a relative newcomer to the tape format. The owner's manual was judged to be well-presented, although some of the explanations might be more explicit and some of the language smoother.

General Info: Supplied in metal case with front dress-panel. Supplied with demo Elcaset (Type II, LC-30 size, 15 minutes per side), and blank Elcaset (Type I, LC-60 size, 30 minutes per side); small tool to take up slack in tape; two pairs of signal cables. Dimensions are 17 inches wide; 6¾ inches high; 12⅝ inches deep. Weight is 23 lbs., 2 oz. Suggested retail price is $629.95.

Individual Comment by N.E.: The EL-5 performs as well as, or perhaps a bit better than, its specifications claim. This bespeaks good design-engineering and careful manufacturing, as well as honest published ratings. But, like many others, I am left wondering what exactly does this product offer that cannot already be obtained from among the top-ranking standard cassette models? Perhaps the answer will come later, with the costlier EL-7 recorder (which employs three heads, three motors and some added refinements not found on the EL-5), and with the "Type III" tape that is reportedly soon to be made available. In the meantime, the EL-5 stands as a product of undeniable virtue but one with a less-than-clear marketing goal, in terms of the various segments of the tape recording public.

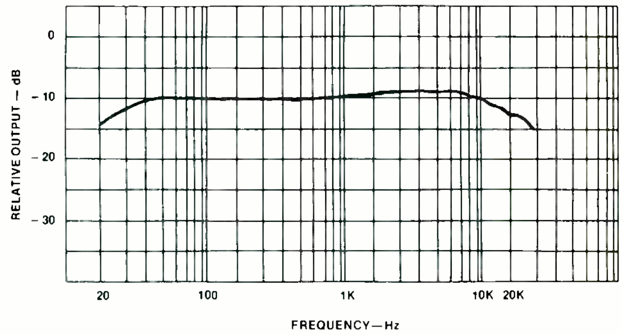
I must comment, by the way, on the demo tape supplied. I feel Sony could produce something better. Some of it sounded peaky in the highs, and I wasn't terribly intrigued by the musical content. The mixed-bag arrangement of the "2001" theme from *Also Sprach Zarathustra* really annoyed me (I wonder how that arrangement will go down among classical music buffs, and especially Europeans). I sort of wiped that listening experience out, though, by recording a string quartet from the air, using Type I Elcaset tape and the indicated EQ and bias settings on the deck. It sounded, on playback, indistinguishable from the broadcast. So one can make musically valid tapes with the EL-5.



Sony EL-5: Record/play response using Type I tape (low-noise, ferric oxide, high output).

One final word: I find the single output level control at the rear less than convenient.

Individual Comment by LF: I must confess that after having been all primed and excited for the introduction of this new tape format by the various press releases which have covered this new development from Sony, Teac and Technics by Panasonic, I was somewhat disappointed when finally confronted by this first Elcaset deck. Not that it failed to meet any of its specifications. Quite the contrary, frequency response, signal to noise and even harmonic distortion were all somewhat better than claimed. What bothered me was that in this admittedly "low end" Elcaset unit (much higher-priced models are already planned or in production from Sony and other firms), many of the features which had excited my curiosity over Elcasets in the first place were absent. The Elcaset package is more easily adapted to three-tape head configurations, but the EL-5 is a two headed machine, with no tape monitoring facilities. The Elcaset package has sensing notches by which a machine can detect the type of tape being used (contrary to the situation with standard cassettes, where the formulations are almost numberless, the originators of the Elcaset standardized on three standard types, numbered I, II and III), but the EL-5 machine has the usual bias and EQ switches which must be set to their proper positions with each type of tape used. The Elcaset format makes provision for two narrow control tracks besides the four audio tracks, but the EL-5 provides no access to this area of the tape cross-section for application (even by external electronics) of trigger or synchronizing signals, cueing signals, or whatever other control signals one might want to apply to a given tape.



Sony EL-5: Record/play response using Type II tape (ferrichrome type).

We fully understand that all these extra features can and will be made available in future, more expensive Elcaset machines, but that leads us to the basic questions concerning this EL-5. Why did Sony bother to produce the unit in the first place? How does it show up the Elcaset format to best advantage? Why would anyone buy an EL-5 when, for very little more money, they could purchase, say, a three headed standard cassette machine from Nakamichi or Tandberg, each of which would provide as good frequncy response. Admittedly, the signal-to-noise ratios obtained with the two types of tape tested were a few dB better than can be obtained from standard cassette tape, and the headroom or dynamic range beats out the ordinary cassette by a couple of dB as well, but is that sufficient justification for the Elcaset's existence—or for the existence of the EL-5? We tend to think not, especially since the Elcaset was intended to "bridge the gap" between "better cassette decks and open-reel machines."

As far as wow and flutter is concerned, I have in recent weeks measured standard cassette units which have as low w&f figures as those measured on the EL-5. I have also experimented with standard cassette units whose transport systems are fully as automated and logic actuated as the EL-5's. In short, judging by this first Elcaset deck, I get the feeling that the format has very little to justify it. Admittedly, my opinions may change if and when I see Elcaset decks that really take fullest advantage of the new tape format, but as matters stand now, I prefer to use a good cassette deck for less serious recording work and a decent open-reel machine for my more serious recording efforts. While it is possible to edit Elcasets more easily than standard cassettes, I was not able to do any serious and precise editing because of the way in which the tape is handled by the Elcaset mechanism. There is no way (nor is there likely to be) in which one can "cue up" a point on the tape to be edited and then find that precise point once the Elcaset is ejected from the machine in which it is functioning. I certainly don't intend this to be my final statement on the Elcaset tape format, but as far as the EL-5 is concerned, I am not about to trade in my other recording equipment for it.

SONY EL-5 ELCASET RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response, Type I tape	±3 dB, 22 Hz to 19 kHz
Type II tape	±3 dB, 22 Hz to 21 kHz
S/N ratio, Type I tape	61 dB
Type II tape	62.5 dB
THD at 0 VU record level, Type I tape	0.7%
Type II tape	0.65%
Recording level for maximum 3% THD,	
Type I tape	+8 dB
Type II tape	+9 dB
Mic input sensitivity	0.3 mV
Line input sensitivity	65 mV
Line output level	700 mV
Headphone output level (8 ohms)	90 mV
Wow and flutter (WRMS)	0.045%
Fast-wind time (LC-60)	75 seconds
Bias frequency	160 kHz

CIRCLE 2 ON READER SERVICE CARD

Lux Model PD-121 Turntable

General Description: Lux Audio (whose products also are known by the term Luxman) recently introduced the model PD-121, a two-speed (33 and 45 rpm) manual turntable for use with a tone-arm of the buyer's choice. The Lux turntable is supplied less an arm but with a unique arm-mounting platform that goes onto the unit's baseplate by means of a bayonet-fit arrangement. This particular piece of hardware is carefully engineered to assist the owner in mounting the arm correctly, with due attention to proper geometry and operating angles. The platform also has an attachable arm-rest. The owner can, if desired, keep more than one such platform on hand with specific tone arms pre-mounted and ready for relatively quick interchanging for experimental or other special applications. No drilling of holes in the turntable base is required for either the tone-arm or the arm-rest.

The turntable itself is a direct-drive type, powered by a brushless DC servo-controlled motor. The platter is die-cast aluminum weighing 5.4 pounds and is fitted with a fairly thick cover that has a rubberized bottom and a top surface of a smooth-nap firm fabric. It measures 11⅞ inches in diameter. The entire turntable assembly is supported on an aluminum die-cast chassis which is integrated with the unit's housing, itself a handsome rosewood-panel surround. The unit sits on four adjustable shock-mount "feet." Supplied with the PD-121 is a cover, made of extra-heavy gauge clear plastic, which may be fitted into hinges at the rear that provide a preset "stop" for raising the cover. The fit is excellent, and the cover may be left over the turntable during play.

A completely manual single-play design, the PD-121 offers no automated features. Its controls consist of an off/on power switch, the 33/45 rpm speed selector, and two fine-speed adjustments (one for each speed). These are located just under the front edge of the unit. On the base is a strobe readout of speed, which shows the actual numbers (33 and 45) for the speed selected at either 50 Hz or 60 Hz power-line frequencies.

Test Results: With its extremely quiet, shock-resistant, and smooth rotation, the Lux PD-121 is, in the view of MR's testing staff, a splendid example of a superior turntable that could be of interest to both the professional and the amateur audio perfectionist. S/N (rumble) came in a shade better than specified (72 dB down as compared to the rated 70 dB down). Similarly, wow and flutter—measured by MR as 0.025%—was better than the 0.03% claimed. Needless to say, these are "state of the art" figures and of course are the most critical factors from the standpoint of pure turntable performance. The speed adjust range, specified as ±4%, was measured in our lab as +3%, −4%. In MR's view the 1% discrepancy for the fast-adjust range is negligible. Start-up torque is excellent, with the platter coming up to nominal speed in about half-rotation of the platter.

The bayonet-fit mounting disc for tone-arms can be ordered precut to fit many of today's popular separate arms such as the Shure SME, Grace, Denon, Fidelity Research, Ortofon or Stax. As supplied, the PD-121 includes one platform precut for the SME, plus an uncut platform for any other arm. The material of the platform (aluminum bonded to hard plastic) is rather difficult to work with, however, and so if the arm the buyer decides to use is one that fits a precut platform-mount, MR advises buying it (cost is an extra $20).

General Info: Dimensions are $18\frac{5}{8}$ inches wide; $14\frac{11}{16}$ inches deep; $5\frac{11}{16}$ inches high (including feet but without the cover). With cover in place, height is $6\frac{3}{16}$ inches; maximum height for cover up is $16\frac{3}{4}$ inches. Weight is 29 lbs. Advertised retail price is $495; additional bayonet mounts, $20 each.

Individual Comment by N.E.: By the basic criteria which I myself have described in an earlier issue of *Modern Recording* ("What Is A Professional Turntable?," Oct/Nov 1976, page 29), the Lux PD-121 surely is just that. Rumble is about the lowest I have yet encountered and is utterly inaudible even through a high-powered playback system going at full tilt. The wow and flutter we measured are similarly very low and inaudible. Speed accuracy is assured of course by the sophisticated and effective built-in fine-speed adjustment and its associated readout strobe indicator. Speed also tends to "hold" once the adjustment is made on a given day or time of day. Quick cueing is facilitated by the fast start-up time and high torque of the platter. The PD-121 also appears to be quite immune to external shock and jarring effects.

Which brings me to my final thought here on the price of the PD-121 which, in view of the fact that one still has to buy an arm to go with it, may cause some raised eyebrows. On this point I can offer no meaningful comment in terms of whether "it is worth it or not." At over $500 (if you include the cost of an arm), the PD-121 is one of the higher-priced models around, but not the highest-priced. I do have a feeling though that if you can afford it, you are going to be very tempted to get it once you inspect and try it.

Individual Comment by L.F.: It has been said that all a good turntable has to do is rotate, unerringly and unwaveringly, at correct speed and with as little vibration or noise as is possible. The LUX PD-121 comes as close to that definition as I have ever seen. The test-result numbers speak for themselves, but they cannot convey the feeling of elegance and stability which this rugged (and handsome) turntable provides. The solid, die-cast chassis that forms the foundation for the device is as heavy as any I know of, and its reinforcing ribs and fins seem to have been carefully calculated to provide the ultimate in stability and rigidity. This structure and the four insulating "feet" below the base help reduce to an all-time low the possibility of external vibration from reaching the pickup/arm combination used. The insulators are of a triple-structure design, and the spring element in each "foot" is damped by rubber while the whole mechanism is suspended in silicon-grease. The sturdy dust-cover (0.157-inch thick) is perfectly hinged to the base and is able to remain partially open without slamming down.

In addition to its excellent performance, the Lux PD-121 is visually appealing with its low-profile dimensions, fine wooden panels, and general refinement of finish and detailing. It is an exceptionally fine product.

LUX PD-121 TURNTABLE: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Turntable rumble (DIN "B" weighted)	−72 dB
Wow and flutter (WRMS)	0.025%
Speed adjust range	+3%, −4%
Start-up to full speed	1/2 rotation
Speeds available	$33\frac{1}{3}$ and 45 rpm
Motor and drive system	D.C. brushless motor; direct drive
Platter type and weight	Die-cast aluminum; 5.4 lbs.
Power consumption	6 watts (20 watts, start-up)
Speed indicator	Direct numerical readout

CIRCLE 13 ON READER SERVICE CARD

GROOVE VIEWS

Reviewed by:
SEDGWICK CLARK
CHAS FARRELL-KIMBRELL
NAT HENTOFF
JOE KLEE
RUSSELL SHAW

POPULAR

RACING CARS: ***Downtown Tonight.*** [Bill Price and Racing Cars, producers; Bill Price, Rick Stokes, engineers; recorded at Wessex Studios, London, England.] Chrysalis CHR 1099.

Performance: **Smooth and flowing**
Recording: **Unmasks their abilities**

You know it the time you see the prodigy win the science fair, or watch the young tyke tear up the Little League. They reflect potential, that elusive quality; an attribute which Racing Cars possesses in droves.

When looking for this positive trait in new recording acts, one demands versatility, uncommon skill on instruments, and a smooth sense of segue between various musical personalities. Yet most often, potential is best unmasked by a basically abstract quality—that of originality.

Racing Cars, a British quartet, forces a trip to the Thesaurus to look for both adjectives and superlatives. With a lead singer sounding a bit like a mellow Paul Rodgers, and with three-part harmonies resembling the Eagles, the new band is a curious hybrid of intricate melodic constructs coupled with lyrical twists and turns which indicate a wide-ranging palette of allegorical and poetic references.

A non-hostile, but somewhat cynical stoicism pervades most of their work. "Hard Working Woman," a ballad, describes a one-night fling and how the heroine had to leave the next morning right in the middle of hanky-panky to go to work. "Pass The Bottle" also deals with the sphere of alienation.

Cinching the plaudits of Racing Cars being one of the best new bands in some time is the superb collage of instrumentation present. The lead guitars of Graham Williams and Ray Ennis constantly weave in and out with delicious parallel lines, contrapuntal harmonies, and generally fruitful interchanges reminiscent of some of the better dual lead workings in rock. "Moonshine Fandango" is a good example of this. The chording in the left speaker of Williams is nicely shadowed by the energetic lead of Ennis coming from the right.

Indeed, save for the unfortunate overmix of Robert Wilding's rather pedestrian drumming on "Calling the Tune," the blend of sounds is most perfect. Material recorded at Wessex has always been known for uncommon clarity. Here, the ability of several percussion instruments to be individually recognized on the instrumental "Four Wheel Drive" attests to that. In addi-

RACING CARS: First place winners

tion, the complex interworkings of the guitarists are captured with crystal clarity.

It's always a joy to watch a new band flash images of embryonic brilliance. With the wealth of creativity and technique available to Racing Cars, further growth is to be expected.

R.S.

GENE CLARK: *Two Sides to Every Story.* [Thomas Jefferson Kaye, Gary Legon, producers; Joel Soifer, Roger Nichols, engineers; recorded at Fidelity Recording Studios, Los Angeles, Ca.; remixed at ABC Studios, Hollywood, Ca.] RSO TS 1-3011.

Performance: **Eloquent**
Recording: **Flawless**

Gene Clark's conceptions are getting more poetic with each album. The imagery of such tunes as "Kansas City Southern," with its discussion of freight trains as a symbolism of home has been attempted before. But there is something in the vocal delivery, important contributions by a variety of pickers, and the album's overall sincerity that turns traditional themes as these into newly delivered messages.

GENE CLARK: Evocative and memorable tunes

Possessed with vocal equipment that is at once soft and biting, Clark delivers most of the tunes on this disc with unmitigated conviction. Each of the songs is populated by events or personages familiar to us all; "Silent Crusade" evokes the notion of life as an ocean voyage; "Sister Moon" is sung to a lady leaving her man due to a wish for freedom.

All throughout, the exceptionally fluid lead lines of ace session picker Jerry McGee add a healthy dose of spunk to the proceedings. "Marylou," a traditional blues revived by Clark, gets an exceptionally lively treatment while "Lonely Saturday," a mournful ballad, is treated with lines of winsome twang.

The production of Kaye and Legon is quite equal to the task. The instruments are mixed perfectly. The contributions of guest vocalist Emmylou Harris are stressed but not given dominance, and the more morose tunes are better for their instrumental sparseness rather than layers of soapy strings which other producers would be tempted to add.

No great unexplored vistas are charted here, but for evocative and memorable tunes done with both taste and grace, *Two Sides To Every Story* is highly recommended. R.S.

LEON REDBONE: *Double Time.* [Joel Dorn, producer; Bob Lifton, Vince McGarry, Neil Brody, engineers; recorded at Regent Sound Studios, New York, N.Y. and Village Recorder, Los Angeles, Ca.] Warner Bros. BS 2971.

Performance: **Purposely uninspired**
Recording: **Ditto**

Leon Redbone is a figure who is impossible to classify; only broad parallels exist between him and anyone else working today. What other contemporary singer chooses to work almost exclusively with a bunch of ancient tunes, culled from the country, blues, and cinematic themes of fifty years ago?

The wide choice of material is, in itself, quite compelling. In their time, movie songs such as the "Sheik of Araby," the Dixieland compositions of Jelly Roll Morton, and the "Mississippi River Blues" of Jimmie Rodgers had audiences as disparate as any three you would want to find.

In transposing this material to the modern disc, Redbone is singularly sparse. These selections contain precious little amplification; main instrumentation is Redbone's seemingly disinterested, apathetic guitar which is lazily strummed with only interceding darts of brass instruments called upon for occasional embellishment. Over this sloth-like pace presides Leon's voice, which in fact is a combination of a scat, a borderline emphysema case, and a gargle. At times, *actual* words protrude from behind the affected slur.

Recording such an uncharacteristically ancient minstrel mode, requires production techniques that contain a

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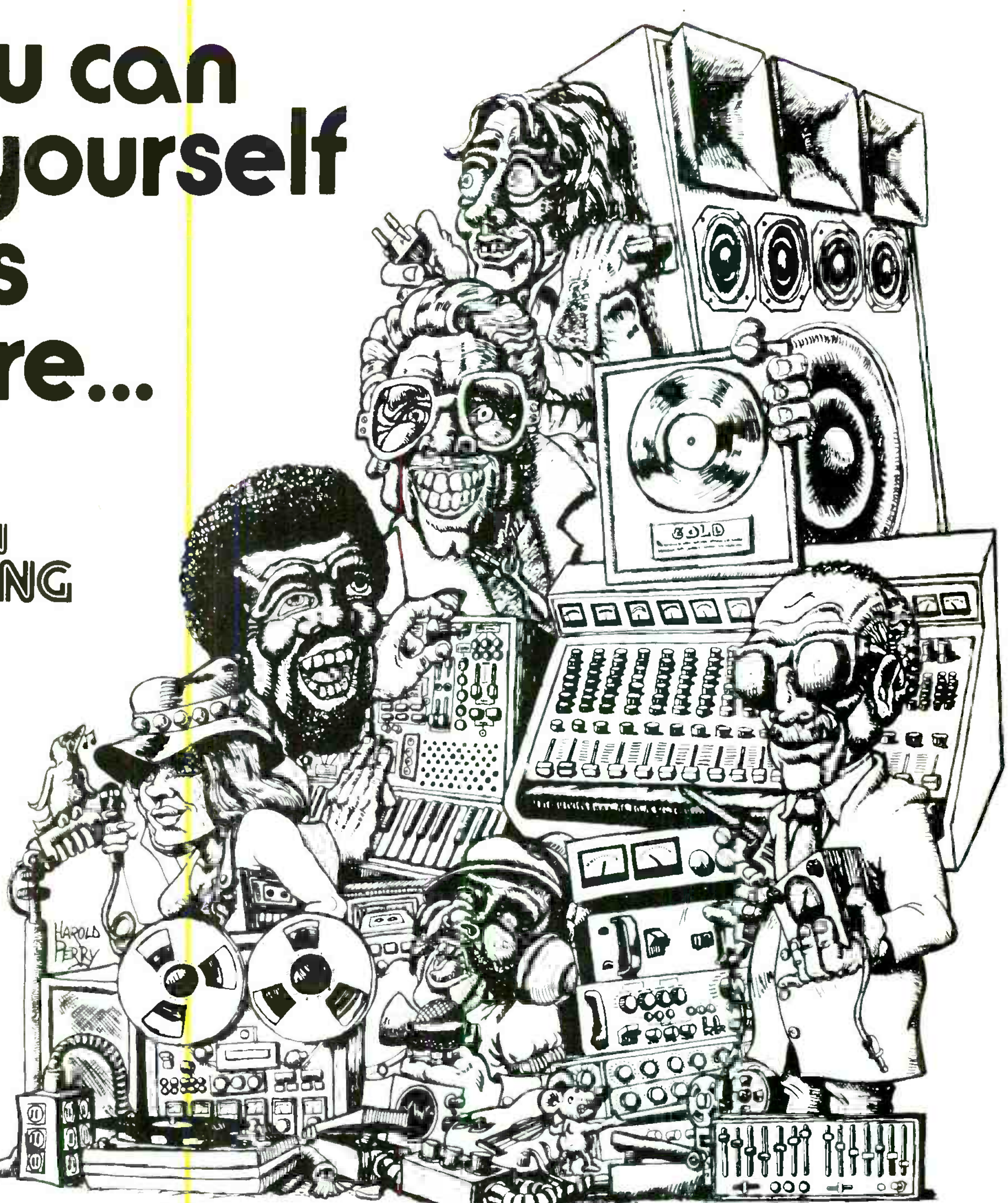

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M-12

A Classical Modernist and the Newest Frontiersmen

By Nat Hentoff

If I were a jazz musician with free choice as to where to record, I'd book lots of time at Contemporary Records' studio in Los Angeles—provided Lester Koenig, the boss, produced the date. Through the years, Koenig and his engineers have established what I consider the optimum criteria for recorded jazz sound. Included is an immediacy, a fulness of *presence*. Not doctored presence by which technology so envelops the musicians that it is as if they had two souls—one their own and the other one fabricated in the control room. Koenig works the other way, putting his highly sophisticated equipment wholly at the service of the players so that their own authentic, hard-won sound comes through with naturally vivid clarity. Also, the resilient interplay that is the essence of collective improvisation is projected with as close to perfect balance—as if the listener were *in* the band—as any studio has ever attained.

A new illustration of Koenig's truth-in-recording is Art Farmer's *on the road.* Farmer, once a trumpet player and now a most mellow (but often dramatic) flügelhornist, has been based in Vienna since 1968 after a diversified, distinguished American career which brought him much acclaim among other musicians but little bread and security. Farmer is returning to the States more often now because there has come to be a growing audience for classical jazz modernists. Farmer long ago distilled the jazz advances of the '40s and '50s into a deeply personal style which, in recent years, has become ever more flexible and multi-colored.

In *on the road*, Farmer is stimulatingly complemented by alto saxophonist Art Pepper, pianist Hampton Hawes, bassist Ray Brown (himself the very model of a classical modernist) and alternating drummers Steve Ellington and Shelly Manne. The playing by all is lyrical; incisive; and on some tracks (especially Duke Ellington's "What Am I Here For?") possesses a sustained quality of collective invention that will lead to this set being played decades from now. This is jazz maturity, captured whole and intact by Koenig and his associates who ought to run master classes for other jazz engineers and a&r men.

From time to time, in the months ahead, I shall be exploring in some detail here what the post-classicists, the newest frontiersmen of jazz (or black music) are doing. Meanwhile, one of the best single introductions to these largely uncharted sounds is the two volume *In-Sanity* (Black Saint) by a startling array of musicians on the ascendant—all of them grouped here under the rubric, The 360 Degree Music Experience. Drummer Beaver Harris and keyboardist Dave Burrell direct this co-op musical unit, and among the other players is unquestionably the most original and powerful baritone saxophonist in this new jazz—Hamiet Bluiett.

Throughout, the dense, often explosive horn textures laid over steaming, complex rhythms encompass a wide variety of moods and nascent forms—with oases of contemplative lyricism. I wish someone like Les Koenig had engineered this because the balance is sometimes askew and there could be more totality of presence. Nonetheless, put up the volume and plunge right in. When the music starts getting to you, you'll not be thinking about the engineering.

ART FARMER: *on the road.* [Lester Koenig and John Koenig, producers; engineer not named; recorded at Contemporary Records studio, Los Angeles, Ca.] Contemporary S7636.

BEAVER HARRIS, DAVE BURRELL/ THE 360 DEGREE MUSIC EXPERIENCE: *In-Sanity.* [Giacomo Pellicciotti, producer; Timothy Marquand, engineer; recorded at Generation Sound Studios, New York City]. Black Saint BSR 8006/7 (distributed by New Music Distribution Service, 6 West 95th Street, N.Y., N.Y. 10025).

MODERN RECORDING's PRO SHOP

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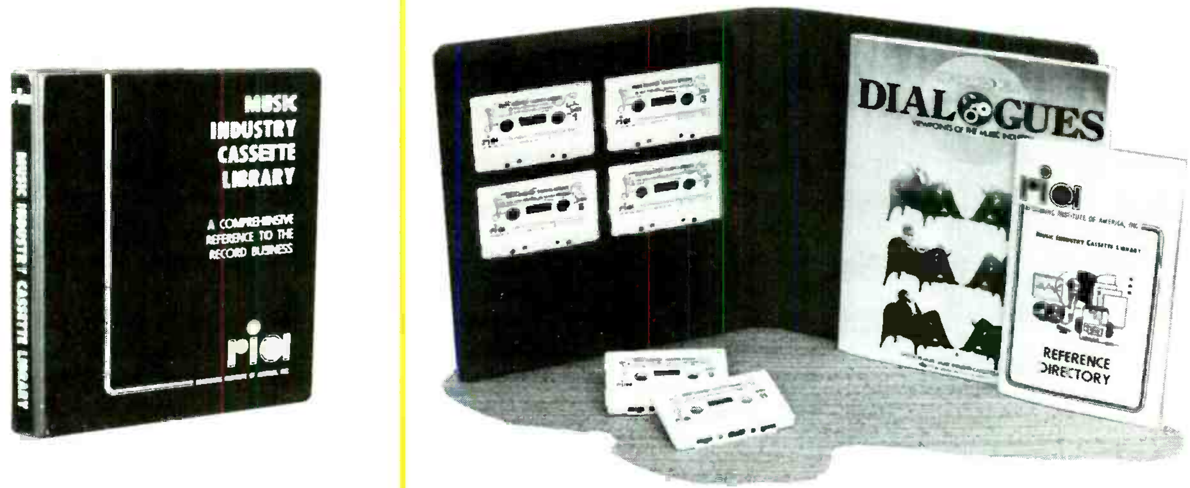

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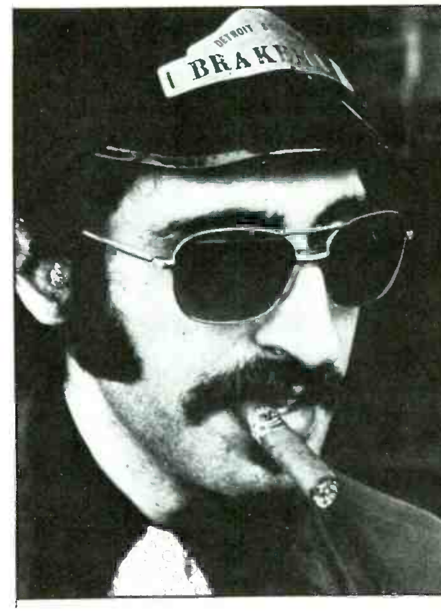
minimum of frills. Producer Joel Dorn and his crew of engineers have once again done just that. The mix is not especially good; at least by modern definition. The simple guitar licks dominate the mumbling vocals, and when trombone and trumpet are employed, Redbone's intonations seem to be completely buried. Yet, given



LEON REDBONE: Unique and authentic.

Dorn's track record, these "shortcomings" are not to be taken as evidence of sloppiness. Technical hands obviously went into the sessions with a mind to make them as "authentic" as possible.

By virtue of his sheer uniqueness, Redbone is an artist for whom all customary rules of auditory decorum should be waived. Therefore, the album's many technical flaws should be regarded in that light. R.S.

JAZZ

LOUIS ARMSTRONG: *Young Louis Armstrong 1932-1933.* [Frank Driggs, producer; Don Miller, remastering engineer; originally recorded at RCA Studios in Camden, New Jersey and Chicago, Illinois between 12/8/32 and 4/26/33.] RCA Bluebird AXM 2 5519.

Performance: **Classy classics**
Recording: **Not bad for thirties mono**

Let's begin with one fact in mind. From April 5, 1923 when he recorded his first sides with King Oliver's Creole Jazz Band until his unfortunate passing in July of 1971, Louis Armstrong made enough records to fill a discography with more than a thousand titles of beautiful joyful music. That's quite a legacy to leave behind. These recordings were made between December of 1932 and April of 1933, sandwiched between Louis' jaunts to Great Britain and the Continent. They include Louis playing wih Chick Webb's band and Charlie Gaines' band recorded in Camden, New Jersey in December of 1932 and Louis playing with a band put together for him in Chicago in 1933; a band that included Teddy Wilson on piano and Sid Catlett on drums. There's not a bad side among them. Armstrong could dignify a pop hit like "Hustlin' and Bustlin' for Baby" or clown his way through hokum like "Laughin' Louis." Of course, there were always the good old ones from New Orleans such as "High Society" and "Mahogany Hall Stomp." So here are the complete Louis Armstrong 1932-1933 Victors, including alternate masters of "That's My Home" and "I Hate To Leave You Now."

The transfers are clean and clear, most of them being made directly from vinyl pressings from the original metal parts. This made the desperate search for collectors whose 78 rpm copies showed the least amount of wear and tear unnecessary.

LOUIS ARMSTRONG: Re-released clear and strong.

Rudi Blesh's authoritative and complete liner notes would have been even better if Rudi had refrained from quoting the song lyrics. They sound better than they read, especially when Louis sings them. —J.K.

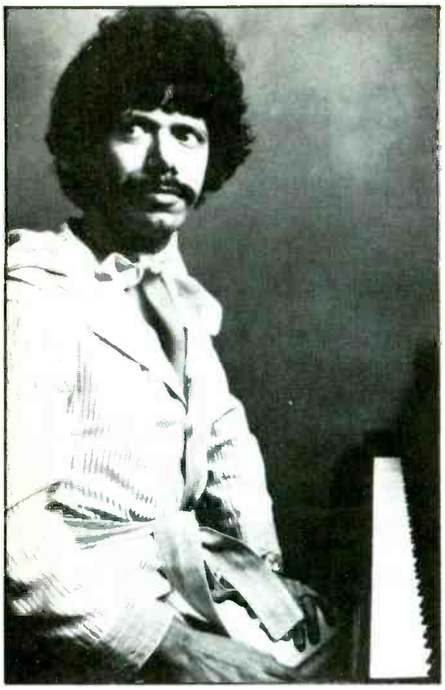
CHICK COREA: *My Spanish Heart.* [Chick Corea, producer; Bernie Kirsh, engineer; recorded at Kendun Recorders, Burbank, Ca.] Polydor PD 2 9003.

Performance: **Exquisite yet redundant**
Recording: **Purposely mystical**

On *My Spanish Heart*, four whole sides have been devoted to a thorough exploration of Chick Corea's Latinesque musical preferences, a leaning evident throughout his entire career from his days as a pianist for salsa bands to his early seventies work with Airto and Flora Purim.

As opposed to the rhythmic drive of his early exercises in the idiom, the vogue on *My Spanish Heart* seems to be a concentration of melodic, flamenco-type narratives with occasional wafts of trumpets creating torreador imagery. In fact Chick is photographed in such clothing on the outside album jacket.

Taking this package as a whole, the results have to be mixed. The initial impression is that of a moody, restless piece that segues from the sensuous and flowing acoustic ivory of "Love Castle" and "My Spanish Heart" to the percussion-infused stomping of "Day Danse"



CHICK COREA: Some cutesy ticklings.

and "Armando's Rhumba." However it all bogs down by the third side with nearly ten minutes of Moog doodling on a three-part work called "El Bozo." There are some cutesy ticklings in the first movement, but only obscurities throughout the rest.

The calling card of this record, "Spanish Fantasy" (Parts I-IV) occupies another side all by its lonesome. As with the other material, it definitely has its moments. Some are inspired plucking by Stanley Clarke, and several bars of a frantic beat chase by peerless session drummer Steve Gadd. Yet as time wears on, the central idea becomes just a bit overdrawn, and we begin to hear the same, reduntant "flamencoisms" that dominate the disc.

On a technical level, *My Spanish Heart* manages to impress even when the imagination runs aground. Corea and Clarke are always worthy of praise, and Stanley's all-too-rare bass bowing on "The Gardens" adds a touch of the surreal. Yet even here, there are flaws. A four-piece brass section added on several of the tracks, sounds as if they were recorded in the next county. Muted to inaudibility, they sound like kazoos, or more seriously, Craig Anderton's synthesizer-trumpet sounds like those early seventies Linda Cohen albums.

Coupled with the occasional drum overmix, the overall judgement is one of an LP that should have been only one album. It might have done better without the meandering filler which tragically trivializes the otherwise compelling and creative material. R.S.

CLASSICAL

SHOSTAKOVICH: *String Quartets Nos. 7, 13 & 14. Nos. 8 & 15.* Fitzwilliam String Quartet. [Peter Wadland, producer; John Dunkerley, engineer.] L'Oiseau-Lyre DSLO 9 and DSLO 11.

Performances: **Admirably sustained**
Recordings: **Superb**

The late Shostakovich quartets, like his *Fourteenth Symphony* and Mahler's last two symphonies and *Das Lied von der Erde* are the works of a man awaiting death. Broadly paced and introspective, the music aches with irony and quiet intensity. The frivolity and sardonic humor of the earlier works has soured, replaced by acid cynicism and quivering expectation. Shostakovich's music was always, first and foremost, intensely human, but his obsession with death was hardly confined to his own mortality, as the sadness and despair of the autobiographical *Eighth Quartet* (1960) demonstrates. Dedicated to the memory of victims of war and fascism, the *Eighth Quartet* is representative of Shostakovich's art at its most profound.

That the Fitzwilliam is an accomplished group goes without saying, but one must register particular note that such young musicians have so admirably sustained the morbid, almost painfully concentrated inspirations of a composer who has, so to speak, seen beyond the pale.

These two discs are notable examples of string quartet recording. The musicians appear to be in a rather resonant hall, yet there is no loss of clarity and the three-dimensional perspective is excellent. Some U.S. producers mic so tightly that the individual players appear to be in separate rooms, which may produce clarity but little sense of interaction or ensemble—and certainly no intimacy. Also, close miking is rarely flattering to a string instrument, which is prone to harshness and tonal grittiness unless allowed space for the sound to expand. This British production team has judged well, giving the Fitzwilliam Quartet a recording of ample warmth and clarity. S.C.

PROKOFIEV: *Symphonies Nos. 1 & 7.* London Symphony Orchestra, Walter Weller cond. [Michael Woolcock, producer; James Lock, engineer; recorded April 1974 in Kingsway Hall, London.] London CS 6897.

Performance: **Excellent**
Recording: **London's best**

Prokofiev's first and last symphonies under a conductor who is little known in the U.S. may not send buyers running to the store, and that's unfortunate. Walter Weller was formerly concertmaster of the Vienna Philharmonic and leader of the Weller Quartet; to my knowledge, he has never appeared in the U.S. as a conductor. Already released on London are noteworthy recordings of Rachmaninoff's *First Symphony* (CS 6803) and Shostakovich's *First* and *Ninth Symphonies* (CS 6787), and this new disc is even more distinguished.

There are many good recordings of Prokofiev's popular "Classical" *Symphony No. 1;* but for wit, lightness and orchestral polish, one would be hard pressed to better this performance. (Decca/London's production is, after all, only one element in a successful record!) The *Symphony No. 7* is considered in some quarters a feeble retread of rejected tunes from the composer's ballet, *Romeo and Juliet.* Yet I find the second (the most balletic) and much of the final movements irresistible, and the symphony as a whole worth hearing. The Rozhdestvensky pairing of these works on Melodiya/Angel is let down by coarse playing and recording. Interpretively, Weller is equally as distinguished as the Russian conductor, and his superior recording quality makes the choice obvious.

This is Decca/London's engineering at its best. The orchestra is realistically balanced, with exemplary clarity of each instrumental choir and no exaggeration of individual solos through multi-miking. Unlike this company's recent Solti/Chicago recordings (under a different production team) in which the perspective and dynamic range are flattened out by claustrophobic miking and careless mixing (due to quadraphonic recording technique, I would guess), this disc presents a three-dimensional, front-to-back image of an orchestra, warmly recorded in a fine acoustical surrounding. Surfaces were okay. And it's a pleasure to note that London dutifully lists the names of the producer and engineer, and recording site and date on the album.

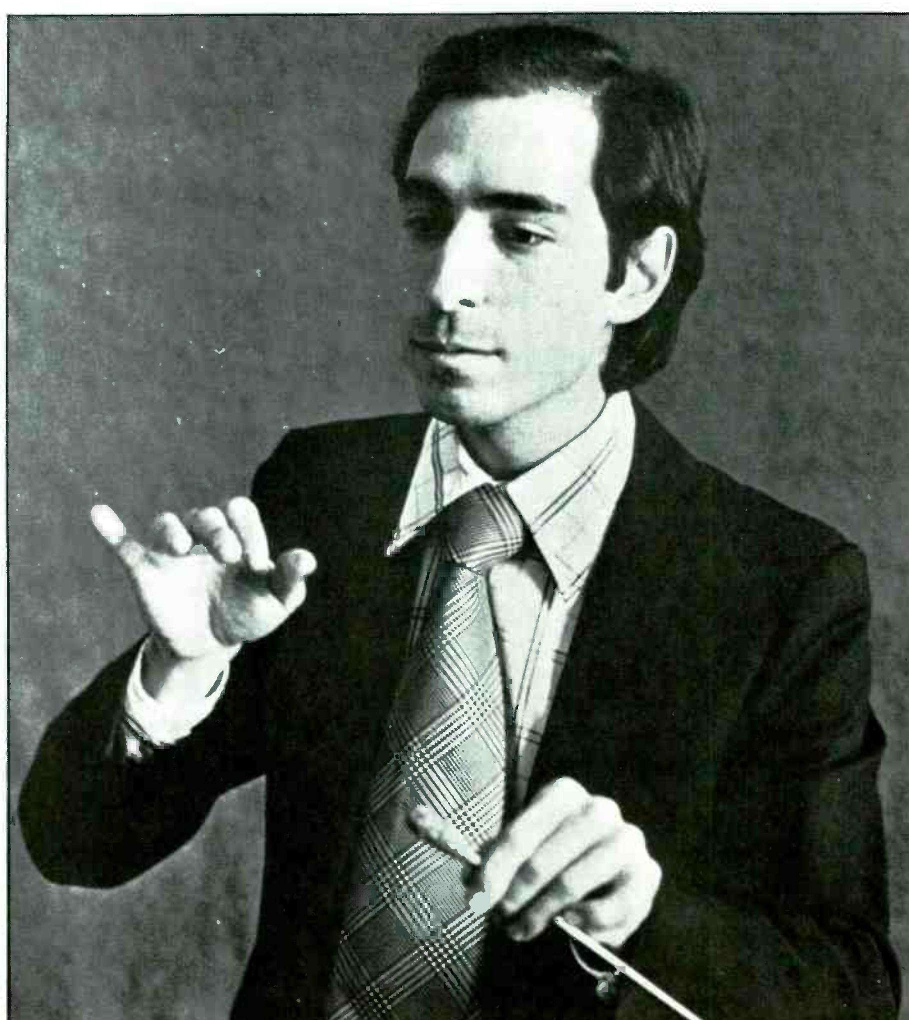
This disc, the first in a Decca/London Prokofiev cycle conducted by Weller, was released in England in May 1975. Since then, his recordings of the Fifth and Sixth have been issued there, and one hopes that the wait on our side of the Atlantic will not be so long. S.C.

SCHOENBERG: ***Serenade for Seven Instruments and Bass Voice, Op. 24.*** Kenneth Bell, bass; The Light Fantastic Players, Daniel Shulman cond. [Marc J. Aubort, Joanna Nickrenz (Elite Recordings, Inc.), engineering and musical supervision; recorded March 1976, New York, N.Y.] Nonesuch H-71331.

Performance: **Assured**
Recording: **Clean**

Nonesuch's director, Teresa Stern, is always careful in her a & r choices, so it comes as no surprise that this disc—only the fourth record of Schoenberg's music on this label in recent years—is also a winner from the vantage point of performance and recording. The earlier records, all necessary items for anyone interested in this repertoire, are: *Pierrot Lunaire, Op. 21*, performed by soprano Jan DeGaetani and the Contemporary Chamber Ensemble, with Arthur Weisberg conducting (H-71251); the complete piano music, in masterful performances by Paul Jacobs (H-71309); and Miss DeGaetani singing *The Book of the Hanging Gardens, Op. 15*, with Gilbert Kalish accompanying (H-71320, paired with Schubert songs, so technically that makes only three-and-a-half discs).

The *Serenade* is one of Schoenberg's most accessible works. Scored for clarinet, bass clarinet, mandolin, guitar, violin, viola and cello, this witty piece contains many parodistic and ironic jabs at compositional trends of the time (1923). The seven movements, containing dance, march and lied elements, are arranged around a central "Sonnett Nr. 217 von



DANIEL SHULMAN: A notable Schoenberg interpreter.

Petrarca" (sung here by Kenneth Bell), and span moods from the lyrical to the grotesque. The Light Fantastic Players, a group of young musicians formed in 1971 by Daniel Shulman who specialize in contemporary music, reveal a full understanding of the music and a secure technique in the face of Schoenberg's tricky rhythms—no small praise in lieu of the alternative versions. The late Bruno Maderna's recording with the Melos Ensemble on L'Oiseau-Lyre is consistently below the composer's metronome marks but his players savor detail more than on any other recording. Pierre Boulez's direction of the Domaine Musical Ensemble is particularly graceful and Everest's price is right, although Columbia really should get him into the studio for an up-to-date recording. Only the Nonesuch, by the way, prints the text and translation of the central "Sonnett" on the liner.

The crisp recorded sound on the new Nonesuch disc makes the most of the work's instrumental color, although occasional balances seem oddly vague. These appear to be conductorial decisions, rather than the engineering, but no one need hold back from purchasing this record. As usual, Nonesuch's liner notes (by Robert Erich Wolf, who contributed the particularly fine translation for this company's *Pierrot Lunaire*) are especially informative. Surfaces were so-so. S.C.

SHOWS and SOUNDTRACKS

DICK HYMAN: *Music From the Motion Picture, "Scott Joplin."* [Ettore Stratta, producer, Bob Simpson, engineer; recorded at RCA Studios, New York City.] MCA 2098.

Performance: **Cinematic but musical enough to stand on its own**
Recording: **Simple but honest**

As a jazz pianist and arranger, Dick Hyman is historically dedicated enough to begin with the original sources of the music. Yet he has too much creativity to just leave off where the pioneers stopped. His updates of the music of Jelly Roll Morton, James P. Johnson and Louis Armstrong stirred up no small amount of controversy among jazz fans and critics with regard to authenticity versus creativity. This made Dick Hyman the logical choice to compose the background music for the Universal film, "Scott Joplin" starring Billy Dee Williams and Art Carney. Hyman had previously recorded all of Joplin's piano music played straight from the published score on RCA Red Seal. Having proved he can play it as written, Dick Hyman is entitled to play around with the music and have some fun doing it his own way. What Dick Hyman has given us this time around, is a dramatic score to underline the plot of a film about Scott Joplin using themes from Joplin's music.

This is not simply an LP dubbed from a soundtrack. Some of the original segments are only fragments of a minute or so in the film version. Hyman has extracted from the film score some of the better and more dramatic excerpts. He has expanded them to fill a whole LP which was recorded in New York with a top notch studio orchestra made up of classical players such as flutist Julius Baker and jazz stars like drummer Bobby Rosengarden.

The record listener even has an advantage over the film viewer. In the film Dick Hyman overdubbed the second piano part on "Cutting Contest." On the record it becomes a real cutting contest between two excellent masters of jazz piano, Dick Hyman and Hank Jones.

—J.K.

LED ZEPPELIN: *The Song Remains the Same.* [Peter Grant, Jimmy Page, producers; Eddie Kramer, engineer; recorded "live" at Madison Square Garden, N.Y.; mastered by Lee Hulko at Sterling Sound, N.Y.] Swansong SS 2 201

Performance: **Tiring**
Recording: **Well done**

Upon first listening to Zeppelin's movie soundtrack album, I felt that the album title was incomplete. It seemed that the title should have read, "The Song Remains the Same for Too Long." This feeling was supported largely by the track "Dazed and Confused"—which takes up all of side two—with Page running out of things to say on his guitar long before he stops playing. As I continued to listen, I began to take into consideration that this is a soundtrack album for a movie, which, I'm sure, accounts for much of the dead space and the seeming self-indulgence of the players. With a band as visually exciting as Zeppelin, I'm sure that the pregnant pauses and

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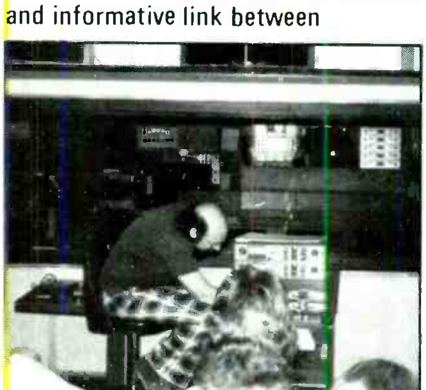

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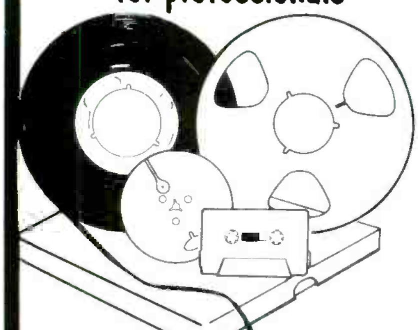

overplayed lead passages can be joyously overlooked in a "live" situation. However, without the visual properties, this recording can tend to be tiring.

The engineering is very clean with surprisingly great separation for a "live" recording of a band that plays

LED ZEPPELIN: The song is too long.

as loudly as Zeppelin. Of special note is Kramer's panning on Page's guitar during "Dazed and Confused." The bass does tend to rumble at times, which could as easily be Jones' fault as Kramer's since Jones plays a very full, rolling style on bass. John Bonham's drum solo, "Moby Dick" is especially well-recorded and a must for headphones. Each drum has it's own distinct place in the mix. Bonham is still, without a doubt, the best all-around *hard* rock drummer on the scene. One truly strong plus on the album is Robert Plant's vocals. His improvisations are interesting and his voice is in fine form. The vocal mix is easily comparable to Zeppelin's studio sound.

Overall, the worst problem this album suffers is that it *is* a soundtrack album, so there are lags. However, let it be known that during the moments when the band is kicking together on the body of a song, there is no questioning Zeppelin's right to claim the title as the world's top rock band. Perhaps another "live" album, recorded for the express purpose of standing alone as a "live" performance and intended only for audio presentation, would be much tighter and more like the Led Zeppelin I expect to hear on vinyl. C.F.-K.

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