#06691 (F)
$1.50

Profile: Drummer Nigel Olsson

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

VOL. 4 NO. 9
JUNE 1979

A Session with B.B. King

Unraveling the Cable Problem

LAB REPORTS:

Crest Audio Model P-3500 Power Amplifier

dbx Model 2BX Dynamic Range Enhancer

Harman Kardon hk-3500 Cassette Recorder

HANDS-ON REPORT:

Otari MX-5050-B Recorder

NEW PRODUCTS

RECORD REVIEWS

YAMAHA'S NEWEST TOURING PROFESSIONAL.

Yamaha's new PM-2000 Mixer. Ideal for professional sound reinforcement, it's the kind of full production console pros have always had in mind, but never in hand.

The PM-2000. The touch is solid, smooth, consistent. It feels like the professional console that it is.

The knob, switch and slider placement anticipate where your hands will naturally fall.

With 5-position, 4-band equalization and six independent sends on all 32 inputs, plus a full function, 14x8 matrix, the PM-2000 has everything you would expect from the consummate professional console.

And if the PM-2000 looks and feels like a custom console, and seems to have read your mind, it is no accident. Because Yamaha spent two years on intensive research and prototypes based on input from professionals. One touch and you'll realize: the PM-2000 feels how you think.

Available soon on a limited basis, through select Yamaha dealerships.

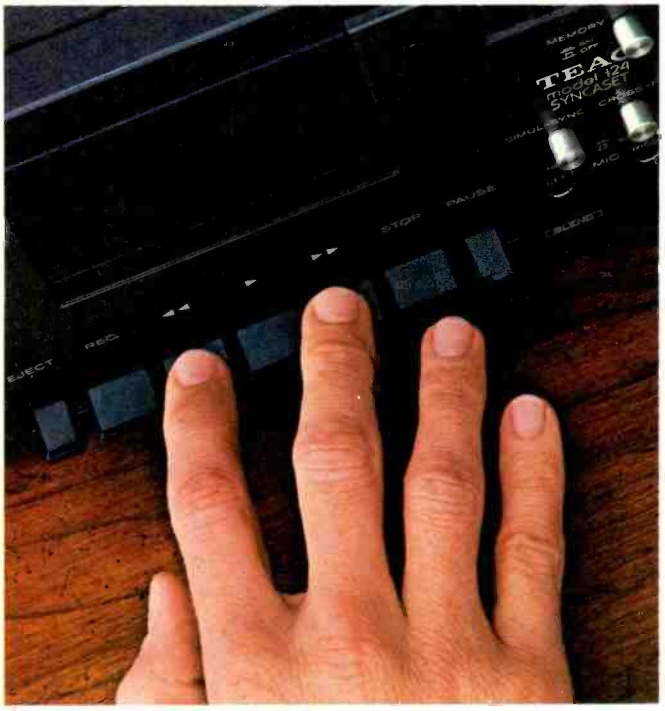
YAMAHA

P.O. Box 6600, Buena Park, CA 90622

Write for complete information on the PM-2000.

CIRCLE 99 ON READER SERVICE CARD

EVERY MUSICIAN SHOULD PLAY THIS KEYBOARD.



It controls the TEAC Model 124 Syncaset™. Our first cassette deck that lets you record one track, then overdub the other to get two musical parts in perfect time. Later, you can mix live material with these two tracks and hear all three parts through your home sound system.

With the Model 124, you can accompany yourself or an existing piece of music, and record the result. Rehearse a tune or create one. Sharpen your ear for harmony and phrasing.

And develop your timing and playing skills while you're at it.

*Dolby is a registered trademark of Dolby Laboratories, Inc.

After you've worked on your own music, enjoy the sounds of others. The Model 124 is an outstanding stereo cassette deck. High signal-to-noise performance. Low wow and flutter. Wide, flat frequency response. There's Dolby* NR (disabled in the "Sync" mode). Memory rewind for fast tape checks. And illuminated VU meters for easy level adjustments.

Probably better than anyone, we know the Model 124 can't give you all the multitrack flexibility and open reel performance you want. But at a third the cost of an open reel multitrack recorder, it could be the start-up tool you need. And when you consider the savings on tape alone, you'll find the Syncaset a handy, economical instrument to work with.

So try out the keyboard every musician should play. See your TEAC Multitrack dealer today for a demonstration of the Model 124 Syncaset.™

TEAC®
Multitrack Series

CIRCLE 56 ON READER SERVICE CARD

MODERN RECORDING

JUNE 1979
VOL. 4 NO. 9

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
A Session with Kansas Musicians: Where & How to Hook-up All Those Special Effects Devices
Profile: Boston's Tom Sholz

Modern Recording (ISSN 0361-0004) is published monthly by Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050. Design and contents are copyright by Cowan Publishing Corp., and must not be reproduced in any manner except by permission of the publisher. Second class postage paid at Port Washington, New York, and at additional mailing offices. Subscription rates: $12.00 for 12 issues; $22.00 for 24 issues. Add $3.00 per year for subscriptions outside of U.S. Subscriptions must be paid in American currency. Postmaster: Send Form 3579 to Modern Recording, Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050.

THE STAPLES

Cover photo & B.B. King B&W photos: Lynn McAfee
B.B. King color photos: Bob Khoury
Nigel Olsson photos: Murray M. Silver, Jr.

MODERN RECORDING
SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

H.G. La TORRE
Editor

PAM HIGHTON
AUDREY KURLAND
Assistant Editors

NORMAN EISENBERG
LEONARD FELDMAN
JIM FORD
JOHN MURPHY
BRIAN ROTH
Technical Editors

ROBERT ANGUS
NAT HENTOFF
DAVID MOYSSIADIS
FRED RIDDER
PETER WEISS
Contributing Editors

SEDGWICK CLARK
JOE KLEE
RUSSELL SHAW
Music Reviewers

LORI RESSA
Production Manager

BONNIE BRENNAN
CHRIS KELLY
KIM SMITH
Production Staff

BILL TRAVIS
Art Director

KAREN JENSEN
HAROLD PERRY
LIZ RYAN
BARRY SIMON
Art Staff

JANET KURTZ
Circulation Manager

MELANIE DEUTSCH
Assistant to the Publisher

BILL SLAPIN
West Coast
Advertising Representative

MYLES GROSSMAN
Advertising Director

VINCENT P. TESTA
Publisher

Editorial and Executive Offices
Modern Recording
14 Vanderventer Ave.
Port Washington, N.Y. 11050
516-883-5705

COWAN PUBLISHING CORP.
RICHARD A. COWAN Chairman of the Board & President
CARY L. COWAN Vice President
JACK N. SCHNEIDER Vice President, Marketing
RICHARD A. ROSS Vice President, General Manager
MARC L. GILMAN Credit Manager
AMY C. GILMAN Secretary/Treasurer
SANFORD R. COWAN Founder & President Emeritus

Editorial contributions should be addressed to The Editor, Modern Recording, 14 Vanderventer Ave., Port Washington, N.Y. 11050. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

LETTERS TO THE EDITOR

Pro-Analog Concerns

In one of your recent interviews with a successful producer/engineer, when asked what he planned to do when "digital" comes in, the producer replied, tongue-in-cheek, "Hopefully get out of the business by then!"

I, for one, *seriously* agree with that statement, or rather, more honestly, I am worried about the forthcoming digital trend, for two reasons: One is purely economic and the other is in terms of general and specific understanding.

It has taken me years to develop a real working knowledge of audio circuitry to the point where I really understand design and function of audio electronics. However, I know absolutely *nothing* about digital! Therefore, the "new wave" will effectively set me back six years (the amount of time it's taken me to master traditional audio electronic theory).

Furthermore, I'm just about to open a small studio. What happens when I invest my life savings in gear that will be considered obsolete in the immediate future?

I am sure there are many deserving, able engineers in your readership who, like me, can't afford to set up shop all over again every time a new "earth shattering" technology comes along!

My question is, in your opinion, exactly to what degree will digital make analog obsolete? Will conventional multi-track boards (analog) be of no use in tomorrow's world, like their tape machine counterparts? Will analog master tapes be sneered at by the elite of the record industry?

I am, I suppose, one of those "horrible few" audio enthusiasts who is actually very comfortable and satisfied with today's analog recording capabilities. Now, as much as that statement may disgust some, I don't apologize for it, and feel that my point of view has substantial merit.

Everything I read, including your articles, seems to be pro-digital. Well, digital may dig your grave as well as mine! Because if digital becomes the only real way to do things in the future, that'll not only eliminate the thousands of "semi"-pro studios, but the publications that serve them, as well.

I'd like to see comments from other readers who feel as I do. I have a feeling that "digital" may go down the same path as "quad" of a few years back. I, for one, certainly hope so.

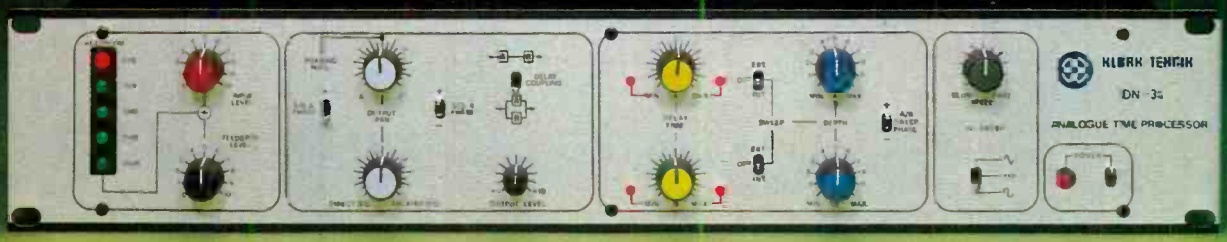
—Reginald Parks
Augusta, Ga.

We think you might be missing the point and over-reacting. When digital recording becomes more prevalent, one can expect to see many "hybrid" studios, where the console is traditionally analog and the master two-track recorder is digital.

The chief advantage of digital is the recording, *and in the tape itself, the storage medium. The digital process offers high dynamic range that is not degraded with successive overdubbing to create a master tape. Also, distortion levels are as low as 0.03%. The mix-down process is the most lucrative place to go digital—and the entire studio does not have to follow suit. The analog multitrack recorder will continue to be around, we do not doubt, for a long time to come.*

We do hope that our May 1979 issue cover story on a highly am-

The new DN34 analogue time processor.



Think of the effect you'll have.

The new DN34 analogue time processor is an exceptionally versatile signal processing and special effects unit, designed around two discrete, independently controllable delay sections.

Like other Klark-Teknik products, the DN34 is the result of intensive research and development – the best there is in state-of-the-art analogue delay technology.

With a product of this stature you can achieve all these effects cleanly and noiselessly.

- Positive flanging.
- Negative flanging.
- Double tracking.
- Resonant flanging.
- Triple tracking.
- Loudness enhancement.
- Pitch detune.
- Pitch shifting.
- True vibrato.
- Chorus.
- 'Cardboard tube' echo.
- Doppler/Leslie effects.

And, if this isn't enough, the DN34 can give you such new effects as:–

- Crossover flanging.
- Time-related frequency synthesis.
- Complex Doppler effects.

The DN34 analogue time processor also offers you:-

- A dynamic range better than 90dB.
- A time sweep range of 70:1.
- T.H.D. at less than 0.3%.
- Numerous exclusive features including full 'on board' mixing and phase reversal facilities.
- Amazing performance and value for money.

The DN34 is unequalled in the signal processing field today.

And we're not just saying that for effect.

KLARK-TEKNIK

You know it's the best.

For further information about the DN34, our new DN70 digital time processor, and also our DN27 and DN22 graphic equalisers: **Hammond Industries Inc.** 55 Michael Drive. Syosset. New York 11791 (516) 364-1900; West Coast Office (213) 846-0500; Canada (416) 677-0545

CIRCLE 96 ON READER SERVICE CARD

bitious and sophisticated PCM project was enlightening and not further alienating. Please, don't go out of business yet!

P.S. With regard to your understanding of design and function—Consider what happened a number of years ago when technicians familiar with tubes were faced with transistors. . . .

Small Change, Greatly Pleased

I am currently chief engineer for a moderately sized college radio station in Eastern Pennsylvania. I am also quite involved in a small recording studio that I recently hanged together in the basement of my own home.

For these and other reasons, I think I can call myself a good judge of audio and semi-pro recording equipment as well as of the periodicals in publication that accompany this field. I hold subscriptions to *Billboard* and to *Broadcast Management/Engineering*, and up until now I have been very satisfied with them. Just a week or two ago, though, while in a local bookstore, I spotted your magazine and, for the sake of getting rid of loose change, shall we say, I bought a copy. Upon reading it, I discovered it was one of the most informative, descriptive and best-worded magazines I have ever read. Congratulations on a great magazine raised far above all other "audio" (*MR* is much more) magazines.

The point of this letter is not only to give praise where due, but also to add my name to your list of subscribers. My money order is enclosed.

Once again, congratulations on a terrific magazine.

—Clive R. Chittick
Bethlehem, Pa.

Loose change, huh? We were really hurt before we read you liked what you found. Welcome to our computer, and thanks for writing!

Misses Reader Service

In addition to the products circled on the enclosed Free Information Service card (from the February 1979 issue of *Modern Recording*), I am *extremely interested* in receiving material from Micor Audio Products Div. regarding the new Coupland Digital Synthesizer. This particular product was listed as Number 17 in January 1979's issue of *MR*, but unfortunately the copy I purchased had no reader service cards, which frustrated me greatly since I've been hearing about the Coupland for a long time, and have continued to be unable to obtain their literature or even locate an address where they might be contacted.

Thank you for any and all assistance.

—Mark Pettigrew
Sioux City, Iowa

Sometimes the problem of missing reader service cards does crop up (and it always involves the issue with the product that you're dying to find out more about) and we are sorry for the inconvenience. Micor Inc., Audio Products Division, can be reached at P.O. Box 20885, Phoenix, Arizona 85036; phone (602) 273-4381 for information on the Coupland synthesizer.

Canadian Railroad Interest

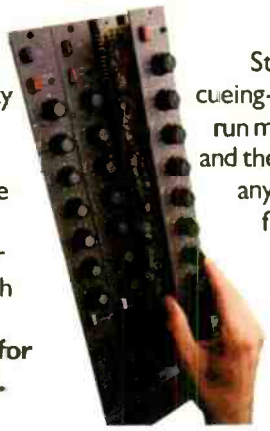
It was with extreme interest that I read Len Feldman's recent cryptic and enticing remarks ("Ambient Sound,"

The new M12.
It eliminates the three deadly sins of live performance mixing.

If a mixer can't deliver a complete range of functions, high-quality electronics and rugged reliability, it can't bring a live concert to life.

But now, Fender® introduces the M 12 Live Performance Mixer—a fully expandable 12 in/4 out mixer with staggering possibilities for both live concerts and live recording.

Functions you want for the options you need.



Start with submasters, limiters, cueing-talkback, hi-level in/outs that run multiple effects simultaneously and the capability of assigning signals anywhere on the board. They're functions you'd expect to find on mixers costing a whole lot more.

Mixing drums, keyboards or vocals is a job in itself. You can take advantage of the M 12's full-function design by mixing as many mike or direct inputs as you need on one submaster, then mixing the rest of your band and patched-in effects on the other.

Full cueing-talkback. Listen to Program or any of the three monitor mixes or any independent channels via earphones. And communicate to Program, monitors or the cue/stage monitor mix with talkback.

High-performance electronics. All Lo-z input and output channels are transformer coupled and floating. High slew rate, low-noise op amps are used throughout. Continuous gain controls allow input impedances to remain unaltered. Equivalent input noise is −128 Dbm.

Rely on it. The M 12 was definitely designed to perform *every* time. The rigid extruded front panels and built-in case keep the M 12 mixing down when other mixers give up. And modular construction makes a rare servicing a snap.

Check out the M 12's complete value story at your authorized Fender dealer. With functions, electronics and reliability like this, the M 12 just might be a whole new standard in live performance mixers. Check it out today.

NEW from Fender®
MADE IN U.S.A.

Professional Sound Products
1300 E. Valencia Drive
Fullerton, CA 92631

CIRCLE 101 ON READER SERVICE CARD

December 1978) on Bob Carver's magnetic field amplifier in *Modern Recording.* This was (is) good copy!

I am interested in more and would like to follow this development very closely. Thus, can you provide me with the name and address of Carver's company? Also, please continue to update *MR* as you and Mr. Feldman receive more information. Carry on!

—"Railroad"
Vancouver, British Columbia

The Carver Corporation can be reached at P.O. Box 664, Woodinville, Washington 98072.

MR *will be testing and evaluating the new amplifier as soon as we can get our hands on it—keep watching Lab Reports for this piece of equipment.*

Big on Buyer's Guide

I found your Winter '79 *Buyer's Guide* very informative, especially David Moyssiadis' "The Making of a Record." I had no idea how many steps were involved in cutting a record, let alone the great costs involved. The author did a fine job covering every aspect of making a record, from preparing material and finding a studio to promotion and distribution. This article will prove invaluable to anyone even considering any type of professional recording. Congratulations on some great work.

—Jim Tormey
Northport, N.Y.

Hopes to Oscillate in Nashville

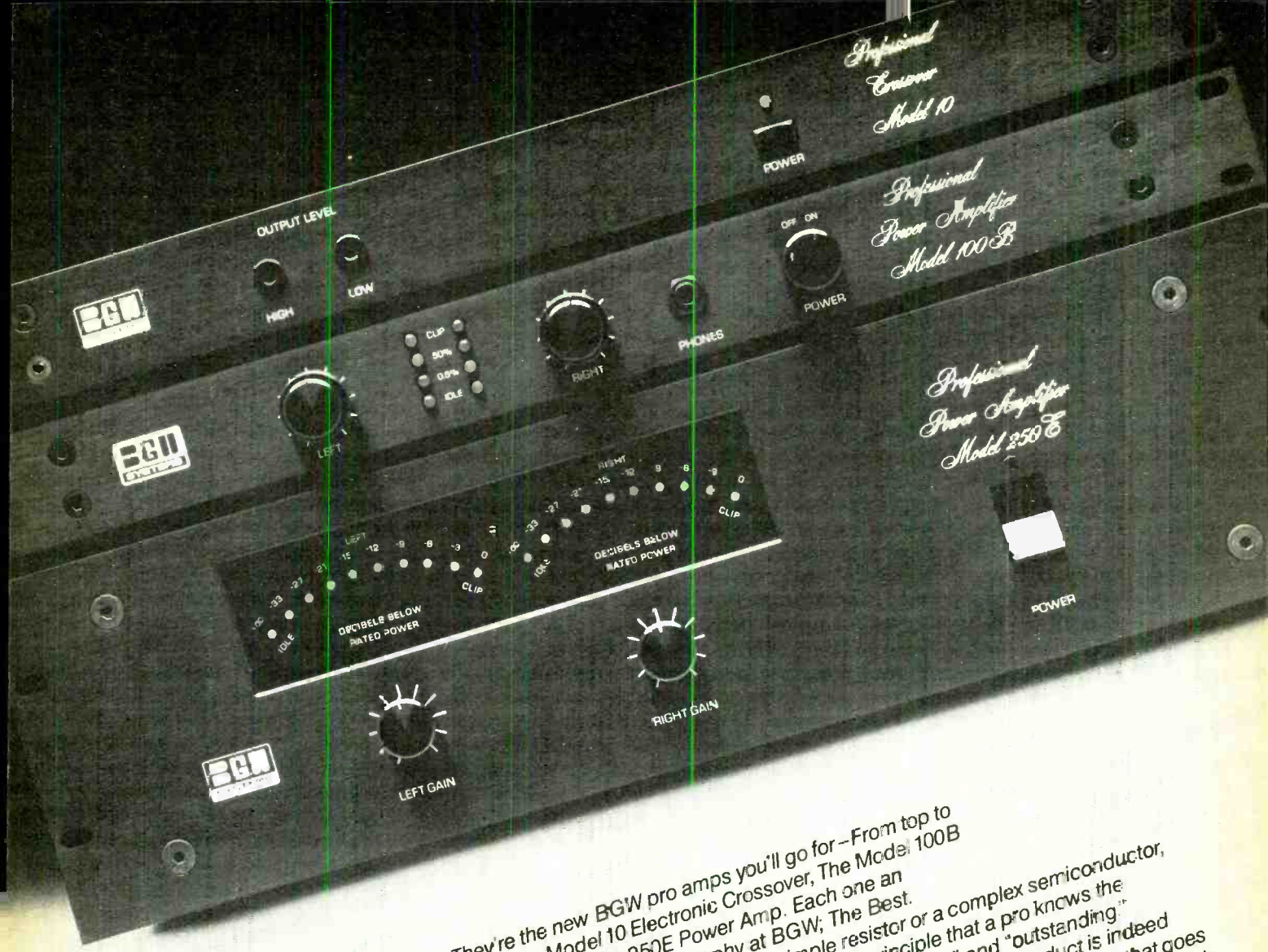
I am a senior at the State University of New York Fredonia campus and am enrolled in the Sound Recording Technology program here. Just recently, I borrowed some back issues of your magazine from my instructor, and was amazed at the wide spectrum of readers that your magazine effectively reaches. Being heavily involved in electronics, I was especially pleased with the articles on this particular subject; there seems to be a general lack of knowledge of basic electronics amongst budding engineers. I am continually repairing things such as broken cables, direct boxes, etc., simply because people won't use their heads or are afraid to trust themselves to try *any* kind of repair. (I don't, of course, mean trying to fix a bias oscillator when you don't know what the bias oscillator is or does!) But please, encourage your readers who are just entering the recording field to learn as much as they can about electronics and not to be afraid to try, once they've acquired the knowledge to begin.

A primary reason I am writing to you, though, is because in the April '78 issue, on page 41, you have a "Newsical" on the Govox Inc., adjustable guitar nut. Could you please print the address of the company?

Also, when I graduate next spring, I am planning on going to Nashville to try and get a job as a technician in a studio, or with a recording-related company. Where could I get a list of some of the bigger studios and some of the companies that are located in Nashville? Any help or advice you can give me would be greatly appreciated.

—Steven J. Hebrock
Fredonia, N.Y.

The address we have for Govox Inc. is 1318 W. Oak St., Kissimmee, Florida 32741, phone (305) 847-8570. For Nashville information such as what you desire, Billboard *Publications (as men-*



They're the new BGW pro amps you'll go for—From top to bottom, The Model 10 Electronic Crossover, The Model 100B Power Amp, and the 250E Power Amp. Each one an extension of our basic philosophy at BGW; The Best.

Whether its a simple resistor or a complex semiconductor, BGW is dedicated to the principle that a pro knows the difference between "good enough" and "outstanding." Because we know that quality in a pro product is indeed something you can see, touch and hear, everything that goes into a BGW product, everything we do in its manufacture, is designed to back up the man who depends on our gear to make his living. And, like all BGW products, the new 250E with our exclusive arc-scale power indicators joins with the new 100B and Model 10 in answer to your demands. They're worthy additions to the solid-gold reputation that's been earned in recording studios, discos and concert stages throughout the world.

If you're not already a BGW Pro we invite you to compare any of our products with anybody else's. Anybody's. You'll then see why so many of your pro friends swear by us. And our competitors at us.

Professionals Swear by Them, Competitors, at Them.

Go For The Best

BGW SYSTEMS

BGW Systems, Inc.
13130 South Yukon Avenue,
Hawthorne, California 90250
In Canada: Omnimedia Corp.,
9653 Cote de Liesse
Dorval, Quebec H9P 1A3

CIRCLE 72 ON READER SERVICE CARD

tioned in our April 1979 issue) has a number of annual directories of value. The one most suited to your needs probably would be their Country Music Sourcebook. *Write (or call) for more information:* Billboard *Publications, 1515 Broadway, New York, N.Y. 10036; phone (212) 764-7300. (Those eagle-eyed readers who saw our April reference to* Billboard *with a Los Angeles address please note: L.A. harbors main offices for the publishing company; there are quite a few regional offices, including the one in New York City.)*

Candid Remarks

I wanted to write in reference to the *Modern Recording* issue featuring Fleetwood Mac, about which I can truthfully say for the first time I really enjoyed a studio session article. It was somewhat unorthodox in that it dealt more importantly with the attitudes of the bandmembers and engineers, and I can appreciate Mr. Dashut's candid remarks about himself and the industry.

Second, I would like to see the same type of article on some of the techno-rock bands like Styx and Kansas. Another good idea would be into other areas of music, such as jazz-rock (i.e., Brand X, Stomu Yamashta). Reading about how to record rock and roll month after month is great, but I am looking for new, innovative ideas, as well. Perhaps if not a recording session, how about a profile on Peter Gabriel, Brian Eno, Robert Fripp or Larry Fast and their crowd? Let's see something different, occasionally. Thanks,

—Bennett Check
Atlanta, Ga.

Interesting names there. We'll check out the possibilities of these.

Trouble Finding Work

Each month I read your magazine from cover to cover and have been doing so for some time. I find each issue enlightening and enjoyable. In recent issues, I have noticed a growing concern for a listing of schools which teach recording engineering or subjects of a similar nature. My concern goes one step further—where do you find a place to practice your newly acquired knowledge of engineering?

For several years now, I've worked with musicians both on the road as a soundman and in the studio. The guitarist and bass player that I travel with have both logged studio time and traveled with different bands. Our guitarist is also a recent graduate of the Jon Miller Sound Studios (Jon submitted a letter in your March issue). Even with our good foundations in both music and recording, though, we've experienced much difficulty in finding employment with recording studios. It's not the kind of work that you can find in the "Help Wanted" section of any newspaper. On many occasions, we've been forced to seek employment in the regular job market. That takes a lot of time from practice and study, not to mention what it does for stifling creativity. My question to you is this: How, if at all, can a person with a good working knowledge of sound, music and recording find employment with recording or sound reinforcement companies? Is it really as easy as Richard Dashut found it (truly a stroke of luck and genius)? Are there any openings? If so, how do you go about finding and landing one?

One last thought—it might be helpful if studios and sound reinforcement companies in need of experienced personnel would send an ad to your classified col-

Introducing the system that can tape itself. Tune itself. All by itself. For up to a week.

Up to now using a tape deck to record Tchaikovsky's 4th at 4 PM on one station and Beethoven's 5th at 5 PM or another was easy. As long as you were home. Now Technics makes it just as easy when you're not, with the ST-9038 quartz synthesizer FM stereo tuner and its matching SH-9C33 microprocessor.

When used with the ST-9038 the SH-9038 microprocessor can be programmed to tune eight FM stations in any order, at any time, on any day for one week. In fact, the SH-9C33 can be programmed to remember 32 individual steps. Starting with the day of the week, the time the FM station and AC line on/off.

All you do is simply select the "write" mode and the SH-9038's computerized memory does the rest. Then select the "read" mode for a readout of the programs you have selected. You can also override your preselected program by switching to the manual mode. What's more, the SH-9038 can be programmed to turn on or off three other components in addition to the ST-9038 tuner.

That's what the SH-9038 microprocessor can do. What the ST-9038 quartz synthesizer tuner can do is just as impressive. Unlike conventional tuners which use a series of variable capacitors to tune in FM frequencies, the ST-9038 uses the quartz synthesizer tuning system. With this system the quartz crystal, one of the world's most accurate reference devices, becomes the reference for the local oscillator frequency and the broadcast frequency. The results: Only the frequencies on which a broadcast signal might exist can be received. At precisely spaced 200 kHz steps. And that means you don't have to worry about drift or misalignment due to temperature, time or mistuning.

The SH-9038 microprocessor and the ST-9038 tuner. Because you can't beat the memory of a computer or the accuracy of quartz.

SENSITIVITY: 1.2 μV (75Ω), 50dB QUIETING SENSITIVITY (New IHF): Mono 18.1 dBf, Stereo 38.1 dBf. THD (100% modulation): Mono 0.1%, Stereo 0.15%. FREQ. RESP.: 20Hz to 18 kHz +0.1dB -0.5dB. SELECTIVITY: 75dB STEREO SEPARATION: 45dB (1 kHz), 35dB (10 kHz). IMAGE REJECTION AT 98 MHz: 105dB.

Technics SH-9038 and ST-9038. A rare combination of audio technology. A new standard of audio excellence.

Technics
Professional Series

CIRCLE 98 ON READER SERVICE CARD

umn. Finding an opening might involve travel but that's a small price to pay.

If by chance any other readers could pass on any knowledge, my friends and I could use the help. Great job on a great magazine! Any help that you can give us will be greatly appreciated.

—Michael Newsome
Arnold n Miller Band
currently, Panama City, Fl.

Michael, not to come down hard on you, but we can't quite believe you might be as naive as you sound. Employment in any creative field, especially in this day and age (to coin a phrase) naturally is extremely limited.

There are overnight success stories and there are long, hard climb stories. Richard Dashut, as he described to our Fleetwood Mac session story writer, in the March 1979 issue, swept floors and cleaned toilets for a period of time—not quite an immediate stroke of luck and genius, as we see it, anyway.

The openings you're looking for wouldn't likely be listed in daily newspapers, but in papers for the trade, regional music publications, etc. The MR *classified section is a valid forum for job openings.*

Consider that you have a good foundation in music and recording; that you have been working in your field for several years. You are well ahead of countless others and may get that break any moment. Or maybe after a couple more lean years.

Again, please don't take this response "wrong." We're afraid we haven't quite been helpful (or gentle, for that matter). But thanks for writing. We hope you continue to enjoy Modern Recording, *and we wish you luck and genius!*

Note of Praise

I am a recording addict and love your magazine and someday would like to work in a studio. I'm sure all of the other recording freaks who like your mag appreciate your providing information on recording schools and courses that are offered; I certainly do!

So please, if possible, keep printing any additional info you get on schools.

From all of us who are trying to learn something, thanks for the info and keep up the good work.

—L. Mirand
Bloomington, In.

Left "Dingling"

My thanks and congratulations on "A Session With Fleetwood Mac" in the March issue (1979). I must admit that I was surprised to find out that the band uses a $7.86 microphone to record on, but Ken Klinger mentions it is not meant to be used as a microphone. What is it really, and where can I get one? If it is as good as any Neumann, then I sure could save a lot of money. So I would appreciate your filling me in on this.

Thank you for a great issue.

—David Dial
Atlanta, Ga.

Allow me to be the 3,183rd reader to write you concerning a reference in your March issue to a microphone called the "Dingle." $7.86? If Ken Klinger's claims for this device are true, he's preventing someone (or himself) from becoming very rich, very fast.

As a drummer in a local rock band, the problems of getting my sound into the audience seem endless. Good mics are starting to rival good drum equipment in price. I feel sure that us poor clods strug-

You want your audience to hear your music clearly. Naturally. And without distraction.

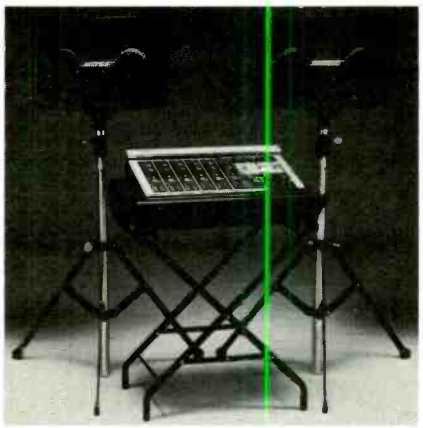
This Bose PA system will do it for you.

Bose 802 speakers cover your audience uniformly and smoothly. Which means your sound will never be boomy or piercing. And the small enclosures won't distract the audience from your performance.

Perfect mate to the Model 802 speakers is the PM-2 Powermixer. An all-in-one unit that incorporates a low-noise mixer, parametric equalizer, and 400-watt professional power amplifier.

Powerful yet compact, the Bose PA System fills the room with sound, yet fits in the back of your car.

So many audience-conscious groups in countries all over the world let Bose deliver their message.



Ask your Bose Professional Products dealer for a demonstration and find out why.

Bose Corporation, Dept. MR
100 The Mountain Rd.
Framingham, MA 01701

Please send me a copy of the Bose Professional Products Catalog and a complete dealer list.

Name _____
Street _____
City _____
State _____ Zip _____
Tel. ( ) _____

BOSE®

When music is your message, let Bose deliver it.

gling in clubs across the country would buy these "Dingles" up by the millions, if they truly do what Klinger says they do.

My obvious question is—Can Murray Silver, or *Modern Recording* get us some information on just what this thing is, how it works, and how *we* can buy them by the crate, like Mr. Klinger has?

Thanks for a fine magazine. My friends and I learn something from every issue. Keep up the good work.

—Bob Veazue
Brockton, Ma.

Well, we'd like to publish the information, but we've been asked not to, at least not at the present time. The nature of the "Dingle" is considered a trade secret by Ken Klinger, who claims to have been the first to discover its use and who has dispensed, we understand, with all conventional mics. We've been informed that other studios are on to the technique, though, and that it's not quite so well-kept a secret anymore.

Murray M. Silver, Jr., assures us that in the near future, possibly in a session story presently in the works, he and we will be able to disclose the specifics.

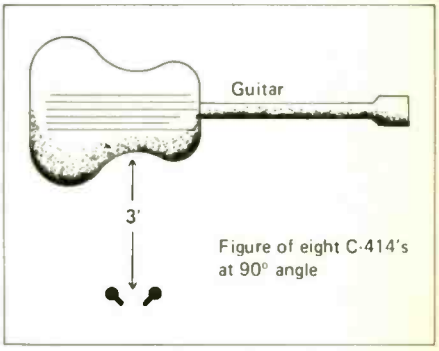
Natural Noises

As regards the review by Joe Klee of the Larry Coryell *European Impressions* album ("Groove Views," April 1979), on which I had the pleasure of working, I should like to make a few additional observations.

As shown in the photograph accompanying the review, Larry played an Ovation Electric-Acoustic guitar with a built-in stereo pickup. This rather unusual pickup splits the first, third and fifth strings to the left and the second, fourth and sixth to the right.

Larry and Michael (Cuscuna) wanted to mix in the direct injected sound of the stereo pickups with the sound from the mics, which accounts for the closeness, and some of the noise to which Mr. Klee referred, of the guitar.

Actually, the mics (two AKG-414s) were a good three feet in front of the instrument; in Figure-of-Eight pattern; and at right angles to each other (see illustration).



As Mr. Klee rightly pointed out, some people do like the little finger squeaks and other similar "natural" noises and I must admit that I am very much one of these people.

Thanks for a most pleasant review, and for saying my side sounded better!

—William Paul Wittman
Senior Balance Engineer
Soundmixers
New York, N.Y.

Joe, who had stated in his review that his efforts to find out what sort of guitar Larry plays were met with frustration, was glad to have this information and added that the achievement in getting such a natural sound was impressive. Thank you for writing and giving us the details.

FOR THE SMALL STUDIO OWNER WITH BIG IDEAS.

If you're a small studio owner, you may have a problem: your ideas are far beyond your present equipment.

Maybe you're an engineer dreaming of an automated 24-track studio. Maybe you're a producer, searching for "the next big thing." Maybe you're an artist, trying to land a record contract. What you need is something that will get you from where you are to where you'd like to be.

Sound impossible? Not to us. At dbx, we're committed to make professional recording technology available to anyone with the determination to make use of it. We make a line of rack-mountable signal-processing devices designed and priced especially for the small studio.

Our tape noise reduction systems, the 155 (4-channel, switchable), the RM-155 (8-channel, switchable) and the 158 (8-channel, simultaneous), all offer the same 30 dB noise reduction and 10 dB headroom improvement as our state-of-the-art units and are fully compatible with them. They enable you to make master quality tapes, instead of demos, on your present equipment.

Our 161 and 163 compressor/limiters feature true RMS signal detection, which closely resembles the response of the human ear, and feed forward gain reduction, which allow for infinite compression capability. The 163 employs "Over Easy" compression, the most natural-sounding you've ever heard, and its "one knob" operation is the simplest around.

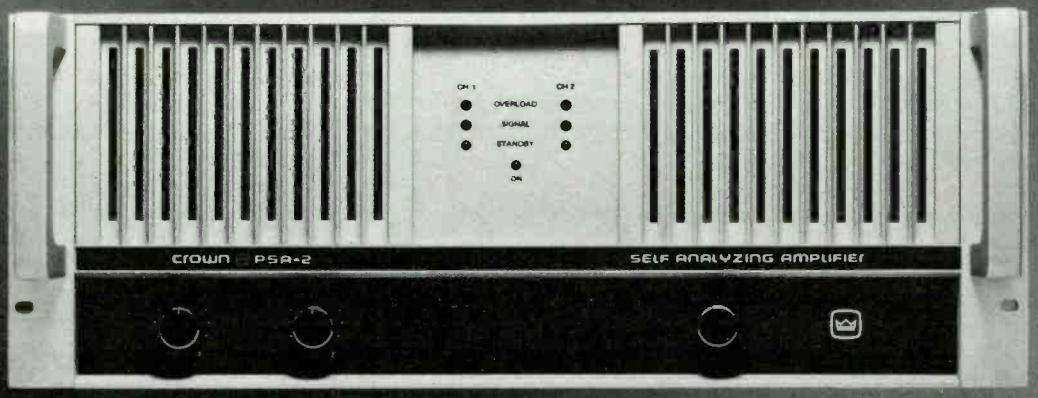
We can't guarantee our products will make you a star. But if you've got the talent, they'll take you as far as you want to go. dbx, Incorporated, 71 Chapel Street, Newton, MA 02195 617-954-3210.

dbx

UNLOCK YOUR EARS

CIRCLE 92 ON READER SERVICE CARD

Five years from now,



The Crown PSA-2 Professional Power Amplifier*

220 watts per channel minimum RMS (both channels operating) into an 8 ohm load, 20Hz-20KHz at a rated RMS sum total harmonic distortion of 0.05% of the fundamental output voltage. (tested per FTC specifications)

250 watts ±1dB per channel, 20Hz to 20KHz into 8 ohms with no more than 1.0% THD (EIA Std. SE-101-A).

400 watts ±1dB per channel, 20Hz to 20KHz into 4 ohms with no more than 1.0% THD (EIA Std. SE-101-A).

685 watts ±1dB at 1K per channel into 2 ohms, with no more than 1.0% THD.

*Designed for installation and use in professional sound systems

the Crown PSA-2 amplifier will still be unique.

There is unique technology built into the new Crown PSA-2 amp that is not available to other amp manufacturers. Our competitors may try to copy the PSA-2, but only the Crown label guarantees you access to that technology.

You will experience that technology as reliable, long-term performance of the PSA-2. No other amplifier combines such power and dependability.

Here's why.

For over ten years, Crown has tested every output device manufactured for us. We built an electronic wizard — the SOAR III Transistor Analyzer — to determine for ourselves the safe operating area (SOA) of each type of output device. Designers have long understood that the SOA changes as operating conditions change, but until now there has been no way to define and compensate for these changes. The SOAR III has changed all that — exclusively for Crown.

As a result, we can include in the PSA-2 analog computers connected to sensing units which constantly monitor the operating circumstances of each output device. These self-analyzing circuits are programmed at the factory with Crown's data on the SOA. For the first time, the protection circuit actually follows the changes in transistor SOA resulting from operation of the amplifier. If an output transistor exceeds its SOA for any reason, the self-analyzing circuit limits the output, preventing its destruction. If the SOA is not exceeded the output devices are not limited in any way.

What good does that do you?

The Crown PSA-2 provides more usable power from each output device. There are no arbitrary voltage or current restrictions on the output.

You get reliable power for less money than you might expect. Output devices are expensive. Only Crown has learned how to use them at maximum efficiency.

In the PSA-2, you'll also find

- . . . . a two-speed fan and completely enclosed high-efficiency heat sinks
- . . . . balanced variable gain (XLR) inputs on a back panel plug-in module
- . . . . switchable high and low pass 3-pole Butterworth filters that are factory-set for 50Hz and 15KHz, with other roll-off points available
- . . . . a push-button test-tone generator
- . . . . an adjustable-threshold compressor to limit output at the user's discretion
- . . . . switch selectable low-frequency load protection
- . . . . switch selectable turn-on delay
- . . . . thermal-sensing power supply protection to eliminate premature fuse-blowing
- . . . . stackability (without a cabinet)

The Crown PSA-2 is a unique professional component. With the PSA-2, the amplification systems you are bidding today will still be state-of-the art years from now. Call us for spec or delivery information at 219/294-5571.

crown

1718 W. Mishawaka Road, Elkhart, Indiana 46514

American innovation and technology...since 1951.

CIRCLE 104 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.*

Definitive Device Dossier

In a P.A. system of 100 watts or more, what is the difference between an L-pad, a rheostat and a pot? What are the characteristics of each? When would an L-pad be used instead of other variable resistance components?

—Ron Finizio
Riviera Beach, Fl.

Perhaps the best way to answer this question is to provide, for each device, a basic description, an explanation of the "theory" of operation, and some general comments on applications.

Since the theory discussions will involve some manipulation of formulas and symbols (nothing hairy, just multiplication and division), as a first step it would be a good idea to establish the symbols and notation to be used. Traditional circuit theory symbols will be fine, so in the following discussions voltages will be represented by Es, currents by Is, and resistances by Rs. These symbols, when referring to a specific circuit, will have two-letter subscripts indicating the lettered circuit points across (or through) which the quantity is measured (e.g., the voltage across points A and B or across the resistance R_{AB} in Fig. 1 is E_{AB}). Other subscripts will be easy to identify, such as E_L for voltage across a load, or E_S for source voltage.

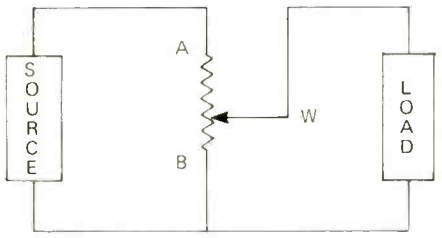


Figure 1

As an exercise in the use of these symbols (without subscripts) here are the three "versions" of Ohm's Law:

$$E = I \times R \qquad I = \frac{E}{R} \qquad R = \frac{E}{I}$$

These formulas state, reading from left to right; the voltage drop (in volts) across a resistance is equal to the current (in amperes) through the resistance multiplied by the resistance (in ohms); the current through a resistance is equal to the voltage drop across the resistance, all in appropriate units; the resistance in a circuit (or portion of a circuit) is equal to the voltage drop in the circuit (or portion) divided by the current through the circuit (or portion), again in appropriate units.

The preliminaries having been dealt with, the next step is to define and describe the operation of a potentiometer. A potentiometer (literally "measure of potential") or "pot" is a variable resistance connected in such a way as to provide a means of determining what fraction of a source voltage is delivered to a load.

In audio circuitry, potentiometers can be used "across" signal paths between stages of amplification or at the input to amplifier stages as volume controls. This is probably the most common use of potentiometers in audio equipment. However, since potentiometers are essentially voltage controls, they can be used to control power supply voltages as well. Potentiometers are generally *not* used for applications such as mixer faders, at least not in well-designed equipment.

A potentiometer (see Fig. 1) performs its function of dividing up the resistance element R_{AB} into two portions R_{AW} and R_{WB}, the moving contact or "wiper" (W in Fig. 1) being the point of division. As shown in Fig. 1, the source is connected across the entire resistance element, while the load is connected across that portion of the element between the wiper and the end of the element marked B. The fraction of the source voltage that appears across R_{WB}, and consequently across the load, will be equal to the fraction $\frac{R_{WB}}{R_{AB}}$. The following theory discussion will show why this is so.

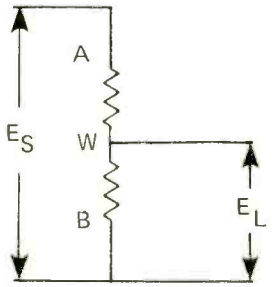


Figure 2

Figure 2 is a re-drawn version of Figure 1, using the previously defined symbols and notation, and with the single resistance element R_{AB} represented as two distinct resistances, R_{AW} and R_{WB}. It can be said that $R_{AB} = R_{AW} + R_{WB}$. Some other relationships that can be seen in Fig. 2 are,

$$E_S = E_{AB} \quad E_L = E_{WB}$$

According to Ohm's Law, the total current, I_T flowing in the circuit consisting

of the voltage source, R_{AW} and R_{WB} is

$$I_T = \frac{E_S}{R_{AW} + R_{WB}} \quad \text{or} \quad I_T = \frac{E_S}{R_{AB}}$$

Another fundamental law of electric circuits is Kirchoff's Law for voltages. This law, stated simply, says that in a circuit such as the one in Fig. 2, the sum of the voltage drops across the individual resistance ($E_{AW} + E_{WB}$) is numerically equal to the source voltage (E_S). That is, the entire source voltage appears across the entire resistance element. Written out,

$$E_S = E_{AB} = E_{AW} + E_{WB}$$

Re-writing E_{WB} and E_S using Ohm's Law to express these voltages as products of currents and resistances,

$$E_L = E_{WB} = I_T \times R_{WB}$$

$$E_S = E_{AB} = I_T \times R_{AB}$$

We are interested in forming the fraction $\frac{E_L}{E_S}$, since this fraction represents the fraction of the source voltage that appears across the load.

$$\frac{E_L}{E_S} = \frac{I_T \times R_{WB}}{I_T \times R_{AB}}$$

The I_T's "drop out" and what remains is

$$\frac{E_L}{E_S} = \frac{R_{WB}}{R_{AB}}$$

Since $\frac{R_{WB}}{R_{AB}}$ represents the fraction of the total resistance R_{AB} that exists between the wiper and the end of the resistance element marked B, it is this fraction that determines the fraction of the source voltage appearing at the load.

A rheostat (see Fig. 3) is also a variable resistance and differs from a potentiometer only in the way it is connected in a circuit. While a potentiometer is connected across or in parallel with a source, a rheostat is generally placed in series to act as a current regulating control. The rheostat shown in Fig. 3 is connected as

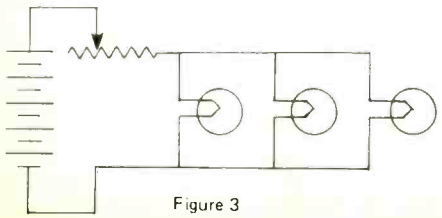


Figure 3

a dimmer for the lamps in the circuit. This is a very common application of rheostats.

The variable resistances available at electronic supply outlets (the ones that look like volume controls) can be installed either as potentiometers or rheostats, depending on how they are wired into a circuit. However, those variable resistances sold as rheostats can only be used as rheostats since they provide a means of connection at only one end of the resistance element and at the wiper. To be used as a potentiometer, a variable resistance must have connections provided for *each* end of the resistance element as well as for the wiper.

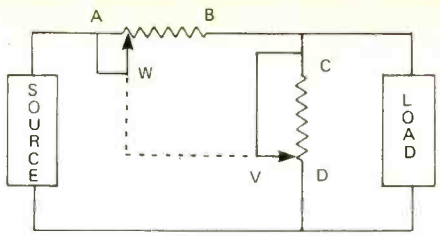


Figure 4

An L-pad (see Fig. 4) is a device, also based on variable resistances, with the characteristic of maintaining a (more or less) constant resistance across the voltage source. The total resistance across the source at any setting of the L-pad is equal to the sum of the resistances R_{WB} and R_{CV}. The segments R_{AW} and R_{VD} are shorted out by the wired connection of the wipers $_W$ and $_V$ to points $_A$ and $_D$, respectively.

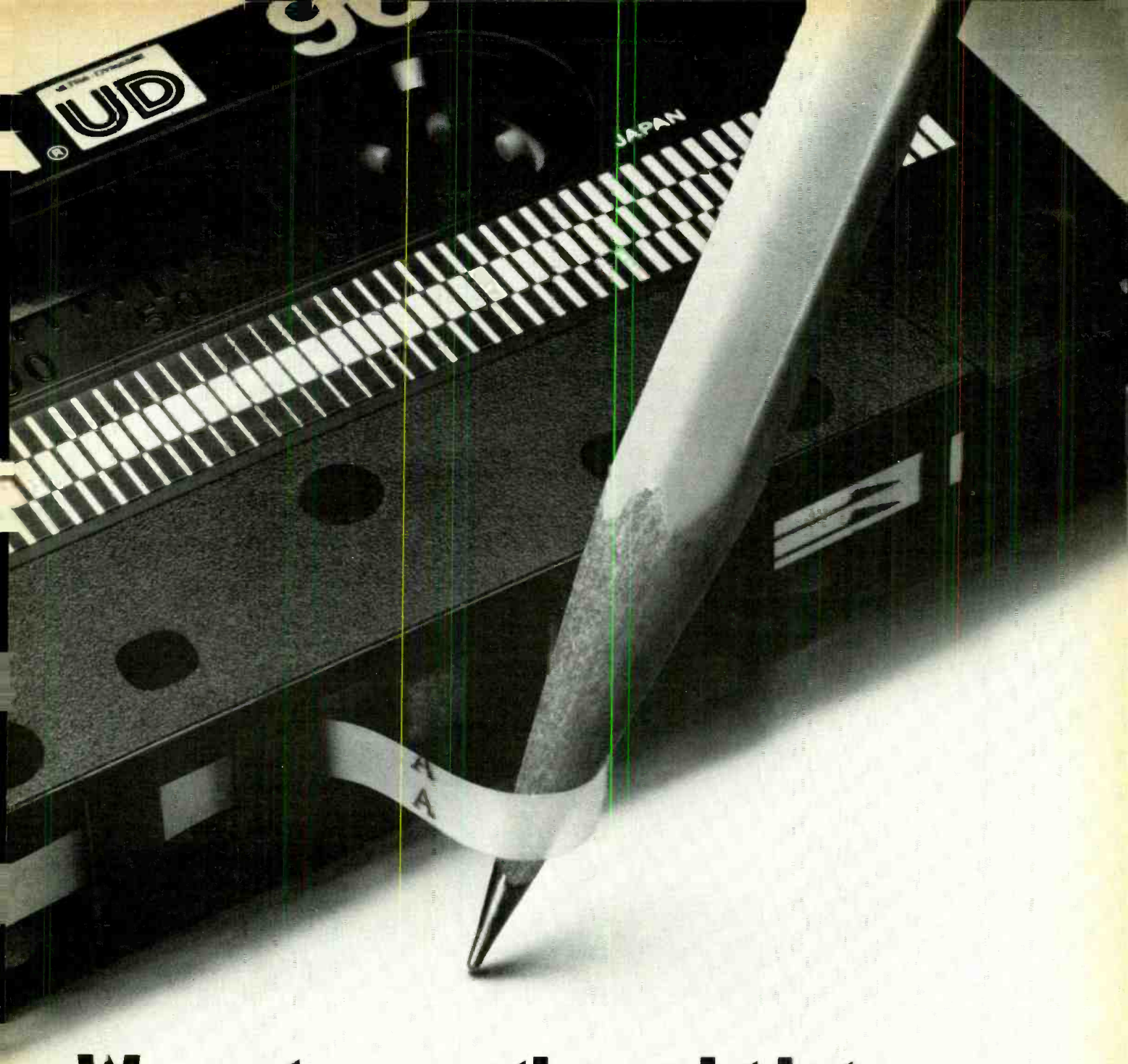
Note the dotted line connecting the two wipers. This is meant to indicate that the wipers move together. However, it is also shown that the wipers are at *opposite extremes* of their total travel. Thus, R_{WB} is at a maximum while R_{VD} is at a minimum or zero resistance. As the L-pad shaft (which moves both wipers) is moved from one extreme to another, the *individual* resistances will vary from this condition, but their *sum* will remain a constant.

This has the dual effect of (1) controlling the total power (power=current x voltage) delivered to the load through the current-controlling action of R_1 and R_2 (actually connected as a rheostat), and (2) maintaining a constant total resistance across the source ($R_{AD} = R_{WB} + R_{VD}$). L-pads are used in audio power control circuits. For example, the tweeter and midrange controls on speaker systems are generally L-pads. Although an L-pad will provide constant resistance across a source, there is no such provision going the other way. That is, the effective resistance of R_{VD} can be *zero*. Therefore, if a commercially manufactured L-pad is to be used in a high-power audio circuit, extreme care must be used in installation to prevent the L-pad from being wired in reverse. That is, the elements labeled R_1 and R_2 in Fig. 4 should always be connected to the source. Otherwise a high power audio signal will be short-circuited, with results that may be something close to spectacular.

—Peter Weiss
Contributing Editor
Modern Recording

Upright Equalization

What advice can you give me in regard to using a Tapco 2200 equalizer properly, i.e., where can I find literature that can best guide me in the use of EQ? I am using the Tapco 2200 for a lounge show. One of the problems I have is that because of space limitations, I am forced to mount a speaker column with a horn attached to the top of the column



We put more thought into our leader than most manufacturers put into their tape.

One of the reasons Maxell has such a great following is because of our leader.

It has a built-in non-abrasive head cleaner designed to remove the oxide residue other tapes leave behind, without damaging your tape heads.

It also points out what side of the tape you're on (A or B) as well as which direction the tape is traveling. So it's almost impossible to make a mistake.

It even gives you a five second cueing mark, so you can set your recording levels without wasting tape. Or time.

Obviously, all the thought that went into our leader was designed to help you get more out of our tape.

So if you think our leader sounds impressive, wait till you hear what follows it.

maxell

CIRCLE 79 ON READER SERVICE CARD

Maxell Corporation of America, 60 Oxford Drive, Moonachie, N.J. 07074.

on its side. For best efficiency, must the column stand upright with the horn at the top? The speaker column I am referring to is the biggest model Shure makes that has a horn at the top.

—Ron Finizio
Entertainment Director
Hilton Inn of the Palm
Beaches
Riviera Beach, Fl.

In answer to your first few questions, we recommend that you do the following: If you do not already have the owner's manual (from the sound of your question, you must not), write to Tapco immediately and request one. They—and they alone—know best the parameters of their products and can instruct you as to how to get the most from your 2200 EQ. (Tapco is located at 3810 148th Ave. N.E., Redmond, Washington 98052, telephone number 206-883-3510.) In the meantime, look up some sound reference materials and see what they have to say on this subject. We suggest Home Recording for Musicians, *by Craig Anderton (Guitar Player Books, P.O. Box 615, Saratoga, Ca. 95070),* Howard M. Tremaine's Audio Cyclopedia *(Howard W. Sams and Co., Inc., 4300 W. 62nd St., Indianapolis, Ind. 46268),* The Recording Studio Handbook, *by John M. Woram (Sagamore Publishing, Inc., Plainview, New York 11803), or* Modern Recording Techniques *by Robert Runstein (also published by Howard W. Sams and Co.). Each of these presents specific information on equalization (equipment, definitions, usage, etc.) and can instruct you in the judicious use of this process.*

I believe you are referring to our SR 108 loudspeaker. This loudspeaker was designed as a linear array to be operated in an upright position. Dispersion patterns in the vertical plane (speaker upright) are somewhat narrow compared to the horizontal dispersion (speaker upright). When the SR 108 is put on its side, the above situation is reversed. The wider dispersion will now be in the vertical plane and the narrow dispersion will be in the horizontal. Problems that arise from this will be that most of the sound will be dispersed to the ceiling and the floor, not the audience, and the areas on either side of the ends of the speakers will have severely limited coverage.

I would recommend that if at all possible, you use the SR 108 in an upright position. If you have a low ceiling, you may be able to mount it on a wall (upright) with the horn slanted out from the wall. The horn should be aimed somewhere in the center of the "width" of the listening area and somewhere between the first quarter and the center *depth* of the audience area nearest the speaker. If you have a high ceiling, mounting the SR 108 *upside down* with the horn five to six feet off the floor will give good results. Here the horn should be aimed in the same manner, only this time it will be the woofer portion of the speaker that hangs out from the wall.

—Paul Bugielski
Manager, Professional Sound Products
Shure Bros.
Evanston, Ill.

The Significant Factor

What is the importance of the "damping factor" in amps? I heard somewhere that a high damping factor allows better speaker protection. How would the

difference between the new Heath amp, with a damping factor of 50-77 and a Phase Linear, at 1 K, affect performance in a system being run under all the standard professional use/abuse? Thanks very much for an answer to this—and for being there!

—Charles Ross
Cincinnati, Ohio

I think we'd best deal with the three questions here in order.

(1) What is the importance of "damping factor" in amplifiers?

(2) Does damping factor effect speaker safety?

(3) What might one expect from a system with damping factors of 50 - 77 as opposed to 1000?

Before we can discuss the "significance" of a characteristic of an amplifier, we must know exactly what the characteristic is and why it is built in. This topic has caused a great deal of confusion in other circles, so I'm going to try to lay it out in enough detail that we'll avoid all confusions. Well, that's a nice idea anyway.... Doing this will probably result in telling many of you more than you ever wanted to know about the amplifier-speaker inferface. For others, I'll be repeating what you already know. To both, no apologies.

"Damping" is a word that floats up at many points on the audio sea. We hear of damping room resonances. Everybody knows about those dampers on the piano that stop the strings vibrating. Both concepts are relevant here. We, the audio people, have stolen the word "damping" from the physicists, from descriptions of physical motions.

Everyone has experienced damped vibrations. Strike a tuning fork and listen to it slowly stop vibrating or place your fingers alongside a tine and stop it more quickly. What you're doing is increasing the damping of those vibrations. Your finger dissipates some of the energy from the vibrating body. The harder you push, the more energy withdrawn from the system. The more energy withdrawn, the faster the fork stops vibrating. What you're doing is increasing the damping factor.

If all of this seems silly and obvious, let me say that this is exactly what we do at the speaker-amplifier interface. We do it electrically as opposed to mechanically, but we're damping the physical motion of the speaker. Part of the electrical circuit for this damping is the output section of the amplifier. Of particular importance is the Z_s, the source impedance of the amplifier; that is, the impedance of the amplifier output to an AC signal in. Yes, it's necessary for good quality audio that the amplifier output actually see a signal from the speaker. Things starting to get interesting?

Just so we don't get too excited, let's slow down a minute and look at why we might want to dampen "vibrations," call them oscillations if you want. When we get the speaker "vibrating," say with a good kick drum, it doesn't want to stop. It wants to slowly die out just like the tuning fork. This means that the nice tight thud of the kick will sound like THUDUUUUUuuuuu. Further, that thud will shift in pitch to the natural resonance of the speaker rather than stay at the pitch of the sound which excited the motion in the first place. We want the speaker to respond fully to the audio of the amplifier, but we also want it to stop as soon as the audio stops. We need to dampen the

speaker vibrations.

This is accomplished by using a by-product of the speaker coil's motion as an aid to damping it. Every move that the speaker coil makes through the magnetic field of the speaker magnet generates a current. This alternating current appears at the speaker terminals. The speaker terminals are connected to the output of the amplifier. If the amplifier has a low source impedance, then that impedance acts like a short across the coil. When you put a very heavy load, like a short, on a generator it becomes much harder for it to move. The motion is damped.

The AC created by the speaker when it moves is shorted by the Z_s of the amplifier, thus quickly stopping any speaker movement. You can demonstrate this for yourself by tapping a speaker which is not connected to anything and listening to its sound. Most will thump with the sound being at the frequency of resonance. Now put a piece of wire between the speaker's terminals. Tap it. No thump, no sound at the frequency of resonance because you've damped the speaker motion with the wire. The wire represents the output of your power amplifier. Pretty neat, huh? (Don Davis might call it synergistic.)

To be completely honest, let me state and not explain that the amplifier-speaker interface is a more complex system. Amplifier impedance varies with frequency. So does the speaker's, but not the same way. Most amplifiers have load isolation networks at their outputs to compensate for those changes. Further, they exhibit differing characteristics with large signals than they do with small signals. In particular, the whole feedback circuit can be a part of the small signal response. The interface is a delightful combination of variables.

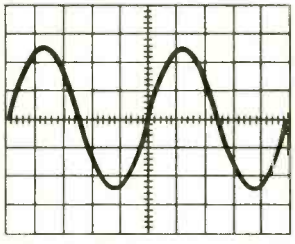
It should be obvious, though, that the lower the Z_s of the amplifier, the closer the system behaves to a straight wire short on the speaker's back emf output. The closer to a short, the greater the damping. A modern solid state amplifier can have an output impedance of .1 or even .01 ohms. This would mean a damping factor of 64 or 640 respectively. This corresponds to the Institute of High Fidelity (IHF) specification for damping factor as equaling $8/Z_s$ (Eight being the most common speaker impedance in high fidelity use today.) There are other methods of determining damping factor but the only one that you might guess was being used on a spec sheet would be the IHF, unless some other is stated.

Yes, I said that the output impedance (more correctly let's refer to the source impedance) of a power amplifier is a fraction of an ohm. They are not 8 ohms! They are designed to work into an 8-ohm reactive load. Forget the notion of impedance matching; it has no meaning in this context! Speakers bridge the outputs of power amplifiers. We are not concerned with maximum power transfer but with a constant voltage output from the amplifier.

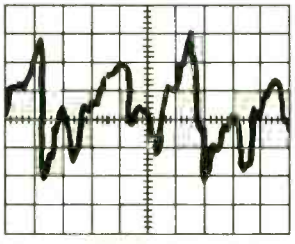
Okay, now we know what damping factor is and we know how to compute it if we know the amplifier's source impedance. We also know why the system works as it does, that is, how the amplifier not only drives the speaker but shorts it out, thus damping its resonances. Some might say that this "tightens up" the sound. That speaks to the "significance" of "damping factor." But not to what to expect in terms of sound from a specific amplifier when connected to a specific speaker. This

Many amps can deliver pure sound. The Sansui AU-919 delivers pure music.

Today's audio engineering has reached the point where you can select among a number of affordable high-power amplifiers that have virtually no "total harmonic distortion." That's good. But THD measurements only indicate an amplifier's response to a pure, continuously repeating, steady-state test signal (below, left). They don't tell you how the amp responds to the never-repeating, rapidly-changing transient waveforms of real music (below, right). And only an amplifier designed to reproduce the demanding dynamics of music signals can satisfy the critical audiophile. An amp like the Sansui AU-919.



SINE WAVE



DYNAMIC MUSIC SIGNALS

Because low THD without low TIM is like sound without music, the Sansui AU-919 is designed to respond well to both simple sine-wave test signals and also to handle the jagged, pulsive edges required for realistic reproduction of music—without imparting that harsh, metallic quality known as "transient intermodulation distortion" (TIM).

The Sansui AU-919 sounds better than conventional amps because Sansui developed a unique (patent pending) circuit that is capable of achieving both low THD and low TIM simultaneously.

Our DD/DC (Diamond Differential/DC)* circuitry provides the extremely high drive current necessary to use proper amounts of negative feedback to reduce conventionally-measured THD (no more than 0.008%, 5Hz-20,000Hz into 8 ohms at 110 watts, min. RMS) without compromising our extraordinary 200V/μSec slew rate, ensuring vanishingly-low TIM, as well. The power amplifier frequency response extends from zero Hz to 500,000Hz.

Since ultimate tonal quality depends on more than the power amplifier alone, Sansui also uses its DD/DC* circuitry in the phono equalizer section—where current demands are also particularly high—to prevent TIM. ICL (input capacitorless) FET circuits are used throughout the AU-919, and a "jump switch" is provided that will let you run pure DC from the Aux. input to the output.

Visit your authorized Sansui dealer today, and he'll show you a lot more that the AU-919 has to offer. Like twin-detector protection circuitry and our Penta-Power Supply system. Two-deck monitoring/recording/dubbing facilities. And a high-performance ICL/FET pre-preamp for moving-coil cartridges.

Then listen to the AU-919 with the most demanding music you can find. You'll hear the way the music should sound. Like music. Not just like sound.

*The Diamond Differential/DC, Sansui's (patent pending) totally symetrical double ended circuitry with eight transistors, is named for its Diamond-shaped schematic representation.

SANSUI ELECTRONICS CORP.
Lyndhurst, New Jersey 07071 • Gardena, Ca. 90247
Sansui Electric Co., Ltd., Tokyo, Japan
Sansui Audio Europe S.A., Antwerp, Belgium
In Canada: Electronic Distributors

Sansui

CIRCLE 111 ON READER SERVICE CARD

cannot be answered simply.

The amplifier speaker interface presents a number of delightful variables of which damping factor is only one, and indeed, utilizing current definitions for damping factor, it is only a great big number which has reference to the amplifier's source impedance. The other variables singly or as a group can dwarf the effect of damping factor. The best we can do is suggest that a number for damping factor that is lower than 20 probably should be questioned for serious monitoring purposes. Indeed, I'd be inclined to think that a number below 50 is quite unusual and might not represent the most thoroughly engineered amplifier output. That is a considerable generalization and hence cannot be considered a statement that is true or false or in any way absolute. Further, I make it in the context of "standard professional use/abuse." The literature will allow as "adequate" damping factors of 4, 16, or 20. No help there. That means you cannot accurately predict how amplifier H or amplifier P will sound in your system. You can only make a guess about how it will respond to rapid changes in level.

To pick up the second question, speaker safety. Yes, it should be obvious that if a speaker rings on because of some high-level transient and then it sees an additional signal you can get an adding effect which might get the speaker vibrating too far. But even a relatively low damping factor should provide a sufficient protection against that eventuality. Still no help in making you choice. Maybe "damping factor" as currently utilized in the industry is just another way to help sell amplifiers. It certainly is satisfying to have a damping factor of 200 or 1000. Is it better? Maybe.

—Ed Rehm
The Ken Nordine Group
Chicago, Ill.

The Mic Matters

I have a question regarding sound reinforcement in small applications such as nightclubs. Wouldn't it make sense, when equalizing a system with an analyzer, to use the particular mic that is being used on stage? Of course, this would only apply when all the mics being used on stage are the same, let's say, Shure SM 57s. If you use a calibrated mic to attain a "flat" response, you'll still have to contend with the peak characteristic of whatever mics you use during the performance (i.e. a "presence rise"). This could be why sound companies who painstakingly equalize a system still end up with a shrill sound when they start the show. They've compensated for the room and the speakers but not the mics.

—Charles Ross
"Somewhere on the road in Indiana"

Using the system microphone for analyzing and equalizing a sound system can be a very useful and time-saving technique since it will automatically compensate for the mic's frequency response. However, some hints for using this technique may be helpful.

First, know the bandwidth limits of your mic so that you don't over-equalize your system at these extremes. Trying to flatten the low and high ends of the spectrum using a mic that is really not capable of reproducing these frequencies will only result in excess hum and hiss without adding much sound quality.

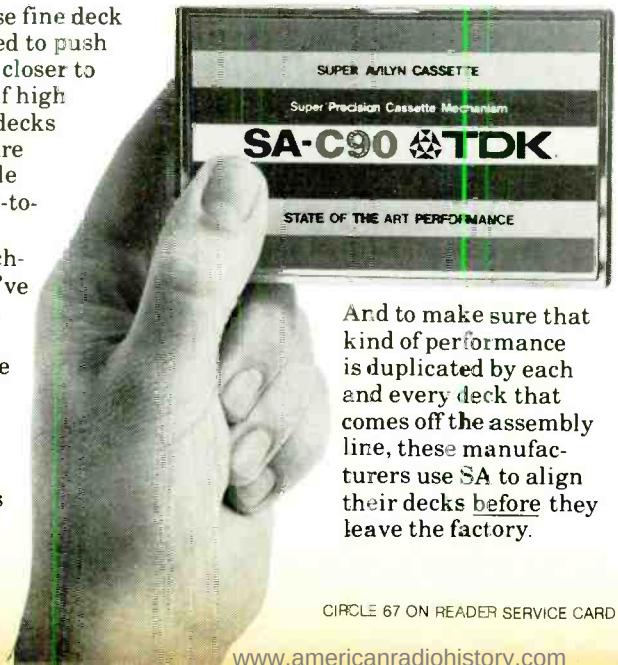
Second, certain mics do exhibit a presence rise in their response. This is

The standard bearers.

The high bias standard.

In the past few years, these fine deck manufacturers have helped to push the cassette medium ever closer to the ultimate boundaries of high fidelity. Today, their best decks can produce results that are virtually indistinguishable from those of the best reel-to-reel machines.

Through all of their technical breakthroughs, they've had one thing in common. They all use TDK SA as their reference tape for the high bias position. These manufacturers wanted a tape that could extract every last drop of performance from their decks and they chose SA.



And to make sure that kind of performance is duplicated by each and every deck that comes off the assembly line, these manufacturers use SA to align their decks before they leave the factory.

Which makes SA the logical choice for home use; the best way to be sure you get all the sound you've paid for.

But sound isn't the only reason SA is the high bias standard. Its super-precision mechanism is the most advanced and reliable TDK has ever made—and we've been backing our cassettes with a full lifetime warranty* longer than anyone else in hi fi—more than 10 years.

So if you would like to raise your own recording standards, simply switch to the tape that's become a recording legend—TDK SA. TDK Electronics Corp., Garden City, NY 11530.

TDK®

The machine for your machine.

*In the unlikely event that any TDK cassette ever fails to perform due to a defect in materials or workmanship, simply return it to your local dealer or to TDK for a free replacement.

CIRCLE 67 ON READER SERVICE CARD

sometimes desirable for added vocal intelligibility and thus should not be compensated for in the equalization.

One technique we have had great success with when equalizing systems, is using a laboratory quality nondirectional mic or one which has been selected for a particular analyzer. Set this microphone in the audience area connected to your analyzer. Then connect your system mic as it normally would be on stage. Begin by feeding pink noise through the system. The "raw" response of the audience area is observed through the analyzer. Flatten out the observed response with your equalizer, but don't equalize for a flat high frequency response. Listener preference usually dictates rolling off the system at 3 kHz at about 3 dB per octave. Average three or more typical audience locations.

Now, without turning off the pink noise, slowly bring up the system mic level until a feedback tone is heard. Note this frequency on your analyzer and attenuate the corresponding equalization control 2 to 3 dB. Repeat this until many feedback tones are heard together.

Last, check the system for intelligible and natural vocal quality as further equalization trimming may be needed. The final results should be immediately apparent; more system gain without feedback and a smoothed-out system response by having compensated for the system's microphone characteristics.

Why certain systems sound "shrilly" after the system has just been equalized, may be due to the fact that the audio engineer has equalized the system "flat." Generally in sound reinforcement, a "flat" system does tend to sound high-ended or "shrilly." Remember it is usually preferred to roll-off the high end for a more natural sound. In any case, always use your ears as well as your instruments in equalizing any system.

—James L. Warwick
Vice President
and David Tkachuk
Sound Engineer
Capron Lighting and Sound
Needham Heights, Ma.

Recommended Measurements

In the case of a center-channel subwoofer which gets everything below 200 Hz, can't I make smaller enclosures for my mid-bass speakers? If I can, what size? The speakers are Altec 416As, seeing from 200 to 800 Hz. The recommended bin for full bass range is a reflex-type, 40″ × 26″ × 18″.

—Colin Wedgwood
West Vancouver,
British Columbia, Canada

In order to provide you with accurate specifications for an enclosure for your Altec 416A mid-bass speakers, I would have to measure all of the parameters of the speakers. Unfortunately, because the 416A's have been discontinued for some time now, we did not have any workable speakers here which would be suitable for measurement. However, if all you need is 200 Hz low-end, you would do very well to build a 2 cubic foot box for them.

For your further information if you're looking for a good subwoofer you should try the Altec 416-8B speaker in a 10 cubic foot box with a 13½ square inch port. You can achieve 35 Hz with this without displacement limiting up to 57 watts.

—Jerry Siciliano
Research Engineer
Altec Lansing International
Anaheim, Ca.

PHASE LINEAR SETS THE STANDARD FOR HIGH POWER.

Current loudspeaker design theory takes for granted the availability of a high power reserve. An idea once considered frivolous by many. Today it is considered essential for the best possible reproduction of recorded material.

To reproduce a musical peak, a loudspeaker requires up to 10 times the average power being delivered. If the amplifier lacks a sufficient power reserve, it will clip, producing distortion and audibly destroying sonic quality.

The Phase Linear Dual 500-Series Two Power Amplifier is capable of delivering in excess of 505 watts per channel from 20Hz-20kHz into 8 ohms, with no more than 0.09% Total Harmonic Distortion. That's unsurpassed power for unsurpassed realism.

CONVENTIONAL OUTPUT TRANSISTOR.

DUAL-500 OUTPUT TRANSISTOR.

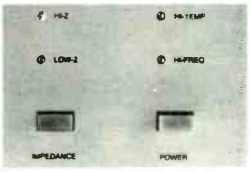
The Dual 500 utilizes an advanced design in output devices to overcome the problems associated with amplifier clipping at realistic listening levels. As a result, the power handling capability is greatly improved. In fact, the power semiconductor complement of the Dual 500 features the highest power handling capability in the audio industry.

A massive rear mounted extruded aluminum heat sink assembly protects the 36 output devices against overheating and includes a self-contained, thermally activated forced

air cooling system. You don't have to worry about over-heating under normal operating conditions.

Instantaneous indication of output activity is easily maintained with an exclusive 32-segment LED display,

while a special 4-segment clipping indicator warns of hazardous overloads. High/Low Impedance Operation modes are automatically, or manually activated for increased amplifier efficiency when using low impedance speakers.



HEARING IS BELIEVING.

See your local Phase Linear dealer for the most powerful argument for the DUAL 500: a demonstration.

SPECIFICATIONS

OUTPUT POWER: 505 WATTS MINIMUM RMS PER CHANNEL 20Hz-20,000Hz INTO 8 OHMS, WITH NO MORE THAN 0.09% TOTAL HARMONIC DISTORTION.

CONTINUOUS POWER: 1000Hz per channel, with less than 0.09% Total Harmonic Distortion:
8 ohms - 600 watts
4 ohms - 800 watts

INTERMODULATION DISTORTION: 0.09% Max (60Hz: 7kHz = 4:1)

DAMPING FACTOR: 1000:1 Min

RESIDUAL NOISE: 120uV (IHF "A")

SIGNAL TO NOISE RATIO: 110dB (IHF "A")

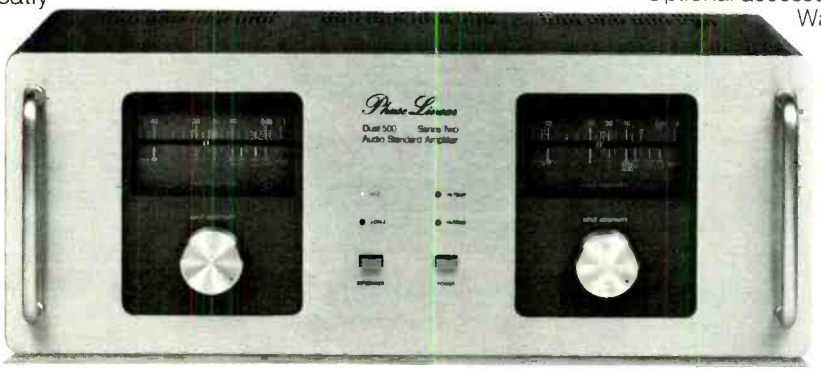
WEIGHT: 65 lbs. (32 kgs.)

DIMENSIONS: 19" × 7" × 15" (48.3cm × 17.8cm × 38.1cm)

Optionally available in E.I.A. standard rack mount configuration.

Optional accessories: Solid Oak or Walnut side panels.

AGAIN.

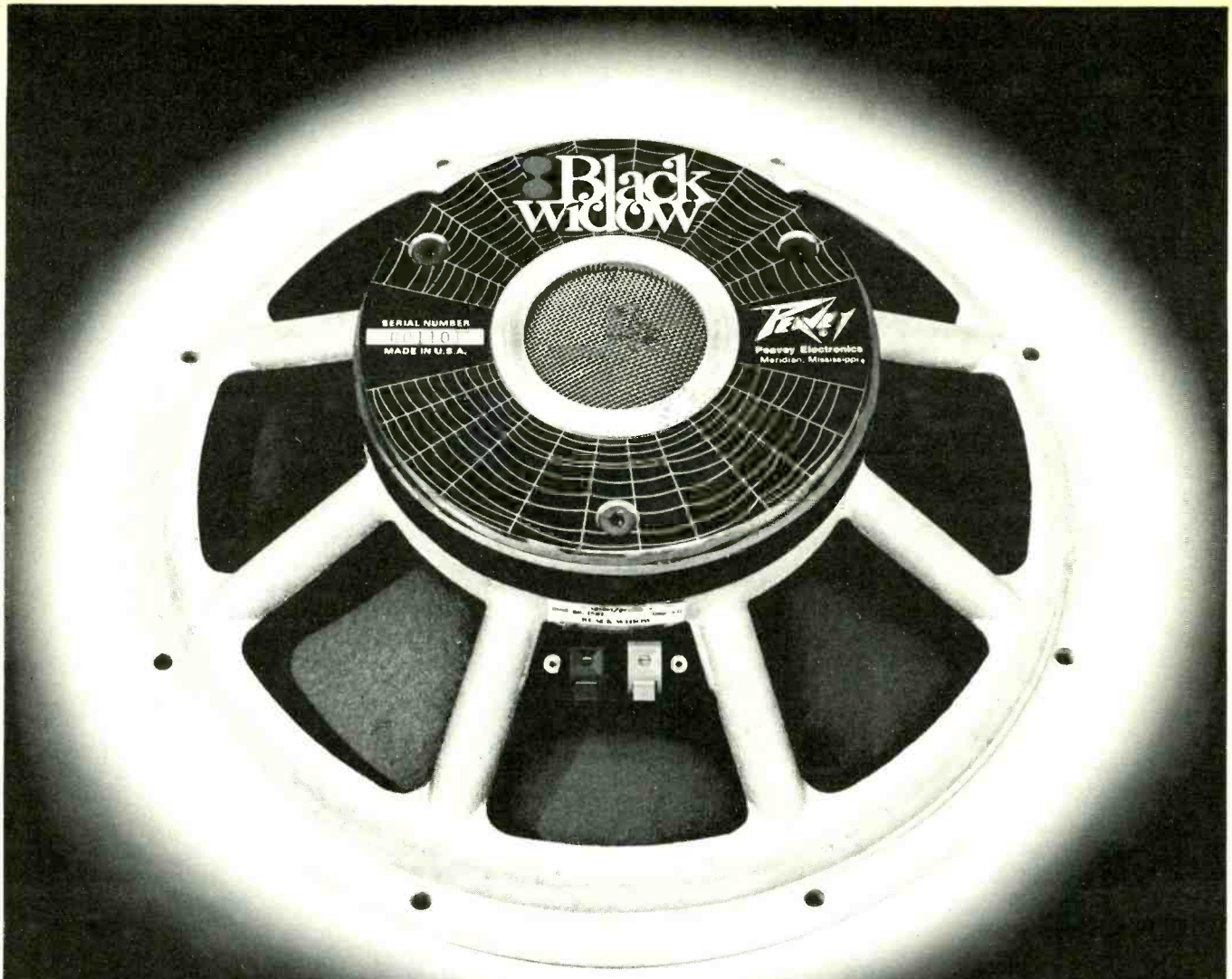


Phase Linear Corporation
20121-48th Avenue West
Lynnwood, Washington
98036

Phase Linear®
THE POWERFUL DIFFERENCE

MADE IN U.S.A. DISTRIBUTED IN CANADA BY H. ROY GRAY LTD. AND IN AUSTRALIA BY MEGASOUND PTY. LTD.

CIRCLE 97 ON READER SERVICE CARD



THE BLACK WIDOW

...because the best wasn't good enough.

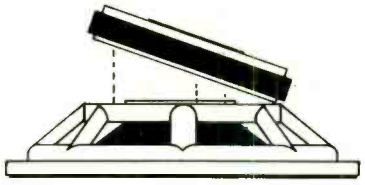
You're looking at one of the finest loudspeakers in the world...the Peavey Black Widow. They were created to fill a serious void,...speakers that could match the sophistication of today's sound reinforcement technology. For years we have employed the finest speakers from the most respected manufacturers in our equipment and through years of experience, have re-discovered the value of that old cliche', "if you want it done right, do it yourself." We did.

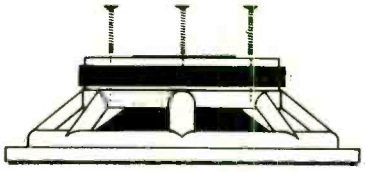
Since its introduction several years ago, the Black Widow has been praised by sound experts and musicians for its excellent efficiency, bandwidth, and power handling capabilities in applications that range from high powered concert sound reinforcement to studio recording.

The Black Widow's unique characteristics are the result of optimized procedures and concepts in design and manufacturing that provide a complete integration of form and function.

Unlike the other established manufacturers who are still building the speakers they designed back when a 100 Watt amp was a big deal, Peavey has designed the Black Widow with today's technology for today's high powered music.

The combination of a rigid cast-aluminum frame and high-efficiency magnetic structure is a feature found in many professional quality loudspeakers. What places the Black Widow Series far ahead of its competition is its field replacable basket assembly.





This feature, usually found only in high quality compression drivers, allows the user to be "back in business" in a matter of minutes, rather than days or weeks.

The high efficiency and high power handling capabilities found in the Black Widow make each model the best choice for its sound reproduction application. Again, what separates the Black Widow from other high quality transducers is its unique integral coil form/dome structure. When a loudspeaker is subjected to very high power levels, the voice coil temperature rises very rapidly, causing the loudspeaker impedance to increase. The result of this increase is a loss of efficiency. The Black Widow Series provides a most effective method of minimizing any impedance increases due to heat by utilizing the one-piece coil form/dome as a heatsink. Just as high power amplifiers use aluminum heatsinks to dissipate heat, the Black Widow coil form/dome is produced with low mass, high rigidity aluminum.

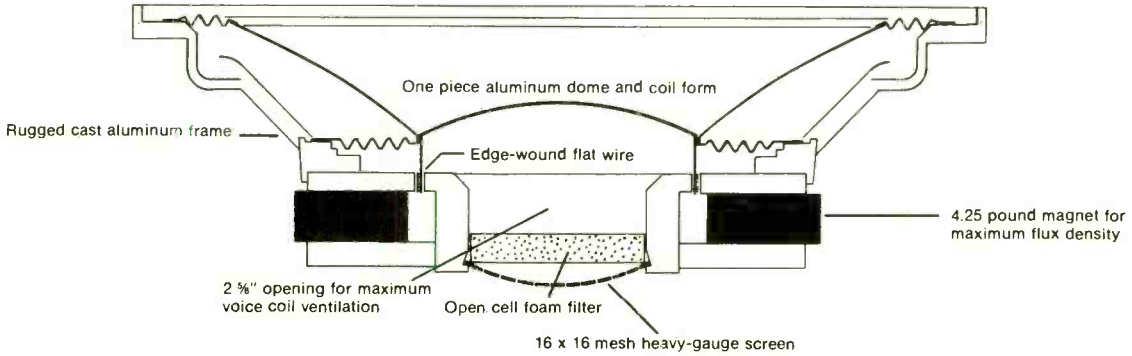
Each Peavey Black Widow is subjected to extensive quality control procedures to insure long field life and high reliability. The manufacturing methods employed by Peavey, such as numerical and computer controlled machining equipment, allow the Black Widow to maintain the close tolerances necessary for previously unattainable levels of quality and consistency.

Each Black Widow has a four-inch edge-wound aluminum wire voice coil to provide maximum energy conversion. The cone assemblies provide the required frequency response shapes with minimum weight and maximum structural integrity for high mechanical reliability. Each magnetic structure is fully removable and will provide minimum flux density of 12,000 gauss with very precise operating clearances. The magnetic structure uses a large rear vent to assist in further voice coil temperature control.

The Peavey Black Widow is now offered as standard equipment or as an option in most Peavey enclosures and will soon be available "over the counter" at selected Peavey Dealers.

The Peavey Black Widow,...for those who can't accept less than maximum performance and reliability from their speakers.

PEAVEY ELECTRONICS
711 A Street
Meridian, Mississippi 39301



MODEL NO.	DIAMETER	NOMINAL IMPEDANCE	POWER HANDLING CAPACITY CONTINUOUS	PROGRAM	SENSITIVITY 1w, 1m on axis	VOICE COIL DIAMETER
1201	12"	4/8	150W	300W	101 dB	4"
1501	15"	4/8	150W	300W	103 dB	4"
1502	15"	4/8	150W	300W	101 dB	4"
1503	15"	8	150W	300W	102 dB	4"
1801	18"	4/8	150W	300W	99 dB	4"

CIRCLE 89 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

ADS ACOUSTIC DIMENSION SYNTHESIZER

The company known as Analog and Digital Systems, Inc., has introduced its model ADS 10 01 which is described as a third generation digital time delay system. Derived from the previously introduced model ADS 10 Digital Time Delay, in the new model the built-in power amps and speakers are deleted. So, the new version is designed for listeners who already own suitable additional power amplifiers and speakers. Like the earlier model, the new ADS 10 01 represents a third generation digital time delay system of low noise (−83 dB), wide bandwidth (13 kHz) and low distortion (.3%) design. The model 10 01 features totally non-coherent outputs, high echo density and up to four delayed rear channel signals. The ADS 10 01 is designed to allow the listener to select the hall size and ambience preferred and offers controls for stage depth and act as a listener's seating distance from the proscenium. When used with electronic or disco music, the full bandwidth of the delayed rear channel offers complete delayed surround sound. The ADS 10 01 measures 15¾ inches wide, but it can be mounted in a standard 19-inch rack with optional brackets, or it may be equipped with optional walnut side panels. Tentative price is $675.

CIRCLE 11 ON READER SERVICE CARD

PHASE LINEAR RANGE EXPANDER

The new model 1000 Series Two from Phase Linear is a Dynamic Range Expansion/Noise Reduction System, actually an improved version of the Phase Linear Autocorrelator. The new 1000 Series Two has second generation low noise, high-slew rate integrated circuitry for quiet, distortion-free performance. According to Phase Linear, the device can remove up to 10 decibels of hiss, hum and rumble without affecting the music. It also is said to be the only noise reduction system with dynamic range recovery circuitry that can expand the dynamic range of unencoded material by a full 7.5 dB. As music is reproduced, the 1000 Series Two analyses

the incoming waveforms to find signals similar to a sine wave—a highly "correlated" waveform with periodic repetition. A guitar note is a correlated waveform; so is a vocal note. The 1000 Series Two electronically analyzes the signal to find fundamental musical tones and their harmonics. Where these are missing, there is no music and the 1000 Series Two can safely assume there is noise. If the 1000 Series Two identifies a fundamental waveform, it instantly orders one of its silent bandpass gates to open. If no music is present, the gates remain shut, removing a full 10 dB of hiss, hum and rumble. The new device is easy to interface with any stereo receiver, integrated amplifier or preamplifier and power amp combination, and is claimed to be usable in conjunction with Dolby and dbx systems. Suggested price is $349.95.

CIRCLE 12 ON READER SERVICE CARD

THREE NEW SPEAKER SYSTEMS FROM CELESTION

From Celestion Industries comes word of what apparently is their first major product change in some years. This refers to three new speaker systems known as the Ditton 662, the Ditton 551 and the Ditton 442. These systems all use direct drivers, and the 662 utilizes Celestion's Auxiliary Base Radiator (ABR) system for smooth bass response. The speakers also feature a unique crossover design which Celestion says "results in significantly better response and cleaner sound."

CIRCLE 13 ON READER SERVICE CARD

BEYER INTRODUCES NEW MICROPHONE

From Beyer there's word of a new hypercardioid dynamic microphone, the model M 88. Frequency response is listed as 30 Hz to 20 kHz, with an off-axis attenuation of greater than 23 dB at 120 degrees. Its EIA sensitivity rating is −144 dBm; output level is −51 dBm, and a special humbucking coil reduces hum sensitivity by 20 dB (to between 4 and 5 microvolts per Tesla). Impedance is 200 ohms, and the M 88 is said to be ideally suited for all recording studio and commercial sound applications. A special suspension system prevents transmission of hand-held noise. Unlike ordinary directional mics, the M 88 needs no vent hole on its shaft; as a result it can be hand-held without the possibility of altering its directional characteristics by covering the vent-hole. Weighing 10.8 ounces and equipped with a standard Cannon XLR 3-pin male connector, the M 88 is 7.8 inches long and has a head diameter of 1.9 inches.

CIRCLE 15 ON READER SERVICE CARD

NEW PIONEER CASSETTE DECK

The model CT-F800 cassette recorder from Pioneer features a three-head source/monitor design and the use of a dual capstan system and two-motor drive. Meters are quick-response, showing peak or average values. Bias and EQ selection are provided with a separate bias adjust control and automatic switching for CrO_2 cassettes. Dolby, of course, is built in, and the pushbutton system for the transport is described as soft-touch. The CT-F800 may be used with a separate electronic timer clock for preset recording and/or playback and it has full audio stop protection. Separate inputs are provided for left and right microphones. But since one level control is available, no input mixing on the deck is possible. Some more important specs listed include: frequency response with chromium dioxide tape +3 dB from 30 Hz to 17 kHz; signal-to-noise ratio with Dolby on is given as better than 64 dB.

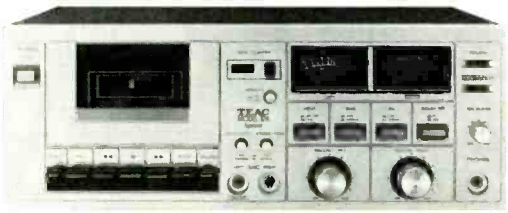
CIRCLE 14 ON READER SERVICE CARD

SOUND WORKSHOP REVERB SYSTEM

Sound Workshop's new model 262 Stereo Reverb System utilizes the latest in spring design for professional studio applications. The 262 features extended high and low frequency response and a fullness of sound which the manufacturer says was associated previously only with systems of substantially higher cost. The Sound Workshop 262 features a versatile and competent equalizer section. Two channels of EQ are provided with each channel allowing a ±15 decibel range over the high and low frequency bands. Frequency selection is fully sweepable from 50 Hz to 1000 Hz on the low band, and from 500 Hz to 10,000 Hz on the high band. The EQ bandwidth is optimized for proper contouring of the reverberant signal. The model 262 also allows dry/wet mixing (for broadcast and disco applications), full drive level up to 600-ohm loads, LED level indicators (for optimum dynamic range), active balanced input and matched bi-FET preamps for ultra-low noise performance. Mounting in a 3½-inch rack space, the unit sells for $700. It also is available with transformer balanced outputs and XLR connectors for $750.

CIRCLE 16 ON READER SERVICE CARD

SIMUL-SYNC OFFERED IN CASSETTE DECK



Something of TEAC's Simul-Sync for multi-track recording is incorporated in the firm's new model 124 Syncaset, a two-head recorder that employs tape/mic and line/mic mixing plus a cross-feed switch for blending. Says TEAC, with this system it is possible to record on the left channel first, then go back to the beginning and record on the right channel while listening to the left. This gives the recorder "one-man duet" and "one-man trio" capability. Compatible with all standard Philips-licensed cassettes, the TEAC 124 uses a servo-controlled DC motor and Dolby-B noise reduction. It has two-step bias and EQ switches, VU meters, memory rewind and other usual features. Specs include overall response of 30 Hz to 16 kHz, S/N without Dolby of 55 dB, wow and flutter of 0.07%. Price is $449.

CIRCLE 17 ON READER SERVICE CARD

NEW DIGITAL DELAY SYSTEM

From Advanced Audio Designs of Eugene, Oregon, comes word of the model D-250 time control which is said to employ the latest technology available in analog-to-digital circuitry. The D-250 may be bypassed without affecting the internal memory. Input display and gain control allow proper front end adjustment of sensitivity. Regeneration controls quickly adjust the amount of feedback and the hold function provides infinite repeat with no degradation of signal quality. The voicing function introduces random variations of the delay signal which is especially useful in creating a more natural vocal doubling effect. Delay time is selectable in one-millisecond intervals and is displayed in large bright numerals. Three-way mode switching includes straight delay, VCO effects and flange. The output stage offers mixing, phase reversal and gain control. The D-250 is priced at approximately $950.

CIRCLE 18 ON READER SERVICE CARD

UNI-SYNC DISCORAMA

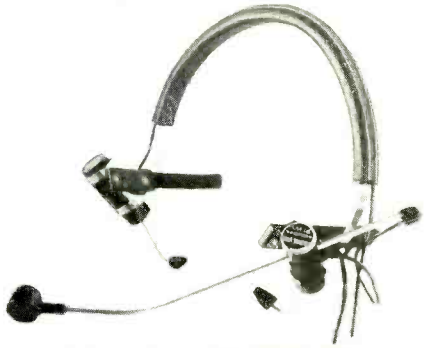
Uni-Sync has announced its Discorama, described as a professional mixing console. It accepts five individual sources (two turntables, two auxiliaries and one microphone) and is able to fade to or from any combination of these sources. The full list of specifications and features is available from the manufacturer.

CIRCLE 19 ON READER SERVICE CARD

SHURE DUAL MONITORING HEAD-WORN MIC

Shure announces a new lightweight, head-worn microphone with dual monitoring capability for use in various studio and remote professional broadcasting applications. The new Shure SM14 consists of a headband, unidirectional dynamic mic and two integral earphone assemblies to permit the monitoring of separate sound sources. Each of the twin earphones assemblies has its own transformer as well as its own phone plug.

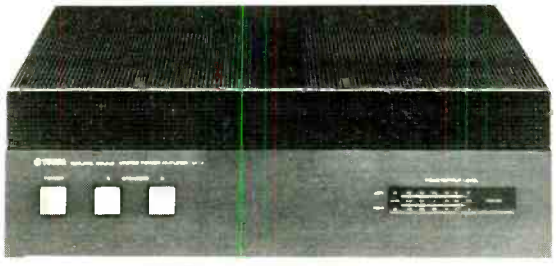
The SM14's dual monitoring feature provides several distinctive advantages over a single earphone system. Besides enabling the user to monitor two separate sound sources, such as program material and studio directions, the double receiver system helps prevent background noise interference. In addition, having two monitoring systems provides a reliable back-up should one system fail. A low impedance microphone, the SM14 allows extra-long lengths of cable to be used. The unit is constructed of stainless steel, aluminum and high impact thermal plastic, and it is mounted on a lightweight cushioned headband with an adjustable knob to permit it to extend or pivot to fit any face or head. User net price is $135.



CIRCLE 20 ON READER SERVICE CARD

YAMAHA SHOWS NEW AMPLIFIER

Yamaha's model M-4 is a new power amplifier with a rated output of 120 watts per channel into 8 ohms. The unit is said to employ new power transistors, which Yamaha itself developed. They are driven at low impedance in the complimentary push-pull output stage and exhibit no disturbance in current waveform, even under severe operating conditions. The power supply is said to be unusually stable even at very low frequencies, a very important point in a DC amplifier such as the M-4. A low impedance detecting circuit is activated the moment the load impedance falls below 2 ohms. In addition, there is a DC detection device that cuts off the speaker from the amplifier the moment a DC voltage is generated to the amplifier's outputs. The M-4 includes a peak level meter, an overdrive indicator, AC/DC input switch selection, front panel speaker switches, separate left/right input level controls, and is housed in an "elegant enclosure." Suggested retail price is $650.



CIRCLE 21 ON READER SERVICE CARD

DUST COVER POSITIONS RECORD BRUSH

A Lenco brand rotating brush for cleaning disc recordings and simultaneously drawing off electrostatic charges that cause surface noise has been announced by Neosonic Corporation. The device mounts on the dust cover of a single-play turntable so that when the cover is lowered, the brush is automatically positioned at the record groove's outer edge. As the record rotates, the brush does its job. No liquids are used. The charges are conducted to the turntable spindle via the brush and a spring contact. Associated with this system is a conductive foil pad to be placed under the disc for removing charges on the flip side. Price is $19.95.

CIRCLE 22 ON READER SERVICE CARD

TAPE TOPICS

Metal recording tape, now coming off production, is being researched by the 3M Company for other applications. One other area, says 3M, is in digital audio tape where large quantities of data must be stored which include both program information and data for error correction. This requirement means that larger quantities of conventional oxide tape are needed to record digitally as compared to what is needed for analog recording. Several hardware manufacturers have satisfied these requirements by developing equipment that would record at high speeds—up to 120 inches per second. Most analog reel-to-reel studio recorders use 7½ or 15 ips and audio cassettes, of course, are recorded at 1⅞ ips. Other equipment manufacturers have taken a different route, by developing equipment that records several signal tracks—two to eight—per channel of information. In both cases high tape speeds or recording several tracks more tape is used. According to the 3M Company Scotch Metafine represents an attractive alternative because the tape itself is capable of much higher recording densities than ferric or chromium oxide tapes. This is a result, says 3M, of the high coercivity and retentivity characteristics of the metal tape. Says a spokesman, "A metal particle digital tape will allow equipment manufacturers to reduce the tape speed, and/or upgrade the quantization levels of the digital system. This means more efficient use of recording tape without loss in sound quality."

Ampex has announced its own newly developed metal particle tape to be offered at first in audio cassette form. Says Ampex, it improves high frequency saturation capability by more than 10 dB over standard gamma-ferric oxide when measured at 10 kHz, and more than 5 dB over high-bias formulations. The improvements are even more dramatic at even shorter wavelengths.

DuPont has entered the home video market with video cassettes for home recorders used in the Betamax format. Under the "Primetime" label, the DuPont video cassettes will be made with a magnetic coating of chromium dioxide especially developed for home video use.

For its part, Maxell Corporation has made its debut as a video tape supplier with its first branded VHS products for home use. The tape on Maxell's new VHS video cassettes is an exclusive Epitaxial crystal formulation, consisting of a cobalt crystal encapsulating a ferric oxide particle core.

MUSICAL NEWSICALS

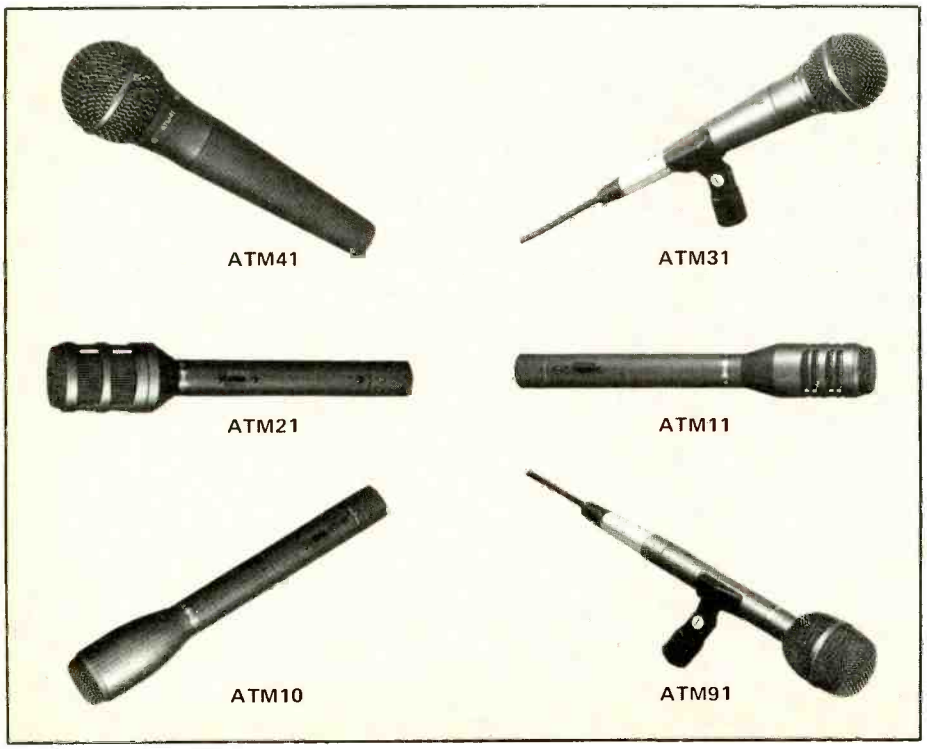
SOUND REINFORCEMENT EQUIPMENT

Audio-Technica U.S., which is perhaps best known for its phono cartridges, has introduced a new line of microphones for the performing musician. The new line is called the Artist Series and comprises six models, three of them designed specifically for vocal use and three of them designed primarily for miking instruments. The three instrument models are the ATM10, an omnidirectional fixed charge condenser mic; the ATM11 which is a fixed charge condenser mic with cardioid directional characteristics; and the ATM21, a cardioid dynamic microphone. All three of the instrument mic models are available with an optional full suspension shock mount. The three vocal models, the ATM31, ATM41 and ATM91 are all cardioid pattern mics; the ATM41 is a dynamic model while the other two models are fixed charge condenser designs. All three vocal models feature a special triple-layer windscreen for ruggedness—a heavy wire outer mesh, a finer inner mesh and a fine brass screen are combined to withstand rough handling, and the assembly is soldered rather than glued for extra ruggedness. The "fixed charge" construction used by Audio-Technica is a type of electret construction which places the permanently charged (electret) element in the back-plate of the capsule rather than the diaphragm, allowing the use of a thinner, lower mass diaphragm for lower distortion and improved frequency response and transient reproduction.

CIRCLE 2 ON READER SERVICE CARD

Electro-Voice has introduced a new shock-mount mic stand adapter, designated the 313A. The new stand clamp has the outward appearance of a standard type unit, but actually is a full-suspension shock mount. The unit is manufactured from metal and polycarbonate plastic with four replaceable urethane bands suspending the microphone itself. The microphone is retained via a metal latch or a supplied set screw for temporary or semi-permanent microphone mounting. The clamp is made to accommodate all microphones whose barrels are approximately three quarter-inch in diameter, which includes the majority of E-V's mic models. The small size and unobtrusive appearance of the 313A make it ideal for broadcast and sound reinforcement applications where conventional, spider-type shock mounts might be ruled out from an appearance standpoint, and also facilitates mic placement in the studio.

CIRCLE 3 ON READER SERVICE CARD



MUSICAL INSTRUMENT PICKUPS

All-Test Devices Corp. manufactures a full line of musical instrument transducers for a variety of acoustic instruments. Their line-up includes models for guitar, banjo, violin, string bass, cello, drums, flute, reed instruments, and a four-pickup system with six input preamp/mixer for piano, most of which attach to the instrument with special adhesive. Of special note are two models for acoustic guitars. The first is the PK-1 permanent mounting transducer; this model mounts in a single three-sixteenth-inch hole which may be located in the bridge or anywhere the wood is at least one-quarter-inch thick. The transducer was specially designed with a mylar and silicon capsule for user installation in a matter of minutes using the drill bit supplied. The other model is the PP-1 semi-permanent transducer

which is built into a bridge pin. One pickup can be used for full range reproduction, or two pins may be used for the first and sixth strings for greater control of the tonal balance. The pins come in two sizes and in black or white to suit most any guitar.

CIRCLE 4 ON READER SERVICE CARD

Ibanez has announced another new addition to their line of magnetic pickups for electric guitars. The latest model is the V-2, which is a high-output pickup with reinforced upper midrange and slightly rolled-off bass response for a bright, cutting tone quality. The high output of the pickup was achieved by increasing the number of turns of wire in the coils, and this was combined with a specially-designed magnet to produce a tight, punchy sound that is effective even at very high volume levels. The V-2 is a direct physical replacement for any large humbucking pickup. It is an open-coil design which comes in black or cream-colored coils and mounting rings. The V-2 is also available with the Ibanez Tri-Sound option, which gives the guitarist the choice of single-coil, humbucking or reversed-phase operation with the touch of a mini-toggle switch.

CIRCLE 5 ON READER SERVICE CARD

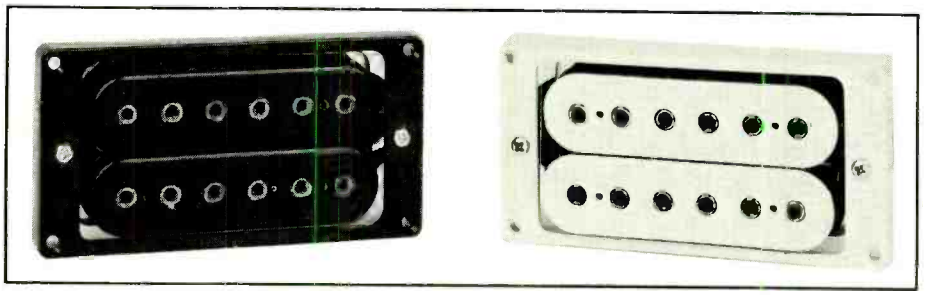
MUSICAL INSTRUMENT AMPLIFIERS

RolandCorp US has three new guitar amplifier models, two compact models and a top-of-the-line addition to the Jazz Chorus series of amplifiers. The two compact models are the Cube 20 and the Cube 40 which feature a single 8-inch and a single 10-inch speaker, respectively. Both amps include normal and overdrive input channels with input volume, and bass, mid and treble tone controls, plus reverb and master volume controls. An additional feature of both models is a headphone jack which disconnects the speaker when a plug is inserted for private practicing or tuning. The Cube 40 also features line in/line out jacks for connecting effects devices or for use in recording. The new Jazz Chorus model is the JC-200, which delivers 200 watts RMS into a pair of twin 12-inch speaker cabinets. Each of the JC-200's two channels has high and low sensitivity inputs, bright switch, and volume, bass, mid and treble controls. The effects channel of the unit includes the chorus-vibrato section, a distortion control and reverb, each of which is selectable or cancellable by footswitch according to front panel footswitch controls.

CIRCLE 6 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

A problem that is all too familiar to multi-keyboardists who use a Fender Rhodes piano is how to safely balance another keyboard on top of the Rhodes' curved top cover. Linett Unlimited has come up with a very useful solution in the form of the Flat Top—a replacement top cover for either 73-note or 88-note Fender Rhodes pianos. In addition to being flat, which is a great advantage in itself, the Flat Top is constructed from three-eighths-inch plywood which is much stronger than the original molded plastic top. The plywood of the Flat Top is covered with black Tolex and is a perfect match for the rest of the piano. The Flat Top fits exactly in place of the original top; no modifications are required and the original travel cover still fits perfectly.



CIRCLE 7 ON READER SERVICE CARD

Star Instruments, Inc. has announced two useful, new accessories for their Synare electronic drums. First is a new stand which adapts a Synare 3 electronic drum for use as a foot-pedal operated bass drum. With the new BD-2 bass drum stand it is now possible to assemble a totally electronic drum kit except for cymbals. Any Synare 3 unit may be used for the bass drum thanks to the wide range of sounds available from the units; with proper tuning a Synare 3 can sound virtually identical to a standard bass drum, or a nearly limitless-range of special effects may be produced. The other new product from Star is a road case for safe and convenient handling of the Synare 3. The case is built from furniture grade plywood covered with formica-type laminate. Heavy aluminum edging strips and steel corners add strength and durability to the case, and all hardware has been selected for reliability and ease of operation. In addition to the model which accommodates two Synare 3 drums, two other models are available to fit the Synare 1 and Synare 2 percussion synthesizers.

CIRCLE 8 ON READER SERVICE CARD

Barcus-Berry Sales Corp. has a new, low-cost model pickup for fretted instruments. The B-B Jr. is a professional-quality transducer which is said to yield optimum performance without the necessity of experimenting with physical placement of the pickup on the instrument. The B-B Jr. has a high output level which is sufficient to drive most musical instrument amplifiers. The output impedance of the unit is high (recommended load impedance is 1 megohm), but a full range of studio quality preamplifiers is available if it is necessary to drive a lower impedance load. The B-B Jr. is said to have a natural-sounding, flat response from 5 Hz to 50 kHz and a virtually unlimited dynamic range.

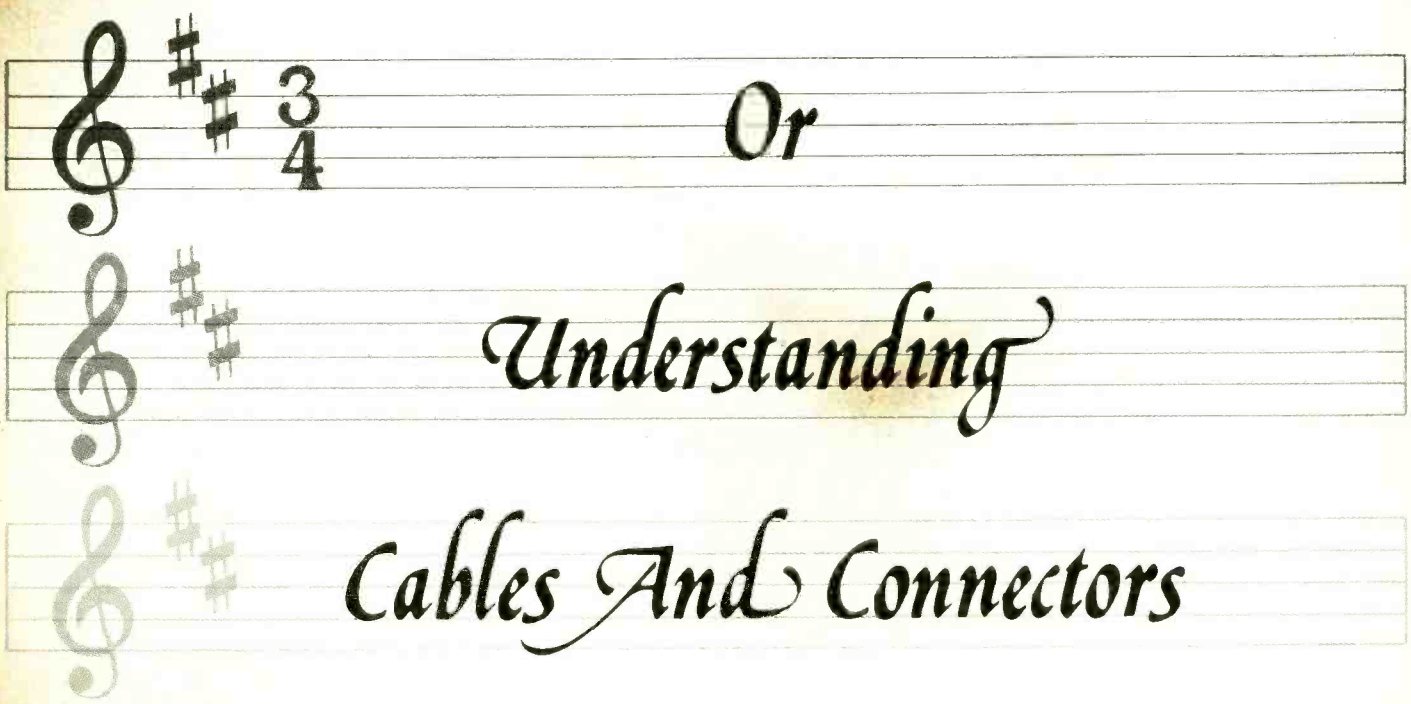
CIRCLE 9 ON READER SERVICE CARD

Fender Musical Instruments has expanded its line of Fender Strings with the introduction of the new Fender Studio Bass Strings. These new strings are double-wound with nickel windings over a hexagonal core of Swedish steel, which Fender has found to yield remarkable responsive and consistent strings. The hexagonal shape of the core grips the windings and allows Fender to achieve a very tight, consistent wrap despite the hardness of the nickel wire used. The result for the bass player is a string that is more brilliant, more responsive throughout the string's tonal range, longer lasting and more consistent from string to string. Fender Studio Bass strings are available in both flat wound and round wound types to suit all playing styles.

CIRCLE 10 ON READER SERVICE CARD

Spaghetti Sonata

Words and Music by
Brian Roth



The cables and connectors utilized in a P.A., recording, or even a stereo system are truly the arteries of the sound equipment. If any of the cables fail, the signal path is broken and the audio program will usually die, often accompanied by noisy comments from the loudspeakers. Consequently, a good understanding of this critical portion of a sound system is very important.

This dissertation has been divided into two sections: (1) Cables; and (2), Connectors. Within each of these parts, there is a further division of information to separately examine the requirements for both high- and low-level cables and connectors. By way of definition, low levels describe signals of negligible power such as that produced by microphones, guitars (and similar instruments), as well as the line level inputs and outputs of mixers and peripheral gear. High level signals are found in loudspeaker or AC electrical power lines; the voltages and currents are much greater compared to low level lines.

So, continue on, dear reader, and hopefully your questions concerning the subject of cables and connectors will be answered.

Cables

Shielding. Any cable carrying low-level audio signals requires an effective electrical shield. Without one, all sorts of nasty noises from motors, dimmers, CB transmitters, cash registers, etc., can intrude upon the vulnerable audio program that is trekking through the cable. *It is most important to remember that cables carrying microphone or phono pickup signals MUST be shielded if electrical interference is to be avoided.* This same rule applies to line-level wiring as well, although the shielding requirements are not as rigorous due to the stronger signal levels involved. The catalogues of the wire manufacturers often give a specification for shielded cable that is called "percent of shield coverage." This number can range from 0%, for completely unshielded wire, to 100% for cable with a foil-type shield. Generally speaking, the higher the "percent of coverage," the better the shield.

Now, that last statement may suggest that foil shielded cable is the ultimate answer for all signal level applications. This is not always the case. The foil, which is actually a thin film of plastic with a layer of vaporized aluminum deposited upon it, is not particularly strong. It is not intended for tough service such as hand-held microphone connection cords, guitar cords, or any other cord that is subject to lots

of flexing, twisting or stomping.

The majority of the multi-conductor "snake" cables have individual foil shields around each signal-carrying wire (or, more generally, around each *pair* of signal carrying wires for low impedance, "balanced" microphone lines). Due to the foil shields, it is important to "Be nice to your snake!" Don't roll it up too tightly when packing it, and never put undue mechanical stress on it (such as backing over it with the forklift). Snake cables are much more expensive to replace than an individual mic cable since there is the equivalent of nine, twelve, fifteen or even twenty-seven separate mic cords crammed into one jacket. The convenience of a snake is obvious, so just give it proper care and it can be used for many years before the foil gives up the ghost.

So, if foil-shielded cable can't take on-stage rock antics, what should be used for mic or guitar cords? The answer is found with a cable that has a braided or spiral wrapped wire shield. Both of these shields consist of a multitude of tiny bare wires that are woven or twisted around the inner signal carrying wire(s). This arrangement results in a shielded cable that is more flexible and durable than foil types. The percent of shield coverage can range from below 75% up to 100% for the fancier grades of cables using braided shields.

Ah, you say, why not use 100% coverage braid instead of foil in a snake? First, it is almost impossible to find. Second, foil shielded cable is much less expensive to produce. If you think that the price of foil shielded multi-conductor cable is high, just be sure you are sitting down before checking the price of cable with individual braided shields around each pair of signal-carrying wires.

An alternative design for multi-conductor cable uses only one shield that surrounds all of the pairs at once. This will help keep the garbage out of the signal just as any equivalent shield would, but it does not protect against crosstalk between all of those wires. If only microphone level signals are being transported to the mixer, crosstalk between mic lines will probably be negligible. However, it is common practice (although definitely not the best) to send the line level output from the mixer back down the snake to the on-stage amplifier racks. If this is done with the "one-overall" type of shield, it is very possible that the strong signal returning to the stage will "leak" into the mic lines that are running to the mixer's input. This can cause the board to go into oscillation and generate some really rude high frequency noises that are guaranteed death for tweeters. It is good policy to keep the stronger line level signals away from mic level wiring. If possible, don't use the mic snake for the line level return to the stage, particularly if the cable doesn't have individual shields around each pair of wires.

For the rough service requirements, the choice has been narrowed down to spiral wrapped or braided shields. As mentioned earlier, the better quality braid can give 100% coverage; the spiral wrapped shields don't go much over 90%. Additionally, spiral shields can "bunch up" inside of the outer jacket and thus reduce the shielding effectiveness. This problem depends upon the overall construction of the cable, so it's more of a problem with some types of spiral wrapped than with others. Also, braided shields can have as low as 80% coverage due to the type of weave. These can be successfully used for microphone applications, but it's better to play it safe by using cable with 95 to 100% shield coverage. So, for best results, use a cable that has a braided shield with a high percent of coverage for the cords that attach to the microphones or guitars. The other styles of braid or spiral shielded wire with lower coverage will be quite acceptable for line level connections—mixer outputs, limiters, equalizers, etc. Of course, there's nothing wrong with using 100% shield coverage cable for *all* applications, that's really playing it safe.

High shielding effectiveness doesn't come cheap. Premium mic cable is much more expensive per foot than foil shielded wire, plus it is much more difficult to attach the shield to an audio connector. This is because the ground termination for foil shields is made via a separate, bare "drain" wire that is in electrical contact with the foil throughout the length of the cable. All one has to do is strip the outer plastic jacket, tear the foil away, and solder the bare drain wire to the connector. This in direct contrast to braided shields that require much more preparation prior to soldering. Spiral shields are somewhere in between foil and braid as far as ease of shield termination is concerned. Due to its connection simplicity, foil shielded wire is extremely popular for applications that do not involve rough service.

Loudspeaker connection cables seldom require a shield. An exception might be in dense RF (radio) fields such as broadcast stations. Otherwise, shielding is an extravagance for speaker lines. AC power cables are also unshielded in most applications. If the wire is run through a conduit, this can be called a type of shield, but obviously we are not too concerned about interference getting *into* the power cable. A conduit can help reduce the hum field that radiates *from* all AC power lines. Again, certain situations that involve strong RF fields may dictate the usage of shielded power coverage, but this is relatively unusual. Braided shields are generally utilized in this case. One thing to keep in mind: *Never* use regular microphone cable for AC power transmission. It is not rated for the voltages and currents involved and can go up in a blaze of glory. So, to those people that use foil shielded cable for 120-volt power, I suggest that your insurance premiums had better be paid up, and keep an eye out for the fire marshall!

Wire size and stranding. The metal wires in any cable are rated by *gauge*; the smaller the number, the larger the diameter of the current-carrying conductors. Low-level audio cables can use small wire gauges since only minimal amounts of power are involved. #22 AWG (American Wire Gauge) is acceptable for mic and line level, but larger wire (such as #18) will usually have greater mechanical strength. If the lead vocalist enjoys swinging the microphone around by its cable, #18 AWG wire is in order (along with secure mechanical attachment of the cord to the microphone's connector) to prevent waffle prints on the forehead of an adoring fan.

Loudspeakers require high amounts of power and a large diameter wire is necessary to avoid power loss though the wire. #16, #14, #12 and #10 AWG are common sizes for loudspeakers cables. Within limits, bigger is better, particularly for lines that interface the woofers in a biamped speaker system; a large cable can improve the "tightness" of the bass range.

AC power cords should never be smaller than 18 gauge, and only then, if the power requirements are very low. An amp rack housing half-a-dozen 400-watt power amplifiers needs something like #12 or #10 AWG connection cords to avoid overheating the cable as well as to reduce the voltage drop

through the cable.

The next decision to be made concerning the construction of the cable is stranded versus solid wire. If the wiring is being installed into a conduit or wall and consequently is not going to be flexed, solid wire is acceptable. Otherwise, use stranded conductors. When buying wire always check on the stranding; it is common for a wire manufacturer to offer two versions of an otherwise identical wire—one stranded and one solid.

A very large sound system may have to use really hefty wire to handle the high AC power demands. #6, #4, and larger are not uncommon for the main power feed to the amplifiers. For these larger sized systems, some people use "welding cable" which is an individual, standed conductor with a rubber insulation jacket. It is quite flexible considering its size and is durable. Just be certain that the insulation is rated for the voltage involved (120 or 240 volts AC). 600 volts is the usual rating for wiring that carries AC power to give a good margin of safety.

Insulation and jackets. Thanks to "better living through chemistry," there is a multitude of choices for insulation materials that includes vinyl, synthetic rubber, and, yes, even Teflon. In addition, there is a large range of insulation thicknesses available. Unfortunately, the variables make it impossible to choose one type for all situations. Rubber is recommended for heavy-duty requirements such as microphone, guitar and power cords due to its flexibility and durability. A plastic insulator, which is often less expensive, is usually stiffer and does not have the abrasion and chemical resistance of high-quality rubber insulation. Of course, there are some nifty types of plastic available that can outperform rubber, but these tend to be a high-priced proposition. Teflon is a very expensive material to use for insulation, but it has superior heat resistance. A less expensive choice that still has a high melting point is irradiated PVC. For unknown reasons (except perhaps its higher price), it is not commonly used in audio cables. This is unfortunate since irradiated PVC insulation won't melt and shrivel like ordinary vinyl when soldering connectors to the wire.

After selecting the material for insulation, the next step is to determine the required insulation thickness. A heavy jacket gives more mechanical protection and generally allows higher operating voltages, but naturally tends to be less flexible. So, the choice becomes a compromise between electrical or mechanical protection and handling characteristics. A popular type of loudspeaker connection cable is designed with two individually insulated conductors with an overall plastic jacket. While it is not quite as durable as high-quality rubber jacketed cord, this style of cable offers better protection of the current-carrying conductors than ordinary lamp or "zip" cord.

Cable capacitance. A capacitor (or, as the automobile mechanics say, a condensor) is made up of two metal surfaces in close proximity with insulation between the two. It should be apparent that two (or more) insulated conductors in a cable will form a capacitor of sorts. So, what does this mean in the real audio world? At very low audio frequencies, the effect of this capacitance can usually be ignored. However, the cable capacitance will result in something equivalent to a "short circuit" between the conductors at very high frequencies, and thus can cause loss of treble response. Cable capacitance is the main reason why high impedance microphone or phono pickup lines should be kept very short. It is advisable to use cable with a low capacitance characteristic (given in picofarads, or pF, per foot) in these instances. Better yet, don't use high impedance lines at all!

Capacitance between conductors or a conductor and shield can affect lines opening at low impedances if the cable run is long. This is due to the fact that the total capacitance between conductors in the cable is directly proportional to the length of the cable. In other words, the longer the cable the higher the capacitance. This may cause instability in the output circuits of the equipment that is feeding audio down the long line; it just depends upon the circuit design.

Since most of the popular foil shielded cables have relatively high capacitance per foot, try to keep these as short as possible (that's *always* a good rule, anyway).

"High Definition" loudspeaker cables. The latest rage in the hi-fi salons is loudspeaker cords with unusual physical construction. Some look more like an orderly sheet of spaghetti than a speaker cord. All are claimed to improve the transparency, punch, and so forth of the audio signal. Other than their appearance, what sets these cables apart from more conventional styles is the fact that they have been designed for low resistance, capacitance, and/or inductance. Any of those three factors can affect the ability of the cable to transfer the signal to the speaker without degrading the audio. However, some observers have suggested that switching to a large gauge cable of "normal" construction will give much the same improvement, which is often very minor. The improvement in sound quality will be most noticeable if you are changing from something like 22-gauge "speaker" cable (that cheap stuff with the clear insulation) to the "super" cables or large (#12) regular cable.

Most of the new cables appear to be rather delicate. It is very doubtful if they will hold up under rock 'n' roll stage conditions. They may be beneficial for connecting studio or control room monitors. There have been some reports that the new breed of cables can cause power amplifiers to fail due to the capacitance and inductance being *too* low for that particular amplifier. This would be appear to be more the fault of the amp rather than the cable. Nevertheless, consult the amplifier's manufacturer if the special speaker cables are to be used.

Connectors

Audio connectors for low level applications. The most common connectors in this category are the RCA phono plug, the quarter-inch phone plug and the "Cannon" or "XLR" type.

The phono plugs and jacks are the least expensive and also the least reliable. They rely on a friction fit, and the center "hot" pin is connected before the ground sleeve when inserting the plug (and the reverse upon removal of the plug). If your power amplifiers are activated, this sequence of connections can cause loud buzzes and screeches (and maybe blown cones).

Phone plugs are definitely higher class; they "click" into position when inserted in the jack, and the connection sequence causes the grounded sleeve to be connected prior to the "hot" tip of the plug. The result is fewer surprising blasts from the loudspeakers. A phone plug is also usually much easier to attach to a cable than an RCA plug.

The 3-pin "XLR" (which is actually the series designation for connectors made by Cannon) is the Cadillac connector for low level audio. They are readily available from manufacturers

Three hours set-up time. Four amps, a bunch of mics, monitors, etc...and what did you get?

Spaghetti.

Now you're laughing. But maybe the fact that you can relate to this picture means something. Chances are that if you've ever played in a band, engineered, or had the pleasure of setting up equipment you probably know this scene all too well. The fact is, though, it's not funny when you consider that the groups livelihood depends on that equipment. And when you think of all the money you spent on mics, amps, monitors, and the rest of it, you can get sick. It's too bad, but the entire system is no stronger than its weakest link.

As the complexity of your music system increases so can your headaches. The overall success of your show can be jeopardized by bad connectors, cables and a big mess on stage. And what about the time element? A well organized act not only looks but sounds better. Being efficient saves time, and that saves money.

At Whirlwind, we have been long aware of the music industries needs for quality cables and connectors. We are continually developing new designs and products for the market we serve.

Whirlwinds name was built on the strength of our basic cord line. We were first with a rugged cord, which is fairly priced, and yet carries a full two-year warranty. The cable is custom made to our specifications by the worlds best manufacturers. They are durable, and stay silent for years. Our Snake and Ultra-Snake cords are available at different standard lengths or at custom lengths by request. The Cobra is the first truly noise

free retractile cord made from thick and tough cured neoprene, covering a double shielded cable which is tipped with various connectors, depending on your needs.

Our famous Medusa was developed as the result of years of experience in order to bring simplicity to the multiple wiring of PA systems. They are currently being used by many of today's top touring bands whose schedules and professional reputations demand the utmost in equipment performance and reliability. They are available in nine basic configurations or custom made to your specifications for flexibility.

Our connectors represent the finest in American-made hardware technology to provide positive contact and give the greatest strain relief. Our own Whirlwind connectors are constructed of solid brass with stainless steel tips for definitive contact and durability. Our **SK Series Cords** are designed to provide positioning flexibility for amplifiers and speakers. They can take the abuse of *on the road* conditions. Our **MK Series Microphone Cordsets** can withstand years of the kind of treatment that mic cables get, and still pass noise-free signal consistently. There is an MK cord for almost every mic, in high and low impedance.

The Whirlwind Imp can be successfully used to match impedances and for direct line access. We also distribute assorted accessories to meet your requirements: including stage tape, heavy duty AC cords, cable ties, speaker load protection systems, and many various adapters.

Whirlwind products are available at leading music stores and pro-sound outlets. Please write us for our free catalog.

whirlwind

Whirlwind Music, Inc.
P.O. Box 1075/Dept. B
Rochester, New York 14603

CIRCLE 110 ON READER SERVICE CARD

that include Switchcraft, Amphenol, Cannon and Neutrik. This style of plug is rugged and reliable due to its construction. The plug locks into the socket which makes accidental disconnection extremely unlikely. Since XLRs have three independent electrical contacts, attaching a balanced line pair with shield is no problem (that's impossible with RCA plugs). (It should be mentioned that "stereo" quarter-inch phone jacks can also be used for balanced line applications.) The 3-pin locking connector is the standard of the professional audio industry, and is the most expensive of the three popular connectors because of its more complex construction.

In addition to those discussed above, there are a virtually infinite number of other connectors that are suitable for low level audio. Most of these are multi-pin connectors; as many as two hundred pins are available in some of the larger ones. (That's enough for sixty-six balanced line pairs with individual shields!)

When choosing any connector, be sure that it is designed for very low power "dry" level operation, especially if microphone level signals are passing through the pins. Frequently, the various metal surfaces that make the actual electrical connections are gold plated to resist deterioration of the contacts. Other metal platings, such as nickel and tin, can be suitable as well, but always check with the connector's manufacturer if in doubt.

Gold plating is also available on RCA plugs and on the contacts of the "XLR" connectors (at a premium price, of course). Assuming that the plating doesn't scrape off, it can be worth the investment, particularly for connectors that are left plugged-in for long periods of time.

Quarter-inch phone plugs can be purchased with brass contact surfaces. These "military" style plugs are very rugged, but also require periodic cleaning to eliminate tarnish.

One other connection system is widely used in professional audio equipment; it is the so-called "barrier" block which consists of screw terminals mounted on a plastic bar. Bare wires from the interconnection cables can be wrapped around the screw and tightened down, but it is a better idea to attach a lug to the wire. In most cases, the fork-shaped lug is crimped onto the wire by means of a relatively inexpensive (about ten to fifteen dollars) crimping tool which is specifically designed for this application. Also available are lugs that can be soldered onto the wire. Either the crimp or solder type lugs will result in a much more solid termination (and a tidier installation) as compared to merely tightening the bare wires under the screws of the barrier block.

Connectors for high-currents. Many of the same connectors can be used for both loudspeaker as well as AC power interconnections (but *never* use the same type for both applications in one sound system, or else you may find yourself accidentally pumping the 120-volt power line into your favorite pair of woofers). Twist-Lock (trademark of Hubbell, one of the manufacturers of this variety of power connectors) plugs and sockets are quite popular since they must be given a quarter of a turn to mate or unmate the connectors. Consequently, the plugs are not likely to fall loose from their sockets.

There are many configurations of Twist-Lock connectors. The differences are mainly the quantity of contacts (ranging from two to five) and their current ratings. Beware, however, of the fact that many apparently identical Twist-Locks are not interchangeable because of subtle differences in the pin geometries. Before you plunk down your hard-earned cash at the electrical supply house, double check and ascertain that the male and female connectors will actually interface with each other.

The catalogues of the connector manufacturers list a multitude of other plugs and sockets that are suitable for loudspeaker and AC power applications. The selection procedure boils down to the number of pins, the current rating and the durability of the shell and contacts. If possible, compare notes with other users of this type of connector so you can benefit from their experience.

This article is far from being all-inclusive due to the extent of the subject. However, it is hoped that the basics have been thoroughly discussed so that the reader will be better informed about cables and connectors.

It is highly recommended that the wire and connector manufacturers be contacted to obtain copies of their literature. There is a wealth of suitable candidates for audio interface applications as judged from the catalogues.

If any reader out there in audio land has some experiences, good, bad or indifferent, concerning cables and connectors, please drop a line to *Modern Recording* and share your comments. Through the active exchange of ideas, the entire audio industry can benefit.

The following is a short list of a few popular cable and connector manufacturers (original equipment manufacturers); there are many others in addition to these. Contact the companies to secure literature and information on their product.

WIRE

Alpha Wire Corp.
711 Lidgerwood Ave.
Elizabeth, N.J. 07207
Phone: 201-925-8000

Belden Corp.
Box 1327
Richmond, In. 47374
Phone: 317-966-6661

Columbia Electronic Cables
11 Cove St.
New Bedford, Ma. 02744
Phone: 617-999-4451

Manhattan Electric Cable Corp.
225 Passaic St.
Passaic, N.J. 07055
Phone: 201-779-3331

CONNECTORS

Amphenol, Div/Bunker Ramo Corp.
900 Commerce Drive
Oak Brook, Ill. 60521
Phone: 312-986-2700

AMP, Inc.
Eisenhower Blvd.
Harrisburg, Pa. 17105
Phone: 717-564-0100

Bendix Corp.
Sidney, N.Y. 13838
Phone: 607-563-5321

Harvey Hubbell, Inc.
State Street & Bostwick Avenue
Bridgeport, Ct. 06602
Phone: 203-333-1181

ITT Cannon
666 E. Dyer Rd.
Santa Ana, Ca. 92702
Phone: 714-557-4700

Switchcraft, Inc.
5537 Elston Ave.
Chicago, Ill. 60630
Phone: 312-792-2700

Winchester Electronics
Main Street & Hillside Ave.
Oakville, Ct. 06779
Phone: 203-274-8891

Your keyboard solo has to be hot.

Peter Robinson of Brand X

When you play an ARP Odyssey your solos sizzle

On stage and in the studio, the Odyssey lets you perform musical tricks that other synthesizers imitate, but never duplicate.

A patented two-voice keyboard is just one distinct advantage. ARP's famous "human engineered" control panel puts bells, screaming guitars, and lilting flutes at your fingertips. The Odyssey's musical options include everything from bombs and machine guns to sweet strings and soulful saxophones. Plus hundreds of original sounds which you can create.

Since the Odyssey came on the scene, it's blown away every other solo synthesizer around. Sea Level, Boz Scaggs, Chick Corea, the Starship, Kansas, Jean-Luc Ponty, Brand X, and countless others use the Odyssey on recordings and in performance. Each of these bands plays its own kind of music, yet they all depend on the Odyssey to give them their kind of sound.

Now it's your turn.

Step into a Certified ARP Dealer for a complete demonstration and musical tour.

Step up to an ARP.

It's the sound.

For a copy of ARP's full color catalog, and the names of Certified ARP Dealers in your area, send $1.00 to: ARP, 45 Hartwell Avenue, Lexington, Massachusetts 02173

CIRCLE 44 ON READER SERVICE CARD

a session with B.B. KING

By Ed Masciana

One would think that after recording over seventy albums in a thirty-year period, a musician wouldn't really have much more to say 'bout the blues. After all, B.B. King has recorded some seven hundred songs, so you wouldn't think that this recording session would be any different than the hundreds which have preceded it, right? Wrong! This particular session is one of the final sessions. The basic tracks have been laid down as well as most of the lead and background vocals. As the tapes are being played back, fine guitar licks as well as soulful vocal phrases are punctuated by control room shouts of: "Did you hear that?" and "Oh, fantastic!" Most of the hard work that has gone into the making of B.B. King's new album, *Take It Home* (almost two months of recording), is coming to an end as the principals gather around to put on the finishing touches. Much the same way as a painter stands back and looks at his creation, these artists are constantly listening and probing, waiting to brush on the last strokes of what they say will be "B.B.'s best ever." Although the session takes place in an unmarked building in Hollywood [unmarked but not unknown Hollywood Sound], once inside you could be in Pasadena or Long Beach. There are none of the ego trappings here associated with tinsel town. These guys have a job to do and they do it well.

The Principals Herein

Producer Stewart Levine and engineer Rik Pekkonen have worked together for over ten years. This is their second B.B. album. The first one, *Midnight Believer*, is one of the best-selling albums B.B. has had in years, approaching 500,000 units in sales. The team of Levine and Pekkonen have done all of the Crusaders' albums over the past nine years—some fifteen in all—and since they are on ABC's jazz label, Blue Thumb, and B.B. is on ABC Records, it was just a matter of time before the two parties got together. And, get together they did.

The Crusaders not only played on the last album, as well as the new one, but pianist Joe Sample, and lyricist Will Jennings, co-authored some of the songs on both albums.

"We did it mainly to have fun," states Levine. "We all grew up listening to B.B. and it just seemed inevitable that we would do something together; I've known and admired him for many years."

These past two albums have been somewhat of a departure for King, according to Levine, since the songs were written specifically for him and do not follow traditional blues progressions.

"We put these two songwriters together to write blues song forms. Rather than just write twelve-bar blues, we took the blues and extended it and put bridges and sections on it. We tried to get away from the same old format," emphasizes Levine.

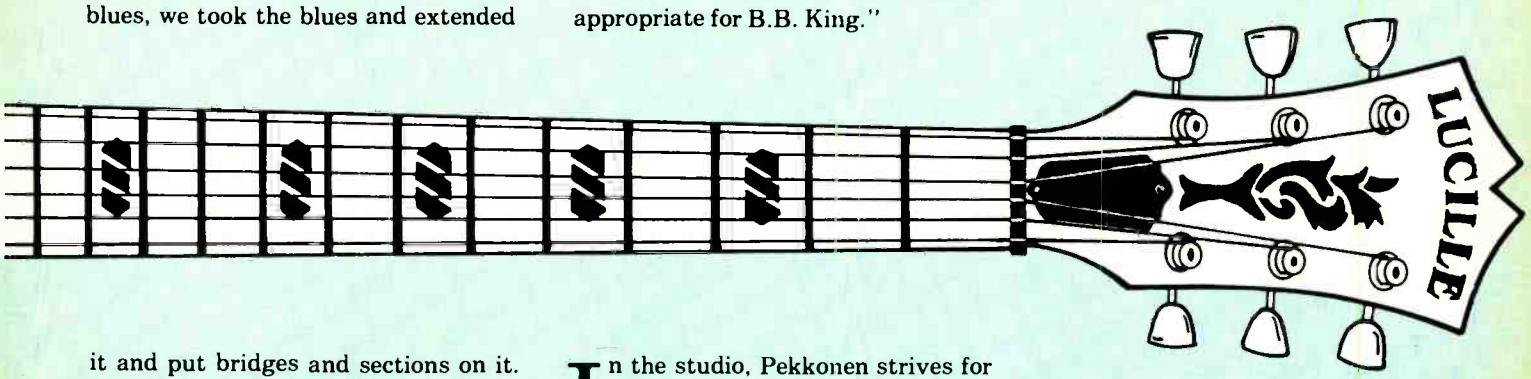
The generalization that people go to a B.B. concert to hear *him* and that it really doesn't matter what songs he does can be supported by the fact that many of the same faces can be seen at each different performance. [A classic case of: "It's the singer not the song."] This leads one to wonder if the record producer and engineer try to capitalize on this popularity by recreating a "live" feel on record.

"Not really," says Levine. "With this album we are trying for a little more of a 'live' feel than the last one, mainly because B.B. is playing and singing the takes 'live' as opposed to tracking one and then the other."

They are also striving to broaden his audience base. "When you're making a record with B.B. King, you're not really only after those who go to see him. We were attempting to open up a newer and younger audience to him. He has sort of peaked out in terms of drawing black audiences. But, we weren't concerned with what he had done in the past. We were mostly concerned with how to take this first album, open up a new groove and continue it." Levine also pointed to B.B.'s strong white cult following as a result of the renewed interest in blues guitarists during the late sixties.

Strictly speaking, *Take It Home* will not be on the "down side of the blues," according to Levine. There will be some fun parts and even humor.

Many stars, in order to attract a larger audience, try to get more "commercial" and step out of their mold and into something else—a phenomenon called "crossover," which is not a favorite word in Levine's vocabulary. "It's taking two good things and making them into one mediocre thing. B.B. is very comfortable where he's at, but, that doesn't mean he can't extend the blues form. However, we want to do it with taste and in a manner appropriate for B.B. King."



In the studio, Pekkonen strives for the best sound possible. He doesn't re-mix to attain the limited dynamic range and hyped-up high frequencies of so-called "AM Sound." "We just try to make good records. The music will dictate the sound of the record." Comparing their Crusader sessions to B. B. King's uncovers some interesting differences and similarities.

"By trying to get different guitar sounds, more of a 'dirtier' sound, we can strive to capture the feeling of the music," explains Rik. "This would be a little different than the Crusaders where we try for a crisper jazz sound."

The actual recording technique, however, is generally similar. Whatever differences are inherent in the music, the emphasis is on the performance with the engineer doing very little, if anything, to change it. "Occasionally they'll get a little too trashy (distorted) and I'll have to say, 'Hey, let's get out of the garage,' but that doesn't happen too often."

While not going after the big productions, Levine does use strings and horns as a "pad" with B.B. in the forefront. "We always have to remember who we're working with. B.B. is a very trusting guy and makes one point when discussing how we should approach his albums, and that is, 'Remember who I am.' He tells us what he wants; we try to sweeten it up."

Occasionally, a momentary lapse can result in a rather humorous anecdote. For instance, on the new album, the word "stirred" was in one of the songs. B.B. remarked, "I wouldn't use a word like that. It would be like me saying 'lackadaisical.' "

Interesting Business

These recording sessions, however, are not all fun and games. When it's time to get to work, it becomes serious business. For one thing, charts are written for each member. Unlike the [unfavorable] impression many have of rock musicians, these guys go over the material much the same way a concertmaster will hold individual sessions with each section of a symphony orchestra. Also, everybody shows up on time, or early.

At the board, Pekkonen believes in a straightforward, no-hype sound with a minimum of extra gadgets. He uses a modified Bushnell (so modified, in fact, that they have taken the name off the

panel) with a capability for 20 inputs. This is fed into a 3M 24-track tape machine running at 30 ips with no Dolby or dbx. "They [noise reduction systems] sometimes do 'strange' things to cymbals and piano as well as affect the ambience and harmonics."

For mics, Rik uses a basic studio array (see diagram) in a standard set-up, but with a few interesting changes. For example, he doesn't mic the drums or bass in the "center channel;" both instruments are double miked for stereo, the bass guitar utilizes both a miked bass amp track and a direct input into the board. His reasoning is that the kick-drum, when mixed for the center channel, rises about 3 dB over what it should.

"A lot of records want a 'hot' kick drum and ours is as hot as anybody's. However, I don't want it to dominate the record," explains Rik.

"When they're playing your record, especially on AM, the mono-mixed kick drum gives you a build-up in the middle. I happen to like separating the sound a little to avoid that build-up."

As far as the positioning is concerned, Rik also explained that he positions the bass [guitar] and kick drum mid-left and mid-right [in the stereo spectrum]. Both he and Levine agreed that the bass sounds from these instruments are not non-directional as many have thought. "It's a fallacy," says Levine flatly. "You can localize the bass sound in many cases, which is why we treat them the way we do."

As for stereo image, Pekkonen believes that the song dictates how much separation he should strive for. "If a song demands it, that's how I'll mix it. Everything is isolated on tape so it's just a matter of mixing down." He explained that in some instances, after hearing a song on the final 2-track master, he'll decide that it has too much or too little separation and he will go back and re-mix. "We try to create a picture."

Levine has a definite idea of how the recording session is going to be scheduled after he has heard all the songs which will be included in the album. Unlike with many other producers, everything is done in advance so that the possibility of problems is held to a minimum. He and Rik outlined their basic schedule for the majority of their sessions:

1) The first tracking date lays down the basic rhythm tracks—guitar, bass, drums and a scratch vocal.

2) Horns and/or strings are added to the rhythm section.

3) Background vocals. At this point B.B. may not even know the songs. Someone, usually the writer, will sing them while B.B. works on his guitar leads for the songs.

4) After the album is recorded, Rik takes about four days for mixdown.

5) Cut a reference disc to see how it relates to the actual master tape. At this point they may go back and remix.

6) After the best reference disc is approved, a test disc is made which is "supposed" to sound like the final pressing of the album.

7) After the test is approved, the master disc is laquered, mothers made, and finally the stampers and record disc are made.

What rejects a test print? Skipping, noise, distortion. (This is a vital nerve with both Levine and Pekkonen.) "The state-of-the-art is atrocious in pressing," sneers Levine. "We have loads of sophisticated equipment in the studio, up-to-date and modern, and yet the pressing techniques haven't changed in fifteen years. The equipment hasn't been up-dated to keep pace with the changes made in the studio. The better the master, the more pressings we reject."

The test discs are heard on all of Levine's four systems and Pekkonen's three. They go back and forth to each other's homes, listening on big and small systems to see how the disc relates to the original master. They listen on the big systems to see if they can hear what they put on the original, and on the small ones to see if everyone else can.

Pekkonen admits to a small amount of limiting and gain riding, but tries to keep it to a minimum. "We strive for as open a sound as possible. I try to stay away from it as much as I can, but sometimes I'm looking for a certain kind of sound that only limiting gives, and it's the only way to get it."

Modest Beginnings

Both Levine and Pekkonen show a tremendous amount of concern and appreciation for the artist frequently referred to as "The King of the Blues." It might have something to do with the total absence of a "superego" and "Hollywood Flash" impression that many "stars" frequently give to the casual observer.

For one thing, how many stars show up for a recording session wearing a three-piece pin striped suit with co-ordinated tie and shirt instead of spats, leather and a cape? B.B. King looks as if he should be walking into a Hollywood bank, not Hollywood Sound recording studios. His present modest lifestyle is closely tied to his modest beginnings.

King left his Mississippi plantation home just after the war while still in his teens. "I came to Memphis in '46 or '47, he recalls. In '48 I learned that an all black radio station had been started." B.B. was hired to do ten and then fifteen minutes of singing and playing "live."

Engineer Rik Pekkonen taking a moment out during the sessions.

"When one of the disc jockeys left, I became a disc jockey. I would fool around on the radio and talk and play guitar. I had to learn to mix and so forth, cue up the commercials on these big reel machines because automation wasn't what it is today. Commercials were on these large LPs, so we had a little of everything."

King's down-home style evidently paid off because his popularity grew to the point where he began getting requests to play "live" concerts in the Memphis area. He became so popular, in fact, that he attracted the attention of talent scouts with Modern Records in Los Angeles.

After five or six records, he had attained a measure of success and by 1950 had recorded what could be called some qualified hits, among them "Three O'Clock Blues."

King played the smaller clubs in the South, referred to as the "Chitlin' Circuit," some holding a mere 200 people. Even after thirty years and several hit records such as "The Thrill Is Gone," "Hummingbird," "Ain't Nobody Home," and others, King still shows a preference for the smaller clubs.

"There are a lot of people who enjoy going to those places that wouldn't enjoy going to the [L.A.] Forum. I like those people too. I want to play for them and I don't even mind the noise as long as they're not heckling me or disturbing others."

Performing "live" versus recording? "They both have their rewards," explains King. "When I sit behind that [recording] mic and I make a mistake, I'll say, 'Wait a minute!' and go back. If I'm out there in front of an audience and I make a mistake, I say to myself, 'Oh God, I hope they didn't hear it.'"

King's "live" performances have changed over the years. He recalled an outdoor concert [a sprawling, Woodstock-like affair] he appeared in nearly a decade ago in Atlantic City, New Jersey. After several hours of pulsating heavy metal rock, King came on stage with one Fender Twin Reverb amp and could be heard, and enjoyed, throughout the entire race track grounds.

"I don't knock the guys that use the big loud amplifiers, because I think it's good. For the guys that can use it, great. I don't know that much about electronics, and a small amplifier is all I want." King uses the Gibson SG amplifier system on both his "live" and recording dates as well as his prize posession, Lucille, a custom Gibson guitar valued at $7,500 and the only model in existence.

"In my case, being a blues singer and musician, I have to carry more people. We're not exposed in the media as much as a lot of the rock or soul groups. Therefore, I try to make up for it in the 'live' performance. I can't afford to carry that many people *and* a lot of sound equipment."

Most rock groups feature four or five, whereas King carries ten or more musicians with him on his "live" engagements. "I can't get away with the big equipment so we substitute musicians for the big amplifiers."

King's new album will feature the same basic musician set-up: two guitars (plus B.B.), bass, drums, piano and organ. An occasional background vocal, horns and strings will also be used on some cuts.

Still Unmistakable

While all admit that B.B.'s style has changed little over these past thirty years, there is still that unmistakable sound that pulls the listener into the song like a magnet. "I've Always Been Lonely," is one such example. It begins much the same as many other blues songs, yet there is a little different feeling here. It might be that the musicians know that this is a B.B. King song, before the listener does, so it has to sound a little different, a little better. He need only sing the first line and now you know this is not just another blues record, it's B.B. King.

Few performers can point to the longevity, seventy albums over thirty years, of a B.B. King. Why? "I remember reading something that the late President Kennedy said: 'Anything you don't know how to do yourself, get somebody who does.' And, I've got Stewart Levine and Rik Pekkonen and Will Jennings and the Crusaders and other good people around me."

They haven't been around King for thirty years, however. "But, I had somebody around at that time who was suitable at that time just as they are now. I've recognized my limitations and have always depended on others who can do the job for me. I guess my old horse sense has paid off," B.B. says. I guess it has.

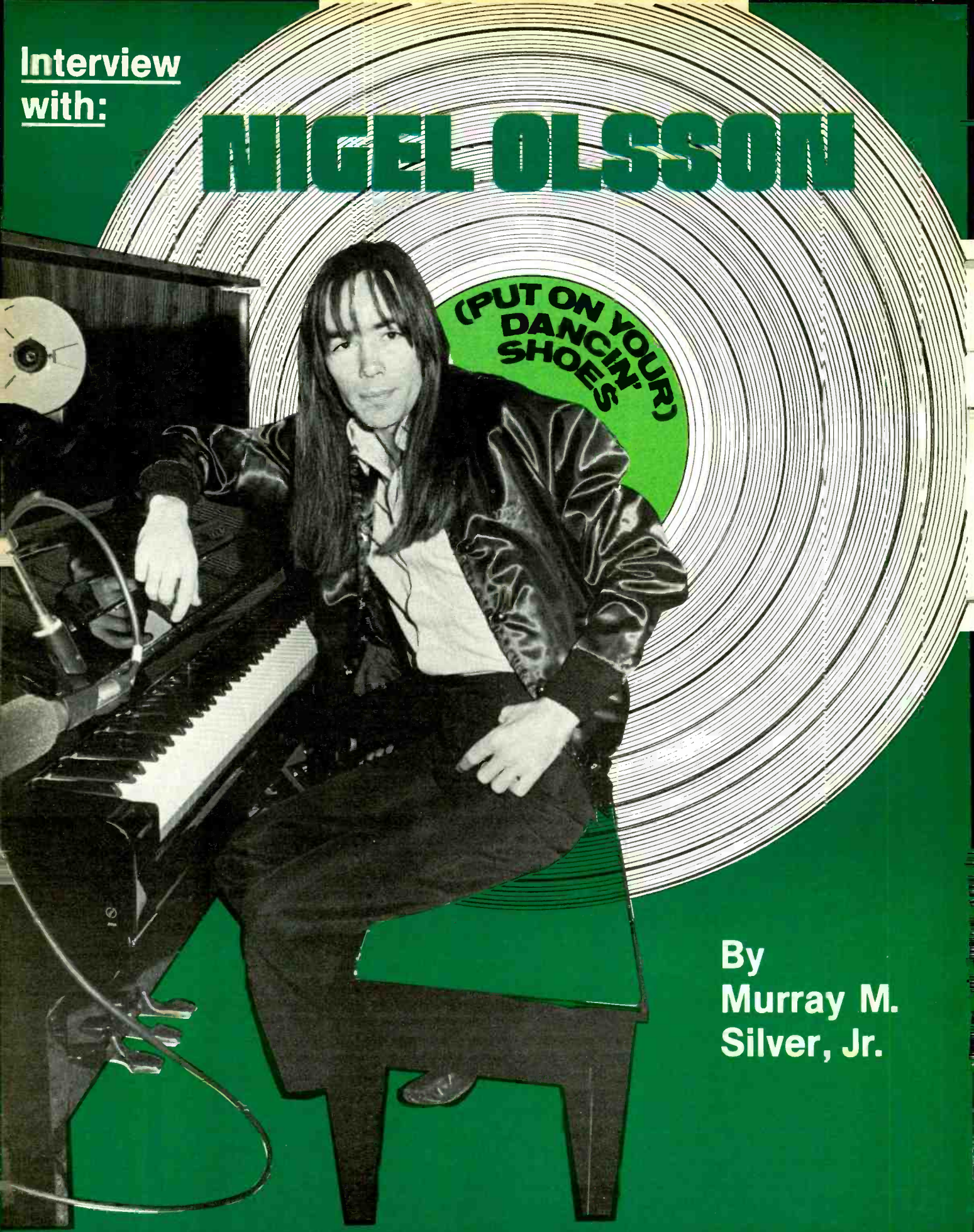
B.B. King Session Mic List

B.B. King:	Vocals—Neumann U-87 Guitar amp—Neumann U-87
Acoustic piano:	Highs—AKG C-12 Mids—Shure 545 Lows—Neumann KM-88
Rhodes:	Direct
Clavinet:	Direct
Guitar (Jackson):	Sony P54 (amp) Shure 545
Guitar (Parks):	Sony P54 (amp) Shure 545
Bass:	Direct and Sennheiser 421 on the amp
Drums: Kick—	Shure 545 Sennheiser 421
	Overheads—AKG 414s
	Toms—Shure 545s
	Hi-hat—Sony P54
	Snare—Shure 545

Producer Stewart Levine, B.B. King, and lyricist Will Jennings (l. to r.).

Interview with:

NIGEL OLSSON



By Murray M. Silver, Jr.

It was Curtains for the Captain. Say Goodbye to Hollywood. After living in a fantasy as drummer with the finest edition of the Elton John Band for six years, Nigel Olsson bid goodbye to the yellow brick road paved with recorded gold, and, at the ripe old age of 26, was faced with starting all over again. His dismissal was a shock; the situation seemed incomprehensible.

Did Nigel feel like shooting the piano player? Hardly. After all, the pageantry of the Elton John parade was what his fondest memories were made of. Whatever fleeting feelings of anger he experienced initially were replaced by a deep sense of loss and the sobering prospect that he would have to mount a solo effort not seen since Ringo stepped out from behind his Ludwigs.

The return of Nigel Olsson to center stage took its time and perhaps it is remarkable that it happened at all. An exemplary drummer, and appreciated by many for his background vocals that so closely resembled Elton's, he had no natural flair for writing lyrics and practically no other musical ability aside from that of percussive timekeeper. He seldom drew direct attention to himself and therefore seemed so perfect in the role of providing back-up to the stellar attraction. Incomplete when working alone, Nigel sought the inspiration of family and friends, and with the help of singer/songwriter Paul Davis, he has reentered the top of the charts in his own name.

MR: The logical place to begin is Cheshire where you grew up. Lonnie Donnegan and "skiffle" was the happening thing and I understand that your first musical interest was in learning to play the acoustic guitar. It was probably some passing interest.

Nigel Olsson: I had a basic interest in music but not in any particular instrument. While we were living in Africa, my older brother and I were given guitars as presents and we learned the three basic chords you needed to know in those days to play the blues. So, I was a rhythm guitar player. I played guitar and sang in early groups but was forced into playing the drums when our regular drummer quit on the eve of a gig. I've been there [at the drums] since.

MR: The family business moved you to Sunderland [a seaport on the northeastern coast of England] where you studied seamanship in school and you learned motor maintenance as a vocation. It doesn't seem that you approached music as a profession. What were your intentions?

NO: I always wanted to be a performer. I always wanted to be on stage and sing but I felt I had to follow in my father's footsteps as a sea captain because he wanted it that way. When I initially wanted to break away from piloting river boats he was pissed. He didn't like that at all. When the Beatles brought their influences on hair styles he was flabbergasted. He was in Africa when I left Sunderland to go to London for an audition and my mother went up the wall. I walked in one night and said, "I'm going to London tonight for an audition and there's nothing you can do about it; there's nothing you can say to change my mind, I just have to go off and do it." And I went. And here I am.

MR: Before we get to here you are, there were the Hondas, the Fireflies and Plastic Penny. Memphis Slim and Graham Bond were influential in your early sound. I understand that you were very shy and didn't like being frontman, that it embarrassed you.

NO: It still does. When I performed at the Agora in Atlanta on New Year's Eve I was frightened to death. Before going on stage, I was always the one who would pace the floor because I was so nervous. People would comfort me by saying I was hidden behind the drums where no one could see me if I made a mistake or dropped my sticks, but it wasn't that way at all. Just being up in front of 25,000 people . . . .

MR: After all the touring you've done certainly you've gotten over this stagefright by now.

NO: No, I'm still nervous.

MR: I think that's healthy. If you weren't it would probably show in your presentation. Nervous energy can easily be transformed into something more positive.

NO: Being up front again will take a few weeks to get used to, especially not being able to hide behind the drums.

MR: Who is coaching you in developing stage presence?

NO: I'm doing it myself.

MR: Chronologically, you worked with guitarist Mick Grabham [Jethro Tull, Procol Harum] in a band called Fallout. Then you and Mick joined a band called Plastic Penny in 1966 and that lasted for a year and a half. What happened next?

NO: I was with Spencer Davis for a few weeks to do his last tour—which was my first time in the United States. The band split up after that. Dee Murray played bass with Spencer and we sort of stayed together after that.

In between Spencer Davis and the first Elton John sessions was a brief stint with Uriah Heep. I recorded two tracks on their first album [*Uriah Heep*] under the name Ollie Olsson. I don't know where that version of my name came from.

MR: It was probably an administrative error. Perhaps when they were compiling the album credits the secretary didn't know your real first name so she made it up.

NO: I was with Uriah Heep nine days before Elton asked me to do a promo gig at the Roundhouse and I said yes, and went immediately into rehearsal. As soon as we started to rehearse I knew it was the kind of music I wanted to do because I could play from the heart.

MR: On Elton's first four studio albums, he used four different drummers on each album. There was Roger Pope, Barry Morgan, Terry Cox and yourself. You played only one cut on *Empty Sky*, on "Amoreena" on *Tumbleweed Connection* and on "All the Nasties" on *Madman*. Here and there you picked up a few background vocals. Yet, on *11-17-70* it was you that Elton asked to perform. What took so long to settle on one drummer?

NO: Most of the early stuff, especially the *Elton John* album, was done by charts. The music was written out so the drummer had to be a reader—which I'm not. I still can't read music. The first of what we call "the group albums" was *Honky Chateau*. Before then, the music was written out and Elton tested a number of different drummers.

MR: How were actual introductions made between you and Elton?

NO: Dee and myself and Elton and Bernie were all working for Dick James. Elton and Bernie were like staff writers, and Dee was playing in a

band called Mirage while I was working with Plastic Penny, both of which were handled by Dick James. We all knew each other, we all hung out together, so when it came time to go on the road, Dee and I were the obvious people to ask because we were there and we knew each other.

MR: Well, the odd thing here is after you were selected to do *11-17-70*, Elton returned to the studio to use four drummers again on *Madman*.

NO: I think Elton felt that Dee and I did not have enough studio experience to go with just us. On the road we were adequate.

MR: Elton was rather slow to credit the band's input. When listening to an early Elton album, the background vocals were a very important part of his sound. To this day, people are amazed to find out that Elton did not overdub his own background vocals and that it is you they are hearing. The pitch was so close. You were used sparingly in the earlier albums, but by 1975 you were singing on most of the cuts. How was the decision made as to which vocals you would contribute?

NO: After we had layed all the basic tracks, Elton would say, "Ok guys, I'm off," and he would leave us to do our parts. Davey [Glover] would do his guitar overdubs or I would do my drums or tambourine overdubs or whatever. Then we would sit down with [the producer] Gus Dudgeon and discuss which songs needed reinforcing with background vocals. On a couple of occasions, "Crocodile Rock" and "High Flying Bird" in particular, Elton would ask me to sing along with him because he couldn't get up that high. I made every effort to effectively duplicate his sound.

MR: I suspected that, but there was no way to find out for sure until a concert rolled around.

NO: Well, most of what you hear in the way of background vocals was my work. Elton used to think I was crazy the way I used to stack vocals, and if you listen to "Someone Saved My Life Tonight" and "Curtains" on *Captain Fantastic* they were multistacked. There were all sorts of things happening. We would lay a basic part over the top and then underneath there would be answering things back and forth. We used to go on and on, and I love to do that.

MR: Several critiques liken your last solo effort to *Captain Fantastic* as far as mood and tempo is concerned; the finale, "Au Revoir," approaches the effect of "Curtains." During which period of your involvement with Elton was the band performing material which is perhaps most reflective of your personal tastes?

> **"I would love to ... put the old band together and tour just one more time ..."**

NO: *Captain Fantastic*. My favorite Elton project ever was the *Yellow Brick Road* album, but *Captain Fantastic* was closest to what I do now and it was the album that came closest to projecting my ideas. I was able to provide more input on *Captain Fantastic*, and, in fact, a careful listen to the album shows that the drums are the most prominent instrument—or at least the loudest. We finally got, well, I won't say the perfect drum sounds because we have gotten better, but the drum sounds were the best of all the Elton John albums to date.

MR: The quality of the drums on Elton John albums is exquisite to listen to. The sound is thick yet precise. To what lengths did you go to in order to find those sounds?

NO: We worked long hours on those sounds. Gus and I were the first in the studio and the last out. We would spend two days getting the drums to sound the way we wanted. We worked on various mic techniques and fooled around. I would sit and just bang on the snare drum for five or six hours just to get the right sound. Sometimes we would do nothing more than go in a complete circle and have to leave the studio for an hour or two break. On *Captain Fantastic* we perfected the mic placements and tuning.

Listening to my last album, the drums on "All It Takes" are probably the best drum sounds on the album, which came about by working with my producer in L.A. who had also been a student of Gus Dudgeon's. I think we perfected the drum sound on that album. In the early days, there was a lot of what I call "clacking," where there is little tonal quality. By the time we did *Captain Fantastic* we were able to master tonal quality, but it wasn't until last year that we could master tone quality while effectively using delay and echo.

MR: In the studio does it require forty takes and ping-ponging drum bits down to one track in order to get what you want? The precision of your work [past and present] sounds either like the engineer did an incredible amount of work or you practiced the same part for days.

NO: We never spliced anything. Never. In fact, most of the hits that we had were first or second takes. Bernie would send the lyrics to Elton and he would write the melodies. At Caribou, [recording complex] Elton would write the tune in the morning, we would go into the studio in the afternoon, rehearse it a couple of times and tape the first take. "Someone Saved My Life Tonight," "Candle In The Wind," "Rocket Man," "Honky Cat," and "Don't Let The Sun Go Down On Me" were all first takes and we might have played each song only five or six times before hand. Nothing was overdubbed.

All of the takes on my solo albums are overdubbed. The piano track was the first track to go down. I had my own studio at home in L.A. and I would record the basic tracks on piano and then overdub everything else in the studio. Drums were added last.

MR: *Don't Shoot Me I'm Only The Piano Player* was the second of what you call "group albums." I hated it except for "Elderberry Wine."

NO: There are two albums that stand out in my mind as being a bit funky, *Don't Shoot Me* and *Caribou*.

MR: Looking back, it seems that *Don't Shoot Me* was like biding time until the *Yellow Brick Road* sessions and that *Caribou* simply rested on the *Yellow Brick Road* laurels. I have a theory: a hit song should be one which everyone can sing along with while driving in their car or in the privacy of one's shower. Elton's songs defy that. The general public cannot sing along with Elton John and since your pitch approaches his, the same might be said for your material. If you ever heard someone try to sing "Goodbye Yellow Brick Road" you would see what I mean. Yet, the hit quality of these songs is unsurpassed. So I can't make my theory fit the example.

NO: That's an interesting point because you are right in what you are

Soundcraft Series 1s.

The new industry standard.

Series 1s: stereo mixer available in 12, 16 or 20 channel format; walnut console optional.

Soundcraft EX4S.

The ultimate electronic crossover.

EX4S: 24dB/octave Bessel function 4-way stereo crossover.

For full details and specifications, contact
Soundcraft Electronics Ltd
5-8 Great Sutton Street, London EC1V 0BX
Telephone 01-251 3631 Telex 21198

Soundcraft Inc.
PO Box 2023, Kalamazoo, Michigan 49003, USA
Telephone (616) 382 6300
Telex 22-4408 Soundcraft KMZ

SOUNDCRAFT ELECTRONICS LIMITED

CIRCLE 78 ON READER SERVICE CARD

saying. Sometimes when kids do sing along, its usually my background vocals that they sing along with. As in "Yellow Brick Road," most people only remember the background on the chorus. That's a good point.

MR: In 1973 the Elton John Band hit a relative low with *Caribou*. You had been working with Elton four years, going through his evolutions, how were you changing as an individual musician? Did you have any desire to do a solo project?

NO: I guess all along, deep down, I wanted to make my own records, but I didn't necessarily want to go on the road because of that scare factor I mentioned earlier. I really didn't have time to write and I couldn't play any other instrument beside drums ... except for the old three chords on the guitar. The music that I was getting into, which was heavily Elton influenced, had a lot more chords in it than the three I know. I didn't really start writing until the band was over with.

MR: You had two solo projects.

NO: *Drummers Can Sing, Too* was actually just a dialogue album, an interview with songs from the Rocket LP, so it really wasn't an album, just a promo for radio stations. We later found someone selling them in stores so somebody had a scam there. The first solo project was the *Drum Orchestra and Chorus* album in 1970 where I went into the studio with old friends like Mick Grabham and Caleb Quaye and we played around.

MR: When the Elton John Band broke up in 1975 there was not a whole lot in the way of an explanation presented to the public. All we heard was that Elton had summarily dismissed the band for "reasons," and that there seemed to be the promise of a reconciliation. There were many who, upon receiving the news, said, "That's impossible, no one can replace Nigel."

NO: To this day, no one in the band understands why. I remember when he called me in LA. to tell me that it was over. The only reason Elton gave me was that he wanted a change in musical direction and a change in his stage appearance. I still don't think he has found his new direction.

MR: Even though they had their moments, *Rock Of The Westies* and *Blue Moves* was largely disposable. "Tonight" was lovely. But I didn't hear anything as poignant as *Captain Fantastic*. Davey left under slightly different circumstances to form a one-time shot with China and Dee backed Kiki Dee on a tour. At the time of the break up, I thought that the Elton John Band might go on without Elton John because the trio had a great sound. Was there any discussion of a project of that nature?

NO: Yes, it was discussed that we should make a record together and we did record together. In fact, we still have the stuff in the can. It was Davey's material. But when Davey formed China two years ago, he did not ask me in. He has postponed a second album to work with Alice Cooper and presently they are planning a tour.

MR: The last Alice album sounded like an Elton John album without Elton, especially with Bernie writing the lyrics. A latter day *Madman* as it were. You stated that *Captain Fantastic* was a pinnacle for you personally. At the time, what were the specifics of your recording technique?

NO: I always mic very close and from underneath the drums. I used two overhead mics and a pair of ambience mics at the other end of the studio to get that delay effect which comes over so well on *Captain Fantastic*. We achieved the perfect delay effect without getting into the electronic special effects. The rest was provided by echo. Digital delay was used only on the cut "Better Off Dead" and the drums were overdubbed on that track only.

MR: Your first solo hit, "Only One Woman" sounded a lot like Elton, but your most recent hit, "Dancin' Shoes" completely erases all traces of an English accent in your voice. Is that by design?

NO: I think that comes with living here for four years.

MR: Let me ask you a personal question. During the Elton years, the money you were making, were you socking it away or blowing it? When you first arrived in Atlanta, I remember hearing that if you want to find Nigel Olsson, look for a blue Ferrari.

NO: Someone took a sledge hammer to it shortly before New Year's and it's in the shop presently. I've always been an automotive fanatic. I love toys like boats and jet skis. In fact, I just had the cast removed from my hand where I broke my fingers while tearing around in a dune buggy. I got overconfident and flipped it over which put me out of commission for five weeks.

MR: Did you wisely manage the money you earned in the golden years?

NO: No, I threw it away. Just blew it. Well, that's not entirely accurate. The cars I bought were worth something and I did buy a 60 acre farm in England which my parents live on.

MR: Was it your design to bring your brothers over here?

NO: My brother Carl is here [Atlanta] and working for Bang Records. He is a musician and a writer and shows great promise. Fantastic, really. Then there is my brother Kai who is now recording with Chrysalis. He had a hit single in England which hasn't been released here.

MR: Will you be helping him with his project, do you think?

NO: Yes, I did play some on the album, and of course, if there's anything I can do ... but he's pretty set in his own ways.

MR: Do you get invitations to do session work?

NO: Not since I moved to Atlanta. If I were still in L.A. I would still be in the session circles. If the right project came up I might be interested, but since I moved I have worked with Paul Davis mostly.

You know, I have even been asked to give drum lessons, but I don't because I am not a technician. I play strictly from the heart. I can't even play a drum roll. If I was asked to play "God Save The Queen" I couldn't do it because I can't do the roll. I play from the heart, even the fills, but I am nothing more than a timekeeper.

MR: Your new album [*Nigel*] released in February, has seven cuts from the previous LP. What was the bright idea behind doing that? Were you editing out cuts that were all wrong for you?

NO: We felt that the last album [*Nigel Olsson*] wasn't promoted properly and this is the album that I am most proud of. It was the best project I ever did. Even though all the others were masterpieces and we had a lot of fun, for me, this album was the best thing I have ever done. I just started writing, I was happy with my songs and I had reached what felt like a personal perfection.

At one time before, I had tried to write lyrics that were just sucko: *I love you, yeah, yeah, yeah*. The writing process used to be very difficult for me, but on this project it wasn't. I would just sit down at the piano and play only the black keys. I can't really play the piano, but something would invariably come up.

Having my own studio in L.A. was

1in. 8-TRACK WITH A PLUS

MX-7800: Full function capability with no compromise in reliability. Performance second to none. • Full-servo constant-tension tape drive • Dynamic braking • 30/15 ips operation • FET switching • $8895 suggested retail USA.

THE PLUS: A unique multi-function remote, the CR-706, lets you do it all from your console • Overdubbing with automatic monitor switching • Silent punches • Fine and coarse speed variation • $995 suggested retail USA • CT-501 real time counter with return to zero • $495 suggested retail USA.

EXPERTISE NEAR YOU: We are proud to have these select dealers franchised to sell, maintain, and service **the only 1-inch 8-track with a plus.**

WEST

Accurate Sound
Call Greg 415/365-2843
Redwood City, CA

ACI/Dave Kelsey Sound
Call Fred 213/851-7172
Hollywood, CA

EAR Sound Consultants
Call Ed 602/968-8675
Tempe, AZ

Express Sound
Call James 714/645-8501
Costa Mesa, CA

Sound Genesis
Call Dave 415/285-8900
San Francisco, CA

Westlake Audio
Call Bruce 213/655-0303
Los Angeles, CA

CENTRAL

ASI/Abadon Sun
Call Galen 512/824-8781
San Antonio, TX

Audio Distributors
Call David 616/452-1596
Grand Rapids, MI

AVC Systems
Call John 612/729-8305
Minneapolis, MN

Ford Audio & Acoustics
Call Jim 405/525-3343
Oklahoma City, OK

Paul Westbrook Audio
Call Paul 512/578-4401
Victoria, TX

Sonics Associates
Call Doug 205/942-9631
Birmingham, AL 35209

Hy James Enterprises
Call Henry 313/994-0934
Ann Arbor, MI

Allied Broadcast
Call Jim 317/962-8596
Richmond, IN

Milam Audio
Call Jerry 309/346-3161
Pekin, IL

Houston Cinema & Sound
Call John 713/933-7180
Houston, TX

Gill Custom House
Call Al 312/598-2400
Palos Hills, IL

Ludwig Sound
Call Charles 713/449-8388
Houston, TX

EAST

Alpha Audio
Call Nick 804/356-3852
Richmond, VA

Audio By Zimet
Call Sid 516/621-0138
Roslyn, NY

Audio Unltd
Call Ed 919/274-4682
Greensboro, NC

Blue Diamond
Call Joe 412/746-2540
Cannonsburg, PA

Cathedral Sound
Call Don 518/465-5689
Rensselear, NY

Dimension Five
Call Charles 215/589-5312
Womelsdorf, PA

Harvey Sound
Call Gene 212/575-5000
New York, NY

Lebow Labs
Call Peter 617/782-0600
Allston, MA

Martin Audio
Call Courtney 212/541-5900
New York, NY

P.A. Palace
Call Ellis 404/636-3044
Atlanta, GA

Professional Audio Video
Call Paul 201/523-3333
Paterson, NJ

Stereo Sonics Labs
Call Alan 305/831-2575
Altamonte Springs, FL

VSM Leasing
Call Vince 716/759-2600
Clarence, NY 14031

Harris Company
Call Michael 305/944-4448
No. Miami, FL

Sounds Fine
Call Dave 703/951-3489
Blackburg, VA

RCI Sound Consultants
Call Tom 301/587-1800
Silver Spring, MD

Recording Services (of W.N.C.)
Call Ted 704/648-6012
Canton, NC

OTARI®

For information on other Otari products and dealers, call Ruth Pruett at 415/593-1648.
Otari Corporation, 981 Industrial Road, San Carlos, CA 94070 TWX 910-376-4890
In Canada: BSR (Canada, Ltd.), P.O. 7003 Sta. B, Rexdale, Ontario M9V 4B3 416/675-2425

fantastic because I could just go downstairs and fool around with the tape recorder on without having to worry about paying engineers. That's why I am building a studio in my new home.

The songs I wrote for the last album were much like the attitudes in *Captain Fantastic*. I was living the moment and the stream of consciousness just flowed. I would get as far as I could on the black keys and then ask David Foster [arranger, pianist] to assist. I've known him several years and he recently worked with Alice Cooper. The best song that I have written to date is "All It Takes" because it is so close to what has happened to me now.

MR: There's is a strong R&B sound on the first album. Will it carry over into your new work?

NO: Yes. "Dancin' Shoes" is obviously of R&B origins. And I am doing a remake of "A Little Bit Of Soap" as if it were being performed by the Doobie Brothers. The new takes are very raw and very basic without a lot of strings and all the background stuff. As far as R&B influences go, there's a lot of Stevie Wonder here.

MR: Why did you cover a Billy Joel tune ["Say Goodbye To Hollywood"]?

NO: When Billy Joel recorded the *Turnstiles* album, Dee Murray and I went to New York to record it with him. When it was on the drawing board, the producer, Jim Guercio from Caribou, told Billy that he sounded a lot like Elton. So, Billy was afraid to use us—he thought he'd be accused of resurrecting the Elton John Band.

MR: Now he sounds like McCartney.

NO: Yeah. But Dee and I went to New York and did the album and I fell in love with "Say Goodbye To Hollywood" because I respect it. Billy reneged on the deal and re-cut the whole album with another band. When it came time for me to do my own album, the song just stuck in my mind.

MR: It has a Beach Boys' sound. America in the '60s.

NO: Yeah, well, Bruce Johnston assisted on the vocals and we used a lot of echo in trying to create the Spector sound. That song was responsible for getting me the Columbia deal. I did it as one of three demo cuts for my manager to take to shop around for a record deal and Don Ellis at CBS in L.A. liked it and said "bring him in." All I wanted was enough money to record where I wanted and how I wanted with whom I wanted.

MR: Why did you have two other drummers [James Stroud, Mike Baird] to assist on the album?

NO: I played on every track, but on some there are two drummers although you may not be able to tell. James Stroud is Paul's drummer. If you plan it properly and if you don't get in each other's way, it's fantastic. Basically, you both have to be good timekeepers playing in the same style and making sure that you give a nod when you plan to take off on a fill.

MR: On your albums, the qualities of drum sounds that I loved so much in Elton albums are almost non-existent. Did you make an effort to keep the drums from coming to the front?

NO: I wanted it to sound like a hit record, not a drum solo. That's why I chose to work with Paul Davis. He's a great referee and mixer. I sat down with Paul in L.A. along with Kevin Beamish and Ed Seay and we mixed it together. I tried to stay away from mixing as much as possible so that I wouldn't get too close to the tracks.

Paul and I co-produced most of the tracks on the first album. On the new material, "Dancin' Shoes" and "A Little Bit Of Soap," Paul takes over which I am very happy about. There are so few people who can record themselves and Paul Davis is one of them. He is not overly self-indulgent. Stevie Wonder is another.

MR: You co-wrote most of the music and lyrics of your songs with relative unknowns. What was the writing arrangement here?

NO: I would get a few ideas and then take them to Jozy Pollock who was my old lady at the time. I would take the song as far as I could and then she would take it. When we split up, I more or less took over the rest of the project myself with the exception of my brother Wayne who wrote a verse here and there. It was the same thing with the music. I would take it as far as I could go and then give it to someone else to work on.

MR: How much input did you allow the "hired help" to contribute?

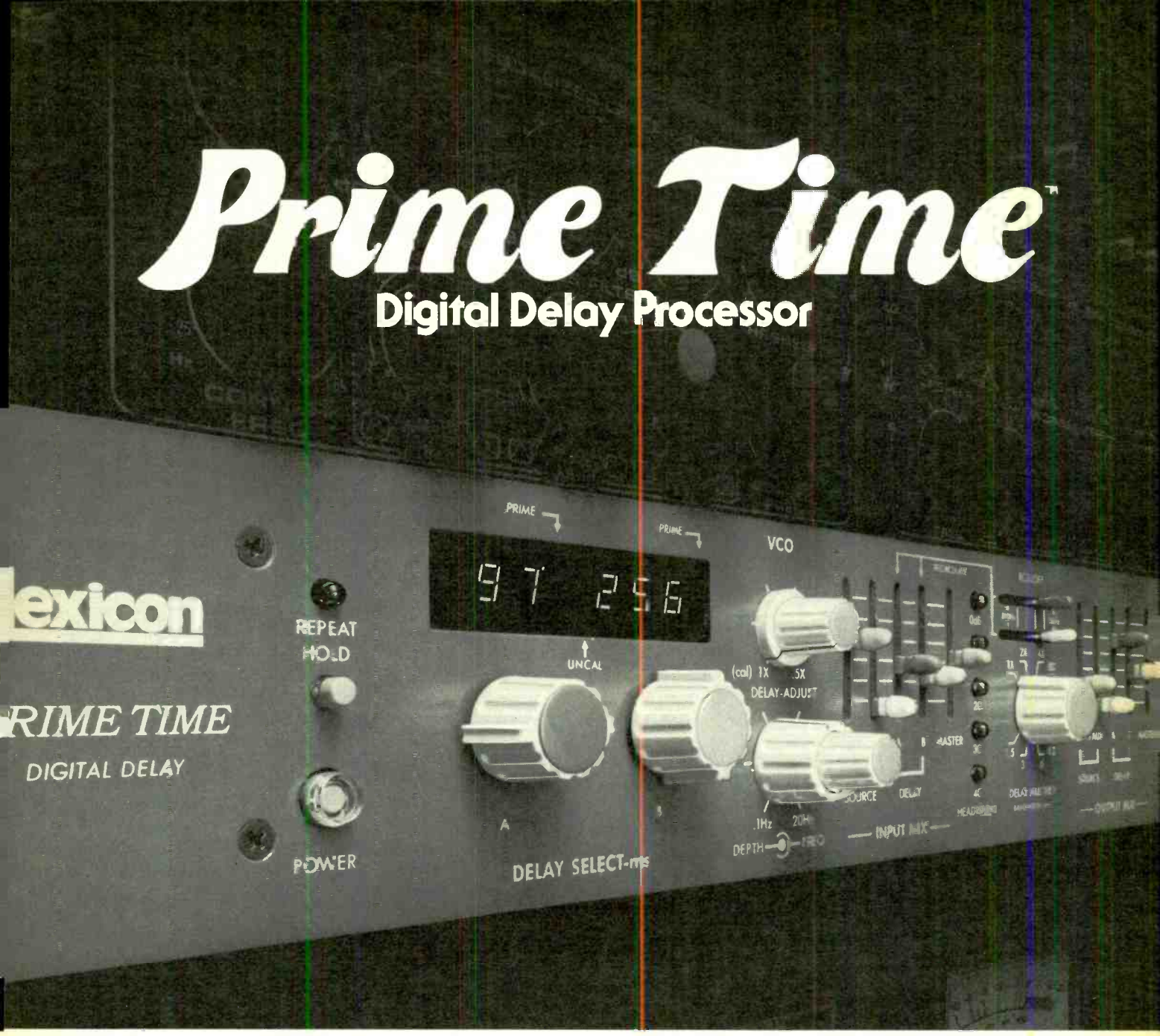
NO: The session musicians added a lot as far as the way the tracks feel. Everybody had his input. It wasn't like: Nigel is the boss and don't go changing any of his songs. If anybody had an idea, it was welcomed.

MR: At the completion of your tour, what will be the next step? Will you form the Nigel Olsson Band?

NO: At some stage I will have to do

Prime Time™

Digital Delay Processor



professional quality delay plus special effects

Lexicon's new Model 93 "Prime Time" digital delay processor gives recording studios and entertainers an easy-to-use professional quality time delay with special effects and convenient mixing all at a price you can afford. It combines a degree of flexibility and versatility never before offered in equipment of full professional quality.

- Two delay outputs independently adjustable from 0 to 256 ms
- Complete mixing for delay and reverb processing, freeing up main console channels and tape tracks
- Integral VCO for special effects like vibrato, doppler pitch shift, flanging and automatic double and triple tracking
- Long delay special effects — up to 2 seconds
- All dynamic functions can be footswitch controlled
- 90 dB dynamic range, total distortion below 0.08% at all delay settings

lexicon Lexicon, Inc., 60 Turner Street
Waltham, MA 02154 (617) 891-6790

Export: Gotham Export Corporation, New York, New York

CIRCLE 71 ON READER SERVICE CARD

that because Paul will be going into the studio and on the road at different times than I will and it will become too difficult to use the same band. Right now, I use Paul's band because we tour and record together. Perhaps that is the toughest thing about starting a solo career—the forming of a new band. I don't have anyone in mind so I imagine I will be holding auditions.

MR: Before the previous album was released, your relationship with Jozy came to an end and a new one with Ilene Berns began. At what point did the one end and the new one begin?

NO: Ilene was visiting me in L.A. while I was finishing the writing of the lyrics to four of the songs. "Chosen Few" and "All It Takes" were written about the things I was going through and they are for Ilene. "Cassey Blue" is Ilene's daughter.

On one occasion I was driving Ilene to the airport and she told me not to walk her to the gate because she hates goodbyes. I told her, "Well, don't say goodbye, say *au revoir*. It only means, "so long, see you soon." I thought as I said that that it was a nice sentiment. When we got to the airport I didn't see her to the gate, but went back to the studio and wrote the beginnings of "Au Revoir." That's the sort of way things come to me.

MR: Nigel, when you sit back at the ripe old age of 30 and reflect over an already brilliant career, your fondest memories, what are they about?

NO: My fondest memories of all time are the first two years of the Elton John Band. The last two years were the beginning of the end. I knew the end was coming and when it came I wasn't really surprised. Money was the sign that the end would be approaching.

The Elton John Band turned into a big business and it still does me in to talk about it. It wasn't the music that was important, it was how much money we were making and how many gigs we were performing.

MR: Was money Elton's concern or the business people around him?

NO: It was the people around him. A lot of people borrowed money that was never repaid, stole it; no names mentioned—because I still don't know who did it and who didn't do it.

The thing was, after awhile, coming off stage, it wasn't, "Hey, the show tonight was great and that little thing you did there during that song was terrific," it was, "Guys, do you know how much money you made tonight?" Which is great you know, everybody loves money. I'm not saying I hate the money because here I am with three houses and cars and gold and platinum all over the walls. It's just that the music had lost something and I could see the beginning of the end. And Elton had a lot of pressure on him, God bless his heart. I don't know how he went through all that he did. The pressures of having to do an album and then as soon as we're finished off to a major American tour and then off on a major European tour with maybe two weeks off a year. We never had time to sit down and say, "God, we've done all this, why don't we just take six months off?

MR: Were you happy with that pressured schedule?

NO: Yeah, I was happy touring and cutting albums and getting up on stage to that roar. That sound of all those people, when I hear it I just . . . even when I listen to a "live" album, particularly Natalie Cole's "live" album, when she's finishing off and she says, "I'd like to thank you and I love you all" and the band suddenly goes into one of her hits without any introduction except, "This is from me to you," and the audience goes absolutely wild. The hairs on my arms stand up—I know that feeling when the audience just goes wild. I used to love that.

But the sixty-five days on the run . . . after three or four weeks you just burn out. And I always felt that when you are burning out you are not giving your all to the people out there who are buying your records and making you what you are. And without them folks out there, who've put "Dancin' Shoes" where it is, there wouldn't be any Nigel Olsson aside from the fellow on a few old Elton John albums.

MR: What was your down time like? Were you big party goers?

NO: No. We didn't party or go to drug orgies. In fact, many said we were the straightest band they ever knew. I mean we weren't like the Who or the Stones where you come into hotels and smash them up. We came off stage, straight to the limos and to the airport, onto our private jet and on to the next place. We would get to the hotel and sleep all of the next day, go

KELSEY®

PRO-CLUB SERIES

HERE AT LAST!

From the people who specialize in high quality mixers. "Kelsey" introduces its new line of "or-stage Mixers". The New Pro Club Series! And in the true Kelsey tradition, each Pro Club Mixer comes built in its own Road Case. If you're looking for the finest quality in any price range check out the new Pro Club Series.

Pro Club Series Mixers are available in mono and stereo with these standard features:

- **L.E.D. on each input**
- **Transformer Balanced Inputs**
- **Variable Input Gain Control**
- **3 Bands of EQ**
- **Monitor Send**
- **Fader Style Level Controls**
- **Built-in Reverb with Additional Effects Send**

Dallas Music Industries

150 FLORENCE AVE., HAWTHORNE, NJ. 07506 201-423-1300

DMI

CIRCLE 88 ON READER SERVICE CARD

do the gig at night and that was it. We had no time for parties. We might stay up and bullshit, but I didn't smoke or do drugs, that sort of thing.

MR: So much has been written about Elton John personally and about all of you gentlemen individually, how did you actually perceive the reality of who you were?

NO: It was hard to understand, especially in the early years, exactly what was going on. The Elton John Band was *the* touring band to see and Elton's records were *the* records to buy. Because of our heavy schedule, we didn't have the time to sit down and think about it all and say, "Look at what we've done." It was only in the later years that I had the chance to sit back and take it all in. It is all very easy to fall into the ego trip, sloughing people off on the street.

I always swore that I would remain natural and that's a very hard thing to do in this business. I notice that when I go home and see an old school friend that it's very hard to hold a conversation. Talking to you, it's fine if I mention Ringo or Mick Jagger or Farrah Fawcett, but when I go home it's very hard to hold a conversation with someone who will find it hard to relate to. If I say, "Farrah and I were at dinner with Lee," they say, "Oh God, Nigel's gettin' a bit flash on us, he's got the ego trip!"

MR: How do you want to relate to the crowds now? Do you think that you can reach a point where you could play Wembley again to 50,000 as Nigel Olsson and walk out to that roar?

NO: It would be nice to be able to pull that many people to a concert but it's not necessarily my aim. I like to do gigs like Carnegie Hall, the Greek Theatre in L.A. and the Fox in Atlanta. Small 3,000 to 4,000 seat halls where you can still be intimate with the audience. When you're up on stage you can only see the first five or ten rows anyway. I like to feel the audience, whereas in most halls you can't even see them. If the album takes off I don't know how I will be able to continue doing 4,000 seaters and do the big halls as well. I'll have to stay in each city three nights . . . .

MR: Is your relationship with Elton such now that a collaboration is out of the question?

NO: I think so, yeah. I would love to—and I haven't mentioned this to anybody but Dee and Davey, seeing that the band split up before we could tour *Captain Fantastic*—go out with Elton on tour. Put the old band together and tour just one more time for no money. I mean, all expenses paid, obviously, but go first class, do it in style. But after all expenses were paid, just give the money to someone so the big money boys that screwed everything up earlier couldn't say, "Well, Nigel's just trying to get out and make little bit extra." I would love to do something like that.

I know that I could never get through the barrier of people that surround Elton. I could never get in to talk to Elton about it. But it would be incredible. Just one more major tour.

MR: That's religious. That's a marvelous wish. I would hope that a musician who has had a phenomenal career would do something like that for his fans even if he didn't want to do it for himself.

NO: That's why I say I don't want to make any money off of it. We never got the chance to tour our best work. That's all I want.

The notes were always there.

The Electro-Voice S18-3 Stage Keyboard System now lets you hear them!

Unless they were willing to spend $2,000 or more for a sound system, most keyboard and synthesizer artists were never able to hear the full range of sound their instrument was producing.

Electro-Voice has changed this! The 40 Hz note a synthesizer puts out can now be heard. The upper harmonics that sounded muddy because they were squeezed through a lead guitar speaker sound muddy no longer. That "honky" sounding midrange won't sound "honky" anymore.

The S18-3 Stage Keyboard System sells for under $1,000, and was specifically designed for those applications that require an accurate sound over a wide frequency range – exactly the performance required by keyboards and synthesizers.

Electro-Voice uses the famous EVM-18B loudspeaker in an optimally vented enclosure to achieve frequency response and high acoustic output that is virtually flat to 40 Hz. The ST350A tweeter not only is flat to 16,000 Hz, but radiates its output over a solid 120° angle. Gone is the "beaminess" at high frequencies that restricts the audience's ability to hear the highs or the artist's ability to hear himself accurately. The VMR (vented cone midrange) couples a 6-inch cone driver to a 16 lb magnet structure in an integral vented enclosure. The result – the clarity of a cone driver without sacrificing the efficiency of a horn.

It's conceivable that many keyboard artists don't even know how good their instrument can sound. If you are one of these artists, prepare yourself for an emotional experience. Audition the Electro-Voice S18-3.

Electro-Voice®
a gulton company

600 Cecil Street, Buchanan, Michigan 49107

CIRCLE 82 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

Direct-to-Disc Vs. Digital Recording

"Which is better, direct-to-disc recording or digital recording? Is there an *audible* difference? How about laser recording?"

That's the content of a letter received from Francisco Rosas, of San Angel, Mexico. And when the editors of *MR* posed these questions to me, I decided immediately that a two word answer really wouldn't do. The questions posed are complex and deserve an entire "Ambient Sound" column.

Leaving aside the entire subject of "laser recording" (by which I presume the writer means the as yet unavailable optical/digital disc which may someday come as a spin-off from the just introduced optical video disc), let's concentrate on the first two questions posed. First, a few definitions. The so-called digital recording to which Mr. Rosas refers is *not* a digital recording in the true sense of the word. It is as much an analog recording as any mass-produced LP disc you have owned since the invention of recording 100 years ago. A stylus riding in a record groove still has to respond to mechanical wiggles in that groove and must translate those wiggles into electrical voltages. What *is* digital about these new discs is the intermediate tape recording process by which the master tapes are mixed down from multi-track tape recordings. Those individual tracks may also be digitally recorded.

In digital tape recording (at least as it is now being practiced in a very few select recording studios), the audio signals to be recorded are first translated into a digital code consisting of millions of "bits" of digital information, in the form of constant amplitude pulses. It is these pulses that are recorded onto the tape, rather than the continuous waveforms corresponding to the original music signals. There are three important advantages gained from mastering tapes in this manner. First, given enough "bits" of information, there are no limitations upon dynamic range that can be stored on the tape.

Next, since the recordist is no longer concerned with tape saturation or tape background noise levels, harmonic distortion of the resultant master tape now depends strictly upon the sampling rate and the accuracy of the digital system, and not on the tape or tape deck's quality. Wow-and-flutter, too, become insignificant, since the pulses recorded on the tape are "released" by a crystal controlled clock for translation back to analog audio signals.

Finally, no matter how many subsequent dubbings are needed to mix-down to the final two-track master, such dubbings have no effect on overall signal-to-noise ratio. Thus, the final master is as noise free as any of the digitally recorded individual tracks.

All of which would be sonic utopia if the process stopped at that point. In fact, if any of you have ever heard a digitally recorded *master tape* played back over a top-quality sound system, the experience is thrilling. There is no audible background hiss or noise. There is not evidence of overload or audible distortion. There is only the music—pure and unmasked. Unfortunately at this point in the process, we must (for now) abandon the digital process and revert back to analog techniques. These magnificent master tapes are translated back to analog audio and the resultant signals power the same kind of cutting head we have been using for years to cut master lacquers from which stampers are made, which in turn are used to press the discs we buy.

Naturally, the smaller companies who have begun issuing so-called "digitally mastered" discs take great pains to use the finest cutting equipment, the best grade vinyl and pressing masters which are not too overused to stamp out high quality discs. But here we have variables that depend upon the company involved and the economics of record production. There is another problem, too. While the carefully engineered digital master tapes may indeed have music recorded with dynamic ranges of 80 or 90 dB, it is not possible to preserve that dynamic range in even the best vinyl disc. At the bottom of the dynamic range scale we still have some amount of surface noise, while at the high end of the scale, groove modulation must be restricted so that the modulation in one groove does not break through the groove wall and run into the next groove. Of course, variable-pitch grooves help somewhat here, but you can't keep pushing the grooves further and further apart if you intend to get a reasonable play-time out of an LP disc.

Now, let's look at the direct-to-disc recordings. Here we have no intermediary tape process whatever. The performers do their thing and the music signals are directly cut into a master lacquer disc. If the artists goof, it's "take two" or "take five" or as many takes as are needed to get a "straight-through" performance. Some will argue that such recordings have a sense of spontaneity that is lacking in any multi-track originated studio recording; I can't argue with that, except to say that the success of a direct-to-disc recording depends upon *all* the technical and artistic contributors doing a first-rate job all the time. If the performance is perfect but the real-time miking is found to be unbalanced, will the album be released or will the company spend the time and money to try again? Conversely, if there is one sour note in the fifth band on side A but the engineering is perfect, will the producer call for an entire repeat performance?

There are, of course, other limitations. Since no more than a few cutting lathes will ordinarily be found in direct-to-disc cutting rooms, the number of stampers that can be created from the few master lacquers is limited. And so is the number of discs that can be produced, making each direct-to-disc release a sort of "limited edition" album, with prices to match. No chance for a gold or platinum record here! And since big numbers, high profits and artist royalties all figure largely into the record business equation, few top artists thus far have been attracted to the direct-to-disc format. Interestingly, the percentage of classical music recorded using the direct-to-disc process is much higher than it is in the world of mass-produced discs. That's probably because 50,000 or 100,000 sales of any classical disc is deemed a success in any case, and the direct-to-disc process is capable of turning out that number of individual discs of a real-time performance.

What surprises (and annoys) me a bit is the pricing of digitally-mastered records. At the moment, at least, they seem to be priced as high as direct-to-disc records. Since the number limitations do not really apply to digitally-mastered discs (in fact, perfect dubs of digital tape masters can even be shipped around the country by record companies that have multiple pressing facilities), I cannot justify the $14 to $16 price tags associated with these discs. I can see asking a dollar or two more per disc because of the special care taken in the use of top-quality vinyl and the more frequent discarding of overused stampers, but to ask double the price of a mass-produced disc (which such digitally-mastered discs are, in every way) seems to me to be something of a rip-off, prompted, no doubt, by the success of the direct-to-disc records (which can justifiably sell for as much as they do) among knowing audiophiles.

Which Is Audibly Better?

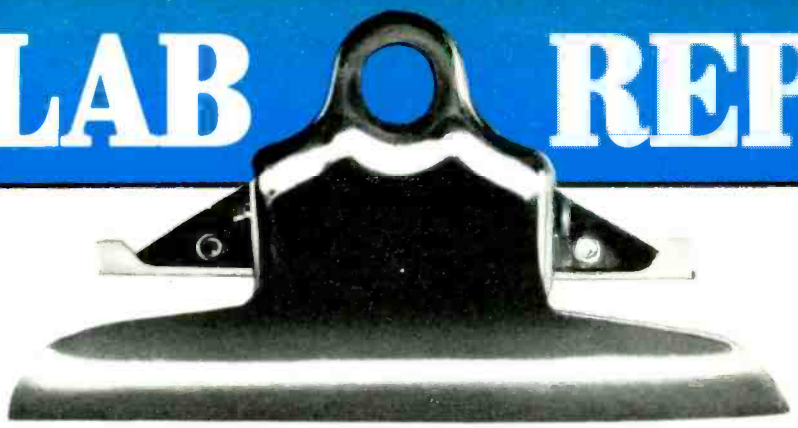
There really is no general answer to our correspondent's second question. Yes, there are times when I can hear an audible difference between direct-to-disc and digitally mastered discs. But it isn't always in favor of one or the other. I have run into some direct-to-disc records where in the rush to capitalize on the direct-to-disc craze, recorded velocities were so high that no cartridge I know of was able to track the peaks when adjusted for reasonably low downward tracking forces. In fact, examining the grooves of some of these records under a microscope, there was evidence of groove-wall breakthrough. Such records may have inherently high dynamic range musical content, but what good is it if you can't find a pickup that will reproduce those dynamics without distortion and mistracking. On the other hand, I would hasten to add that examples of such poorly pressed direct-to-disc records are few and far between. Most of the ones I've listened to are so much better in sound quality and dynamic range than conventionally mass-produced discs that even a tin-eared listener would appreciate their superiority.

As for the few digitally mastered discs that I have heard, (and right now there are fewer of them than there are of direct-to-disc albums, so perhaps the comparison is unfair), I have not found any which deliver lower background surface noise levels than the best of the direct-to-discs. I have heard one or two which exhibited what I can best describe as an electronic sort of coloration of musical instrument sound, but for the most part, those I have listened to are totally "musical" sounding and offer a high degree of realism and accuracy of reproduction.

What About Laser Recording?

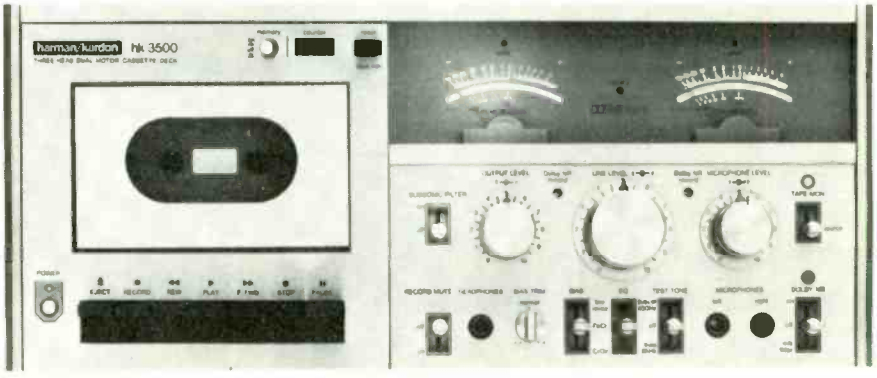
I must confess that it's been more than a year since I had my one and only encounter with what our correspondent calls a "laser recording." That was a demonstration of an optical disc player which Teac displayed at a trade show. My sonic memory, though fairly good over the short term, cannot be relied upon to make a meaningful comparison between that listening experience and more recent encounters with direct-to-disc and digitally mastered analog recordings. I have, however, listened to a variety of program material recently which was reproduced directly from digitally recorded tape and this much I can tell you. If the direct-to-disc format turns out to be an interim recording method because of the growing availability of digital tape mastering systems in recording studios, then even the digitally mastered analog disc may well prove to be short lived. For, without a doubt, once you have heard digitally recorded and reproduced sound which does not have to be reduced to analog disc format for playback you will be as sold on it as I am. But reducing digital audio to disc form involves a lot more than technology. We are, at the moment, faced with a lack of standard disc format (should it be mechanical tracing, capacitance pickup or laser optical pickup), possible obsolescence of countless millions of dollars worth of conventional record making equipment, plus the usual problem of which comes first, the hardware or the software.

How about Laser recording? I think it's great, but I wouldn't consider discarding my turntable, cartridge and LP collection just yet!



NORMAN EISENBERG AND LEN FELDMAN

Harman Kardon hk3500 Cassette Recorder



General Description: The hk3500 is a front-loading stereo cassette recorder employing three heads and two motors. The record and play heads are electrically separate but contained in one housing so there is no need for periodic realignment of the record head.

The cassette well is covered by a swing-down door with a transparent section. The full cover may be removed, if necessary, to get to heads and related parts for degaussing or cleaning. The reset button for the three-digit tape counter is larger than usual and actually lights up with small vertical bars that indicate tape motion. The memory switch, if activated, will stop the tape in rewind when the counter reaches 000.

The transport keys operate through mechanical linkages. It is possible to key directly between rewind and fast-forward or from play to either of the fast wind modes. But to go into record, or into play, the stop button must be pressed first.

The deck's peak-reading meters are amply sized and calibrated from −40 to +6 dB. Above each meter is an LED that shows transient peaks which may be too sudden or brief for the meters to follow. A third LED, between the meters, serves as a record mode indicator.

All signal level controls (line in, mic in and output) are dual-concentric, permitting individual or simultaneous channel adjustment. The two input controls are fitted with a numbered ring that may be rotated for reference points. Line and mic inputs may be mixed using the deck's controls.

To adjust the hk3500 for different tapes, there are separate switches for bias and for equalization, with markings for low noise, FeCr, and CrO_2. In addition, there's a bias trim knob which HK recommends using each time you record on a tape type different from what you used before. The adjustment is made with the aid of a test signal of 8 kHz (built-in and activated by another switch) and a bias-calibrate portion included on the signal meters.

The Dolby NR switch includes positions with and without multiplex filter. To calibrate the Dolby NR system, a test signal of 400 Hz (activated by another setting of the test-tone switch) is available. Screwdriver adjustments are made for suitable readings on the two meters.

Additional front-panel controls include a subsonic filter, the tape/source monitor switch, and record-mute switch that disconnects the incoming signal from the tape during record for deliberate sections of silence or "pre-editing." Left- and right-channel mic inputs, and the stereo headphone output are on the front panel.

Line inputs and outputs are on the rear of the deck. There are two sets of inputs—"high" and "low." The high inputs are for the usual signal levels from typical program sources. The low inputs may be used for added gain from a source, if needed.

Also on the rear are Dolby NR playback calibration adjustments, factory pre-set but possible to change as explained in the owner's manual. A final adjustment at the rear permits varying tape speed on playback, if desired. The AC power cord completes the picture.

Test Results: Performance of the hk3500 in our tests was excellent with both low-noise (standard or LH) tape and with chrome-bias tape. The former tape

exceeded the specification for response; the latter confirmed it exactly. The chrome tape had better S/N; the standard tape had a dB more of headroom in the midrange, and a jot less distortion—although distortion with either tape was very, very low.

With its two line level input options, and its ample line output of 1.5 volts, the hk3500 should interface readily with a wide variety of signal sources and playback systems.

Although the transport is mechanically operated (rather than solenoid operated), it appeared in our tests to be rugged and reliable. Its speed acuracy was near perfect; its wow-and-flutter figure was very low.

The subsonic filter switch on the front panel suggests that the low-frequency record/play capability of the hk3500 is unusually good, and indeed it is. The data summarized in our table of vital statistics only partially tells the performance story. For example, our measured wow-and-flutter figure of 0.04% WRMS which, in itself, is rather spectacular for a cassette deck in this price category, is actually more impressive than it seems. What small amount of wow-and-flutter exists is of an extremely "smooth" character, with virtually no cyclical spikes or peaks. Such peaks, though they might not alter the WRMS wow-and-flutter readings, would be audible to the critical listener. The readings observed without the weighting network in the circuit (0.08%) are significant in this respect.

Figure 1 is a plot of frequency response, using a TDK AC-337 test tape, recorded for playback using a 120-microsecond EQ setting. Although a continuous curve is shown, this tape contains a series of spot frequency checks from 40 Hz to 12.5 kHz. Deviation from "flat" response was no more than 1.0 dB in either direction from 40 Hz to 8 kHz, and was down 2.0 dB at the 12.5 kHz test point.

Record/play response, using Maxell UDXL I sample tapes, was within ±3.0 dB from 19 Hz to 19 kHz. Thanks to the three-head configuration of the hk3500, we were able to plot this response directly on our spectrum analyzer, as shown in Fig. 2. The lower trace of this 'scope presentation represents a record/play level of −20 dB. Harman Kardon offered us some of its own data, in which the record level was at −30 dB and under those circumstances, response extended all the way out to 21 kHz for the −3 dB roll-off point. Since it is our normal practice to measure response at −20 dB relative to "0 dB" on the supplied record-level meters, we did so here. Obviously the difference in response represents the beginnings of tape saturation (at the high end) rather than any problem of the record electronics or head capability or alignment. Bear in mind that "0 dB" on these peak-reading meters corresponds to a full 200 pWb/mm (Dolby level) rather than the 185 or 165 pWb/mm used as an arbitrary 0 dB reference by some other tape deck makers.

The advantage of the UDXL II sample tape (again, by Maxell) shows up in the signal-to-noise readings (61 dB without Dolby, 69 dB with Dolby "on") though

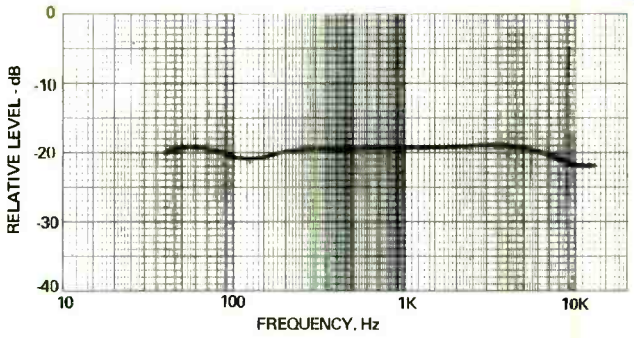


Fig. 1: Harman Kardon hk3500: Playback response (120 usec. EQ) using TDK test tape AC-337.

some diminution in high-end response is experienced as a tradeoff, with the −3 dB point occurring at 17.5 kHz when record level is at −20 dB. A plot of response using this tape is shown in Fig. 3.

Upper traces of both Figs. 2 and 3 represent record/play response at a full 0 dB record level and are simply intended to show the tape saturation characteristics of the two types of tape tested, at the high frequency extremes.

Individual Comment by L.F.: I was, frankly, amazed at the number of control features which Harman Kardon has managed to incorporate in its new model hk3500, a unit which retails for well under $600. Here is a 3-head, dual-motor design in which at least as much emphasis has been given to accuracy of musical reproduction as to control features. Tapes recorded on this machine *sound* good.

The twin peak-reading expanded scale record level meters help, of course, with their expanded (−40 to +6 dB) scales, as do the peak overload LEDs, the continuously variable trim bias control (which operates as a vernier, over and above the three fixed bias settings),

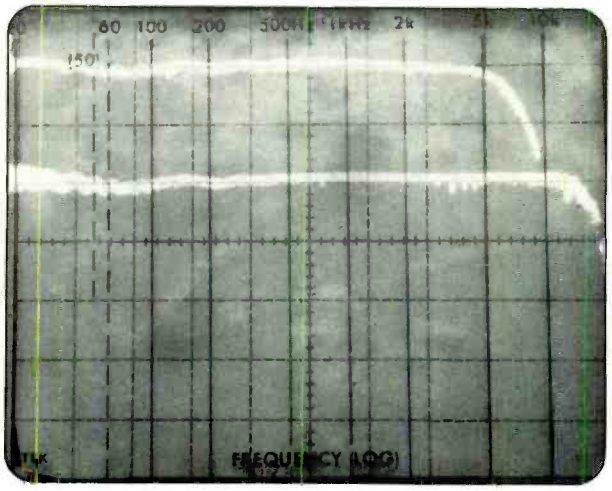


Fig. 2: Harman Kardon hk3500: Record/play response at 0 dB and −20 dB record level using Maxell UDXL I tape.

the built-in low-frequency (400 Hz) and high frequency (8 kHz) test tones for optimization of Dolby and bias settings, and the fact that tape monitoring "as you record" is possible. Harman Kardon uses a twin (but separate) record and play head in a single housing, which means that azimuth alignment is not necessary, as it is with decks using physically separate record and play head shells. Another advantage of this approach is the extremely short time lag which occurs between actual recording and "read-out" of recorded results by the "play" section of this dual head. When A-B'ing between source and tape, the real-time difference is negligible, enabling the recordist to make precise aural comparison between the two sounds.

I mentioned earlier that the sound quality of recordings made on the hk3500 was particularly good. Harman Kardon maintains that this is largely attributable to its having engineered the unit for wide-bandwidth and extremely accurate phase response. HK has been espousing this design approach for its other audio products for many years and only in recent years has its intuitively correct design approach been quantified and characterized by such terms as "lower TIM distortion" and the like. In an effort to confirm some of the reasons for the excellent sound quality we heard in our listening tests, I recorded a 500 Hz square wave on tape and played it back, photographing the results in Fig. 4. In this 'scope photo, the 'scope face was used as a regular oscilloscope, and the frequency notations should be ignored. Readers who are accustomed to seeing 'scope photos of square waves taken at the output of preamplifiers or amplifiers may find the results (lower trace is playback of square wave, upper trace is input signal) somewhat unimpressive. Let me assure you, however, that these results are very impressive for a cassette deck. You can easily confirm that fact by trying to record a square wave at this frequency on competitive cassette decks in the same price range and you'll soon see what we mean.

The availability of the vernier bias control on the hk3500 not only lets you optimize the bias for any given tape but gives the user some measure of choice with respect to recording parameter preferences. For example, we could easily have extended response for the UDXL II tape had we been willing to sacrifice distortion and signal-to-noise somewhat by slightly underbiasing for the tape in question.

When you read the description of controls and features provided on this deck and add to that the excellent measured results of our tests and the superb listening quality, you must conclude, as I did, that Harman Kardon has indeed come up with a worthy successor to their highly acclaimed hk1000 and hk2000 decks of bygone days.

Individual Comment by N.E.: With its better-than-average audio response, very low wow-and-flutter, excellent signal-to-noise characteristics, lower-than-average distortion and very ample headroom, generous headphone signal output and equally full line output, combined with very high speed accuracy, full three-head operation but without the need for alignment of the record head, expanded peak-reading meters, built-in facilities for both Dolby and bias calibration and more—the hk3500 at its price (which is low for today's market) may just be what a lot of quality-minded but bargain-seeking recordists have been waiting for. This model seems to embody a cumulative know-how in terms of audio technology and product design that would be something to take notice of in any event, and is almost unbelievable in view of the price the unit is tagged with.

Our test measurements of the 3500 speak for themselves. They either confirm or exceed the manufacturer's specifications. But measurements aside, there is the sound of the hk3500 which, even at 1⅞ ips speed and given the relative limitations of the cassette format vis-a-vis open-reel tape, is simply splendid. HK's "secret ingredient" in this area is, of course, the design philosophy known as "wideband response," first introduced some years ago in their first Citation amps and preamps, and since becoming a well-known, if debated,

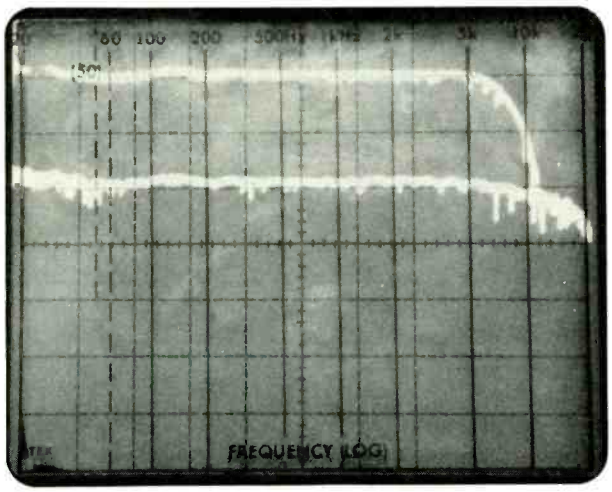


Fig. 3: Harman Kardon hk3500: Record/play response at 0 dB and −20 dB record level using Maxell UDXL II tape.

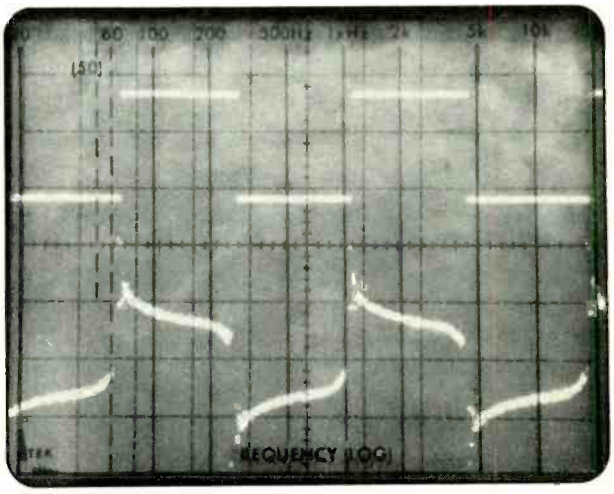


Fig. 4: Harman Kardon hk3500: Lower trace shows playback of 500 Hz square wave recorded on cassette tape using the hk unit.

design approach. The general idea behind this approach is to provide an audio bandpass that well exceeds the "normal" audio range. There are both "hard technical" as well as "psychoacoustic" arguments for this approach, and much discussion (both verbal and published) has ensued for and against it. I believe that it is an important element in audio circuit design, related to such desiderata as improved phase characteristics and measurably lower TIM distortion.

To return to the hk3500 recorder itself—the only aspect that might not appeal to the more advanced recordist is its lack of fast-buttoning for the transport. Apparently, this is a major area of product design in which manufacturing economies can be realized and passed on to the buyer. But the absence of solenoid feather touch controls does not penalize the transport in terms of its basic requirements for accuracy of speed and for very low wow and flutter. It does mean of course that you must press the stop button before changing transport modes, and you do not have the convenience of run-in or flying-start recording. In view of all the other sterling performance aspects and useful features of this deck, and especially in view of its price, the lack of fast-buttoning may be overlooked by many a buyer. Surely, I know of no machine in this price class that rivals the hk3500.

HARMAN KARDON hk3500 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, standard tape	±3 dB, 20 Hz to 17.5 kHz	±3 dB, 19 Hz to 19 kHz
Chrome	±3 dB, 20 Hz to 17.5 kHz	±3 dB, 20 Hz to 17.5 kHz
Signal-to-noise, Dolby off std/CrO_2	NA/NA	58/61 dB ("A" wtd)
Signal-to-noise, Dolby on std/CrO_2	65 dB/NA	67/69 dB ("A" wtd)
THD at 0 VU, std/CrO_2	1.3%/1.3%	0.7%/0.8%
Record level for 3% THD std/CrO_2	NA/NA	+6 dB/+5 dB
Bias frequency	105 kHz	105 kHz
Mic input sensitivity	0.5 mV	0.5 mV
Line input sensitivity	200/50 mV	190/58 mV
Line output level	1.5 V	1.5 V
Headphone output level	NA	430 mV/8 ohms
Wow and flutter (WRMS)	0.05%	0.04%
Fast wind time, C-60	60 sec.	67 sec.
Speed accuracy	0.75%	0.01%
Power consumption	NA	15 W

CIRCLE 23 ON READER SERVICE CARD

Crest Audio Model P-3500 Power Amplifier

General Description: The Crest Audio model P-3500 is a stereo (two-channel) power amplifier. Power output, over a bandwidth of 1 kHz to 20 kHz is rated at 250 watts per channel into 8-ohm loads, or 400 watts per channel into 4-ohm loads.

The front panel contains two meters made up of peak-reading LEDs. Each meter uses twelve LEDs of which there are five green, four yellow and three red. There are separate level controls for each channel. A display control enables the peak power readings to appear as a "bar" (all LEDs up to the peak level will flash) or as a "dot" (only the highest reading LED flashes, based on peak program levels). Finally, there is a sensitivity control which provides peak readouts even at very low levels of gain. In conjunction with a meter calibration switch on the rear panel, this control provides for either 8-ohm or 4-ohm calibration.

In the 8-ohm position zero-dB equals 250 watts. In the 4-ohm position zero dB equals 400 watts.

The rear of the amplifier contains both an XLR and a quarter-inch phone jack connector for inputs on each channel. Outputs are binding posts on each channel. A switch changes the amplifier from stereo to single channel operation with a rated output of 800 watts. In the monophonic mode the P-3500 can drive any load, including 70-volt lines in commercial applications. The rear panel also contains an individual circuit breaker for each channel for protection without need for fuses.

Inasmuch as the P-3500 uses two completely separate power supplies, it is effectively two separate mono amplifiers. Amplifier circuitry is direct-coupled. The P-3500 contains a ventilating fan which is fairly audible. However, in most professional situations the amplifier would be used far enough away from loudspeakers and the audience so it would not be heard.

General construction of the amplifier is quite

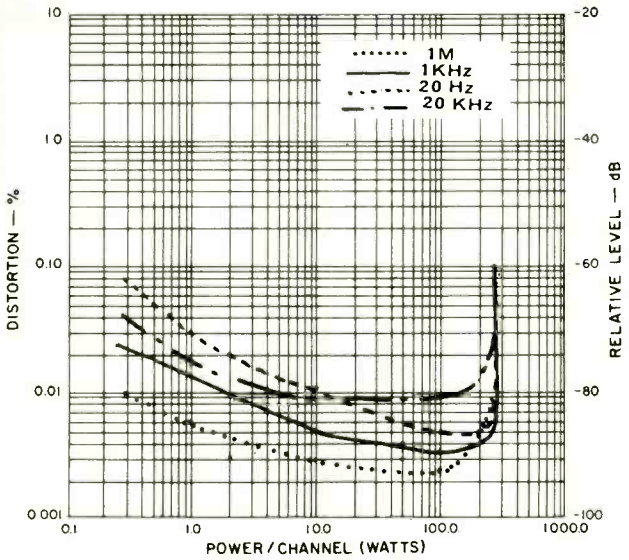


Fig. 1: Crest P-3500: Power output vs. distortion, 8-ohm loads.

rugged. The chassis is made of 16-gauge cold-rolled steel. The front panel is a standard 19-inch rack mount width using $^{3}/_{16}$-inch thick aluminum. Internal modular construction facilitates servicing and repairs.

Test Results: In our tests, the P-3500 easily made its rated power output—and then some—and with lower distortion than specified. The residual levels of distortion in our Sound Technology 1700 analyzer are about as high as the readings obtained for rated output, so in all probability the amplifier is doing even better than the extremely low distortion measured.

Figure 1 shows power output versus THD (at three frequencies) and IM for 8-ohm loads. The clipping point (0.1% THD) for a 1-kHz signal with 8-ohm loads and both channels driven measured 276 watts. With 4-ohm loads it went as high as 470 watts.

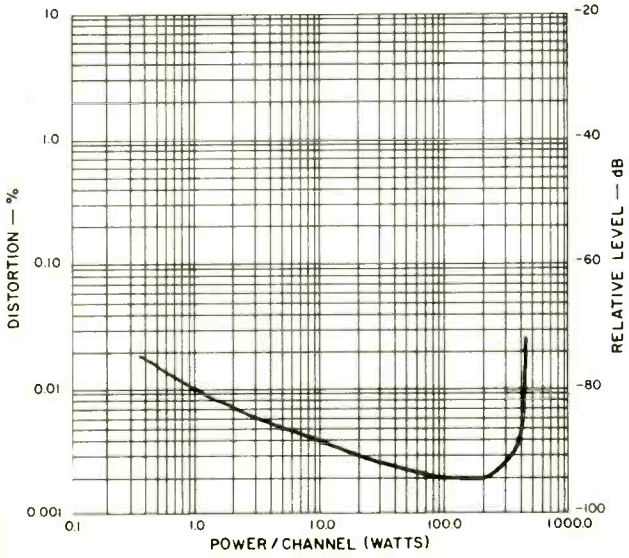


Fig. 2: Crest P-3500: Power output vs. distortion, 4-ohm loads.

Figure 2 shows power output versus THD for the 4-ohm condition and a 1 kHz signal. Figure 3 is a graph of harmonic distortion versus frequency at rated output (250 watts per channel) with 8-ohm loads. As the graph shows, THD remains extremely low even down to 20 Hz and it rises only slightly at the high frequency end of the spectrum, confirming the amplifier's published high slew rate.

General Info: Dimensions are 19 inches wide, 3½ inches high; 17 inches deep. Weight is 54 pounds. Price of the amplifier is $1,059.

Individual Comment by L.F: I never cease being amazed at the amount of audio power that can be crammed into a relatively small cubic volume these days. Crest's P-3500 is an excellent example of this

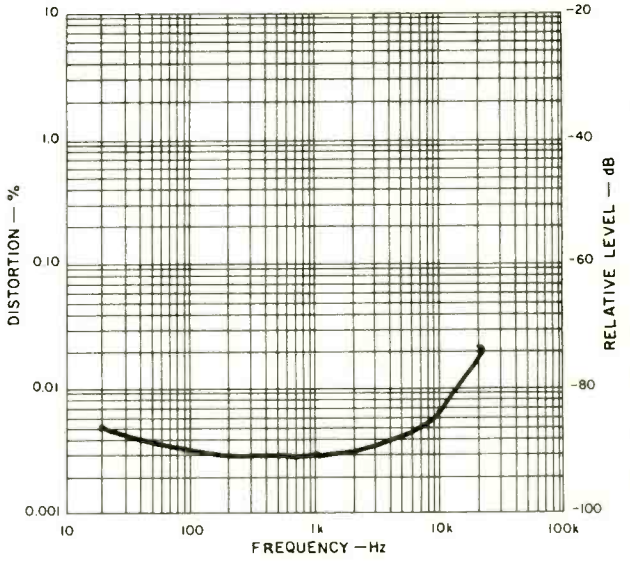


Fig. 3: Crest P-3500: Harmonic distortion vs. frequency, 8-ohm loads, at 250 w/ch. output.

trend, occupying no more than 3½ inches of vertical panel space in a standard 19-inch rack. The amplifier also is another in the growing list of high powered amps which employs two completely separate power supplies so that aside from the common power cord and on/off switch, you are dealing here with two separate mono amplifiers.

I particularly liked the LED metering system which enables you to display peak power levels either as a "bar" or as a "dot." The convenient sensitivity adjustment on the front panel eliminated the frustrations I have experienced when trying to use LED power displays that only begin to flash when you approach rated output, or the ones which have so many dB between adjacent LEDs that you really can't tell where you are at. This also is the first amp I've encountered in which the LEDs can be made to read accurately with either 4-ohm or 8-ohm loads.

Circuit breakers (one per channel) instead of fuses are a feature that I think every professional amplifier should have. The Crest P-3500 has them, and I'll just bet that many a sound engineer will bless Crest for not

having to carry around a spare set of fuses (or several sets) in case the "inevitable" occurs on location or in a studio. I couldn't confirm Crest's claim for TIM distortion simply because I have not as yet set up my lab to make this measurement (the industry hasn't agreed completely on how to measure TIM though the square-wave/sine-wave method used by Crest seems to be gaining in acceptance), but I can tell you that based upon my listening tests with the amp, using some of my most recently acquired direct-to-disc program material, I have no reason to disbelieve their claims. Transient response was superb and even when I deliberately tried to push the amp into clipping levels, recovery was so fast that I was able to detect neither ringing nor overhang.

Individual Comment by N.E.: The Crest Audio P-3500 power amp apparently combines the ruggedness and operational features of professional grade equipment with the kind of sonic performance that would appeal to the true audiophile. There is much to admire in this amplifier. Especially commendable are the LED displays—their accuracy and versatility; the use of circuit-breakers instead of fuses; the facilities for quick changeover from 8 ohms to 4 ohms, and for quick changeover from stereo to mono.

CREST AUDIO MODEL P-3500 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power output to 20 kHz 8 ohms/4 ohms	250 W/400 W	260 W/416 W
THD from 250 mW to rated output, 8 ohms/4 ohms	0.03%/0.05%	0.003%/0.004% (see text)
1-kHz power at clipping, 8 ohms/ 4 ohms	275 W/450 W	276 W/470 W
IM distortion up to rated power, 8 ohms	0.02%	0.004%
Transient IM, 8/4 ohms	0.01%/0.03%	NA
Frequency response	−3 dB @ 100 kHz	−3 dB @ 60 kHz
Slew rate	35 V/μsec.	40 v/μsec.
Rise time	3 μsec.	2.8 μsec.
Damping factor	300 or better	260 (see text)
Hum and noise re: rated output	−100 dB	−105 dB (−100 "A" wtd)
Input sensitivity for 250 W out	2.2 V	2.35 V
Input impedance	20 K ohms	confirmed
Meter sensitivity	0 dB to −52 dB	confirmed

CIRCLE 24 ON READER SERVICE CARD

dbx Model 2BX Two-Band Dynamic Range Expander

General Description: The model 2BX is a stereo dynamic expander that divides the audio frequency spectrum into two bands and offers an increase in dynamic range of up to 50 percent (ratio of 1 to 1.5). The device has applications in listening to music as well as in tape-recording from other sources, the latter activity involving the use of a suitable noise-reduction system such as the dbx 120 or 150 series. The 2BX is patched into a sound system at the tape monitor signal level; the exact hookups to make depend on what other equipment is in the system. Details are given in the owner's manual.

Operating controls on the front panel include the AC power off/on switch; a slider for the amount of expansion; another slider to adjust transition level for allowing signals to be upward or downward expanded. Between the two sliders are two rows of LEDs that indicate the amount of gain change in each of the 2BX's two frequency bands. Red LEDs show upward expansion (volume increase); yellow LEDs show downward expansion (volume decrease). These LEDs are used in adjusting the transition level control.

Additional front panel controls include four push-switches to select signal sources and to select expansion of a program before recording it.

All signal jacks and the power cord are at the rear.

The 2BX uses RMS detection, which senses the root-mean-square value of the input signal. This method is claimed to be capable of responding accurately and precisely to all changes in input level while not over-reacting on musical transients or noise spikes. Linear expansion is used, which means that the same expansion ratio (as selected on the slider) is applied to the incoming signal regardless of its dynamic content or level. The manufacturer calls this unit a "true stereo" expander. This is based on the use of one control voltage (derived from the sum of left and right input levels) to provide the same amount of linear expansion to both channels so that the stereo image is preserved.

Test Results: Manufacturer's specs for the 2BX were confirmed or exceeded in bench tests of the unit. Distortion figures were lower, and frequency response was wider, than dbx claims.

To study some of the action of the 2BX we fed a fixed tone (1 kHz) to its inputs and monitored the results on our spectrum analyzer (Fig. 1). The analyzer

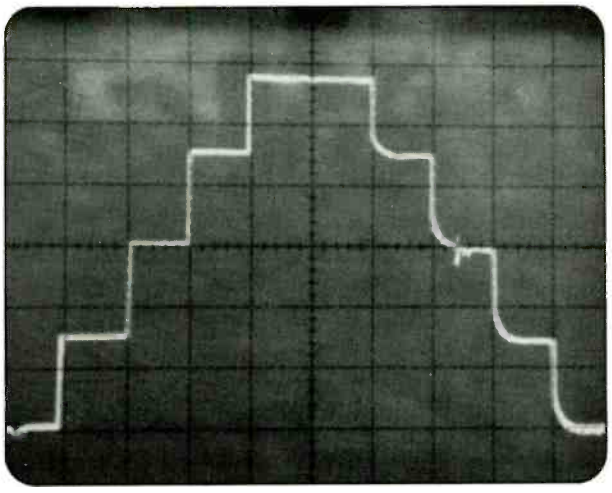
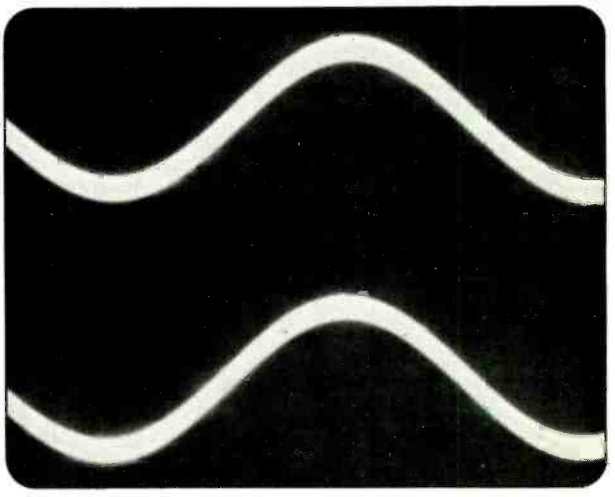
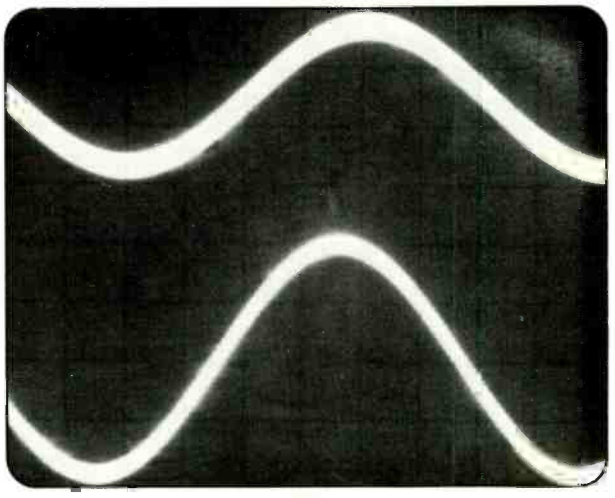


Fig. 1: dbx 2BX: Amplitude "steps" at left were measured at output of 2BX as we manually increased input levels in 10 dB increments. Each output increment change equals 15 dB (expansion control was set for maximum). Decay rate (at right) is seen to differ from attack rate.

was "tuned" manually to that fixed 1-kHz frequency (non-sweeping mode), so that the traces seen represent level changes only, rather than changes of frequency. Each vertical division on the 'scope graticule represents a change of 10 dB. As the trace began to sweep from left to right, we periodically *increased* the input signal level in 10 dB steps. As you can see, the output, however, increased in exact 15-dB increments (the expansion rate was set for maximum, or 1.5). Note the sharp attack time for the upward steps at the left. After traversing an input range of 40 dB (which becomes almost exactly 60 dB as seen in the output of the 2BX), we reversed the procedure, stepping back down in 10 dB increments. Again, each downward step resulted in a 15-dB downward level shift at the output of the 2BX, but this time, the decay time required to



(A)



(B)

Fig. 2: dbx 2BX: Lower trace in (A) shows output complex signal (low and high frequencies in 4:1 ratio) is identical to input with 2BX control set for no expansion 1.0:1.0 In (B) output (lower trace) has undergone upward expansion of larger, low-frequency component while high-frequency signal component has been "downward" expanded. This action would not be possible with a single-band expander.

reach stability is noticeably longer. While our 10-dB level transitions were manually accomplished, it is clear that with different wavefronts or "envelopes" other results might have been obtained and changes in attack as well as decay times would have been observed. It is clear that the attack and decay times need not (and should not) necessarily be identical.

To illustrate the ability of the expander to treat different frequencies in a different manner, we used a readily available test signal—the familiar low/high frequency mix used for checking IM distortion (60 Hz and 7,000 Hz, in a 4:1 ratio). Since the 2BX can be adjusted for specific thresholds (below which downward expansion takes places and above which upward expansion occurs), we set the threshold so that the 60-Hz signal was already great enough to be in the

upward expansion range. Since the 7 kHz signal was ony one-fourth as great in amplitude, that automatically placed it in the downward expansion region. In Fig. 2A, the upper trace represents the input signal, as described. With the expansion ratio set for 1:1 (no expansion), the lower trace of Fig. 2A, which was taken from the output of the 2BX, is an exact replica of the input, as it should be with expansion set to 1:1.

Next, the expansion control was set to maximum, or 1:1.5. The upper trace in Fig. 2B again represents the input signal. But now, the output signal consists of an increased-amplitude low-frequency signal (which has been upward expanded, based upon our setting of the threshold control) and a diminished-amplitude high frequency signal (which has been downward expanded). Obviously, the 2BX has applied different modes of expansion to the two frequencies contained in the test signal. It doesn't take much imagination to realize how effective this action can be when applied to actual music signals, and our own listening tests confirmed that feeling.

General Info: Dimensions are 17¾ by 3¾ by 10½ inches. Weight is 8 lbs., 5 oz. Price is $450.

Individual Comment by L.F.: Ever since I first heard dbx's model 3BX expander more than a year ago, I have been convinced that multiple-band expanders are the way to go if you really want to experience the benefits of dynamic expansion without having to suffer the side-effects of intrusive pumping and breathing. The 3BX still stands out as the best example of a linear expander for home use that I have ever heard or tested. However, not everyone can afford $650 for an add-on expander.

Having now tested and listened to the dbx 2BX I can honestly say that it is the closest thing to the 3BX that I have tried and, the savings afforded ($200) in opting for this lower-cost linear expander are substantial. As most readers probably know, dbx expanders, such as the 2BX we tested, employ RMS level detection and provide *linear* expansion. For this reason, they are extremely easy to set up, requiring only that the user adjust threshold (the level at which the action of the device changes from downward expansion to upward expansion) and degree of expansion (in this case, variable from none to 1.5, in dB).

Part of the reason why less sophisticated expanders tend to "breathe" or "pump" has to do with their attack time and release time. No matter how sophisticated an expander circuit may be, it is, after all, acting upon "instructions" which are based upon signal levels reaching it, and cannot act instantaneously.

While a rapid attack and release time may be suitable for many types of music, different attack and release times may be desirable. As the dbx owner's manual explains it, a smooth classical string quartet might sound best when expanded with slow attack and release times. Attack and release times should be allowed to vary according to the program content, for the most natural sound. The 2BX's attack and release times actually follow the rate of change of the program envelope. Furthermore, the attack and release times are scaled differently in each of the 2BX's two frequency bands.

Individual Comment by N.E.: Listening tests, in addition to the lab measurements, confirm the effectiveness of the 2BX. Doubtless, a very important contributing factor to its "unobtrusiveness" when used in a music-reproducing system is its two-band design. The frequency spectrum is divided into two bands which are treated separately. This can become very germane to expanding music that has strong percussives. By separating the lower frequencies from the middles and highs, the 2BX does not allow the bass to influence vocals or mid-range instruments. With single-band expanders, in contrast, the entire signal content would be expanded when something like a low-pitched drum roll came along, since these devices are governed by the overall signal level. The 2BX, which senses and acts on signal levels in two different bands, can do its expansion more naturally. And this becomes especially significant for music that varies widely in dynamic range to begin with, and which is complex in its orchestration or vocal mix.

dbx MODEL 2BX TWO-BAND DYNAMIC RANGE EXPANDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Expansion ratio	1.0 to 1.5, linear, in dB	confirmed
Transition level range	30 mV to 3 V	15 mV to 3 V
Frequency response at 1.0:1 expansion	± 0.5 dB, 20 Hz to 20 kHz	± 0.5 dB, 20 Hz to 35 kHz
THD at 1.0:1 expansion	0.1%, 20 Hz to 20 kHz	0.022% at 20 Hz; 0.02% at 1 kHz and at 20 kHz
THD at maximum expansion	NA	0.10% at 20 Hz; 0.037% at 1 kHz; 0.045% at 20 kHz
IM distortion (no expansion)	0.15%	0.02%
IM distortion (full expansion)	NA	0.18%
Input impedance	50 K ohms	confirmed
Max output level	6 V	7 V
Power consumption	20 W	18 W

CIRCLE 25 ON READER SERVICE CARD

Otari MX-5050-B Open Reel Tape Recorder

By John Murphy and Jim Ford

General Description: The Otari MX-5050-B half-track recorder is the successor to the MX-5050 open reel recorder and offers several improvements and new features. Since we had been favorably impressed by the MX-5050 we were eager to check out the new "B" version and see what changes had been made.

Our initial impression was quite positive. Otari has retained the front panel bias, EQ and record level calibration adjustments which made the MX-5050 such an easy machine to set up, and they've added a pitch control, a peak reading LED, a memory rewind and transformer balanced floating outputs.

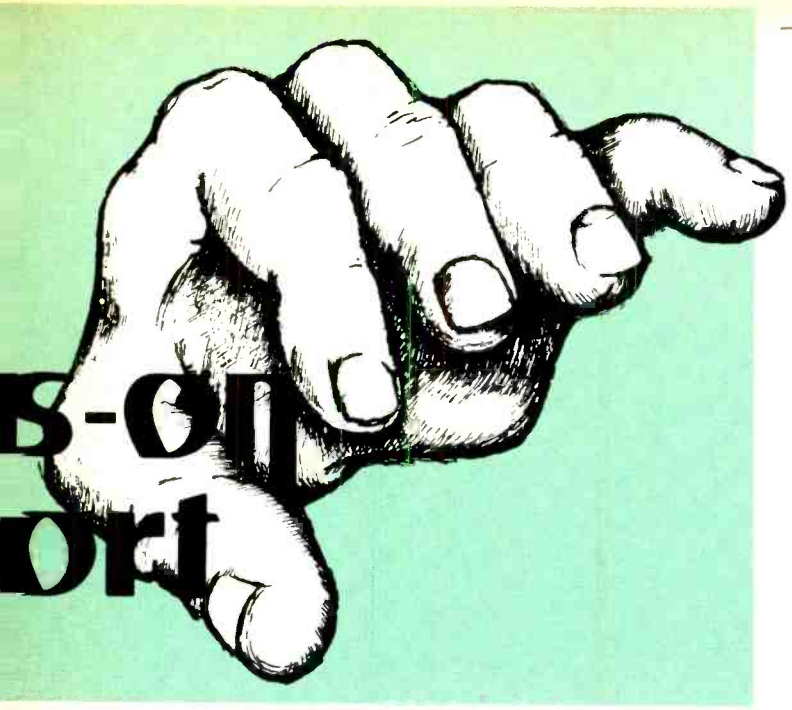
Editing is easy on this machine because of the edit control and the splicing block right on the front panel. The head cover is hinged to allow easy access to the heads for cleaning or alignment.

The tape transport system utilizes two 6-pole induction motors to drive the reels and a DC servo motor for direct capstan drive. Tape speeds of 15 ips or 7½ ips can be selected by way of a front panel switch. Motion sensing is used in the transport control logic so that the machine can be switched between fast wind modes and play without having to go through stop and without an annoying time delay between modes.

The combination of front panel adjustments and a built-in 1 kHz test oscillator make it rather easy to optimize bias and record level settings for any particular type of recording tape. However, an outboard 10 kHz generator is required to calibrate the record EQ. There is an extra playback head which can be selected for the playback of 4-track tapes.

The front panel layout is very logical making it easy to use the machine, and easy to get excellent results.

The pitch control is located just above the head cover and provides ±7% speed change. It is activated by pulling the control knob out and turning it clockwise or counterclockwise to increase or decrease pitch (tape speed), respectively. When the control is active a red LED next to the knob lights up. When the knob is pressed in, the LED is extinguished to indicate that the transport is operating at the calibrated tape speed. This control may be used during either record or play and is quite useful for synchronizing the pitch of previously recorded music with the pitch of instruments to be recorded on the other track. For example, say track one contains an excellent guitar performance and you want to record a piano part on track two. You set the machine up to monitor the guitar track off the record head (by punching the "sel. rep." button for channel one) so that the piano can be recorded in time synchronization with the guitar. Then you notice that the guitar was tuned flat with respect to the piano (*rats!*). Normally you would have to either tune the guitar to the piano and then re-record the guitar track (taking a chance on not being able to recreate that inspired performance) or re-tune the piano to the flat guitar. Absurd! However, with a pitch control on the



recorder you simply increase the tape speed slightly to bring the flat guitar up to standard pitch and you've salvaged your inspired performance without having to re-tune your piano. What a lifesaver! (And it's *much* easier to "tune" the recorder than to tune the piano.

Just below the left reel there is an adjustable cue control which defeats the tape lifters during the fast wind modes. This allows the operator to hear the tape and search for selections while fast winding. But be careful. With the lifters fully defeated the output of the machine is *loud* and can eat your tweeters.

At the bottom of the cue control is a tape counter and memory on-off switch. When engaged, the memory works in conjunction with the counter to stop the transport when the counter passes "0000" in the rewind mode. This feature is very helpful in applications where it is necessary to return to the same tape location many times. At the top of the head cover is a switch allowing for selection of either the 2-track or 4-track playback head.

On the left side of the machine and just below the head cover there is a row of buttons which control (from left to right): POWER (on/off), SPEED (high/low), REEL SIZE (large/small) and EDIT. The machine comes set up for 15/7½ ips operation but can be internally switched over to 7½/3¾ ips operation. The reel size switch is used to select tape tension for either 2¼- or 4½-inch diameter reel hubs. Otari states that the same size reels should be used on both turntables. When the EDIT function is selected, a green LED EDIT indicator to the right of the button lights up. The take-up reel and tension arm switch are defeated and tape is allowed to spill between the capstan and take-up reel. The EDIT mode can be entered from either the play mode or from stop. When in the stop mode, the EDIT button is depressed first, then the PLAY button is pressed. Exact edit points can be located by rocking the reels back and forth. Splicing operations are then facilitated by the built-in splicing block on the head cover. These features make the machine highly suited for production work.

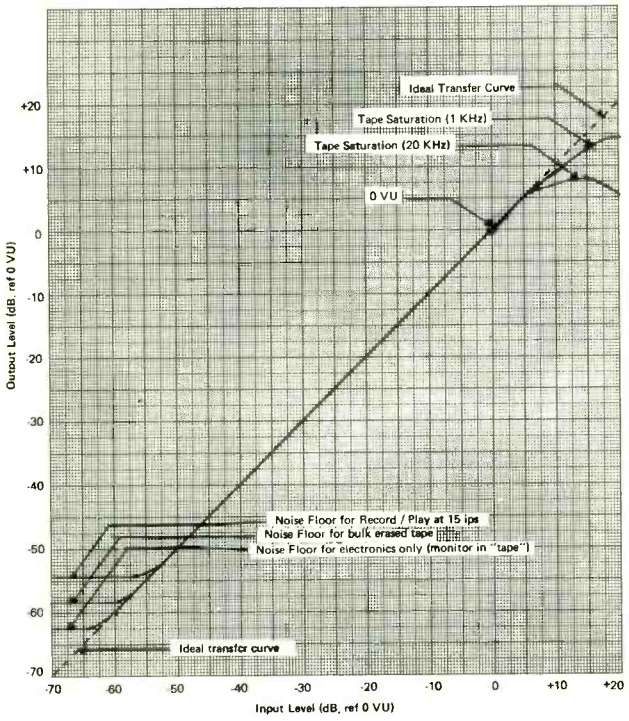
On the right side of the machine just below the head cover are the transport operating controls. From left to right these are: RECORD, PLAY, STOP, REWIND and FAST FORWARD. To the left of the RECORD button is a red LED record indicator. (More about this indicator later.) Below the push buttons on the left side of the machine are the input level controls. There is a pair of concentric mic/line level controls for each channel and mic and line inputs can be mixed if desired. To the right of the input level controls are a pair of VU meters. These are calibrated from −20 to +3 VU and each meter is equipped with a red LED peak indicator. To the right of the VU meters is a pair of concentric output level controls, one control for each channel. Just below these is a slide switch for selecting either "standard reference level" (SRL) output or variable output level. With the switch in the SRL position, the output level controls are defeated and the output is fixed at +4 dBm for a balanced 600-ohm line (when the signal level is at 0 VU). The output level can be reduced to −10 dBm by a rear panel switch to facilitate interfacing with low-level equipment. There is also a record level switch on the back panel for selecting low, medium or high record levels. These represent a 0 VU flux level of 185, 250 and 320 nWb/m, respectively. Otari gives a chart of recommended settings for various types of tapes in the owner's manual. For the low record level setting, 0 VU corresponds to Ampex operating level. The medium and high record level settings are +2.6 dB, and +3.8 dB with respect to the low record level. These are actually record level calibration adjustments to fine tune the machine to various types of tape. (Some types of tape can tolerate higher record levels than others.)

At the bottom of the front panel and to the far left are a pair of red push button switches which separately enable recording in the two channels. Each switch has a red LED just above it which lights whenever recording is possible for that channel. Whenever either channel is ready for recording, the record indicator (next to the RECORD operating control) flashes to indicate that at least one of the channels is prepared. Recording does not begin until at least one channel is prepared and the RECORD button is depressed, along with the PLAY button at the start of tape motion, or after the PLAY mode has been entered if "punch-in" recording is desired. In either case the record indicator changes from the flashing mode to continuous illumination to indicate that recording is in progress. The machine can be taken out of the record mode by either depressing the STOP button, in which case tape motion ceases, or by depressing the PLAY button ("punch-out") in which case recording stops but tape motion continues. Punch-in and punch-out are used for

overdubbing in conjunction with the synchronous recording feature.

Just to the right of the record preparatory switches are a pair of switches labeled SEL REP (for "selective reproduce"). By depressing either of these, the user selects synchronous playback of that channel. That is, the tape is played back through the *record* head of the channel for which SEL REP is selected. This enables the operator to record on the opposite channel in time synchronization with the program monitored through the SEL REP channel. Recording is disabled on any channel for which SEL REP is selected regardless of the position of the record preparatory switch. A green LED lights up above either switch when SEL REP is selected for that channel.

Across the bottom of the machine to the right of the SEL REP switches are small round holes which provide screwdriver access to adjustments for record bias



(Ch.1/Ch.2); record EQ (high speed/low speed, Ch.1/Ch.2); and record level (Ch.1/Ch.2). Otari provides excellent instructions in the owner's manual for the proper calibration of these adjustments. To the right of the calibration adjustments is a phone output jack for driving 8-ohm (or higher) headphones. Next to the phone jack is a momentary push button for the test oscillator. When depressed, a 1 kHz test signal is injected into the line input. The level of the signal is controlled by the line input level controls and can then be used to perform bias and record level calibrations. It can also serve as a level calibration tone for other equipment interfaced to the output of the recorder. At the bottom and far right of the front panel are source/tape monitor switches for the two channels. In the *out* (SOURCE) position the input signal is fed to the outputs and the meters. In the *in* (TAPE) position the tape playback signal is fed to the outputs and the meters of the machine.

The rear panel has a remote control connector to the far left through which all of the transport operations can be remoted. There is a chassis ground terminal beside this connector and input/output connectors on the right side of the rear panel. Input/output connections are made by way of XLR-type connectors. From left to right these connections are: Line Output (Ch.2/Ch.1); Line Input (Ch.2/Ch.1); and Mic Input (Ch.2/Ch.1) There are four slide switches along the bottom of the back section. The first of these is line output level low/high (−10 dBm/+4 dBm). Just to the right of the output level switch is an equalizer switch for selecting either IEC or NAB equalization. Further to the right is the record level switch for selecting low, medium and high record levels as discussed earlier. In the far right bottom corner is a three-position mic attenuator switch; the three positions are labeled 0 dB, −20 dB and OFF. Otari recommends leaving this switch in the off position when the mic inputs are not in use; this practice will keep any noise from the mic preamps out of the system.

Field Test: We had no problems interfacing the recorder with our reference system. We wanted to give the unit a tough test of audio quality so we chose as our program input a very good direct-to-disc recording with which we were already familiar. It would be quite an accomplishment if the recorder could reproduce such excellent music without audible degradation. We put the disc on our turntable and switched in the recorder. First we kept the recorder's monitor switches in the source position and switched the recorder in and out at our preamp. This allowed us to evaluate the recorder's electronics separately from its tape system. We were careful to set the recorder's line input level controls for the same loudness level with the machine switched in as with it switched out of the system. Listening with our best ears we were hard pressed to detect any sonic differences when the recorder was introduced into the system. However, we did agree that there was something slightly "different" about the low bass with the machine in the system. The change was so slight that we can't really say more than that with confidence. So, the audio quality of the electronic section is quite good.

Next we loaded some Ampex 406 tape on the machine so that we could perform the same listening test on the machine's tape system. (We had previously set the machine up for this tape.) Record levels were adjusted so that the peak indicator just flashed occasionally. We then adjusted the output level controls for the same loudness level when the machine was switched in and out at our preamp. We could then listen to the disc directly through our preamp or, at the touch of a button, introduce the recorder (tape *and* electronics this time) into the chain with no change in loudness level. This would have been a challenging test

Jaco Pastorius | Acoustic Amps

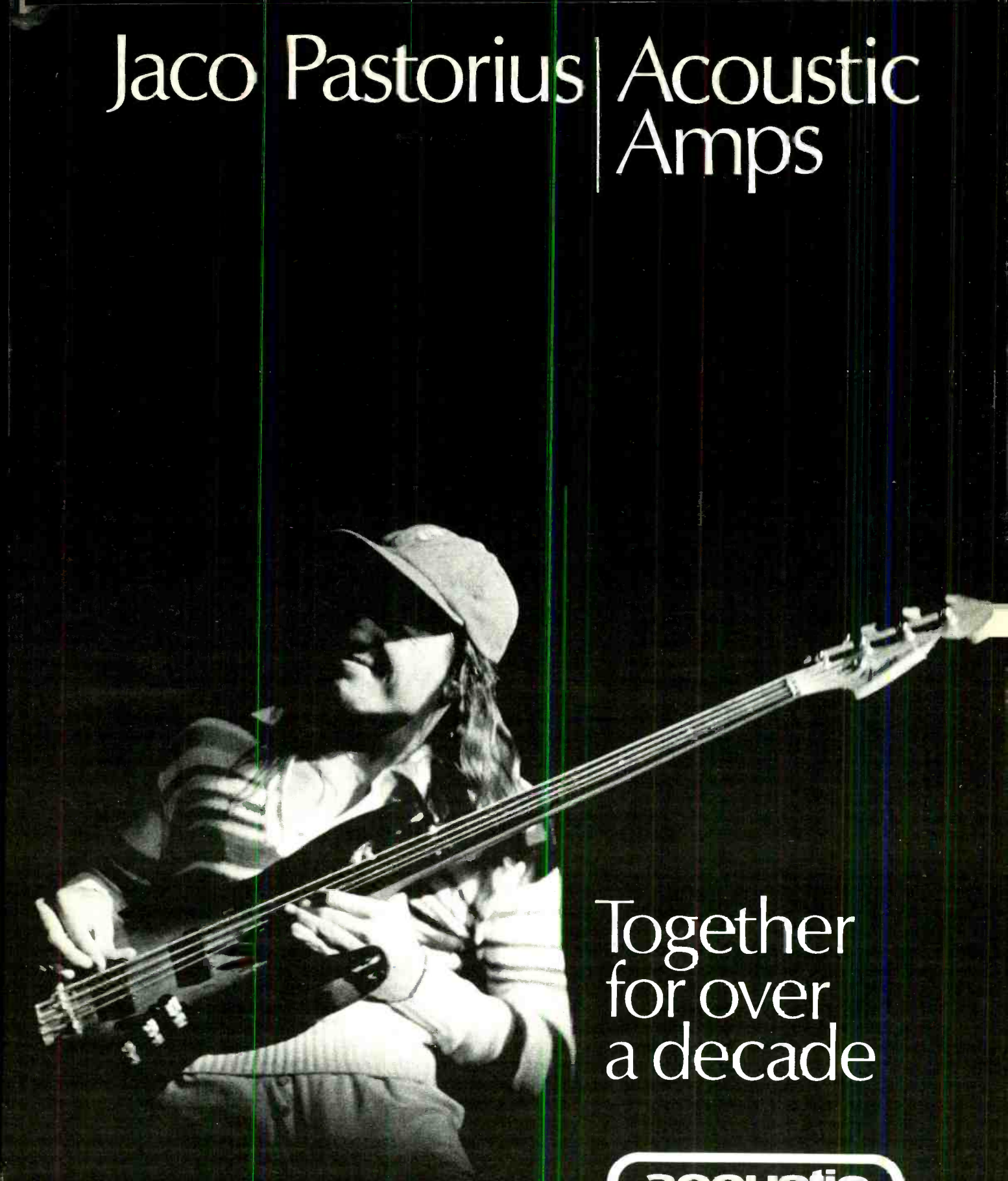


photo: Gorm Valentin

Together for over a decade

acoustic

professional sound equipment

On stage: 330 top/408 enclosure/361 bass system
Off stage: 727 stereo amp/648 & 626 studio monitors

7949 Woodley Avenue
Van Nuys, California 91406

CIRCLE 54 ON READER SERVICE CARD

for any piece of audio gear. Again, we listened critically for any audible differences when the recorder was switched in. Except for the slight time delay the recorder introduced (time for the tape to travel from record head to play head) we couldn't be sure if we were switching in the recorder or not. It was that good. Even the cymbals (which are usually very difficult for a recorder to reproduce without coloration) sounded the same with the recorder in as with it out. There was still something changing in the low bass when we switched the Otari in but it didn't seem much different from when we listened through the recorder's electronics alone. This constitutes truly first class performance for any piece of audio equipment but especially for a tape machine. The only time it became easy to detect when the recorder was in the system was during quiet parts in the music and between cuts on the disc when the tape hiss became audible.

Up to this point we had kept our record levels kind of conservative so we decided to push the levels and see what would happen. We increased the record level enough to make the peak overload indicators flash regularly and peg the VU meter occasionally. Performing the comparison again we noticed a clearly audible degradation in sound quality. We concluded from this that it's best to heed the warning of the peak LEDs and avoid pegging the VU meter. We think it's worth sacrificing a few dBs of signal-to-noise ratio (by using more conservative record levels) to preserve the high audio quality that the 5050-B is capable of.

The machine's controls had a good feel and the LEDs provided good visual indication of the various modes of operation. All of the machine's features performed as expected and there were no surprises. We were really quite impressed.

Lab Test: After getting the recorder on the test bench we calibrated the bias, record EQ and record level (for Ampex 406 tape) according to instructions in the owner's manual. From there we proceeded with the usual tests. The test results can be found in the "Lab Test Summary" section. The record/playback frequency response of this machine is outstanding at either tape speed. The signal-to-noise ratio is strictly tape dependent. Referenced to 3% THD level, the S/N ratio is about 62 dB at 15 ips, and 64 dB at 7½ ips. The dynamic range of the machine can best be seen in the representation of the input/output transfer curve. Distortion is also strongly dependent on the tape used.

The 0 VU distortion level was about half a percent, increasing to about one percent at +3 VU. The three percent distortion level was reached at around +7.5 VU with the peak indicators lighting at +6 VU.

The input and output levels are standard (+4 dBV) and the output level can be reduced to −10 dBV if desired. Thus, interfacing with other pro and semi-pro equipment should pose no problems.

Although Otari states that either high or low impedance mics can be used with the machine, we recommend that low impedance mics be used with an in-line transformer to provide a high impedance input to the machine. Using a typical low impedance mic (Shure SM58) directly into the unit and talking about six inches from the mic we found that the mic preamp noise was only about 33 dB below 0 VU. Using an in-line high impedance transformer this figure improved to 47 dB. A recorder of this caliber deserves to be used with a high quality mic mixer in order to take full advantage of the machine's capabilities.

Serviceability of the unit looks pretty good as there is good access to the circuit boards.

Otari provides an excellent owner's manual with detailed explanation of the various operating modes and features of the machine.

Conclusion: The MX-5050-B is a truly excellent half-track tape recorder. The new features, such as pitch control and memory rewind make it even more flexible than the previous version. It is well suited to production work and can do synchronous recording. We give it our highest recommendation. Good work, Otari!

LAB TEST SUMMARY

(*Note:* All dBV values are referenced to .775V)

Frequency Response for Playback Only

15 ips tape speed
Two-track head: (Ch. 1) ±1.2 dB 30-15 kHz
(Ch. 2) ±1.4 dB 30-15 kHz
Four-track head: (Ch. 1) ±1.5 dB 50-15 kHz
(Ch. 2) ±2.0 dB 50-15 kHz

7½ ips tape speed
Two-track head: (Ch. 1) ±2.6 dB 50-15 kHz
(Ch. 2) ±2.5 dB 50-15 kHz
Four-track head: (Ch. 1) ±2.5 dB 50-15 kHz
(Ch. 2) ±2.6 dB 50-15 kHz

Playback Through the Record Head (Sel. Rep.)

15 ips tape speed (Ch. 1) ±3 dB 50-15 kHz
(Ch. 2) ±3 dB 50-15 kHz
7½ ips tape speed (Ch. 1) ±3 dB 60-8 kHz
(Ch. 2) ±3 dB 60-8 kHz

Record/Playback Frequency Response

(*Note:* All tests were made with Ampex 406 tape with the record level switch in the medium position)

15 ips tape speed (0 VU record level)
Ch. 1: ±1 dB 25-29 kHz
Ch. 2: ±1 dB 26-29 kHz
7½ ips tape speed (−10 VU record level)
Ch. 1: ±1 dB 20-22 kHz
Ch. 2: ±1 dB 20-22 kHz

Record/Playback Signal-to-Noise Ratio (unweighted)

(*Note:* All S/N values are referenced to 0 VU)

15 ips tape speed (1 kHz test signal)
Ch. 1: −54.1 dB
Ch. 2: −54.6 dB
7½ ips tape speed (1 kHz test tone)
Ch. 1: −56.8 dB
Ch. 2: −56.8 dB

(continued)

Innovate your sound

with sunn®

For more than a decade we've pushed the limits of electronic amplification and sound reinforcement equipment. We've established ourselves as the sound innovators. Our commitment is to develop imaginative concepts that give you the opportunity to discover your own unique sound. But it's really more than that. Our amps and sound systems *are* different. They're built for you. But, we go at it like they're built for us.

That's why we applied C-MOS (Complimentary MOS) circuitry to our Beta amplifiers. C-MOS was first used in the computer industry. We had no idea there was music in C-MOS until we tried it. The result is a unique overdrive circuit that is rich in harmonic content, much like tube distortion. But C-MOS does more than tubes. Distortion can be added in any amount, at any volume, so you don't destroy your ears to get that hot sustained sound you want. Yet in the clean mode the Beta is clearer, tighter than tube sound, so the characteristics of your instrument shine through.

Most amplifier people use standard speakers. We don't. We make our own so we can capture the subtle sounds that make each of our products unique. And that is what makes your sound unique. Because you can count on those differences to bring out your best. Play the fantastic Beta Series at your Sunn Sound Dealers.

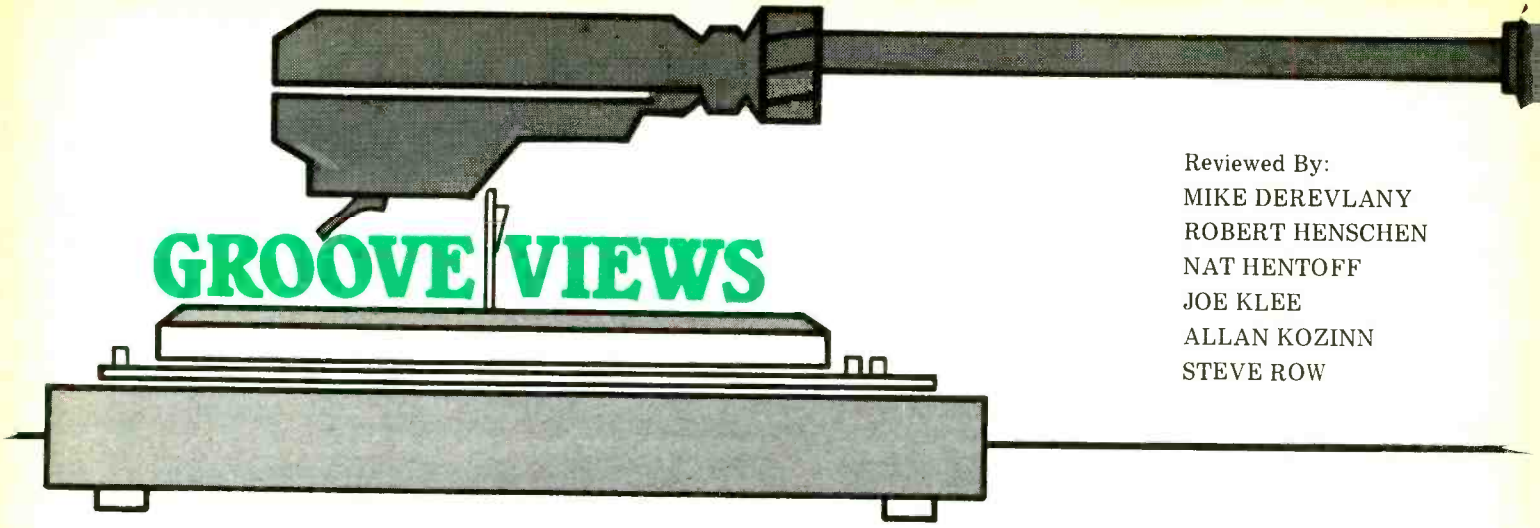
Sunn. The Sound Innovator.

Beta Series Self-Contained Amps

Sunn endorsees include Chris Squire of Yes (shown above), John Entwhistle with The Who, Steve Buslowe with Meat Loaf, John Deacon with Queen, Joaquin Lievano and Jamie Glaser with Jean Luc Ponty

Sunn Musical Equipment Company A Hartzell Corporation Company Amburn Industrial Park Tualatin, Oregon 97062

CIRCLE 40 ON READER SERVICE CARD



Reviewed By:
MIKE DEREVLANY
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE
ALLAN KOZINN
STEVE ROW

POPULAR

PAGES: ***Pages.*** [Bobby Colomby, producer; Michael Verdick and Scott Singer, engineers; recorded at Location Recording Service, Burbank; Heider/Filmways Studio, Hollywood; Sound City A, Van Nuys; Larrabee Sound, Hollywood; A&M Studios Mix #1, Hollywood, Ca.; CBS Recording Studio B, New York, N.Y.] Epic JE 34459.

Performance: **Natural pop groove**
Recording: **Buttery warm**

This debut album took five and a half months to record and approaches studio perfection. Pages got their official start as the backup band for Andy Gibb's first tour, but the individual members of this band have built an impressive list of support credentials over the years. Their abilities as a group were immediately obvious to former Blood Sweat & Tears drummer/producer Bobby Colomby, who subsequently turned Dave Grusin, Philip Bailey (of Earth, Wind & Fire) and the Brecker Brothers onto them. The finished Pages product is definitely commercial (packed with potential AM radio hits) but also solidly constructed and highly enjoyable.

Buttery warm cuts like "Clearly Kim," "This Is For The Girls," and "If I Saw You Again" are immediately contagious, subtle mixtures of light rock and a touch of soul. Richard Page is an excellent vocalist capable of near-falsetto highs and mellow lows; his superb voice arranging has been influenced by years of close work with Modernaires great Allen Copeland. Richard's duet with Lani Groves on "It's Alright," and Bailey's contribution to "If I Saw You Again," are cases in point.

Instrumentally, Pages is equally enticing and the boys hint at hidden strengths. "Love Dance," for instance, is a two-minute non-vocal tease, with Dave Grusin's introductory strings making way for a broiling Latin percussion jam and then just a glimpse of electric jazz-rock fusion. The piece is reportedly several minutes longer in concert. "Room At The Top" is heftier, led by bassist Jerry Manfredi and guitarist Peter Leinheiser. Steve George and leader Page handle keyboard duties, blending lightweight, tasteful synthesizer colors with a glowing abundance of Fender Rhodes. Grusin's string "Interlude" adds yet another dimension to side two, and his other charts are compatibly groovy.

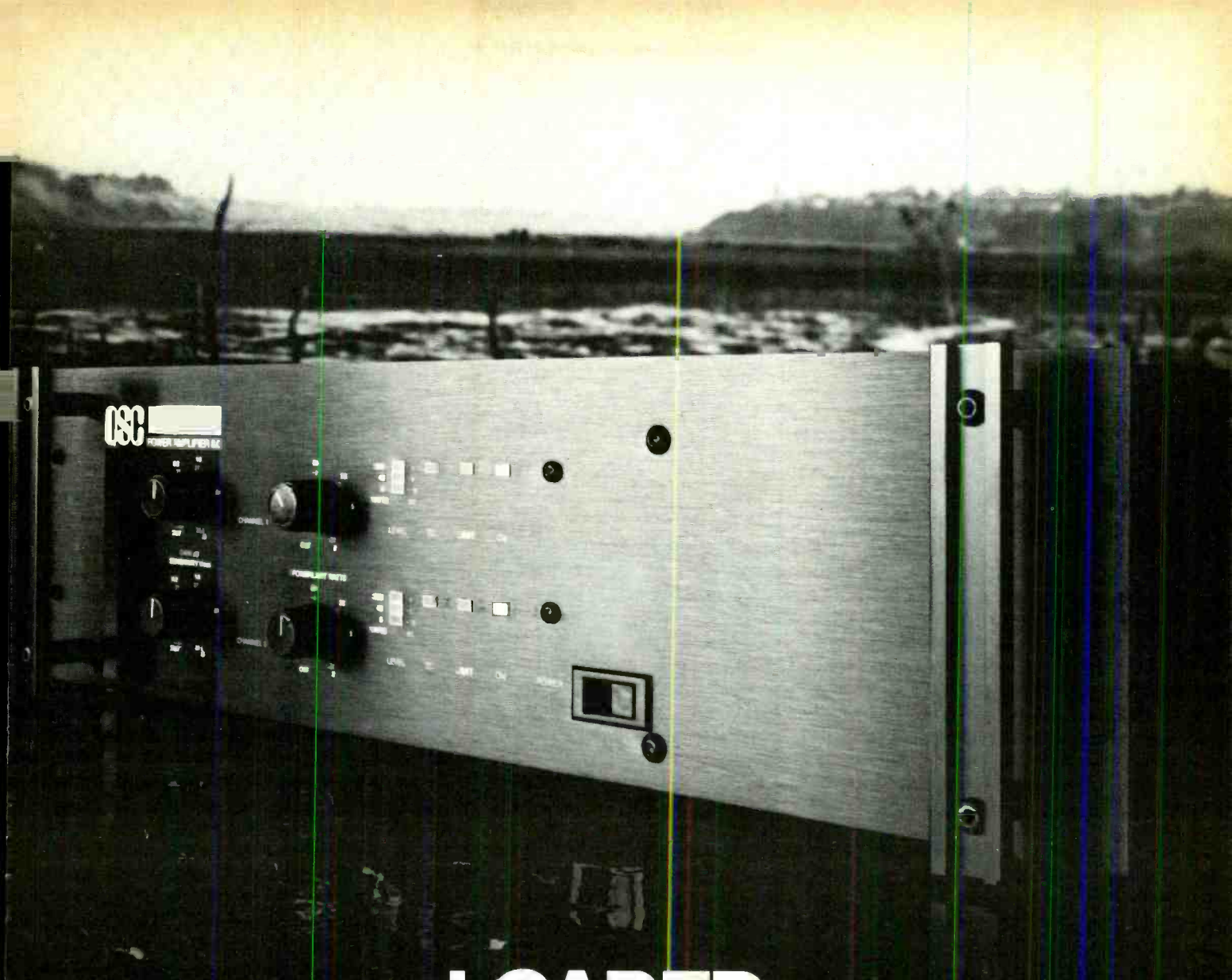
Producer Colomby has guaranteed that the melodic attraction and skillful musicianship are rendered in attentively precise fidelity. There's a relaxed, natural aura surrounding the Pages sound, and Verdick's silky mix helps maintain the buoyant, hip appeal that elevates this platter above mere crossover commerciality. True, Pages will be enjoyed by fans of relaxed soul and catchy pop alike, but the melodies have enough depth to encourage many listenings. R.H.

PAGES: Buttery-warm, solidly constructed and enjoyable

10CC: ***Bloody Tourists.*** [Eric Stewart and Graham Gouldman, producers; Eric Stewart, Keith Bessey, and Tony Spath, engineers; recorded at Strawberry Studios, South Dorking, Surrey, England.] Polydor PD-1-6161.

Performance: **9cc's**
Recording: **100cc's**

When 10cc split in half a couple years back, it was immediately apparent that the two ensuing aggregations (Lol



LOADED... THE 600 WATT NO-OPTION AMP.

People kept asking us "How about a high-power amp with low distortion that's loaded with options and doesn't cost an arm and a leg?" We listened to them and set out to build "The Complete Amp" with reliability, power, specs, features, and price. We've succeeded. Our reputation has been built on the design and construction of cost-effective gear combining maximum performance with simplicity and reliability. Now QSC offers a package you can't find in any other amp, REGARDLESS OF PRICE OR OPTIONS. The A 8.0 delivers 300 watts of clean power to each channel (20-16 kHz with less than .09% THD rising gradually to 0.2% THD at 20 kHz into 4 ohms) and 600 watts into 8 ohms with the same specs in the bridged-mono operation.

Features include: PowerLimit Controls; Fan Cooling; 3-way Load Protection; LED displays for level, distortion, and limiting indicators; Balanced Inputs with XLR type 3-pin connectors; and Outputs with 5-way binding posts, phone jacks, and speaker protection fuses. Ask your Pro-Audio Dealer about the A 8.0 or write directly to us for a free brochure detailing the incredible features and specifications of this exceptional new power amplifier from QSC.

QSC
AUDIO PRODUCTS
1926 Placentia Avenue
Costa Mesa, CA 92627
714/645-2540

CIRCLE 80 ON READER SERVICE CARD

Creme and Kevin Godley still perform as a duo on Mercury) wound up with about 50% of the original group's comedic ability. While Eric Stewart and Graham Gouldman *have* managed to keep 10cc's formidable engineering prowess on an even keel, they haven't quite matched the utter brilliance and hilarity of *Sheet Music* (London AUKS 53107), *The Orginal Soundtrack* (Mercury SRM-1-1061), or the single "I'm Not In Love." And yet, the Stewart-Gouldman tandem has come back to within one or two cc's of their previous proficiency, and they still demonstrate the legendary 10cc studio skills.

The best cut on the album, "Dreadlock Holiday," starts *Bloody Tourists* off in strange musical surroundings. Although 10cc has shown some prior history of island music ("Hotel" from *Sheet Music*), never before have they unleashed a straight ahead reggae tune with this kind of clout. "Dreadlock Holiday" not only shows off a classic Gouldman one-liner in "I don't like reggae. . . I love it," the cut also features a potent instrumental side of today's 10cc: Gouldman (bass, guitar, percussion); Stewart (keyboards); Rick Fenn (guitars, keyboards); Stuart Tosh (drums); Duncan MacKay (keyboards); and Paul burgess (percussion). This opening track is absolutely infectious, the most danceable Caribbean beat since Marley went disco.

At every turn of the dials, 10cc exerts contined engineering mastery over their diversified product. The melodic ode to "Tokyo" is technologically sophisticated but also romantic and mystical. "Old Mister Time" and "Everything You've Wanted to Know About Exclamation Marks)" flaunt great instrumental mixing of odd key and speed changes. "From Rochdale To Ocho Rios" is Latinesque and breezy, "For You and I" captures gorgeous Polymoog and pop harmonies, and other tunes take clever turns. The whole LP is quite listenable. But when the wit suffers occasional letdowns, or becomes a bit too obvious ("Shock On The Tube"), it's the marvelous sound that carries this album. R.H.

DEVO: *Q. Are We Not Men A. We Are Devo.* [Brian Eno, producer; Conrad Planck, engineer; recorded at Conny's Studio, Koln, Germany and at Different Fur Studios, San Francisco, Ca.] Warner Bros. BSK 3239.

Performance: **Different but unispired**
Recording: **Ordinary**

Q. Are we not conscious?
A. No, we are victims of Devo hype.

And amidst the turbulent throes of the hype is a bunch of clones from Akron, Ohio who dream of the day that they can become so much animated plastic and conquer the world with their own verions of Muzak. Yet in the face of what amounts to an overt claim of conformity with their prefabricated sound, the music of Devo more than occasionally manages to somehow rise well above the kinetic energy level of lead.

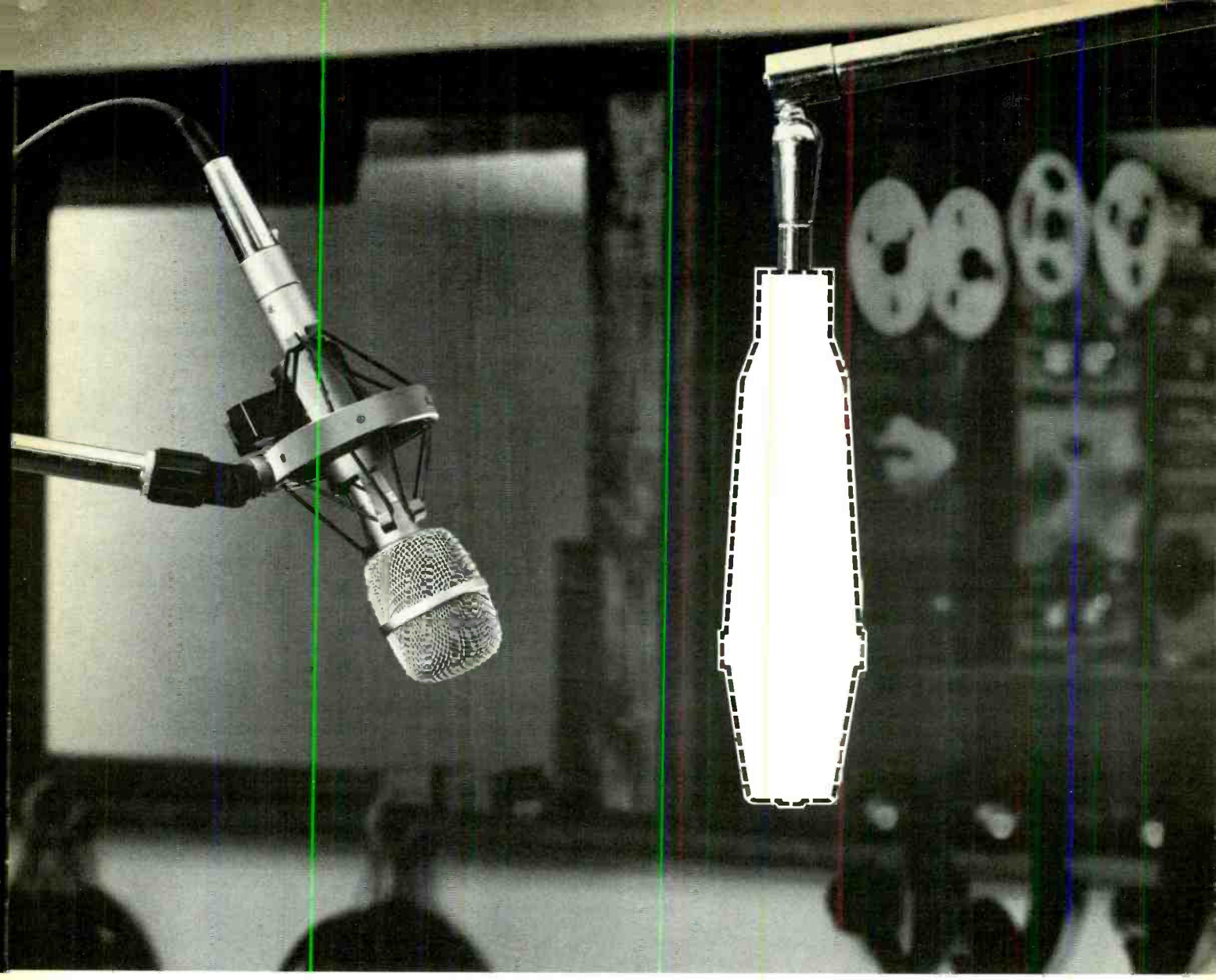
And that's the strange thing about this album. Despite vocals that range from mediocre to downright terrible, inappropriate mixes, and miserable judgement in their selection of material (which unwittingly led to Devo's lobotomized abortion of the Stones' "Satisfaction"), Devo is interesting, if only because of their novelty, and in their most enlightened moments become an amusing form of entertainment. The reason Devo has this unexpected quality is that much of their material is uninhibited solid rock or almost as good spaced-out electronic rock, accompanied by assorted interesting drek with a lesser quantity of just plain boring drek and horrible filler. Maybe it shouldn't work at all (and it probably wouldn't in a slightly different situation), but with a healthy dose of beginner's luck and the helpful expertise of producer Brian Eno, a fair amount of it does.

The Devo songwriting formula isn't hard to figure out. Take the rasping, gasping screams of a turtle being run over by a 10-ton Mack truck, throw in a few randomly chosen chords strong-armed from various instruments, add a few well-garbled lyrics with a vague semblance of social relevance that have been hijacked from the Sex Pistols and/or the Ramones, and mix thoroughly at high speed. Add mania to taste and presto! Devo!

As you may have wondered (that is, if you're numb enough to have the amount of interest in five genetic factory rejects to wonder about them), the name Devo actually has a meaning. Of sorts. It stands for a doctrine (joke? threat? therapy?) known as de-evolution. It's hard to extrapolate any real sense of what Devo means by de-evolution but it appears to have something to do with the decay of society and, more importantly, the decline in mass musical taste which Devo (quite rightly) believes it represents. As punk rockers on a space kick they prove that even mediocrity can have some value. But then I suppose it's because this de-evolution process takes time. Maybe if Devo de-evolved another couple of million years. . . M.D.

DEVO: Representing a de-evolution in mass musical taste?

JIMMY BUFFETT: *You Had to Be There.* [Produced by Norbert Putnam. Mixed by Marty Lewis on the Enactron Truck, with assistance from Stuart Taylor, Bradley Hartman, Lon



10 to 1 you'll like ours!

Every studio needs a $1,000 microphone. It tells everyone you're serious about good sound, and it impresses the talent.

But when the session gets under way, all that counts is results. Not price tags. And judged *only* by what you hear, the new AT813 Unidirectional Electret Condenser from Audio-Technica is going to truly impress you...at about 1/10 the cost you might expect.

Recently a recording engineering class at a major university made simultaneous multi-track tapes comparing our AT813 side-by-side with some of the best microphones money can buy. The informed and critical students did the judging. Surprisingly, in most cases they couldn't find a nickel's worth of difference. And some preferred the AT813 sound to the expensive competition.

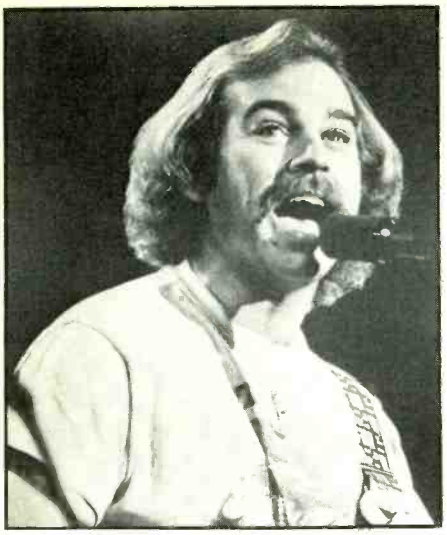
You may agree with these student engineers when you hear the AT813. Or you may not. But at these odds, you can't afford to overlook us. And for new studios just getting underway, who can't afford a roomful of top-dollar microphones, the AT813 is an effective way to achieve full multi-mike capability. Write today for the AT813 spec sheet and A-T microphone dealer list. Then see your Audio-Technica sound specialist for great sound...right from the start.

audio-technica.

INNOVATION □ PRECISION □ INTEGRITY

AUDIO-TECHNICA U.S., INC., Dept. 69MR, 33 Shiawassee Avenue, Fairlawn, Ohio 44313

CIRCLE 102 ON READER SERVICE CARD



JIMMY BUFFETT: In the pocket

Neuman, Donivan Cowart, Martin Cowart; recorded "live" at the Fox Theater, Atlanta, Ga., Aug. 8-10, 1978, and Maurice Gusman Cultural Canter, Miami, Fl., Aug. 14-16, 1978. Remixed by Elliot Scheiner at Air Recording Studios, London.] ABC AK 1008/2

Performance: **What "laid back" is all about**
Recording: **Great "live" feel**

If nothing else, a Jimmy Buffett concert is fun, and this double "live" album has captured just about all the fun one could imagine from his concerts. Salty language, relaxed playing, good-natured patter between songs and some occasionally potent lyrics are all part of *You Had to Be There.*

Buffett's lyrics are superior to his melodies, and he has included some of his best lyrics here: "Margaritaville," "Come Monday," "God's Own Drunk (a lengthy talking blues that may just be the highlight of the album), "A Pirate Looks at 40" and "Changes in Latitudes, Changes in Attitudes." Taken without the music, these verses can stand on their own and even won points for their whimsy and for their sensitivity.

The music, however, can drag the whole song down a bit. "Pencil Thin Mustache," an up-tempo novelty-blues song, starts out just like "Keep On Trucking" and sounds pretty much like that chestnut all the way through. "Son of a Son of a Sailor" is an interesting song, but "Wonder Why We Ever Go Home" is built on virtually an identical melody line which lessens the latter's effect.

Buffett is strong on light-hearted novelty songs, interjected with wry observations as typified by "Why Don't We Get Drunk" and a new song entitled "Morris' Nightmare." Basically, it is good time music with a regional flair about the little people of the American Southeast and Gulf Coast. There is a sun-baked lightheadedness to much of his material, and the music is pushed by a roadhouse beat.

The music has a decidedly blues feel to it, too, because of the prominence of Greg "Fingers" Taylor, a fine harmonica player. On more than one song, Taylor works some musical magic with his playing and adds a different dimension to the material.

The album also features several hot guitar solos, and Buffetts's acoustic guitar playing on side three is the best of the recording. In fact, the material on that side—"Grapefruit-juicy Fruit," "God's Own Drunk," "He Went to Paris" and "The Captain and the Kid"—is the best on the album.

When Buffett moves into higher gear, the effect is that of a rollicking good time, such as on "Tampico Trauma" and the rousing finale, an instrumental version of "Dixie Diner" that features a call-and-response between the guitar and

harmonica at the outset.

Where this album is strongest, however, is the excellent way in which the feel of a "live" concert has been captured on record. A lot of Buffett's talk between songs is preserved (much of it some good-natured, X-rated comments and asides) and the listener also senses the presence of an audience during the musical material. The audiences for the concerts from which the recordings were made were in Buffett's pocket just about the entire time (perhaps because of the geographical affinity with him and his material), and the listener can't help but get caught up in the positive concert atmosphere.

Buffett has been knocking around for quite some time, and it's taken a while for him to emerge from cult status to that of an established recording star. This album is a well-made, well-conceived and logical step for him and is one of the better recorded concert albums produced. Buffett's fans won't be disappointed, and he might even win some new ones with this. S.R.

1994: *1994.* [Jack Douglas, producer; Jay Messina, engineer; Sam Ginsberg, assistant engineer; recorded at Record Plant Studios, New York, N.Y.] A&M SP 4709.

Performance: **White-hot, pulsing**
Recording: **Dense**

Nothing delicate can be found in the quartet known as 1994, from Karen Lawrence's wailing vocals to Steve Schiff's remarkable guitar work, this group has fashioned a heavy metal album that just does not quit. One may hear a little of Lydia Pence and Cold Blood, or Ann Wilson and Heart (particularly as on "Barracuda") or even Led Zeppelin or Black Sabbath in the dense, guitar-heavy songs here, but the effort is not without its own merit.

The band has been together just a little more than a year. Its members knocked around primarily on the West Coast performing in other bands. John DeSautels, the drummer, and Karen Lawrence wrote the song "The Prisoner" that Barbra Streisand eventually recorded for use as the theme of the movie, "The Eyes of Laura Mars." Lawrence was a background singer on Frankie Miller's most recent release, *Double Time.*

Schiff's background isn't as well known, at least from the studio publici-

1994: A heavy, heady dose of hard, raw-edged rock

ty, except that he played guitar by himself a lot, and Bill Rhodes had worked with DeSautels once before in a band called Good Thunder (one record for Elektra). They came together last year under the guidance of Jack Douglas (he produced the Frankie Miller album Lawrence sang on), and now their debut album has been out for several months.

The album deserves more acclaim than it is getting. The four are successfully bridging the technical mastery of Tom Scholz's Boston and the raw energy of punk. But the sweetness of Boston is absent here, and the anarchy, depressing tunelessness and anger of punk is also absent. Instead, the listener is given a heady and heavy dose of hard, raw-edged, substantial rock that is both technically proficient and satisfying.

Schiff and DeSautels carry the instrumental burden on this record, and Schiff's guitar work never falters. He delivers power chords, fuzzed tones, fluid single-note playing and lower-keyed acoustic playing with equal skill. DeSautels provides a variety of drumming and percussive styling, too.

Lawrence's singing style may be the main attraction for most listeners, however. It sounds as if she throws herself bodily into every song, and in the upper registers, she tends to border on a banshee-like wailing (some may say near-hysteria). She doesn't lose track of the melody line, however, and her vocals often are overdubbed with a harmony line. "Sing to Me" offers a good example of how close she comes to losing control of the vocal line, without ever losing it.

The album has nine cuts, none of which could be considered anything as slow or soft as a ballad. "Once Again" has a sinister sound and fine vocal and guitar harmonies, while "Shoot to Kill" is basically a blues number with a steady and pronounced rock tempo. The guitars in this cut do a lot—solo, accompany, counter and provide rhythm— and a sexual innuendo in the lyrics adds to the earthy feel of the song.

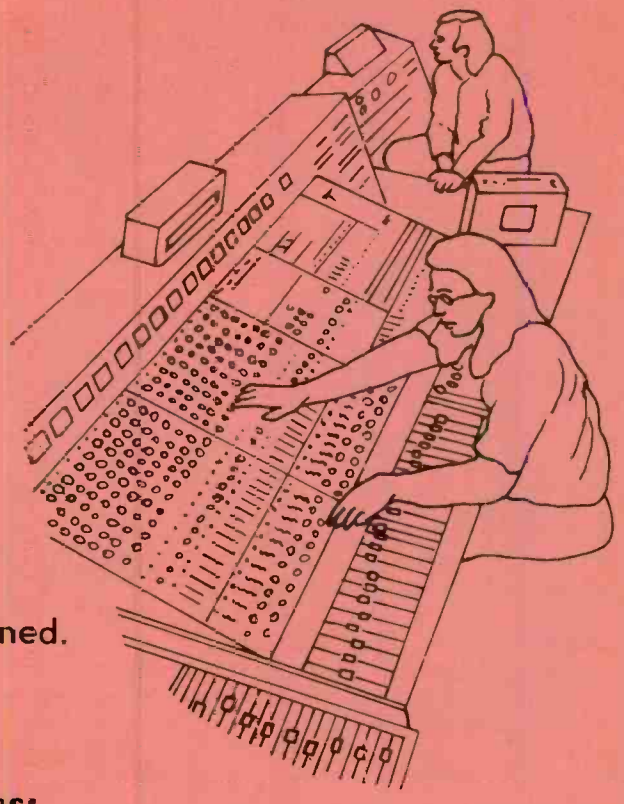
"Sing to Me" has a melody line that could have been written by Renaissance, but the rhythm and arrangement is pure heavy metal, although clearer and less fuzzy than some of the other cuts. "Heleana" has an unusual melody and a chorus that sounds thin and distant.

Schiff uses his acoustic guitar on "Bring it Home" and "Heleana," but only for brief moments on each. The soft vocal opening of the former ends abruptly when the drums signal a change to full-blown electrified (and electrifying) music. "Radio Zone" is a blend of 1950's raw rock and New Wave raw rock, with crashing power chords and a brilliant guitar solo. (This is one number I'd like to see performed "live.")

The opening strains of "Hit the Hard Way" immediately remind the listener of the opening to the Beatles' "Come Together," but the rest of the song has guitars just about everywhere. Schiff uses a voice box in an interesting manner here. "Read Up" and "Anastasia" close the album, the latter a tale of the daughter of the czar with an interesting blend of what sounds like a balalaika and Schiff's guitar. The vocal line on the former has the feel of a 1940's female

RIA, the largest and most respected network of studios offering courses in the art of multi-track recording.

When today's music conscious society made recording the new art of self-expression, the RIA created the nationally acclaimed ten week course in the art of multi-track recording, entitled **Modern Recording Techniques.** All classes are conducted on location at 16 and 24 track recording facilities. Under the guidance of professional recording engineers as instructors, the students see, hear, and apply the techniques of recording utilizing modern state of the art equipment. The course includes live recording sessions so that the student may apply the techniques learned.



For information on RIA's Modern Recording Techniques course, call our local representative in the following cities:

AMES, IOWA
A & R Recording Studio
(515) 232-2931

ALANTA, GA
Apogee Recording Studios
(404) 522-8450

BALTIMORE, MD
Sheffield Rec's Ltd., Inc.
(301) 628-7260

BIRMINGHAM, AL
Solid Rock Sound
(205) 854-4160

BURLINGTON, VT
Starbuck/Ashley Record
(802) 658-4616

CHARLOTTE, NC
Reflection Studio
(704) 377-4596

CHICAGO, Ill
Universal Recording Studios
(312) 642-6465

CLEVELAND, OHIO
Agency Recording
(216) 621-0810

COLUMBUS, OHIO
Rus - I - Col Rec'g
(614) 267-3133

COLUMBUS, GA
HNR Studios
(304) 327-9431

DALLAS, TEXAS
Sound One
(214) 742-2341

DENVER, COLO.
Applewood, Studios
(303) 279-2500

DETROIT, MICH.
Recording Institute
(313) 779-1360

HONOLULU, HAWAII
Audissey Sound
(808) 521-6791

HOUSTON, TEXAS
Wells Sound Studios
(713) 688-8067

KNOXVILLE, TN
Thunderhead Sound
(615) 546-8906

L.A./ORANGE COUNTY, CA
United Audio
(714) 547-5466

NEW HAVEN, CT
Trod Nossel Productions
(203) 269-4465

NEW ORLEANS, LA
Knight Recording
(504) 834-5711

NEW YORK, N.Y.
RIA
(212) 582-0400

NORTHERN N.Y. STATE
Michele Audio
(315) 769-2448

PADUCAH, KY
Audio Creations
(502) 898-6746

PHILADELPHIA, PA
Starr Recording
(215) 925-5265

PHOENIX & TUCSON, ARIZ
Lee Furr Studios
(602) 792-3470

PITTSBURGH, PA
Audio Innovators
(412) 391-4220

RICHMOND, VA
Alpha Audio
(804) 358-2852

SANTEE/SAN DIEGO, CA
Natural Sound
(714) 448-6000

SEATTLE, WASH
Holden, Hamilton
& Roberts Recording
(206) 632-8300

TULSA & OKLA CITY, OKLA
Ford Audio and Acoustics
(405) 525-3343

CANADIAN REPRESENTATIVES

MONTREAL, QUE.
RIA
(212) 582-0400

OTTAWA, ONT.
MARC Production
(613) 741-9851

TORONTO, ONT.
Phase One Recording Studio
(416) 291-9553

ria

CIRCLE 65 ON READER SERVICE CARD

trio or quartet, of all things.

If you had just about given up hope on whether a decent hard rock band would emerge from the expanding pile of punk rockers, then find 1994. Then turn the volume up loud only if you dare, if the kid is awake and/or the apartment building is empty. S.R.

JAZZ

ART HODES: ***Friar's Inn Revisited.*** [Robert Koester, producer; recorded in Chicago between 1968 and 1973 at Stu Black or Dave Antler Sound Studios.] Delmark DS-215.

Performance: **Magnificent traditional jazz ensemble playing**
Recording: **Good natural sound with presence and definition**

ART HODES: Verve and excitement

I wish Bob Koester's Delmark Records would take as much care with liner note information as they do in their selection of musicians and finely balanced, ungimmicked recording. Recording dates are not listed anywhere on the album. The 1968 date comes from the knowledge that Barney Bigard and Art Hodes had made a previous Delmark recording at that time and the notation on the liner that these are, in fact, alternate masters from the session that produced Delmark DS-211. The sides from that session include "Tin Roof Blues" and "Sensation." All the other tracks include legendary Chicago jazz clarinetist Volly De Faut, so we know that these dates had to be made prior to De Faut's passing in May of 1973.

Volly De Faut was no ordinary clarinet player. He was an artist of remarkable taste and virtuosity. He also had the misfortune, as did Morey Bercov, Bud Jacobsen and a few others, to be a contemporary of the great Frank Teschmaker, so very few people noticed him. Tesch's passing came just in time for Benny Goodman to gain the attention that he might have missed had he not escaped the Teschmaker comparison syndrome.

George Brunis, on the other hand, was one of the first of the white jazz players to emigrate to the frozen Northern wasteland of Chicago from New Orleans. Brunis had also died since this recording so this LP is a much needed testament to the greatness of two early jazz giants who are no longer with us. Hodes, thank the Lord, is still alive and well in Chicago and playing piano as great as ever. The rest of the band has jazz credits of long standing all the way from former Earl Hines bassist, Truck Parham to Barrett Deems, who played drums with Joe Venuti, Louis Armstrong and so many other greats. The repertoire comes from The New Orleans Rhythm Kings who played at the Friar's Inn in Chicago. Actually the Friar's Society house band, The New Orleans Rhythm Kings, didn't record all these tunes for Gennett in 1922/23. Even the tunes they did record back then were not exclusively identified with them. "Sensation," for example, was one of the tunes they had in common with Nick Larocca's Original Dixieland Jazz Band as was "Clarinet Marmelade." Tunes like "Jada" and "Angry" were pop tunes of the day. (Brunis, by the way, was co-composer of "Angry" which made him a great deal of money in the '30s when it was revived.) The fact remains that in 1973 (which is the latest this record could have been made) Brunis was in his seventies and Hodes his late sixties and they both played with all the verve and excitement of teenagers. In the presence of such giants as DeFaut, Hodes and Bruins, it is too easy to overlook a player like Nappy Trottier, a man so sorely neglected that I can't find his birth date listed in the history books to figure his age. Nap's too good a player to let that happen. I do wish that Brunis had sung the lyrics that Dud Meacum wrote to the tune in 1925 rather than making up a new set about

Jazzmen Worthy of Wider Acclaim: Chris Woods and Andrew Cyrille

By Nat Hentoff

It has long been a commonplace among most jazz critics and historians that if a musician really has something to say, he'll eventually break through. That's not the way it always works, however, and a clear case in point is Chris Woods (alto and baritone saxophones, and flute). This hard-swinging, incisive, and abundantly imaginative improviser is 54; and yet *Modus Operandi* (Delmark) is his first newly recorded American album (as a leader) in more than twenty years.

Woods, to be sure, has had to take day jobs from time to time but even so, his sideman's credits are impressive: among them, work with Dizzy Gillespie, Clark Terry, Ted Curson, and the Jazz Composers Orchestra of America. I first became aware of Woods several years ago when he was working with Curson, and I found myself shouting in pleasure at the man's fire, crackling beat and leaping imagination. All those qualities are vividly present in this set—along with intriguing, precise, multi-colored arrangements of bold originals that are mostly by Woods.

The other players are even less well known than the leader—trumpeter Greg Bobulinski, pianist Jim McNeeley, bassist Roland Wilson, drummer Curtis Boyd—but they too are authoritative, inventive, and often exhilaratingly full of surprises. The recorded sound is appropriately bright, clear, spacious, sizzling with presence. Once again, Bob Koester of Delmark (4243 N. Lincoln, Chicago, Illinois 60618) has taken a chance on musicians who, despite limited renown, very much merit a forum—for the record.

Andrew Cyrille is better known than Chris Woods—largely as a result of his many years with Cecil Taylor. But this remarkably resourceful percussionist has yet to receive the full recognition due him. Since 1976, he has been leading his own group, *Andrew Cyrille & Maono;* and finally, an album by this powerful unit is available in the United States. *Metamusicians' Stomp* is on the Black Saint label, an adventurous Italian-based company which has created a valuable catalogue specializing in the frontiers of jazz conception.

Making up the rest of the rhythm section with Cyrille is bassist Nick DeGeronimo whose tone is big and resonant and who lays down a resilient enough beat to support legions of swingers. In the front line are trumpeter Ted Daniel, a singular lyrical player; and the forceful, continually absorbing tenor saxophonist, David Ware, whose ideas are intensely contemporary but whose huge sound recalls the fabled approach to the tenor of such titans as Coleman Hawkins who came on like forces of nature.

The quality of the recording is full bodied, crisp, and well-balanced. Black Saint is available in most stores that stock jazz, and is distributed in this country by Rounder Records, 186 Willow Ave., Somerville, Mass., 02144, from whom you might ask for a Black Saint catalogue. As always, in the history of jazz, even when major labels are active in nurturing the music, the smaller ones keep listening ahead and document the future.

CHRIS WOODS: *Modus Operandi.* [Robert Koester, producer; Phil Prevette, engineer.] Delmark DS-437.

ANDREW CYRILLE & MAONO: *Metamusicians' Stomp.* [Giacomo Pellicciotti, producer; Carlo Martenet, engineer.] Black Saint BSR 0025.

the Art Hoe-deez band. His vocalized bass-line accompanying De Faut's clarinet solo that follows is in keeping with the kind of crazy man who used to play trombone with his foot manipulating the slide . . . and once led a chorus of "The Saints" marching in to the ladies' room at Jimmy Ryan's. He was quite a character. He was quite a musician, as Volly De Faut was and Art Hodes still is. J.K.

TOM SCOTT: *Intimate Strangers.* [Tom Scott and Hank Cicalo, producers; engineered by Hank Cicalo with Tony Becker; recorded at Crimson Sound, Santa Monica; Sound Labs, Los Angeles, Ca.; Hit Factory, New York, N.Y.] Columbia JC 35557.

Performance: **Trendy, but energetic**
Recording: **Clean and crisp**

STAN GETZ: *Another World.* [Stan Getz, producer; David Richards, engineer; Martin Pierson, Andre Gouchat and Eugene Chaplin, assistant engineers; recorded at Mountain Recording Studios, Montreux, Switzerland.] Columbia JG 35513.

TOM SCOTT: Super-smooth

Performance: **Fine Getz and great ensemble!**
Recording: **Also clean and crisp**

What may be the most interesting way to compare these two albums is to look at the approach youth and age take toward synthesized reed playing. Tom Scott, 29, has used the Lyricon before, described here as an electronic woodwind sounding something like a synthesizer, organ and clarinet, and he tosses a few passages for the instrument about mostly for effect, without really exploring its possibilities.

Stan Getz (52 this year), on the other hand, gets his hands on a digital delay device with Moog Echoplex for what he says is the first time, and although *Another World* is spread over two discs, he plays with the device only once, very tentatively and not always successfully. It is as if he were experimenting in his room somewhere but forgot to turn off the tape recorder.

Most of Scott's playing is super-smooth and well-organized, while Getz's playing, as skilled as it is, conveys a greater sense of improvisation (sometimes even a roughness) that makes his music more alive. Scott tells a story on the "A" side of *Intimate Strangers*, and in telling the story he resorts to pop rhythms (funk, disco) to propel the narrative. Getz plays on ten separate and distinct tracks, joined together by little more than the fact that the same quintet of players is present on each.

Scott's release is not without merit, however. The six tracks that comprise his song suite "Intimate Strangers" are well-crafted, good-sounding jazz tracks that tell the story of a dream sequence involving a saxophone player and a

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

Because you are... creatively involved in how sound is produced and reproduced, and are serious about creating & recreating music both live & recorded. Modern Recording can help you develope your talent.

Subscribe Now.

906C

MODERN RECORDING
16 Vanderventer Ave., Port Washington, NY 11050
(516)883-6200

I wish to ☐ subscribe ☐ renew today and save up to $14.00

$12.00 for 12 issues Save $6.00

$22.00 for 24 issues Save $14.00

check ☐ money order ☐ enclosed for the amount of $ _____

(Foreign subscribers add $3.00 for each year Foreign subscriptions payable ONLY in U.S. funds by international draft or money order)

My account number is ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐ ☐Bank Americard ☐Mastercharge

NAME _____

ADDRESS _____

CITY _____

STATE _____ ZIP _____

beautiful woman in his audience.

Scott is given splendid backup from the session men who pop up on just about every Columbia jazz release these days—Eric Gale and Hugh McCracken on guitars, Richard Tee and David Paich on keyboards, Gary King on bass, Steve Gadd on drums and Ralph McDonald on percussion. With Scott in the lead, a lineup like that can't go but so wrong.

The second side features four unrelated songs and taken as a group, they are better than those in the suite. "Breezin Easy" is most enjoyable, with good tenor sax work, drums and percussion (and listen for the birds near the close!), while "You're So Good to Me" has a slower, shuffling beat, embellished by a harmonica, but without much melody. Scott's Lyricon works on "Puttin' the Bite on You," while Scott, Paich and Steve Porcaro perform on synthesizers they programmed for "Beautiful Music." The last piece also has a good, quick-tongued flugelhorn solo by Chuck Findley.

Getz has included two real chestnuts on his new album, the third in a series that has become known as "Stan Getz Presents," and his readings of "Willow Weep for Me" and Ellington's "Blue Serge" are both quite good. The latter features a traditional jazz combo setting right out of the 1940s and 1950s, with a soft sax, bass and piano playing in chords. The former opens with a bluesy touch and some very husky sounding sax playing. The track is almost eleven and one-half minutes long, and the middle is given to a long improvisational passage, with returns to melodic phrases from the song.

On these two songs especially, the other four players get to show their stuff, and Andy LaVerne on keyboards, Mike Richmond on bass, Billy Hart on drums and Efrain Toro on percussion produce some exquisite ensemble work. Toro, a young Puerto Rican who has studied at the New England Conservatory of Music, is a percussionist to watch, turning in some outstanding work throughout the album.

Although most of the tracks work, some do not. "Pretty City," which opens the album, is almost abrasively raw, with a very nervous tempo, and is not a very good dedication to the city of Jerusalem (which was the intent of its composer, LaVerne). That cut is followed by "Keep Dreaming," which takes a long time to unwind, and even then does not give the listener much of a melody. The string synthesizer and exotic electronic devices used here almost make the track sound like "Star Wars" gone to the local jazz club. Technically, though, this is one of the album's most exciting cuts, with the sound bouncing quickly from one channel to the other in such a way as to almost force the listener to move his eyes from one side to the other!

Getz's experimentation with the delay device and Echoplex comes on

STAN GETZ: Improvisation and experimentation prove exciting and life-giving

"Another World," the title track, and some of the effects are most intriguing. As he explains in the liner notes: "The saxophone had limited me to playing one note at a time, although I always heard chords and larger contexts in my head as I was playing those notes." With the new devices, he could play chords, and some of the results are genuinely haunting. Other sounds, however, are like so many geese honking, and another even sounds like a trumpet fanfare. And when one least expects it, out come some bars from "Alice Blue Gown."

Scott's best work, in my book, were his arrangements and backup of Joni Michell in 1974's *Court and Spark* and her subsequent *Miles of Aisles*. He never quite matches in *Intimate Strangers* the subtleties and nuances of sophisticated jazz that he had in those charts, although much of the briskness and brightness is here.

Getz, except for a few lapses on *Another World*, has provided the better album for tenor sax fans, combining hot and cool readings, up-tempo and low-keyed rhythms, solo and ensemble playing into a neat package. Getz, too, has provided for his audience some of the best album liner notes for a jazz release in all of 1978. S.R.

CLASSICAL

COPLAND: *Appalachian Spring* (original instrumentation); **IVES: *Three Places in New England.*** The Saint Paul Chamber Orchestra, Dennis Russell Davies, conductor. [Tom Voegeli, producer; Tom Jung, Bob Berglund, Scott Rivard, engineers; recorded using the 3M Digital Audio Mastering System.] Sound 80 Records, no number given.

Performance: **Inspired**
Recording: **Unbelievably clear**

When audiophile recordings began turning up in the stores a couple of years ago, it was evident that sound quality was of primary and performance of secondary interest. When star performers such as the Cleveland Orchestra and its conductor Lorin Maazel went into the direct-to-disc studio, it was with a program of short works chosen to dazzle the ear with audio

clarity; but most of the time the performers were not particularly well-known. Here is an audiophile recording that is a treat on all counts. Rather than several short selections, we are given two integral contemporary American works, both of which call for a variety of timbral resources with which the 3M Digital Audio Mastering System can be tested and displayed. Dennis Russell Davies is a conductor with a proven ability to make this music sound its best, and his Saint Paul Chamber Orchestra plays the music not only with insight and respect but with inspiration and energy. And the quality of the recording is almost too good to be true. Stereo separation is excellent, and instrumental placement as it is reproduced here is exactly what it would be onstage. The sounds—the attacks and various instrumental colors—are lifelike, and the clarity of reproduction at both the high and low end is unprecedented.

Furthermore, the approach Sound 80 has taken—what producer Voegeli calls the "direct-to-disc philosophy" of recording each side in one complete take with no editing—makes for a kind of "honest" recording which makes the performance and sound quality seem all the more unbelievable.

To describe the clarity of each instrumental timbre as it comes at you from, first, the Copland score (a more systematic presentation than the Ives) would take pages. Suffice to say that the strings sound full-bodied; the flute has a lifelike "airiness;" and the keyboard, particularly in the high register and when it is doubling a string or wind line, is reproduced so clearly that you can feel the percussive aspect of the hammer hitting the strings. "Appalachian Spring" is a perfect choice for this sort of treatment, for the score calls for smooth legato lines, sharp staccato attacks, thin almost chamber-music-like textures and glorious full-orchestral pronouncements all in the space of twenty-five minutes.

The Ives may be somewhat less accessible than the Copland; his sense of tonal manipulation still thought by many to be more on the perverse side. The textures he creates with his clashing tonalities (brass bands battling it out with each other) and often nebulous scoring create special problems for the engineers. For example, the beginnings of the first and last movements must sound sort of misty and obscure to create the atmosphere Ives wants. Yet the clarity of the instrumental sounds which make up this texture cannot be sacrificed, and when the scoring gets cluttered later on, clarity of detail must be maintained—that being, after all, the point of this recording effort.

On all counts, the recording is a success. However, to bring a small cloud over an otherwise unqualified rave, the listener ought be aware that for all the perfection of the digital recording process, there is still a major weak link: that is, the record is pressed in conventional vinyl and played with a stylus. Even with the best of pressings—which this, sadly, is not—this can mean trouble. In light of the excellence of the recording and performing though, I don't find the small bit of surface noise audible in the Ives, on my copy, too much to bear. Overall, this is an overwhelmingly successful package. A.K.

RUDOLF SERKIN: ***Rudolf Serkin on Television. Haydn: Sonata, No. 49; Mozart: Rondo, K. 511; Beethoven: Sonata, Op. 81a ("Les Adieux"); Schubert: Sonata in B-flat, Op. Posth. D. 960.*** Rudolf Serkin, piano. [Thomas Frost, producer; engineer not given; recorded in concert at Carnegie Hall, New York, December 14-15, 1977.] Columbia M2 34596.

Performance: **Exquisite**
Recording: **"Live," but clean and mostly quiet**

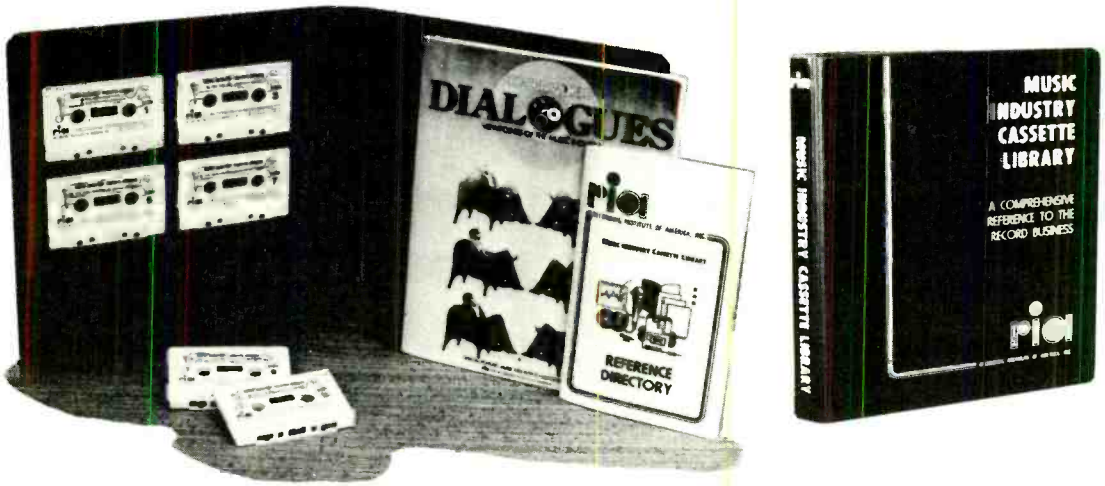
SCHUBERT: ***Sonata in B-flat, Op. Posth. D. 960; "Landler" from D. 790.*** Grant Johannesen, piano. [Recording information not given.] Golden Crest CRS 4178.

Performance: **Sensitive**
Recording: **Disgraceful**

Last year marked the 150th anniversary of Franz Schubert's death, and in commemoration we have heard more Schubert in the last eighteen months than in the preceding ten years. For some reason though (possibly because record companies felt stung by too many leftover Beethoven discs after that composer's 200th birthday in 1970), the Schubert festivities seem to have been more apparent in the concert halls than in record stores. Pianists in particular (the song cycles and symphonies, after all, have always been standard repertoire) have found

MODERN RECORDING's PRO SHOP

Here's a Guide to the Record Business by the Pros!



Compiled from Recording Institute of America's interviews with key executives and "hit-makers", plus Reference Directory and Dialogue's Viewpoints of industry "stars"

Listen to the industry "pros" describe the workings of the Music Business. Hear the most respected attorneys of the entertainment field define and discuss the legal terminology of Recording Contracts, Songwriter Contracts, Professional Management Contracts. Over 3½ hours of professional reference . . . could be the most important 200 minutes of your life!

Plus . . . RIA Reference Directory, including sample songwriter affiliation forms, sample artist contracts, writer contracts, etc., in addition to a Directory of Record Manufacturers, Music Publishers, Personal Managers, Producers and Booking Agents. Also **Record World's "Dialogues"** with over 50 candid interviews from **Record World** magazine, and a cross-section of "star" personality interviews.

You get all the above (regularly $49.95) for only $39.95 for MR readers.

Home Recording Course

—a must for every *creative* tape recorder owner!

HOME RECORDING TECHNIQUES is three hours on two 7-inch, 7½ ips, 4-track tapes, including a booklet of diagrams on:

- sample microphone placement
- instrument set-up
- console and machine patching for special effects
- and MUCH MORE!

ONLY $19.95

ORDER FORM – MAIL TODAY 906C

MODERN RECORDING Magazine
14 Vanderventer Ave., Port Washington, N.Y. 11050

☐ Please send [ ] copies of "Music Industry Cassette Library" at $39.95 each

☐ Please send [ ] copies of "Home Recording Techniques" at $19.95

Please Print

Name _____

Signature _____

Address _____

City _____ State _____ Zip _____

Total Amount _____ Check/Money Order for $ _____

10 Day Money Back Guarantee DATE ___ / ___ / ___

something in Schubert's Sonatas these past few years that they have either missed or ignored before, and the powerful Sonata in B-flat, D. 960 seems to have emerged the favorite of pianists old and young.

Among several recently released tributes to Schubert are these two recordings of the B-flat Sonata, one a "live" recording which takes up the second disc of Rudolf Serkin's two-disc seventy-fifth birthday concert taped at Carnegie Hall, and the other a studio recording by Grant Johannesen who is about twenty-five years Serkin's junior. Both are musicians who have in the course of their careers opted to forgo egocentric flash in favor of entirely artistic sensitivity, and both have served—Johannesen continues to serve—as directors of two of America's major conservatories (Curtis for Serkin, the Cleveland Institute for Johannesen) where they have strongly influenced coming generations of pianists.

While these two interpretations differ, sometimes considerably, both are distinguished readings of a work that has it all—the depth of feeling of the *andante sostenuto*, the breezy playfulness of the two *allegro* movements, and of course the marvelous Schubertian flair for melody and drama that informs the work throughout. For several reasons though, the Serkin recording emerges by far the more satisfying. There are a couple of audible mistakes and a few places where Serkin takes an interpretive liberty that may raise an eyebrow, but the performance on the whole is fluid and well considered. Compared with the Johannesen, Serkin's reading is a bit slower, but this gives the work room to breathe without making it drag and certainly without robbing it of the sense of energy that pervades the last two movements.

One *major* difference between the two readings is that Serkin takes the first movement repeat while Johannesen has chosen to skip it. It is often argued that this repeat is a superfluous one, given the length of the movement, and to some the repeat is even tiresome. Yet it cannot be denied that cutting the length of the movement by about 33% will inevitably alter the intended architecture not only of the movement in question but of the entire work. In the eight minutes he saved by not observing the repeat, Johannesen has programmed one of Schubert's "Ländler," a rarely heard little piece, and charming as it goes its waltzy way.

Then there's the question of recorded piano sound. You would think that Johannesen's studio recording would have the edge over Serkin's Carnegie Hall sound, but quite the opposite is true. Yes, there is an unavoidable "airiness" about the Serkin sound, and there were even a few coughers in the audience those December evenings. But the piano sound is crystal clear, and the subtleties of Serkin's dynamic shadings can be heard and comfortably enjoyed. The Johannesen, by contrast, sounds as if it was recorded in the early 1950s and reissued on a budget label. Were that the case, all might be forgiveable or at least understandable. But it's *not* the case, and there is no excuse for either the extremely loud tape hiss, the shrill distorted high notes which shatter Johannesen's attempts to vary the dynamic, and worst of all a wavering note (third movement, 41st measure during the first repeat) that could have only been the result of someone hitting a tape reel during mastering or recording. Beyond that, the album cover identifies one of the works with a Koechel number (as used in the Mozart catalog) instead of a Deutsch number! I don't know how all this got past the pianist, but given the distinction of his playing he certainly deserves better than this—as do we who are asked to pay $7.98 for it.

To return to the Serkin, while the Schubert is the centerpiece of the set, this recording also includes a Haydn Sonata, Op. 49, the Mozart Rondo, K. 511 and the Beethoven Sonata Op. 81a, "Les Adieux," all performed with the exquisite taste and grace which sets Serkin's playing apart from that of his colleagues of any age. The Beethoven is a particular treat, as his earlier recording (made in 1951) of the work has been unavailable for some time now. The package is a deluxe one, with appreciations of the pianist by producer Thomas Frost and music critic Irving Kolodin, an interview, several pages of photographs and a lengthy discography detailing Serkin's Columbia recording history complete with recording dates, artists and repertoire and consequent LP numbers. For Serkin fans, this is an obvious must.

Grant Johannesen fans, however, must either slice through these awful sonics, or hear him in concert until he signs with a more conscientious label.

A.K.

John Woram's The Recording Studio Handbook

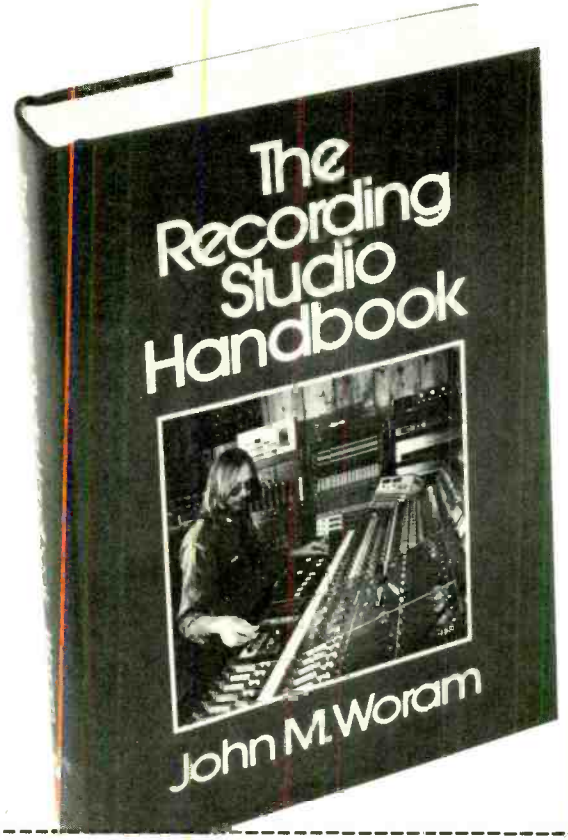
FOR RECORDING ENGINEERS, TECHNICIANS AND AUDIOPHILES

The technique of creative sound recording has never been more complex than it is today. The proliferation of new devices and techniques require the recording engineer to operate on a level of creativity somewhere between a technical superman and a virtuoso knob-twirler. This is a difficult and challenging road. But John Woram's new book will chart the way.

The Recording Studio Handbook is an indispensable guide. It is the audio industry's first complete handbook that deals with every important aspect of recording technology.

Here are the eighteen chapters:

- **The Decibel**
- **Sound**
- **Microphone Design**
- **Microphone Technique**
- **Loudspeakers**
- **Echo and Reverberation**
- **Equalizers**
- **Compressors, Limiters and Expanders**
- **Flanging and Phasing**
- **Tape and Tape Recorder Fundamentals**
- **Magnetic Recording Tape**
- **The Tape Recorder**
- **Tape Recorder Alignment**
- **Noise and Noise Reduction Principles**
- **Studio Noise Reduction Systems**
- **The Modern Recording Studio Console**
- **The Recording Session**
- **The Mixdown Session**



In addition, there is a 36-page glossary, a bibliography and five other valuable appendices.

John Woram is the former Eastern vice president of the Audio Engineering Society, and was a recording engineer at RCA and Chief Engineer at Vanguard Recording Society. He is now president of Woram Audio Associates.

This hard cover text has been selected by several universities for their audio training programs. With 496 pages and hundreds of illustrations, photographs and drawings, it is an absolutely indispensable tool for anyone interested in the current state of the recording art.

Use the coupon at the right to order your copy of *The Recording Studio Handbook*. The price is only $35.00, sent to you with a 15-day money-back guarantee.

MODERN RECORDING MAGAZINE
14 Vanderventer Ave., Port Washington, N.Y. 11050

Yes! Please send ___ copies of **The Recording Studio Handbook** at $35.00 each. On 15-day approval.

Name _____

Address _____

City/State/Zip _____

Total Amount _____

N.Y.S. Residents add appropriate sales tax _____

Enclosed in payment for $ _____

906C **Outside U.S.A. add $2.00 for postage**

Advertiser's Index

CLASSIFIED ADS

FOR SALE: Mellotron, excellent mechanical condition. Violin, Cello & flute, tape bank included, $2,495.00. Tascam mod 10-8 channel board w/pan module, excellent, $1,750.00. Acoustic 88012 channel board, stereo w/sep. monitor chan., EQ on each, new, $1,495.00. Duke Spinner's Music, 375 E. Main St., Rochester, N.Y. 14604. (716) 546-2041.

WANTED: 8 track equipment, all makes. Pay cash. Hall, 10819 Minnette Dr., Cupertino, Ca. 95014. (408) 446-4480.

RECORDING ENGINEER CAREER? Or arranging/writing for TV & films? Send 15¢ stamp to Attainment Research, Box 45333MR, Dallas, Tx. 75245.

SEMI-PRO AND CONSUMER AUDIO EQUIPMENT: TEAC/Tascam, dbx, AKG, Maxell, Delta Lab, Technics, others. Best prices! Prompt delivery! WDI, P.O. Box 340, Cary, NC 27511. (919) 467-8122.

ASi SUPER SPECIAL: Buy an 80-8 with DX-8 and a Model 5 at their regular price and get a 3300SX2T free! You'll save over $1,000. This is for a limited time only, so call now. ASi Pro Audio, (512) 824-8781.

Send for our catalog of studio, musical and stereo equipment: Tascam, EV, Technics, BiAmp, BGW, C-V, dbx, Kelsey, SFW, Tangent, more. Recording Sound Co., 1871 Seminole Trail, Charlottesville, Va. 22901. (804) 973-1110.

TASCAM 80-8 with DX-8 dbx. Four months old. Asking $4,000. Mike Fuller (215) 687-3731.

DOKORDER 8140 four channel deck. Box, manual. $250. (202) 363-6364.

Used Yamaha PM-1000, like new with 1 year warranty. Call or write Professional Sound Labs, Inc. 42 N. Franklin St., Hempstead, N.Y. 11550. (516) 486-5813.

LOWEST PRICES on only the finest American-made equipment. Oberheim, Alembic, Biamp, Furman, Morley; EV and Shure professional mics and more! Rocky Mountain Audio, Box 2577, Silver City, New Mexico 88061. (505) 388-1123.

Scully 1" 8-track, great machine, went 16! Wells Sound, 2036 Pasket, Houston, TX. 77018. (713) 688-8067.

A.B. Systems, Audioarts Engineering, Ashly Audio, Altec, AKG Pro., dbx, HM Electronics, Ivie, JBL Pro., Loft, Lexicon, Northwest Sound, PSL Products, Shure SR, SAE Pro., Tapco C-12 Dealers, Teac Multitrack, Tascam Model 15, Klark Teknik, Technics Pro., Showco and many more. Call or write for a quote. Professional Sound Labs, Inc., 42 North Franklin Street, Hempstead, New York 11550. (516) 486-5813.

STUDIO READY? Tascam 80-8 recorder, Tascam Model 3 mixer, Dx-8 noise reduction, Tapco stereo reverb, plus Tascam cords. Exc. cond. Must sell. $4,300. Retail $5,500. Scott Hybl, P.O. Box 230073, Tigard OR. 97223. (503) 760-4342.

REVOX A-700, 2-track w/remote control. Mint. Call Ken. (612) 925-3595, 941-8171.

Reinforcement, recording equipment—Tapco, EV, Atlas, etc. Write or call for low mail order prices. Sonix Company, P.O. Box 58, Indian Head, MD. 20640. (301) 753-6432.

REEL MEALS—complete entrees of Teac, Tascam, Otari, Studiomaster and other delectable studio packages. Side orders of Ampex and 3M tape. Call or write for our free menu. Man-O-Mann Productions, 1300 N. Triphammer Rd., Lansing, N.Y. 14882. (607) 533-7122 or (607) 277-3971.

If you have a professional multi-track studio, and are interested in becoming a licensed representative for R.I.A.'s Modern Recording Techniques courses, call or write Mr. Chas Kimbrell, R.I.A., 15 Columbus Circle, N.Y., N.Y. 10023. (212) 582-0400. A large profit potential with low operating costs.

Pickups, Parts, Performance

For a complete listing of all hardware and a catalog, send $1.00 to:

DiMarzio®
Musical Instrument Pickups, Inc.

Dept. MR
1388 Richmond Terrace, Staten Island,
NY 10310 USA

CIRCLE 107 ON READER SERVICE CARD

(212) 981-9286

Imagination fuses two powerful illusions.

Experimentation leads to the discovery of new realities.

Introducing the MXR Flanger/Doubler.
For more information see your MXR dealer or write us.

MXR Innovations, Inc., 247 N. Goodman Street,
Rochester, New York 14607, (716) 442-5320

MXR Professional Products Group

CIRCLE 94 ON READER SERVICE CARD