Profile: Elvin Bishop

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

#06691 (F)
$1.50

VOL. 4 NO. 5
FEBRUARY 1979

BOB DYLAN "LIVE"

Deciphering Mixer Specs

Inside "Live From Lincoln Center"

Lab Reports:
B.I.C. T-3 2-Speed Cassette Deck
Crown RTA-2 Analyzer
Heath AA-1600 Power Amplifier

New Products

Record Reviews

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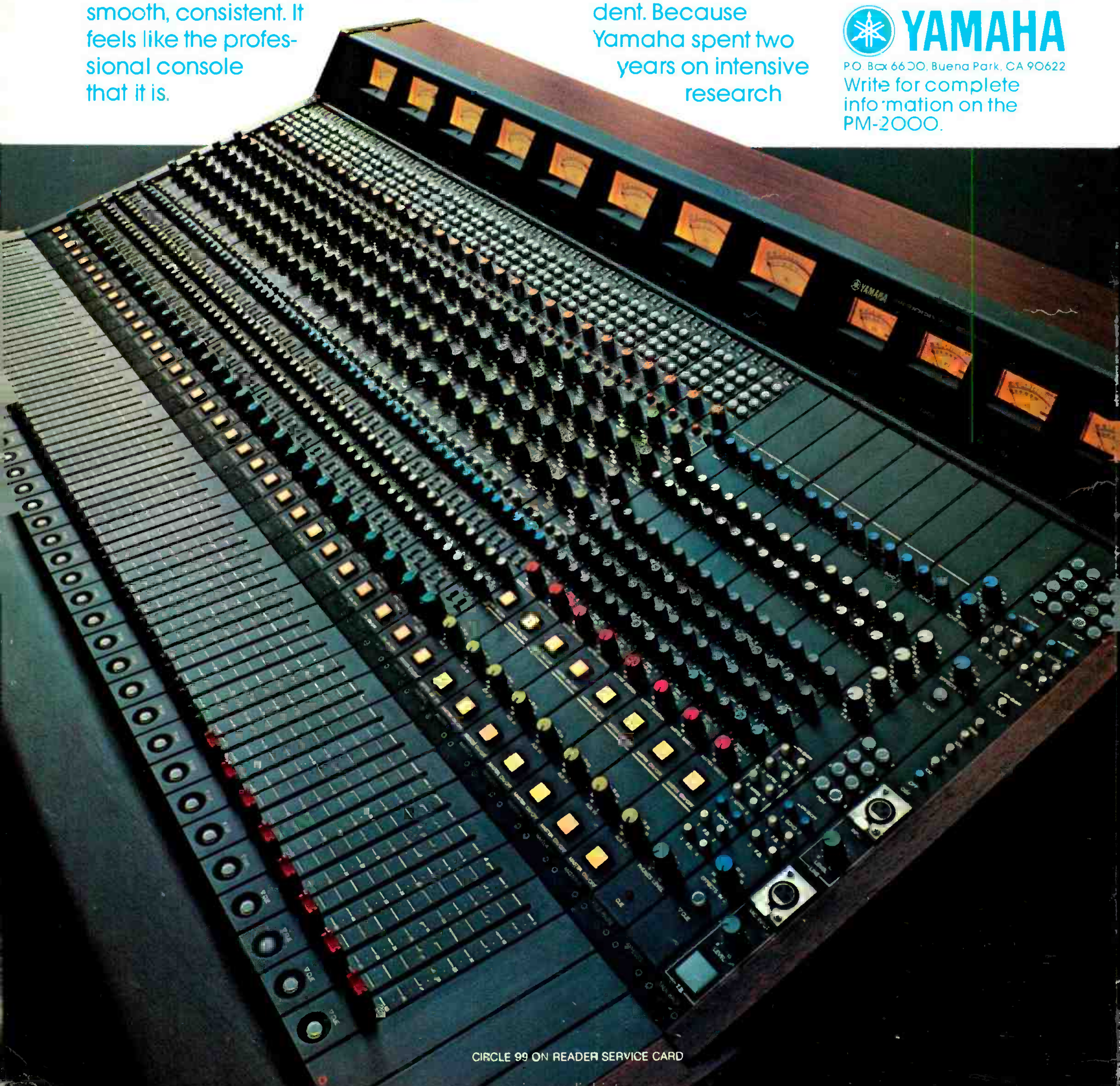
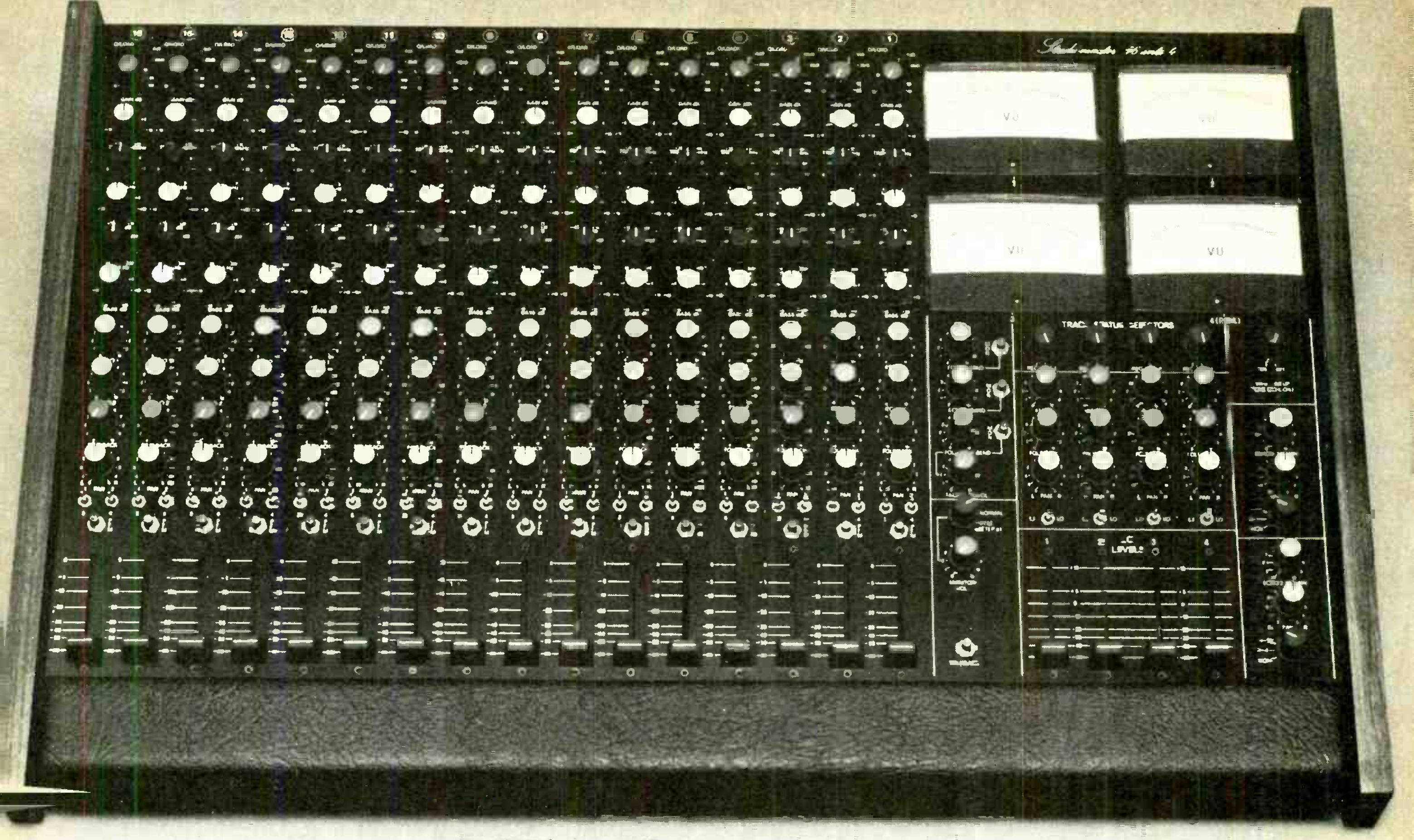
Write for complete information on the PM-2000.



CIRCLE 99 ON READER SERVICE CARD



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Studiomaster has a limited number of openings for qualified pro sound dealers in areas not covered by our dealership list above. For information or recommendations, please contact Studiomaster, 885 S. East Street, Anaheim, California 92805.

No problem.

That's what TASCAM SERIES mixing consoles are all about because whatever stage of recording development you're in, we provide the solution, not create another problem.

Our attention to "human engineering" is what keeps TASCAM SERIES number one and the functions included in our mixers demonstrate just that.

Monitoring. If you already have a mixer, do you have to use the input section for monitoring? You don't if you have a Tascam. *Any* Tascam, not just the Model 15. So when we tell you you've got 8-in on the Model 3 or 5A, you've got 8-in. *Plus* enough monitoring combinations to satisfy anyone in a control room or studio.

Mixing Groups. Problem solving is an orderly pattern of thought. Thinking it through, one element at a time, is the most logical approach. In design you find our mixing groups do just that. Each element on every console that makes the total Tascam mixer has been positioned as a group. This means that each time you operate any Tascam console you are able to logically think about what you are doing, not which knob to twist or button to push.

Meters. Tascam meters are not options. They are an integral part of every mixer we make and they're less visually confusing than a lot of multi-color LED displays we've seen. Compare and see what we mean.

Patch Points. How flexible is your mixer? We have the simplest external patching system in the business. Period. And every Tascam mixer is designed with the same flexibility to provide access to signals and increase your ability to influence and change them to suit your individual ideas.

TASCAM SERIES BY TEAC

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

Channel Assignments (Busses). We believe that channel assignments should be simple and comprehensive. Simple in number and visually comprehensive. All TASCAM SERIES consoles have identical channel assignment color coding, making it easy to see to which output channel any particular input is assigned. Additionally, we have a unique panning system. Whenever any two or more channel assigned buttons are depressed, the pan is automatically engaged. If only one button is pushed, then pan is totally out of the system. Easy, simple, and terribly logical.

A Special Note About The Model 1. Our model 1 is really an inexpensive compact eight by two line-level mixer. It gives you the additional submixes you'll need without costing you a fortune.

Where are our specs? In the equipment, where they belong. Because hearing is believing. So be a skeptic. Pick up the phone or drive to a TASCAM SERIES dealer near you. He has all the information and a personal systems planning brochure from us to you. And remember, whatever your recording needs, the TASCAM SERIES mixers are no problem.

TEAC Corporation of America • 7733 Telegraph Road • Montebello, CA 90640

FEBRUARY 1979
VOL. 4 NO. 5

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

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THE STAPLES

COMING NEXT ISSUE!
A Session with Fleetwood Mac!
Recording Horns & Strings
An Interview with Phil Ramone

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

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Advertising Representative

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Advertising Director

VINCENT P. TESTA
Publisher

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

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Letters to the Editor

Education for Audio

In a letter to *Modern Recording*, Nov. 1978 (pp. 14 & 16), Larry Siedentop comments on the problem of finding recording science programs at colleges and universities. I am a little surprised that you were apparently unaware that the Audio Engineering Society has recognized the problem and is trying to do something about it. In the special Centennial Issue of the *Journal of the Audio Engineering Society*, October/November 1977, issued to commemorate the 100th anniversary of sound recording, Dewitt F. Morris (pp. 864-72) has an article, "The Audio Engineer—circa 1977; What Does He (or She) Do?". In the March 1978 issue, we published (pp. 160 & 162) preliminary results of our Survey of Educational Opportunities for Audio, and appealed for more information. Prospective students may write to the AES at 60 East 42nd Street, New York, N.Y. 10017 for copies of these articles and of the latest edition of our Directory of Education Institutions. This covers all aspects of the industry's needs, including both schools oriented towards engineering technology and those oriented more towards music and recording techniques.

The compiler of any directory is dependent on the information he or she receives, and we do not claim to know about all, or even the majority of educational programs that are offered. Anyone who is associated with or knows of a school that is not listed is invited to write with the address to the AES, so that a copy of the questionnaire may be sent. Alternatively, it may be reproduced from the March issue of the *Journal.*

—(Dr.) Geoffrey L. Wilson
Chairman, AES Education Committee
Audio Engineering Society, Inc.
New York, N.Y.

Our intention was to get feedback and input from readers, which to a heartening extent, we have achieved: Your letter is most welcomed, and we hope our readers look into the AES Directory for what we feel is as complete and current a listing as can be hoped for. It includes codified information signifying types of degrees and major areas of study offered. The Directory, updated periodically, is destined to receive a more simplified format; still, in its present state it will be quite helpful.

Hardhats Out of Work

First, congratulations on a fine magazine. *Modern Recording* really does fill a definite gap in the literature and is written and presented quite well.

Last, having just examined Peter Weiss' article, "Building a Power Supply" in the October issue, I see you have planned several more related construction projects for future publication. Those specifically mentioned were a mic splitter, spring reverb, analog delay and graphic equalizer. Assuming the quality of these devices to be as excellent as the power supply, I am very interested in building each of them. I would like to know if it would be possible to obtain copies of these articles and/or schematics and

Why settle for a copy...

when Tangent gives you the original?

Why settle for an imperfect copy of your sound, when Tangent will give your audience the original?

Tangent's crystal-clear transparency allows your original to flow cleanly to your listeners, with only the coloration that **you** add.

And beyond this foundation of solid quality, Tangent's "a" series mixers give you these features found previously only on recording consoles:

SOLO
Listen to any input by itself, or preview an entire group. Pushing ANY solo button automatically puts that channel into the headphones, no matter what signal was there before. Electronic FET switching makes this possible.

MODULARITY
Tangent consoles are totally modular for servicing ease. Take a spare module on the road for no down time.

100mm SLIDERS
Tangent uses 100mm long-throw faders for extra control and visual feedback. Compare these to the competition's usual 45mm or 60mm length.

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Effects, Reverb, and Monitor sends on each channel act as three independent mixers-within-a-mixer. Compare Tangent's three send busses to other mixers having typically just one or two.

CHANNEL PATCHING
Each input has a pair of access jacks for patching external effects into a single channel or sending a direct feed to a multi-channel tape machine.

LOTS OF EXTRAS
Tangent also has totally balanced inputs and outputs, buss access jacks for slave mixer expandability, and an optional reverb that is one of the smoothest sounding spring units available.

Compare these features to those on consoles costing twice as much and you'll see what a value Tangent is.

As for comparing the quality, well, you just can't get better than the original.

Write or call for your nearest dealer.

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musical engineering
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CIRCLE 75 ON READER SERVICE CARD

parts lists prior to publication, as I am spending a few months soon in a rather remote part of northern India and will have no access to your magazine. Realizing this to be an unusual request, I will appreciate anything you can do to help. Thanks again for a fine mag.

—A.K. Baruah
Sheridan, In.

I am building the power supply written up in your October 1978 issue. I want to put this all into one console. You probably don't have all the articles fully written, but if possibly you could send me the parts lists, schematics, simple parts layout...

—Jim King
Pittsburgh, Pa.

Author Weiss is turning out those construction articles as fast as he can, but we're afraid all of you who are sitting on the edges of your seats will have to bear with this situation with a degree of asceticism. (Ya gotta wait.) For A.K. Baruah we can only suggest that a friend save the magazines until his/her return from remote northern India. In the meantime, meditate!

Ursa Major Price Not Astronomical

Modern Recording *of November 1978 included, in "Product Scene," (p. 39), a description of a versatile digital reverb system, the SST-282 "Space Station," from Ursa Major of Belmont, Ma. We've been advised that the $3000 quoted price was incorrect, by about one thousand: the correct price is $1995.*

School Reports

In your November '78 issue, I read Larry Siedentop's letter about his search for schools with audio curricula. I can sympathize with him, because I went through the same search two years ago. Considering the information available, I was very lucky to have made an intelligent decision.

Here at Eastern Washington University, we have a complete and fuctionally designed 16-track recording studio. We also have a video production facility.

All of our hardware is located in the Department of Radio/Television. The building is five years old and was designed as both an audio and video center. It has plenty of floor space with very high ceilings. We feel the design is a perfect compromise.

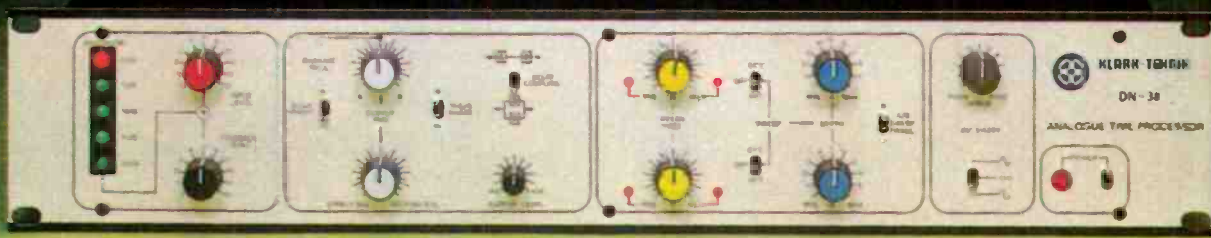
There are programs offering a Baccalaureate in Multi-Track Audio Recording, Television Production, Radio/Television Management and Radio/Television News.

—Harvey R. Gilkerson
Recording Studio Supervisor
Eastern Washington University
Cheney, Wa.

Following, you will find updated information on the Recording Industry Management program here at MTSU (Middle Tennessee State University). This information is in response to Larry Siedentop's letter which appeared in the November 1978 issue of *Modern Recording.* Let me add that Mr. Siedentop is right about *Modern Recording* getting a cover-to-cover reading (in our program, at least).

The purpose of the MTSU Recording Industry Management program is to prepare students for entry and middle level positions in virtually any phase of the recording industry (from the marketing of recordings to audio engineering). The program is designed

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.** 155 Michael Drive, Syosset, New York 11791 (516) 364-1900; West Coast Office (213) 846-0500; Canada (416) 677-0545 in cooperation with and consultation and advice from the National Academy of Recording Arts and Sciences Institute, the National Association of Record Merchandisers and numerous industry executives. Though closely tied with many of the record industry firms located in Nashville (just 30 miles away), the program now has internships and placement on a national level.

The 132-semester hour program leads to a Bachelor of Science Degree in Recording Industry Management. It is housed in the Dept. of Mass Communications.

The degree requirements include 39 hours of University-required core courses, 27 hours of Recording Industry Management courses, 18 hours of Mass Communications courses, and 21 hours of Business Administration courses. Students may select from elective options totalling 24 hours to further concentrate their study in audio engineering, music or business.

Individual courses in the program include those on copyright law; promotion of recordings; merchandising of recordings; recording technology; advanced recording technology; studio production; legal problems of the industry; etc.

—Christian L. Haseleu
Dept. of Mass Communications
Middle Tennessee State Univ.
Murfreesboro, Tenn.

Splitter Separation

Can you tell me the amount of separation that the splitter network described in Talkback of your November 1977 issue (p. 14) has to offer?

—Tom Young
S. Salem, N.Y.

A resistive network such as that described in that Talkback piece cannot be counted on for more than several dB separation (isolation between inputs). For better isolation, a transformer or active splitter is required (see article, "Building a Mic Splitter," Modern Recording, December 1978, p. 58).

For the Class Roster

We've acquired more information on schooling that many of our readers will be interested in: The State University of New York (SUNY) College at Fredonia offers a Bachelor of Science Degree in Sound Recording Technology. This program is designed "to involve the student in an interdisciplinary examination of recording studio technology, emphasiz-

ing acoustics, electronics and music." Its basis is the concept that a professional sound recording engineer should be first and foremost a musician. Admission to the program requires an audition on some professional medium. For additional information, contact Prof. Carter Thomas, Director of Sound Recording Technology, Mason Hall, SUNY Fredonia, New York 14063 (phone) 716-673-3249.

Also: Institute of Audio Research (64 University Place, New York, N.Y. 10003; 212-677-7580) School of Multitrack Recording Technology offers a one-year program with the objective of explaining audio recording technology through emphasis of scientific principles. With New York University's Department of Music, the Institute also offers a four-year Bachelor of Science program, the Music Technology Program.

Professional Goings-on

My associates and I are "very into recording." We try to stay on top of the goings-on in today's recording and music industries, so we read a lot of magazines about the industry.

By *far, Modern Recording* is the best—and most professional—magazine for this industry on the market.

I had to send you this letter giving you congratulations on a job very, very well done.

—Ron Baird
LaRon Promotions
Washington, Pa.

Theoretically Wise

I hope to attend a reputable college for recording arts in San Francisco next year. Having already started studying in preparation for the Recording Engineering course, only one aspect leaves me in the dark; namely, Music Theory. Is it necessary for the recording engineer to be as familiar with music theory as the musician is?

Up until (and including) now, I've found *MR* to be the most informative recording magazine available on the stands.

—R. Michael Randt
Prince George, British Columbia

Well, no. It's not necessary, *but it might come in handy. Then again, not all musicians are as knowledgeable in music theory as they could or sometimes ought to be. Your intended college would give you more information on what it expects or recommends regarding basic music theory.*

Apologia by Explanation

Regarding "Reverberation," the opening letter to the editor in your November 1978 issue, first let me say that I apologize for neglecting any credit due Sami Uckan of Atlantic Studios, New York for the letter from me published in the August '78 issue. This was due to my being a little overanxious in writing a "comment." At least, that's what I had originally intended to write about, but just continued writing.

The heart of my letter in August was concerned with the fact that no mention was made of the ground-lift switch or its function in the article, "Building a Direct Box," April, 1978.

I am a soundman, not a writer per se; I apologize to *Modern Recording* for any inconvenience. Maybe I didn't quite understand what can be contributed and what cannot.

I've been actively involved in

While the others were catching up, TDK was moving ahead.

Shortly after it was introduced in 1975, TDK SA, the world's first non-chrome high bias cassette, was accepted by most quality deck manufacturers as their high bias reference standard. This advanced, new cassette enabled their decks to perform to the limit of their capabilities. And because the decks are set in the factory to sound their best with SA, music-loving consumers made SA the number one selling high bias cassette.

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But making the world's most advanced cassette was nothing new for TDK's engineers. They pioneered the high fidelity cassette back in 1968 and for more than a decade they've led the way in cassette tape technology. Over the last three years, they've refined SA and made it clearly superior to the '75 version.*

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So if you'd like to raise your own recording standards, step up to TDK SA, the high bias reference tape backed by high fidelity's original full lifetime warranty.**

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*Today's SA has a maximum output level (MOL) more than 3dB better than that of 1975 SA at the critical high frequencies, and improved sensitivity across the entire frequency range. **In the unlikely event that any TDK audio cassette ever fails to perform due to a defect in materials or workmanship, return it to your local dealer or to TDK for a free replacement. ©1978 TDK Electronics Corp.

CIRCLE 87 ON READER SERVICE CARD

"sound" for the past seven years, with a musical background that goes back 14 years. I've not gone to any school for training, since in the beginning, they didn't exist or apply to live sound, which is another field altogether.

I taught myself (and continue to) by reading books and magazines—such as this one— as well as learning "on the job." So sometimes, if I'm discussing a subject or answering a question, I may quote from a book or an article simply because it's easier to answer in the terms and phrases I am familiar with and learned from.

Let me end by giving credit to Peavey Electronics and the *Audio Cyclopedia*, for they were other sources for my initial comment.

—Robert L. DeMoss
Dark Star Sound
Framingham, Ma.

Okay, okay. You're forgiven. Thanks for writing again.

Drop a Line to Drum Drops

I have lost the address to send for the "Drum Drops" record advertised in your magazine last summer. Is it still available? If so, please inform me of where to get one.

Your magazine is the best, and greatly appreciated by those who work with sound equipment on a professional basis. Thanks for your help.

—Keith Franzen
Shenandoah, Ohio

Drum Drops™, Vol. 1, a stereo album or cassette of drum tracks, was advertised in our July 1978 issue as "suitable and desirable for musicians for composing, practicing and making demo tapes," can be had by mail for $9.95 (plus 75¢ postage and handling) payable to Drum Drops Vol. 1, at Dept. MR, P.O. Box 3000, Woodland Hills, Ca. 91365. We suggest writing first (before ordering) to find out if the album is still readily available.

Super Tape

I am a fairly new subscriber and I look forward to each month's issue. I'm really glad you folks have two different sections for letters in which your readers can get answers or give input.

I'm just becoming acquainted with the technical aspects of multitrack recording and was reading about 3M's new "super tape" in the July 10, 1978 issue of *Business Week* magazine. Could you tell me a bit more about this new kind of tape?

Also, in the same article, they mentioned two different processes of recording; analog and digital. Analog, I know, is the system used now, but what is digital recording, and just exactly how does it differ from analog?

I'm trying to absorb all I can about the music industry from top to bottom. That's one reason I've subscribed to *Modern Recording.*

—Richard Kocinski
Winona, Minn.

Try as we might, we were unable to find anything on "super tape," 3M, analog or digital, or recording, for that matter, in the Business Week *issue you refer to. But we didn't read word for word, so it might be an oversight.*

3M had introduced, over the summer of '78, a new type of tape, "Metafine," a recording tape which makes use of pure metal particles (as opposed to metal oxide compounds) as the magnetic medium. It could very well be the "super tape" you read about in Business Week.

Modern Recording *ran an "Ambient Sound" by Len Feldman back in September 1978 on the new tape. Also in that monthly feature, in July 1978, Len discussed analog versus digital recording. Unfortunately, we can't supply you with a July issue, but hopefully you were a subscriber at that time. Many moons ago (just about two years) Bob Angus wrote a piece for us: "The Future of Recording—Is it Digital?" It is a very informative article and, at last look, we still had a limited number of copies of Feb/Mar 1977, the issue it appeared in. You can obtain it by sending $2.00, plus 50¢ for postage and handling to Back Issue Dept., Modern Recording, 14 Vanderventer Ave., Port Washington, N.Y. 11050.*

Mobile is not Remote

I have subscribed to your magazine for a year now, and *MR* has been a great help to me in developing a career and clearing up the sometimes tangled world of recording and sound reinforcement. Keep it happening.

But there is one request I would like to make of your publication: Please do a feature on the Mobile Recording Unit.

—Julius Coramiglia
Perry, N.Y.

A valid suggestion. We've already started looking into mobile units; check

Tandberg Presents the Next Generation

out the "Meat Loaf 'Live' and Recorded" cover story of our January 1979 issue, in which the use of the state-of-the-art remote truck of New York's Record Plant plays a large part.

Voracious Readers Voice Void

Help!! I am looking for some books on recording, and I'm not getting too far. I own Martin Clifford's *Microphones; How they work and how to use them*, and Audio Technica's "A Brief Guide to Microphones." Do you know of any places that I could get or send for some more? Book stores in town, well, they just don't have anything on recording. Without your magazine, God knows where I would be. If you can help me, many, many thanks!

By the way, the miking articles— in August and November 1978—were great; keep up the good work.

—David Riddle
Towson, Md.

How can I get the book *Modern Recording Techniques*, by Robert Runstein, as well as a list of and prices of other books about multitrack recording?

—Andre Boulet
Montreal, Quebec

Two places to start—by mail— would be Sagamore Publishing Co., 1120 Old Country Rd., Plainview, N.Y. 11803, and Howard W. Sams & Co., Inc., 4300 West 62nd St., Indianapolis, In. 46206. Sagamore's books include John Woram's Recording Studio Handbook, *and substantial others. Howard W. Sams publishes a wealth of volumes geared to the electronic hobbyist and technician, including the Runstein* Modern Recording Techniques *book, the* Audio Cyclopedia *by Howard Tremaine, and a number of other audio books you would do well to look into. Write the two companies directly for further information.*

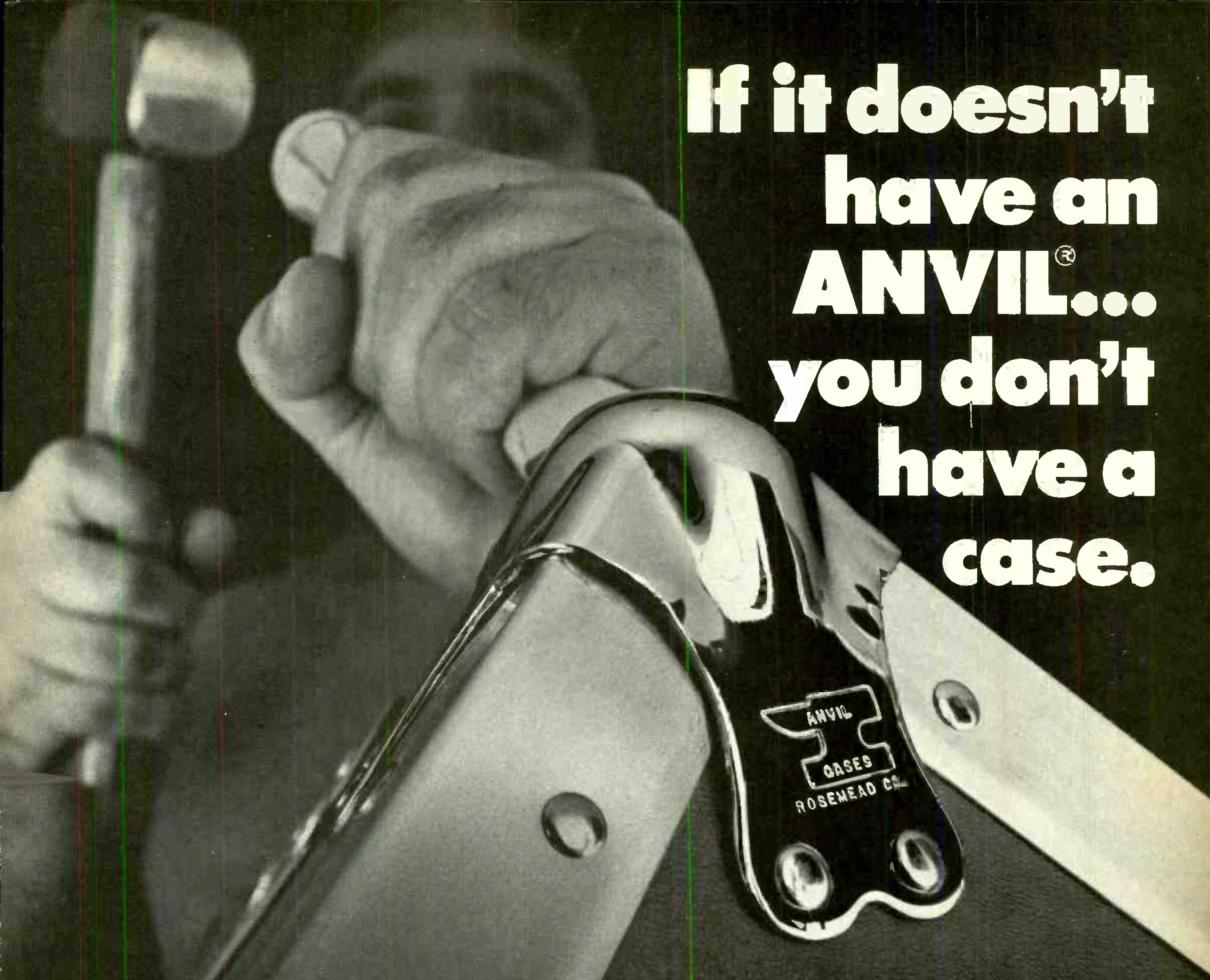
Also, a great place to get a "hands on" idea of what books exist and how to purchase them is The Library. Try even libraries you personally *might not be able to borrow from but can browse in (with no thoughts of stuffing books in a satchel surreptitiously) and collect titles, authors, and publishers. Check out some Journals that are published, and always pay attention to bibliographies: Very often, the author of a work that doesn't quite meet your needs has used as references works that will meet your needs. Then either write the publishers or ask a full-service bookstore (they'll probably want a deposit) to order them for you. To speed things along, first check* Books in Print *(also,* Paperback Books in Print), *bookstore and library bibles published by R.R. Bowker, to see if the book(s) you are interested in owning are actually available. By the way, The Library is free.*

Direct Box Spectre

With regard to the article, "Building a Direct Box," by Peter Weiss *(Modern Recording*, April 1978): No manufacturer is specified for the two transformers named in the parts list accompanying the article. I've tried a few major electronics stores and can't find the types named.

I'm also interested in fitting a TEAC Tascam Model 5A with the direct box. Could you print instructions?

—Stephen Robertson
Baldwin Park, Ca.



A lot of musical instrument and amp cases are going around these days trying to pass themselves off as ANVIL® cases. And a few dealers are adding to the confusion by using the word "anvil" to describe cases that may look like ours on the outside but don't even begin to measure up on the inside.

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I have been trying to build a direct box ever since I read the article in your April issue. However, I've encountered a problem: parts availability.

Line matching transformers are not as readily available as it would seem. I have tried to locate one with a 100,000 ohm rating, as specified in the article, and I haven't been able to find a retail outlet that can even special order one. Where can I find one?

Also, with reference to the parts list, what is meant by an "XLP-3" connector? Is this the same as an XLR connector, the three-pronged type?

Your help in this matter is greatly appreciated. Send any information, or print a reply in the "Letters to the Editor" column and I'll read about it. Terrific magazine! At your mercy,

—Steve Ballard
Glen Ellyn, Ill.

Could you tell me where I can get the transformers mentioned in the direct box construction article? I have been all over town, and no supplier knows of or has heard of them.

Also, why was no provision made for a ground lift switch?

—Ken Gray
Victoria, British Columbia

The transformers were manufacturer-specific. UTC Model A27 is manufactured by the United Transformer Company, with headquarters at 150 G Varick St., New York, N.Y. 10013 (phone) 212-255-3500 and a good number of regional offices (UTC is a division of TRW). The ADC-11-4F had been manufactured by the Audio Device Corp., which, as we discovered, is now out of business. Never worry, though, because the ADC transformer's function can be duplicated by either a Stancor #A-4407, a Stancor #A-4350, a Thordarson #20A07, or any other multiple primary/multiple secondary audio line-to-line transformer. Stancor Products is at 3501 Addison St., Chicago, Ill. 60618 (phone) 312-463-7400. Thordarson-Meissner Inc. is in Mt. Carmel, Ill. 62863 (phone) 618-262-5121. As far as fitting to a particular unit is concerned, the direct box is compatible with whatever (within reason) is at the other end, according to author Weiss.

The XLP-3 connector does have the three-pin configuration Mr. Ballard is familiar with, but differs in that it is a larger, professional type of connector.

Now about that ground lift switch. There is one, although its use is not detailed in the text of the article. Check the schematics (and parts list, too) for the symbol "S_1," which represents an SPST bat-handle switch with 7/16" bushing.

You'll find radio and TV parts distributors unlikely sources for the transformers. Try more industrial, rather than "consumer" electronics parts distributors. It's usually true that these don't cater to what is called the "hobbyist" trade, so direct contact with the manufacturers may become necessary or preferable.

A further note of import: Purchasers of MR's Buyers Guide *(Winter '79) will find a reprint of "Building a Direct Box" therein. That old April parts list (listing the unavailable ADC transformer) slipped in though, during the typical* Buyer's Guide *mayhem, and though we've braced ourselves for a postal deluge regarding it, we'd like to stem that as much as possible. Thus, we suggest that letter-readers to whom "Building a Direct Box" is an unfamiliar and stimulating idea and who are now sending for or searching the newsstands for* Modern Recording's Buyer's Guide *save this column for future reference.*

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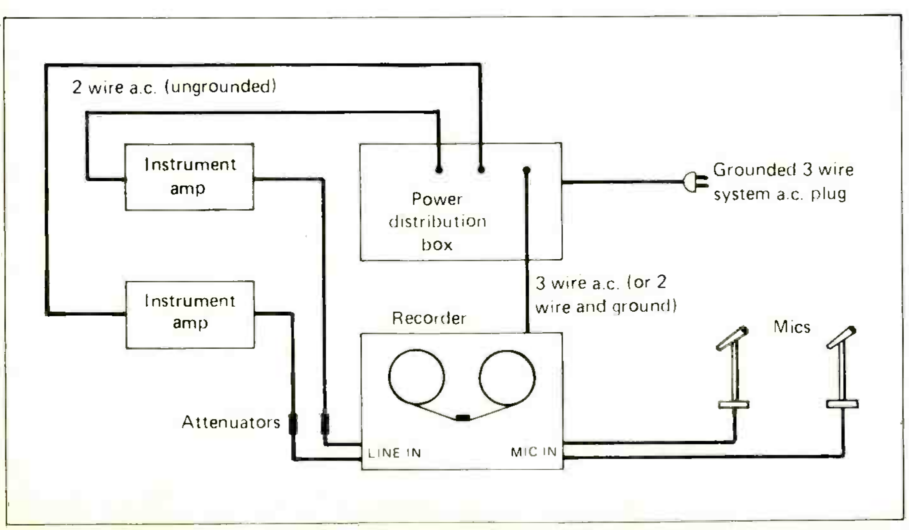
TALK BACK

"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.

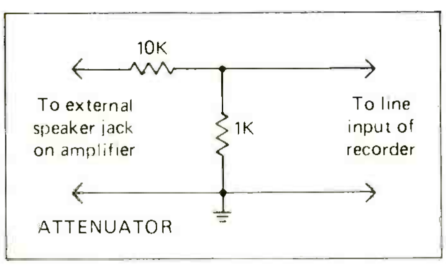
We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel that we are skirting any issues, fire a letter off to the editor right away. "Talkback" is the Modern Recording *reader's technical forum.*

No Faults To Be Found

I am planning to do some low-budget recording using direct feeds from instrument amplifiers as sources into the line inputs of a four-track home-type recorder. After reading the articles in the April and May 1978 issues on direct boxes and grounding problems, I don't understand why the following set-up won't work:



Running the entire system off a single power distribution box would insure that a single circuit was in use. The instrument amplifiers are grounded through the direct lines at the tape recorder. The attenuators should provide about the right signal levels at the recorder line inputs. Although the lines between the instrument amplifiers and the tape recorder aren't balanced, they're certainly low impedance (about 10 ohms maximum) so I wouldn't expect hum pickup to be a problem as long as the attenuators are at the recorder end of the line. Any instrument amplifiers used that aren't run direct would be grounded at the recorder. Since a resistive attenuator has a higher fidelity and a lower cost than a transformer, why use the latter? It seems that the common-mode noise rejection (if that's the right term) of the transformer and balanced line set-up is no big advantage with a relatively short line having such a low impedance. Also, the inability to isolate the grounds of the sound system and recording system using resistive attenuators seems no big disadvantage with a simple set-up such as the one I have described. Should the system work okay or am I missing something?



—Charles Cummings
Bozeman, Mont.

According to contributing editor Peter Weiss, author of one the articles that prompted your letter, "Building A Direct Box" (page 48, April 1978) you're not missing anything! Immediately upon receipt of your letter, we forwarded it to Peter, feeling that he would be the best person to play devil's advocate and give you some insights into the possible pros and cons of your system. To your credit, Peter feels the system you described will work fine, as long as lead lengths are kept under six feet in length. Peter cautions that for longer runs, you should be sure to use extra-low capacitance cable (10-20 pf per foot). Peter adds that transformer-type direct box installations and balanced line feeds are used primarily for and needed most specifically in professional recording situations. Apart from that, you have kudos from Peter and assurance that your method should work just fine.

How To Halt The Hiss

A few months ago I purchased a dbx 155 and I am having trouble with it. Whenever I reprocess the decoded mix through the 155, and/or when I record a bass instrument, I get more hiss than if I recorded without dbx. The hiss occurs only between the attacks of each note and stops at the

moment of each attack. This happens on all four channels. Another problem I have with it is program modulation after a few mixes. Am I doing something wrong or did I purchase a faulty piece of equipment?

—Taylor Sappe
Hazelton, Pa.

Any encode/decode tape noise reduction system is only as good as the record/play accuracy of the tape recorder. Deviations throughout the record/play cycle will result in decode errors. We suggest that bias, tape recorder head alignment, head wear, and frequency response of the tape recorder be checked. We are assuming each channel of the tape is being individually decoded before the channels are combined.

The record levels used weren't mentioned. If the nominal recording level was low, or improper miking techniques were used, the basic signal-to-noise ratios of the tape recorder and the mixer could be sufficiently degraded, causing excessive modulation noise. (Modulation noise is the noise generated by gaps between the magnetic particles of the recording tape.)

The type of tape used, while not mentioned, is important. The new calendered tapes, having extremely smooth surfaces, result in less modulation noise. Without noise reduction, program materials containing passages suffering from modulation noise are almost always lost in the hiss associated with the recording process.

The monitor system, not described, if overly sensitive at high frequencies, will exaggerate modulation noise beyond what would be reproduced on an average record.

From the information provided, we believe the dbx 155 is functioning correctly. The solution probably lies in one of the areas mentioned above.

—Harold Cohen
Manager, Customer Services
dbx, Inc.
Newton, Ma.

As Unique As Larry Feeney

Please help me. I'm looking for a previous article or any information on what the "typical" equalization settings are for some typical instruments used in today's rock recordings. At what frequency and how many decibels should I peak or dip the snare, kick, toms, hi-hats, ride cymbals, lead guitar, rhythm

guitar, bass, piano, organ, trumpet, trombone and violin?

I know this varies widely between studios, but I am a small studio and relatively new to precise mixdowns and need a good starting point. I have Tremaine's *Audio Cyclopedia,* Everest's *Handbook of Multichannel Recording,* Runstein's *Modern Recording Techniques* and various other technical publications, but they do not offer information of this sort specifically.

—Larry Feeney
Wilmington, Ma.

The reason that publications do not offer this information specifically is because this sort of information is not in existence. In order for anyone to answer your question, you'd have to give a bit more information to the guy you're asking the question of.

Take the guitar for example. You'd have to know what kind of guitar the guitarist is playing, what kind of strings it is strung with, how he is playing the guitar, what kind of music he's playing, how fast or how slow he is playing it, how he is picking it (fingers, finger picks, hard, medium or soft plastic picks), whether he is playing nearer the bridge or nearer the fingerboard, what register he is playing in, what studio he is in, where in the studio he is, what other instruments are in the room, where they are, what kind of microphone you are using, where you have placed it, at what angle, whether the diaphragm has been exposed to twenty-five years of polluted air or just taken new from a shipping carton and what mood the guitarist is in. Then, when you get out of the studio and into the control room, you have to know what kind of console you're using, what kind of equalizers, how sharp or how soft the Q (how narrow or wide the bandwidth) of the equalizer is, what other peripheral equipment or other signal processing equipment is going to be used, what abnormalities, if any, exist in the signal chain, then to top it all off, you have to find out that the producer wants the guitar to sound like. When you can give all that information about each of those specific instruments, then *possibly,* just *possibly,* someone *may* be able to suggest how many decibels he might prefer to peak or dip.

In essence, don't waste your time looking for previous or future articles with information about "typical" EQ settings. There are none. No more than there is a typical Larry Feeney—each one is unique. Obviously, no recording engineer goes running around with a checklist covering all of the questions above. But all of these questions must be answered.

The process is quite rapid in an experienced sound mixer's mind. Usually one quick listen and the old gray matter computer gives him the answer. But it really is on the level of a computer-programmed question, mixed with a wealth of human experience.

Of course, this answer won't make you happy, so let's try to give you some encouragement to help you cope with the enormity of the problem. When you're in your control room, listen to each instrument individually and make each one sound good to *you.* Not by counting notches on the equalizer, but by turning the controls until you get something you like, then remember what you did. Close your eyes if you like. Do that for each instrument then compare what you did with similar instruments on records from your own collection. Then go back and analyze what was wrong or different, if

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CIRCLE 86 ON READER SERVICE CARD

anything. But you won't find this easy, because if you listen to each of your records, you will find that each instrument on each different artist sounds different. There may be some sounds that appeal to you and some that don't. The ones you like, I suspect, are the "typical" sounds you refer to in your question. Others may disagree with you as to what sounds "good." There is no right or wrong, or any clear-cut good or bad. No one can tell you what you want to know but Larry Feeney and it will take him a good couple of years to get around to it. It took David Moyssiadis quite a few years to tell me what the typical EQ settings are for each session I do, and I'm still not sure he told me everything. If there was an answer to your question, equalizers wouldn't be calibrated in frequencies and decibels, they would be marked trumpet, guitar, hi-hat, etc. Besides, life would be too easy if we could go out and buy an equalizer like that. And if we could, the button that pushes the engineer (out of a job) wouldn't be far behind.

It's a lot of hard work, but I hope this gives you some food for thought and a direction to go in.

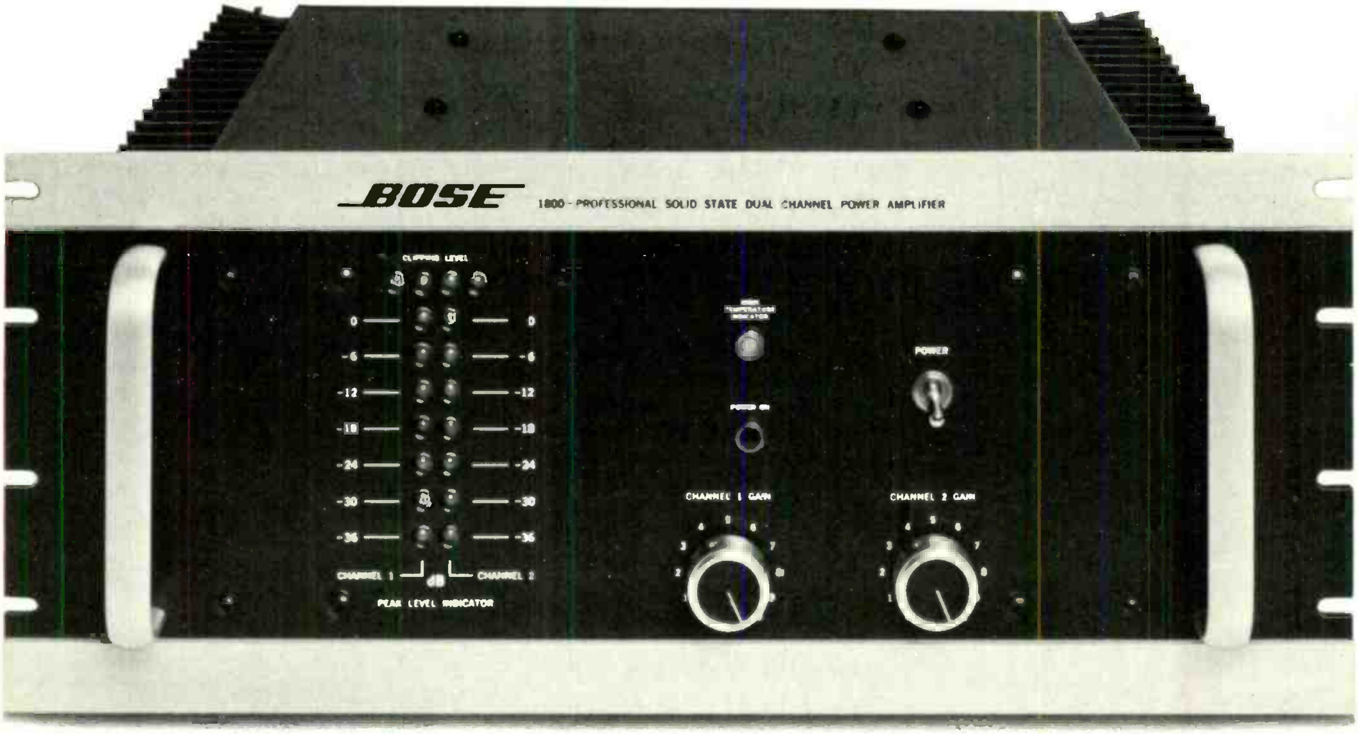
—David Moyssiadis
Contributing Editor
Modern Recording

[We do most certainly agree with David's statement that no hard and fast rules have been written down governing the number of dBs you should peak or dip an instrument, but we feel he might have inadvertently underrated a good reference piece that he contributed to the 1977 Modern Recording's Buyer's Guide *("What Freqs to Tweak or: Finding Frustrating Frequencies For Freq Freaks," pages 54-59). While it will not solve all your problems, a reading will certainly give a bit of added insight into your dilemma—and maybe that little bit of encouragement that David so wisely noted we all need from time to time. Ed.]*

Curing College Sound Shortcomings

Before getting on to my question, I would like to praise *Modern Recording* for its fine articles and the invaluable information it shares with pros and semi-pros alike.

I am presently a sound engineer for a college in the Seattle, Washington area. I have been doing sound for three



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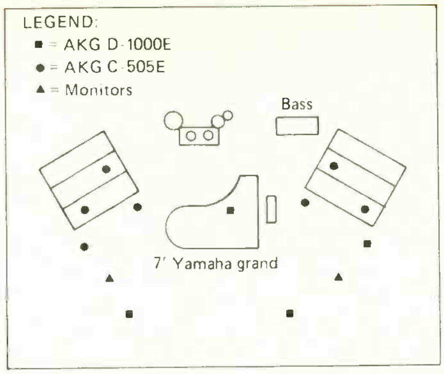
years and consider myself a fair engineer, although I realize I still have a lot to learn.

In doing sound reinforcement for this college, I encounter many different situations. The main work I do, though, is with a vocal jazz ensemble consisting of sixteen voices, piano, drums, and bass, with other instruments included on occasion. The voices are in four parts with four people on each part.

We have an excellent system consisting of the following pieces of equipment: a Uni-Sync Trouper I 18-input board, a Tapco 220B graphic equalizer, a Furman sound crossover network, Spectro-Acoustics power amps, Community Light and Sound PBL speakers with Electro-Voice components and specially designed monitors. We use eight AKG C-505E mics for the singers, and six AKG D-1000E mics—two for solo work, one for piano, with the others used only when needed (see diagram). The singers are on split risers (two on each mic) with the rhythm section in the middle. This set-up has worked extremely well for us and we intend to stay with it.



We have gotten excellent sound through this set-up with the exception of two things. We get an extremely high amount of leakage through the piano mic from drums and singers. Feedback from monitors is present in almost all situations. Also, with the piano there seems to be an unusually high amount of ringing. What can be done to minimize these shortcomings? Any help will be greatly appreciated.

—Kevin Knight
Edmonds College Music Dept.
Seattle, Wa.

The use of a piano pickup (Helpinstill or similar) will immediately eliminate bleed through from the vocal sections. However, although pickup from the drum kit will be reduced, complete removal of this bleed through is another matter, which unless relocation of the piano is possible (180 degrees), can only be solved by a series of compromises.

Assuming that the piano is played with the lid open, the drums are not only being picked up by the piano mic, which is causing your present problem, but also by the soundboard. The reason for this is that the open lid forms what is in effect a parabolic "trap" for pressure waves, which will be most noticeable from the kick. Because piano pickup relies on the resonance of this soundboard, any external interference of this nature will be sensed. Preventing these pressure waves from reaching the piano can be achieved in a number of ways.

When a "live" (miked up) drum kit is used, it is not uncommon to use a sheet of clear plastic in front of the drums at an angle of about 45 degrees. This would obviously deflect any waves away from anything directly in front of the kit. Unfortunately, since no drum mics are shown in the stage plot, you are now left with the problem that the audience will not be able to hear the drums.

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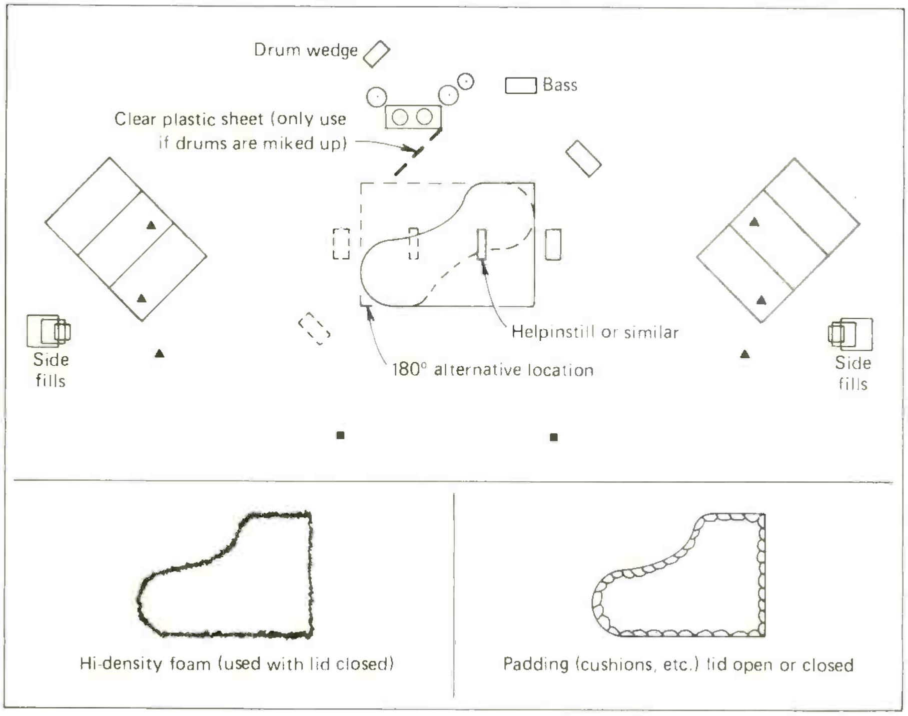
It is possible to "deaden" the piano by one of two methods. The easiest and least effective is to literally pad the inside with sound absorbent material—cushions, etc. The other method is to affix a layer of high-density foam all the way along the top of the piano and close the lid. This will then form a "dead" vibration free enclosure. (This won't affect the performance of a piano pickup.)



As mentioned above, turning the piano through 180 degrees would prevent any noise from the drums from reaching the soundboard. (N.B. You omitted to say whether or not the piano was an "in house" unit.)

With regard to monitors, use of a graphic will enable higher levels to be reached before feedback occurs. In addition, two wedges are not really sufficient for the wide angle of coverage required. A better method would be to locate one wedge behind the drummer and the other by the pianist, and to employ "side fills" for the remainder of the artists especially when taking the location of the vocal section into account. A basic side fill comprising of a JBL 4560, JBL 2482, mid-range driver attached to JBL 2395 lens and a JBL 2440/20 high-range horn, should serve admirably (per side).

If you have any further problems or need further advice, please do not hesitate to call me at 805-448-1966.

—Steve J. Griffiths
Chief Technician
Tasco Sound Ltd.
Newbury Park, Ca.

CAN YOU TELL THE EXPENSIVE IMITATIONS FROM THE AFFORDABLE ORIGINALS?

When Phase Linear introduced the first real high-power, high-fidelity amplifier in 1971, the philosophy of audio component design was abruptly changed. Almost overnight, Phase Linear became synonymous with high-quality, high-technology, high-powered high-fidelity.

The entire audio industry was forced to take notice.

Most Hi-Fi companies had not expected high power to become generally accepted as the optimum means of achieving sonic realism. Suddenly, companies found themselves spending large amounts of money in R&D, Engineering, and Marketing in a very short period of time.

Who do you think picks up the tab?

Contrary to what some manufacturers would like you to think, Hi-Fi companies are not philanthropic organizations. They can't absorb the cost. This rapid expenditure of money must be factored into the retail price of the component.

The result?

You pay a higher price. You most certainly receive the performance as advertised by these expensive imitations. But you also pay a "penalty."

Take a good look around the next time you visit a high-fidelity store. You'll see a lot of models on the shelf. All sizes. All shapes. All colors. All costing more than the original.

The Phase Linear 700 started it.

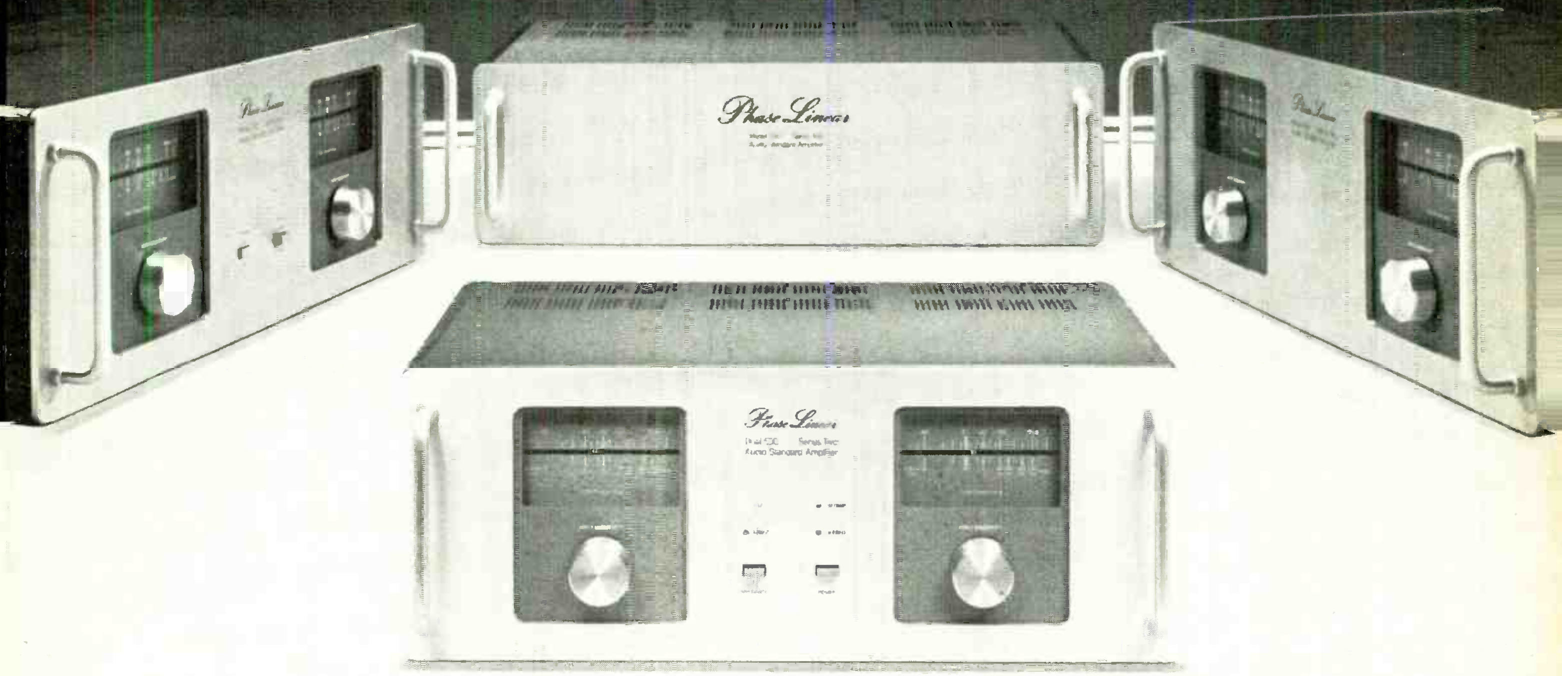
Since we started it all, we were able to advance in an orderly, cost-effective manner, and improve on our original technology. There's no tab to pick up. No penalty to pay.

The result is the Phase Linear Series Two.

The line is highlighted by the new D-500 rated at over 500 WATTS PER CHANNEL, with distortion at an incredible 0.09%. Designed to handle speaker impedances down to 2 ohms, the D-500 easily adapts to rugged professional use, or demanding home applications. A self-contained, thermally activated cooling system, combined with an instantaneous LED display incorporating built-in output clipping indication, allows for precise power control. Electronic energy limiters and independent fusing of the power supply prevent the possibility of damaging overloads.

Affordable Technology.

This same dedication towards exceptional performance, design and affordability is found in each Series Two Power Amplifier. From the "Kilowatt," to the fully complimentary output, BI-FET process and high-loop gain technology of the Phase Linear 200-Series Two, there is an affordable original that easily fulfills your most demanding requirements.



	D-500-Series Two	700-Series Two	400-Series Two	200-Series Two
OUTPUT POWER PER CHANNEL*	505 WATTS	360 WATTS	210 WATTS	120 WATTS
Intermodulation Distortion (60Hz: 7kHz = 4:1)	0.09% MAX	0.09% MAX	0.09% MAX	0.09% MAX
Signal to Noise Ratio (IHF "A")	110dB	110dB	110dB	110dB
Residual Noise (IHF "A")	120uV	120uV	120uV	120uV
Forced Air Cooling	YES	OPTIONAL	OPTIONAL	NO

Optionally available in Standard E.I.A. rack mount configuration. Solid Oak or Walnut side panels optional.

*Output Power Minimum RMS per channel into 8 ohms from 20Hz-20,000Hz with no more than 0.09% Total Harmonic Distortion

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 76 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

RACK MOUNTABLE POP AND CLICK KILLER

Claimed to be offered in response to many requests for the convenience and styling of rack-mounting is the model TNE 7000A, the transient noise eliminator by KLH Burwen Research. The device is designed to eliminate ordinary "medium and small" ticks and pops transmitted from disc recordings as a result of scratches, dirt, static buildup and surface imperfections. Additionally, a music simulation circuit in the TNE 7000A is designed "to keep the musical signal smooth without introducing its own noise when taking out impulse noises."

CIRCLE 1 ON READER SERVICE CARD

NEW SONY CASSETTE DECKS

Four new cassette recorders—all front loading—have been announced by Sony Industries, which now operates as one of three companies under the wing of Sony Corporation of America. The new models include the TC-K1A, with DC servo-controlled motor and Dolby noise reduction, list price $180; the TC-K4A, whose DC motor includes a frequency generator and which uses a ferrite-and-ferrite head, list price $280; the TC-K60, with liquid-crystal display of peak signal levels plus automatic "Music Sensor" for automatic playback of up to nine tape selections, list price $550; and the TC-K96R, a microprocessor-controlled deck that offers a variety of programming options as well as automatic reverse capability in both record and playback, list price $620.

CIRCLE 2 ON READER SERVICE CARD

COMPATIBLE VIDEO CAMERA

A lightweight video camera, said to be usable with all existing videotape and videocassette systems on the market, has been announced by Akai. The model VC-8300 is described as a high-resolution black-and-white camera "that is compatible with both Beta and VHS formats as well as with Akai's own VT-300 and VT-350 systems." Of modular design, the VC-8300 consists of the basic camera

with 16-mm lens; a universal C-mount permits use with other lenses. An 8:1 zoom lens is offered by Akai as an option. The camera includes an optical viewfinder, but it also can be used with an electronic viewfinder. Built into the camera is an omnidirectional electret microphone for sound pickup. Also featured is an automatic light control and a warning light to prevent operator error. Weighing only 1.75 pounds, the new camera is priced at $395.

CIRCLE 3 ON READER SERVICE CARD

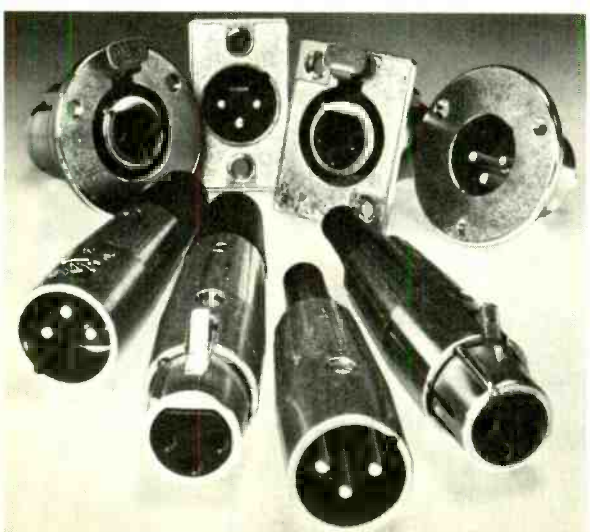
NEW TEST INSTRUMENTS

Leader Instruments Corp. of Plainview, N.Y. has announced a new product group consisting of oscilloscopes, frequency counters and "high-performance" audio instrumentation. Leader has called special attention to its model LBO-308, a 3-inch compact battery-operated 'scope described as a 20-mHz device with 2 mV/Div sensitivity selling for about $1000, and to its new 520-mHz Frequency Counter which features a large, bright 8-digit fluorescent display, zero blanking, gate time and overflow indicators in addition to 20 mV/RMS sensitivity and pushbutton attenuation.

CIRCLE 4 ON READER SERVICE CARD

NEW AUDIO CONNECTORS

A new line of low-impedance audio connectors for broadcast and professional use has been released by ADC Products, Minneapolis, a division of Magnetic Controls Company. Input and output three pin/contact connectors are electrically and mechanically designed to meet EIA RS297 standards, and are claimed to offer high reliability, total interchangeability and complete compatibility with all other audio connectors on the market. A separate ground terminal is electrically integral with the connector shell. Mechanical design includes alignment keying for correct plug/receptacle mating, secure fitting and ground continuity. The company, operating since 1935, also produces transformers, jacks and jack panels.



CIRCLE 5 ON READER SERVICE CARD

PHASE FILTER

The Phase Filter from Symetrix, Inc. of Seattle, Washington is described as a studio-quality signal processing device intended for recording or "live" performance applications. Phasing effects are accomplished by frequency-spectrum notching. Eight 90-degree phase delays in the signal path create four notches. Unlike flanger designs, in which the notches are spaced in musical octaves, the Phase Filter's notches are related by constant-frequency bandwidths. This technique is credited with achieving a very strong phasing effect with "none of the objectionable pitch-blending found in flanging devices." Controls include input level (with LED overload indicator); two variable low-F oscillators with LED rate indicators; a manual/auto sweep selector for LFO 2; blend; and depth and resonance. Low-level sources such as electric guitar and bass may be jacked into the front panel; line level inputs and outputs are at the rear along with a low-level output for onstage use. Also at the rear are jacks for an external sweep control and for an in/out footswitch (available as an optional accessory). Supplied in a rack-mount (19 × 1¾ inches) case, the Phase Filter is priced at $299.

CIRCLE 6 ON READER SERVICE CARD

KENWOOD DEBUTS TEST SYSTEM

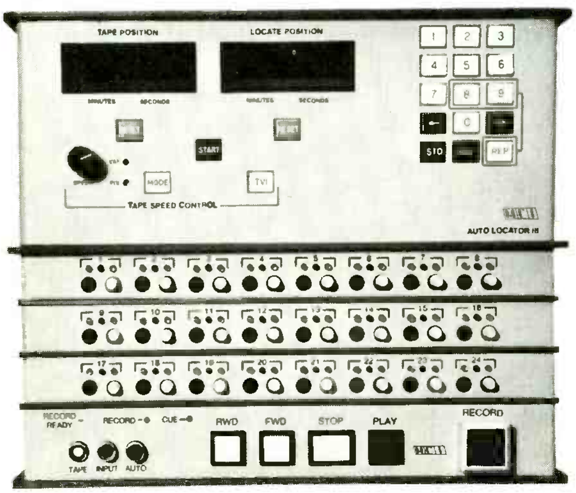
From Kenwood there's news of the model SE-3000 Acoustic Measuring System. A three-meter device with automatic chart printout, it is designed for making sound-field measurements and for measuring audio equipment performance. In the former category, the SE-3000 may be used for measuring transmission frequency response, reverb characteristics, sound insulation characteristics and noise level. In the latter application, the device can measure amplifier characteristics, tape deck frequency response and phono cartridge frequency response and crosstalk. Supplied in a carrying case, the SE-3000 consists of five functional sections plus power supply. The sections are: signal generator, level meter, reverb meter, recorder and microphone — the last being an electret condenser rated for response of ± 3 dB, 20 Hz to 20 kHz.

CIRCLE 7 ON READER SERVICE CARD

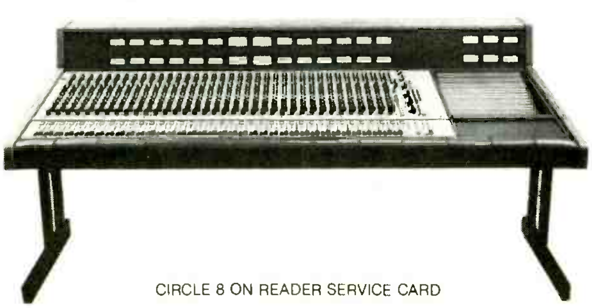
MCI SHOWS TAPE INNOVATIONS

Described as presenting the first change in recording tape since 2-inch tape was introduced in 1969, the new model JH-32 from MCI uses 3-inch tape, has 32 input tracks and introduces the new tape speed of 20 inches-per-second. The wider tape is credited with improvements in noise and transient response. In addition to the new speed, the unit also has 15 and 30 ips speeds.

Also new from MCI is its JH-600 series console available in two frame sizes (18 and 36 in and out buses). The console, which is equipped with MCI's JH-50 Automation as standard equipment, includes 24 channel buses with panning, six send buses and a true parametric EQ with variable Q and high- and low-pass filters.



Other MCI offerings include an Autolock unit for syncing various combinations of audio and video (including film) equipment; the Autolocator III which provides ten memory locations and a shuttle function; and the RTZ III which provides four tape position memories for 1/4-, 1/2-, and 1-inch transports. The RTZ III/M, designed for the JH-110M systems of tape-to-disc transfer, controls lathe functions by means of twenty additional memories.



CIRCLE 8 ON READER SERVICE CARD

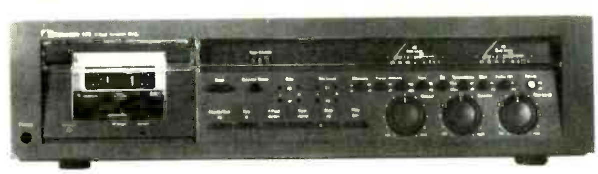
CAMEO PLANS BOOK

The new group CAMEO (Creative Audio and Music Electronics Organization) is planning a 160-page book on pro audio basics, to appear in the next few months. The book will cover sound, microphones, mixing consoles, signal processing, power amps, speakers, recording systems, sound reinforcement, and MI interface. A glossary will be included. CAMEO, formed about mid-1978, already has twenty-eight member companies and is inviting other organizations in the expanding pro sound industry to join.

CIRCLE 9 ON READER SERVICE CARD

INNOVATIONS IN NAKAMICHI DECK

The Nakamichi 580, a three-motor, two-head cassette recorder, boasts some technical innovations that add up, says the company, to better sound. One is a patent-pending "breakthrough" known as the direct-flux erase head which — instead of relying on flux leakage to penetrate the tape coating — converts erase current directly to flux at the point of tape contact. A single pass over this head, says Nakamichi, is equivalent to using a bulk eraser. Another departure from conventional design is the "diffused resonance" transport of the model 580. In this double-capstan design the flywheel, capstan and pressure roller on the take-up side all are different in diameter and mass from their counterparts on the supply side. Since they rotate at different speeds, they do not contribute to resonant peaks at the same frequencies. This dispersion of resonances, combined with the use of new non-resonant materials in critical transport areas, helps reduce modulation noise.



Electronic "firsts" for the cassette format include a circuit (for both record and playback amplifiers) known as the double NF (negative feedback) configuration. The record circuit is totally "DC" (direct coupling to the head). Results claimed are lower distortion, reduced phase-shift and added headroom

The model 580 uses Nakamichi's r/p "superhead" whose specially formed 0.9-micron gap is said to allow flat response to 20 kHz. Price is $650.

CIRCLE 10 ON READER SERVICE CARD

MC PICKUP NEEDS NO BOOSTER

The first moving-coil phono cartridge that needs no signal booster (transformer or "pre preamp") is the Dynavector 10X, made in Japan and marketed here by ESS, Inc. The high output voltage of this pickup is the result of its use of finer coils of wire with many more turns than in the past. Average tracking force is 1.5 grams. Retail price is $120.
P.S.—It really works.

CIRCLE 11 ON READER SERVICE CARD

TAPE TOPICS

Having roused widespread interest (and not a little concern over an implied need for equipment modification) with the recent announcements of metal-particle tape, the industry seems now to have cooled it somewhat.

What we're getting these days is not new word of metal tape, but news of improvements in oxide tapes and some development in video cassettes. For instance, BASF has introduced its improved studio cassette series which claim higher coercivity, higher bias and better performance due to a recently developed oxide particle which delivers 1 to 2 dB more output at the high end. No increase in price is planned at this time.

CIRCLE 12 ON READER SERVICE CARD

Memorex has a new high-bias audio cassette line that replaces its chromium-dioxide line and features an advanced ferrite-crystal oxide formulation for use with high bias (chrome) and 70-microsecond EQ. An improved album cover has been designated to go along with the new tape.

CIRCLE 13 ON READER SERVICE CARD

Maxell has announced two new cassette lines. One is the LN (ultra low noise); the other is UD (for ultra-dynamic). Both employ new magnetic formulations for improved dynamic range and greater output in the high frequencies. Package design of both types also has been changed.

CIRCLE 14 ON READER SERVICE CARD

From Irish Tape there's word of new "snap pack" modules containing its series 2000 cassette tape, this company's premium line in which special effort has been made to assure against head wear and tape clogging.

CIRCLE 15 ON READER SERVICE CARD

On the video front, BASF—which last spring announced it was preparing chromium-dioxide cassettes for the Betamax system—has been joined by TDK in this newly developing field. The TDK video cassettes for the Beta-format will use Super Avilyn tape. TDK also offers the same tape for the VHS video format.

CIRCLE 16 ON READER SERVICE CARD

The 3M Company, which introduced cassettes for the Beta format in mid-1977, now is also offering cassettes for the VHS format. 3M's announced prices for the VHS videocassettes are $17.95 and $24.95 for 1 to 2 hour, and 2 to 4 hour, lengths respectively.

So what's happened to metal tape?

CIRCLE 17 ON READER SERVICE CARD

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

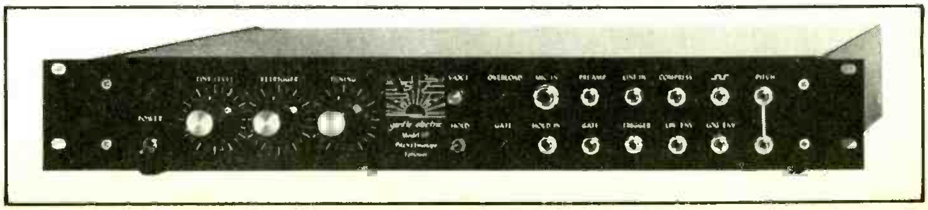
Vocoders seem to be the hot item in the synthesizer world at the moment, and the latest entry in the field is a sophisticated unit from Moog (Norlin Music). In simple terms, a vocoder continuously analyzes the tone-color of an audio input and applies the articulation of tone-color that occurs in this audio signal to a second audio signal. In the most common application, the first signal is a human voice and the second signal comes from a synthesizer and the result can be thought of as a "talking synthesizer" or as an "electronic voice." (A well-known example of this application is the voice of a Cylon warrior on the Battle Star Galactica television series.) The Moog Vocoder analyzes the program input and processes the carrier in sixteen separate frequency bands. An unusual feature of the Moog unit is that the analyzer outputs and synthesizer inputs for each of the sixteen channels is available on front panel jacks to allow cross-patching for unusual effects or connection of individual channels into an external synthesizer system; a patch select switch is available on the panel or as a foot switch to allow instant selection of normal or patched operation for greater versatility. A sample-and-hold function is also built in to allow a particular instantaneous articulation to be held for an extended period. Level controls with overload indicators are provided for the line level carrier input and the program input which may be either mic or line level. For improved intelligibility the Moog Vocoder treats high frequency signals above 5080 Hz differently than lower frequency signals which go through the sixteen analyzers; the high frequencies may either be fed through directly or they may be gated off except when there is high-frequency information present in the program input, and there is a balance control to tailor the mixture of vocoded low frequencies (termed "buzz") and high frequencies or "hiss."

CIRCLE 18 ON READER SERVICE CARD

Oberheim Electronics has announced the availability of a cassette storage interface for Oberheim programmable synthesizers. Either sets of patches or program sequences may be stored on standard tape cassettes using a standard portable cassette deck, allowing the user to develop a library of programs which are then available in a matter of minutes at a later time. When recording data, it takes approximately 45 seconds to dump sixteen complete programs from an Oberheim Four Voice synthesizer or approximately 24 seconds to dump eight patches from an OB-1 lead synthesizer. The unit has a check switch and error light to allow verification of error free recordings before the program is lost from the unit's memory.

CIRCLE 19 ON READER SERVICE CARD

Gentle Electric has introduced a pitch and envelope follower which can be used to interface any monophonic instrument to a synthesizer. The pitch of the instrument generates a control voltage on a 1 volt per octave basis, while envelope control voltages are generated on both linear and logarithmic bases. Additional outputs from the unit are provided for gate and trigger with variable sensitivity, plus a fundamental-frequency pulse wave. The unit includes a low-noise preamp so that it can accommodate almost any input signal whether from a microphone, instrument pickup or tape recorder. A compresser circuit is also included in the unit to allow a variety of unique effects not available with other synthesizer interface units.



CIRCLE 20 ON READER SERVICE CARD

MUSICAL INSTRUMENTS

RolandCorp US has announced a new electronic piano, the MP-600. This new model has a 64-note keyboard which utilizes a new mechanical action harpsichord, which can be mixed in any proportion with the variable voice sliders for a wide range of sounds. The

basic range of sounds can then be further modified by the six-band graphic equalizer which is built into the unit. Another useful feature is a variable decay time slider which ranges from poppy, percussive articulation to long, slow decays.

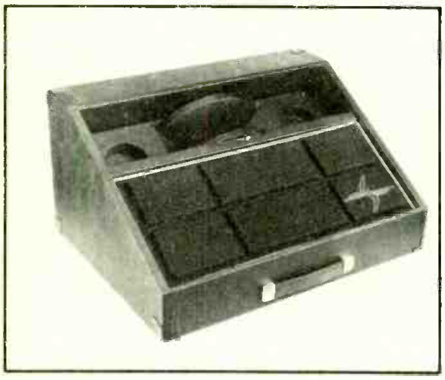
CIRCLE 21 ON READER SERVICE CARD

Novaline Piano Co. has announced the introduction of a new line of electronic pianos featuring Concertouch action. With this new system, the pianist has fingertip control of dynamics over an extremely wide dynamic range. Two models are currently available with Concertouch, one a 64-note model and the other a full 88-key version.

CIRCLE 22 ON READER SERVICE CARD

SOUND REINFORCEMENT EQUIPMENT

Shure Brothers, Inc. has introduced a new stage monitor speaker with the unique feature of variable high frequency dispersion. The Model 703 is a compact, two-way system using two 8-inch woofers and a special high-fre-



quency driver which has a horizontal dispersion of 120° by itself, or of only 60° when two removable acoustic wedges are inserted. This allows the speaker to cover a wide area for several performers or one performer who prefers to move around, or to be narrowed down to minimize spill and feedback or to provide more localized coverage. In addition, the speaker cabinet is designed to be used either in an upright position for long-throw or side-fill applications, or tilted back for placement at the musician's feet. The frequency response of the Model 703 is specially tailored to emphasize the presence range which is the most crucial in stage monitoring applications, and to eliminate excess bass which often leads to boominess or muddiness in the on-stage sound. The 703 is an 8 ohm speaker system and is rated at 100 watts of continuous program. The efficiency is also quite high at 97 dB SPL at 4 feet with 1 watt input so that it does not require large amounts of amplifier power to cut through a high on-stage sound level.

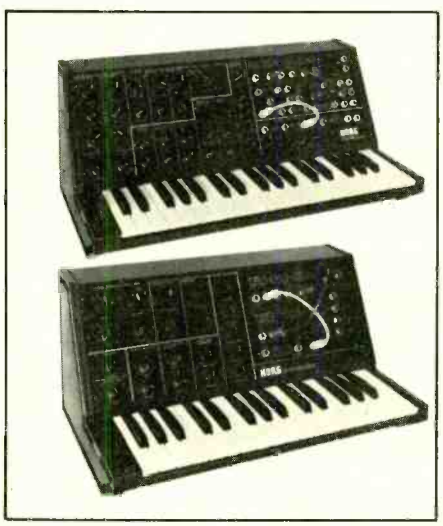
CIRCLE 28 ON READER SERVICE CARD

Dynacord Electronics offers the EC 280, an electronic echo/reverb unit. The system is basically an analog delay line and thus has no mechanical moving parts to wear out. Among the effects possible with the EC 280 are voice doubling or multiplying, phasing, true flanging, chorus effects and stereo operation. Two sets of four independent pushbuttons control the delay times allowing the user to program a variety of echo and reverb effects. Maximum delay of the system is 280 milliseconds.

CIRCLE 23 ON READER SERVICE CARD

SYNTHESIZERS

Unicord, Inc. has expanded their line of Korg synthesizers with the addition of the MS series, which are designed to provide a maximum of features at a reasonable cost. The MS-20, for example is a fully variable, patchable (and hence expandable) synthesizer with a $750 list price. The MS-20 includes two 10-octave VCOs with selectable, mixable waveforms, two VCFs with resonance controls, variable modulation controls, ADSR envelope generator with a variable hold control, LFO envelope generator with variable delay, a patchable VCA and pink and white noise generators. In addition, the unit has a built-in External Signal Processor, which is basically an envelope follower allowing

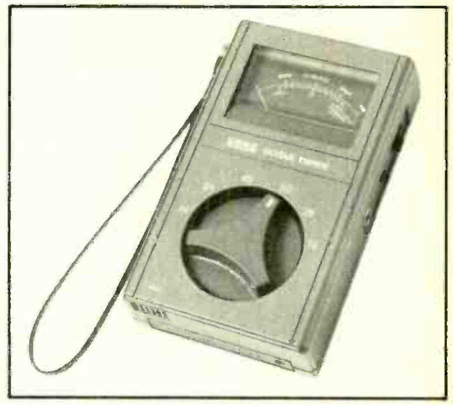


an external instrument to control the MS-20 without the need for a separate interface unit. Another model in the MS series is the MS-10, which is a single-VCO unit which can be used by itself or as an expander for another synthesizer.

CIRCLE 24 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

The Korg GT-6 guitar tuner is a recent addition to the product line from Unicord, Inc. The GT-6 is a compact, battery-powered unit which is small enough to fit in the accessory compartment of most guitar cases. The unit is operated by selecting the desired note



on the selector switch; then reading the illuminated meter when the string is played. The GT-6 has a built-in microphone for tuning acoustic instruments plus in and out jacks for tuning electric guitars or basses while still listening to the instrument through the amp.

CIRCLE 25 ON READER SERVICE CARD

New from Ross Musical Products is the Ross Distortion unit. The unit is a high-gain amplifier (40 dB of gain) which is capable of generating clipping type distortion with inputs as low as 1.5 mV peak. The unit is housed in a rugged, cast aluminum chassis with recessed knobs and is powered by a 9-volt battery or an external AC adaptor.

CIRCLE 26 ON READER SERVICE CARD

Advanced Audio Designs has introduced two, new, high-quality preamplifiers, one for guitar and one for bass guitar. Both models comprise two, series-connected preamp stages for maximum control of noise, distortion and overdrive characteristics, and are very compact designs which mount in 1¾" of rack space. In addition to gain controls for each of the two preamp stages and a master gain control, the Guitarist Model 101 and the Bass Guitarist Model 101B feature extensive equalization facilities. Three bands of equalization are provided, each with ±15 dB boost or cut and switch selection of two different center frequencies, plus low frequency and high frequency boost switches. For the Guitarist, the center frequencies are 125 Hz/250 Hz bass, 1 kHz/2 kHz midrange and 3 kHz/4 kHz treble, while the Bass Guitarist has lower center frequencies of 75 Hz/150 Hz bass, 500 Hz/1 kHz midrange and 2 kHz/3 kHz treble. The design is all solid state for low noise and reliability. Signal-to-noise ratio is given as 70 dB below a 2 volt RMS output; the output impedance is 500 ohms, and distortion is specified as ranging from .01% to 85% total harmonic distortion depending on the settings of the gain controls.

CIRCLE 29 ON READER SERVICE CARD

JHD Audio has three related products for users of Fender reverb-type amplifiers. All three models plug into the back of the amplifier in place of the reverb unit and create a powerful overdrive sound controllable by the reverb footswitch and the reverb knob on the amp which varies the amount of boost. The basic model is the Ice Cube II which has no controls on the box itself. The Super Cube allows use of the reverb at the same as the boost effect and has a blend control to vary the effect between boost only, reverb only or any mix in between while the front panel control still varies the overall

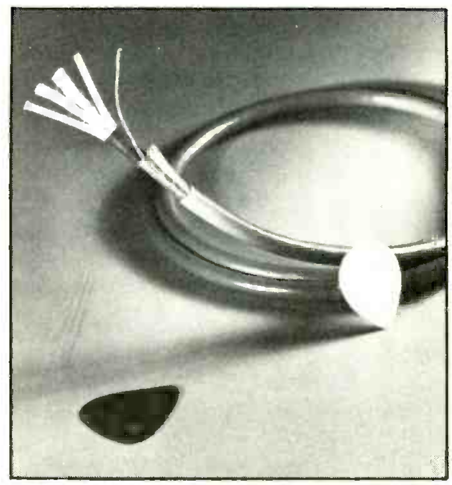
strength of effect. The Super Cube is for use with pre-1978 models while the new Super Cube II is for use with the latest model Fenders.

CIRCLE 30 ON READER SERVICE CARD

In a sense, the ultimate musical instrument accessory is information, and that is exactly what is being offered by *Device*, a new newsletter for electronically-oriented guitarists and musicians. Several newsletters already exist for the synthesizer user/enthusiast, but until the advent of *Device* periodicals for the guitarist have been almost exclusively performance-oriented. *Device*, on the other hand, is equipment-oriented and features new product reviews, test reports, construction projects, feature articles on circuit design, and interviews. The newsletter is edited by Roger Clay and Craig Anderton. While *Device*'s avowed purpose is to improve communications between manufacturers and musicians, it will do this in a non-commercial, no-advertising format. One year subscription rates are $15 for U.S. residents, $16 in Canada and Mexico, and $18 internationally.

CIRCLE 31 ON READER SERVICE CARD

The most basic, yet most overlooked accessory item for electric guitarists and other amplified musicians is the guitar cord or hook-up cable. The earliest types of guitar cords were basically hi-fi cables and had many shortcomings for performance use—hum shielding was not very good, molded-on connectors lacked reliability, and the cables were prone to triboelectric noise, which is the noise caused by the friction between individual strands of wire within the cable. More recently, cords have become available commercially made from a variety of cable types including microphone cable, coaxial cable and instrumentation or computer cable, yet each of these types has its own shortcomings; coax is usually too stiff, instrumentation cable has excellent shielding but doesn't hold up under constant handling, and many cable types have too much capacitance to be used with the high impedances of guitar pickups without significant loss of treble frequencies. The Electronic Division of Belden Corporation has announced a new, low capacitance, high reliability, 100% shielded cable designed with musical instrument applications in mind. Belden Corp. is well known throughout the electronics industry as a manufacturer of highest quality wire and cable, and many of the better commercial guitar cords use Belden cable of one type or another. The new cable type is known as No. 9211, and was designed to combine the excellent shielding of computer type cables with the

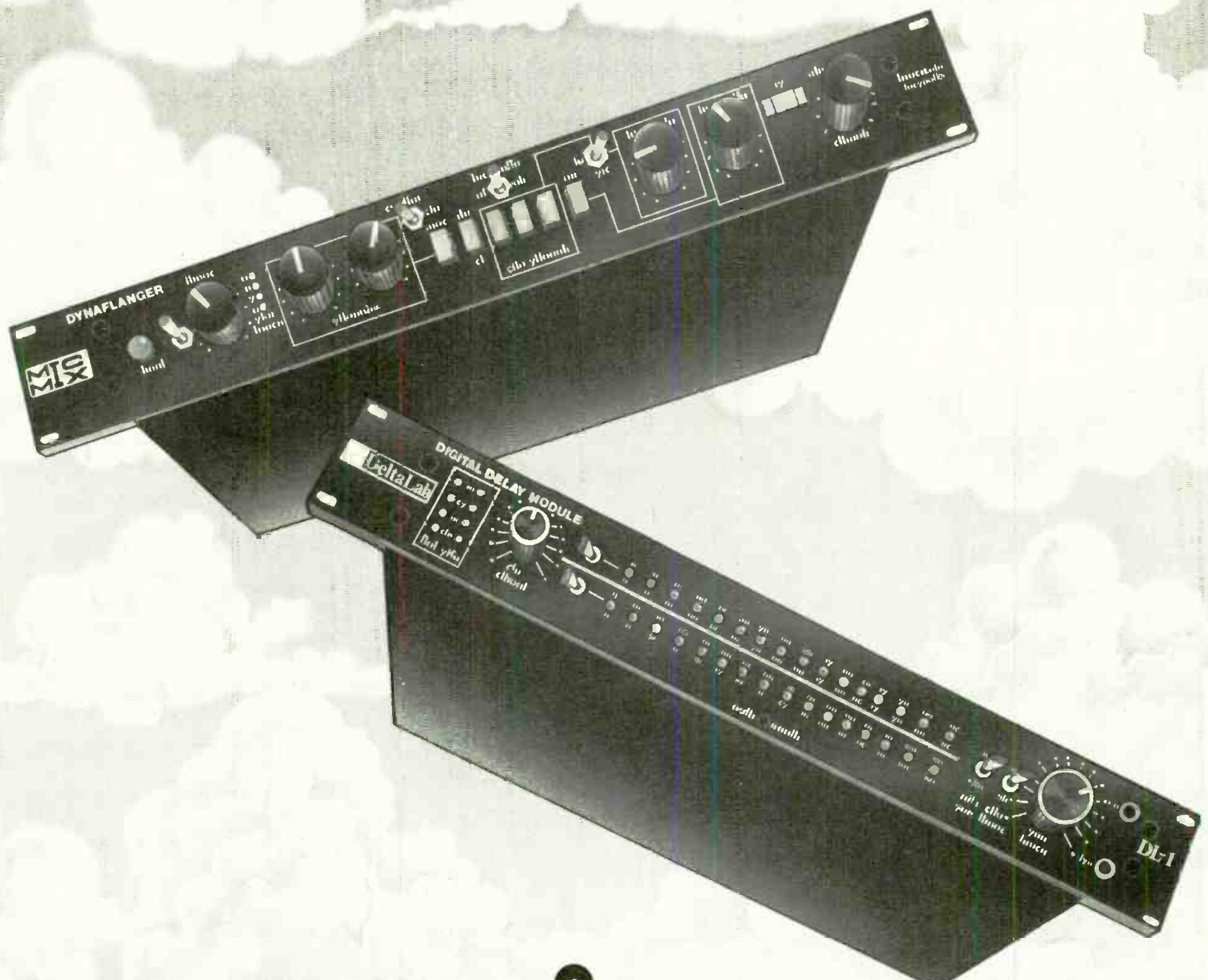


flexibility, ruggedness and low handling noise required for instrument-to-amplifier connections. The construction details of 9211 include two 25-gauge stranded tinned copper conductors (one of which is redundant in normal, unbalanced applications); each of the two conductors is insulated with a polyethylene jacket and the two of them are cabled or twisted with a conductive cloth tape wrapping to minimize handling noise. Over this tape layer is a Beldfoil® shield with 22 gauge tinned copper drain wire and a black or light blue vinyl jacket. Beldfoil is a laminate of aluminum foil and polyester film which yields 100% shield coverage with minimim bulk and maximum cable flexibility. The nominal outside diameter of 9211 is 0.247 inch, and the nominal cable capacitance is 12.2 pF/foot.

CIRCLE 32 ON READER SERVICE CARD

DEFY TIME & SPACE

THE MIC MIX DYNAFLANGER AND THE DELTA LAB DL-1, DL-2, DL-3



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MICROPHONES, MIXERS, MACHINES, SIGNAL PROCESSORS, AND A HOST OF MUSICAL EQUIPMENT.

CIRCLE 69 ON READER SERVICE CARD

Inside "Live From

Not so long ago—even at the beginning of this decade—the TV set was just about the last place anyone would turn for a satisfying musical experience. After all, even if the performers were exciting, the sound quality made it hardly worth the aggravation, particularly with the wide repertoire available on comparably high quality stereo disc recordings.

In 1973, however, the first of a series of experiments was quietly staged during a Metropolitan Opera performance of *Tales of Hoffman.* Under the direction of John Goberman, head of Lincoln Center's Media Development Department, and Mark Schubin, a young technical wizard who had signed on with Goberman a year earlier, the performance was taped, problems were noted and solutions were found. The idea was to film an event under actual performance conditions—that is, with no restaging, with an audience in the house and with absolutely no interference from the camera, lighting and sound crews.

After just more than two years of such experimentation, on January 30, 1976, "Live from Lincoln Center" was born, the premiere show being a New York Philharmonic concert conducted by Andre Previn, with guest soloist Van Cliburn. This past September, "Live from Lincoln Center" began its third season—one that promises the usual complement of Philharmonic concerts, solo recitals, operas from both Lincoln Center opera houses, ballet, and for the first time, chamber music concerts.

Of course, the broadcasting of classical music on television is nothing new. In the forties and fifties, when the medium was young and the quality of its audio not that much worse than the "high fidelity" of the time, broadcast concerts were an important part of local programming in certain parts of the country. But by the early sixties, the gap between TV sound and standard hi-fi/stereo sound had grown rather wide. At the same time, network executives began to see that they could attract a much larger audience (which meant, naturally, more advertiser dollars) with witless situation comedies than with tinny sounding orchestra shows that even music fans were no longer enjoying.

Ironically, the same TV/hi-fi technology gap that took music off the small screen made its return possible; the parallels between the development of stereo FM and stereo television, which is on the way, are noteworthy. Just as the early experiments in stereo required the listener to use two

Lincoln Center"

By Allan Kozinn

Photos by Susanne Stevens

radios—one to receive the station broadcasting the right channel and the other to receive another station broadcasting the left—current simulcasts require the viewer to use his TV screen for the visuals and his FM receiver for the stereo soundtrack. Goberman and Schubin foresee, and in fact are working toward, a time when the simulcast will be as obsolete as the dual station stereo broadcast, and when "Live from Lincoln Center" will be available only via a subscription cable. But that's all sometime in the future. For now, "Live from Lincoln Center's" primary objective is to bring the sights and sounds of Lincoln Center's various performing institutions into homes coast to coast, free of charge.

"What we're doing," says Goberman, "is transmitting a performance—and event—without interfering with that event or with the audience which has come to see that event 'live.' "

The operational word is *transmitting,* which Goberman uses in the most passive possible sense, as opposed to *producing.* As Mark Schubin clarifies, "We want to give the lie to McLuhan, to show that the medium is *not* the message. We feel that we should be nothing but a pipeline from the stage to the viewer, and that the less electronic magic the viewer perceives, the more effective the performance is."

Back in the experimental days, Goberman and Schubin knew that to comply with their own non-interference dictum, they would have to dispense with what Schubin calls "a television mentality"—one that calls for scaffolds full of extra lighting equipment, cameras and cameramen, microphones strung all across the stage, and consequently, an invited audience (since, as Schubin puts it, "A paying audience just wouldn't put up with that nonsense."). Cameras would have to be placed where they would least interfere with the viewing pleasure of the in-house audience, and the lighting (particularly in opera, where lighting is crucial) had to deviate as little as possible from normal concert lighting. With recently developed low-light cameras, the lighting aspect wasn't much of a problem.

But another aspect crucial to the success of the venture seemed more likely to pose a challenge to Schubin and Goberman: the sound quality. In something as complex as a staged opera, for instance, where singers would be wandering around the stage, the ideal solution might seem to be to use body mics. Yet, the producers feel that body mics would be an intrusion on the artists, interfering, at least psychologically, with the performance.

"It's very difficult," Shubin says. "What we try to do is mic the sound that's being presented to the house rather than miking individual instruments or singers. Our microphones tend to be outside the stage area, so that they pick up what is essentially the sound that the house audience is hearing. When we transmit a symphony concert, we use only four mics—two fairly close to the orchestra, two farther out in the house to pick up the ambience."

The microphones generally used are Sennheisers, two 415s for the orchestra, and two 815s for the ambience. No artificial reverb or delay is used. However, Schubin feels that absolutely nothing is sacrificed in the sound even though the miking system is fairly simple. In fact, during one of their early experiments, they taped the Philharmonic and had a record pressed from the concert tape. They then took the disc to several producers at RCA, who were convinced that it had been recorded in the studio.

Nor is the sound at all doctored. Even with two orchestral mics, it would be possible—although a complex procedure—to highlight instruments so that an oboe, for example, could be boosted during the oboe solo. This is something done on other concert programs. The Boston Symphony Orchestra's "Evening at Symphony," for instance, is recorded on a 16-track tape which is specially processed so that when an instrument appears on the screen, it is also given a slight boost on the soundtrack. "Evening at Symphony" producer Jordan Whitelaw calls this process "sweetening" of the sound, and was originally designed to help the audio (particularly in "Evening at Symphony's" non-simulcast days) overcome the limitations of TV sound. Goberman, however, doesn't believe in "sweetening."

"It destroys the authenticity of the performance," he insists. "When you sit in a concert hall, the sound isn't enhanced for you. Also, 'Live from Lincoln Center' is just that; it's a 'live' program, a sharing of an actual event. The viewer at home can feel the same tension as the 'live' audience and the performer. We are of a transient nature, transmitting an event that will not exist next week."

Video switching area of the 40-foot mobile television studio during a recent broadcast.

In fact, the event may exist on videotape next week, and it may be shown once more after the original broadcast date, but no more. "Live from Lincoln Center," unlike "Great Performances" and "Evening at Symphony" is not out to document events or performances, just to convey them. The benefit to the performer, says Goberman, is that "they don't have to feel that this performance will haunt them for years to come. If our intention were to produce a tape, we wouldn't go about it the way we are. A different set of aesthetics would apply."

In transmitting opera, two additional mics are usually used. The stage sound, however, is transmitted in mono, while the music and ambience are sent out in stereo. This is done to avoid confusion that might otherwise be caused when camera angles changed but voices coming out of the left and right speakers stayed in the same relationship.

In the early days, all the mics used during the performance were mixed through a Sony MX-20 monitor amplifier [MX-20 now designates Sony's top-of-the-line mixing console], but they have recently switched to a Yamaha. Solo instruments are occasionally miked separately. In the premiere broadcast, for example, when Van Cliburn performed with Andre Previn and the New York Philharmonic, an AKG-451 (with a swivel cardioid capsule, to capture the reflected sound) was placed inside the piano.

If Goberman and Schubin feel somewhat constrained in their manipulation of the audio, they—and their director, Kirk Browning—feel freer with the visual side of the show. If they were going to greet the video the same way they present the audio, the show would be shot with one camera strategically placed at the back of the hall, with no special camera angles, close ups, fade-ins or other intricate cinematic devices.

"But television is not a good enough medium for that," Schubin says. "If there was a way we could use one camera to produce an incredibly large image so that the effect for someone sitting in his living room would be the same as sitting in the first row, yes, that's how we'd do it. Right now, television, with its 525 scanning lines, is too primitive a medium. It works best when it's specific—a closeup of a singer, for instance—and least well when dealing with a mass."

"The key," adds Goberman, "is variety. In opera and ballet, your pictures are there, you just have to select them. With an orchestral concert it's different; you have to create your pictures from what is basically a non-visual melange of men and women. And those pictures must be the ones which best represent the music. You must have a sense that you are seeing what you are hearing, and there are lots of ways of accomplishing that. One way is to find the focus of energy in the orchestra and concentrate on that. But you can't do that the whole time, sometimes the opposite is more

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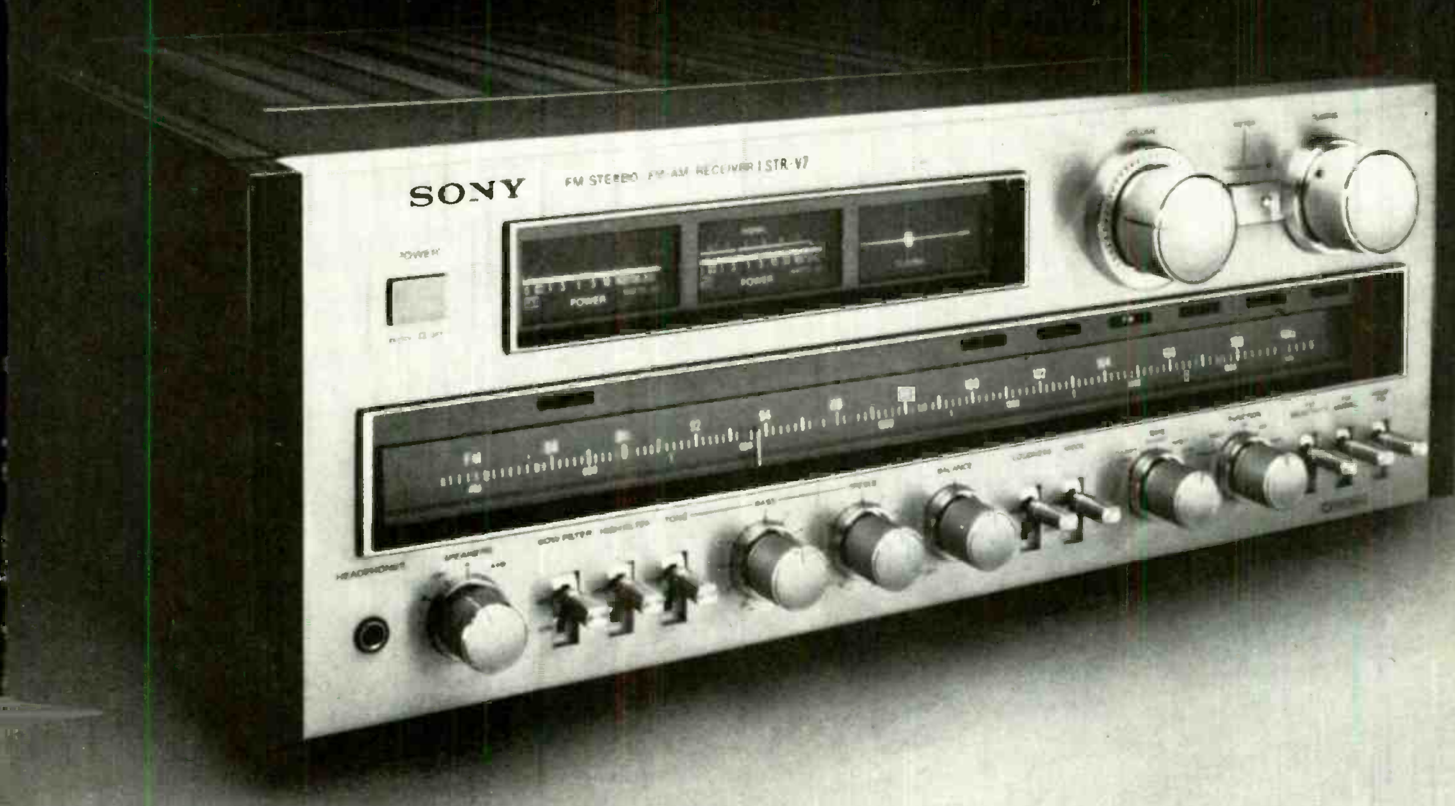

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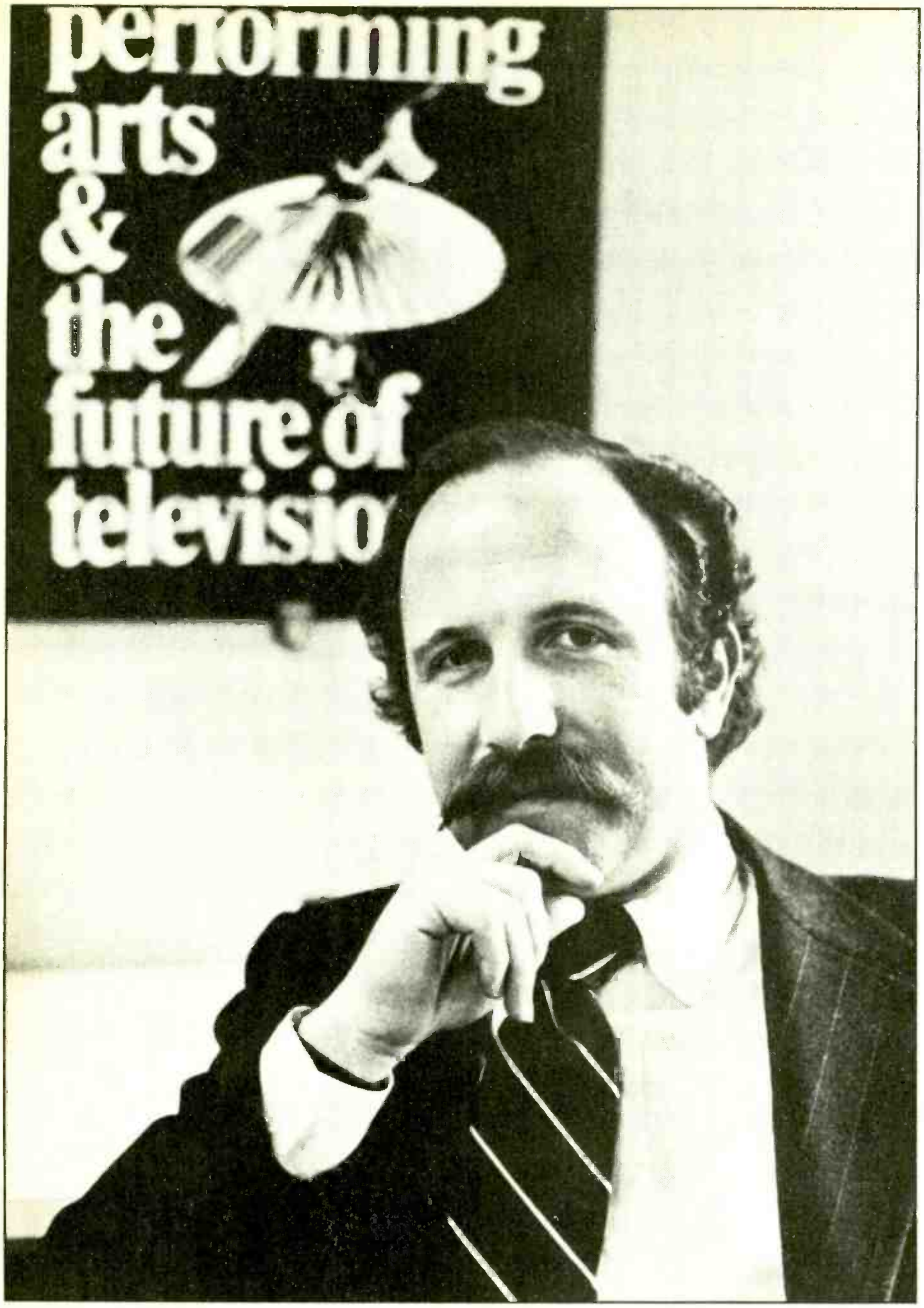

John Goberman, head of Lincoln Center's Media Development Department.

like putting together NBC from scratch," he says, although the stereo FM aspect is an extra side of broadcasting that NBC usually doesn't have to deal with. (The exception, of course, is the recent Mehta/Horowitz broadcast carried on NBC with a stereo simulcast).

There would be no need for this extensive network, of course, if television sets were made to carry stereo sound. According to Goberman, the major obstacle to stereo TV has been the phone company. "It's a vicious circle," he says. "If you ask the television manufacturers, who all have patents for stereo sets, they'll tell you that they aren't producing them yet because the stations are not broadcasting in stereo. If you ask the stations, they'll tell you that the networks are not feeding them stereo. If you ask the networks, they'll say that the phone company is incapable of feeding stereo through its lines. The phone company's attitude has been, 'Nobody has a stereo television set, so why should we bother?' And can you imagine the amount of inertia a company like that is capable of?"

Lincoln Center has tried to overcome Ma Bell's resistance to change by offering to pay for the equipment that would make stereo transmission possible. The offer was refused.

Satellite technology, however, is not only capable of transmitting stereo, but can deliver a much cleaner signal than Bell at a relatively low cost. At present, PBS is preparing to use its own satellite, the DATE (Digital Audio for Television) system, a system vastly superior to that which Lincoln Center has been using.

Lincoln Center's own satellite experimentation goes back to 1975, when the American Association for the Advancement of Science asked the Media Development Department to prepare an exhibition that would demonstrate television's value as a servant of the arts.

"One of the things we did," recalls Schubin, "was to set up a receiving station outside the Vivian Beaumont Theater and a projector on the stage inside. The Colorado Concert Ballet, in Denver, sent us the images of two dancers via a NASA ATS-6 satellite. This was not only the first transmission of ballet by satellite, but the first satellite TV signal received in the City of New York. Once the images of the dancers appeared on the screen of

effective. Sometimes it's best to see a person; other times it's best to see an instrument. When we did the *Rite of Spring* last year, we concentrated on the fingers, because we felt that that was the best way to show how Stravinsky has created his sounds, his magic.

"You have to use your imagination. There are things you can do with television to create an impact on the screen, and particularly with orchestral programs, we use those techniques a lot. Yet, again, we don't interfere. Our cameras are hidden, we in no way alter the activity on the stage. And what we find happens is that the performers forget that we're there and that they are doing a television broadcast. From the audio end, we believe that the conductor should have 100% control over the sound of the performance, so we don't tamper with it. It's like the old Toscanini sessions: he used to go into the recording booth, tell the concertmaster to have the orchestra play its loudest passage, have the engineers set the levels and insist that they leave them that way."

The real high technology—and it is quite literally that—involved in "Live from Lincoln Center" is the satellite transmission network Mark Schubin assembles before each broadcast. "It's

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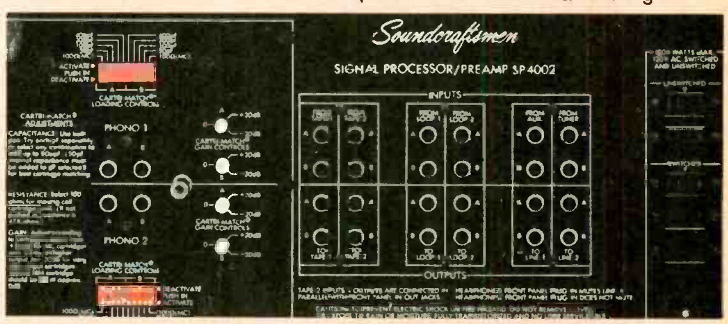

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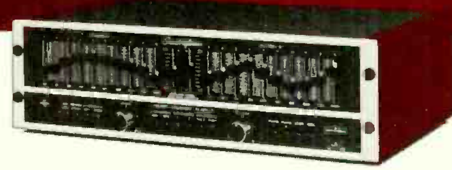

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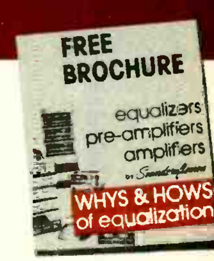

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the Beaumont Theater, we had a ballet master teach them the *pas de deux* from *Giselle.* He would tell them to stop, and they would stop. But he got so caught up in the demonstration that he forgot that the dancers couldn't see him. He kept saying, 'No, no. Do it like *this*!' "

A year later, on the occasion of the first "Live from Lincoln Center," Schubin achieved another "first." Using lines from ATT, Western Union and more than a dozen other carriers as well as a Western Union satellite, Schubin set up the first network capable of sending a stereo signal across the country.

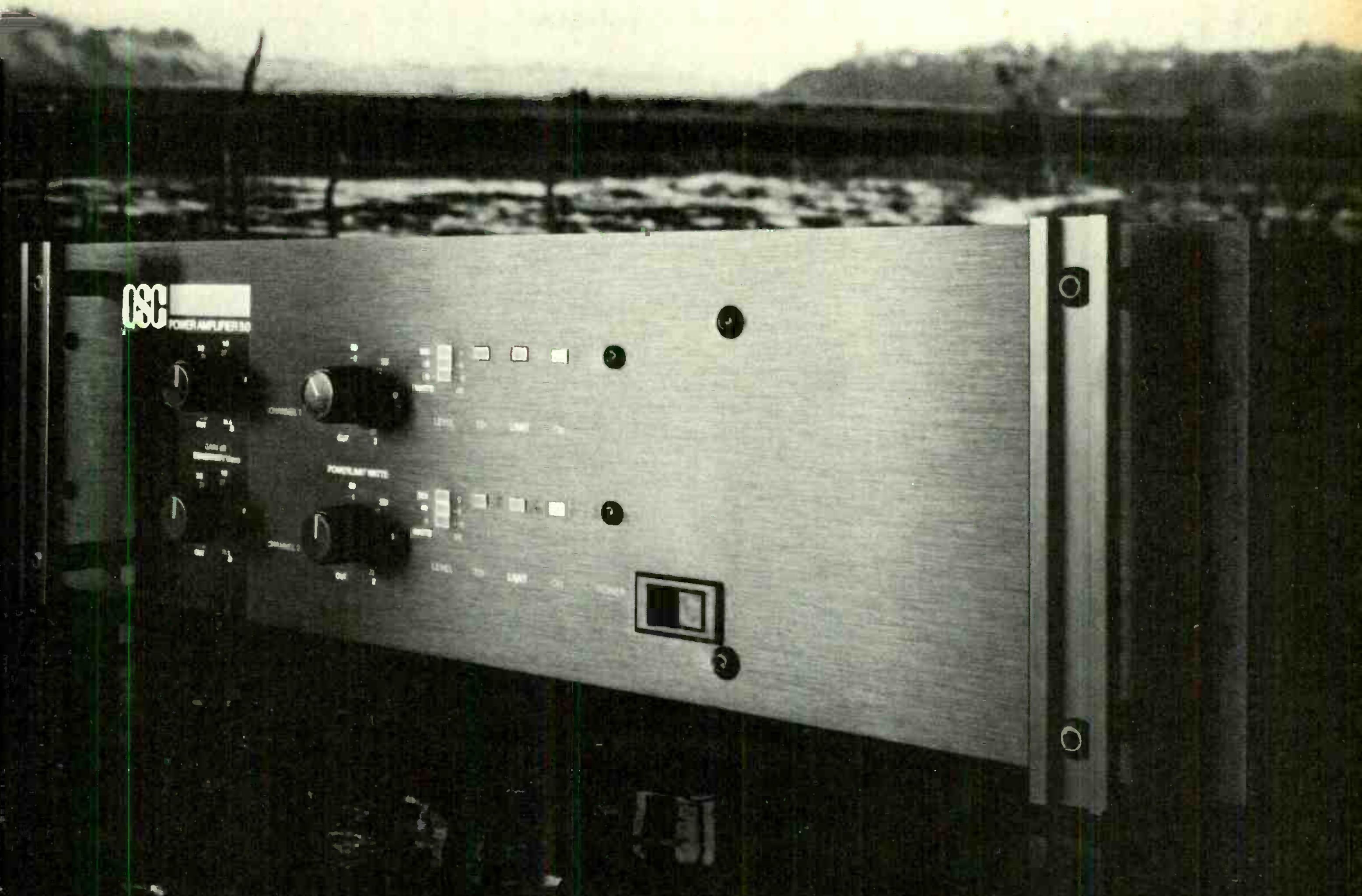
"Since that first show," he says, "everything we've done has gone out by satellite. The biggest problem, it turns out, is that the video signal—which travels via telephone lines—arrives one third of a second before the audio signal, which travels a 45,000 mile path to and from the satellite. To remedy that, we take the video and audio, and send them to the satellite together. The signal then comes back to New York before being fed to the rest of the country. Now, the signals, audio and video, arrive everywhere at the same time. Which means that 'Live from Lincoln Center' isn't really 'live', it's 'live' three seconds later."

The implementation of the PBS DATE satellite in early 1979 should have a great effect on the quality of "Live from Lincoln Center" audio. As Schubin says, "Any digital transmission system (which the DATE is) comes close to perfection. You can transmit a signal literally millions of miles without any loss of quality."

In terms of specifications, this means that when a signal is transmitted with a signal-to-noise ratio of 70 dB, it will be received with the same 70 dB S/N no matter what happens between the point of transmission and the point of reception. A comparable signal sent via an analog transmission system, such as the Western Union satellite Lincoln Center has been using these past few years, will be degraded to a noticeable extent, so that a signal transmitted with an S/N ratio of 70 dB may be received with an S/N of only 55 dB. According to Schubin, "Live from Lincoln Center" has generally done better than that: he has measured signals from his console that have reached to the transmitter in Los Angeles (via analog satellite) with a signal-to-noise ratio of 65 to 70 dB. That is not only a better specification than you'll get from a well-pressed LP record, but also better than any FM station is capable of transmitting.

"Right now," concludes Schubin, "our sound is certainly respectable. But the technology behind DATE will take our improvements in sound transmission even farther. It will help us reach places in the country that we can't get to with our own satellite. We welcome the DATE system, and can't wait until it is fully implemented."

That will be soon. But of course, DATE cannot be used to its fullest potential until homes are equipped with stereo televisions to receive its high quality signals. Until then, all we can do is tune in the "Live from Lincoln Center" broadcasts, turn down the TV sound, switch on the FM stereo, and enjoy.



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By Peter Weiss

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Generally, it is standard policy to write articles on concert sound reinforcement in a fairly impersonal, non-critical manner. But this time I'm going to step out from behind the shield of journalistic anonymity and take my chances with personal impressions and opinions. I feel justified in doing so in this case because Bob Dylan is not a recently arrived rock star. He is a veteran show business personality with well over 15 years exposure, experience and success; and I believe it is from this perspective that his current efforts should be viewed.

Too often, in both the print and electronic media, Bob Dylan is portrayed as "rock poet," "rock prophet," even "a god in the pantheon of the generation that grew up and out listening to his material." However, though Dylan's success is the result of his own talent, he is also primarily a product of the music business, as much packaged and marketed for selling as any other commercial success.

If this view is accepted, then it seems naive of Dylan's current critics to base their criticism (as many do) on a comparison of today's Dylan with the Dylan of a decade and a half ago. Certainly such comparisons cannot deal with the nature or quality of the man's talents, since these (it is almost universally agreed) have grown, or remained undiminished over the years. Rather they seem to be reactions to the new "style" that Dylan has acquired, especially in concert. If this "styling" is intended to insure continued commercial success, and does so, it should be welcomed by veteran Dylan fans and by critics who share the time perspective of those fans. It was the quest after the goal of commercial success, in a *package that suited the style of the times*, that initially made Dylan's messages available to such a wide audience, and allowed those messages to achieve the social impact that they unquestionably did.

Let us examine the style in which Bob Dylan is now presented to concert audiences. First, there is the designer-designed stage and drapings for monitor speaker enclosures. Also, the clothing worn by all of the performers on stage is carefully designed and coordinated. Visual requirements also dictates the placement of performers on the stage.

Madison Square Garden concert set-up featuring Stanal Sound's "flying P.A." system.

Flying Arrays

The sound reinforcement system contributes to the clean stage appearance since the main house speaker arrays are "flown" (hung) above the stage perimeter.

These arrays consist of pairs of Altec-Lansing 604 HPLN 15-inch radiators with coaxial horns, mounted in double 817 cabinets. The 604s are supplemented by companion arrays of horns driven by Altec-Lansing 291/16B drivers. Included in the horn arrays are 203B "long-throw" and 329 "short-throw" types. This main speaker system (817s and horns) is mounted in a framework of gimbal mounts, allowing each element to be positioned exactly as desired (see photo). Tim Charles, chief of the Stanal Sound crew and house mixer, said that with the gimbal mounts "you can be anywhere in the house and be looking down the throats of at least two horns and two cabinets."

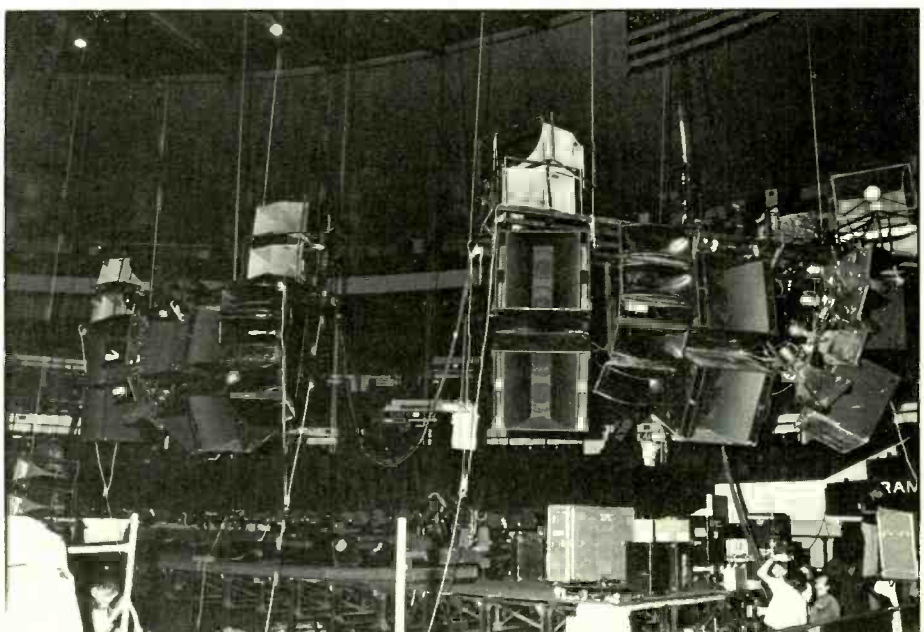
Since "flying" speaker systems have become popular, reinforcement operators have been working hard to substitute for the help they used to get in the bass region of the spectrum from stage mounted speakers. Stanal's answer is to employ two huge sub-woofer cabinets, each containing two Altec-Lansing 421-8LF 15-inch low-frequency loudspeakers. These elements are fed only that portion of the reinforcement mix which consists of frequencies 75 Hz and below. The effect of these sub-woofers, under good conditions, can be *literally* breathtaking. (Tim Charles' family was present at one of the concerts and Tim's mother complained of heart palpitations. Actually, to everyone's relief, she was feeling the impact of the sound of Ian Wallace's bass drum as augmented by the sub-woofers.)

The main house speakers including the sub-woofers are fed by Altec model 9440 800-watt dual channel power amps. These amps receive their input signals from Stanal Custom crossovers equipped with De Coursey crossover cards. The crossover frequencies are more or less fixed at 800 Hz and 6 kHz, but can be changed on the cards.

The nerve center of the house system is a brand new Yamaha PM-2000 "Proto 2" not yet generally available. This board has several practical features, one of the more interesting being the calibration of the input pads in terms of the output level of the source. That is, the pad controls are continuously variable and the calibration markings around them run clockwise from +4 dBm to −60 dBm. This wide range of acceptable input levels combines the functions of input padding and mic/line switching.

Each input on the board has phase reversal as well as phantom microphone power available by pushbutton selection. The equalizers for each input are 3-range (low, mid, hi) detented types. Each equalizer also contains low (50 Hz) and high (15 kHz) cut-off filter sections.

The Yamaha PM-2000 is a 24-input console with a multiplicity of possible output configurations. Tim Charles sends a "stereo" feed from the console but the mix is mono. Helping out on the input side of the console are four Yamaha PM-180 4x1 sound reinforcement mixers. These auxiliary mixers are necessary to expand the total num-



Close-up view of the "flying P.A." speakers being hoisted on cable network.

ber of inputs available to the required *thirty-five*.

Sharing the rack with the PM-180s are four dbx Model 160 compressor limiters, an Orban 1118 dual reverberation unit, a dbx "Boom Box" and two Stanal custom graphic equalizers.

Another Yamaha mixer, this one a PM-1000, is used by stage monitor mixer Chris Coffin to feed *eleven different mixes* to the performers on stage. So complex is the job of juggling monitor feeds that Chris relies on keyboard player Alan Pasqua to achieve a balance (for monitoring purposes only) among his three electric keyboards—Clavinet, Fender Rhodes, CS-80 (Yamaha synthesizer). This mix is sent to Chris, who in turn sends it back out through the monitoring system. The PM-1000 has multiple-grouping outputs very similar to those available on the PM-2000. Since he requires eleven differently mixed feeds, Chris takes greater advantage of the flexibility of the PM-1000 than Tim does of that of the PM-2000.

Stage monitor speakers are mostly S-4 cabinets housing two Altec-417s each for low end, and an Altec 288 wide angle radiator for the remainder of the spectrum. In addition to these combinations there are Altec-Lansing 1221 systems, and SS-1 (Stanal Sound) monitors.

The monitor speakers are powered by a very new type of power-amplifier system recently developed and introduced by Altec. Altec calls it an "incremental power system." Basically these systems (of which four were in use) consist of a rack-mounted holder containing up to twelve modules, eight of which are individual 75-watt power amplifiers. Of the four remaining module slots, one is taken up by a power supply module, and the others can be occupied by a combination of balanced input modules, driver modules, and crossover modules. These modules (the power amplifiers, drivers and crossovers) can be interconnected in a variety of ways. First, the crossover cards are switch-selectable either 2-way or 3-way, with crossovers available at 625 Hz, 800 Hz, 1250 Hz, 1600 Hz, 3150 Hz, 4000 Hz, 5000 Hz and 8000 Hz. These crossover cards have output stages that contain drivers that are suitable for driving power amp modules directly. However, if required, driver cards with either two or four outputs are available. By means of

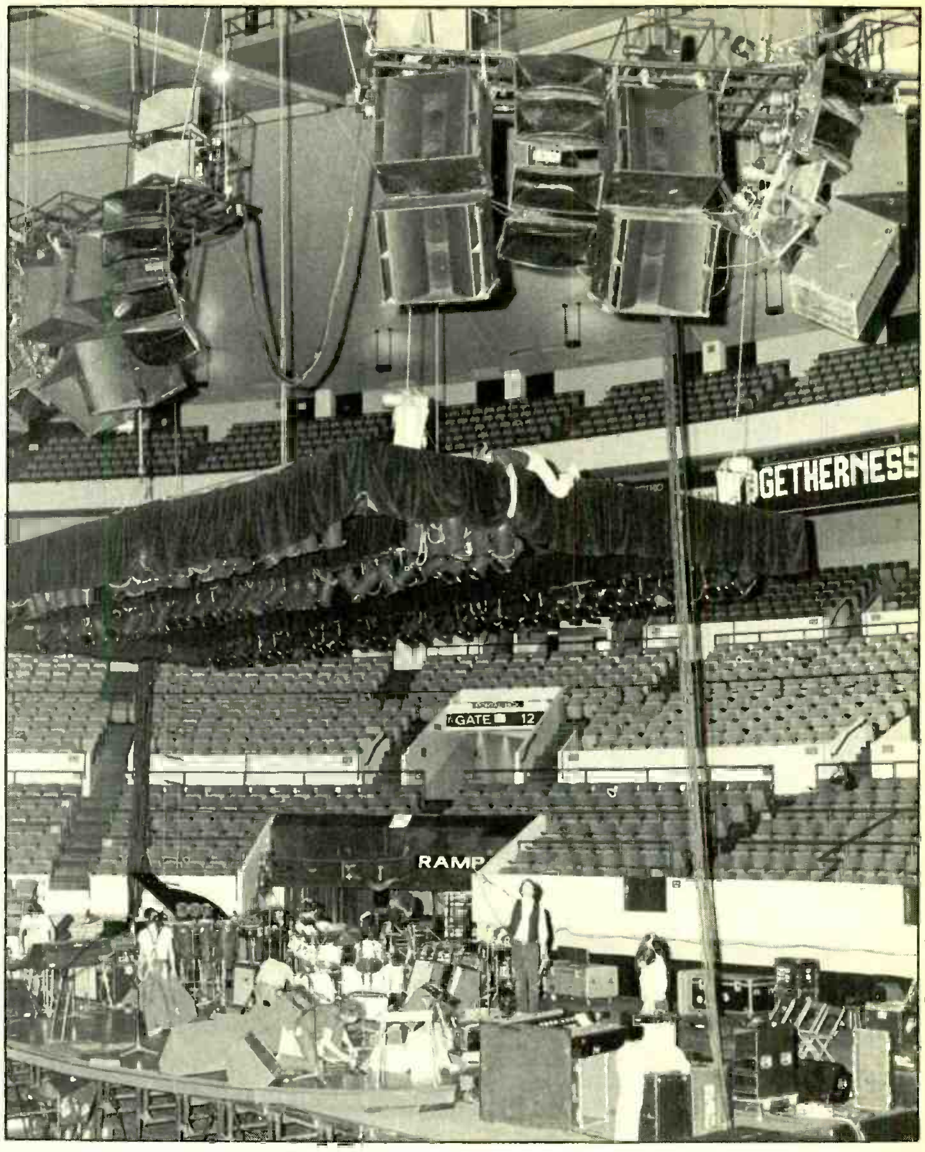


Photo by Doug Hanewinckel

Sound crew setting up stage area for concert at Madison Square Garden.

miniature switches on the various module cards, any combination of individual power amp modules may be driven by any desired source. For example, a crossover card in the two-way configuration may feed two groups of four power amp modules.

Each group of four power amp modules can feed either four independent loads, per group, or the outputs from four can be paralleled together to provide 300 watts of power amplification to a single load. Or, the groups of four can be broken up into pairs of parallel amps. Any desired combination of amplifiers can be achieved from eight single, independent 75-watt feeds, to a 300-watt-per-section bi-amp. Each rack of input, crossover and power amp modules has the sort of interconnecting flexibility that is usually associated with consoles, with the additional convenience of outputs that may be connected in parallel.

(As an added note, to point out just how new this Incremented Power System is, the literature from Altec accompanying the systems shipped to Stanal have the words "temporary operating instructions.")

While dealing with the subject of stage monitoring, it should be mentioned that checking out and adjusting the stage monitor system took up most of the time devoted to sound checks. The reason for this is the previously mentioned large number (11) of different, independent mixes that the monitor system must handle.

As for the sound checks themselves, all of the band members, including Bob Dylan participated, and the sound checks were not considered finished until everybody's requirements were

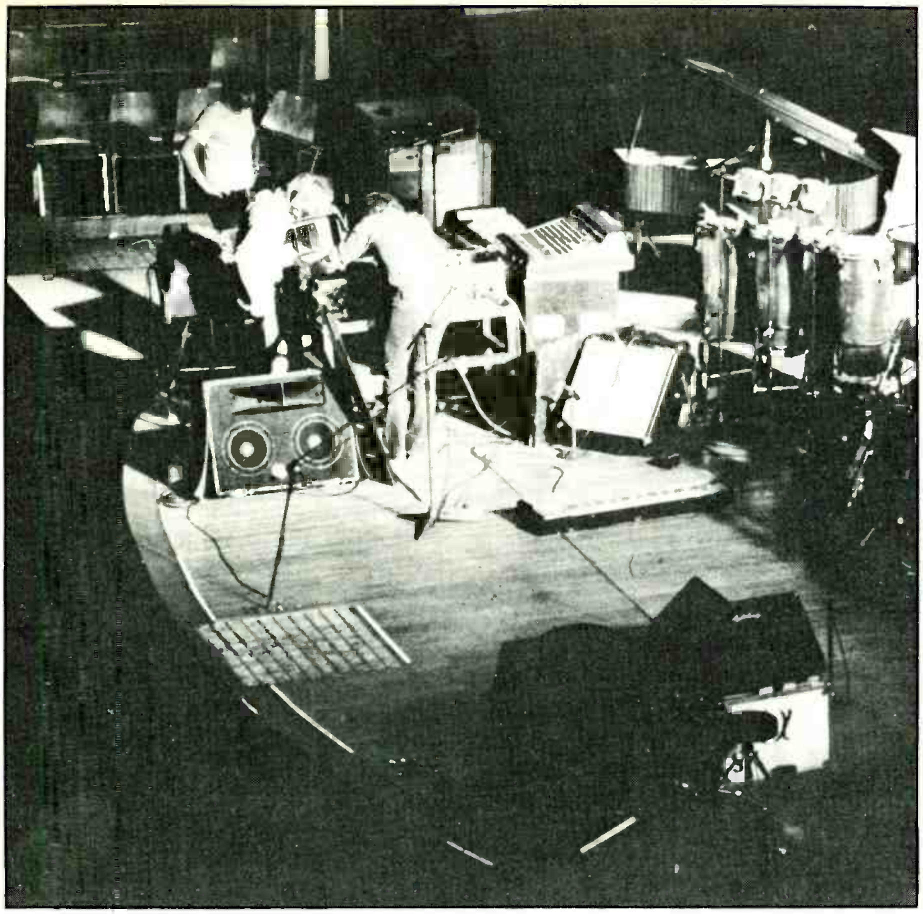


Photo by Stan Miller

The Stanal Sound crew putting finishing touches on the set-up.

met, both on stage and out in the house. During the final ten minutes or so of each pre-concert sound check, Tim Charles, having pre-tuned the house system using pink noise and a real-time analyzer, and, having gotten a mix that was acceptable as heard from the house mixing location, made a walking tour of the concert location with a hand-held SPL meter in order to detect any "dead" spots.

EQUIPMENT REFERENCE

604-HPLN—15″ speaker with coaxial horn and special Altec "Tangerine" phasing plug

421-8LF—15″ low-frequency loudspeaker

817—Cabinet—combination horn/loaded bass-reflex enclosure for two 15″ lf loudspeakers

1650—One-third octave equalizer

417-8LF—12″ lf loudspeaker

288-16G—Mid/hf driver

1221—Slope-type monitor system

Covering this assignment provided an opportunity to observe how a sound reinforcement system operator handles different locations with the same equipment. The first concert observed was at the Nassau Coliseum in Uniondale, N.Y. and the second was at Madison Square Garden in New York City.

During the sound check at the Coliseum, Tim Charles commented on the very bright reverberation characteristics of that location. Even after pink-noise/RTA pre-tuning, the actual mix as heard from the house mixer location seemed pretty shrill. The bottom end was fine, but the midrange and high frequency portions were dominant beyond acceptability. Tim was able to turn all of the short-throw horns *off*, and still get enough of the upper half of the spectrum. He decided to leave the short-throws on, feeling that the addition of a few thousand sound absorbing bodies would help smooth things out. Unfortunately, although there were plenty of bodies that night at the concert performance, the audible characteristics of the house mix grew worse (at first). Not only did the objectionable shrillness remain, but the bottom end seemed to disappear, despite the sub-woofers. These problems were corrected after the first two numbers, but they did cause Tim Charles to suffer several anxious minutes of readjustments.

At "The Garden," it was apparent at the sound check that the difficulties encountered at the Coliseum would not occur. This impression was borne out as correct during the concert. The kind of balance that a performer like Bob Dylan requires is different from that of a typical (if there is such a thing) rock situation. Since much of the appeal of Dylan's work is in lyric content and his unique vocal style, the reinforcement mixes tend to be very vocal-heavy, with an emphasis on clarity and articulation. Such was the case at both the Garden and Coliseum concerts, but at the Garden the lead vocal was clear and strong and well supported by the band and back-up vocals. The mix at the Coliseum contained plenty of vocal, but the band seemed to have been overpowered by it. Tim Charles was philosophical about the difficulties encountered at the Coliseum, saying that each location was unique, presented its own set of problems, and that a professional crew, equipped with flexible sound reinforcement tools should be able to overcome or at least minimize any difficulties.

From Levi's to Sassoon

Getting back to personal opinions and viewpoints, as I watched and listened to both concerts, and took in the "new look" that Dylan's concert presentation has taken on, I was once again struck by the superficiality of most of the complaints about the flashiness or slickness of the presentation. If our readers will permit a bit of amateur philosophising let me point out that Dylan's original fans and followers—during the last 15 years—have been a-changing some themselves. Levi's and workshirts have given way to wool worsted and Sassoon; cross-country rides now usually terminate at Las Vegas; Woody Allen is more revered today than Woodie Guthrie.

I personally see and hear little difference between present and past Dylan, at least at the core. The changes that have been made, other than true artistic growth, are no more significant than the art on the back side of an LP cover.

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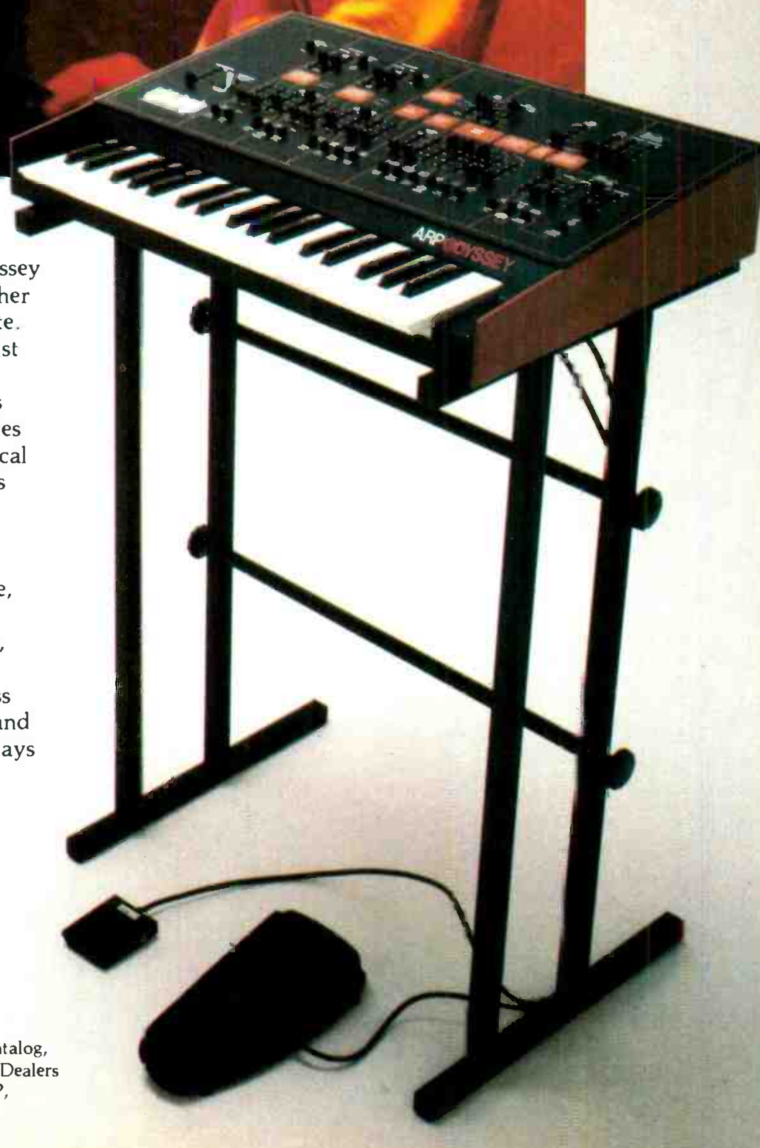

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DECIPHERING MIXER SPECS

BY BRIAN ROTH

Anyone wading through the massive stacks of mixer literature is certain to become confused by all the varieties of specifications. Some things, like the number of inputs or physical dimensions, are obvious. The electrical performance specs, however, are a different story.

This article is intended to shed some light on the technical mumbo-jumbo, particularly in the three key areas of frequency response, distortion and noise. While aimed at mixer specifications, this discussion will also apply to many other types of electronic audio equipment.

We will attempt to simplify the technicalities of these specifications so they may be more easily understood by the average purchaser of a mixer. However, many of the principles will require careful reading. So without further verbiage, here we go.

Frequency Response

The average human ear is capable of detecting audio frequencies over a range of 20 Hertz (Hz) to 20,000 Hz (which is the same as 20 kHz). If full signal fidelity is to be maintained, the mixer must be capable of passing this frequency band. Because of this, just about every spec sheet quotes, "Frequency Response — 20 Hz-20 kHz."

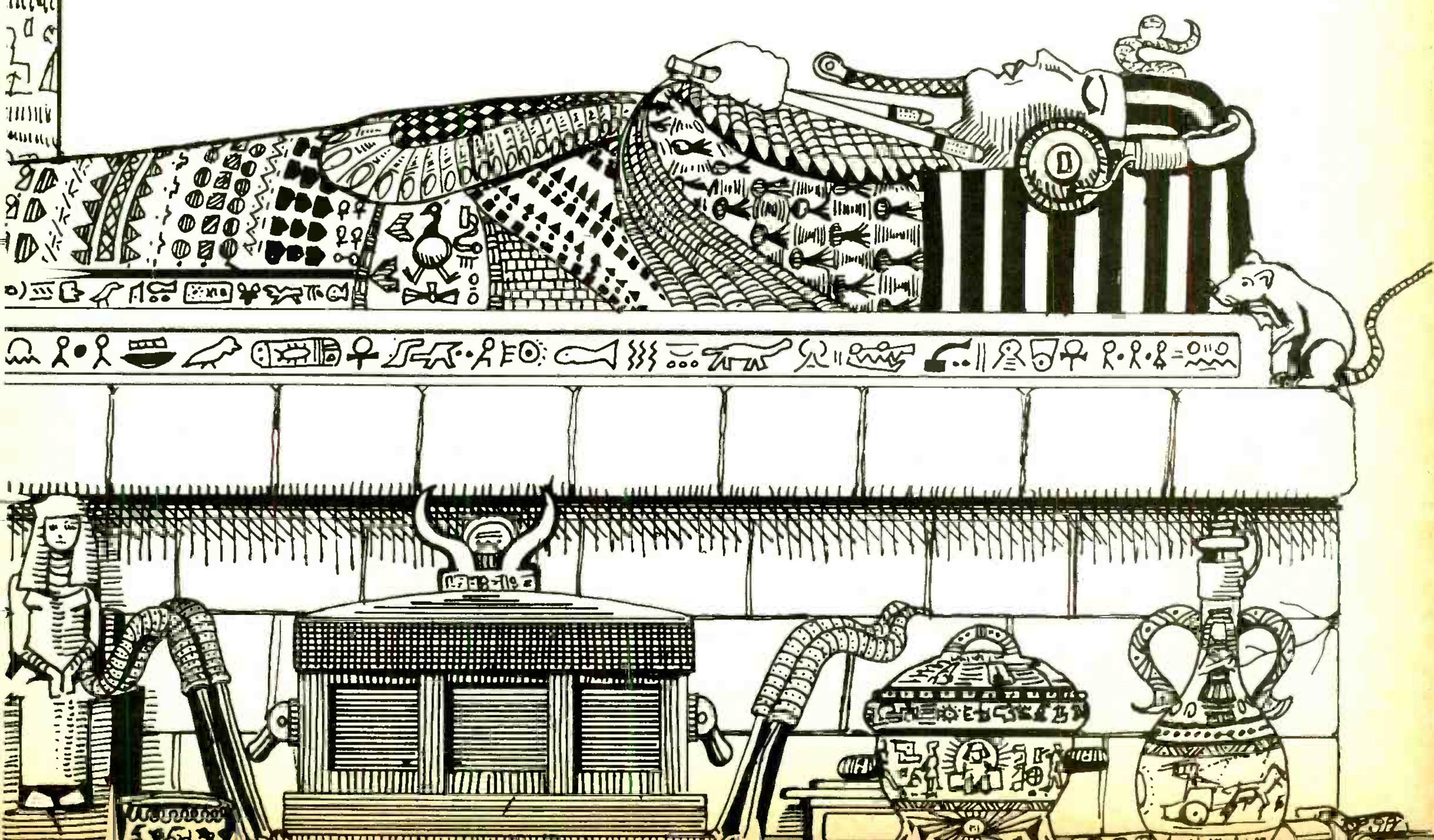
Now if that is the entire extent of the response spec, you had better watch out. To be more than just a random collection of numbers, the spec needs to be completed with a tolerance. This might be expressed as: "20 Hz-20 kHz, ±2 dB." This states that all frequencies over the band of 20 Hz to 20 kHz will be amplified by the same amount *within* a four decibel tolerance (plus *or* minus 2 dB). Thus, the bass portion of an audio input signal receives the same amount of amplification as the midrange or treble (within that ±2 decibel level variance). No part of the frequency range will be significantly emphasized at the expense of the rest.

Even if two mixers are quoted as having an identical frequency response, it is still possible for one mixer to sound differently from another. This can sometimes be attributed to the overall *flatness* of the frequency response, particularly in the critical midrange. Unfortunately, a written response spec cannot express how flat the response truly is. For this reason, frequency response graphs are widely used.

Referring to Figure 1 and 1a, here is the response of two mixers each having the same quoted spec. Both of them fall within the allowed range of ±2 dB, but mixer A goes through some extra gyrations that mixer B doesn't. Since these aberrations are located in the midrange, they would probably be audible, particularly in a properly set up A versus B comparison.

These graphs demonstrate that a picture can be worth 1000 specs. However, even a graph can be misleading. For example, Figure 2 shows a rather useless graph. The decibel scale is much too wide to allow an accurate examination of the response. The line could squiggle 5 dB without being overly obvious. When comparing response graphs, always check the decibel scales on the vertical axes to be sure the graphs are comparing apples to apples.

Of course, all of this assumes that the printed graphs are honest, and most probably are. Sometimes, the factory's art department will "beautify" a somewhat lumpy curve, making mountains into molehills. This seems to be the case more with loudspeakers than with mixers (remember that when compared with electronic gear, loudspeakers will generally have a much more uneven frequency response).



An often overlooked factor when determining the frequency response of a transformer coupled microphone input is the impedance of the signal source used to conduct the measurement. Figure 3 shows how a signal generator is substituted for the normally used microphone. If care is not taken, the test set-up in 3a will give confusing results.

The reason for this is due to the fact that the input transformer in the microphone preamp will yield different frequency responses depending upon the output impedance of the signal generator. This is an inherent characteristic of input transformers. Figure 4 shows a typical case of this. Notice that with a 600-ohm-source impedance, the high end rolls off rather rapidly, while with a 50 ohm the response is wider, but with a peak just before the roll off point. A 200-ohm source looks much better overall.

Consequently, it is important to know what the signal generator's output (or source) impedance is, regardless of whether you are examining a published response graph or conducting your own tests. 150- and 200-ohm-source impedances are often used since these represent the impedance of an average microphone. (There can be a considerable variation of microphone impedances; American mics tend to be on the low side, like 75 or 100 ohms, while European mics can range up to 250 ohms or higher.)

It is unfortunate that many response specs don't state the signal generator's source impedance. Of course, sometimes this may not be necessary as in the case of an "active" transformerless input which generally will exhibit excellent response stability, even over the 50-ohm to 250-ohm range.

Since a transformer coupled input will almost always yield a poorer frequency response, be sure that the spec or graph is for the microphone input and not for a transformerless line input which will usually have a flat and extended response.

While on the subject of extended frequency response, let's examine some current trends. It is not uncommon to see a piece of audio electronic equipment (such as amplifiers or mixers) with a response much greater than 20 Hz-20 kHz. Some specs really go out there, like 50 or 100 kHz on the

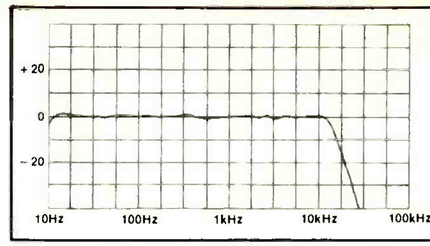


Fig. 2: This frequency response graph is hard to interpret due to the extremely wide dB scale.

top end and maybe down to 5 or 10 Hz on the bottom. What's the use of this if we only hear from 20 Hz to 20 kHz?

Probably foremost is "specmanship," and trying to be one kHz up on the guy down the block. However, there also are legitimate reasons why this can be desirable.

If the frequency response extends way beyond 20 Hz or 20 kHz, it is generally true that the response over the audible range will be much flatter when compared with a response that rolls off at the extremes. Figure 5 demonstrates this. Mixer A has much fewer response variations over the 20 Hz-20 kHz region than mixer B, and consequently would be more sonically accurate (all other things being equal, of course).

It can be argued, of course, that the variations in response of mixer B wouldn't be particularly audible since the human ear is least sensitive at the frequency extremes. In many applications, such as "live" P.A., this can be quite true since so many other factors (loudspeakers, microphones, room characteristics, etc.) will introduce their own roll-offs. In fact, a response roll-off may be desirable to eliminate extraneous noise pick-up, such as low-frequency stage rumble. It is not uncommon to see low- or high-cut filters used in a P.A. system. In these situations, extended frequency response is rather useless.

In critical situations, like high-quality recording, the wide response can be desirable, and not just because of frequency response flatness over the 20 Hz-20 kHz region. In general, whenever a piece of audio equipment exhibits a frequency response roll-off, it is accompanied by a phase shift. This really isn't the same kind of phase shift as those "whoosh-o-matic" effects boxes make, but it is a distant cousin.

Passing Phase

This phase shift causes some frequencies to be slightly delayed in

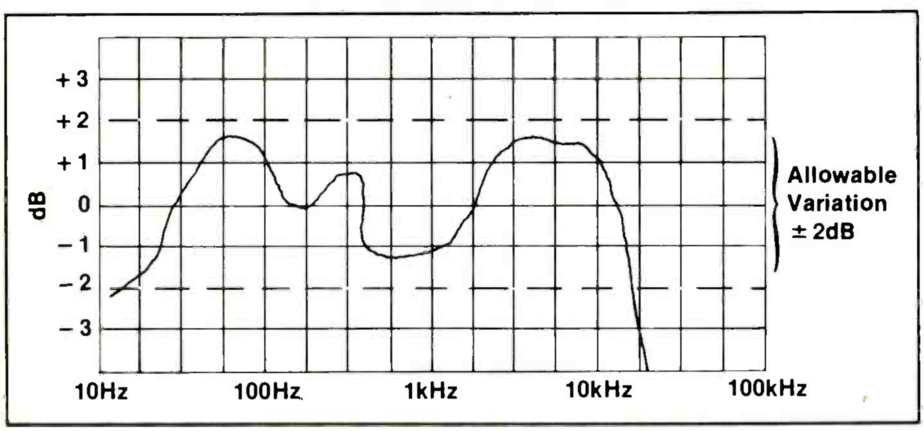


Fig. 1: Frequency response of "Mixer A—20 Hz-20 kHz ± 2 dB."

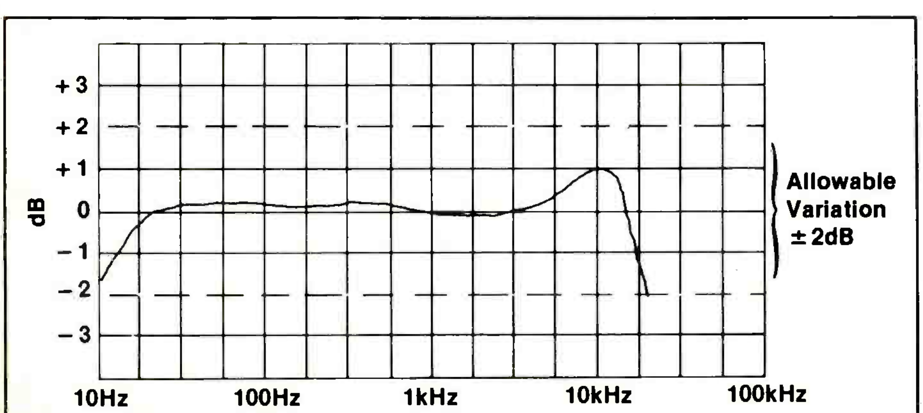


Fig. 1a: Frequency response of "Mixer B—20 Hz-20 kHz ± 2 dB."

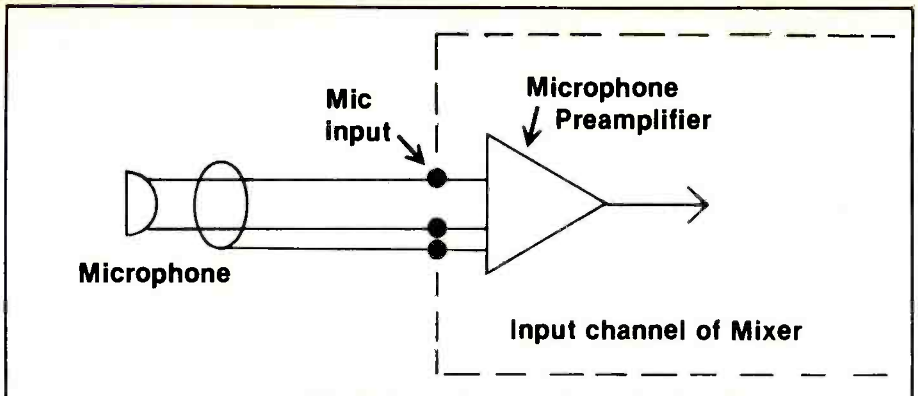


Fig. 3: Normal connections of microphone to mixer.

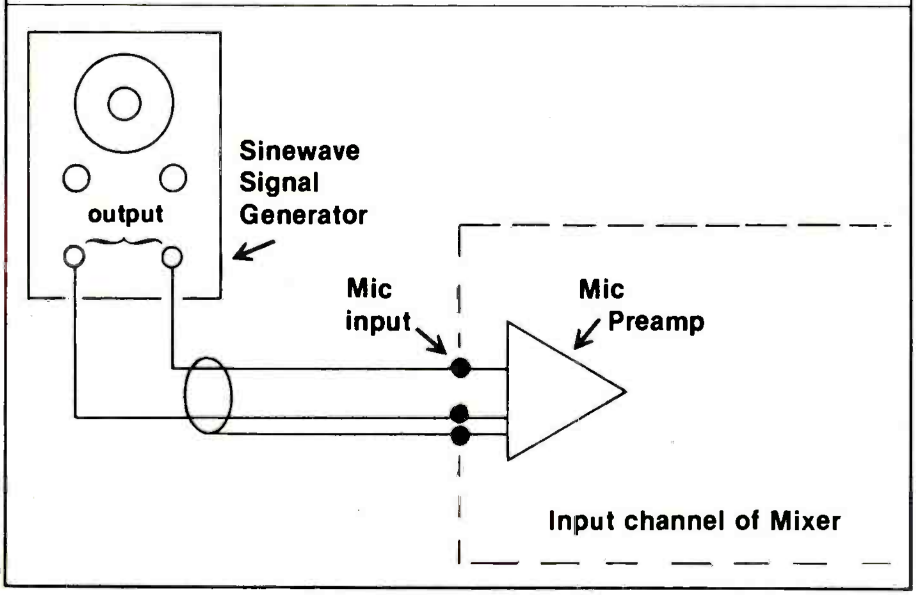


Fig. 3a: Sinewave oscillator is substituted for microphone to allow frequency response measurements.

time when passing through the mixer. Thus, with complex program material, the high-frequency harmonics will leave the mixer a split second after the fundamental tone. Phase shift is measured in degrees with 0 degree phase shift representing no time delay and 180-degree phase shift corresponding to complete reversal, or an out-of-phase condition.

The degree of audibility is subject to disagreement. On one hand, we have the "golden ears" types who desire frequency response out to blue light so that phase shift is non-existent, while on the other hand some noted scientists have conducted tests that indicate phase shift is essentially inaudible.

So what's the answer? Perhaps it's setting a reasonable phase shift level of 10 or 20 degrees at 20 kHz. If a mixer's response is essentially flat out to around 50 kHz (1 dB down) then phase shift will generally be acceptable at 20 kHz. Extending the bass end down to 10 Hz or lower will also eliminate excessive phase shift at 20 Hz.

Phase shift specifications are seldom given for mixers (although they are often seen in power amp spec sheets). Part of the reason may stem from the rather lousy phase shift characteristics that many mixers possess. One well-known brand of studio console that costs around $100,000 was found to have a 45-degree shift at 10 kHz!

Let it be made clear that the frequency response and phase shift characteristics should be examined in the light of the intended application. 40 Hz to 15,000 Hz ±3 dB can be perfectly acceptable in even a high-quality P.A. System. 20 Hz to 20 kHz ±1 dB may cause audible phase shift delays in a critical recording situation utilizing high-quality condenser mics and loudspeakers.

Here's a couple of final notes before moving on to other areas. It is difficult to extend the low-frequency response with transformer coupled inputs, and thus a response of −1 dB at 20 Hz is very respectable with this type design. Transformerless inputs can go down to DC (0 Hz) in some cases. This is not to say that transformerless inputs are the total answer since they *can* sometimes be noisier than a transformer coupled preamp, or more subject to outside interference from CB, electrical noise, etc. It all depends on the quality of the preamplifier design. Current trends are to eliminate all transformers in the audio path since transformers tend to be expensive and response limited. However, in certain situations, a transformerless input runs into real problems, particularly in P.A. systems where the transformer is still king. A high-quality transformer can have amazingly good fidelity and will have good input noise immunity.

Distortions of Many Flavors

Next on our agenda is distortion. An uneven frequency response can be called a type of distortion, but what we are referring to here is the addition of

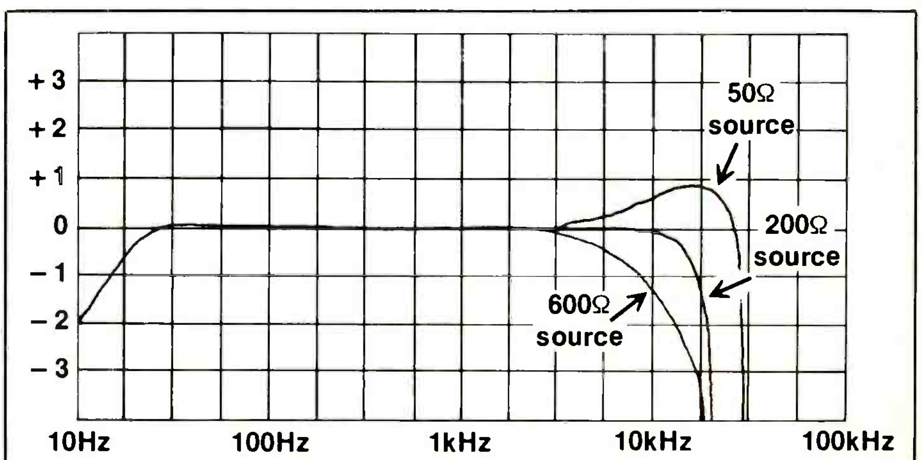


Fig. 4: Frequency response variations of transformer coupled microphone input due to impedance of input source.

extraneous frequencies to an audio signal.

For instance, assume that a pure 1 kHz tone is being applied to the input of the mixer. Upon careful examination of the mixer's output signal, we find not only 1 kHz, but also a small amount of 2 kHz and 3 kHz signals. These undesirable multiples of the original frequency definitely weren't present in the test tone. The mixer is thus guilty of generating harmonic distortion (and they *all* do to some extent).

For this reason, most mixers' spec sheets quote a harmonic distortion figure, almost always expressed as a percentage. In a typical spec of "Total Harmonic Distortion — Less than .5%," it is stated that the level of all the undesired harmonics will add up to being less than .5% of the total signal.

However, simply stating a percentage of THD (Total Harmonic Distortion) is far from being a complete and useful spec. In many cases, the THD is measured at 1 kHz, where the results tend to be much better than at 20 Hz or 20 kHz. Unless the specification indicates that the mixer is capable of meeting the stated percentage of THD over the entire frequency range, take the numbers with a pound of salt.

A Total Harmonic Distortion measurement implies, by its very name, that all the harmonics of the signal are being measured and "totaled." This allows a standardized measuring technique, but unfortunately it does not necessarily correspond to what is heard during a listening test. The reason for this lies in the fact that the measurement method responds to all of the harmonics equally while the ear is sensitive to the higher numbered harmonics. A piece of gear with lots of second and third harmonic distortion can sound "big and fat" (like a tube-type guitar amp). On the other hand, even a small amount of fifth or seventh harmonics can give the sound a harsh texture since the odd numbered harmonics are the least "musical."

Thus, it is useful to know what harmonics are being produced and in what proportions. This information can be given in the form of a spectrum analysis graph (*See Figure 6*) which allows us to see the relative levels of all the harmonics.

Another factor that will affect the THD characteristic is the input and output signal levels of the mixer being tested. High input levels will almost invariably result in higher distortion levels due to the limitations of the mic preamp (and in the case of transformer coupled inputs, the increase in THD will be most acute in the low-frequency region due to the saturation of the transformer's core).

Triggering SR

High input or output levels can also trigger another type of distortion called *slew rate limiting*. This type of distortion is generated by the electronic circuitry's inability to handle high signal levels at high frequencies. Slew rate limiting can be noted as a large rise in harmonic distortion when strong input or output levels are used at the high end of the audio band (10 to 20 kHz). If the signal is monitored on an oscilloscope, the observer will note that the test oscillator's sine wave signal turns into somewhat of a triangle-shaped waveform.

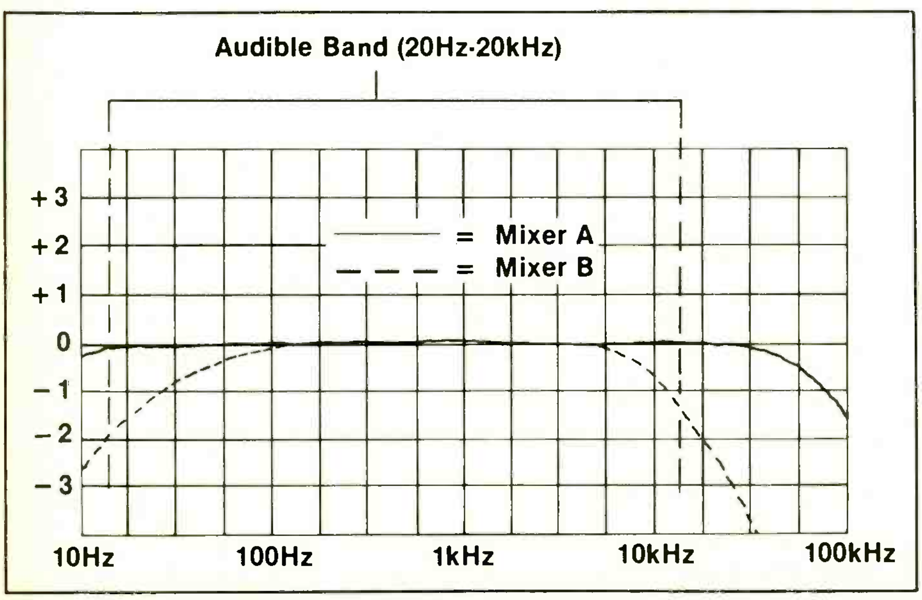


Fig. 5: Comparison of two mixers; notice Mixer A has much fewer response variations over the audible band.

Slew rate limiting can occur in any equipment, but some suffer much more than others. It is a very obnoxious form of "gritty" distortion and can be easily generated in some mixers with the right combination of input/output levels and a "fast" signal with lots of high end content.

The slew rate (SR) of a piece of equipment is expressed in volts per microsecond, or abbreviated, V/μs. A figure of 5 to 10 V/μs is generally adequate for most all mixer applications. Lower SR figures can be tolerated in certain applications (rock & roll P.A.) since the higher levels of distortion can be masked by fuzz guitars or overloaded power amps. The perfectionist will seek the highest SR numbers, which is O.K. as long as other things — like a signal to noise or circuit stability — aren't lost in the quest for a high slew rate.

Again, slew rate figures are seldom given for mixers. The harmonic distortion specs can help *if* a THD measurement is given for 20 kHz at high output levels (like 15 or 20 dB above 0 VU) and high input levels (such as a couple of dB below preamp clip). When a mixer can handle 20 kHz at the same input or output levels as at 1 kHz (and assuming the THD remains low at 20 kHz at these elevated levels), slew rate limiting probably won't be a problem.

If slew rate is an important sonic factor, why isn't more attention given to it? In low cost mixers, price is the big problem since a high slew rate requires premium circuitry. Fortunately, integrated circuits (I.C.s) such as used in most mixers have improved drastically, so perhaps high levels of slewing induced distortion will become a thing of the past.

Testing IM

Another technique of measuring distortion utilizes two tones rather than just one as in harmonic distortion measurements. The idea is that music is more than just one single frequency, so the test signal needs to be more complex. This type of distortion test measures Intermodulation Distortion (IM), which is often significantly higher than harmonic distortion.

Imagine if all clubs were built for live music; that clubowners spent as much on sound systems as they do on decor; and all you had to do was set-up and play. Well, forget it. There is only one Hollywood Bowl and chances are it's not your next gig. More likely, the acoustics at your next room will be just as bad as the last, maybe worse. More likely, the next clubowner's "vocal smasher" is older than the last one, and as usual it will be you and your group that suffers. All too familiar? Well relax. Acoustic, with over a decade of live music experience, is introducing an exciting new line of Sound Re-enforcement products, designed for turning problems into opportunities. Quiet, versatile mixers with low distortion amps built-in for fast, easy set-ups. Features like dual-sensing overload indicators, 9-band graphic equalizers, built-in reverb and light bar output displays. Rack mountable power amps that boast fan cooling, and extensive circuit safeguards. Even the compact solid-plywood speaker systems include a driver protection circuit that will handle power overloads without program interruption. Acoustic has carefully matched these components to perform in the most adverse conditions, and continues to offer the exclusive Lifetime Protection Plan. So why suffer through another night of feedback and blown horns? Don't expect "good acoustics," take them with you.

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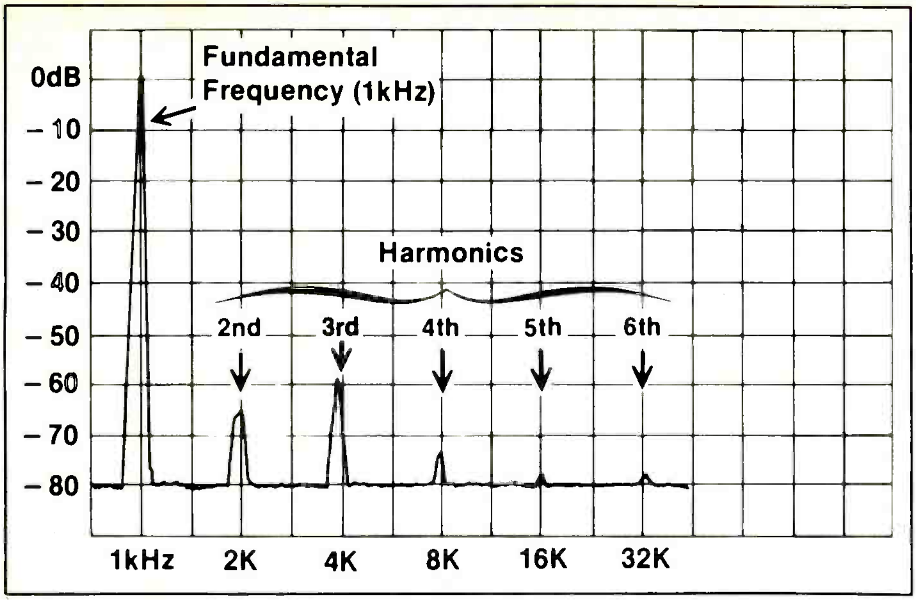

Fig. 6: Spectrum analysis of output of mixer shows relative levels of harmonics due to distortion.

Intermodulation distortion will result in the generation of spurious frequencies that are *not* harmonic multiples of the original signals. For instance, if 2 kHz and 3 kHz are mixed, the intermodulation products would be 1 kHz and 5 kHz (the sum and difference of the two original signals). By comparison, the spurious products caused by harmonic distortion are frequency multiples and can be musically related to the original signal; for example, the second harmonic of a signal is one octave higher than the fundamental and it can even be considered as an "enhancing" effect. Intermodulation distortion products are not musically related to the desired audio and can cause aural migraines.

What is probably the oldest standard intermodulation technique is the SMPTE (Society of Motion Picture and Television Engineers) method. This test uses frequencies of 60 Hz and 7000 Hz mixed 4:1. It is useful in determining how much distortion a low-frequency signal will cause to a high frequency signal. An extreme example of this can be created by cranking up the volume on a cheap audio tape deck until the bass guitar causes the vocals to gurgle. This would be a gross and extreme case of IM.

Modern circuit designs have reduced the levels of SMPTE IM to very low levels, so this test tends to be less informative than it used to be with older tube-type equipment.

No rule says that IM measurements must be made with frequencies of 60 Hz and 7000 Hz, so several other techniques of two-tone testing have evolved. Nearly all of these are intended to examine the high frequency IM characteristics of the equipment being evaluated. Frequencies of 10 kHz and 11 kHz mixed 1:1 are often used, while other tests use a pair of swept frequencies with a constant frequency difference. All of these are useful in detecting any "nasties" caused by a combination of high-frequency signals such as would be found in real music.

Fashionable TIM

Yet another distortion test has come into vogue. It is referred to as Transient Intermodulation (TIM). Like the previously discussed IM test tones, TIM measurements use two signals. However, one is a square wave and the other is a higher frequency sine wave. The composite signal looks like a Figure 7. The square wave causes rapid shifts in signal voltage similar to the instantaneous variations found in dynamic music.

TIM measurements are very seldom found in spec sheets due to their very recent development and the lack of

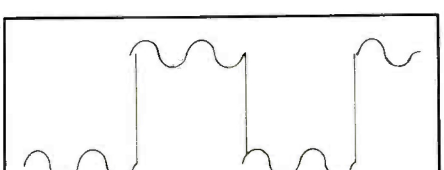

Fig. 7: Representative waveform of signal used for transient intermodulation distortion (TIM) test.

suitable test gear. Some authorities have also stated that a proper collection of high frequency THD and IM measurements will give an equally informative view of a piece of equipment's electrical performance. So, it remains to be seen whether or not TIM becomes a regular member of the spec sheet cast.

Having discussed the most popular distortion measurement methods, here is a hopefully realistic view of the entire scene. Everyone wants their equipment to have a distortion spec of .00000001%, but it appears there is a point of diminishing returns. A THD or IM percentage of .05% or less over the entire frequency and signal level range will satisfy the vast majority of ears, regardless of application. Higher levels of THD at low frequencies due to input transformers can be tolerated since it is generally lower ordered distortion and not as audibly apparent. The purists will cry that these levels of distortion are too high and will go for that last ounce of performance: "If you can measure it, it ain't good enough." Oh well, that's what fidelity is all about!

Hum, Hiss and Other Noises

It is an unfortunate fact of life that all audio equipment must generate noise. As the temperature of any substance rises from absolute zero, the random activity of all the little atoms becomes more frenzied. Consequently, there is also electron movement in the substance. In audio equipment, this movement of the electrons generates a noise signal of random nature that the ear calls hiss.

No matter how hard we try, noise levels cannot go below a certain level determined by several factors. First, the temperature of the electronic equipment must be considered. Sophisticated scientific instruments are sometimes cooled in a liquified gas to lower the noise floors. In audio, this is rather impractical since not only the mixer, but the tape recorders and even the mics must be cooled to make an overall reduction of hiss. Yes, that's right, don't forget that a microphone, even in a perfectly silent room, must generate a noise output if its temperature is above absolute zero.

Now before you buy a walk-in cooler from a bankrupt tavern, you should know the following fact. If it is capable of being inhabited by humans, it's not cool enough to make a significant difference in noise levels. So let's examine

the other two factors that play the important roles in noise performance.

The impedance or resistance of the electronic circuits will have a big effect. The lower the impedance, the lower the noise. A zero ohm impedance theoretically will have no noise.

So what does this mean in the real world? Well, start with the microphone. If it has an impedance greater than 0 ohms (they all will except for that one you dropped last Saturday), it *must* generate a certain amount of noise regardless of any acoustic noise. The higher the impedance, the higher the internally generated hiss level.

This could be interpreted as stating that high impedance mics are inherently noisier strictly due to their impedance. While the self-generated noise from the microphone is greater, so is the level of the desired audio. Thus, the ratio of the signal-to-noise levels remains basically unchanged between low and high impedance mics. Of course, the low impedance mics have their virtues like the ability for long cable runs, etc., etc., which makes them the choice in professional applications.

We'll come back to the effect of impedance after examining the third factor that determines the hiss level that will exist in audio equipment.

If a person were to listen to hiss reproduced via a high quality loudspeaker, he would notice a marked difference in the perceived hiss volume if a high and/or low cut filter were switched into the audio system. This happens because the hiss, called white noise by the scientific types, consists of all frequencies from sub-audible to supersonic. Thus, if we reduce the frequency response of the system, we are chopping out some of the noise.

Bless the Calculator

This third contributing factor is referred to as bandwidth. The wider the bandwidth, the higher the noise level. This is one reason why telephone lines are limited to a 200 Hz to 2 kHz response since all those miles of cable have enough of a noise pickup problem by themselves. A telephone with 20 kHz response would probably bury the conversation under a ton of hiss.

Years ago, the three factors of noise generation were put together into a little formula that would give the noise voltage created with a particular circuit. It is: Noise voltage = $\sqrt{4KTBR}$ (equation 1).

The T stands for the absolute temperature in degrees Kelvin, B is for the bandwidth in Hertz, and R represents the resistance of the circuit.

The 4K part makes the answer come out right (trust me). K stands for Boltzman's constant, a mathematical unit which approximately equals 1.38×10^{-23}.

Let's take some real numbers and slug them into the equation to see what happens. Assuming a 200-ohm microphone (and neglecting that the impedance of the microphone will vary over the frequency range), a bandwidth of 20,000 Hz and a typical room temperature of 295° Kelvin, we enter everything into our calculator.

$$\text{Voltage} = \sqrt{4\times1.38\times10^{-23}\times295\times20{,}000\times200}$$

$$= 2.5522\times10^{-7}\text{ volts}$$

$$= .2552\text{ microvolts}$$

(Thank Goodness for scientific calculators!)

The microphone with a 200 ohm output impedance *must* generate at least this much noise. A real microphone will have an actual noise level several dB higher since it's mighty darned hard to make something absolutely perfect. The random voltage signal will be amplified along with the desired audio signal when processed through the mixer.

The voltage amplification factor of a mixer is expressed in decibels. When a signal voltage is doubled, this would be a 6 dB gain. An amplification of 10 times is 20 dB, 100 times is 40 dB, 1000 times is 60 dB, 10,000 times is 80 dB, and so forth.

The formula for determining voltage gain in dB is: dB gain=20×common Log of voltage gain. It's all quite elementary if you have a logarithm key on your calculator!

If the previously discussed 200-ohm microphone were to be connected into a mixer set for 60 dB gain, we would find that the output noise from the mixer (assuming that it added absolutely no noise) would be 255.22 microvolts since a 60 dB gain will multiply the signal level by 1000. The noise level at the mixer's output cannot be lower than that figure with 60 dB of gain and a 200 ohm source.

Now let's assume that the mixer does add some noise and causes the output level to increase by 3 dB over what the calculations project. This mixer would be said to have a "Noise Figure" (sometimes abbreviated as NF) of 3 dB.

A check of various mixers' spec sheets will reveal that noise figure is seldom used, which is unfortunate since NF gives us an exact value for the amount of noise added to the signal by the mixer's electronics.

What is usually seen is equivalent input noise (EIN) with figures like "−126 dBm." It appears that the popularity of this kind of spec might be due to the fact that the numbers are larger than with NF type specs. EIN requires more interpretation than NF, and is more subject to manipulation by utilizing sly measurement methods.

Before proceeding, let's define that ubiquitous unit; the dBm. It is based on a standard reference created years ago for telephone work and corresponds to 1 milliwatt of power being fed into a 600-ohm load. It happens to work out that an audio signal of .775 volts RMS across a 600-ohm resistor will result in 1 milliwatt of power being dissipated by the resistor.

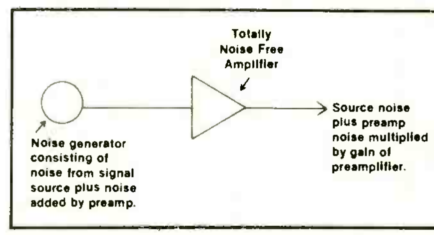

Fig. 8: Idea behind equivalent input noise (EIN) calculations.

Consequently, if the voltage across the 600-ohm resistor is increased by ten times to 7.75 volts RMS, the signal has increased by 20 dB (referring back to the dB equation). The power level would be ±20 dBm. 0.0075 volts across 600 ohms would be ten times less than the reference, or −20 dBm.

Now let's see how all of this relates to EIN specs. Figure 8 gives the premise for this measurement. Once again, use your imagination and pretend that a noiseless microphone preamp actually exists. Connected to the input of this little wonder is a noise generator which represents the random voltage created by the signal source (microphone) *plus* the noise added by the audio circuitry.

If the output level of the noise generator is −126 dBm and it is boosted by 60 dB in the preamplifier, simple math will reveal that the noise output from the preamp will be −66 dBm (−126+60=−66). In other words, under these conditions, the

noise level will be 66 decibels less than an audio signal of 0 dBm. (Since many mixers are calibrated so that 0 VU on the meters corresponds to a level of +4 dBm (about 1.23 volts), the noise would be 70 dB under a signal that causes a 0 VU deflection.)

So is an EIN of −126 dBm good, bad or indifferent? Let's examine it further to see.

Since the noise generator in figure 8 accounts for both source noise and amplifier noise, we will look at only the signal source's contribution for a moment.

Earlier we had calculated that an ideal 200-ohm microphone must generate a noise signal of .2552 microvolts. This happens to be 129.65 dB smaller than the standard reference of .775 volts. So, if the noise generator has an output of −126 dBm, the electronic circuitry must be adding 3.65 dB of noise.

It would be all very simple if all we had to do was subtract an EIN spec from −129.65 dB to determine the NF. However, there are several other things that will muddy the waters.

First, all of the previous discussions assumed a 200-ohm signal source. If 50 ohms is substituted into equation 1, the minimum theoretical noise would be .12761 microvolts which is −135.65 dB referred to 0 dBm. A mixer with a 3.65 dB noise figure can be stated as having an EIN of −132 dBm under these conditions. The noise figure of the mixer is identical in either case, and yet the numbers are 6 dB apart. Human nature tells us that the "better" figure is what will show up in the literature.

Consequently, it is most important to know the value of the source impedance used by the manufacturer during the noise measurements. If it is not stated, the spec is about as good as a Confederate dollar.

Since the bandwidth also plays a key role in determining the noise level, it is important to know whether or not the 20,000 Hertz bandwidth was used in the measurements. A smaller bandwidth results in a lower noise floor which naturally looks better in print. You could take this to an extreme and come up with EIN of −180 dB or something like that, but that's *really* cheating. Fortunately, most manufacturers don't resort to this cheesy manipulation.

By the way, an EIN of −126 dBm with a 200-ohm source is quite respectable, even for recording applications. It could be slightly better, but not more than 3.65 dB since anything beyond that would be impossible.

dBm: Commonly Incorrect

After going all the way with the dBm unit, now we're going to say it is not correct to use that term in the fashion commonly found in EIN specs.

Remember that the dBm definition was based on .775 volts *across 600 ohms*. Yet, the examples used 200-ohm or 50-ohm microphones. Consequently, the reference will get fouled up since it specifically calls for 600 ohms.

So, rather than using dBm, it would be more correct to use dBv. This gives a reference of .775 volts across *any* impedance, including 600 ohms. Now the terminology is accurate. It is possible to properly use dBm in an EIN spec, but due to the mathematics of it all, the numbers end up looking much worse, for that reason dBm is almost never applied properly. In nearly all cases mentally substitute dBv for dBm when reading equivalent input noise specifications.

Other Noise Sources

The microphone preamplifier is not the only source of noise in a mixer. Equalizers, mixing amps, and all the rest of the electronic goodies add their share, too. Consequently, it would be nice to know how badly these other portions will wreck the signal-to-noise ratio. This can be found in several ways.

A measurement of output noise with the mixer's master volume control set for minimum may look like a totally useless spec, but it will tell us how noisy the output stages are. Checking noise with all the inputs turned off and the master at its normal setting indicates the quantity of noise contributed by the mixing bus amplifiers, and perhaps part of the input channel electronics (depending on the circuit configuration). With the microphone inputs short circuited by a jumper, the mic trim pots turned to minimum and the channel level pots set normally, we can determine the overall noise level of the entire circuitry except for the mic preamps. In most all cases, the preamps *are* the greatest noise source, but it's worth checking the mixer under these other conditions just to be sure. Hopefully, the noise will be at least 70 or 80 dB under 0 VU due to all factors except the mic preamp.

All of the noise measurements discussed up to this point assumed a 20,000 Hertz bandwidth. However, the human ear has a hard time detecting low or high frequency information when the volume level is very low. Since the noise output of a mixer is quite small (hopefully!) compared to the normal program audio, the "flat" 20,000 Hertz bandwidth doesn't totally correspond to what the ear perceives.

For this reason, weighting curves were developed for noise measurement. These all roll-off the frequency extremes to give greater emphasis on the midrange noise which is most audible. This is fine except for the fact that numerous different weighting curves are in use, making comparisons most difficult. Keep in mind also that weighted specs will be almost always better than unweighted 20,000 Hertz measurements; this is one reason why they're used — to impress you.

Odds 'n' Ends

When examining the input and output impedance specifications of a mixer, it is preferable that the spec sheet states more information than simply "High" or "Low" impedance. This is quite important with low impedance mic inputs which can cause undesirable colorations of the microphone's signal because of excessive loading (i.e., the mixer can have an input impedance that is too low). Consequently, it is recommended that low Z mic inputs have an impedance of 1000 ohms or greater. It is also important for this input impedance to remain as constant as possible over the audible frequency range of 20 Hz to 20 kHz. Transformer-coupled mic inputs typically can vary drastically in this respect. If the impedance stays over about 750 ohms or so, even at the frequency extremes, the performance of a so-called low impedance microphone will not be significantly degraded because of excessive loading effects from the mixer's mic inputs. A high-impedance mic input should be 50,000 ohms (50 K ohms) or higher to insure minimal loading of this type of microphone.

Line inputs are utilized for interfacing outboard tape recorders or processing equipment with the mixer. The input impedance should be at least 10,000 ohms (10 K ohms) so that the mixer will be compatible with just about any piece of audio gear.

Older mixers were equipped with 600-ohm line input impedances which is fine for most pro outboard equip-

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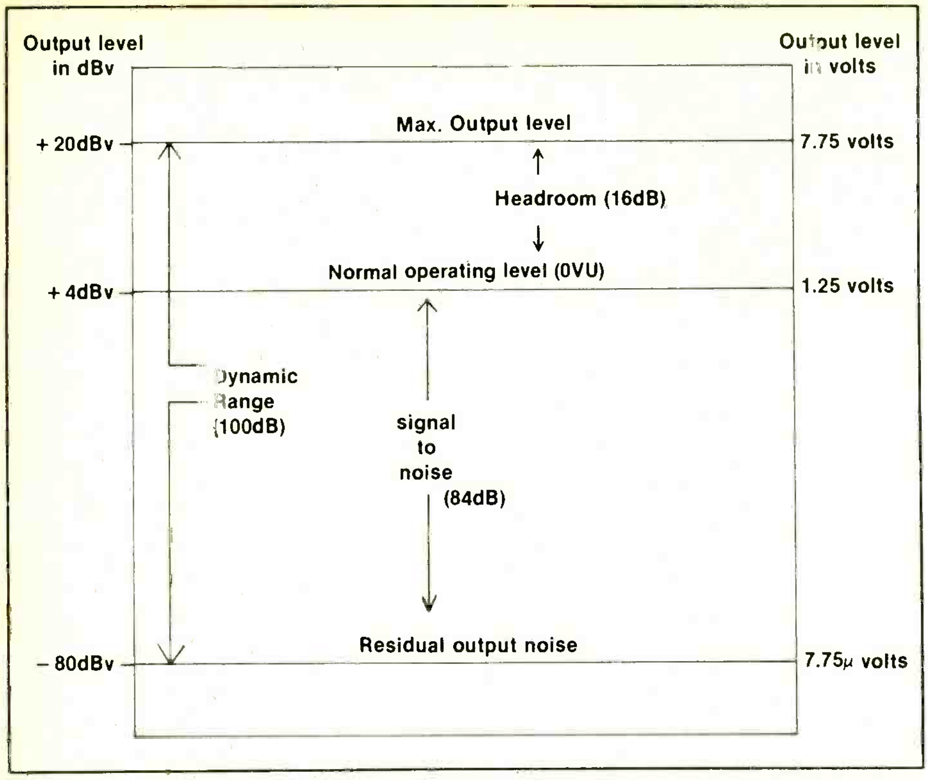

Fig. 9: Relationships of signal levels in a "typical" mixer.

ment, but will generally cause heavy loading of semi-pro, hi-fi or musical instrument equipment. A balanced line input is often desirable (particularly in P.A. applications) since it can bail your tail out of some noisy ground loop problems.

The outputs of most modern mixers have a very low output impedance, often less than 1 ohm! This doesn't mean that the equipment can drive a 1-ohm load, but rather that sensible loads of 600 ohms or higher can be used with no ill effects to the fidelity or the mixer's innards. Additionally, a very low output impedance allows the use of long output cables without high-frequency loss.

Even though an output may have a very low impedance, it may not be capable of proper operation with a 600-ohm load. It is important to know the minimum impedance that mixer will drive because many pieces of pro audio equipment still have a 600-ohm input impedance. Also, the outputs should be balanced in P.A. applications to minimize those nasty grounding noises.

While on the subject of mixer outputs, the prospective buyer of a mixer should determine the maximum voltage that the output can produce before clipping. In a mixer calibrated such that a 0 reading on the VU meter equals 0 dBv or +4 dBv (.775 volts or 1.25 volts, respectively), the line outputs need to be capable of generating at least +20 dBv (7.75 volts) to ensure ample headroom for peak transients that are too fast to produce an accurate reading on the VU meters. A 15 dB margin between the 0 VU output level and the clipping level is mandatory, and 20 dB gives an even more acceptable amount of headroom.

Many mixers cannot provide that much output headroom at high frequencies due to slew rate limiting (discussed earlier). Output transformers *can* limit the maximum output voltage at low frequencies, but high-quality transformers won't impose a significant low frequency limitation. Also, the maximum output level of some mixers is much lower when driving a 600-ohm load compared to a high-impedance (10,000 ohms or higher) load. If equipment with a 600-ohm input impedance is to be connected to the mixer's output, the user should determine if the amount of headroom is sufficient for the intended application.

Figure 9 shows the relationship between maximum output levels, normal operation levels and the residual noise floor of a typical mixer. Note that the range between maximum output and noise is called dynamic range (the largest and smallest signals capable of being produced at the mixer's output). This is sometimes deceptively used for a signal-to-noise spec since it is greater than the ratio between normal operating level and the noise floor, and we all know that "better" numbers are what spec sheets are all about.

Outputs aren't the only part of a mixer that have maximum signal level restrictions. Any of the inputs on the mixer can be overdriven by a sufficiently strong audio blast. In rock 'n' roll applications, low-impedance mic inputs need to be capable of accepting levels of *at least* 0 dBv, and a higher figure (usually obtained by attenuating the incoming signal with a mic pad) is even better. Situations that do not produce higher sound pressure levels at the microphones can be handled by a mixer with a lower maximum input level specification.

Line inputs should be able to accept signal levels up to +20 dBv (7.75 volts) or more to be compatible with professional-type outboard equipment.

An often overlooked characteristic of mixers is the phase relationships between the various inputs and outputs. It is desirable that all inputs and outputs be in phase since an out-of-phase signal can play havoc with the proper operation of a complicated recording or P.A. system. This can be particularly true when interfacing outboard special effects units with the mixer.

Conclusion: If you made it all this way without a brain failure, congratulations. Electrical performance measurements are a complex subject, and we have only hit upon some of the high (and low) points. There are an almost infinite number of factors to be considered when evaluating a mixer, but remember that the ultimate judge of the quality of audio equipment is *your ears.*

For more details, check these references:

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PROFILE:

ELVIN BISHOP

By Craig Anderton

Elvin Bishop's latest Capricorn release, Hog Heaven, *has an element not present on any of his previous albums: Elvin Bishop as producer, as well as player and songwriter. While new to the field of production (and somewhat apprehensive about it at first), his excitement about the project showed that he was more than satisfied with the results. He was particularly quick to share the credit for his excitement with the other players, and technical personnel, involved in the making of the album.*

The nucleus of Elvin's studio band is Mac Cridlin on bass, Scott Mathews on drums and Phil Aaberg (from Elvin's previous band) on piano. For this project, Elvin also expanded his personnel with some well-known, and some not so well-known, session players. As a result, while the new album retains much of Elvin's familiar sound, there are some additional surprises which Elvin feels will delight his old fans and pick up some new ones at the same time.

Elvin's music may be unpretentious and relaxed, but underneath it all is a constant concern for professionalism. Not surprisingly then, these same characteristics showed up in the interview. While Elvin was generally loose and affable, on some subjects he would become very emphatic, phrasing his replies as precisely as possible to get his meaning across. He seemed to enjoy the opportunity to talk to an audience that is knowledgeable about recording.

As Elvin says later, he'll "lie, cheat or steal" to get the sound he hears in his head. He freely admits that he's no technical genius behind the board; but he also made it clear that there's much more to an Elvin Bishop session than technical considerations. As producer, he delegates technical matters to those whom he respects, which leaves him free to concentrate on the subject closest to his heart—his music.

Elvin was not concerned with impressing anyone or creating any mystique, preferring to be as open and candid as possible. His comments were refreshingly unspoiled for this era of super-high technology in recording; he never strayed from his premise that music is the reason for making an album, and that all the technical tricks in the world will never replace a talented musician.

The interview was held at a local San Francisco restaurant.

MR: Your last studio album was *Hometown Boy Makes Good*, and you followed that up with the double album "live" package, *Raisin' Hell*. How did this album come about?

EB: I disbanded the band I had on *Raisin' Hell*; certain members wanted to go off on their own projects anyway. It took a little while to collect my thoughts and my material, then I got a bunch of guys and started jamming in the studio.

MR: How did you reconcile recording and concert schedules?

EB: All during the time I was recording, I'd be flying off to do gigs. You know, that old pragmatic approach . . . if something comes up and there's enough bread in it, drop everything, run and get the bread, then come back and pick up recording where you left off.

MR: Who were the other players for this album aside from the rhythm section?

EB: Amos Garrett, of "Midnight at the Oasis" fame, played guitar; he's a hell of a player . . . he played a bunch of stuff on this album that would have busted my fingers! Melvin Seals played additional keyboards, and he's a monster player too. For vocals, a friend of mine named William Schuler sang one tune on the album ("Waterfalls"); Maria Muldaur sang lead on "True Love" and duets with me on a couple of others; and a gospel quartet called the Gospel Clouds sang background on a few tunes. For the other instruments, Johnny Vernazza played a little guitar on one song, Rick Jaeger played some drums, and "Applejack" played harmonica on "Let's Break Down," a song he wrote.

MR: Who played horns?

EB: Horns were two players from my old band, Jerry McKinney and Terry Hanck. We also had a local horn player named Jules Broussard and a trombone player in his band, Danny Armstrong. They're real good . . . we did all four horn parts "live" and avoided doubling as much as possible to get a more natural sound.

MR: In other words, this isn't a typical "Elvin Bishop Band" album but a bit of departure from the norm?

EB: You might say that. I'm sort of trying to have my cake and eat it too; got a few old rowdy typical Bishop songs on there, and got some of the smoother side too . . . more AM-oriented, I'd say.

MR: Who was the main engineer for your sessions?

EB: A fellow named Mike Fusaro. He's really hot, and a good man to work with.

MR: Why did you want to produce this album yourself?

EB: Well, I had a little talk with myself. Producers we had previously were sort of running me to the poorhouse; they'd had their minds adjusted to this real high budget stuff. God knows this album ended up costing enough with me doing it, but other producers were into 16 and 18 hour sessions and things like that . . . I'll just say that when you produce your own record, and realize that you're financially accountable for all that stuff, things tighten up a little bit and you do things more efficiently.

MR: So, you were more motivated into producing by practical reasons instead of musical reasons?

EB: With a lot of my records, tons of my suggestions ended up being adopted by the producers anyway. Producing myself also eliminated other problems — you know, sometimes people start holding on to their ideas just cause they're *their* ideas, and you get these little conflicts going. I'm not saying any of my producers were really bad about that, but to a certain degree it always happens in any project I've ever seen. With me being the producer and the artist, it automatically eliminated one possible source of conflict.

MR: How do you view the role of producer? Do you feel more like a coordinator, or do you have very specific ideas and the players are simply the vehicle . . .?

EB: I think the basic trick to production is getting the right people in the first place, and then 99% of your problems are solved — as long as you have a pretty good idea of what you want to do with those people. For this project I tried to surround myself with players who were talented enough so I could throw out a basic sketch of a song, and let them go to it. And a lot of times, they came up with some very pleasant surprises. If they ever got bogged down, or too far away from my basic idea, then I had a backlog of specific ideas I could bring in to keep 'em on the right track.

MR: So you're very open to input from the people you select?

EB: Yes.

MR: Were there any creative conflicts, or did things flow pretty smoothly?

EB: Well, that's exactly what I'm talking about: picking the right people. Most of the players were fairly seasoned studio-wise, and the ones who weren't were pretty nice people . . . not hard-headed or nothin'. Everybody had the idea in the back of his mind that the buck stopped with me and the final say would be mine, but it never got to the point where I had to overtly come out and say that; it was just kinda understood.

MR: Were you aiming for a specific audience?

EB: Every motherfucker in the world! It's not a concept album or aimed at a certain part of the market; I'm just trying basically to satisfy my old fans and get some new ones.

MR: Do you do any writing when you are already in the studio?

EB: No, I don't write much in the studio. I'll sometimes think up some new words or something, but basically it's all written before we start.

MR: Do you also have your arrangements completed before you start rolling tape?

EB: So-so. I have the songs mostly together; sometimes I don't have any specific arrangement in mind. Arranging happens more at rehearsals before the record. I have my own little 8-track studio out where I live, and discovered it was much cheaper to get all those little practical problems out of the way at zero dollars an hour than $120 or whatever they're charging in the studio these days.

MR: You know, there are a lot of semi-pro 4- and 8-track home studios out there in the world . . .

EB: It's a wonder more records don't come out of that.

MR: . . . I think they will before too long. Could you talk a little more about your home studio?

EB: The room is really well done. There was a car port there when I moved in, and the control room is on a separate foundation. The studio is about 20′ by 20′ and the control room is 20′ by 10′. I have a Tascam board, which presents difficulties; you always have to be drastically re-patching just to play back what you finished recording. But I have good mics — Sennheisers and stuff, earphones, a little cue system . . . it's all together. If I get some hits out of this album, I just may go 16 track (gets that look in the eye common to musicians contemplating a bigger studio setup) . . . I'm seriously thinking of doing some home recording for the masters.

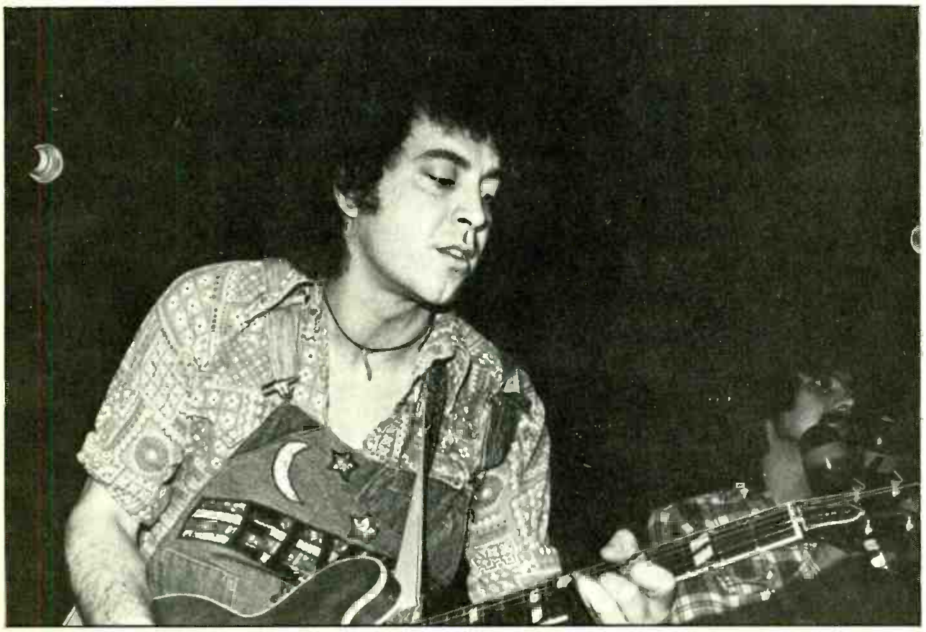
My studio's real good for rehearsals, and getting ready for the record; it saved me a lot of money in studio time. You can not only try out different parts and see if they work like they do in your mind, you can do things like try out different musicians — just invite them over to a jam and see who's the best. Then you don't have to dish out a bunch of session checks for guys that don't appear on the record, or pay for things that don't work out.

MR: How long have you had the home studio up and running?

EB: I started about two years ago. Every two or three months I'll get back and get some licks in on it. Haven't had a whole lot of time to get serious about it, though.

MR: With all this pre-rehearsal, did it take long to record the tracks?

EB: It varied a lot, depending on the overdubs; most of the basic tracks went down pretty fast. We had one problem tune; I guess you always have to have one . . . we heard it, decided about three weeks later that it wasn't up to par with the other stuff, recorded it again with a different drummer, and we did two takes. Both had their points, but we finally decided on one. So we overdubbed horns and vocals only to find out we'd overdubbed on the wrong take and had to do the horns and vocals all over again.



MR: Why did you choose to cut your album at San Francisco's "Automatt?"

EB: Well, the rhythm section I got together for the project had worked with an engineer there a lot. He really knew the room well — I should say rooms, cause we used three of them. I figure once you get past certain obvious flaws or defects in certain studios, most of them are pretty much the same; it's important to find a guy who really knows the room you're going to be working in.

MR: So you don't ask that certain monitors be carted in or that type of thing?

EB: No, my stereo ain't even what you'd call hi-fi. I grew up listening to music recorded in people's basements and churches and stuff. I still put my stereo on mono when I listen, anyway . . . I'm not into refinements, you know. I'm not much of a quadraphonic type of fellow (laughs).

MR: I assume you used the automated mixdown. Did you find that using it was helpful?

EB: Yes.

MR: Did you take mixes home and come back later to make changes?

EB: No, we didn't have the luxury of doing that cause we were on a real tight release schedule. I think we maybe remixed one song.

MR: How interested are you in all the high technology aspects of the recording studio?

EB: To tell you the honest truth, I'm only interested in what directly bears on me. If a guy gets a real good vocal sound on me, which is a pretty rough struggle — I wasn't given a magnificent vocal instrument when I was born — and I find out there's some digital delay on it, next time I'll ask for some digital delay. I also had to learn about reverb, tape delay, chamber echo . . . stuff like that. I wouldn't know how to get a drum sound to save my ass; I just know voice and guitar.

MR: So you work very closely with an engineer . . .?

EB: Yes, and Mike was a really great guy to work with. Engineers in general got a big load if there's a guy like me producin', cause I'll just stand over the poor bastard and say, "That don't sound right," until he gets it right, and can't hardly tell him what he's doing wrong. As far as touching knobs, most knobs look pretty much the same to me, you know? But if I see him turning a knob and something starts sounding good, I'll say "Yeah! That's it; what are you doing?" so next time I can communicate with him a little better.

MR: Are you interested in, or considering producing other acts?

EB: Only if the music was something really close to home to me, something I understood really well. To tell you the honest truth, producing is a pain in the ass 'cause you've got to sit there on your butt for hours on end. Man, I must have gained 15 or 20 pounds! I'm used to being on the action end of things, you know; sitting there making decisions does not sit well with my mentality. The only reason I do it is because I *got* to do it. You know the old saying, "If you want something done right, you got to do it yourself."

MR: Were you trying for a specific "style" of sound?

EB: Most of the sound comes from my own head. Steely Dan might sound good, or certain punk records might sound good, anything might; but most of 'em just don't apply to me. I hear people getting great sounds out of Telecasters and Stratocasters and Les Pauls all the time, but I can't make them sound like nothin', man. All I can do is play a Gibson 345 and I have no business foolin' with the other ones.

MR: How do you resolve the "feel vs. perfection" dilemma?

EB: I kind of walk the line. What *sounds* spontaneous on a record may be something that appears on the 100th take, you never know. When it sounds right to me, I leave it. I ain't

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ST-9030. THD (stereo, 1 kHz): Wide—0.08%. Narrow—0.3%. S/N (stereo): 73 dB. FREQUENCY RESPONSE: 20 Hz—18 kHz +0.1, -0.5 dB. SELECTIVITY: Narrow—90 dB. CAPTURE RATIO: Wide—0.8 dB. IF, IMAGE and SPURIOUS RESPONSE REJECTIONS (98 MHz): 135 dB. STEREO SEPARATION (1 kHz): Wide—50 dB.

SU-9070. PHONO MAX. INPUT VOLTAGE (1 kHz RMS): MM—380 mV. MC—9 mV. S/N (IHF A): MM—100 dB (10 mV input). MC—72 dB (60μV). FREQUENCY RESPONSE: Phono 20 Hz—20 kHz (RIAA ±0.2 dB).

SE-9060. POWER OUTPUT: 70 watts per channel (stereo), 180 watts (mono) min. RMS into 8 ohms from 20 Hz to 20 kHz with no more than 0.02% total harmonic distortion. S/N: 120 dB (IHF A).

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going to leave nothin' that's out of tune or out of time. Then there are those little in-between cases where something is *almost* in tune or *almost* on time; what I'll do is stop right there, and fool around with a little echo or something and see if I can doctor it up to where it will sound right in the mix. But I ain't going for none of that "fix it in the mix" bullshit. I'm going to stop right there and find out for sure; if it ain't right, I've got the guy right there who can do it right, even if it takes ten more takes.

MR: Do you clean up tracks as you go along, or wait for the mix?

EB: In order not to give myself a horrendous job in the mix, I'll erase fluffs, talkovers and stuff we decide not to use *on the spot* . . . *if* I'm not interfering with the flow of things. If there's somebody out there hitting a peak, and had just had the one drink of whiskey that's really going to make 'em sing their ass off, I'm not going to stop and fool with details like that . . . first things first. Or, if a guy's just figured out how he should play a part, I won't let it get cold.

MR: Do you have any tricks to keep a "live" feel in the studio?

EB: No. Just get the best players you can.

MR: Do you cut "live" in the studio?

EB: We set up the rhythm section so they can see each other good, which helps the tightness of the playing; and I'll stand out there and sing a guide vocal, and jump up and down and point and holler and stuff. We do pretty much standard procedure; we have a room ambience mic which we can add if we want it.

MR: Do you have separation problems due to the lack of extensive baffling?

EB: Mike Fusaro and I had some running battles about things like this 'cause naturally he tended to be more technically oriented, and I'm more musically oriented. We ended up kind of meeting in the middle somewhere; he gave up some of his technical perfection, and I had to go for some of it. For example, when we first started I sort of had this irrational thing about limiters . . . I just didn't like the idea of limiters. I said, "As I understand it, the purpose of a limiter is for when you have guys with a real uneven touch; I associate limiters with unprofessional musicians. These players know what they're doing; I don't think they are going to need limiting." Mike said, "Yes they will." I said, "No they won't." So Mike said, "Wait a minute, let's try it out." He showed me there's more to limiting than I thought; there are sometimes sound, legitimate musical reasons for using it.

MR: I think limiters are more a concession to the tape recorder than the players. When did you have to use limiting?

EB: We had some trouble with Amos Garrett sometimes because he has a tremendous dynamic range. He can play really soft to great effect "live," but it wasn't being picked up on the tape until we started using limiting.

MR: Do you have any preferences in noise reduction systems?

EB: I can't tell all that much difference between 'em.

MR: Do you think engineers prefer certain types of equipment more for their own technical reasons than musical reasons?

EB: Well, it must be like being a prostitute and getting a little jaded with the sex act. Engineers hear so much music go by their ears day in and day out, they kind of tend to focus on watching dials instead of listening to what's going on sometimes. The one exception in the world, of course, is Mike Fusaro. I want you to make sure you put that in there [in this article] (laughs). I don't want to get hurt next time I see him (laughs).

MR: Did you find it difficult to produce your own vocals and still be able to be objective?

EB: No.

MR: Have you had any hassles from record companies for having an "uncommercial" voice?

EB: No, I haven't really. Most of the shit I get is from myself. I'm harder on myself than anybody else, but *that's good* because that's how I improve. I had much less problems singing this album than I ever did before, and I was really hard on myself. I'm not sure I technically got the vocal sound exactly right on all the tunes, but I thought it was pretty good and that's saying something.

MR: Generally speaking, do you have a hard time being objective?

EB: No, I know when it sounds right and when it doesn't. Sometimes I'll crack something I think is really hot, and I'll say let's try another on

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But perhaps the best feature of the 9440A is that it's from Altec Lansing—a leader in quality audio products for over four decades.

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another track to see if I can do something a little better. That's when Fusaro and I had some discussions. He'd say, "For technical reasons, if I open up another track now it will make it hard to mix later," and I'd say, "Fuck how hard it is, let's get the magic vocal, that's what you're being paid for."

MR: Did you use all 24 tracks?

EB: Yes, every damn tune just about. Wished we had more sometimes.

MR: Do you generally mic your amps or record direct?

EB: I use my stage amp. It's an old Princeton that was wired by Owsley, the acid [LSD] genius, about ten years ago and it's still going strong. Mike is in the habit of miking amps up close. We ended up a lot of times using a close mic and one back about twenty feet — an overhead mic, you know, gave it a nice sound.

MR: So you tend more towards acoustic ambience than electronic delay processing. . . .

EB: We'll lie, cheat or steal . . . anything that sounds good (laughs).

MR: Anything you'd like to add about getting into production?

EB: I'd like to say I was really uptight about getting into producing at first, but it actually wasn't as hard as it looked from the outside . . . it's mostly just a matter of common sense: keep on pushin' till you get the son of a bitch soundin' right.

It really helped out having studio pros like Mac and Scott, 'cause they didn't just play their tracks, then take their checks and go home like most studio persons. They got involved in the project, and they got real enthusiastic about it. Mac helped with mixing the rhythm section sound, getting the right sound on drums and bass . . . put in a *lot* of hours. Scott was real helpful, so between them and Mike and me, the mixing went real smooth. I've never seen a session with so few personality conflicts; there just about weren't any.

MR: How long did it take to mix down your songs?

EB: About four hours apiece.

MR: You sound like you're pretty happy with the way things worked out. . . .

EB: Yes, I think we've got some hits on this record, if I'm not mistaken. But that's for the people out there to judge, not me.

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CIRCLE 67 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

PCM Update

It's been several months since I last talked about digital, or PCM recording in this column. When we first heard about a studio digital mastering system from 3M we felt that it would be several years before major recording studios would convert their operations to digital audio. It wasn't long after that that we saw prototypes of consumer type digital tape recording systems, most of them in the form of adaptors or attachments to existing video tape recorders, such as Beta and VHS.

Not too long after that we began seeing prototypes of PCM audio discs, many of them based upon systems which were originally developed for analog video disc reproduction. As anyone familiar with the bandwidth needs of video reproduction knows by now, a disc that has enough storage density to be able to store and also reproduce wide-band video signals in an analog fashion also has enough bandwidth to store and reproduce audio in digital fashion.

By now, we have seen at least three approaches to digital audio discs, all of them intended for ultimate use in the home. The three basic types involve optical tracking methods, capacitive tracking and mechanical tracking. In the earliest of these discs shown, a concentrated laser beam is used to trace information contained in a grooveless record. This is the system proposed by Philips (which now owns Magnavox, the company that is slated to produce and deliver the first home digital audio and analog video players), with MCA involved in the production of matching software. Recently, the giant Pioneer Company of Japan has indicated that they will be producing the hardware for these video discs and, ostensibly, the same hardware with modifications could be used to play audio digital versions of the "laser disc."

Last year, Matsushita (Panasonic) startled the audio fraternity with several new disc formats all of which were given the name VISC. On the surface (pardon the pun), these discs seemed to be about the simplest to play since they did feature actual grooves traced by a new strain-gauge type of cartridge or stylus. And of course, the video versions were augmented by an audio-digital version, which they called VISC-AD.

Most recently, JVC Industries unveiled what many industry experts believe to be the best compromise of all the video/audio PCM disc formats. It is one in which there are no physically identifiable grooves in the record (in the usual sense), but in which the subsurface "grooves" consisting of encoded "pits" and "bumps" are traced by a capacitance pickup that is guided across the disc electronically, making it possible to achieve random access to any portion of the video or audio disc (and even to achieve "stop frame" action in the case of video).

In this writer's opinion, the proliferation of disc formats in the last year will probably delay the impending revolution in home entertainment that others have been predicting. It is not likely that three or four systems can exist simultaneously in the case where the user depends upon pre-recorded software for program sources. That situation is *not* true when it comes to digital tape systems, whether one is talking about home tape recording or studio mastering systems. Here, the user need not be that concerned with standardization, because we are essentially dealing with a "closed loop" system. The user has control of the making of the master recording (analog to digital conversion and storage, in digital form on tape) as well as subsequent transfer to a master lacquer (in professional studio activities), or home reproduction via a suitable complementary digital-to-analog audio conversion scheme (in audiophile or semi-pro usage).

On the basis of these considerations, some recently introduced products by Sony Industries (at the most recent Audio Engineering Society convention in New York) promise to give the earlier-announced 3M studio mastering system a run for the money. In their large display area at AES, Sony showed four separate products intended for use by professional recording studios, plus yet another home digital audio disc (their third version in less than three years), this one capable of 2½ hours, all on one side of the disc. Ignoring the home disc system for the moment, let's examine the other products introduced.

The first of these (and the only one available for immediate delivery) is the PCM-1600, a two channel PCM processor for use with Sony's professional U-matic video cassette recorders and editing equipment.

(continued)

The first available digital audio processor (PCM 1600) from Sony.

Using this device, it is possible for a studio to record a stereo master or sub-master with better than 90 dB of dynamic range, distortion of less than 0.05% over the entire audio spectrum. As with all digital or PCM systems, wow-and-flutter is unmeasurable since it becomes a function not of the mechanical irregularities of the tape transport but is controlled by the precision of a quartz sampling-rate clock. According to Sony, there is no hint of tape hiss or print-through.

Editing and dubbing can be performed in a direct digital-to-digital mode through a video editing console. Using this system, separate takes can be reassembled in any order without signal quality loss and generation after generation of lacquers can be cut with direct-disc fidelity. The analog-to-digital and digital-to-analog converter section of the PCM-1600 uses 16 bits of information for each channel, which accounts for the high dynamic range.

Also shown, though in prototype form, was Sony's new PCM-3200 series of digital recorders. These are to be available in models with from 4 to 48 digital channels, plus analog and SMPTE time-code tracks. These decks are comparable in size to conventional studio recorders and are provided with both digital and analog input and output connections for interfacing with both existing analog equipment and the new generation of digital studio systems. Dynamic range, distortion and wow-and-flutter are essentially the same as for the PCM-1600 processor, since 16-bit encoding is used, and frequency response per channel is flat within +0.5, −1.0 dB from 20 Hz to 20 kHz. Tape speed is 22½ ips, but production versions may also run at 15 ips, for playing time of 60 and 120 minutes from a 10-inch reel. 16- and 24-channel versions will use 1-inch tape. Two-inch tape would be used for the 32 and 48 channel machines, while ½-inch tape would be used for the 8-channel version and ¼-inch tape would be used for the 2- and 4-channel models.

One of the stumbling blocks in the creation of an all-digital recording studio has been the lack of available hardware for mixing of input signals, or mixing down from multi-track recordings. Until now, it was necessary to first convert the digital multi-tracks back to analog signals, mix them on a conventional mixer, and then convert them back to digital code for final master-recording, usually with some resultant loss of quality of signals.

Sony's DMX-800, also introduced at AES, is designed to mix 16-bit digital audio signals in real-time, with no analog process involved. Looking for all the world like a conventional mixer (there are the usual fader controls, echo send and receive, analog-type peak-program level displays in the form of "plasma" bar graphs and even inputs and outputs for Sony's equally new DRX-1000 digital reverb unit), the mixer has 8 input and 2 output channels. Designed to mix 16-bit linear quantized digital signals, it can be synchronized with either internal or external sampling rate clocks. The internal clock offers switch-selectable sampling rates of 44,056 (fast becoming the standard sampling rate for both consumer and professional digital tape systems, because of its ideal time relationship to video tape transports and international video standards) and 50,350 Hz. The mixer can also be used with conventional analog equipment, interfaced via an AD/DA converter such as Sony's ADA-1601, another digital-related product introduced at the AES show.

The DRX-1000 digital reverb unit referred to earlier was the final digital product shown by Sony at AES. It offers a range of initial delay times from 0 to 100 milliseconds, with reverb times ranging from 0 to 20 seconds. A built-in microcomputer allows front-panel programming of any of four reverberation modes. Reverb mode selection can be stored in a non-volatile memory. Like the other products discussed, the DRX-1000 accepts digital signals directly, and adds reverberation digitally; there are no analog-to-digital conversions unless the unit is interfaced with a conventional analog system by an external converter.

All in all, companies like Sony seem to be demonstrating a tremendous technological capability in the newly emerging field of digital audio recording, at least at the professional studio level. If nothing else, the increased use of digital equipment in recording studios in the months and years ahead could well signal the early end to direct-to-disc recordings which are only now gaining in popularity and public acceptance. Once complete studio conversion to digital takes place, the chief disadvantage of direct-to-disc recording (having to cut a complete performance in real-time and without the advantages of multi-track recording and mixing) will be a thing of the past while the chief advantages of these expensive pressings (greater dynamic range, better signal-to-noise ratio) will all be retained. Once the digital multi-track and mix-down equipment is amortized, I see no reason why discs mastered from PCM studio equipment should be any more expensive than conventional discs derived from conventional analog master tapes.

These are the "big guns" in "professional" power amplifiers. Each of these amplifiers has individual features and abounds with specifications to impress potential buyers and to satisfy the professional user but they are **not** created equal... especially in reliability under professional (rack mounted) conditions.

Some of these "big guns" have been talking about everybody else being "behind", others are talking about comparator LED's, while others depend mostly on their good looks. The Peavey CS-800 comes out on top when you consider the features, the specifications (which are as good or better than anybody's), total power output, and price per watt of professional power.

Some companies have recently "discovered" LED's and comparator circuitry that Peavey pioneered and has been using for years. These recent "converts" were most vocal in the past against LED's...that is until they updated their "plain Jane" units. Some of the other companies spend a lot on cosmetics but not much on built-in forced air cooling and large numbers of output devices to enable reliable rack mounted operation under continuous professional use

Each channel of the Peavey CS-800 features 10 output devices and 2 TO-3 drivers bolted to massive modular heatsinks that are forced cooled by a 2-speed fan, has special distortion detection circuitry and LED indicator (not simple overload), as well as a functional patch panel on the rear to facilitate the use of plug-in balanced transformer modules electronic crossover modules and speaker equalization modules custom tailored to Peavey's SP-1 and SP-2 speaker systems.

In comparing pro amplifiers, one should apply the old commercial sound "dollar-per-watt" rule. The CS-800 is again "on top" at 81¢ per professional watt. The fact is...Peavey is not behind anyone in power, durability, features or performance.

Below are the respective published specifications of the "heavies" in pro amps. Check for yourself to see how we all stack up. You might be surprised.

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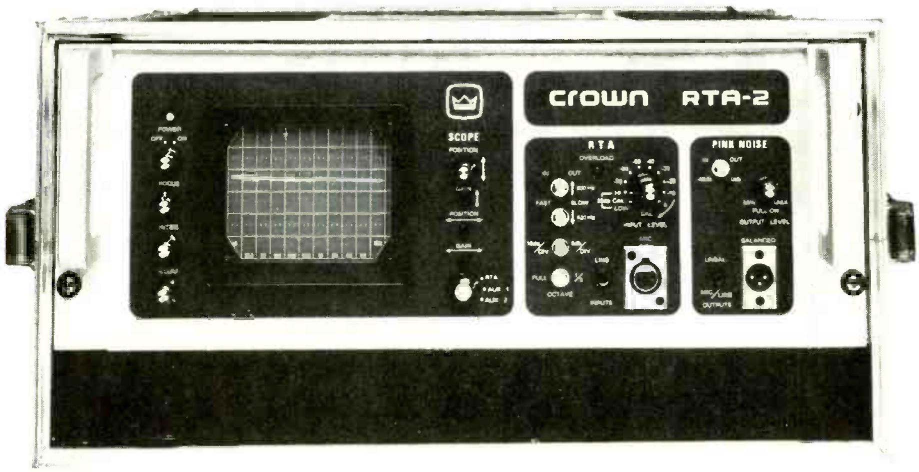
Peavey CS-800	800 W Total 400 Watts/Ch. @ 4 Ohms 260 Watts/Ch. @ 8 Ohms (Both Ch. driven)	20	2 Speed forced air cooling	Yes	Totally Plug-in Modular	None Required	Quasi Complimentary. All rugged NPN Silicon Outputs	Not given. No accepted Measurement standards Presently exist.	$649.50	**$0.81 per Watt** Based on 4 Ohms/Ch. min. load
Crown DC-300A	360 W Total 180 Watts @ 8 Ohms 4 Ohms Not Given	16	Conventional Passive Airflow Only	No	Hard Wired	None Required	Quasi Complimentary. All rugged NPN Silicon Outputs	Not given. No accepted Measurement standards Presently exist.	$919.00	**$2.55 per Watt** Based on 8 Ohms/Ch. min. load
BGW 750 B	720 W Total 360 Watts/Ch. @ 4 Ohms 225 Watts/Ch. @ 8 Ohms	20	2 Speed forced air cooling	Yes	Modular	Relay Circuit	Collector drive Complimentary using PNP & NPN Silicon	.02% No measurement details given.	$1099.00	**$1.53 per Watt** Based on 4 Ohms/Ch. min. load
Yamaha P 2200	700 W Total 350 Watts/Ch. @ 4 Ohms 200 Watts/Ch. @ 8 Ohms	12	Conventional Passive Airflow Only	No	Hard Wired	None Required	Emitter follower drive complimentary using PNP & NPN Silicon	Not given. No accepted Measurement standards Presently exist.	$1095.00	**$1.56 per Watt** Based on 4 Ohms/Ch. min. load

All above figures based on manufacturers' published specifications and minimum recommended load impedances as of 11/1/78

CIRCLE 89 ON READER SERVICE CARD

NORMAN EISENBERG AND LEN FELDMAN

Crown Model RTA-2 Real-Time Audio Analyzer



General Description: The Crown RTA-2 is a device for performing real-time spectrum analysis of the audio range, covering frequencies from 16 Hz to 20 kHz in full-octave or one-third-octave bands. It may be patched into a sound system at any point, from source to loudspeakers, and thus is usable for checking individual audio components as well as the complete system, including on-site analysis of sound-reinforcement setups and general "room response."

It includes a built-in 5-inch oscilloscope (CRT display), its own pink-noise generator, and the requisite inputs and outputs and controls that allow the user to enter a sound system at any point.

The 'scope dominates the left-hand portion of the front panel. To its left are the device's AC power switch and pilot indicator. In addition, there are the knobs for the 'scope display, regulating focus, intensity and degree of illumination. The last control adjusts the brightness of the graticule of the 'scope, which is lighted for viewing in dim light, and for making photos of the display pattern.

To the right of the 'scope are additional controls for vertical position (knob); for vertical gain, horizontal position, and horizontal gain (these three are screwdriver adjustments); and for selecting function (a three-position switch for choosing RTA, aux 1 or aux 2, depending on what other equipment is involved, and in what mode the analyzer is being employed).

The section of the panel marked RTA contains four pushbuttons, two inputs, an input level control and an overload indicator LED. Two of the pushbuttons may be used to slow down or dampen the CRT display to facilitate making EQ adjustments. One button handles the frequency range from 16 Hz to 630 Hz; the other, from 800 Hz to 20 kHz. The third button selects the vertical sensitivity of the CRT display—10 dB or 5 dB per vertical division. The fourth button selects the frequency segment to be shown, either full-octave divisions, or one-third-octave divisions. In either case, the full audio spectrum remains displayed.

One input is a standard ¼-inch phone jack for line-level signals; the other is a locking Cannon connector for microphone input.

The input level control is a dual-concentric pair, with the outer knob serving as a step-switch attenuator over a 70-dB range in steps of 10 dB. The inner knob is a continuously variable vernier attentuator over a 40-dB range. Both controls also are used in vertically calibrating the instrument.

The section of the panel marked "Pink Noise" contains a control knob/pushbutton to activate the pink-noise generator; when this button is pulled out it also serves to adjust the intensity of the pink-noise signal. Another button may be used to attenuate the line-level pink noise to microphone-level pink noise. Two outputs are located here. One is a standard ¼-inch phone jack for unbalanced microphone and line; the other is a Cannon XLR type for balanced output.

Additional signal jacks and adjustments are at the rear. These include phono jacks for vertical out; vertical in (aux 1); vertical in (aux 2) sweep out; horizontal in (aux 1); horizontal in (aux 2) and pink noise out. Both pairs of vertical and horizontal jacks have their

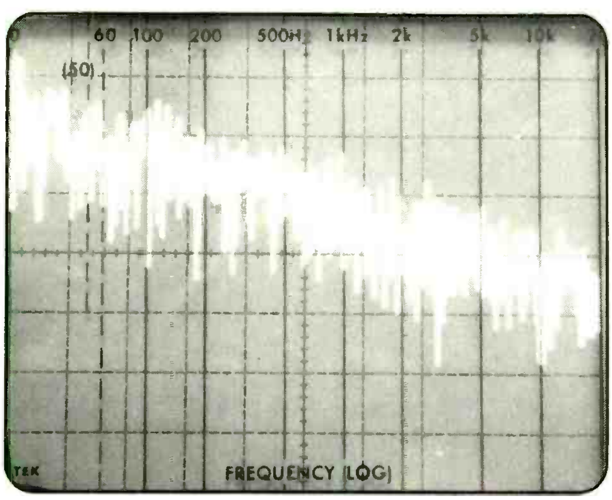


Fig. 1: Crown RTA-2: Pink noise output of RTA-2 as displayed and stored on scope face of our frequency-domain spectrum analyzer.

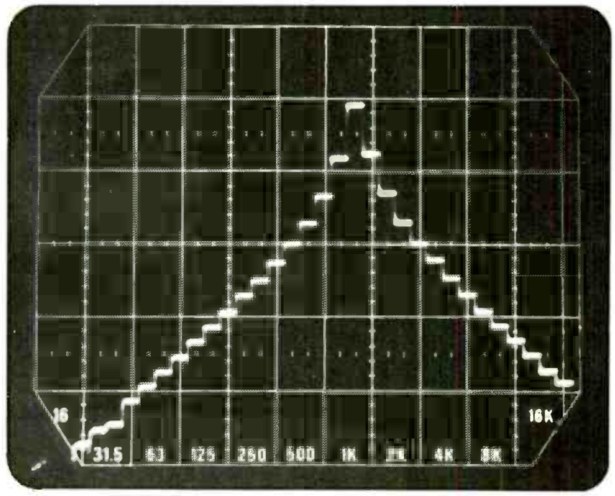


Fig. 2: Crown RTA-2: 'Scope display on RTA-2 when set for 1/3-octave display and fed with 1 kHz signal tone.

own screwdriver adjustments. There also is another line-in phone jack. Finally, sitting atop a power supply extension cover is an astigmatism control which may be used, together with the front-panel focus control, to adjust the 'scope trace for optimum clarity. The outputs permit connecting the RTA-2 to other instruments (such as a storage 'scope or an oscilloscope). The inputs allow the RTA-2 to present an "x-y" display. The rear line-level input, and pink-noise output jacks are overridden whenever their corresponding respective front-panel jacks are used.

Internal circuit jumpers permit using the RTA-2 on AC mains of 100, 120, 100, 220 and 240 VAC. In addition, the device has a built-in phantom power supply which, via another internal adjustment, may be used to supply 15 volts DC for operating those condenser microphones requiring such voltage.

The Crown RTA-2 is of rack-mount dimensions. It is supplied in a sturdy, polished metal case with a grey front panel, clear white markings and a blue 'scope face.

Test Results: Within the limits of our own test instrumentation, we confirmed or exceeded the published specifications for the Crown RTA-2, which we regard as an accurate and versatile "working tool" that can find application in many areas of audio analysis and adjustment, and especially so in the case of audio dealers or professionals involved in selling or installing sound equipment (and particularly equalization equipment).

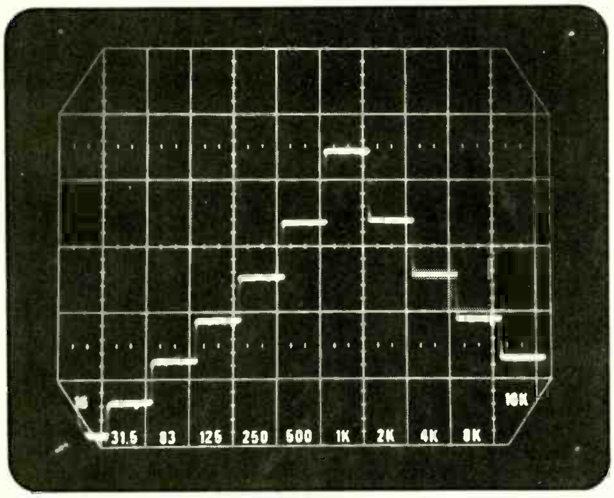


Fig. 3: Crown RTA-2: 'Scope display on RTA-2 when set for 1-octave display and fed with 1 kHz signal tone.

Fig. 1 is a 'scope photo taken of the face of our frequency-domain spectrum analyzer (Tektronix 5L4N in a 5100 Storage Scope Mainframe) after sweeping it from 20 Hz to 20 kHz and feeding the analyzer input with the pink-noise signal provided by the RTA-2. As may be seen from the results, the pink noise versus frequency follows the required 3 dB/octave slope with increasing frequency and energy of noise per octave being quite uniform, as it should be.

Fig. 2 is a 'scope photo taken of the face of the RTA-2's own built-in CRT while applying a 1-kHz signal to the analyzer's input. As can be noted, the filter bandwidth conforms very closely to that called for by ANSI S1.11-1966 Class II and IEC C225-1966 Standards for filters of this type and purpose. The bandwidth of the filters, however, precludes using the RTA-2 for harmonic distortion analysis, a task that is best performed by a frequency-domain spectrum analyzer. In Fig. 2, the RTA-2 was set up to display in the 1/3-octave mode, wheras in Fig. 3 the same signal is displayed using the full one-octave mode. The horizontal calibration accuracy of the sweep is clearly evident in this display, from the lowest to the highest audio frequencies analyzed.

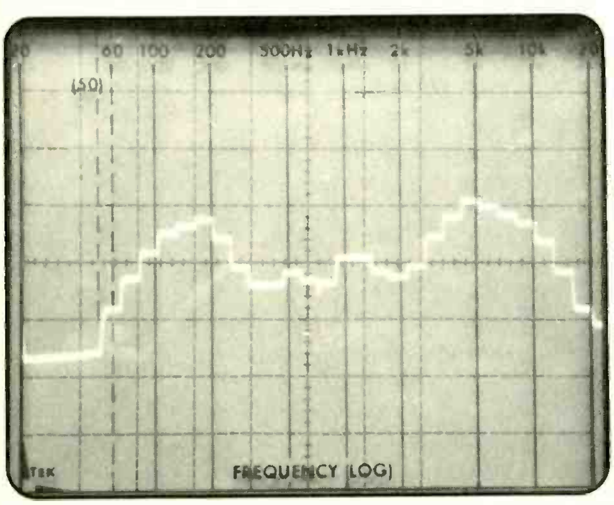


Fig. 4: Crown RTA-2: Typical "system + room" response displayed by RTA-2 before equalization.

As one of the exercises in using the RTA-2, we hooked up a microphone to a system that included a loudspeaker known to be anything but linear in response (no names, please!). Confirming our previous aural reaction to this speaker, the overall response of the system was that shown in Fig. 4. Note that we photographed this display by applying the external outputs of the RTA-2 to a storage 'scope, only in order to avoid a blurry picture from the undulating blips on the RTA-2 screen.

Inserting a one-octave equalizer in this system, it took but a few moments to tweak the controls of the equalizer (while observing the RTA-2 'scope face) to come up with the system response shown in Fig. 5. Still not perfect, to be sure, but orders of magnitude better than before. (With a one-third octave EQ on hand, we could have adjusted the system for even greater linear response.

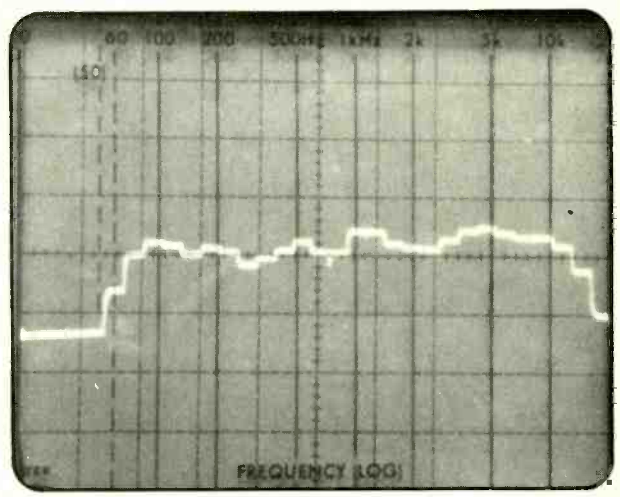
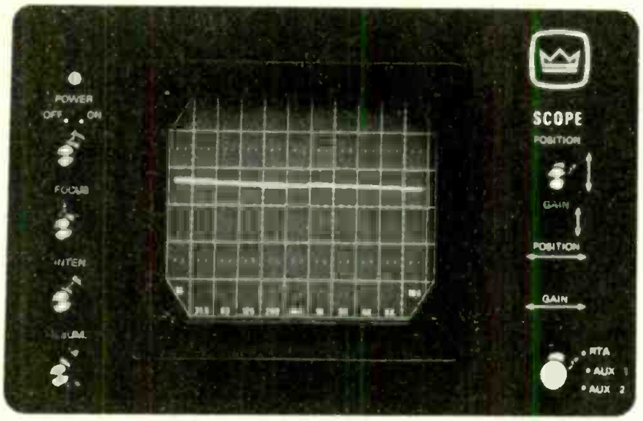


Fig. 5: Crown RTA-2: With the aid of the RTA-2, response shown in Fig. 4 was easily equalized to "nearly flat" as shown here.

General Info: Dimensions are: 19 inches wide; 7 inches high; 15 inches deep. Weight is 22 pounds. Unit is supplied with one pin-plug to phone-plug cable, and one XLR to XLR (male-female) cable. Price is $2595.

Joint Comment by N.E. and L.F.: Crown's newly introduced RTA-2 is part of an ongoing technological revolution that has brought down the cost of real-time analyzers to levels that can be afforded by the serious sound contractor, recording studio owner or hi-fi specialty shop. It was not all that long ago



Crown RTA-2: Close-up view of RTA's 'scope face.

when the facilities and capabilities built into the unit would have cost upwards of $10,000. Moreover, the amount of equipment required would take up many times the amount of space and weight of this neat package.

In a recent issue of *MR* (11/78) we tested and reported on another real-time analyzer from Ivie Electronics. That device had an LED display and its own calibrate microphone. It also ran on batteries.

In contrast, the Crown unit has a built-in CRT display and a built-in pink-noise generator, but to do a complete system analysis and equalization one would need a microphone of known calibration. And while the Crown is compact enough to be carried (it is available in a case with a carrying handle) it requires an external AC source of operating power.

Among the RTA-2's many attributes, one that is particularly appreciated is the ability to separately select the display decay speed for the 16 Hz to 630 Hz range, independently of the speed selected for the 800 Hz to 20 kHz filters, and the slow and fast speeds chosen seemed just right for most tasks for which we might normally use a real-time analyzer.

In doing such work, there are times when we want to be able to photograph response curves obtained under various conditions. Often, especially when the high-speed mode has to be used, the 1/3-octave blips tend to wiggle, plus or minus a few dB, because of the random nature of the pink-noise source. Under those circumstances, it becomes difficult, if not impossible, to get anything but a blurred picture of the display. Happily, the RTA-2 offers external display outputs for hookup to a storage 'scope (or to a larger display), which is exactly what was used to take the photos shown in Figs. 3 and 4.

In addition to its main uses, it should be pointed out that the RTA-2 can also serve to analyze the frequency content of source material itself—such as recorded tapes. It is possible, for instance, to patch the device into a tape-recorder system so that the pink-noise output of the RTA-2 is recorded, and the playback is analyzed by the RTA section of the device. With a three-head recorder (that provides instantaneous monitoring of what is being taken down) this becomes really "real time" analysis.

CROWN MODEL RTA-2 REAL-TIME AUDIO ANALYZER: Vital Statistics

PERFORMANCE CHARACTERISTICS	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency range	16 Hz to 20 kHz	Confirmed
Filters	32 1/3-Octave	Confirmed
Display modes	1/3 or full octave	Confirmed
Sensitivity for full scale		
Unbalanced (sine wave)	15.2 mV to 150 V max.	15 mV to 155 V max.
Balanced (sine wave)	0.76 mV to 3 V max.	0.75 mV to 3.0 V max.
Overall display accuracy		
Full scale to −40 dB	±1.0 dB	±1.0 dB
−40 dB to −50 dB	±1.5 dB	±1.0 dB
−50 dB to −60 dB	±3.0 dB	±2.0 dB
Detector time constants		
slow	15 ms/375 ms	Confirmed approx.
fast	15 ms/47 ms	Confirmed approx.
Scan speed	32 channels in 16.6 ms	Confirmed approx.
Input and impedance		
balanced	XLR (2 K ohms)	Confirmed
unbalanced	¼-in. phone jack (1 megohm)	Confirmed
Pink noise repetition rate	2.1 seconds	Confirmed approx.
Pink noise filter accuracy	±0.5 dB, 16 Hz to 20 kHz	Confirmed indirectly
Pink noise output level	1.1 V rms	1.3 V rms
Pink noise output impedance		
balanced, 0 dB/40 dB	600/50 ohms	Confirmed
unbalanced, 0 dB/40 dB	300/25 ohms	Confirmed

Heathkit Model AA-1600 Power Amplifier

General Description: One of several new units in its new "professional-style" component series, the AA-1600 from Heath is a rack-mount stereo basic or power amplifier rated for 125 watts, minimum RMS power, per channel into 8-ohm loads, across the band from 20 Hz to 20 kHz at a rated THD of less than 0.05%. IM similarly is spec'd at under 0.05%.

The unit presents a rather Spartan appearance, with a front panel that contains only the power switch and four LED indicators: one for power on, one showing high temperature, and two for full power level indication on each channel.

The rear contains, for each channel, a level adjustment, input jack (pin-type) and speaker output terminals—five-way binding posts.

Internal protection circuitry includes an eight-second turn-on delay that allows transients to decay before the speakers are electrically energized. This same circuit disconnects the speakers electrically at turn-off. The safety circuit also will turn off the speaker should a DC voltage appear at the output terminals. In the event of overheating, a thermal circuit breaker disconnects the speakers, and the front-panel high-temp LED lights up. The supply voltage to each channel is fused.

The AA-1600 may be rack-mounted or installed in an optional oak-finish accessory cabinet.

Test Results: All published specs for the Heathkit AA-1600 were, in *MR's* tests, exceeded by fairly wide margins. THD was especially low up to, and beyond, rated power. At 1-kHz inputs, the amp delivered more than 140 watts per channel before reaching its rated 0.05% THD, and more than 146 watts for an IM reading of 0.05%. Results are graphed in Fig. 2, while distortions-vs.-frequency, with both channels delivering rated power into 8-ohm loads, is plotted in Fig. 3.

In addition to these static tests, *MR* also investigated the dynamic headroom of the amplifier in accordance with procedures outlined in the New IHF Amplifier Measurements Standard, and came up with an impressively large 3.75 dB of dynamic headroom. To the uninitiated, this figure means that under short-term musical or transient signal conditions, the AA-1600 can deliver unclipped power levels as high as 296 watts per channel.

The unit tested came pre-wired, although it is sold only as a kit. Based on our examination of its innards, the job of building it from the parts supplied should prove relatively easy and foolproof. As shown in Fig. 1, the amplifier uses three major circuit boards: two identical boards for the amplification circuitry, and one for the power supply. These boards, in the kit, are supplied factory-assembled and tested, so the rest of the job is relatively uncritical, and shouldn't take more than a few hours to complete. There is little point-to-point wiring involved; most of the kit-building chore is mechanical rather than electronic.

General Info: Dimensions are: 19 inches wide; 7⅛ inches high; 13 inches deep. Weight is 38.25 pounds. Price: $359.95. Optional oak-finish cabinet: $27.95.

Individual Comment by N.E.: As Heath has demonstrated in the past, an audio product in kit form can perform as well as a costlier factory-built version. And without doubt, the most successful kit products from the standpoint of "likely to succeed"—even with first-time kit-builders—are power amplifiers. In the case of the AA-1600, as with other of its kits, Heath has rated the product conservatively in terms of specifications just to make sure that even the inexperienced kit-builder (who probably will not dress internal wiring, or perform soldering chores, as expertly as an experienced or trained assembler) will end up with a

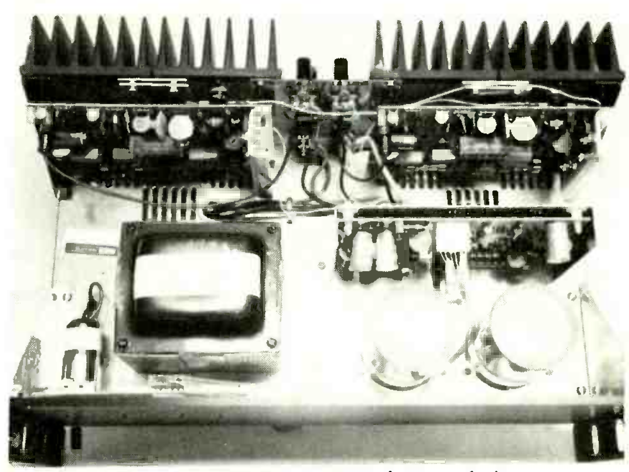


Fig. 1: Heathkit AA-1600: Internal view.

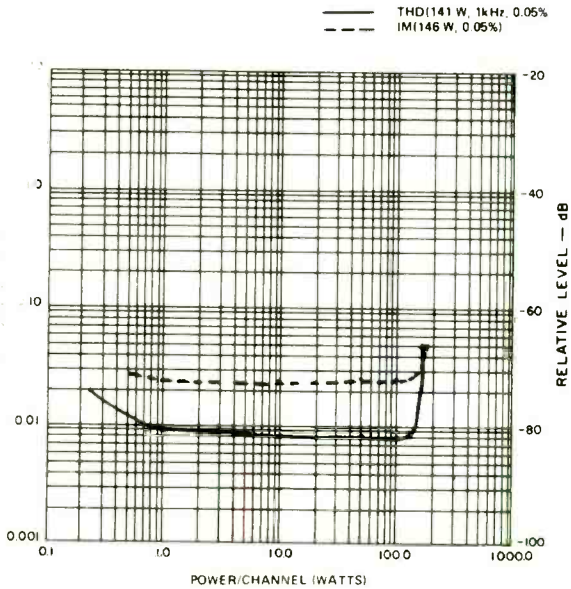


Fig. 2: Heathkit AA-1600: Harmonic and IM distortion vs. power output/channel.

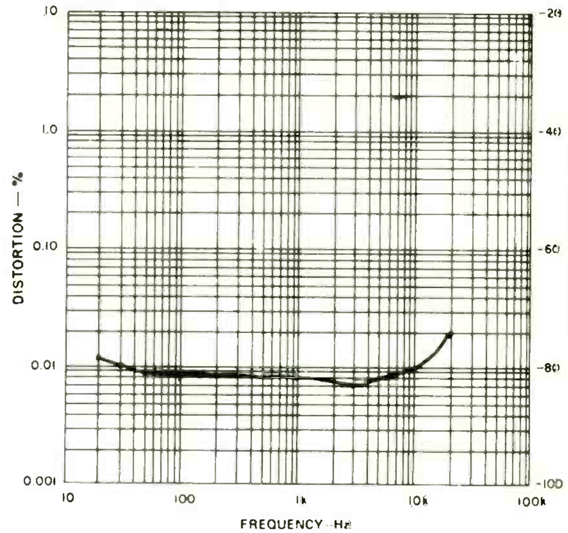


Fig. 3: Heathkit AA-1600: THD vs. Frequency, at rated output, 8-ohm loads, both channels driven.

unit that does meet its claimed performance specs. By the same token, then, it is logical to expect that the experienced assembler will usually come up with a finished product that exceeds specs.

The AA-1600 lacks its own power output monitor, but even if you add the cost of the ancillary device that serves this purpose (the AD-1701, $189.95) you still come up with a system that is very competitively priced.

Individual Comment by L.F.: The AA-1600 should have tremendous appeal for the semi-pro or professional audio person who is always looking for that "sleeper" of a product that delivers performance far above its cost. In the case of this unit, the attractive pricing is obviously due, in part, to the fact that you must buy it in kit form. However, even if you add the few hours of labor (yours or someone else's) it seems to me that you still come out way ahead with the AA-1600. Its rack-mountability seems particularly attractive to *MR* readers.

One little thing did bother me a bit. The left- and right-channel LEDs on the front panel actually "warned" about overload well before overload occurred. The left LED came on with outputs of only 86 watts, while the right one lit up for 102 watts. Could that be more Heath conservatism in design, or was it just the creeping tolerances of the circuit, all working in the wrong direction? In any case, the bandwidth of the amplifier is awesome, and in my listening tests that wide bandwidth paid off, for I heard transients reproduced flawlessly and was able to drive the amplifier a lot harder than one would expect with a "125 watt per channel" rating.

HEATHKIT AA-1600 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTICS	MANUFACTURER'S SPEC	LAB MEASUREMENT
Power output per channel, 8-ohm loads, 20 Hz to 20 kHz	125 watts	131 watts
Rated harmonic distortion	0.05%	0.008%
Rated IM distortion	0.05%	0.025%
Damping factor	>50	77 (at 50 Hz)
Frequency response	± 1 dB, 7 Hz to 50 kHz ± 3 dB, 5 Hz to 100 kHz	± 1 dB, 4 Hz to 210 kHz ± 3 dB, 3 Hz to 350 kHz
Hum and noise	− 100 dB below rated output	− 110 dB (unwtd) − 120 dB ("A" wtd) − 92 dB (new IHF method)
Channel separation	70 dB minimum	80 dB, 1 kHz
Input impedance	20 K ohms	Confirmed
Input sensitivity	1.5 V for rated output	1.5 V (0.13 V for 1-watt output, per new IHF method)
IHF dynamic headroom	NA	3.75 dB

CIRCLE 35 ON READER SERVICE CARD

B.I.C. Model T-3 Cassette Recorder

General Description: The top of the line B.I.C. Model T-3 is the first (and so far only) cassette recorder that contains three heads and runs at either of two speeds. The latter feature, of course, is an innovation which, as our test results show, is responsible for significantly improved audio performance. The two speeds are the standard 1⅞ ips, and double that or 3¾ ips. The same conventional cassette is handled at either speed.

The record and play heads are full-fledged, full-performance heads, but contained in a common housing. As the literature on this model points out, this design obviates the need for realignment since the gaps for both record and play are permanently positioned with respect to each other. Moreover, the very closeness of the two gaps tends to "wash out" any tape misalignment which might occur as a fault of the cassette itself. Our tests of this unit tend to bear out this assertion quite emphatically.

A front-loader, the T-3 is styled with a black front panel and rosewood cabinet. Lettering and numbering on the controls are in white, and the front panel is both neat and very legible. The cassette compartment is at the left behind a swing-down door which itself has a removable cover. The power off/on button is to the left; transport keys are grouped below. These include the usual functions, including a pause control.

Fast-buttoning is not one of this deck's features; you must press the stop key before changing transport direction or transport modes.

Right of the cassette area are the index counter and reset button, plus a memory-rewind button. Next come the two signal meters, peak-reading and calibrated from −40 to +5. Between them is a unique L.E.D. which glows green in the record mode, but turns red to indicate overload.

Below the meters is a dual-concentric knob for setting recording levels individually or simultaneously on each channel. To the right of the right-hand meter is the output level control (both channels simultaneously). Below it is a headphone level output control that operates independently of the line-output level.

Additional controls are ranged across the lower part of the front panel. These include: the two-speed selector switch; the EQ switch (70 or 120 μsec.); the bias switch (hi, norm, lo); the record mode switch (safe, ready and mute); the Dolby-B switch (copy, on, off); an MPX filter switch; the tape/source monitor button; a record calibration button; and the mic/line selector. Finally, there are the input jacks for left- and right-channel microphones, and the headphone output jack.

When the record switch is in "safe" position, recording is not possible. To record, it is necessary to move this switch to "ready" and then activate the play and record transport keys. The "mute" position of this switch is spring-loaded; it must be held down to silence the incoming signal while the tape is running.

The bias switch markings extend to the record switch, a nice touch that indicates that bias settings pertain to recording and not to playback. A red L.E.D. comes on over the record switch when it is activated; a green L.E.D. lights up for the Dolby switch.

Line inputs and outputs, and the unit's power cord are at the rear.

Test Results: The performance of the B.I.C. T-3—at both speeds—was definitely suggestive of cassette decks costing much more than it does. This verdict applies to all tested performance areas: frequency response, distortion, signal-to-noise, recording headroom, wow and flutter, and so on. The specifications for the T-3 are impressive enough; our lab tests confirmed or surpassed them. *MR* tested the unit at both speeds, using two different tape types as applicable, and came up with nothing but superlatives on all counts. The "worst case" response made it out to 19 kHz; the best response extended from below 20 Hz to beyond 20 kHz. S/N varied from 57 dB to 69 dB, depending on speed and tape used. The latter figure, of course, suggests the kind of performance you'd expect of a high-grade open-reel deck. The same story could be told of other tested parameters (see the results in our "Vital Statistics" table).

Interestingly, the manufacturer chose not to "pin down" its three bias settings to specific generic tape types, but instead labeled them low, normal, and high. This tends to encourage the user (and us) to experiment with the different settings regardless of the tape type used. Of course, the owner's instructions do offer suggested "starting points" of EQ and bias for various tapes. Because of this flexibility, we decided to plot response (using our spectrum analyzer) by sweeping frequencies from 20 Hz to 20 kHz. Since the T-3 is a three-head machine, we were able to plot response as the recordings were being made by adjusting the bandwidth of the analyzer so that it could track the slight time-difference between the signal-source and the recorded playback. This time-difference is very slight in the T-3 because both record and play heads are contained in the single housing. Anyway, Fig. 1 shows the response obtained with TDK Type AD tape, for all three bias settings, at the standard $1\frac{7}{8}$ ips speed. Clearly, the "low" bias position is optimum for this combination, and it is the setting we used to derive the frequency response data shown in our "Vital Statistics" table for this speed. Similarly, the "high" bias position yielded the most uniform response when the measurements were repeated for TDK-SA tape at the slow speed (Fig. 2).

When the frequency response runs were repeated for the faster speed, we found that both the AD and the SA tapes yielded the best response curves with the bias set to its "high" position. Lower bias settings would have caused a rising high-frequency response, as shown in the 'scope photos of Figs. 3 and 4. Again, the response data listed in the Vital Statistics table is based on the bias settings that yielded the flattest *overall* response, and they are, to say the least, incredible. If the log-sweep mode on our spectrum analyzer swept up to 30 kHz (instead of stopping at 20 kHz), we would have been able to show how far beyond 20 kHz

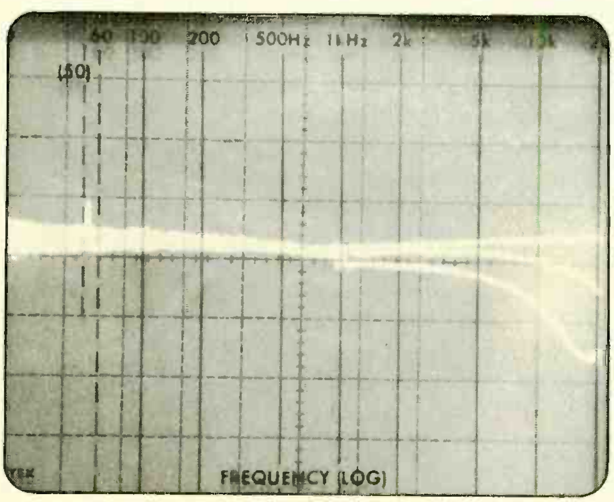


Fig. 1: B.I.C. T-3: Record/play response at 1-7/8 ips (TDK-AD) at three available bias levels.

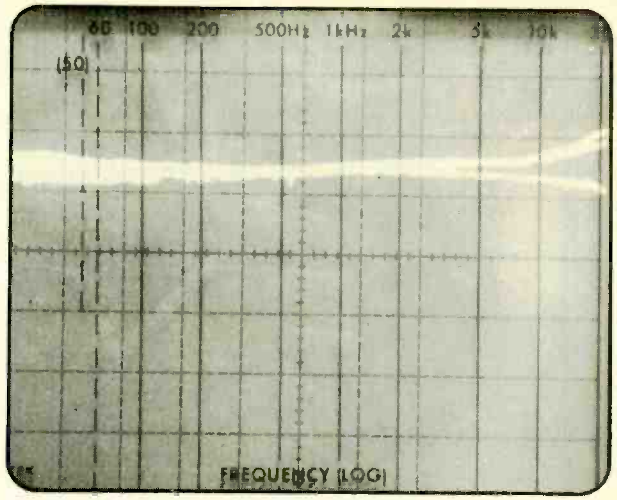


Fig. 2: B.I.C. T-3: Record/play response at 1-7/8 ips (TDK-SA) at three available bias levels.

the response actually extended at the $3\frac{3}{4}$ ips speed.

The designers of this deck—very wisely in our view—chose *not* to utilize the higher speed simply for increased high-end response. (Had they done so they probably could have gotten response out to 40 kHz or better). Perhaps the most important and useful benefit of the higher speed operation turns out to be improved *headroom*, particularly at high frequencies. Figures 5 and 6 illustrate this point most dramatically. Using TDK SA tape, we plotted response from 20 Hz to 20 kHz at 0 dB record level and at −20 dB record level (where one normally measures frequency response of cassette decks) using the slower speed of $1\frac{7}{8}$ ips. Results are shown in Fig. 5 and, as might be expected, the system runs into tape saturation when one tries to record high frequencies at 0 dB recording level (upper trace rolls off above 5 kHz), whereas at the more usual −20 dB record level, response is virtually flat.

Next, with the same sample of tape, we repeated these two response curves using the higher speed of $3\frac{3}{4}$ ips. The results are displayed in Fig. 6 and, even at

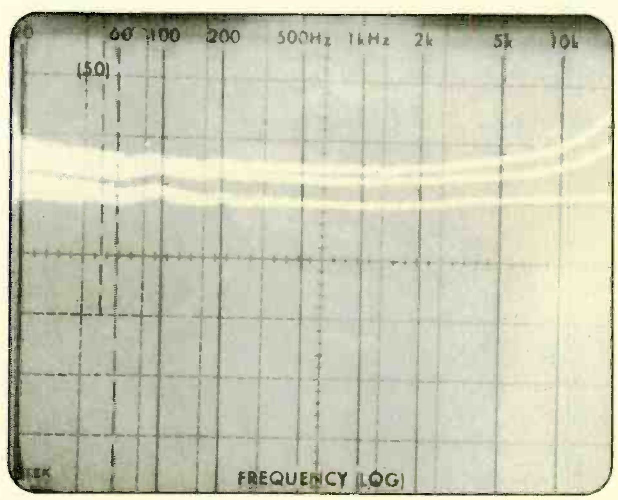


Fig. 3: B.I.C. T-3: Record/play response at 3-3/4 ips (TDK-AD) at three available bias levels.

0 dB record level, response is virtually flat all the way out to 20 kHz.

To round out the picture, we decided to use our spectrum analyzer to examine (or separate) harmonic distortion and noise. The THD figures reported in the Vital Statistics table are, after all, really a single combined reading of THD plus noise. So, we switched the analyzer mode to linear sweep for Figs. 7 and 8 (frequency notations should therefore be ignored in examining the photos) and swept from 0 to 10 kHz *linearly*. First, we used a 1-kHz signal at 0-dB record level and operated the deck at its slower speed. The recorded signal is represented by the tall peak at the left of Fig. 7, while to the right of it are seen the 3rd and 5th harmonic components "read" by the analyzer, as well as the "noise floor."

Using the same tape sample, we repeated the run at 3¾ ips. The resultant 3rd harmonic contribution is obviously lower (relative to the 1-kHz fundamental at the left of the display) than before and there is no evidence of 5th harmonic contribution whatever. The level of the "noise floor" is also obviously much lower.

General Info: Dimensions are: 17$^{15}/_{16}$ inches wide; 6$^{7}/_{8}$ inches high; 10$^{1}/_{8}$ inches deep. Weight is 14.8 pounds. Price is $529.95.

Individual Comment by L.F.: I had read all the material about the advantages of higher-speed operation of cassette decks long before I got my hands on the new B.I.C./Avnet Model T-3 unit, and, intellectually, I understood that faster speed *must* produce better tape performance. It was only after I began measuring and playing with the T-3 that I realized just how *much* better a cassette could work at 3¾ ips, the higher of the two speeds provided by this brilliantly executed product. The Vital Statistics Table accompanying this report tells much of the story, and readers are urged to examine the figures carefully; they are arranged for easy comparison of performance parameters at 1$^{7}/_{8}$ ips versus 3¾ ips.

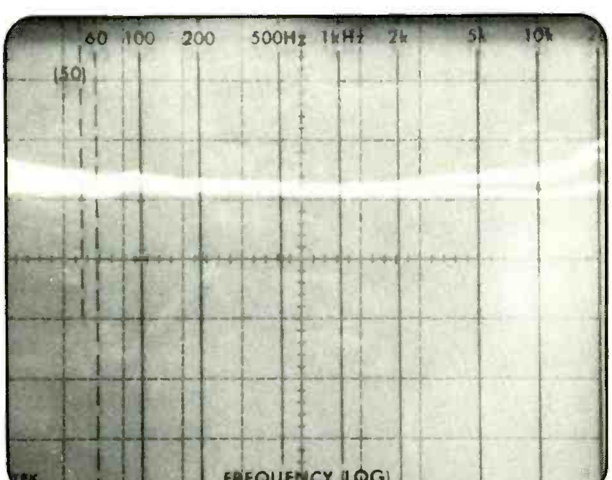


Fig. 4: B.I.C. T-3: Record/play response at 3-3/4 ips (TDK-SA) at three available bias levels.

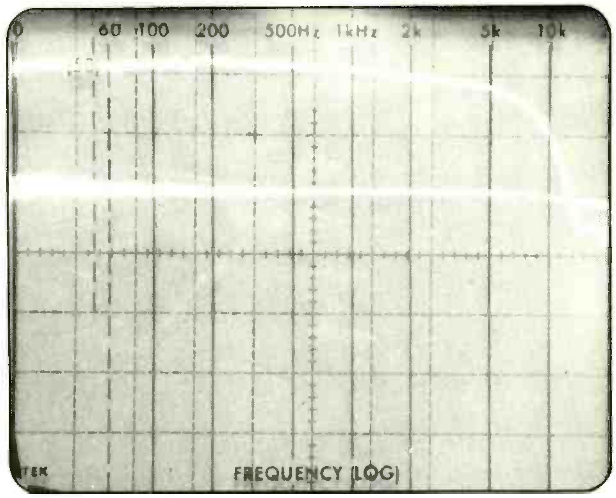


Fig. 5: B.I.C. T-3: Attempting to record at "0 dB" results in high frequency tape saturation at 1-7/8 ips as expected. (Lower trace is at −20 dB, for reference.)

To begin with, let's make one thing clear. This deck is a super-performer even at its standard 1$^{7}/_{8}$ ips speed, and, to my mind, it would be well worth its suggested retail price even if it did not have that higher speed of operation. After all, it isn't every day that you run into a cassette deck that delivers frequency response from 10 Hz to 24 kHz (±3 dB) with 70 microsecond EQ tape (we used TDK type SA tape for all tests involving this EQ setting) or response from 10 Hz to 19 kHz (±3 dB) with "standard" 120 microsecond EQ tape (we used TDK type AD for all tests involving this EQ setting) at its standard 1$^{7}/_{8}$ speed. The wow-and-flutter at this speed was an impressively low 0.045% (WRMS) which is nothing to complain about either. And, at the slow speed, signal-to-noise for the AD and SA tapes (referenced to 3% THD) was 57 dB *without* Dolby (increasing to 65 dB or better with Dolby ON).

But the deck runs away from *all* competition (regardless of price) when one flips that little switch on the front panel to the 3¾ ips setting and begins to repeat the major measurements at this higher speed. Would you believe response from 20 Hz to 23 kHz (±3 dB) using the standard tape? Or how about a signal-to-noise ratio of 69 dB (Dolby on) using the TDK SA samples? Or a record-low wow-and-flutter figure of 0.025% (WRMS)?

Take a look at the "headroom" figures in the Table, and consider what a +11.0 dB figure means in light of the fact that B.I.C./Avnet has calibrated their "0 dB" point on the peak reading meters at 200 nanowebers (Dolby reference level) and *not* at the more typical 185 or 165 nW commonly used as a "0 dB" calibration point by many manufacturers of cassette decks. Talk about recording electronics *headroom!*

As we were coming up with these results, our thoughts turned to "metal particle tape." The higher cassette tape speed of the B.I.C./Avnet T-3 seemed to be delivering all of the benefits claimed for that as yet

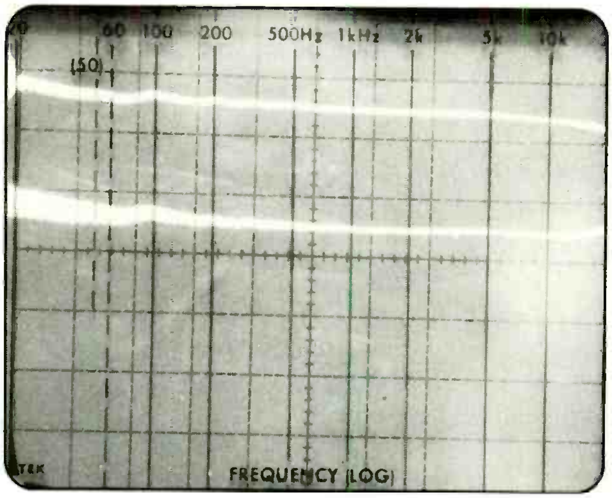


Fig. 6: B.I.C. T-3: At 3-3/4 ips, frequency response of recording made at "0 dB" (upper trace) is virtually as flat as that made at −20 dB.

unavailable tape, and then some. Better headroom at mid and high frequencies, higher recording level capability overall, and even somewhat better signal-to-noise ratio. In other words, *greater dynamic range,* the most important improvement sought for the stereo cassette deck format. Interestingly, the higher consumption of tape at 3¾ ips (a C-90 becomes, effectively, a C-45 with 22.5 minutes of recording time per side) may well be offset by the as yet undetermined higher price of metal particle tape, when it does become available on the marketplace. And of course, erasure is no problem here whereas it is the chief problem facing deck manufacturers who are just now attempting to come up with the first deck models that can utilize the new pure metal particle tapes.

Of course, one could carry the argument even further and speculate on just what kind of performance might be obtained if B.I.C./Avnet were to come up with a two-speed machine that could also handle metal particle tape in the future. The idea boggles the mind altogether!

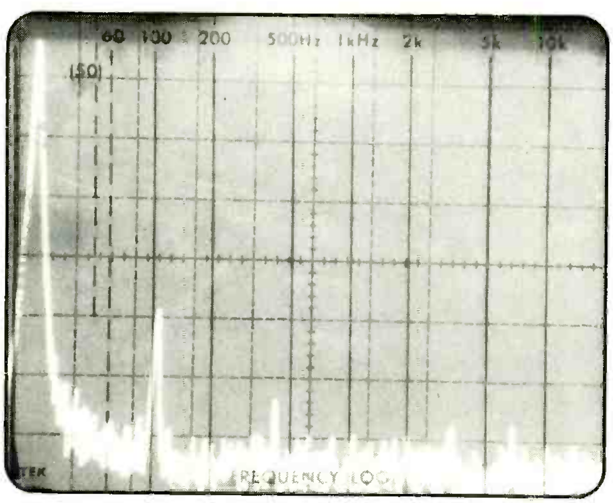


Fig. 7: B.I.C. T-3: Spectrum analysis of 1 kHz signal (peak at left) recorded at 1-7/8 ips separates THD components (3rd, 5th harmonic) from random noise.

There has been much speculation in the audio industry regarding *how* B.I.C./Avnet was able to produce a two-speed deck in the first place. Did they violate Philips license agreements? And if they didn't, can other makers of cassette decks do the same thing? Frankly, I don't really care. At the moment, B.I.C./Avnet is the only company to come up with a two-speed home stereo cassette deck and, operating at its higher speed, there's just nothing around in cassette decks that comes close to it in performance.

Individual Comment by N.E.: Additional testing of the T-3 involved recording from various sources with tapes other than the specific ones used in the lab, including—for the 120 µsec EQ—Memorex MRX3, and for the 70 µsec EQ, Scotch Master III (a ferrichrome). Results were simply outstanding and on A-B tests of source vs. tape, the results at 1⅞ ips were at least the equal of any cassette recording I've yet done (including on some much higher-priced units), while at 3¾ ips, the sound has to be better than anything I have yet got onto a cassette.

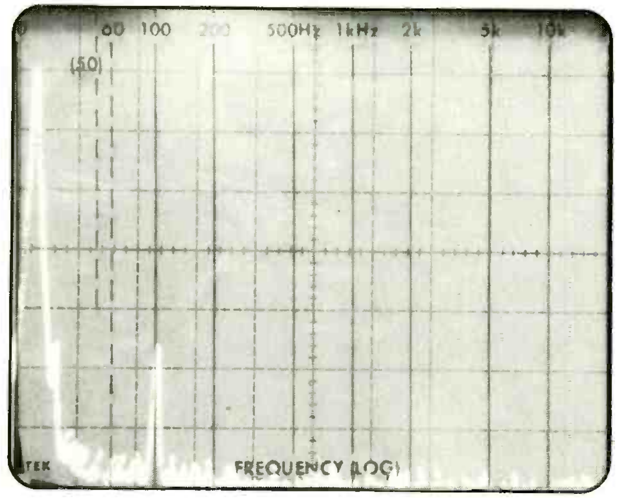


Fig. 8: B.I.C. T-3: When recording speed is increased to 3-3/4 ips, THD components are reduced (no evidence of 5th order harmonic) and noise level is substantially lower. Compare this display with that of Fig. 7.

Obviously, the T-3 does not have some of the features found on other models—such as input mixing or fast-buttoning. The latter feature does lend a "professional" feel to a recorder and, by contrast with those models that do have solenoid-operated "feather-touch" transport controls, the T-3 has an operating "feel" that is reminiscent of cassette decks of a few years ago. But then, look at its low price—low by any standards on today's market, and even the more remarkable in light of the splendid performance of the unit. And although it is a single motor machine, the T-3 has superb wow and flutter measurements.

The 3¾ ips speed raises some interesting questions.

One obvious problem is the fact that at that speed, a cassette recorder uses twice as much tape for a given length of program vis-a-vis a standard-speed cassette deck. Related to this is the fact that there are no pre-recorded cassettes at $3\frac{3}{4}$ ips. Whether there will be depends of course on what the recording industry decides to do. We have, in other words, another case of the hardware preceding the software—which, historically speaking, has always been an iffy situation for the long haul. This question of course came up with the Elcaset format (about which we hear practically nothing these days). However, the Elcaset—which ran at $3\frac{3}{4}$ ips—required an entirely new size of cassette as well as a new speed. The B.I.C. idea is perfectly compatible in terms of existing standard cassettes, so at least on a major part of the problem, there is a major area of accord, without the fear of "sudden obsolescence" that often is raised by changes in a given format.

Another question implied here is that concerning the possibility of *reducing* tape speed in order to realize greater economy in the use of tape but, at the same time, holding performance levels to what we now have at the standard ($1\frac{7}{8}$ ips) speed. I do not mean to deny the obvious accomplishment represented by the T-3, but wouldn't it be really sensational to be able to achieve comparable results at half, rather than at twice, a given tape speed?

B.I.C. T-3 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Frequency response,		
70 µsec-EQ tape		
1-7/8 ips	±3 dB, 25 Hz to 19 kHz	±3 dB, 10 Hz to 24 kHz
3-3/4 ips	±3 dB, 25 Hz to 22 kHz	±3 dB, 16 Hz to 24 kHz
120 µsec.-EQ tape		
1-7/8 ips	NA	±3 dB, 10 Hz to 19 kHz
3-3/4 ips	NA	±3 dB, 20 Hz to 23 kHz
Signal-to-noise ("A" wtd, re: 3% THD)		
70 µsec.-EQ tape		
1-7/8 ips (Dolby off)	55 dB	57 dB
3-3/4 ips (Dolby off)	58 dB	61 dB
1-7/8 ips (Dolby on)	63 dB	65 dB
3-3/4 ips (Dolby on)	67 dB	69 dB
120 µsec.-EQ tape		
1-7/8 ips (Dolby off)	NA	57 dB
3-3/4 ips (Dolby off)	NA	59 dB
1-7/8 ips Dolby on)	NA	65.5
3-3/4 ips (Dolby on)	NA	67 dB
Wow and flutter (WRMS/DIN 45507)		
1-7/8 ips	0.05 / 0.10%	0.045 / 0.09%
3-3/4 ips	0.035 / 0.09%	0.025 / 0.06%
THD (0 VU, 70 µsec.-EQ tape		
1-7/8 ips	1.8%	1.3%
3-3/4 ips	1.5%	1.0%
THD (0 VU, 120 µsec.-EQ tape)		
1-7/8 ips	NA	1.3%
3-3/4 ips	NA	0.8%
Record level for 3% THD		
70 µsec.-EQ tape		
1-7/8 ips	NA	+4.0 dB
3-3/4 ips	NA	+7.5 dB
120 µsec.-EQ tape		
1-7/8 ips	NA	+5.5 dB
3-3/4 ips	NA	+11.0 dB
Fast-wind time, C-60	approx. 45 seconds	45 seconds
Mic input sensitivity	NA	0.16 mV
Line input sensitivity	200 mV*	35 mV**
Line output level	2.0 V	1.9 V
Headphone output level	NA	317 mV/8 ohms
Power consumption	35 W	24 W

*At manufacturer's recommended mid setting. **At maximum gain setting.

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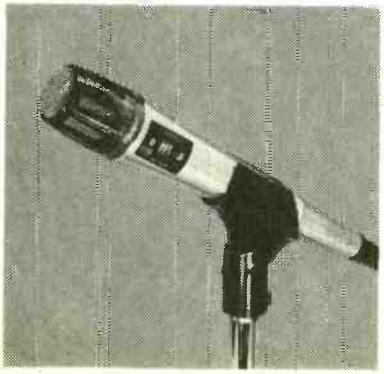

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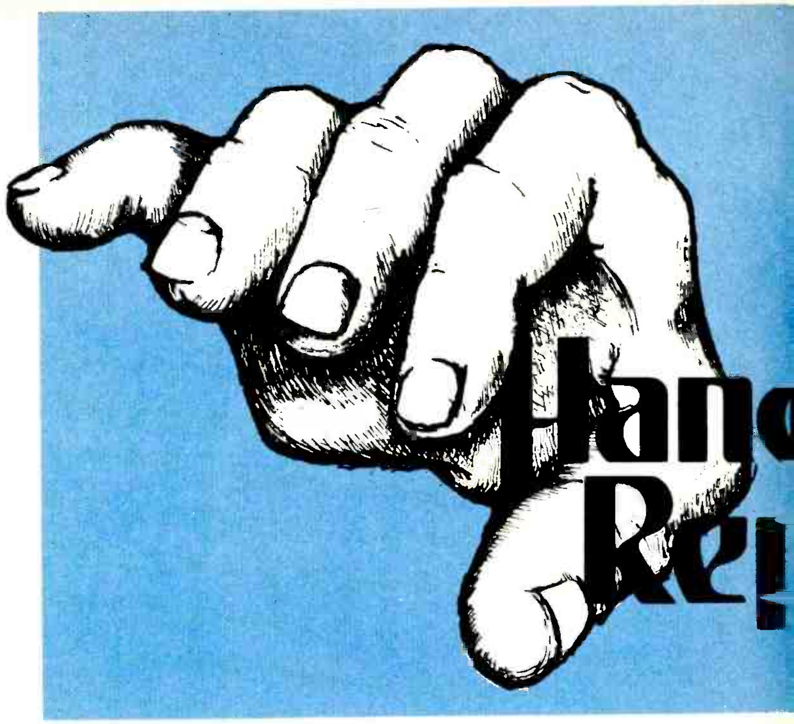
Akai Pro-1000 Open-Reel Recorder



By Jim Ford and John Murphy

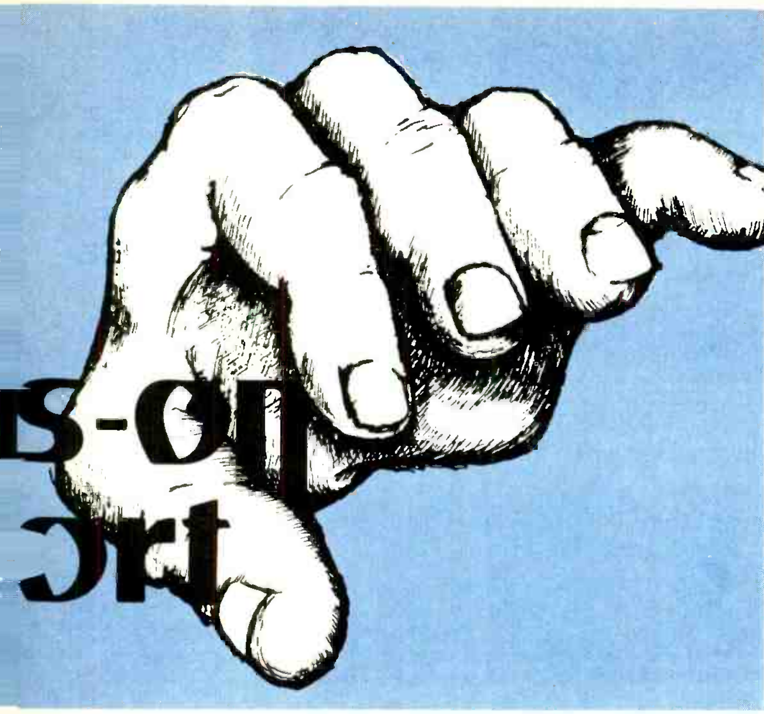
General Description: The Akai Pro-1000 is a two-track open-reel recorder with operating speeds of 15 ips, 7½ ips and 3¾ ips. It can handle 10½-inch reels and has provisions for mixing four inputs (mic, line, or any combination) into stereo. The latter feature makes it particularly well suited for use as a stereo mastering recorder in conjunction with a four-track (sync) recorder. In addition, the unit is equipped with a quarter-track playback head and has connections for using external noise reduction (such as Dolby or dbx).

The transport features a closed-loop double capstan system powered by an AC servo motor. The reels are driven by two eddy current motors. There is an automatic play feature whereby the machine will automatically switch from either of the fast wind modes into play whenever a piece of sensing tape crosses the detector. Although not as easy to use as the more popular "memory rewind," (which simply stops a rewinding tape when the index counter gets to zero) the auto-play feature is much more powerful in that you can index as many points on a tape as you desire.

A front panel switch allows the meters to indicate VU levels, peak levels or bias level. In the peak level mode the meters have a very fast rise time and a slow decay. Because of the slow decay, the meters tend to hold the peaks and exhibit reduced motion, thereby making them easier on the eyes. A hinged head cover makes the heads available for cleaning or alignment.

There are separate input level controls for each of the four input channels and a single master level control for the overall two-track record level. Each of these has a pre-set control which makes it easy to return to a predetermined setting. To the left of the input level controls are a pair of ganged output level controls. Input one is automatically assigned to the left channel, input four is automatically assigned to the right channel, and inputs three and four can be panned anywhere from left to right. Each input channel has a switch for selecting either line, mic or mic attenuated 20 dB. Just to the left of the mic/line selector switches is a headphone jack and its associated level control.

The unit is constructed in two pieces with the transport and transport controls in one section and the level controls, meters and electronics in the other. Each module is built into a heavy-duty vinyl case complete with heavy latching covers and carrying handles. The latter makes the unit readily transportable. The modules can be used either side by side or stacked one on top of the other. Feet on the bottom and back of



each module allow them to be used either vertically or horizontally. The two pieces interconnect by way of three heavy cables, each about 30 inches long. In addition to the usual tape motion controls there is a "cue" switch to override the tape lifters during fast wind operations. This is a useful feature for locating beginnings and endings of program selections. Although tape speed is selected by push buttons on the transport module, the equalizer has to be set for the appropriate tape speed on the other module. Thus, tape speed change on the Pro-1000 involves two separate switching operations.

The electronics module contains front panel bias and record EQ controls for each channel. Akai gives instructions for setting these by referring to a table in the operator's manual. The table lists some twenty tapes by brand name and number and gives bias and EQ settings for each.

The rear panel of the electronics module contains the mic inputs (¼-inch phone plugs) and line inputs/outputs (RCA phone plugs). The connections for external noise reduction are also on this panel. On back of the transport module are a remote control socket, two AC convenience outlets, a ground connector and a switch for selecting 50 Hz or 60 Hz line frequency.

The Akai Pro-1000 is priced at $1,995.

Field Test: After stacking the two units vertically, we made the necessary interconnections, and found this to be a simple task. First we were interested in hearing an A-B test between what goes in and what comes out. This will let our ears tell us something about the fidelity of the recording and playback process. We used a direct disc record for our music source and a set of studio monitor speakers to make the comparison. While in the record mode we switched between input and output (source/tape) and strained our ears to hear any changes in noise, distortion and tonal quality; the results were pleasing. There was the addition of tape noise but this is to be expected. There was no audible distortion but there was a slight tonal change which was not a major problem, and, we suspect, a simple alignment would solve it.

Next we connected the unit to a Tascam Model 5 mixer. Our source this time was a four-track multitrack recorder and the program material was built one track at a time with a few vocals (if you can call our singing vocals) and a few guitar tracks. The mixing was simple and the two-track worked well. It should be noted at this time that the mixer was used to mix the four tracks down to two; however, with the built-in mixer on the tape machine the mixer could have been eliminated. This would work and would save the financial cost of a mixer although there would be many limitations as to E.Q., reverb, panning, special effects, etc. Only for the most simple mixing of four-track to two-track would we consider eliminating the use of a good mixing console.

The unit has mic inputs so we tried them with a standard vocal mic. Each input has a 20 dB pad, and this worked fine to protect against overload due to high input levels.

Since we had produced such a fine "master" tape we decided to do some editing. In this area there could be some additional features that would help the professional get his job done. Some type of edit mode where the take-up reel spills the tape for dumping unwanted footage into the trash can would speed up the operation. In order to do editing the machine would most likely have to be laid on its back.

Another item that might be improved was the lack of individual record selectors for each track. Although the machine is meant for two-track mastering, the majority of serious recording enthusiasts will sometimes want to record on only one channel.

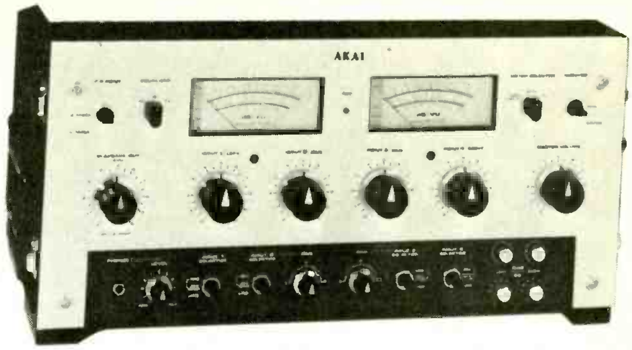
One solid feature that some other manufacturers might look at was that in the fast wind or rewind mode when the tape lifters drop against the heads the output is attenuated by 15 dB. If the engineer is looking for a specific "take" on a master tape he would put the machine into a fast wind mode and drop the tape against the heads. Most tape recorders would put out a full signal and this is a dangerous situation. Because of the high tape speed, all of the frequencies are raised several octaves, and the result is possibly a blown tweeter in the studio monitor speakers. Besides being potentially dangerous to speakers, it is also a nuisance to your ears (and nerves). Akai did its homework on this problem and has an excellent feature.

Overall the machine performed very well and would work very nicely in any semi-pro studio. The quality of construction seemed good and we would expect good reliability.

Lab Test: With the individual and master input level controls at maximum, the required input levels for 0 VU indication were 83 mV (-19 dBV, ref: .775V) and 0.3 mV (-68 dBV) for line and mic inputs, respectively. The line output level at 0 VU was 0.9 V (+1.3 dBV). These are typical levels for tape machines in the "semi-pro" market. Interfacing with most other semi-pro and hi-fi equipment should present no problem.

Playback response was checked through both the 2-track and 4-track playback heads at 15 ips and 7½ ips. At 15 ips the 2-track playback response was within ±2 dB from 30 Hz to 15 kHz for the left channel and ±2.5 dB for the right channel over the same frequency range. Playback response at 7½ ips was checked next. The 2-track head was ±3.5 dB from 50 Hz to 15 kHz for the left channel and ±3 dB for the right. The 4-track head responded nearly the same at both speeds.

Our next test was overall record/playback frequency response. For this and all subsequent tests Ampex 406 tape was used. Since bias and EQ settings were not given for Ampex tapes, Ampex biasing instructions were followed (i.e., the tape was "overbiased" by 1.5 dB at 0 VU with 10 kHz). ("Overbiasing" by 1.5 dB with 10 kHz implies that the bias was first adjusted for the maximum playback level of the 10 kHz tone. Then the bias level was increased until the playback level of the tone dropped 1.5 dB. This should result in a good compromise between the high frequency output level and distribution.) The EQ control was set for flat playback of a 10 kHz tone (4.5/L and 5/R). At 15 ips with the test tone recorded at 0 VU level, the REC/PB response was within ±1 dB from 50 Hz to 20 kHz for the left channel and ±.75 dB for the right. For 7½ ips, test signals were recorded at −10 VU and response was within the same limits as for 15 ips. The record/play bandwidth (−3 dB points) was approximately 28 Hz to 31 kHz for 15 ips at 0 VU and 18 Hz to 28 kHz for 7½ ips at −10 VU. (The bandwidth for



Section containing gain controls, meters and electronics.

3¾ ips at −15 VU was about 23 Hz to 14.6 kHz.) This is excellent frequency response performance.

For the electronics only, noise was −60.2 dB in the left channel and −61.3 dB in the right. (All noise levels are referenced to 0 VU.) When we played back bulk erased tape, the noise was about −57 dB at 15 ips and −59 at 7½ ips. It seems curious that the noise level actually decreased at the slower speed. When we measured overall record/playback signal-to-noise ratios we found that this number also improved at the slower speed. Record/play noise was about −52 dB at 15 ips and −55 dB at 7½ ips. Signal-to-noise ratios referenced to the 3% distortion level were 59.4 dB and 60.1 dB for left and right channels, respectively, at 15 ips, and 61.4 dB/62.9 dB for L/R channels at 7½ ips. Akai specifies only a total S/N ratio of 60 dB.

With a 1 kHz tone recorded at 0 VU, total distortion was .5% in either channel at 15 ips. At 7½ ips, distortion was .65%/.63% for L/R channels. We measured distortion through just the electronics at about 0.1%. Akai states that distortion is less than 1% with 1 kHz at 0 VU. Using a 10-kHz test tone, distortion increased to about 1% at 15 ips, 2.4% at 7½ ips, and 0.2% for electronics alone. The 3% distortion point was reached typically with input levels of +7 VU at either 15 ips or 7½ ips. Onset of tape saturation (with 1 kHz) was observed on an oscilloscope trace at a record level of approximately +11 VU. This should provide adequate headroom for signal peaks.

Track-to-track phase jitter at 20 kHz was within ±15 degrees. Crosstalk at 1 kHz (not specified by Akai) was −50 dB. Wow and flutter was 0.025% at 15 ips, and 0.04% at 7½ ips.

Notably, the operator's manual supplied with the machine contains a good discussion of the use of the meters in the peak level mode.

Section containing transport and transport controls.

Conclusion: This is a good machine that will perform very well for the semi-pro market. The approach of the operational features are somewhat different, and for many cases this machine would provide the needed controls that are not found on most basic two tracks. The specifications and construction are good, and, while the price of $1,995 may seem a little higher than the prices for similar machines, the Pro-1000 offers additional controls and features that make it worthwhile.

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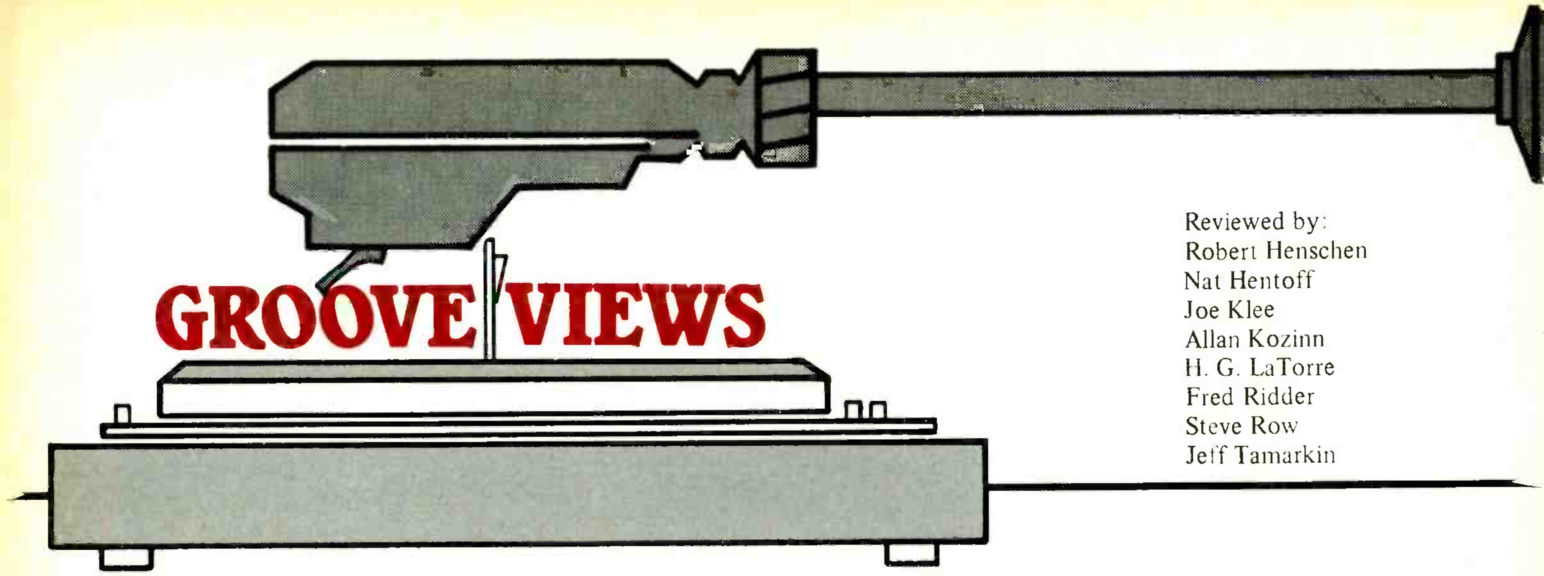

GROOVE VIEWS

Reviewed by:
Robert Henschen
Nat Hentoff
Joe Klee
Allan Kozinn
H. G. LaTorre
Fred Ridder
Steve Row
Jeff Tamarkin

POPULAR

THE KINKS: ***Misfits.*** [Ray Davies, producer; Steve Waldman, engineer; recorded at Konk Studios, London, England. Mastered at Masterdisk, New York, N.Y.] Arista AB 4167.

Performance: **Konsistent of Kinks kwality rekording**
Recording: **Refreshingly sparse**

The title *Misfits* is an appropriate one for a Kinks album, as they've never really enjoyed a secure position in the rock sweepstakes. Having been together as long as their British Invasion contemporaries, the Stones and the Who, and longer than the likes of Fleetwood Mac, it would seem as though the Kinks should either have earned heroic status by now, or have disappeared entirely. But instead, they've constantly remained at a limbo stage of popularity, certainly above cult level, but falling far short of superstardom. And perhaps *Misfits* explains why—the Kinks are a band which refuses to compromise or give in to trends.

The Kinks have been bathing in some renewed respect lately as precursors of the new wave movement, and in some ways this has affected their sound. Like the Stones' *Some Girls, Misfits* is a throwback to the basics. Yet on the other hand, the Kinks continue to evolve here, and lyrically, this is the most complex and touching Kinks LP in many years.

The cut "Live Life" is structurally representative of lead Kink Ray Davies' approach to the whole album. Beginning with a lone electric guitar on one channel, followed quickly by Mick Avory's drums in the other, the song melds together piece by piece. Finally it erupts into a full-blown rocker, with Ray and his brother, lead guitarist Dave Davies, shouting the "live life" chorus back and forth into the final fadeout, which occurs just in time to save the piece from becoming overdrawn. Lyrically, the song exhibits a new form of Ray Davies' optimistic, yet cautiously cynical attitude towards his reality. Davies exhorts that, although life does get depressing, "Don't believe all you read in the headlines." Instead, he says, "Live life see it through/ Carry on it's all you can do." And carry on he does, taking his band through transformations one more time.

As usual, Davies' concerns are erratic and obscure. He touches on subjects somewhat removed from mainstream pop, from hayfever to transvestism and racial chauvinism. Davies is not afraid to stand alone in his opinions, never has been, and unfortunately it is this very strength of artistic character which has been the group's commercial weakness.

The production of *Misfits* is sparse, relying only on the group's basic instrumentation and little additional trickery. The horns and female chorus of past LPs are gone, and that move is a healthy one, allowing the band's development into simple, yet inventive musicians in a contemporary vein to emerge. The mix is clean, abundant with shiny acoustic guitars, crisp lead guitars and percussion, and sensuous organ patterns. The entire record utilizes a haunting and effective reverb, which gives the vocals especially a distant and primitive feel.

The Kinks have always been the least understood of the first-wave British bands. No doubt part of their problem is their lack of accessibility to the generally fickle record-buying masses. The

THE KINKS: Refusing to compromise; continuing to evolve

Kinks' music demands involvement, and additionally, their rejection of technical embellishments over the years has prevented the acquisition of many younger listeners who do not want to make the commitment the Kinks require. And that's a shame, because the Kinks still have more to say, in a more unique way, than any of their descendents, detractors or imitators. J.T.

HEART: ***Dog & Butterfly.*** [Mike Flicker, Heart and Michael Fisher, producers; Mike Flicker and Rick Keefer, engineers; recorded at Sea-West Studios in Seattle, Wa. and "Cook With Fire" "live" by Artisan Mobile Recording at Centroplex Coliseum in Memphis, Tn.] Portrait FR 35555.

Performance: **Dog tries, butter flies**
Recording: **Beauty and the beast**

HEART: Dog bites, butterfly soars

One side of this album is called "Dog" and the other "Butterfly," a critic's straight line if ever there was one. And indeed, the "Dog" side nearly yelps with mangy hard rock, while "Butterfly" unfolds a prettier side of Heart and gives the group more room to soar.

"Cook With Fire" is a "live" rocker electrified by Ann Wilson's razor vocals and the twin guitars of Roger Fisher and Howard Leese, with all feedback intended—exemplary of their raw power in concert. But "High Time" and "Hi jinx" are less impressive, sharing a penchant for underdone rock basics with the side-ending "Straight On." The latter relies more on Ann's great voice than on compositional depth, and there's not quite enough sex and sensationalism to make up for the melodic weaknesses. In all fairness, however, side one's unadorned rock format has its share of primitive appeal.

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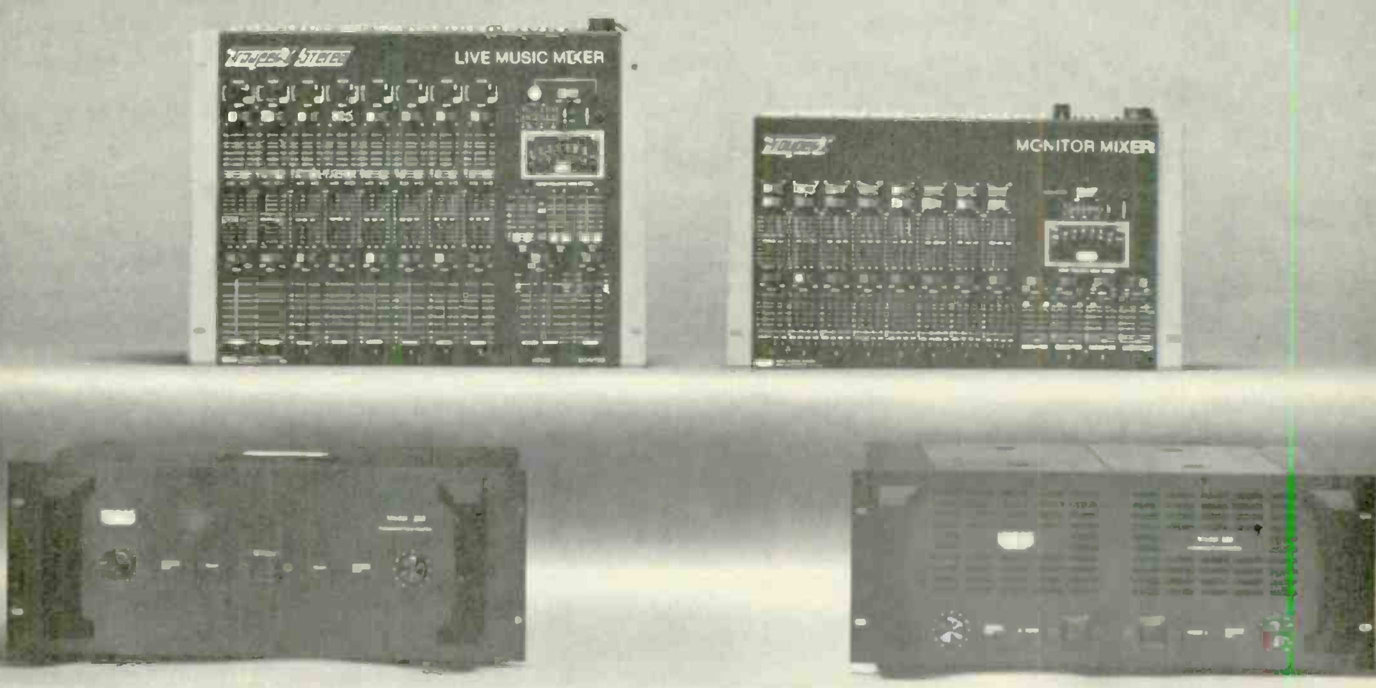

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The second side's "Dog and Butterfly" title cut goes acoustic, with Nancy Wilson supporting her sister on 12-string guitar and harmony vocals. Even more intimate is Ann's mournful solo voice intro to "Lighter Touch," which builds to a powerful lament with great, moaning electric guitar fills and excellent orchestration. Nancy gets her too-infrequent vocal feature on "Nada One," beautiful acoustic guitar, flute, and highly poetic dream imagery.

Now in the classic form, Heart climaxes the album with "Mistral Wind." The song starts slow and heavy, with Heart at their implicative best, then explodes into potent, brooding rock before evaporating mysteriously into lovely musical mist. Finishing with all the strength of *Little Queen, Dog & Butterfly* is another credible effort from an emerging rock superband. R.H.

CHEAP TRICK: *Heaven Tonight.* [Tom Werman, producer; Gary Ladinsky and Mike Beiriger, engineers; recorded at Record Plant, Los Angeles, Sound City, Van Nuys, Ca.; dates unlisted.] Epic JE 35312.

Performance: **Not so cheap . . .**
Recording: **But tricky**

Rick Nielsen in his goofy baseball cap, Bun E. Carlos looking quite the frazzled, seersucker accountant, and two star-sized glamour boys in Robin Zander and Tom Petersson. An unusual bunch to be sure, but one of the most entertaining and innovative R&R bands on today's scene. Their uncanny combination of Fifties yuks and Eighties fusion makes Cheap Trick a pop-rock hybrid, tongue-in-cheek though they may be, able to thrive on several listening levels.

"Surrender" is a case in point. Taking a slightly punkish teen rebellion stance, dealing with communicable diseases and parental lust, this opening cut easily skirts New Wave monotony and comes up with a hilarious chorus that's just catchy as hell. Most other songs here are similar marvels of boffo humor, hard electricity, and insidiously hook-filled melody. On one level, in fact, "Auf Wiedersehen," "Surrender," "Takin' Me Back" and "On The Radio" could all qualify for Top 40 exposure. On another level, though, each of these gems functions as a kind of stand-up cerebral comedy.

Strains of The Beatles and ELO may be faintly recognizable in some of these fast and funny refrains, and there's a Twenties borrowing from Rudy Vallee for the punchy "How Are You." An ironic tint to Rick Nielsen's writing yields urbane laughs à la older 10cc, and "On The radio" cops almost directly from The Tubes' "What Do You Want From Life." But these allusions to other bands and bygone styles just add to the sarcastic merriment, become part of Cheap Trick's varied sound arsenal. "Stiff Competition" goes so far as to blatantly mock the music biz rivalry syndrome. The entire melange sounds refreshingly original.

Heaven Tonight suffers only mild jet lag during portions of "California Man" and "High Roller," but they're absolutely forgotten by the time Cheap Trick is

CHEAP TRICK: Tongue-in-cheek

The Unsung Heroes of 1978—Track One

While the vast majority of music critics will rate the top popular albums for the year in lists containing the likes of Boston, the Rolling Stones, Linda Ronstadt and the Who, some of us are content to seek out the splendid "little" albums—those featuring relatively little known artists or those receiving little critical attention during the year.

I have found several records that fit into this category from among the thousands that the record companies released during the year. My own selections, upon review, lean toward softer sounds, filled with identifiable melodies, and thus you are not going to find a single new wave group in the lot.

Starting with male vocalists, and proceeding on from there, my favorites for the year include:

BRUCE COCKBURN: *Circles In The Stream.* (True North ILTA 9475).

Cockburn is an extraordinarily gifted singer, songwriter and guitarist who has everything necessary to succeed Gordon Lightfoot as Canada's next great musical resource. The double album (recorded during a 1977 concert in Toronto), contains sixteen songs written during the past seven years and is extremely well recorded and the material is filled with some *pure* poetry, burning metaphors and interesting images.

STEPHEN WHYNOTT: *From Philly To Tablas.* (Music Is Medicine MIM 9001).

Whynott is a transplanted New Englander now living in western Washington state who has been working on this album since 1973. He described it as "a completely orchestrated version of my most private dreams and journeys through the corn belts and mountain peaks of this century," a thought that some might find pretentious. But from the opening chords of "Retreat Suite" to the whimsical "Oh Boy, I've Won a Contest At Last," Whynott and his colleagues (including oboist Michael Kamen, formerly of the New York Rock Ensemble) create a fine musical package whose only fault is that some of the songs are too short.

BONNIE KOLOC: *Wild And Recluse.* (Epic 35254).

This is the kind of album that, with the right exposure and promotion, could move the performer out of cult status and into the big time. Backed by the Roomful of Blues band and produced by Joel Dorn, *Wild and Recluse* is a tasteful collection of ten songs firmly based in blues and folk roots. Ms. Koloc has a terrific boice that can growl with the blues or weave a haunting melody through a ballad. Her treatment of "Back Home Again in Indiana" is chilling, and her reading of the Lennon-McCartney song "Golden Slumbers" is simply splendid.

MARY O'HARA: *LIVE AT THE ROYAL FESTIVAL HALL.* (Chrysalis CHR 1159).

This disc marks the return to music by a young Irish woman who was considered to be a singer without peer twenty years ago. After the death of her husband, American poet Richard Selig fifteen months after they were married, she entered an abbey and remained there for the next dozen years. The concert recorded here was a sellout, and she once again is drawing sizeable followings at appearances in England, Scotland and Ireland. The album is a mixture of traditional melodies and popular songs, and her voice is a remarkably pure instrument. Among her covers are "Morning Has Broken," "Bridge Over Troubled Water," "Scarlet Ribbons" and "Tapestry." We haven't had a voice like this since Joan Baez's earlier recordings.

THE NEW BRUBECK QUARTET. *Live At Montreux.*
(Tomato TOM 7018).

This album shows that Dave Brubeck has lost none of his sophisticated approach to the jazz piano, and it also shows that what used to be known as Two Generations of Brubeck (Dave and his sons, bassist Chris, drummer Dan and keyboardist Darius) can play some outstanding ensemble jazz. The interplay among the musicians is remarkable, particularly on two Brubeck standards, "Raggy Waltz" and "Brandenburg Gate."

GO, FEATURING STOMU YAMASHTA. *Go—Live From Paris.* (Island ISLD 10).

This album is from a 1976 concert in Paris, and the lineup is most impressive: Steve Winwood, Michael Shrieve, Yamashta and Klaus Schulze (the nucleus of the group), joined by Al DiMeola, Pat Thrall, Jerome Rimson, Brother James and Karen Freidman. Winwood's singing is a highlight of the album's fourteen tracks, but the stars are many, and the guitar work of DiMeola and Thrall is simply grand. The music is a mixture of jazz and rock, with a large quantity of electronic embellishment added.

RONNIE MONTROSE: *Open Fire.* (Warner Bros. BSK 3134).

This is the rock guitar lover's album for the year, better than Roy Buchanan, Richard Wagner, Lee Ritenour and David Spinozza put together. The album was produced by Edgar Winter and is a showcase for the not insignificant talent of Montrose, who headed a rock group of that name a few years back. He employs a variety of playing styles here, and his guitar always is up front. One recognizable cut: "Town Without Pity."

This leaves me with one final selection, which I have reserved for my favorite record of the year. It may or may not fall into the category of being a splendid "little" album (with at least two hit singles extracted, it undoubtedly became a gold record this year), but it was not one by a supergroup or star of the first order.

GERRY RAFFERTY: *City To City.* (United Artists UA LA840G).

Nearly sixty minutes of pure music and rhythm here, highlighted by the riveting single "Baker Street." But "Baker Street" just happened to get the exposure first; there's not really a bad song on the entire album. "The Ark," with its traditional-sounding introduction; "Maggie's Rag," and "Right Down the Line" all are great songs. The arrangements are skilled, and the backup instrumental work by guitarist Nigel Jenkins and sax player Rafael Ravenscroft, among others, is very good.

Steve Row

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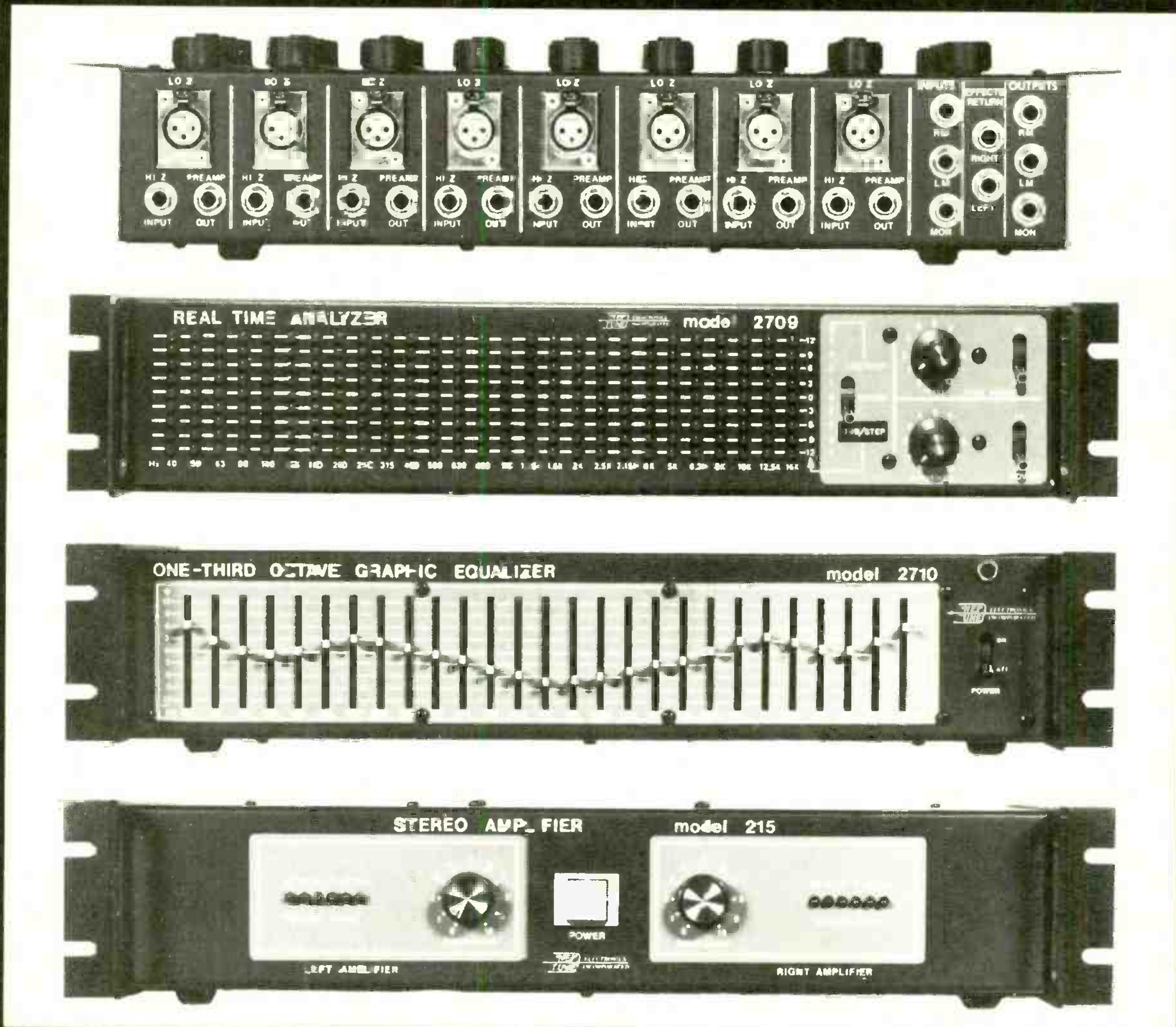

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LEON REDBONE: *Champagne Charlie.* [Joel Dorn, producer; Hal Willner, associate producer; Vince McGarry, engineer; recorded at Regent Sound Studios, New York, N.Y.] Warner Bros. BSK 3165.

LEON REDBONE: Curiouser and curiouser

Performance: **Understated, loving, curiously witty**

Recording: **Good, but too muffled**

Leon Redbone's most recent album is a curious thing, but then perhaps all Leon Redbone albums are curious things. This one, however, is curious for another reason. His earlier releases featured only a handful of easily recognizable songs from many decades ago among a larger number of less-well-known old chestnuts. He gradually became popular (thanks particularly to an appearance on "Saturday Night Live"), and now he has released an album that is dominated by old standards. Isn't this a little bit backward? Perhaps not, but the new release, *Champagne Charlie,* must be calculated to win a larger audience with battle-tested standards before he decides to go back to resurrecting unknown musical antiques.

The titles of the songs on this release will strike a responsive chord in the hearts of many listeners: "Please Don't Talk About Me When I'm Gone," "Sweet Sue," "Alabama Jubilee," "Big Bad Bill," "Yearning," and "T.B. Blues." The title cut is becoming Redbone's theme song for his stage performances.

Because most of the material is well-known, the best evaluation of the album must be directed at its sound. Redbone and producer Joel Dorn score some high marks and some demerits.

Redbone's voice is one of the most engaging musical instruments on record, and rarely has he used such a variety of vocal sounds before. The black blues voice, with its quick fade at the end of phrases, accurate dialect and slight rasp, can be heard on only half of the songs here. On "Please Don't Talk About Me," Redbone sings in a high tremolo and throws in a near-yodel over a hummed chorus. The near-yodel becomes the real thing in "The One Rose." In "Yearning," Redbone sounds old, tired, strained, and just right.

The record begs comparison to Ry Cooder's *Jazz*, reviewed here in the September 1978 issue, and in light of that comparison, *Champagne Charlie* comes out second best. This may be the fault of the recording, because where Cooder's music is crystalline, bright and "up-front," Redbone's has a strangely muffled one-dimensional sound that certainly suits the age of the material, but

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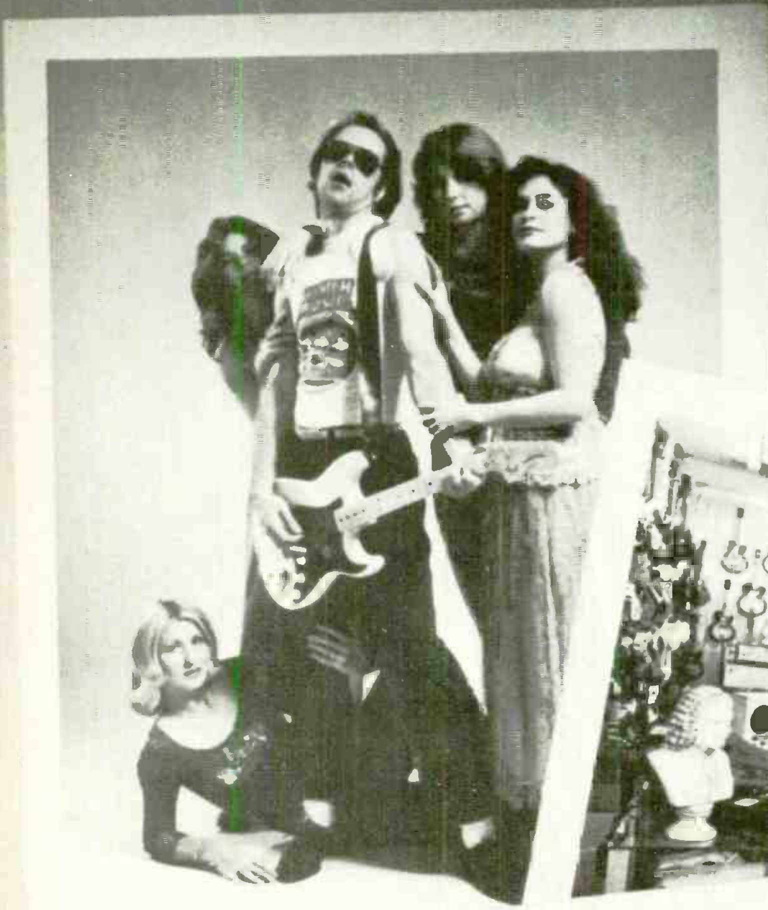

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that also makes the product more of a museum curio than a vibrant evocation of a past age.

The arrangements are faithful, to be sure, with trumpet, violins, clarinet, tuba and ukelele prominently featured. Redbone has not messed around with the rhythms. Some strange instrumentation crops up—note how the occarina sounds like a calliope in "Alabama Jubilee," and how the bass harmonica played by Chris Whitely in "Yearning" matches the oom-pah sound of Jonathan Dorn's tuba. Even the vocal choruses sound as if they came off an old 78.

The sound of the album can be attributed in part to the instrumentation, of course. Whereas Cooder's album emphasized acoustic guitar and percussive instruments, Redbone's is built on acoustic guitar and horns, winds and strings. The ensemble is small (he is using players from a small Toronto band called the Sloth Band), the playing is tight, but the effect is somewhat denser than that of Cooder.

This may be a small quarrel, however, with what is otherwise another interesting and well-executed album by one of popular American music's most curious enigmas. Perhaps we all should quit wondering just who this Leon Redbone really is and start to enjoy his unique music. S.R.

SYNERGY: *Cords.* [Larry Fast, producer, engineer, programmer; Pete Sobel, associate producer; recorded at the Synergy Studio, and at House of Music, West Orange, N.J.; mixed at Studio B, House of Music, West Orange, N.J.] Passport PB6000.

Performance: **Meticulously crafted**
Recording: **Transparent (literally)**

Synthesizer albums are a very risky proposition. Both the artist and the listener must first come to grips with the fundamental nature of the synthesizer as an instrument and decide whether they conceive of it as an electronic imitation of "real" instruments or as a musical instrument in its own right. The artist must choose his musical material to be consistant with his conception of the instrument, the choice ranging from cover versions of popular tunes on the one hand to "serious" synthesizer compositions on the other. Beyond that, the artist must have a clear vision of what he wants the final recording to sound like before he commits the first notes to tape since he is literally responsible for every sound on the record.

One of the more successful practitioners of the synthesizer album is Larry Fast, a synthesizer wizard who records under the name "Synergy." The third and latest Synergy album, *Cords,* shares several virtues with its predecessors. Fast has the good sense to stay away from familiar tunes in favor of original compositions which are intricately structured, but which always seem to avoid being overly serious (i.e. dull). Part of the credit for this must go to Fast's sense of humor; the listener is constantly treated to interesting twists and quirky little effects in the middle of otherwise classically structured compositions. A sense of humor is also evident in the choice of titles, which can be remarkably important to non-vocal music. Fast acknowledges the assistance of Peter Gabriel (in whose band Fast plays

LARRY FAST/SYNERGY: Meticulous

synthesizers) in devising titles like "Disruption in World Communications" and "On Presuming to be Modern" on *Cords,* which is itself a deliberate pun.

One of Fast's greatest strengths is his ability to evoke the textures of conventional instruments without making any attempt at slavish imitation of their sound. All the sounds on his albums are produced electronically, and one never loses sight of that because the sounds never pretend to be anything but electronic. Yet somehow, Fast manages to blend them in such a way that the overall effect of the recording is almost orchestral. The compositions themselves depend heavily on mood and texture, being somewhat akin to classical tone poems, so that it is easy to listen casually to the album as a kind of background music. But for the listener who has the interest and the energy to put into serious listening, *Cords* reveals itself to be meticulously crafted with a wealth of details that will hold one's interest through repeated listenings.

On several of the tracks on *Cords,* Fast is joined by Pete Sobel, who performs on a guitar synthesizer system designed by Russ Hamm. This use of guitar synthesizer serves to add a more human touch to the sound of the record because the guitar has a much greater potential for personal expression or individual touch than a keyboard instrument, even when the guitar is only being used as a control device for synthesizer circuitry. "Sketches of Mythical Beasts" and "Phobos and Deimos Go to Mars" are particularly striking demonstrations of the distinctively different sound of the guitar synthesizer.

Two semi-random observations about *Cords* before closing: First, the record is pressed on clear vinyl, which, besides looking great and making it much more difficult to cue an individual selection, may result in a quieter pressing. (Direct-to-disc enthusiasts have claimed for several years that the black pigments used in most record compounds contributes to surface noise.) And second, the Aphex Aural Exciter was used on at least part of this record, which seems like a more natural application for the device than on conventional pop or rock records where its "liver-than-live" quality must be considered a distortion or a coloration since it certainly isn't fidelity.

Make no mistake: *Cords* is *not* rock and roll, but it *is* consistantly interesting music from a young master of the synthesizer. F.R.

KISS GENE SIMMONS: *Gene Simmons.* [Sean Delaney, Gene Simmons, producers; Mike Stone, engineer; recorded at The Manor, Oxford, England; Cherokee Studios, Los Angeles, Ca.; Blue Rock Studio, New York, N.Y. Mixed at Trident, London, England.] Casablanca NBLP-7120.

KISS ACE FREHLEY: *Ace Frehley.* [Eddie Kramer, Ace Frehley, producers; Eddie Kramer, Rob Freeman, engineers; recorded at the "Mansion," Sharon, Ct.] Casablanca NBLP-7121.

KISS PETER CRISS: *Peter Criss.* [Vini Poncia, producer; Bob Schaper, engineer; recorded at Sunset Sound, Hollywood, Ca., except "I Can't Stop The Rain," "Rock Me Baby," "Kiss The Girl Goodbye," "Easy Thing,"- basic tracks recorded at Electric Lady Studios, New York, N.Y. and engineered by Mike Stone. Additional recording at Sound Labs, Inc. Hollywood and The Burbank Studios, Los Angeles, Ca.] Casablanca NBLP-7122.

KISS PAUL STANLEY: *Paul Stanley.* [Paul Stanley, producer; Jeff Glixman,

The Unsung Heroes of 1978—Track Two

Each year, dozens of great albums are released that nobody ever hears about. Oh sure, now and then a Gerry Rafferty will suddenly break out of the pack and go for the gold. But there's nothing more frustrating for us true-blue music fans than to watch *Saturday Night Fever* hit triple platinum while some highly original new talent has his/her creative life threatened with extinction.

Fraser & Debolt, Biff Rose, Sopwith Camel, David Pomeranz, Essra Mohawk, Milton Nascimento, Nick Drake. Largely unknown, these are just a few of the sleepers from yesteryear that still find their way onto this reviewer's turntable with astonishing regularity. Not so very long ago, of course, most pop music fans couldn't have identified Barry, Robin and Maurice Gibb as The Bee Gees, and names like Dolly Parton and Bruce Springsteen were equally unknown. So what's in a name?

Modern Recording wants to remove the cloak of anonymity from ten of 1978's most neglected artists. Lack of space prevents us from espousing the delights of Karen Alexander, David Pritchard, Pezband, Renee Armand, Bim, Gary Peacock, and other soon-to-be-forgotten greats, so this list will have to suffice. Presenting the Unsung Heroes (and Heroines) of 1978.

KATE BUSH: *The Kick Inside.* (EMI SW-17003).

Actually *singing* the role of Emily Bronte's character Cathy Earnshaw, Kate Bush took her song version of "Wuthering Heights" to number one on British pop charts. Her voice is eerie, her lyrics intense, and these tunes are sensuously unique.

CARLA BLEY BAND: *European Tour 1977.* (Watt/8).

This is a hilarious collection of mind-blowing jazz, including a 19-minute collage of international anthems called "Spangled Banner Minor And Other Patriotic Songs," played in wild keys and crazy combinations.

DEAN FRIEDMAN: *"Well, Well," said the Rocking Chair.* (Lifesong JZ 35361).

This total unknown sounds a bit like Neil Sedaka, but don't let that bother you. Friedman is an amazingly bright songwriter, putting music to lovers' conversations ("Lucky Stars"), afterhours diner dialogue ("The Deli Song"), and non-animate apartment decor like table tops and coffee cups ("Rocking Chair"). Weird, but absolutely contagious and likeable pop melodies.

KAYAK: *Starlight Dancer.* (Janus JXS 7034).

This Dutch group packs *Starlight Dancer* with potential AM radio hits. Accessible pop-rock harmonies, progressive keyboards, and excellent production make for a sound somewhere between Abba and Yes.

PASSPORT: *Sky Blue.* (Atlantic SD 19177).

The German equivalent of Weather Report, Passport plays electric jazz-rock of consistent melodic invention. Saxophonist Klaus Doldinger plays a spacy soprano atop a heady mix of electric keyboards, guitar, and modern percussion. If you've recently disc-overed Doldinger, check out the 1973 Warner Bros. issue that debuted this group, called simply *Passport.*

THE CARS: *The Cars.* (Elektra 6E-135).

Hard rock with the immediacy of New Wave, but offering more imaginative songs and fewer anti-establishment cliches. "You're All I've Got Tonight" became an underground dark horse for radio play, but other cuts are just as potent.

BUDDY HOLLY/THE CRICKETS: *20 Golden Greats.* (MCA-3040).

Forget the Hollywood film hoopla; Buddy Holly *really* lives on this timely reissue by MCA, a collection of early rock classics that will serve as a revelation to those who think "Words of Love" and "Not Fade Away" began with The Beatles and Stones. Tremendous songs, some very familiar.

PAUL PARRISH: *Song For A Young Girl.* (ABC AA 1031).

This mellow singer-songwriter gets full orchestration on his belated return to recording. Emphasis on poetry transforms pretty tunes into seriously good compositions like "Hoedown."

ELVIS COSTELLO: *This Year's Model.* (Columbia JC 35331).

If you're still shying from the puzzling New Wave, here's a good place to jump in. Elvis is less punk, more rock 'n' roll, and singing honest songs like he really means them.

CHARLIE MARIANO: *October.* (Inner City 1024).

Now living in Europe and playing in Eberhard Weber's Colours, reedman Charlie Mariano here reemerges on the contemporary jazz scene with a strong set of modern instrumentals.

Robert Henschen

The Editor's Choices

The list of albums that you see below was compiled from recordings each of which attained an incredible degree of anonymity during the past year-and-a-half (approximately). These albums were ignored by just about "anyone who is anyone," but mostly (much to the chagrin of these artists' families) by the record-buying public. So, to allow these artists to see their names in print, some undoubtedly for the first time, we present here some choices that we feel should have received plaudits for technical and/or musical accomplishments.

PROFESSOR LONGHAIR: *Live On The Queen Mary.* (Harvest SW-11790)

EDDIE PALMIERI: *Lacumi, Macumba, Voodoo.* (Epic JE35523)

DUCKS DELUXE: *Don't Mind Rockin' Tonite.* (RCA AFl-3025)

CHARLIE AINLEY: *Too Much Is Not Enough.* (Nemperor JZ 35080)

SONNY RHODES: *I Don't Want My Blues Colored Bright.* (Advent 2808. Available by mail only from Advent Productions, P.O. Box 635, La Habra, Ca. 90631, telephone 213-449-2992.)

WILKO JOHNSON: *Wilko Johnson's Solid Senders.* (Virgin V 2105. Available by mail only from Virgin Records, distributed by Jem Records, P.O. Box 362, 3619 Kennedy Rd., South Plainfield, N.J. 07080, telephone 201-753-6100 or 18615 Topham St., Reseda, Ca. 91335, telephone 213-996-6754.)

JORGE SANTANA: *Jorge Santana.* (Tomato TOM-7020)

FENTON ROBINSON: *I Hear Some Blues Downstairs.* (Alligator AL 4710)

H.G.L.

Paul Grupp, engineers; recorded at Electric Lady Studios, New York, N.Y. The Record Plant, Los Angeles; The Village Recorder, Los Angeles, Ca. Mixed at Trident Studios, London, England.] Casablanca NBLP-7123.

Performances: **Standard and substandard Kiss to better than expected.**
Recordings: **All good**

Oh, how the critics are going to hate these records. Having to review *one* Kiss album is not a task most rockcritics would look forward to. The group's reputation among the opinion makers of the biz is not, after all, the same one that is shared by the millions who lap up every commercial offering which bears this group's name, especially their records. Having to sit through *four* Kiss-related LPs, however, seemed like it could be the ultimate critic's nightmare, one that might require a few tranquilizers, or at least a stiff drink, to make it through.

Having finally summoned up the courage, and foregoing the sedatives at that, it was cheering to discover that these guys aren't *that* bad after all. They're not the future of rock'n'roll, to coin a phrase, but on the basis of these four simultaneously released solo records, it *can* be stated that there are certainly worse records being recorded today. However, none of these four are convincing enough to have me sending in my bucks for a membership card in the Kiss Army, either.

The four LPs by Gene Simmons, Ace Frehley, Peter Criss and Paul Stanley, were conceived as independent productions (none of the members appear on any of the other's albums), but there are some common denominators. The most striking similarity between them is a technical brilliance on the part of the engineers involved. The sound on all four is uniformly excellent, and despite the varying directions taken by the performers, there is a unity which ties the four recordings together as one multi-set Kiss package. While the individual statements of the group's members are pronounced for the first time, it is also obvious just who is contributing what to the total phenomenon that is Kiss.

Gene Simmon's LP is the most successful and intriguing of the four in that it is the furthest departure from the trademark Kiss sound, showing that there are possibilities available to the group should they ever decide to be something else besides hard-rock clowns. *Gene Simmons* mixes diverse production values within its concept without veering too far from its intended course at any point along the way. Some tracks use the basic rock mix of guitars, bass, and drums without overdubs. Others are overproduced to a point just short of absurdity. "Living In Sin," for example, uses sound effects sparingly enough for the idea to enhance the piece, while the LP closer, the classic, "When You Wish Upon A Star," is sweetened to where it becomes an unintentional parody. Simmons' LP is also the only one of the bunch sporting famous names—Cher, Bob Seger, Helen Reddy, and Donna Summer all appear—but none are featured prominently enough to matter. This is a funny record, whether or not it was meant to be.

The *Ace Frehley* and *Paul Stanley* collections do not fare as well because they are less adventurous and not nearly as funny. However, hardcore Kiss fans will likely feel more at home with the basic blitz-rock approach which fills

KISS: Solo efforts could be on the right track

up most of these two discs. This is the music for which Kiss is noted, and despite the attempts by all four to show a sensitive side by including acoustically-oriented songs, the hard rock of Frehley and Criss seems more honest, if less appealing intellectually.

Paul Stanley begins with an artsy, Genesis-like intro with heavy emphasis on the aforementioned rich acoustics and cymbals in the mix. The intro, part of "Tonight You Belong To Me," then segues into a full-blast blues-based rocker. With the guitar soloing to the right, the drums enter with a pounding 4/4 on the left, and it stays that way. In fact, it stays that way for the rest of the album, which occasionally lapses into run-of-the-mill heavy metal excesses and loses direction. All in all, this is common dime-a-dozen hard rock which Kiss fans will no doubt eat up.

Frehley also takes the storm trooper approach, leaning on fuzzed-out guitar solos front and center. "What's On Your Mind" has the makings of an FM pop hit, and is engineered with sensible restraint. The highlight of *Ace Frehley,* though, is a song by one-time Argent guitarist Russ Ballard, "New York Groove." Not surprisingly, the cut is reminiscent of early Argent in ways, but also contains snippets of sound-bringing to mind acts as diverse as Queen and Devo.

Peter Criss shows the most promise of all as a soloist, however. His R&B-tinged album is a tasteful effort, and as such, is probably the only one of these four I'd consider coming back to more than a few times.

Criss has integrated a variety of styles, from mushy ballad/artsy-orchestral meanderings ("Easy Thing") to righteous, high-energy boogie-rock ("Rock Me Baby"). Criss is held back by his limited vocal range, but he makes the best of his grittiness by enhancing it with reverbs and backing choruses. The highlight of *Peter Criss* for me is the remake of the 60's soul nugget, "Tossin' And Turnin'."

It's hard to say if any of the experimentation of these solo outings will rub off on Kiss, but it couldn't hurt any. While the four records themselves are less than spectacular, the musicians have said that these projects will strengthen them as a group. If these solo LPs were conceived as nothing more than another scheme to put a few million more dollars into the hands of those behind the incredible Kiss marketing scam, nothing musically valuable will have been established with them. If they were done to prove that there is more then hype to Kiss, then they're on the right track. But I'll still need a lot more convincing. J.T.

LINDA RONSTADT: ***Living In The U.S.A.*** [Peter Asher, producer; Val Garay, engineer; assisted by George Ybarra; recorded at The Sound Factory, Los Angeles, Ca. Mastered at the Mastering Lab by Doug Sax.] Asylum 6E-155.

Performance: **Calculated**
Recording: **Sterile**

The word most often used by critics to describe Linda Ronstadt's concerts is calculated. Every step of her way is meticulously planned out to the most minute detail, so that one musical hair thrown out of place would seem likely to cause the poor girl to short circuit. Similarly, sterile is the word most often used in summing up Ronstadt's recordings. And for good reasons.

Linda Ronstadt is a performer who has fashioned a thriving career around the fact that she is predictable and takes no chances. Her arrangements are never any further than a key change away from the original songs she covers, and her voice, perfect as it may be, is totally devoid of emotion. Ronstadt may be the ultimate in 1970's studio mentality, which is to say that she is a product of the technology rather than an artist in control of it. This recording is so wholesomely clean that Marie Osmond could listen to it. It's also about as exciting as a hunk of Velveeta cheese.

A case in point is Ronstadt's whitewashing of Elvis Costello's "Allison" on this record. Out of Costello's material, "Allison" is the obvious choice for Ronstadt to record. It is Costello's prettiest ballad. But underneath the tender music is a vindicative lyric. While Costello's biting tones are frightening in their intense rendering, Ronstadt has trouble even sounding convincing in some passages.

This distance from the music she is putting out carries over into the backing tracks as well. Ronstadt has employed a cast consisting of several of the most respected west coast studio musicians for this LP. From the sound of it she must have been paying them by the hour, because this is not exactly an inspired bunch. They too, like the lyrics she sings, are buried behind *the voice* to serve as some insignificant backdrop. The few instances where one of these musicians is brought up in the mix to where it counts are so rare that each one stands out as a precious moment. Especially noteworthy is the subtle steel guitar playing of Dan Dugmore.

There really is no way anyone can argue about the actual quality of the recorded sound of *Living In The U.S.A.* Basically, the disc is flawless as far as its production. However, that becomes the real flaw which plagues this and other Ronstadt albums, in that it is *too* perfect for rock 'n' roll. Ronstadt and her crew leave no room for the individual to feel anything, to discover, to identify with some part or another. They simply record a record, filter out all substance, and stamp it safe for mass consumption. Then they ship it to the apple pie section of your local record mart, where millions eat it up loyally. It's sweet all right, and inviting, but organic it's not. J.T.

JAZZ

EDDIE HENDERSON: ***Mahal.*** [Skip Drinkwater, producer; Larkin Arnold, executive producer; Jim Gaines and Alan Suddeth, engineers, recorded at Wally Heider recording in San Francisco, Ca.] Capitol SW 11846.

Performance: **Mwandishi lives**
Recording: **Heider fidelity**

Once upon a time, not so long ago but before there were the Headhunters or the Vocoder voice synthesizer, Herbie Hancock had a band called Mwandishi. The band owed a lot to the Miles Davis classic quintet of the pre-"Bitch's Brew" era. The idea was improvisation over rhythmic figures that were often vamp-like in quality, but the improvisation was melodic and tonal and accessible to the average ears. Among the horn players in the band were Julian Preister (one of the most consistently underrated trombonists around then and today), Bennie Maupin (a rising young multi-reed talent at the time who since has flowered into a mature artist) and a trumpet and flugelhorn virtuoso whose name was (and still is) Eddie Henderson.

Eddie has the lushness of tone, the lyric sensitivity and the subtlety which distinguished Miles' playing up until his fling into outer space music, never to return perhaps. They're all here, back together again under Eddie's name this time and on Capitol. This, plus top cats like Hubert Laws on flute and Mtume on second piano and Latin percussion. A lot of them brought along music, too. Herbie Hancock wrote a very Mwandishi chart "Butterfly" and there a couple of tunes by Mtume and one by Bennie Maupin but the two cuts that get to me are the very lyrical "Emotions" and "Ecstacy," both of which are Eddie Henderson originals.

As far as the engineering is concerned, someday they're going to build a monument to Wally Heider. His studios

EDDIE HENDERSON: Lushness, sensitivity and subtlety–together again

have turned out so much good sound that they deserve the recognition they are only just beginning to get. It doesn't sound like a gimmicked recording job to me which means that if it is they did a good job of making it sound natural. I have no objection to judicious and tasteful overdubbing and sweetening if it isn't done in such a way that it becomes annoyingly obvious.

Okay, so Eddie Henderson isn't exactly an original. He grew up listening to Miles and it shows in his playing. But he's not an ape either. He's doing his own thing with the tools he has and if Miles used some of these same tools before him, who can blame either of them? And anyway, Miles Davis doesn't play jazz any longer (by his own admission), so why not? J.K.

ALBERTA HUNTER: *Remember My Name.* [John Hammond, producer; Frank Laico and Bob Waller, engineers; recorded at CBS Recording Studios, New York City, N.Y.] Columbia CS 35553.

Performance: **Fine old wine**
Recording: **Okay but uneven**

In her song "I Got Myself A Workin' Man," Alberta Hunter sings about her boyfriend who even though "he's old and very thin, there's plenty a' good tunes left in an old violin." The same could be said of the singer herself. At 80 plus she's still going strong. Between her first recording for Black Swan (1921) accompanied by Fletcher Henderson's Novelty Orchestra and this latest LP produced by John Hammond for Columbia, this lady has worked with the best of them. In 1924 she took the name of Josephine Beatty to record with the Red Onion Jazz Babies (who included Louis Armstrong and Sidney Bechet among others). She has recorded with accompanists such as Fats Waller and Eubie Blake. She played in Europe and was part of the London cast of *Show Boat* along with Mabel Mercer, Paul Robeson and Sir Cedric Hardwicke. She had a hit on Paramount with her song "Downhearted Blues," as did two other blues singers, Bessie Smith and Mildred Bailey. With all this experience behind her, Alberta Hunter turned her back on show business and went into the nursing profession in 1957. She came out of retirement in 1961 to record LPs for Prestige in their Bluesville series and for Riverside in their Living Legends series. After that... nothing till now.

Alberta Hunter still sings like a woman possessed—possessed by the blues which she told liner annotator Jim Bickhart, "means to me what milk does to a baby." She also has a good deal of instrumental help from Budd Johnson on clarinet, Vic Dickenson on trombone, Gerald Cook on piano, Al Hall on bass and drummers Connie Kay and Jackie Williams. The honors, however, go to Adolphus "Doc" Cheatham whose trumpet accompanies her as superbly as his saxophone playing did Ma Rainey's blues singing in the 1920s.

I'm not as happy with Wally Richar-

essences of swing: Basie, Bags and Barnes

By Nat Hentoff

For some years, the Basie band, while generating formidable power, has been more like a machine than a group of warm, potentially spontaneous spirits. One unkind French critic described it as sounding like "civil service swing." However, in two new sets, the Count and his troops have been regenerated by the addition of one man. The guest is Milt "Bags" Jackson, a vibist whose sense of jazz time is so strong, clear, and compelling that he probably could make a military band swing.

Although Bags has recorded with Basie in a small combo context, this is the first time he has joined on record with the full band. *Milt Jackson & Count Basie & The Big Band, Vols, 1 and 2,* Pablo). The result is one of the most buoyant, sometimes exhilarating, jazz sessions of the year—in terms of getting back to infectious basics. The playing here is the very definition of swinging. And since both Basie and Bags are expert in the uses of silence to dramatize flowing time, their combined presence here leads to an especially lithe, resilient beat that in turn makes the big band come on with a more supple, springy thrust than has characterized it for a long time.

There are blues, ballads, and medium-and-up-tempo swingers—and considerably more Basie on spare, spearing *solo* piano than on most of his previous big band sets. The sound is full, clear, vibrant. I expect the main engineering problem was keeping one's hands steady at the controls while the rest of the body moved to the music.

From the huge, jubilant forces of the Basie band and Milt Jackson to guitarist George Barnes and a rhythm section is somewhat like moving from a celebratory mass meeting to a small seminar. *George Barnes Plays So Good* (Concord Jazz) may not make you shout out loud, as the Basie-Bags encounter will, but there are abundant pleasures on a smaller scale.

The late George Barnes was a guitarist with an exceptionally glowing, singing sound and the kind of clarity of line-making that had its roots in such earlier lyrical improvisers on the instrument as Dick McDonough and George Van Eps. Barnes also was a player of quick, urbane wit; and underneath it all, as Alec Wilder has pointed out, he was "a true romanticist." Much of his best recorded work in the 1970s was with trumpeter Ruby Braff—a kindred passionate melodist. But this set is George Barnes' most fulfilling under his own name.

Consisting entirely of standards—from "I'm Coming Viriginia" to a subtle, penetrating transformation of "St. Louis Blues"—the album is a particular delight for the luminous illustrations it provides of the improvising process. It *is* a seminar, both in swinging and in the improvisatory evolution of melodic and harmonic designs. As is Concord's estimable custom, the engineering is superb. There is a lot of presence, but without any imbalance in the parts. And I've seldom heard a truer guitar sound on record. This set, by the way, underlines the value of Concord Jazz. It's not going to sell much, but it will be indispensable to those who do buy it.

MILT JACKSON/COUNT BASIE: *Milt Jackson & Count Basie & The Big, Band, Vols. 1 and 2.* [Norman Granz, producer; Val Valentin, engineer.] Pablo 2310-822, 2310-823.

GEORGE BARNES: *George Barnes Plays So Good.* [Carl Jefferson, producer; Phil Edwards, engineer.] Concord Jazz CJ-67.

son's amplified guitar. Electric guitar goes fine with the modern blues singers and blues/rock crossovers but I don't think it's in keeping with the other things happening on this record.

This album is supposed to be the original soundtrack recording of the new Robert Altman film "Remember My Name" for which Alberta Hunter wrote the music. If Hollywood finally has had the sense to do a film with a musical background of ten tunes by a classic blues singer with a top notch band really blowing in back of her, it's a sign of maturity that I didn't expect from glitter city. What I wonder is how many people will concentrate on the film itself when the music is this great?

The recording job is passable but not up to the best that CBS can do, and have done in the past. Players leap out of nowhere for a fill in rather than being heard in their proper perspective. Also on some cuts Al Hall's bass is so dominant it's annoying—not Al's fault but the engineers.

Yet if Alberta Hunter could triumph over the horn-on-the-wall acoustic recordings she can surely put enough of value into any modern day recording to make up for the minor technical difficulties.

J.K.

THE LEE KONITZ NONET: ***The Lee Konitz Nonet.*** [Hank O'Neal, producer; Fred Miller, engineer; recorded at Downtown Sound, New York, N.Y.] Chiaroscuro CR 186.

Performance: **In the style of Miles**
Recording: **Cut and dried**

I wish I could be more enthusiastic about the music on this album. It's certainly good music, well played, with some particularly fine work from pianist Benny Aronov and the brass section of Burt Collins, John Eckert and Jimmy Knepper, but I've heard this sort of thing before. Let's face it, any nine-piece band that includes a brass section with a tuba, a two-man saxophone section basically alto and baritone, and a three-piece rhythm section is going to remind the informed listener of the Miles Davis-Gil Evans Birth Of The Cool unit on Capitol. The similarity is further heightened by the fact that this band revolves around Lee Konitz who played alto sax on the Miles Davis sides and is basically playing the same way today he played then. Another is the fact that a lot of the arrangements are by Sy Johnson whose writing in and out of the Charles Mingus band has exhibited a strong Gil Evans influence. Yet if you can just put on the record and sit back to enjoy it, there's a lot to enjoy here.

My particular favorite is "Sometimes I'm Happy" based on the version of the tune that Lester Young recorded for Keynote Records in 1943. Not only is Lester's tenor solo reproduced in the arrangement but also Slam Stewart's bass solo from the same recording. Another plus is Lee Konitz' beautiful reading of Tad Dameron's ballad, "If You Could See Me Now," delightfully set off by a boppish baritone solo by Ronnie Cuber.

Chiaroscuro has done better in regard to sound on other records. True, the spectrum of the nonet from Burt Collins' piccolo trumpet to Sam Burtis' tuba is well recorded and comes through nicely but it's a dry sound without much presence. I'm not looking for rocky echo chamber sound, but I think a little bit more "live" sound would have helped.

Yet, there are moments like the duet between Knobby Totah on bass and Sam Burtis on tuba in "Sometimes I'm Happy" that make me feel very guilty for saying anything critical at all about the album.

J.K.

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MAYNARD FERGUSON: ***Carnival.*** [Maynard Ferguson and George Butler, producers; Mike Delugg, engineer at Mediasound, New York, N.Y.; Stan Tonkel, engineer at CBS 30th Street Studios, New York, N.Y.] Columbia JC 35480.

Performance: **Opportunistic pizzazz**
Recording: **Crisp highlights**

More of the same from MF's horn—pizzazz and more pizzazz. Maynard Ferguson has led his sometimes exciting big band into the rock-jazz era with no uncertain flair, but such moves continue to prove less innovative than trendy. Refurbishing saleable pop hits with contemporary funk-disco beats and souped-up jazz arrangements, this flashy trumpeter frequently subverts his potential for genuine artistry with ostentatious high-note showmanship.

Ferguson has seldom been more opportunistic than on the theme from "Battlestar Galactica"; the album coordinated, apparently, for simultaneous release with the TV series' highly rated debut. Astral schmaltz mixes with a danceable beat and sensationalistic brass to create teen-level excitement. Even more biz-minded are Ferguson's insipid cover versions of both Gerry Rafferty's "Baker Street" and Earth Wind & Fire's "Fantasy."

Weather Report's awesome "Birdland" somehow survives the Midas touch, a delightful composition even in these overindulgent hands. Big horn flourishes and seething rhythms, though too snappy on the chorus, are tightly performed by the whole ensemble and the engineering is refreshingly crisp. More Zawinul influences crop up on the opening "M.F. Carnival," a cut that begins with fine Spanish guitar and superb blues runs by Ferguson—he *can* still play when he wants. A village whistle melody with jungle percussion gears the carnival up for a joyous sax head, but Maynard's eventual high notes debilitate the number into fashion consciousness, transporting the celebration from sunny holiday to slick suburban disco.

An effort *has* been made to play some straight jazz on two standards, with "Over the Rainbow" limping home to Kansas on a high-blowing waste of trumpet glitter. But "Stella By Starlight" is masterfully performed and recorded, with dark clusters of big band sax sectionwork brushing against muted brass. Almost predictably, this is the only chart written by an MF orchestra outsider, arranger Slide Hampton. More of this, and less of that, would bring this big band roaring back to respectability. R.H.

JEAN-LUC PONTY: ***Cosmic Messenger.*** [Jean-Luc Ponty, producer; Ed Thacker, engineer; recorded at Cherokee Studios, Hollywood and Chateau Recorders, North Hollywood, Ca.] Atlantic SD 19189.

Performance: **Simpler, heavier**
Recording: **Fission harnessed**

MAYNARD FERGUSON: Opportunistic

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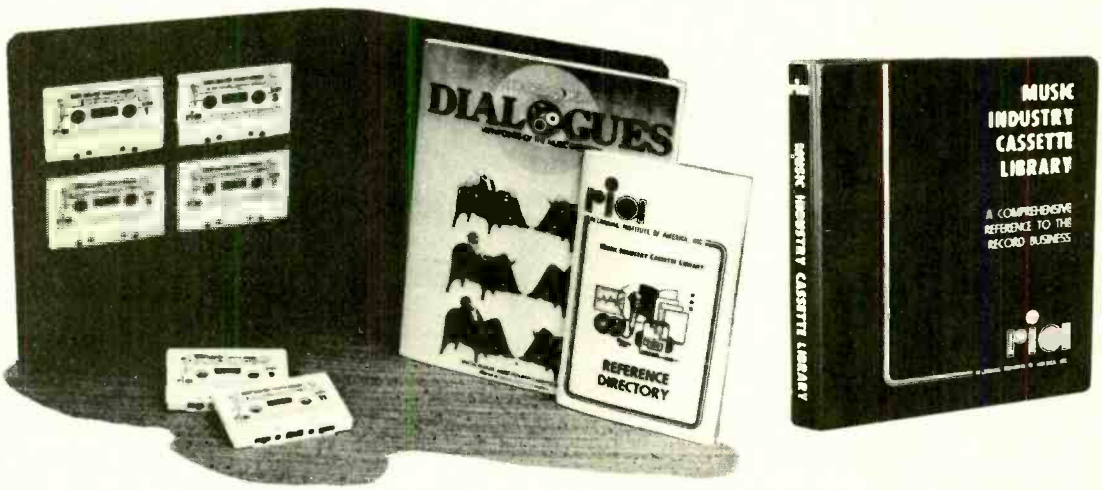

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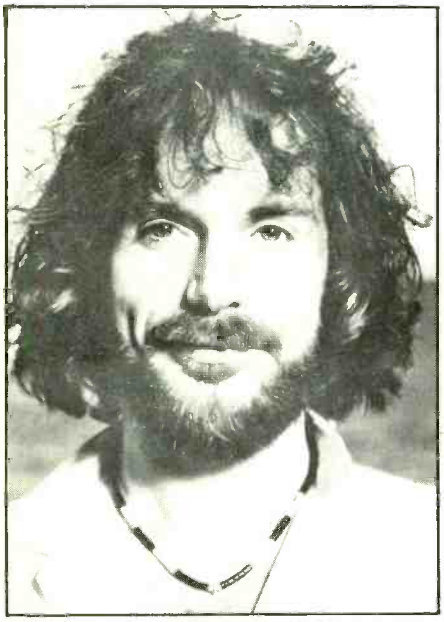
Maybe this sounds like typical fusion from the French giant of the electric fiddle, but it's virtually unflawed and intermittently fantastic. Ponty has been able to keep his band intact and touring for long periods of time, and they've grown accustomed to his pace. Keyboardist Allan Zavod (once with Maynard Ferguson, of all people) and Ralphe Armstrong (from a late period Mahavishnu Orchestra) are still around, though guitarist Daryl Stuermer has left to replace Steve Hackett in Genesis. Ponty has gone to a two-guitar lineup in Stuermer's wake, a move hinted at by Allan Holdsworth's presence on *Enigmatic Ocean.*

Compositionally, Jean-Luc seems to have reached his peak two years ago on



JEAN-LUC PONTY: Experiment, please

the more ambitious *Imaginary Voyage*. The cuts here are more consistently directed than before perhaps, but there are fewer melodic inventions and virtually no European undertones. Then again, cuts like "Cosmic Messenger," "Don't Let The World Pass You By" and "Ethereal Mood" don't *need* to go anywhere, being based on mantra-like tonal tapestries over which the players are meant to solo with understated inventiveness. In this simpler, heavier context, Ponty can unbend futuristic sounds in a manner befitting the prententiously cosmic song titles.

Past infatuations with funky astral flayings can almost be heard on "The Art Of Happiness," where the mutable mirth is given new meaning when Ponty's violin enters like a rough, threatening beast. Atomically energized moments on "Fake Paradise" and "Egocentric Molecules" are temporarily motivated, though it's not really necessary to cook on *every* burner 100% of the time, as Ponty's violin mentor Stephane Grappelli has recently shown on the brilliantly acoustic *Uptown Dance* (Columbia). The occasional non-electric moments of previous concerts and albums are missing from *Cosmic Messenger*, and Ponty could use one or two earthly breathers amidst so much intergalactic retro-fire. R.H.

CLASSICAL

JOHN WILLIAMS: *John Williams and Friends.* John Williams and Carlos Bonell, guitars; Brian Gascoigne and Morris Pert, marimbas and vibraphone; Keith Marjoram, bass. [Paul Myers, producer; Mike Ross-Trevor, engineer; recorded in London, England.] Columbia 35108.

Performance: **Delightful**
Recording: **Very good**

John Williams is unique among classical guitarists not only because he possesses a technique that leaves virtually all competition far behind in terms of clean, accurate sound production, but because he is not afraid to step outside "classical" bounds. In the past, the results of his experimental efforts have been mixed: there was a popsy LP called *Changes* that most guitarists consider an embarrassment, as well as a wonderful jazz disc, *The Height Below* available only on import.

John Williams and Friends consists primarily of baroque music (there are two short Mozart pieces and one by Louis Daquin, the rest are pre-Classic) arranged for two guitars, marimbas, vibraphone and double bass. On paper, that instrumentation may seem a bit hokey. But the effect is absolutely sublime and entails no greater musical compromise than any more conventional transcription.

In fact, these arrangements by percussionist Brian Gascoigne (who also worked on *The Height Below*) often make great musical sense. The opening work, for instance, is a Vivaldi concerto originally scored for two mandolins and strings. When the work is transcribed simply for two guitars and strings—as it is often played—the drop of an octave in the transition from mandolin to guitar results in a registration balance quite different from that intended in the score, and one that doesn't particularly favor the solo instruments. Here, the marimbas replace the strings, compensating for the dropped octave and allowing the solo guitars (placed here on opposite channels) to stand out in much greater contrast. The double bass, likewise, replaces the cello and further serves to round out the sound.

For the most part, the guitars are out front, although the percussionists are no more buried than the string players in more conventional baroque recordings would be. On one track, Bach's "Jesu Joy of Man's Desiring," the guitars play the arpeggioed *obligato* part, giving the brief melodic passages to the marimbas. The guitar parts, by the way, are generally of equal prominence, and Carlos Bonell's efforts here are worthy of note. Bonell was a student of Williams, and now holds a professorship of guitar (as does Williams) at the Royal College of Music. He has recorded two LPs in England, but this constitutes his first appearance on disc in the U.S.

The combination of instruments used here is an especially sympathetic one which yields a delightfully soft, almost ethereal sound unlike anything heretofore available to guitar enthusiasts. The material Williams has chosen reveals a range of interpretive capabilities for this grouping ranging from the humorous (as in the spunky performance of Mozart's *Turkish Rondo)* to the more stately and serious (as in the Purcell *Trio Sonata in F Minor).* It may be pushing it to ask for more of the same, but at the time this was recorded (back in 1976—and it's been available in Europe since then; I wonder why American CBS has sat on these and other Williams tapes for so long) Williams et al were engaged in a series of concerts that included these works as well as some contemporary music written for this instrumentation. A recording of the latter would be more than welcome to these ears.

Williams plays only two solo works here, a Bach *Bouree* (from the third cello suite) and a Mozart *Adagio* (from his works for glass harmonica). As usual, his playing is a picture of clarity and taste. One small point, however, there's a persistent hiss in the *Bouree* that mars the recording, similar to the effect of turning a Dolby on and off in the mix. A small point, as I said, but an unfortunate one.

A.K.

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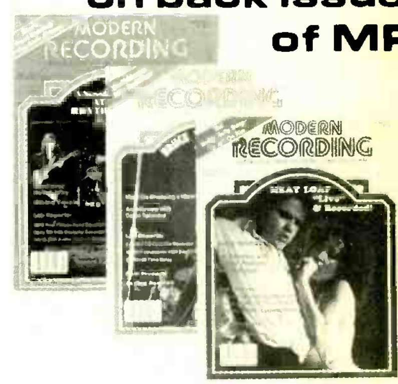

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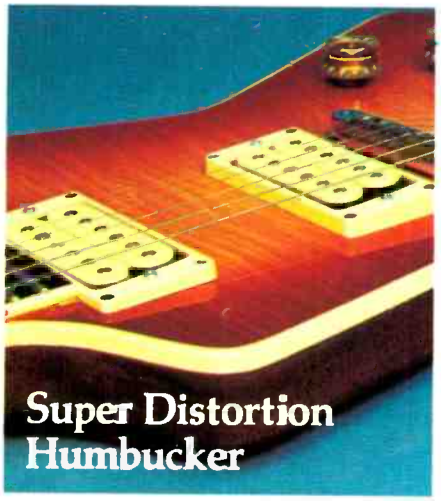

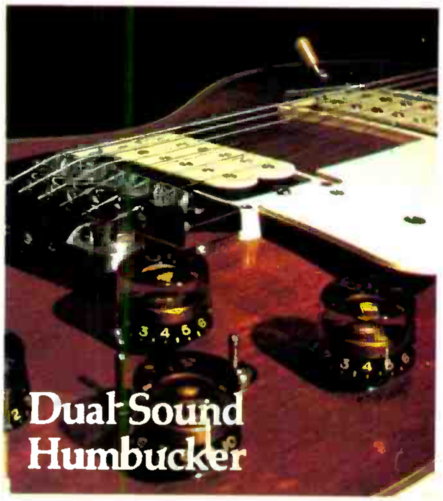

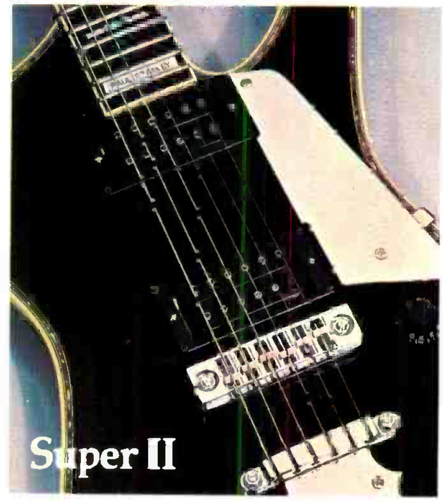

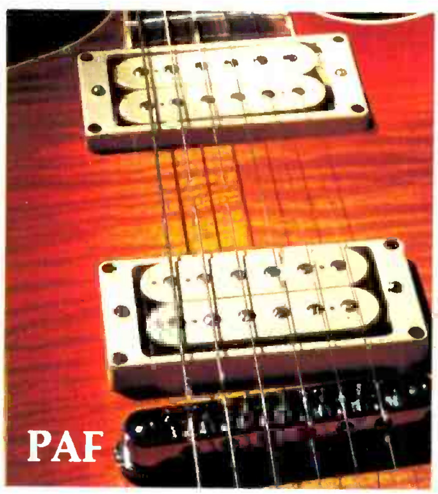

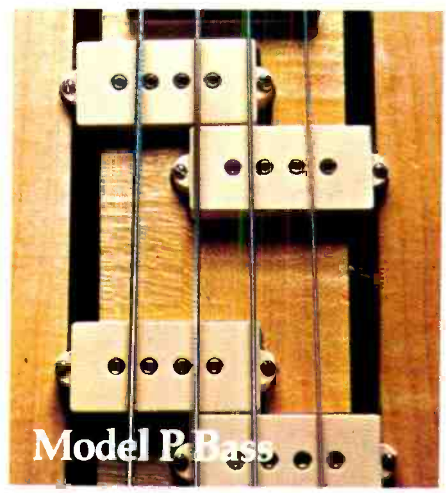

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