Profile: The Brecker Brothers

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

VOL. 4 NO. 8
MAY 1979

PCM Recording with Archie Shepp, Jaki Byard & Others

Interfacing Auxiliary Equipment -Part 2

Lab Reports:
Denon DR 750 Cassette Recorder
Hitachi HMA-8300 Power Amplifier
Spectra Sound 1000B Graphic EQ

Hands-On Report:
Neutrik Audio Delay

New Products

Record Reviews

Years ago this was a flanger.

Incredible, isn't it? But when flanging was first used, it was done like this. Rumor has it that the first time flanging was achieved, it happened by accident. An engineer mistakenly leaned on the flange of a moving reel altering its speed relative to another simultaneously moving machine. The sweeping sound that resulted was one of an enhanced tonality, similar to a phase shift but also having characteristics of its own. This phenomenon became the hottest new sound in the recording industry overnight, but there were problems. In order to duplicate the flanging sound one had to obtain three recording machines, one experienced engineer, and a lot of time.

It was soon realized that this mysterious sound was actually the result of a time delay causing the cancellation of certain harmonically related frequencies whose sweep could be controlled.

Later, it was also discovered that the same sound could be attained electronically by splitting the signal, passing one half through time delay circuitry, and re-combining the signals. The only setback was that this effect could be produced only with expensive electronic equipment, limiting its use to large recording studios.

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CIRCLE 83 ON READER SERVICE CARD

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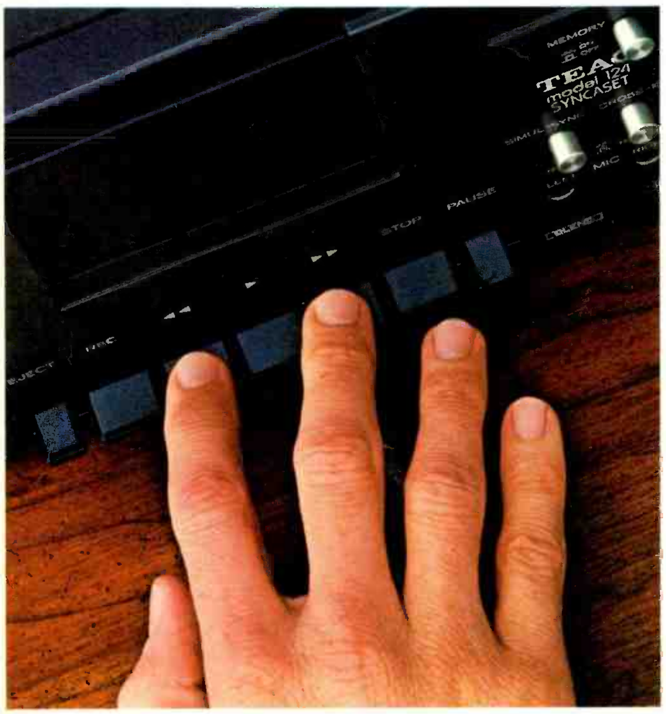

It controls the TEAC Model 124 Syncaset™. Our first cassette deck that lets you record one track, then overdub the other to get two musical parts in perfect time. Later, you can mix live material with these two tracks and hear all three parts through your home sound system.

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

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

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

TEAC.
Multitrack Series

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

MAY 1979

VOL. 4 NO. 8

THE FEATURES

COMING NEXT ISSUE!

A Session with B.B. King

Unravelling the Cable Problem

THE STAPLES

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MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING CONSCIOUS SOCIETY

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Letters to the Editor

Presto Chango

With regard to the letter ("Talkback," page 22, March 1979) from Keith Evans about his old Presto disc recorder, perhaps I can help lead you to the current status of the firm.

A number of years back, Presto Recording Company was acquired by the David Bogen Company to form Bogen-Presto. As a division of Bogen, Presto continued to make recording blanks, turntables, disc and tape recorders, etc. While they had some very advanced ideas (at the time) for state-of-the-art tape recorders (ask an old timer about the Presto 800) and were selling a modest number of Muzak tape players and other specialized units, the Presto operation quietly disappeared into Bogen.

The company is now known as the Bogen Communications Division of Lear Siegler, Inc., and is located at Route 4 and Forest Avenue, Paramus, New Jersey 07652. The company is primarily concentrated in the institutional sound field, having dropped out of the popular-priced hi-fi business some time back.

By the way, the recorder Mr. Evans has purchased is a Model 6N, not a GN. Presto had a well-deserved reputation for solid, sturdy equipment, and this is certainly proof, since the unit must be at least two decades old!

—Lloyd W. Loring
Vice President
J.G. Sullivan Advertising, Inc.
South Bend, In.

In reference to Keith Evans' letter in the March 1979 issue (page 22), I would like to point out that Presto was bought out by Bogen Co. during the 1950s. At the time, Bogen, who now makes low-end P.A. gear, was entering the "new" hi-fi field, and bought out Presto to complement its line of amps. The Bogen-Presto turntable and Bogen hi-fi amps were made until the early 1960s, when Bogen dropped its hi-fi line completely. I would suggest writing Bogen at Box 500 at [same name and address as given by Mr. Loring, above].

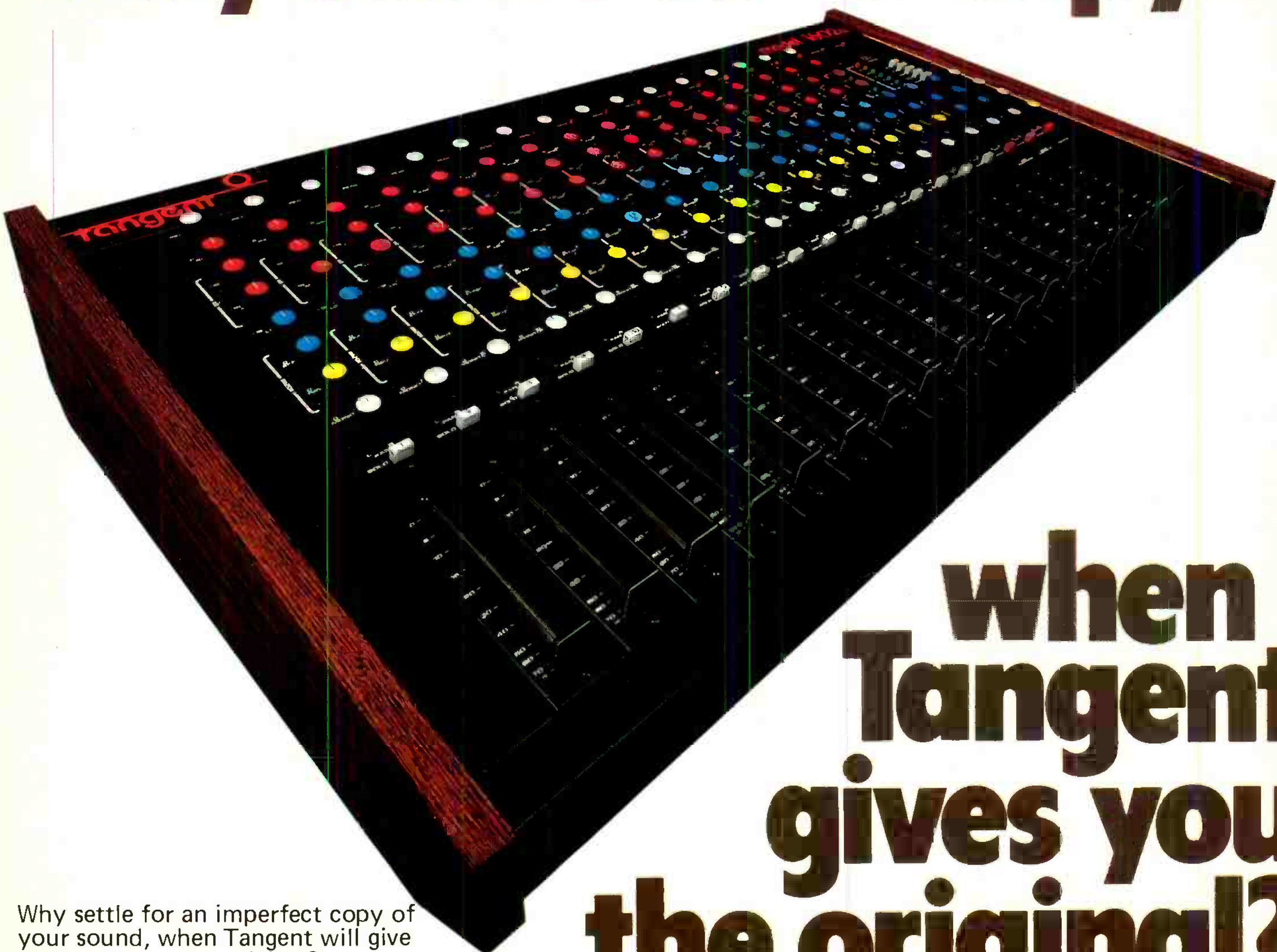
Also, thank you for printing my letter (same issue, page 18).

—Barry Fuerst
Oak Park, Ill.

The above two letters arrived in these offices on the same day and we're most gratified to know that people are willing to take the time to respond to such a dilemma as Mr. Evans'.

The final poop, though, is not too positive: We gave Lear Siegler, Inc.'s Bogen Communications Division a call, fully expecting a response as uplifting as those from Messrs. Loring and Fuerst, but were informed that the company has no supplies such as styli and discs available for Presto units. We do hope that Keith Evans has gotten those accessories, then, from the firms we had made mention of in our original reply in "Talkback," back in March.

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:

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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.

THREE SENDS

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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CIRCLE 75 ON READER SERVICE CARD

Can We Drop It? (The Crew Wants to Know)

Having been an avid reader of your magazine for some three years, I'd like to make a few suggestions. I have been learning a great deal from you. For that I thank you. But you do fall short on some fairly basic points in my opinion.

You have two of the best doing your "Lab Reports" and their work has been fine to date. You've dealt with products the average audio pro person would find interesting. I firmly believe, however, that you need to include some tests you do not presently do. Let's take as an example your recent review of the Heathkit AA-1600 Power Amp. Measurements were taken at its manufacturer-published loads and, as is the case with most Heathkit equipment, they were exceeded by comfortable margins. But as a person with extraordinary needs (in my case, sound reinforcement work), the rating into 8 ohms does me no good. What I need to know is how it will operate into 4 or 2 ohms. More importantly, I need to know how long it will operate into those loads.

What I am asking for is a special series of tests on equipment designed for the needs of the sound reinforcement crew and the studio crew. We need questions answered about durability (Can you drop the amp from the scaffolding to the ground and expect it to work when you plug it in?), performance into non-standard loads (Will it drive some six 4-ohm midranges in parallel at clipping for the whole concert?) and so forth.

I have only dealt with a few of the areas of performance that affect only power amps, of course, but I think you get the message. You have given a lot of us what we have needed up till now. These special tests would give us something we can get nowhere else.

—Jeff E. Heyl
Rocky River, Ohio

Well, you're right. Test reports tailored to sound reinforcement equipment are important to a large portion of our readers. But as a veteran subscriber, you are also familiar with our regular attention to such equipment through the article "Hands On." Whereas "Lab Reports" covers high end audio and racked studio equipment, "Hands On" concentrates on specialized pieces and concert sound units, including amplifiers specifically intended for sound reinforcement use by the manufacturer.

For largely practical reasons, extreme stress tests such as the two you parenthetically suggest can only be speculative on our reviewers' parts, as we receive equipment for review on loan from manufacturers and prefer to return it intact. In most reviews, however, we feel appropriate attention is paid to these particulars.

Cooling Your Copper

In any sound reinforcement systems I have designed I have always had excellent results using standard 10 or 12 gauge industrial copper wire for loudspeaker cables. Spacing the hot and ground conductors about one inch apart eliminates any effects of capacitive reactance of the cable while the cross sectional area of the copper eliminates any significant external voltage losses. This solves most of the major problems associated with speaker cable runs longer than ten feet.

However, the average audio consumer seems to insist on very expensive and complex "high definition cables" which are available at most audio stores. These cables are generally more fragile as the enamel coating on

The Soundcraft 1″ 8-track.

Produced after two years of development, it's sophisticated, easy to use, reliable, and its specifications are superb.
The deck plate is a rigid aluminium casting of extreme dimensional accuracy, ensuring the absolute stability of the tape path.
Tape tension is servo controlled, as is the capstan (which has +15, −50% varispeed control), and wow and flutter is only 0.03%.
Control and monitoring facilities are comprehensive. A simple push-button matrix permits selection of line-in, sync and replay for any of the tracks and led's indicate the selected status.
The tape counter has a plasma display reading in minutes and seconds. There is also a highly accurate search-to-zero facility.

A special feature of the machine is that the whole of this control panel, and the varispeed control, can be used remotely. So once the machine is prepared, a single engineer can work a recording session without leaving the mixing console.
Replay S/N is 76dB (reference 510nWb IEC curve A); erasure is 70dB and record/replay frequency response is +1, −2dB (30Hz to 20kHz).
All signal levels can be adjusted by presets on each channel. The external power supply is fully protected and 19″ rack mounting.
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SOUNDCRAFT MAGNETICS LIMITED

CIRCLE 78 ON READER SERVICE CARD

the fine copper wires may be chipped off with excessively rough handling. Should this happen, the amplifier may short circuit and sustain circuit damage. With this in mind, I suggest using the method mentioned above, or spend a great sum of money and go all the way: Simply use two half-inch hollow copper pipes as the conductors while they are filled with liquid nitrogen to provide the effect of super conductors. The benefits will be inaudible but it may be pleasing to know that every effort has been made to cure a weak link in the system.

—Mr. R.D. Goode
General Manager
Audio Systems Installation and Consulting
Don Mills, Ontario

An interesting suggestion. We wouldn't be surprised if someone put this on the market.

Left, Right, March!

Well, it seems the left and the right hands are not keeping close tabs on each other ... We claimed, last issue (on page 14, answering a letter from one Larry Feeney), that "the third and final Swedien article" appeared within. Wrong. *It was in the* previous issue *(March 1979), and had to do with miking strings and horns. We believe we've gotten back with the program, however, so those of you who delight in watching people slip on banana peels are advised to look elsewhere for such dubious entertainment.*

—Ed.

Keep Those Cards and Letters Coming In

Many of the letters to the editor that we receive include a request for personal replies by mail and often include a self-addressed, stamped envelope or coins for postage costs (which we do appreciate). However, we simply can't answer your letters individually, so please don't send "SASEs" or postage with letters of inquiry.

We've also found that a large percentage of our mail consists of problems, questions, requests for information, etc. that most likely are on the minds of others of our readership who have not taken to pen and P.O. for solutions. Publishing letters and answers serves a wide audience and certainly saves everybody's postage.

Do keep on writing to us—knowing our readers helps us to print the best for our readers—but please don't enclose your stamped envelopes. Use that second stamp and write your mom. Although she might not answer you individually, either. And you can let her think it was your idea, we don't mind.

—Ed.

Interest at a Pique

Would you give some information on the Polymoog Keyboard? We have an August 1978 copy of your magazine (in which there was an advertisement for the Polymoog) which piqued our interest; hence, we're now subscribing. Anyway, even though the ad ran many months ago, we would like to know some more about the product. We are very interested in a keyboard of this type. Thank you very much.

—Lonetta J. Holmes
Dallas, Tex.

Norlin Music manufactures Moog equipment, including the Polymoog. Write them at Advertising Dept. MR4,

7373 N. Cicero Ave., Lincolnwood, Illinois 60646 for further information regarding this and other products.

Any Questions?

I was quite pleased to see *Modern Recording's* recent article on reinforcing Meat Loaf, with a photo showing an overhead array of eight Karlson low-frequency enclosures. These looked as though they may have been built by the late John Karlson himself.

Transylvania Power Company has taken over Karlson's audio operations completely, from his orginal working drawings to the patent rights to his final improvements. If any of your readers have technical questions about these strange-looking enclosures, we would be pleased to answer them.

—Walter Zintz
President
Transylvania Power Company
Walnut Creek, Ca.

Readers who in fact have questions regarding the enclosures may write the company at 260 Marshall Drive in Walnut Creek, Ca. 94598.

Ripe for Rupert

I first became a faithful reader of *Modern Recording* in April of '78 with your outstanding issue of that month. I was delighted to see the "Confessions of an Audio Addict" article by James F. Rupert in your magazine. I first met James several years ago when he and his crew were recording a live concert for Proctor and Bergman of the Firesign Theatre comedy team. He and his crew handled what looked to me like an impossible recording situation in a theatre in the round performance with precision and professionalism. I feel especially thankful since he took the time to patiently answer a couple of hundred ignorant questions of mine before and after the performance.

You have a super mag and my only suggestion for improvement is when are we going to see more articles from Mr. Rupert? As a novice recordist, some of your articles are a tad over my head, but James' piece helped to set me straight in terms I could understand. The humor also helped me to realize that other people have trouble setting up their first studio too.

Otherwise count me in as a lifetime reader and keep up the good work.

—Paul Newton
Lincoln, Neb.

P.S. If Rupert is dead or something please disregard this letter.

Paul, your letter crossed in the mail with our April '79 issue, which just so happened to boast a second piece penned by Mr. Rupert (not at all dead), this one entitled, "Up the Basement, or A Quiz for The Would-Be Recordist." April has proven to be a month highly suited to writings by James F. Rupert, and we would be pleased to continue this trend. If you haven't seen the Quiz, check out pp. 48-9 in Modern Recording, *Vol. 4 No. 7 with a sharp No. 2 pencil.*

Man in Blue Loses the Blues

I was hoping that there was a magazine relating to my interests and the other day I was lucky enough to catch the last issue on the newsstand. I think your magazine is fantastic; it's packed full of all kinds of information and news I've been dying to find and never could until now. Thank you—most sincerely!

—David A. Duhl
Sound Technician
U.S.S. New Orleans

The standard bearers.

The high bias standard.

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

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

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

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

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

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

TDK®

The machine for your machine.

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

CIRCLE 67 ON READER SERVICE CARD

Hot Nashville "Super-Picker" Sessions

Pollaro Multi-Media Advertising & Productions would like to commend you on an excellent magazine. I especially enjoy the articles on the studio and "live" sessions. But I feel you are neglecting a style of music very prominent in today's record market. I'm speaking of Country music.

How about a session with Dolly Parton, Mel Tillis, or maybe Merle Haggard? After all, as dedicated, professional engineers and producers, aren't we expected to cater to anyone who walks in the studio door?

Let's hear what goes on inside those hot Nashville studio sessions with the "Super-Pickers."

Once again, you have a fantastic, extraordinary magazine!

—Sid Whatley
Production Manager
P.M.A.P
Denison, Tex.

We totally agree with you that we've not adequately covered the Country music scene. But it's not a function of motivation: these sessions have so far been relatively closed to us (we did, though, cover and publish "A Session With the Charlie Daniels Band" back in our December 1977 issue) but we keep on plugging. We're always looking to get in on some of those sessions. Please keep watching.

Time for a Feeding

Fantastic job you're doing. Your work demands the best of praise. I would enjoy seeing an article on adjustment and calibration of some of the more popular tape recorders.

Keep the print coming; you're a light in the dark.

—Michael S. Thomas
Rio Music
Troy, Ala.

About a year ago, we ran a feature entitled, "The Care and Feeding of Your Multi-track Recorder." It was written by David Moyssiadis, of Frankford-Wayne Mastering Labs in Philadelphia and a Contributing Editor to this magazine. Get your hands on the issue it appeared in, MR *of March 1978 (which also featured "A Session With Lou Rawls" and our "Interview with Frank Zappa"). We're sure you'll find it all enlightening.*

Flattery Gets Fred To Ga. and Ca.

Being a subscriber, I must take this opportunity to express my belief that *Modern Recording* is one of the few values existing today. However, I must admit that I have ulterior motives for writing this letter.

I would appreciate it if you could send me or print the addresses of Audio Perceptions and Musimatic Electronics, Inc; their equipment was displayed in "The Product Scene."

—Fred Haskins
Rochester, N.Y.

Musimatic is at 4187 Glenwood Rd., Decatur, Georgia 30032. Audio Perceptions is based in Diamond Bar, California 91765, at P.O. Box 4861.

Sound Reinforcement Companies

I am interested in obtaining the names of sound companies across the United States that supply sound and personnel for live concerts. I would appreciate very much your sending me any list that you may have available. If not, do you know where I could obtain such in-

If you want the condenser microphone sound on stage, Electro-Voice gives you that option.

The PL76 and PL77 condenser cardioid microphones are fast becoming the number one choices of vocalists who want to make the "studio-condenser" sound a part of their act. Both mikes give you condenser performance in a package that competes with dynamic microphone durability. Their gutsy, bass-boosting proximity effect adds presence to any voice. The PL76 is powered by a 4.5 volt battery. The PL77 is similar except that it is also phantom powerable. The "77's" output is 4dB down from the "76's" to allow for more flexibility at the mixing board, and it has a recessed on/off switch that many sound men prefer.

For those desiring the more traditional dynamic sound, the PL91 and PL95 fit the bill perfectly. The PL91, with its mild bass-boost and clear highs is a joy to work with. The PL95, the "pro's choice" in a dynamic cardioid offers the best gain-before-feedback of any dynamic mike in the business – a test we invite you to make.

Electro-Voice also offers four superb instrument microphones. The PL5 dynamic omni is the mike to use when high sound pressure levels are encountered, as you would find when miking bass drums or amplified guitars, basses or synthesizers.

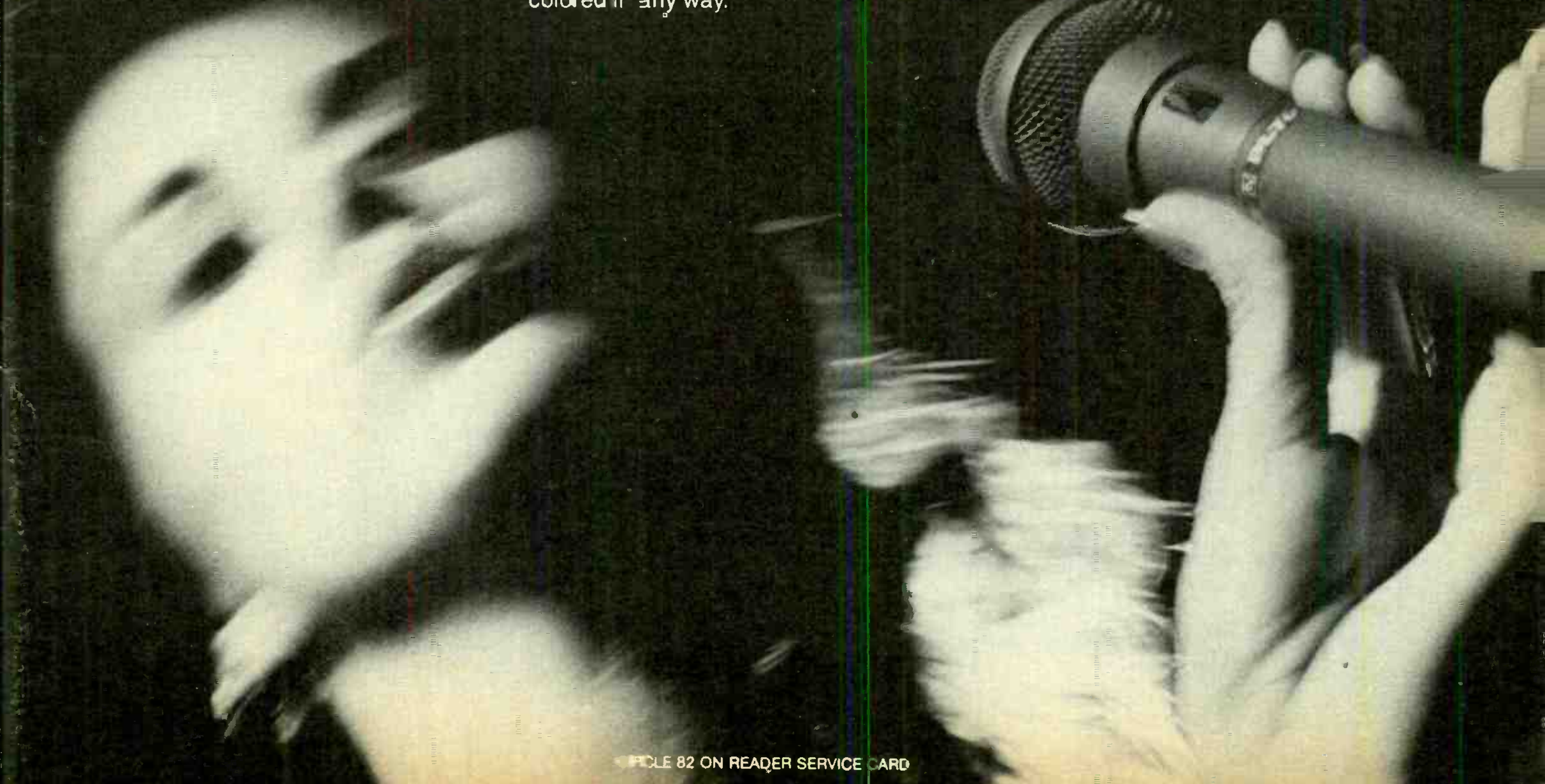
The PL6, with its patented Variable-D construction gives you cardioid (directional) performance without up-close bass boost – perfect for miking brass, reeds, percussion or piano. The PL11, even though it's a directional mike, maintains its response curve off axis. "Leaked" sound from off-axis instruments are faithfully reproduced – not colored in any way.

E-V's PL9 dynamic omni has one of the flattest frequency response curves in the business – from 40 to 18,000 Hz. And its small size lets you mike instruments you couldn't get near with other mikes offering this performance.

All E-V Pro-Line microphones come with super-tough Memraflex grille screens that resist denting. Designed to keep your mikes looking like new for a long time. All have a non-reflecting gray finish that won't compete for attention under bright stage lights.

When the time comes to update your current mike setup, we invite you to A-B Electro-Voice Pro-Line mikes against any others, for any application. If you try them, you'll want them in your act.

EV Electro-Voice
a gulton company
600 Cecil Street, Buchanan, Michigan 49107



CIRCLE 82 ON READER SERVICE CARD

formation? Many thanks for your kind consideration.

—Helene Cohen
Baltimore, Md.

I am looking for the mailing addresses of professional sound reinforcement companies and hoping that you can supply me. If not, could you tell me where I could get that information? I would appreciate it greatly. Thank you.

—Kenneth Feher
Soundbar Sound Co.
Newtown, Ct.

Puzzled though we are by the extreme similarity between these two letters (both writers also mentioned the same two individual companies by name to illustrate their requests, though we've taken the liberty to edit these out so as not to slight others), we suggest getting the Billboard *annual* Talent Directory, *which lists geographically companies that supply sound and lighting. For information on how to obtain the* Talent Directory, *write* Billboard *(boy, are they getting business from us!) at 1515 Broadway, New York, N.Y. 10036.*

Memphis Fan

I discovered your magazine about nine months ago. I like it very much, but I don't understand why you do not feature sessions with more rhythm & blues (soul) artists. Maybe you could shed a little light on this for me—or give a rundown on some features to be in coming issues.

During the mid- to late- 60s, the STAX studios in Memphis, Tennessee turned out some of the biggest hits of all time in R&B. Although the company has gone out of business, I'd like to see an article or feature dealing with their methods, equipment, etc. I'm sure some of their engineers (Ron Capone, Henry Busch, William Brown) are still knocking around Memphis.

Willie Mitchell, famed Memphis trumpeter, is, I understand, one hell of an engineer. He did all the early Al Green sessions (in the early 70s) at the HI recording studios in Memphis. I'd love to see a feature on Willie and his operation. It seems that Willie shies away from solid state equipment and prefers tube type equipment instead.

—Gerald Williams
Rocky Mount, N.C.

A fine, fine suggestion. And you're right! We don't give enough attention to these artists. Please keep in mind, though, that what we tend to print doesn't necessarily reflect what we tend to like. More often than not, our choices reflect the tastes of the majority of the record-buying public. We attempt to give equal time, but sometimes must compromise.

At any rate, your ideas on the Stax and Willie Mitchell sessions are well-founded—we'll get on that case immediately.

Particle Particulars

I am in a sticky situation. After spending hours on the market for a four-channel open reel recorder, I came up with a choice I am very satisfied with. My only problem is related to my concern that a metal particle open reel recorder will come on the market (with its 6-9 extra dB signal to noise) and obsolete all the other decks.

Before I lay my cash on the counter, I would like your advice on a few questions regarding metal particle trends.

First, I think I remember reading that the current recording heads are not compatible with this new tape. Is that true? If that is true, will the companies make replacement heads for their decks or would it also require major modification of the electronics?

Second, does open reel metal particle tape even exist?

Last of all, is your final opinion to buy now, or sit a few months and see what develops on the market? Thank you!

—James Tate
Sharon, Pa.

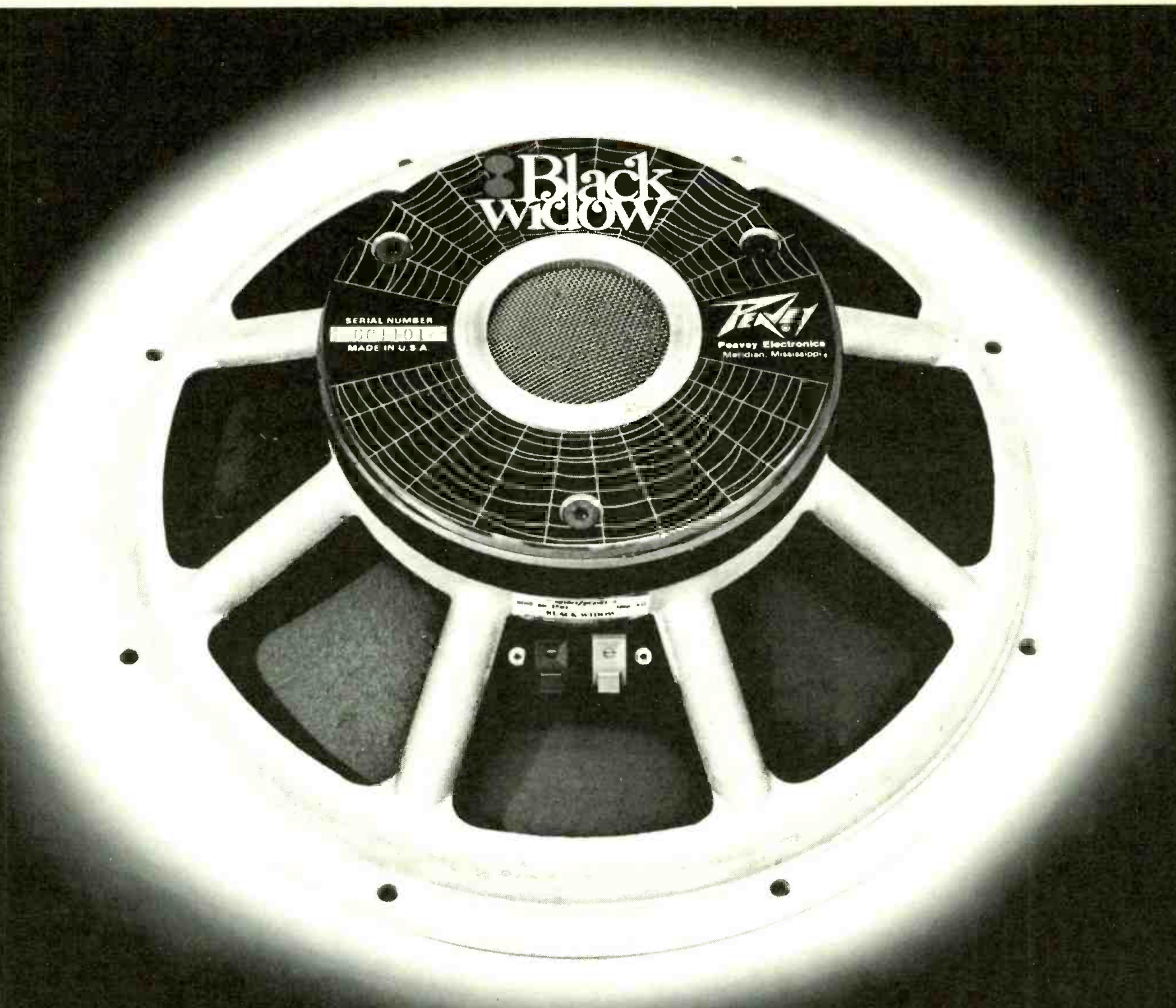
Our first opinion: buy now.

And the answer to your second question: open reel metal particle tape is neither on nor expected on the market, as far as our industry sources state.

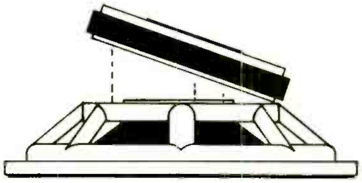
The need for and potential for improved performance with metal particle recording tape is great only in the cassette format; the open reel format would not sustain as significant nor marked an upgrade in performance specs if current high-performance metal oxide tapes were supplanted with metal particle tapes.

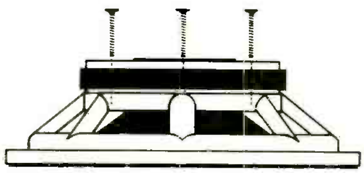
To answer your first question, at least one cassette deck manufacturer (Eumig) has devised a mode of adaptation so existing decks might use metal particle tape. Major modifications are quite necessary.

Our final opinion: buy now.


Black widow
SERIAL NUMBER
MADE IN U.S.A.
Peavey Electronics
Meridian, Mississippi

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





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

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

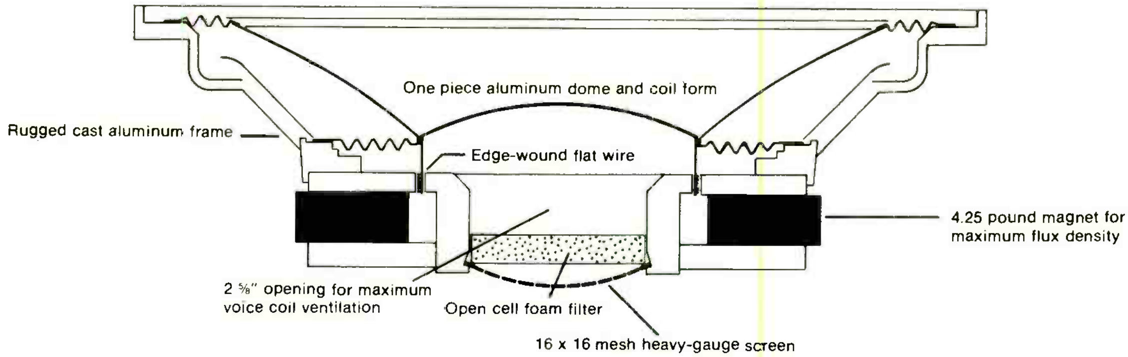
Each Peavey Black Widow is subjected to extensive quality control procedures to insure long field life and high reliability. The manufacturing methods employed by Peavey, such as numerical and computer controlled machining equipment, allow the Black Widow to maintain the close tolerances necessary for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

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

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

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

PEAVEY ELECTRONICS
711 A Street
Meridian, Mississippi 39301



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

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.*

Amplifier Alternatives

I have a question concerning P.A. systems that I hope you'll be able to provide a solution for.

Quite often it is necessary to run paralleled ("Y"-ed) cables from the outputs of electronic crossovers to feed more than one amp. The reasons for this are primarily due to impedance matching (optimum) between power amps and loudspeakers, plus the effects of loudspeaker wiring on the damping factor of power amps. What happens to the voltage (output) of the crossover when it's paralleled (split) to accommodate more than one load (amp/channel)? What happens to the load impedance? Obviously there's a considerable change in both.

Is there any other alternative besides a distribution amplifier system?

—Tom Young
S. Salem, N.Y.

Almost all professional quality electronic crossovers have a low output impedance—usually 600 ohms. This means that they can deliver their full output at a rated distortion into 600 ohms or greater. In multiple amplifier installations, it is common practice to wire the inputs of the power amps in parallel. This is because the input impedance of almost all professional quality amps is very high—50K or higher.

To determine the R_T of all amps on the line with equal or unequal impedances use the following formula:

$$R_T = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} \cdots \frac{1}{R_N}}$$

Using common values:
3 amps 75K input/mp 1 with 50K

$$R_T = \frac{1}{\frac{1}{75K} + \frac{1}{75K} + \frac{1}{75K} + \frac{1}{75K}} =$$

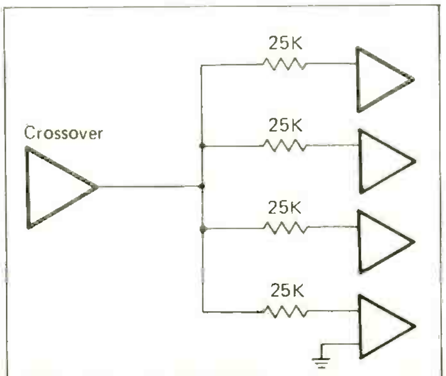
$$= \frac{1}{\frac{1}{.13 \times 10^{-4}} + \frac{1}{.13 \times 10^{-4}} + \frac{1}{.13 \times 10^{-4}} + \frac{1}{.2 \times 10^{-4}}}$$

$$= \frac{1}{\frac{1}{.59 \times 10^{-4}}}$$

$$= 16{,}950\Omega$$

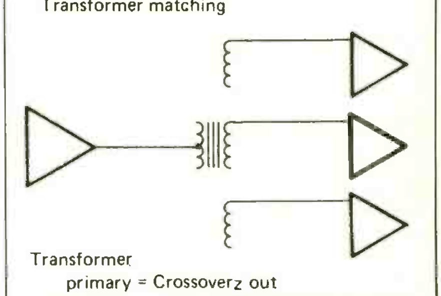
As can be seen, this R_T is well above the output Z of the crossover, assuring that the crossover unit will not be loaded below its rated output Z, and can thus deliver its rated voltage at its rated distortion.

Another method for determining this would be a matching network:

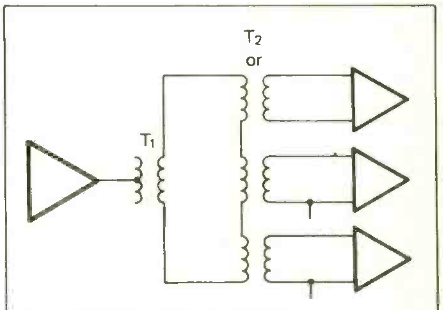


25K resistors should be okay

or transformer matching:



Transformer
primary = Crossover$_Z$ out
secondary = Total$_Z$ of all loads with tap for separate amp at same$_Z$ at each load



T_1 = Primary = Out$_Z$ of crossover
Secondary = Total load$_Z$
T_2 = Primary = In$_Z$ of single load
Secondary = In$_Z$ of single load
1:1 ratio bridging

Remember, resistors are cheap; distribution amps are expensive; and good transformers are very expensive!

—Will Parry
Operating Manager
Herb Slawker
H.D.I.
Maryland Sound Industries
Baltimore, Md.

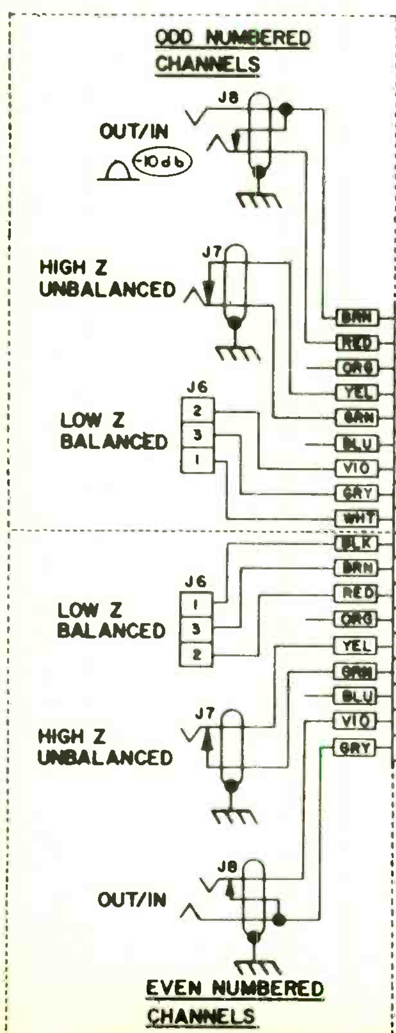
Satisfying Performance

I recently purchased a Uni-Sync Trouper I stereo mixer. I'm more than

satisfied with its performance as it more than adequately fills my application needs. However, one day I tried recording on it and discovered that when the direct jacks are being used, the VU (LED) meters would not operate. Is there something in the circuitry that defeats the VU function while recording or do I have a problem?

—Oscar Webb, Jr.
Toledo, Ohio

The TIS-OCM Live Music Mixing System has eight individual inputs (expandable to eighteen inputs with the TIS-IXM input expander module) that will allow the individual to use either a balanced, low-impedance input (150 ohms) or a high impedance input for mic or line level signals. Each of these input channels has a house pan pot (located at the top of each channel) that pans the acoustic image of the signal to the left or right house summing busses. Located beneath the house pan pots are the echo pan pots that pan the echo signal to the left or right echo send



busses. This enables the operator to pan the echo either towards or away from the acoustic image.

The TIS-OCM is equipped with a built-in reverb system that enables the mixer to adjust individual echo send levels through the use of the echo send controls for each input channel. The levels of the inputs can be monitored through the preview system (solo switch) to enable the mixer to listen in on individual inputs or outputs or combinations of them with headphones. The VU meter follows the preview system for monitoring relative levels through the console.

For individuals that desire to use external processing equipment for each of the input channels, the Trouper Series Mixers have an out/in jack that will allow you to adapt this external equipment by inserting a stereo ¼-inch phono plug. The output signal of the tip is at – 10 dBv with an impedance of 100 ohms. The return impedance is at 4K ohms to the ring of the out/in jack.

The out/in jacks can be used for effects, direct recording feeds, or subgrouping. When used as a direct feed into a tape recorder, the use of a mono ¼-inch phono plug should be inserted to the tip of the out/in jack (this would be the first "click" of the stereo jack). This will sample the signal of that channel and can be monitored by the solo system for that channel. Using the direct feed tip of the out/in jack will not interrupt the signal flow of the channels to the output of the mixer. For direct feeds you must not completely insert the mono ¼-inch phono plug into the out/in jack. This will disconnect the signal to the preview system.

Using the out/in jacks will allow eight (eighteen with the expander) individual direct feeds for taping while still allowing to properly mix the inputs (with reverb) for house and monitor outputs.

—Andy Thompson
Manager, Customer Service
Uni-Sync
Westlake Village, Ca.

Balancing on a Budget

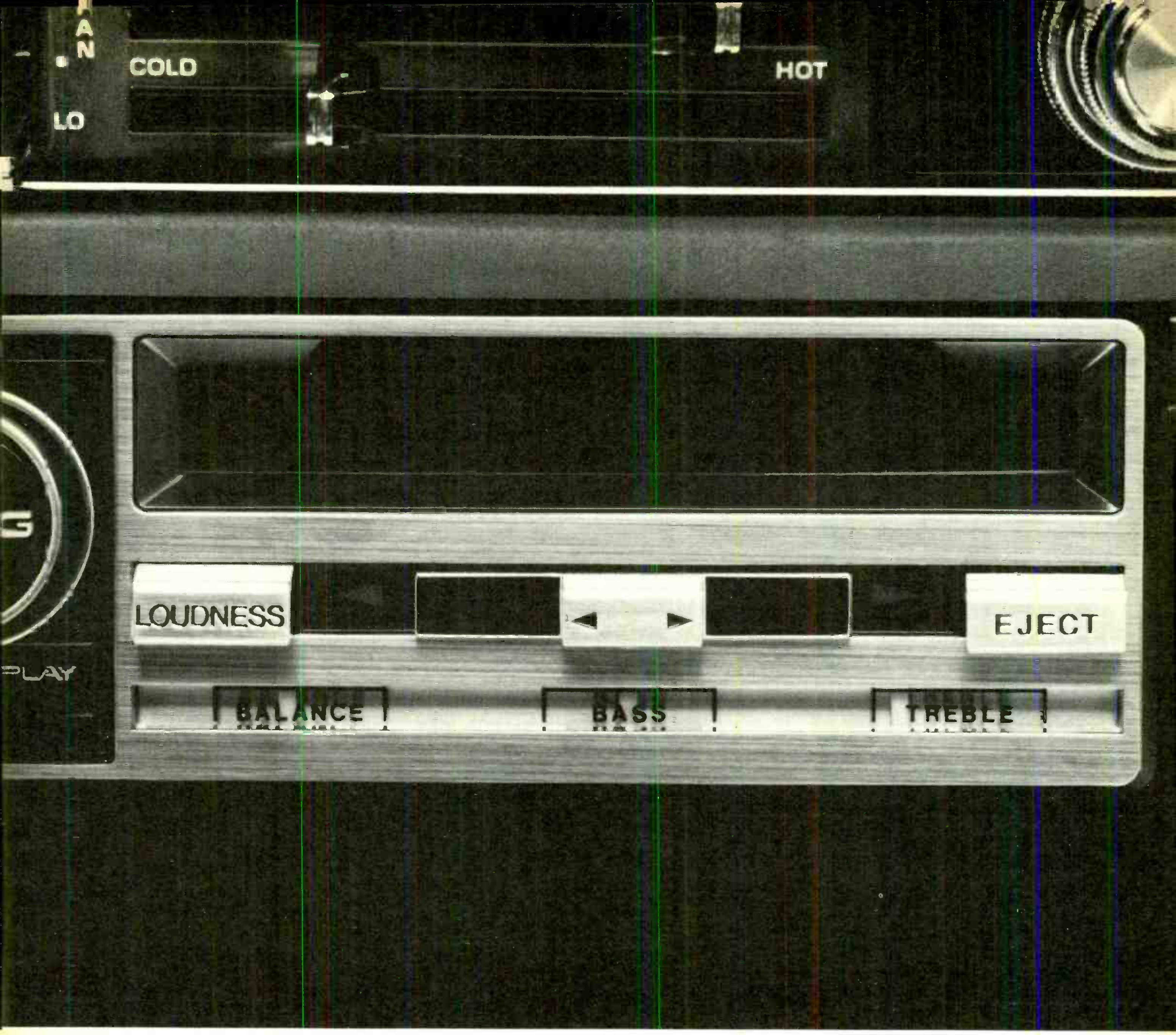
Being a four-track recording enthusiast with a limited budget, I've spent the last few months looking for a modest package of effect boxes. For a private demo project, we are looking for a stereo delay line (including a VCO) with good electronic reverberation, and compression for a tight sound, since I'm working with dbx in the ongoing battle against hiss.

In my search, I've come across a lot of apparatus which is symmetrical, which, as I understand it, would give me some mismatch problems with my system which consists of a TEAC 3340S, a dbx 155 and a Teac Model 3. So far, the pieces that interest me include the Urei LA-4, the Lexicon Prime Time and the Dynacord SRS 56.

Most dealers I've encountered who handle professional lines are not too interested in servicing small studio owners, so I thought perhaps you could give me some pointers on how to work with asymmetrical lines (hints on impedance matching, etc., would be extremely appreciated).

—Bob Smaling
Wormer, Holland

First of all, I assume that when you use the terms symmetrical and asymmetrical lines, that you are referring to what is more commonly called balanced and unbalanced lines. There are two types of commonly used balanced input or output circuits. One type of balanced circuit utilizes a transformer; another type of balanced circuit is electronic.



Fill it up with premium.

What premium gasoline can do for your car, premium tape can do for your car stereo.

And there's no finer premium tape than Maxell.

Every type of Maxell tape is designed to give you the widest frequency response, the highest possible signal-to-noise ratio and virtually no distortion. All of which results in *high octane* sound.

And to make sure our cassettes don't run out of gas somewhere down the road, we've constructed them to tolerances as much as 60% higher than industry standards. We use the finest high-impact styrene, precision pins, polyester and screws.

Because of this, we believe Maxell makes the world's finest cassettes.

And every year, hundreds of thousands of people who own car stereos are driven to the same conclusion.

maxell

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

CIRCLE 79 ON READER SERVICE CARD

There will be no harm done to equipment by feeding a balanced output into an unbalanced input, or by feeding an unbalanced output into a balanced input. The utilization of balanced and unbalanced equipment together is a somewhat common practice today. However, occasionally hums or buzzes can result which usually can be corrected by changing the grounding of one or both units. The various commonly-used grounding procedures are too complex to go into here.

—Larry Blakely
Recording Consultant
Framingham, Ma.

Speaking Out on Monitors

I found the article on stage monitors ("Monitor Speakers," January 1979, page 60) to be informative and well-written, but it has raised some questions in my mind.

Toward the end of the article, Mr. Krause says a "properly adjusted" limiter can improve the overall gain, while a "poorly adjusted" one can foul up the system. Specifically, what does he mean by "poor" and "proper" limiter adjustments?

Also, I've heard singers complain about not being able to hear the monitors that are blasting away directly in front of them and that are plainly audible a few feet away. Sometimes moving the cabinet away from the singer helps, but is there some kind of psychoacoustic phenomenon going on?

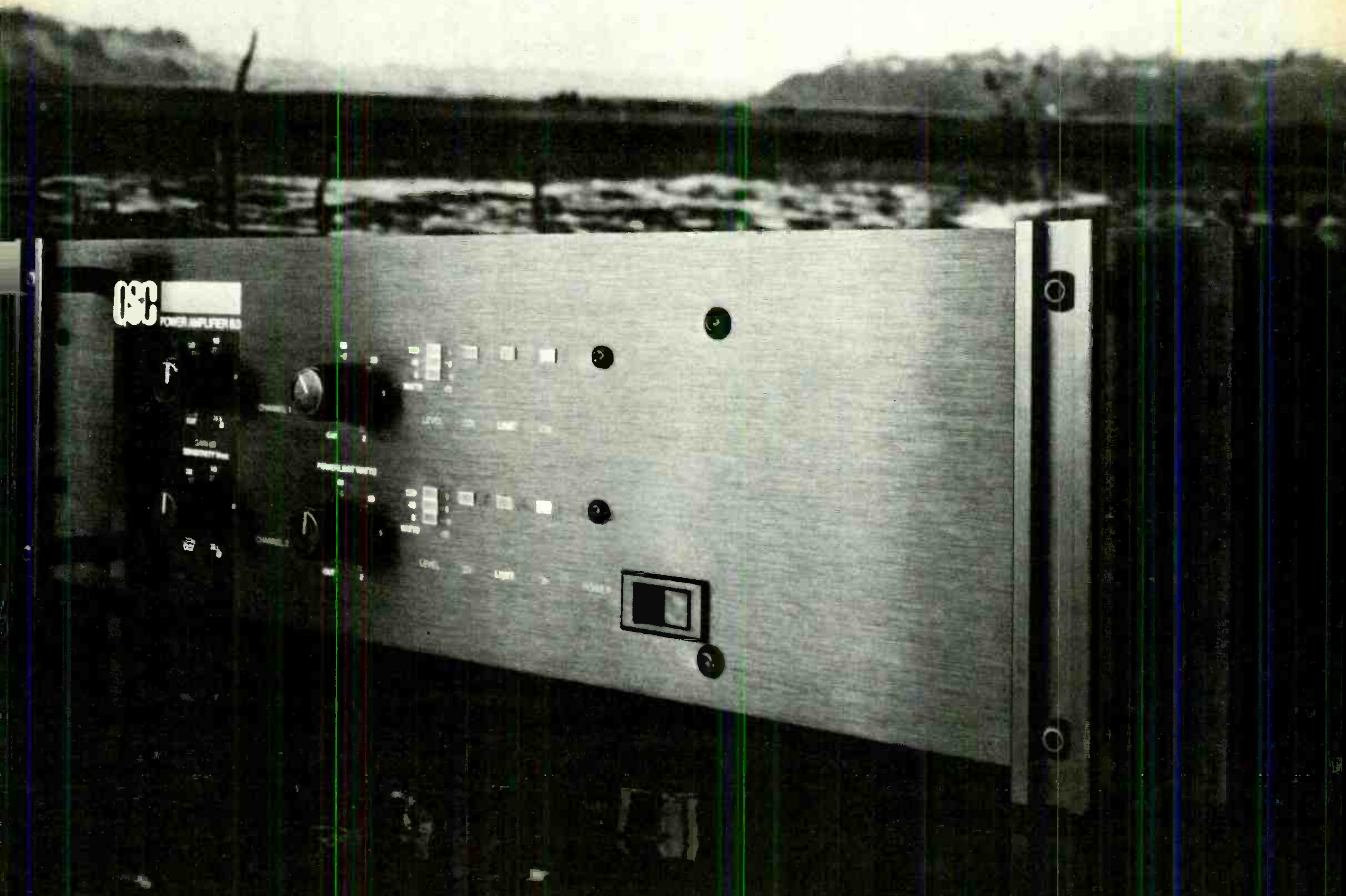
—Philip Adler
Soundsalright
Boston, Ma.

In regard to monitor systems, a properly adjusted limiter will prevent a power amplifier from being driven into clipping. A typical 100 watt power amplifier will have a distortion rating of 0.10% at the rated output power of 100 watts. This represents a voltage of 20.0 volts across a load of 4.0 ohms. The same amplifier will probably generate 10 to 20% distortion if required to deliver 22 or 23 volts across the same load. This small voltage difference results in a great deal of distortion and its undesirable side effects.

Now back to the original question. A limiter is an amplifier whose gain is determined by the amplitude of the input signal. The amount of control is known as the ratio, the point at which gain reduction begins is the threshold, and the time it takes to react to and recover from a gain change are the attack and release times. This rather long sentence is a great simplification of limiter operation, but it will get some readers started in the right direction.

Consider our 100 watt power amplifier. This typical amplifier takes 2.0 volts of input drive to provide an output of 20.0 volts. This is a voltage gain of 20.0 dB or a multiplication factor of 10. If a limiter is used between the mixing console and the power amplifier, the voltage drive to the power amplifier can be limited to the correct 2.0 volts, therefore preventing distortion. An example is that the console may have a peak voltage output of 4.0 volts when the microphone is dropped or thrown on the stage. Without the limiter, the power amplifier will try to deliver 40.0 volts to the speaker system—this is equal to a power of 400 watts. Needless to say, the 100 watt amplifier will deliver an almost perfect square wave. Loudspeakers and ears do not like to reproduce or hear this type of signal.

A limiter whose threshold is adjusted to affect a gain reduction at the



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

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

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

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

CIRCLE 80 ON READER SERVICE CARD

2.0 volt point will begin to reduce the gain as the waveform exceeds the 2.0 volt level. The speed at which this gain reduction takes place will be determined by the attack time adjustment, and in most applications, this will be adjusted to the fastest possible point. If the limiter reduces the gain by 6.0 dB, or a division of two, the limiter output will be 2.0 volts and the power amplifier will be again quite happy. The release time should be adjusted so that the gain will not "come back" too quickly. The best adjustment of attack and release times is obtained by experimentation and the type of program material.

A properly adjusted limiter is one that performs the desired task and cannot be heard in the process. A poorly adjusted limiter makes itself known very quickly.

As to the second question, please consider the following. If you can hear the monitors and the singer cannot there are two possible explanations. You are in the pattern and the singer is not, or you do not have to listen to anything but the monitors. The singer is listening to himself through a very direct path from his vocal cords to his hearing mechanism. It will obviously require more acoustic power from an outside source for him to hear the monitors and himself at the same time.

—Lothar A. Krause, Jr.
Electronic Development Scientist
Peavey Electronics Corp.
Meridian, Miss.

Switching Sequence No Secret

I recently read an article in a well-known audio equipment merchandising journal stating that one should never turn on low-level electronics (mixers, EQs, etc.) after the power amplifier(s) are on. For some time now, I have switched on my mixers and limiter after the amp was turned on. Also, my noise reduction system and graphic EQ (the EQ is connected between pre and main on the amp) are plugged into the two switched receptacles of the amp for AC. By leaving the noise reduction and EQ switches in the "on" position, they get power by simply switching on the amp.

To start, is it wrong to leave these devices on and put the AC power to them by way of the switched outlets instead of engaging each of them separately? Also, is there actually a correct method or order in which each piece of gear should be switched on? If the answers to these two questions are yes, could you please explain the correct procedures for both and the possible consequences of incorrect engagement? Could the piece I originally referred to pertain only to the "pops" one may experience in the loudspeakers when the amp is on before switching the gear on and the damage that may result to the speakers if the amp is at high level? I have used the method I described for some time with absolutely no adverse effects. Also, I keep the main volume of the amp (a Sansui AU-5900) to full "off" until all the equipment is on. Can you please answer all these hows and whys?

—Randy DeFord
Logansport, Ind.

The reason for all equipment to be turned on before the power amplifier is very simple—to protect the power amplifier loudspeaker and your ears.

Much of today's equipment introduces voltage spikes at the output when the equipment is turned on or off. These voltage spikes (sometimes referred to as turn on/turn off transients) will cause a large "pop" in the loudspeaker if the power amp is on and the monitor volume is up.

There is no required order for turning on low-level equipment, but it is


real to reel...
ReVox
Mastering Tape 621

simply good practice to turn on your power amplifier last to protect your loudspeaker.

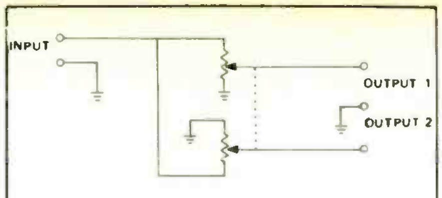
—Eldon Hall
Audio Consultant
Hollywood, Ca.

The Panoramic View

Just what is a panoramic potentiometer (commonly called a pan pot), electronically speaking—is this a switch or a circuit?

—Paul Castaldo
Boston, Ma.

A pan pot is a circuit with a single input and dual outputs. The input signal is divided and sent to the two outputs, either equally to each output, exclusively to one output or the other, or some intermediate ratio. A simple way to achieve this function is shown schematically in the accompanying diagram. (The schematic is for purposes of discussion only, and does not represent an actual working circuit.) Note the dotted line joining the "wipers" of the two variable resistors (which are shown connected as potentiometers). This line is to indicate that the wipers



BASIC PAN-POT CIRCUIT

are connected *mechanically*, not electrically, and that they move together. This mechanical connection can be accomplished by mounting the two potentiometers on the same shaft.

Further examination of the schematic will show that at either extreme of wiper travel one output receives all of the input signal while the other receives none. Also, any intermediate position will result in some signal at each output, the ratio of the two output levels being determined by the actual positions of the wipers. At one specific setting the two output levels will be equal.

Although there are other cues that our hearing system uses to determine the location of a single sound source, the most important cue, and the easiest to control in a stereophonic recording, is the balance of levels of a mix element in each of the two stereo channels. If a mix element is present in both stereo channels at the same level, that element will be heard as if it were located in the "center" of the stereo field. If the left stereo channel is carrying more of a particular mix element, a vocal for example, than is the right channel, then the vocal will be heard as if it were located somewhere to the left of the center of the stereo field. If the vocal signal were exclusively in the left channel, then of course it would be heard coming from the extreme left side of the stereo field.

When the outputs of a device similar to the one described in the opening paragraphs are connected to the left and right channels of a mixing board that is being used to create a stereophonic recording or mixdown, the device is then a means of "locating" a single mix element (*one* microphone signal or *one* track) in the left-to-right stereophonic panorama.

—Peter Weiss
Contributing Editor
Modern Recording

A NEW PREAMP FOR THAT DISCERNING PERFECTIONIST WHO CAN APPRECIATE THE DIFFERENCE.

The new Phase 3000 Series Two was designed for that discerning music-lover who has a passion for accurate sound, an eye for elegant yet functional design, a feel for craftsmanship, and an unfailing determination to maximize return on investment.

The Phase 3000 incorporates the latest technological advancements in preamp design. Transient overloading that plagues preamps has been virtually eliminated whether amplitude, frequency, or slew induced. Now you can enjoy the flexibility performance and features that are priced substantially higher in other equipment.

CMOS LOGIC MEMORY SYSTEM

Most preamps use dated mechanical switching devices that force signals to travel long, noisy, circuitous routes from the inputs, to the front panel, then back to the outputs. *Ours doesn't.* The Phase 3000 uses CMOS-digital logic to energize switching relays located where they belong, at the input jacks. This shortens critical signal paths. Noise, hum and the "crosstalk" that's characteristic of mechanical switching is virtually eliminated.

WANT MORE?

A listening session with a pair of headphones will convince you just how much of a difference a true headphone amp makes. Turn the 3000 around, and see how easy it is to patch in your noise reduction unit.

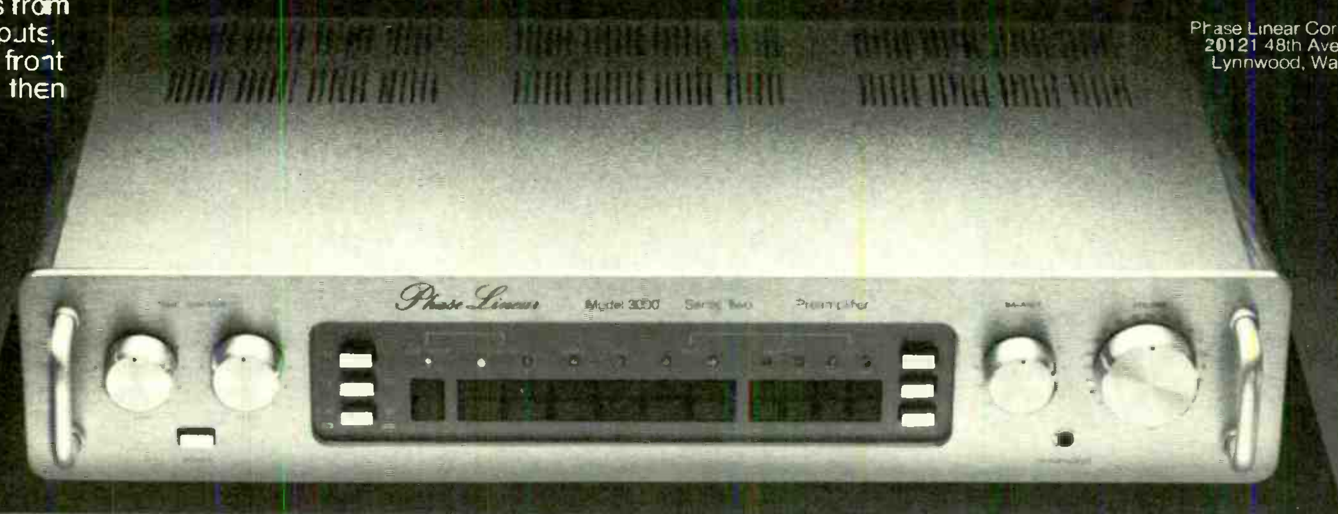
Two complete taping circuits allow you to copy between decks while listening to another source.

But we've done enough talking. If you're serious about state-of-the-art performance it's time for you to do some listening. See your Phase dealer.

SPECIFICATIONS:

Distortion: less than 0.04% (20Hz-20kHz). Typically 0.005% @ 1kHz.
Signal/Noise (IHF "A"): Phono 1—Moving Magnet: greater than 90dB re: 10mV input. Phono 2—Moving Coil: greater than 78dB re: 1mV input
Frequency Response: Phono-1/ Phono-2 deviation: ±0.3dB
Tone Controls: High & Low Frequency controls with switchable turnover points.
Volume Control: 22-position precision attenuator with plus or minus 0.5dB tracking.
Low Filter: 18dB/octave below 15Hz.

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



PHONO CARTRIDGE FLEXIBILITY

The two independent RIAA Phono Stages eliminate all low-level switching. As a result, noise is reduced to theoretical limits. Phono 1 is designed for moving-magnet cartridges and has three selectable capacitance values. Phono 2 is used with moving-coil cartridges and has three selectable resistance values. The expensive outboard head amp usually required for a moving-coil cartridge is already built into the 3000.

Phase Linear®

THE POWERFUL DIFFERENCE

MADE IN USA. DISTRIBUTED IN CANADA BY H. ROY GRAY LTD. AND IN AUSTRALIA BY MEGASOUND PTY. LTD.

THE MACHINE THAT HOLDS THE WORLD TRACK RECORD.

The Tascam Series 80-8 has become the most popular 8-track multichannel recorder in the world. Its reliability has been proven in basements, garages, and recording studios everywhere.

The results produced on the 80-8 are a matter of record. Sometimes gold.

The 80-8 proved a new standard was needed. Eight tracks on half-inch tape. 15 ips only. This new format allowed us to create a combined record/reproduce head, with full frequency response in the sync mode.

The 80-8 proved multichannel recorders could be relatively easy to operate. Our Function Select buttons determine the record, monitoring and dbx* status. One button for each track.

The 80-8 proved that performance and versatility could be affordable. Signal-to-noise is better than 95 dB (weighted) with our integral dbx unit (Model DX-8). Once installed, it's totally automatic. And our new Variable Speed Control**lets you adjust 15 ips ±20% to solve tough cueing and timing problems or add creative effects.

The 80-8 is proving that in professional recording, results are all that count. Because to us, pro means results. On demand. For payment.

If you agree, see your Tascam Series dealer for the machine that can prove it. Because it makes sense to do business with the people who have the track record.

*Registered trade mark of dbx, Inc.
**Installation required: a new DC servo-controlled motor is included.

TASCAM SERIES

TEAC Professional Products

THE PRODUCT SCENE

By Norman Eisenberg

PARIS POWER COMES TO U.S.

Paris Power is the name of a French line of disco equipment now being shown in the U.S. Products include mixers, equalizers, special effects devices, power amps and cabinets. Highlighted in a recent announcement from the manufacturer, Comel, is the PR 1300 broadcast console, which, although primarily designed for local stations, is also claimed to have a place in "high-standard discotheques" in which a radio-style animation is preferred. Comel also calls attention to its PMP 402 mixer which features 100-mm travel sliders for phono fading that are equipped with microswitches allowing instant remote starting of turntables. The mixer accepts any type of source and every input is sensitivity adjustable. It includes two stereo phono inputs switchable into lines or mics, and two stereo tape inputs switchable into four mono mics. The DJ using this device also is offered a special input channel with separate tone controls, pan-pot, bass and presence filters, and an "auto-fade" system that automatically fades the music program leaving both hands free.

CIRCLE 1 ON READER SERVICE CARD

LEXICON OFFERS LOW-COST DELAY LINE

Designed for use in small sound-reinforcement installations and for pre-reverb in recording studios is the new Delta-T 91 from Lexicon, Inc. This device provides digital delay adjustable from 0 to 120 milliseconds, and—says Lexicon—it has all the performance specs and features of the costlier Model 92 including muting of audio outputs during power up/down sequences, automatic bypass, audio input and output transformers and more. The model 91 has a single output, while the model 92 has two outputs. Price is $985.

CIRCLE 2 ON READER SERVICE CARD

AUDIO-TECHNICA ADDS ACCESSORIES

Two new items for disc playback have been announced by Audio-Technica. One is a "Safety Raiser" for manual turntables. This device automatically lifts the tone arm from a record when the stylus reaches the final groove. Price is $19.95.

The other accessory is a "Disc Stabilizer"—a compact, solid brass disc in a rubber jacket that fits over the turntable spindle and on which the disc is placed. Its purpose, says A-T, is to compensate for unsupported areas of a disc (caused by warpage) which create subtle vibrations and sound-coloring resonances that distort the music. Price is $19.95.

CIRCLE 3 ON READER SERVICE CARD

ALTEC MODEL 14 CHALLENGES "SPEAKER MYTHS"

Altec Lansing's newest speaker system, the Model 14, is said to incorporate four recent product developments. Literature describing it attacks what Altec calls "myths" concerning speaker systems.

One of the product developments is an overload device known as Automatic Power Control which monitors power delivered to the speaker and automatically corrects any overload without shutting down the system. Also found in the Model 14 is the "Mantaray," Altec's constant-directivity horn which is said to create a larger listening "sweet spot." Associated with the horn is the "Tangerine" radial phase plug, designed to "essentially eliminate the problem of high-frequency loss found in conventional phase plugs."

The Model 14 also includes a dual-band equalizer for separate adjustment of middles and highs. In this way, says Altec, the Model 14 achieves the performance flexibility of a three-way system within the design simplicity of a two-way (woofer and tweeter) system. The former unit is a 12-inch driver; it is crossed over to the horn at 1500 Hz. The walnut, 30-inch high (by 21 inches wide and 16½ inches deep) enclosure is vented. The system weighs 77 pounds. Operational power range is 10 watts to 350 watts; long-term maximum acoustic output is 114 dB/SPL; rated response is 35 Hz to 20,000 Hz. Sensitivity is given as 95 dB/SPL measured at 4 feet with a 1-watt input.

CIRCLE 5 ON READER SERVICE CARD

LUX DEBUTS CASSETTES AND TWO DECKS

Novel features of Lux Audio's new line of cassette tapes include a skew adjustment for the playback head, an electronic time sensor, an extra-wide (7 mm) pressure pad, a spring retainer for the pressure pad and a guide roller system instead of the conventional guide pins. The time sensor will be of principal interest to owners of Lux's cassette decks, since it provides real-time (in minutes and seconds) readout of tape playing time rather than the conventional numerical index. Luxman cassettes will be available in standard (ferric-oxide) and in high-bias (cobalt-doped) formulations.

The two cassette recorders include the Luxman 5K50 and the K-12. The former is a three-head, three-motor stereo deck with provision for free factory-adaptation to metal-particle tape capability. The record head has user-adjustable azimuth alignment provisions. Circuitry features Lux's new (and reportedly exclusive) BRBS recording system. The letters stand for "Bridge Recording Bias and Signal" amps; and the system is said to virtually eliminate phase-shift and bias signal leakage, while also reducing IM and TIM distortions below conventional circuit levels. Also featured are plasma bar-graph record-level indicators with peak-hold display, the electronic tape counter with real-time readout (when used with the Luxman cassette) as well as 4-digit readout for conventional cassettes, and more. Price of the Luxman 5K50 is $1995.

Lux's other cassette deck is the K-12, a two-motor, two-head model usable with metal-particle tapes. It includes several of the 5K50's features, such as the signal display and direct-coupled DC amps for both record and play. IC logic controls and Dolby are included. Its price is $995.

CIRCLE 6 ON READER SERVICE CARD

SONY ELECTRONIC CROSSOVER

Sony's new TA-D88B electronic crossover is a modular design that allows a choice of twelve crossover frequencies in two-way, three-way or four-way speaker systems. Three different values can be used for the crossover frequency points depending on the plug-in modules selected. Relative output level of each band is adjustable via a detented front-panel control. Eight such controls are provided, one per band for each of the unit's two channels. The left and right channels are completely independent, and are isolated from the power supply and the front-panel controls. Filters are Bessel-function types which combine slopes that are steep enough (24 dB/octave) to avoid adjacent-band interference and accompanying IM, with smooth cut-off characteristics for audio continuity across the spectrum. The filters also are credited with exceptional phase linearity, as well as super-low distortion and excellent transient response.

Available crossover frequencies per module are: Module 1—140, 225 or 280 Hz; Module 2—500, 800 or 1000 Hz; Module 3—1250, 2000 or 2500 Hz; Module 4—5000, 8000 or 10,000 Hz.

CIRCLE 7 ON READER SERVICE CARD

TDK ANNOUNCES CASSETTE CARE KIT

TDK's line of tape accessories now includes the HC-05 kit which contains non-flammable, non-toxic head-cleaning fluid; mirror; brush; felt probes and applicator. All items fit into a standard cassette box. Price is $5.99. Other TDK items in this general area include a head cleaner cassette and head demagnetizer, level adjust tape, labels, and takeup reels in various sizes for open-reel decks.

CIRCLE 8 ON READER SERVICE CARD

HIGH END CASSETTE DECK FROM MITSUBISHI

Mitsubishi's new model DT-30 is described as a three-head, closed-loop, dual-capstan, servo-controlled cassette deck with an automatic spacing pause system and turntable synchronization capability. Distributed by Melco Sales of Compton, California and priced at $650, the DT-30 uses peak meters with a peak-hold switch that displays signal peaks for at least three seconds before changing. Right and left meter bias adjustments operate built-in oscillators at 400 Hz and at 8 kHz to facilitate fine-tuning for optimized response with any tape. When linked with the company's model DP-EC20 turntable, the DT-30 will begin to record as soon as the tone-arm contacts the disc surface. When the arm leaves the surface, the cassette deck goes into the pause mode. If the cassette deck is turned off, the turntable stops and its arm returns to rest. Frequency response of the DT-30 is rated from 30 Hz to 17 kHz with normal tape; up to 18 kHz with "special tapes;" and up to 20 kHz with ferrous-chrome tapes.

CIRCLE 9 ON READER SERVICE CARD

TWO NEW SCAMPS

Scamp model S-02 microphone preamp, and model S-100 Dual Gate have been announced by their British manufacturer, Audio & Design Recording. The preamp is a transformerless design that has 600-ohm line-amp drive on both outputs. Featured are a 30-dB pad and phase-reversal switch; high-pass filter; switchable aux send, pre or post; and 70 dB gain with "optimum modulation indicator."

The Dual Gate, used for low-level source noise reduction and automatic attenuation of non-contributing channels, has balanced inputs, LED indicators, and is keyable. Both devices are fairly compact and are styled vertically.

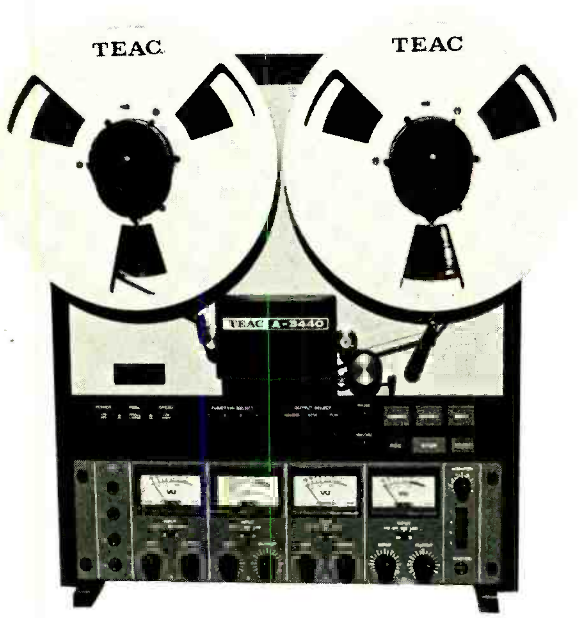
CIRCLE 10 ON READER SERVICE CARD

TEAC UPDATES OPEN REEL LINE

Planned for shipment nationally by this summer are updated versions of well-known Teac open-reel recorders. Succeeding the older 3340 series is the A-3440, a three-head, three-motor, two-speed (15 and 7½ ips) unit with a new transport, circuitry and logic boards. The new Simul-Sync mode is tied directly to the function-select position of each channel either for recording or overdubbing. Optional interface with pro curve dbx is available. Front panel controls are laid out according to "human engineering" concepts. The deck accepts up to 10½-inch reels and will cost $1500.

Two new reversing open-reel decks are the A-2300SR which is Teac's lowest-priced unit in 7-inch reel size ($800); and the A-3300SR, the firm's least expensive unit for 10-inch reels ($1,050). Both models feature a three-motor transport, manual cueing, feather touch control buttons, mic/line mixing, separate bias and EQ switches and both also sport large VU meters.

The fourth updated model is an improved version of Teac's half-track, two-channel stereo deck, the A-6100. Now to be known as the A-6100 Mk II, it will include tension servo, a timer function, NAB/IEC switchable equalization, a 0/20-dB mic attenuator, an impedance roller and the RC-70 remote-control unit. This deck is priced at $1350.



METAL TAPE COMING ON

A few issues back in this column, reporting on some recent developments in recording tape, we asked the question: "So what's happened to metal tape?"

We've been getting some answers. Judging from trade sources, this new tape and the hardware for using it are beginning to shape up as a potentially major trend in recording. The 3M Company, of course, was the first to announce a fairly comprehensive line of metal tapes. Other brands now

known to be moving into this area include BASF, Fuji, Maxell and TDK. Rumor has it that most others will follow suit. At the same time, of course, no one is giving up on oxide-coated tapes.

In recorders, two names that I recall as among the first to become actively involved with metal tape are Nakamichi and Tandberg. Both firms now have definite products for handling metal tape. In addition, there's news of several other recorder brands getting into the act. The JVC KD-A8; the B.I.C. T-4, the Technics RS-M95, the Rotel RD-2200, the Onkyo TA-680D, the Sanyo RD-5372, the Luxman 5K50 and K-12 all are recently announced cassette recorders with metal tape capability. At least one company—Eumig—has announced a metal tape adapter kit to be available for its existing CCD recorder.

Prices for the new decks vary considerably since the metal tape option may or may not be used in conjunction with all the other performance options and features (e.g., number of heads, type of transport, etc.) possible within the cassette format. This means that a prospective buyer will be able not only to choose the metal tape capability, but along with it will be offered pretty much the same range and assortment of controls, features, etc. now found on existing cassette recorders. With this new product trend apparently under way, we will of course test-report the new models as they become available.

The tapes themselves seem to be following a more consistent price trend. In general, they are running a bit less than double the cost of present high-performance metal oxide tapes.

MUSICAL NEWSICALS

MUSICAL INSTRUMENT AMPLIFIERS

Electro-Voice has introduced two bass guitar speaker systems, designated the B115-M and the B215-M. Both models use the EVM-15B 15-inch speaker, one in the case of the B115-M and two in the B215-M. Both models also feature a vented cone midrange driver for a much improved high end over that of conventional, one-way bass guitar speaker systems. The high frequency response is adjustable to the musician's preference with a four-position rotary switch on the front of each unit. The single woofer unit has the clearer, crisper sound preferred by many jazz and studio players and has a power handling capacity of 200 watts, while the B215-M has a heavier sound and will handle 400 watts, making it the choice for most hard rockers. Both models have enclosures constructed with black, vinyl-covered three-quarter-inch plywood, aluminum edging strips and a metal mesh grille. For convenient handling, the cabinets have integral handles, and the larger model has heavy-duty casters.

CIRCLE 12 ON READER SERVICE CARD

SYNTHESIZERS

Tama has just introduced the Snyper Percussion Synthesizer, Model DS 200. Unlike the various other percussion synthesizer systems available, the Snyper does not use special control pads or "pseudo-drums" as the control elements; rather, the drummer's own drums are used to key the effects. The Snyper is a two-channel synthesizer with full and independent controls for each channel. Two special pickups are provided with the Snyper, one for each channel. These pickups are attached to either the top or bottom head of any conventional snare drum, bass drum or tom-tom using either a double-faced adhesive material or special rim-mounting clamps, both of which are provided with the system. The two pickups would normally be attached to two different drums, but both pickups may be attached to the same drum for a unique dual synthesis effect. The output of each pickup triggers a tone generator in the electronics unit. Each tone generator has a tuning control which has some 7½ octaves of range from 30 Hz (lower than the low E on an electric bass) to 6 kHz. Each tone generator has a built-in frequency sweep triggered by the pickup signal; controls are provided for Sweep Range, or how wide a frequency range is swept for a given note, and Sweep Time, which is how fast the sweep will return to its fundamental frequency after being triggered. When triggered, each tone will sustain for as much as five seconds; this parameter is controlled by a Decay Time control. The Snyper is fully touch-sensitive—the harder the drum is struck, the louder the synthesizer output and the wider the fre-

quency sweep range will become. Each pickup has its own sensitivity control, and since each channel of the Snyper has Sweep Range and Output Level controls, a wide range of dynamic effects are possible. A pushbutton switch on the front panel of the electronics unit switches the effect on and off, or this may be done with a remote footswitch which is provided with the synthesizer system.

CIRCLE 13 ON READER SERVICE CARD

The multi-effect String Performer has been introduced as the new top model of the extensive line of keyboard instruments available from M. Hohner. The String Performer produces the polyphonic voicings of piano, clavichord, cello, violin, and viola, and solo voicings of brass, strings, and clarinet. The unit has a 61-note, five-octave keyboard which is split into two bass octaves and three treble octaves which are independently assigned to the various voices with individual function controls and slide bars for the solo voices. An unusual feature is an octave-splitting preset which lowers the solo voices by one octave when engaged. The unit has four outputs enabling the separate amplification and/or processing of the various voicings. The string performer is powered from a 20/24 volt AC adapter, and has a tilt-leg assembly available as an accessory.

CIRCLE 14 ON READER SERVICE CARD

MUSICAL INSTRUMENT PICKUPS

What promises to be a major breakthrough in guitar pickups has been announced by a company called FACT (Frequency Analysis & Circuits Technology) Laboratories. FACT has taken what may be the ultimate high-technology approach to pickup design. The concept is that by analyzing a sample of the particular sound the player wants to duplicate and the sound of the guitar and pickups he normally uses, the FACT computer will come up with the specific pickup design that will produce the desired sound when installed on the musician's guitar. According to the FACT announcement, it will work like this in practice: the musician will bring a record or tape recording of a guitar solo that has exactly the sound he is trying to duplicate; the FACT-trained salesman will play the recording on the FACT computer, which will analyze and display the characteristics of the sound; then the musician will play his own instrument into the FACT computer, which will analyze its sound and prescribe the particular pickup which will come closet to the recorded sample; the musician then will leave a deposit and send the computer readout to FACT, and receive his pickup from them in 14 days or less. According to FACT, this analysis and testing procedure will eliminate the disappointment of installing a pickup just like a favorite star uses only to find that it sounds different on your particular guitar. As far as the computer is concerned, FACT says they will be loaning the units, called SCARF systems, to franchised dealers and that the units should be in selected music stores as you read this. If it all works as it's supposed to, it will be truly revolutionary.

CIRCLE 15 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

RolandCorp US has introduced two sophisticated computer rhythm units which were specifically designed to overcome the "mechanical" feel common to most other rhythm units. The CR-68 and CR-78 accent and syncopate different voices to add feeling to the basic rhythm. In addition, there is a variation or "fill" mode which automatically introduces one of 11 rhythmic variations into the basic pattern; in the automatic mode, these "fills" will be inserted every 2, 4, 8, 12 or 16 measures, while in the manual mode "fills" will only be initiated when the user pushes a button on the panel or a footswitch. Twenty rhythms are provided which include two disco beats and many rock variations. In addition to these features, the CR-78 includes a programmer which allows the user to pre-program four separate rhythm tracks using up to eleven instrument voices each by tapping out each part on a special programming pad. The CR-78 also features a built-in mixer for three add-on voices and a fade in/fade out device, as well.

CIRCLE 16 ON READER SERVICE CARD

Multivox/Sorkin Music has released two new analog delay lines. The MX-99 is a compact model which uses advanced analog circuitry to produce a wide range of effects that belie the unit's dimensions. The MX-99 offers controls for delay time, echo repeat, echo EQ and input gain, and has provision for an optional echo defeat footswitch. The MX-D5 is a sophisticated, rack-mount delay line with a built-in spring reverb. Delay times are variable from 20 to 200 milliseconds and the unit has separate controls for delay time, echo repeat, echo EQ and reverb level. A peak-reading, five level LED meter insures optimum level matching, and there is provision for separate defeat footswitches for echo and reverb.

CIRCLE 17 ON READER SERVICE CARD

Polyfusion Inc., the synthesizer people, have introduced a battery-operated, active volume control pedal. The unit is designated the FP-2, and can be used to control the volume of any amplified instrument. The circuitry of the FP-2 includes a high impedance input buffer for minimum loading of the instrument's output, a low-noise potentiometer and an output line amplifier, and is unity gain so that there is no undesirable loss in signal level. The circuitry was designed for ultra-low power consumption, allowing several months of use from a single 9-volt alkaline battery. The FP-2 is constructed from heavy gauge steel for maximum ruggedness and reliability.

CIRCLE 18 ON READER SERVICE CARD

THE BEST VALUE IN A PROFESSIONAL TAPE RECORDER

When you compare a tape recorder, here are the most important areas to consider for value, quality, and sound.

PERFORMANCE:

Overall Signal-to-Noise: 66 dB unweighted at 520 nWb/m (30 Hz to 18 kHz audio filter).

Playback Signal-to-Noise (electronics): 72 dB unweighted (with audio filter).

Headroom: +24 dB. Maximum Output: +28 dBm.

Overall Frequency Response (15 ips): 30 Hz to 22 kHz ±2 dB.

Playback Frequency Response (MRL test tape): 31.5 Hz to 20 kHz ±2 dB.

RELIABILITY: An unmatched 4-year track record of on the job performance for the original compact professional recorder. Day in, night out. Just ask someone you trust.

ALIGNABILITY: Any tape recorder must be aligned to achieve maximum performance. With the MX-5050-B, all primary alignments including record bias and level are on the front panel. So is a 1-kHz test oscillator. Secondary alignments are inside the bottom panel. You or your maintenance people can align it fast and easy. This saves you time, money, and enhances your reputation.

INTERFACEABILITY: With a flick of the output switch you can plug-in to any system: +4 dBm 600 ohm or −10 dB high impedance. No line amps or pads to mess with. A perfect match everytime.

ADDITIONAL BENEFITS: Three speeds, dc servo ±7%, ¼ track reproduce, full edit including indexed splicing block, over-dubbing, noise free inserts, XLR connectors, NAB/CCIR switching unique 3-position alignment level switch.

PRICE: Suggested retail price $1,945 (USA).

MX-5050-B: THE CHOICE IS OBVIOUS

OTARI®

Call Ruth Pruett on 415/593-1648 for the name of your nearest Otari professional dealer.
Otari Corporation, 981 Industrial Rd., San Carlos, CA 94070 TWX 910-376-4890
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PLAY
STOP
EDIT
PHONES
TEST OSC
VU
VU
RECORD BIAS
RECORD LEVEL
PITCH CONTROL
SLOW
FAST
CUE

Interfacing Auxiliary Equipment:

Where and Why?

Part Two

By Larry Blakely

Many people who have recently started their career or hobby in recording are usually puzzled regarding the use of auxiliary recording equipment. The questions often asked are like a double-edged sword: "Why do I need it and where do I connect it?" This article deals more with the objective "where" question than with the subjective "why," but, as you read, some reasons for why you need auxiliary recording equipment should also become apparent.

In the first half of this article [last month] I mentioned that a recording engineer needs to understand signal flow. The signal flow of any recording console can be obtained from the console schematic or block diagram. The first part of this article also went into great detail on symbols and terms commonly used in schematics and block diagrams, and an explanation of the signal flow through a typical recording console was detailed. Trying to use recording equipment without having the ability to read and understand a console schematic or block diagram is like taking a journey into a strange land without having the knowledge or ability to read a road map. Being able to read a console schematic or block diagram and having an understanding of the signal flow will prove to be one of your strongest assets in your entire career of recording.

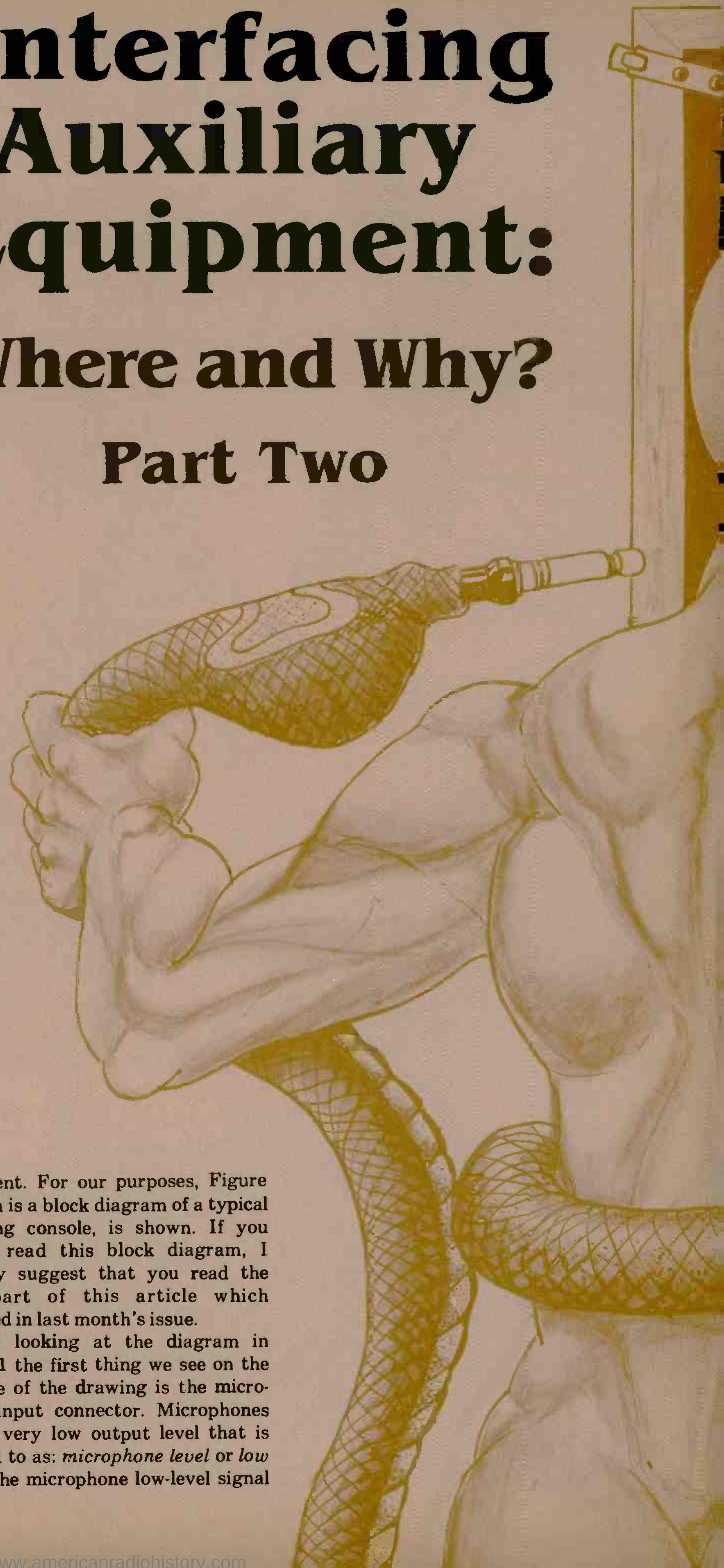
It is difficult for one to reach an intended destination if he doesn't know how to get there. If you can read a schematic or block diagram you will know how to get there.

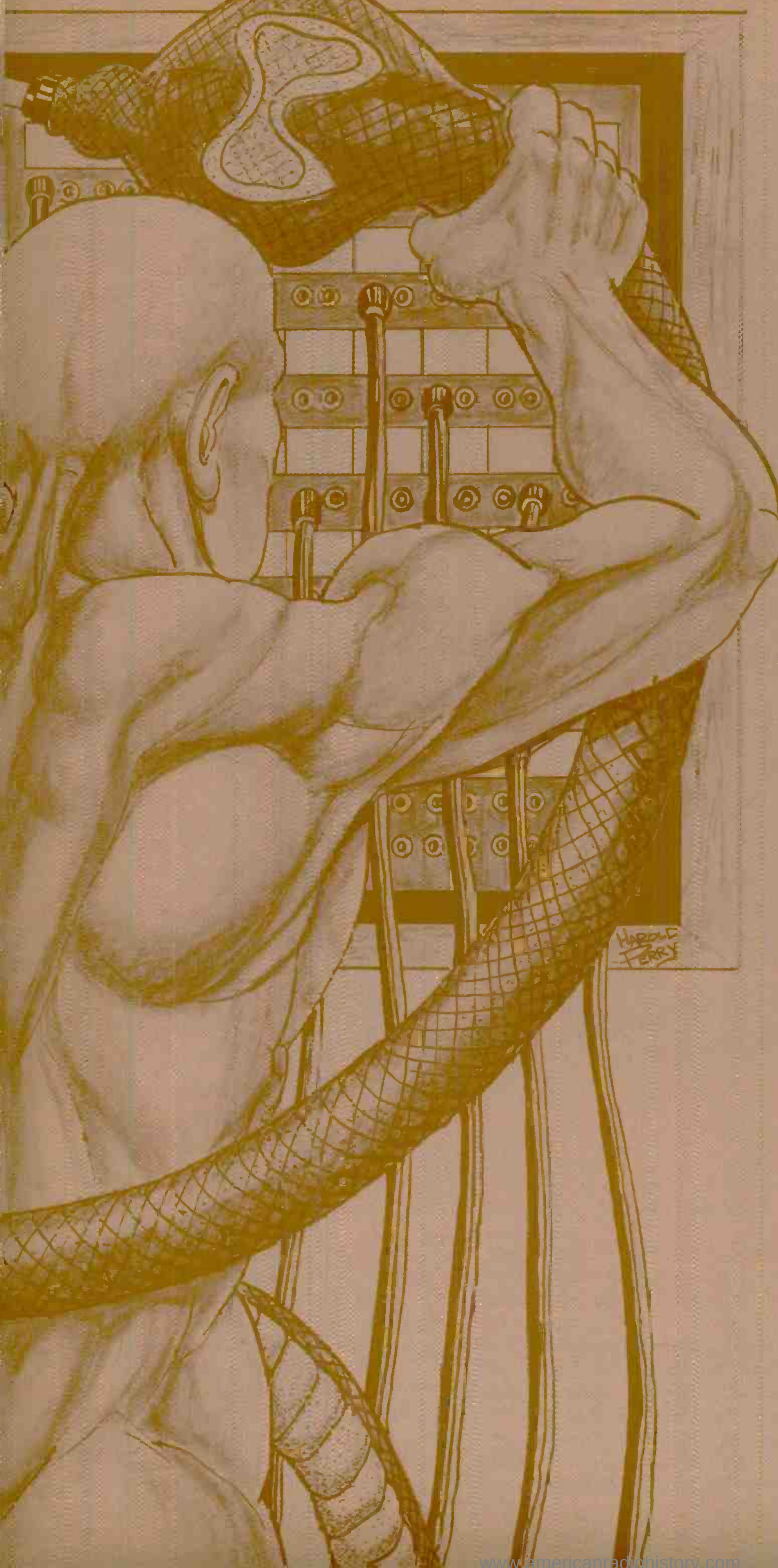
Patch Points were discussed in the first part of this article. These are places where auxiliary equipment can be connected to the recording console. There are typically a number of places in a recording console that have patch points. Each patch point appears at a different place in the signal flow of the recording console; therefore, various effects can be achieved when using the different available patch points, according to your particular needs.

Scrutinizing Points

At this time we will look at each point typically found on a recording console and observe what can be done at each of these patch points with various types of auxiliary recording equipment. For our purposes, Figure 1, which is a block diagram of a typical recording console, is shown. If you cannot read this block diagram, I strongly suggest that you read the first part of this article which appeared in last month's issue.

When looking at the diagram in Figure 1 the first thing we see on the left side of the drawing is the microphone input connector. Microphones have a very low output level that is referred to as: *microphone level* or *low level*. The microphone low-level signal





is amplified (increased in level) to a line-level signal by the microphone preamplifier; there, the output of the microphone preamplifier will have a line-level signal. Virtually all auxiliary recording equipment operates at line level, *not microphone level*.

When to Use *Mic-Pre Out and Line In*

The first place we usually have access to a single microphone at a line level is at the microphone preamp output. This patch point is normally labeled on the patch bay as "mic pre out" [microphone preamplifier output] or "pre out." Typically, another patch point will be located below it and will be labeled "line in." These two patch points give you access to the microphone signal output (now line level) and then access to the position fader through the line in patch point. These two patch points are shown in Figure 2 as normalling patch points, i.e., when nothing is patched in, the signal flows normally. When something is patched into mic pre out the signal path is broken and will flow to the input of the auxiliary device that is patched into that patch point. The console signal flow is now broken. To regain access to the console signal path the output of the auxiliary device must be patched into the line in patch point to complete the signal path so the signal can continue to flow through the console. In Figure 3, an auxiliary device is patched into mic pre out. Therefore the mic pre out signal is routed to the auxiliary device input. The auxiliary device output is not connected and the signal reaches a dead end. In Figure 4 we see that the output of the auxiliary device is patched into the line in patch point and now the signal will flow through the rest of the console. It can be seen that the microphone is amplified by the microphone preamplifier and is then routed (via the patchcord) to the input of the auxiliary device and that the output of the auxiliary device then routes the signal back to the line in patch point to complete the console signal path.

Suppose you had one microphone on a vocal solo and four microphones on background vocals and each microphone occupied its own input position on the console. Likewise, all of the above microphones are assigned to the same output channel. The vocal solo has some loud screaming parts that overload the recording console, while the background vocals have no prob-

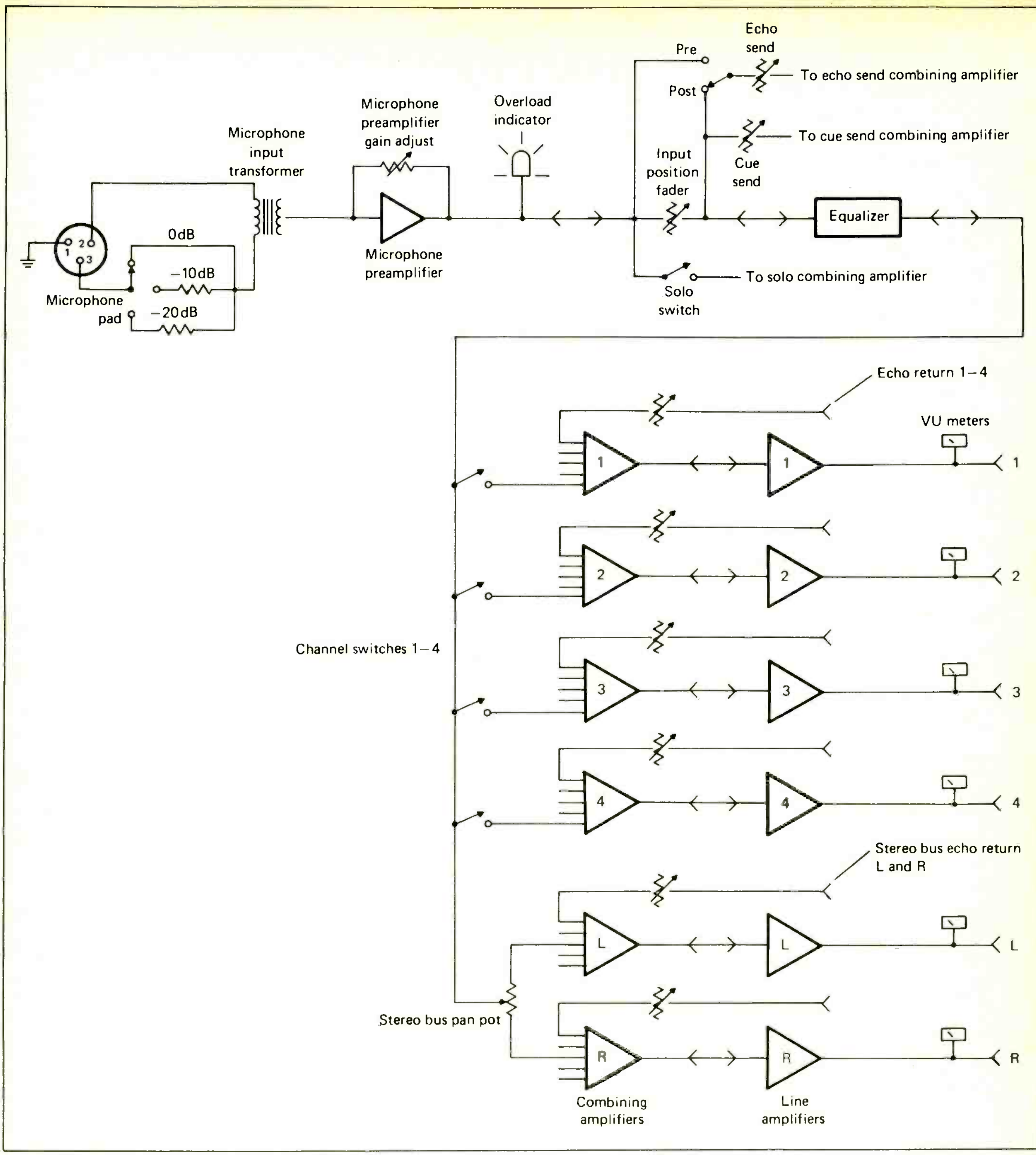
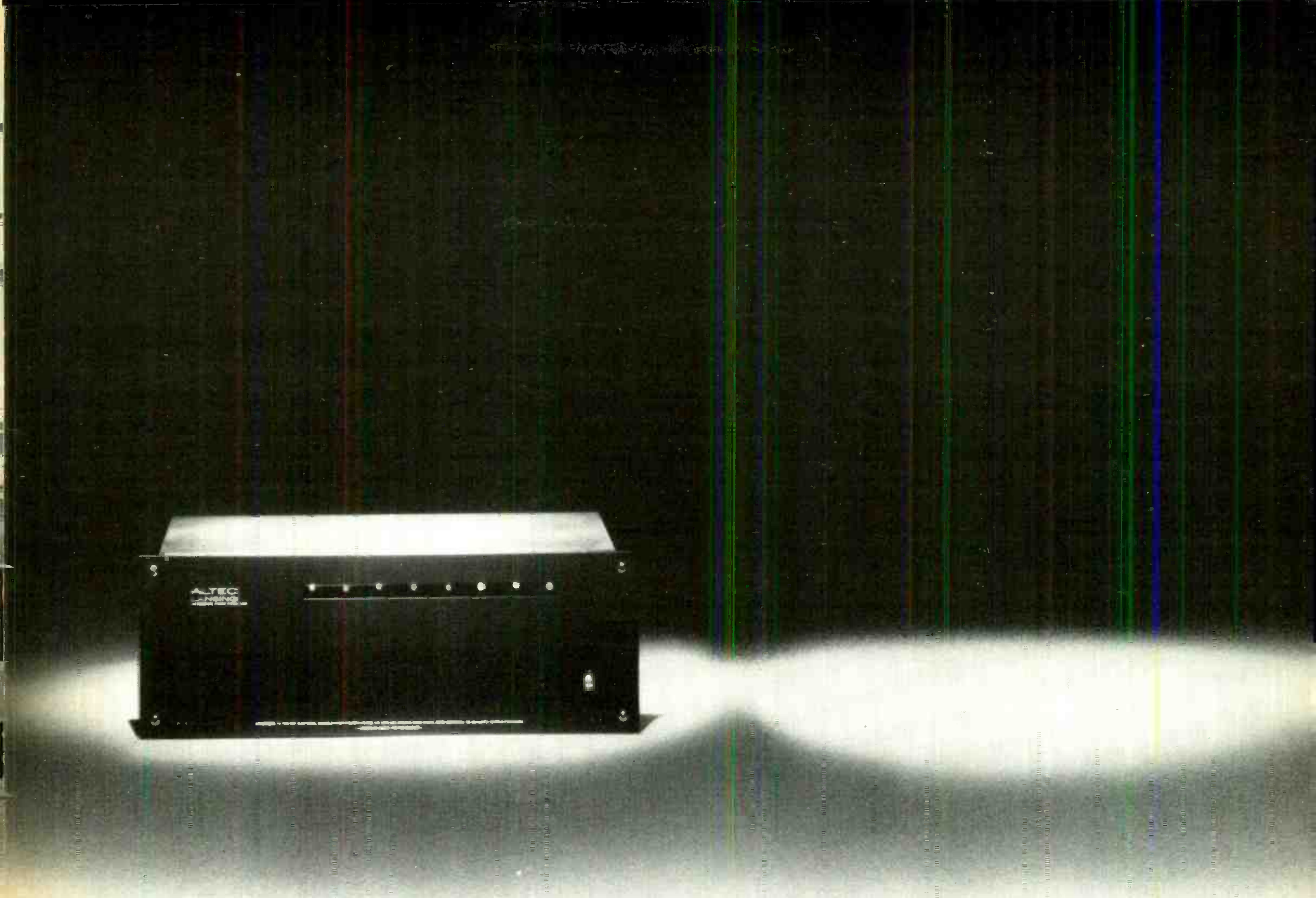


Fig. 1: Typical recording console block diagram.

lems with excessive level at all. You have a compressor/limiter. Where do you connect it? If you connect (patch) the compressor/limiter at the console output channel the compressor/limiter will compress the vocal solo and background vocals simultaneously. However, if we were to patch the compressor/limiter into the mic pre out and line in patch points on the console input position occupied by the vocal solo microphone we could then compress the vocal solo *only* and the compression would have no effect whatsoever on the background vocals. Other specialty devices like parametric equalizers can also be patched in at this point. The mic pre out and line in patch points can be used when you wish to process (via auxiliary device) one individual microphone without affecting any other microphone signals.

When to Use *Line In*

The most common use of line in patching is for mix down from a multitrack tape recorder. For example: If you had an eight-track recorder you would patch the playback outputs of your tape recorder 1 through 8 to the line inputs of your recording console 1 through 8, respectively. Notice in Figure 1 that patching to the line input of a recording console will give you



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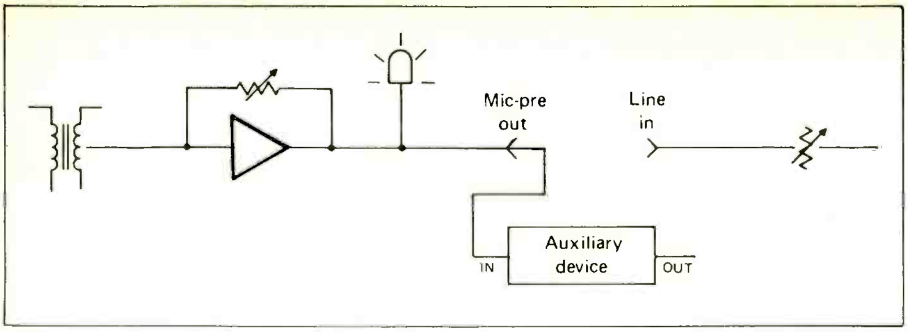

Fig. 2: Patch point normal broken; signal won't flow through console.

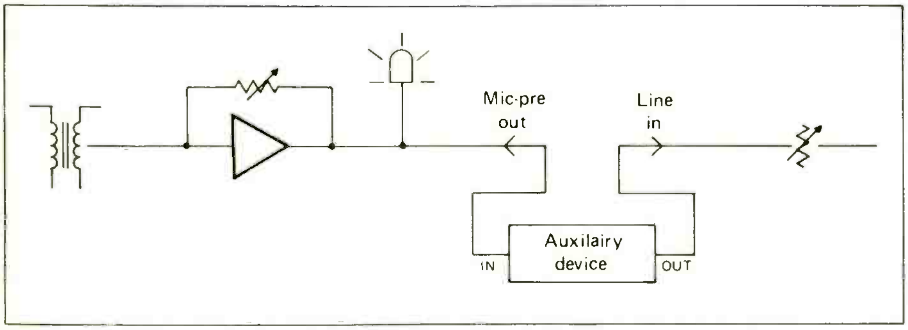

Fig. 3: Patch point normal broken; signal flows through auxiliary device.

access to the entire console minus the microphone preamplifer section. Line-in patch points can also be used for sound effects from additional tape recorders, turntable playback (for turntables with built-in preamplifiers that will provide a line-level output), echo returns, etc.

When to Use *Fader Out*

If you were doing a mixdown the line-in patch points would be occupied by your tape recorder outputs, so if you wished to patch a compressor/limiter in on one specific track it could be done at the fader output shown in Figure 5. I must point out that any reduction in level by the console input position fader will reduce the input level to the compressor/limiter or other auxiliary device. If you don't have a fader-out patch point on your console, the same thing can be accomplished by connecting the recorder output to the compressor/limiter input and the compressor/limiter output to the console line input as shown in Figure 6.

When to Use *EQ In and Out*

This feature is usually found on the more expensive recording consoles and it does have its advantages. For example, if a console had equalizers with selectable midrange frequencies but only one midrange frequency at a time could be used, and you needed to use two midrange frequencies simultaneously, you could cascade two equalizers as shown in Figure 7. Now both midrange frequencies could be used simultaneously inasmuch as there are now two equalizers on that input position. (However, I must warn you that when equalizers are cascaded in this manner they can sometimes add unwanted noise).

Equalizers are rarely found on output channels. Another place that patchable equalizers can be used is to add equalization to an output channel. But, if you refer back to Figure 1 you can see that there are patch points between the combining amplifier output and the line amplifier input that are available for this purpose. Such a patching arrangement can be seen in Figure 8. This will obviously give you a non-working input when you remove an equalizer, but there is even a way around this. Simply grab another patch cord and connect it from fader out to combining amplifier in. Now the input position will function normally, less the equalizer, of course. Also, if you have a noisy equalizer that may make an input position unusable and need to use that input position, you can put a patch cord from EQ in to EQ out which will bypass the equalizer with the wire in the patch cord and thereby remove the equalizer and the noise from the circuit.

When to Use *EQ Out and Combining Amp In*

Frequently, a vocal or instrument during "live" recording or an individual track during mixdown may need a large amount of equalization and limiting for a special effect. For such a situation it is better that the limiter *follow* the equalizer inasmuch as the equalizer does increase or decrease level at certain frequencies and will affect the operation of the limiter.

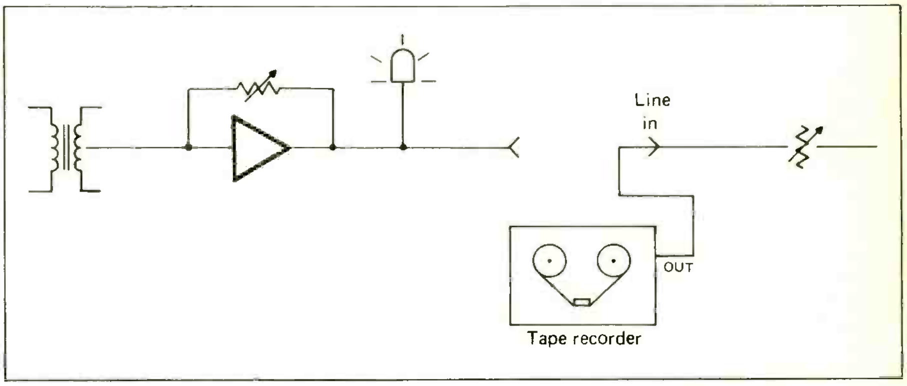

Fig. 4: Patching to line-in for mixdown from multi-track recorder.

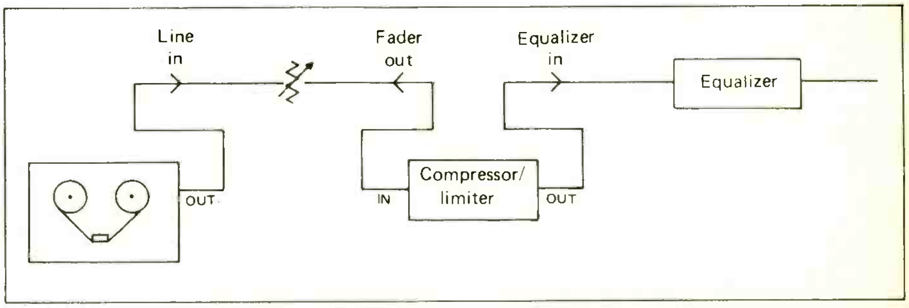

Fig. 5: Patching from fader out to EQ in.

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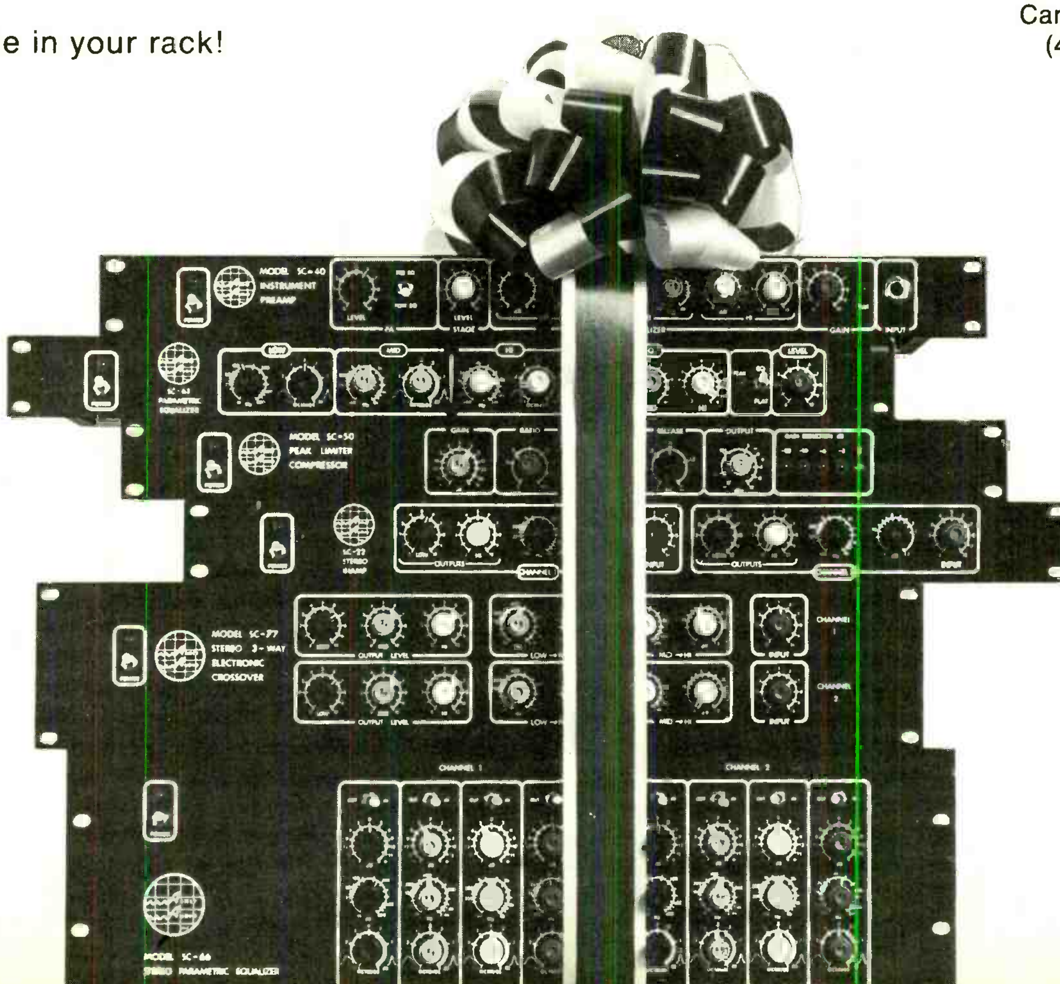

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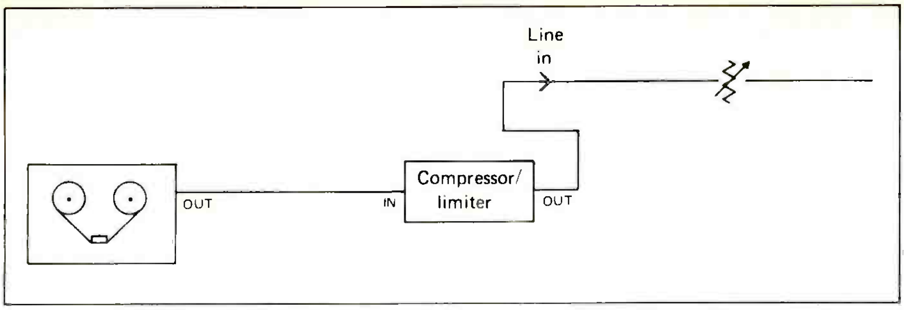

Fig. 6: Using a compressor/limiter for one tape track when doing a mixdown.

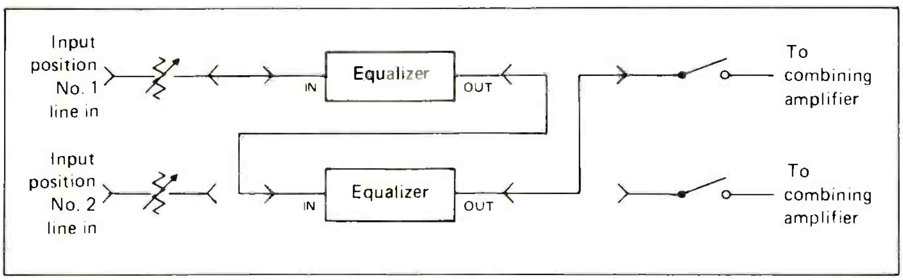

Fig. 7: Cascading two equalizers by using patch points.

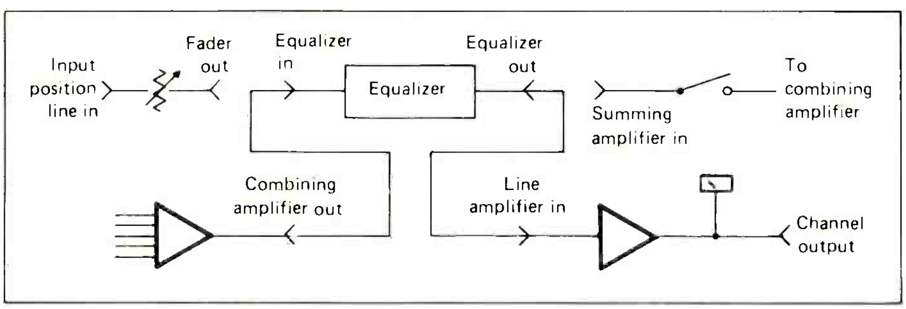

Fig. 8: Patching input position equalizer to an output channel.

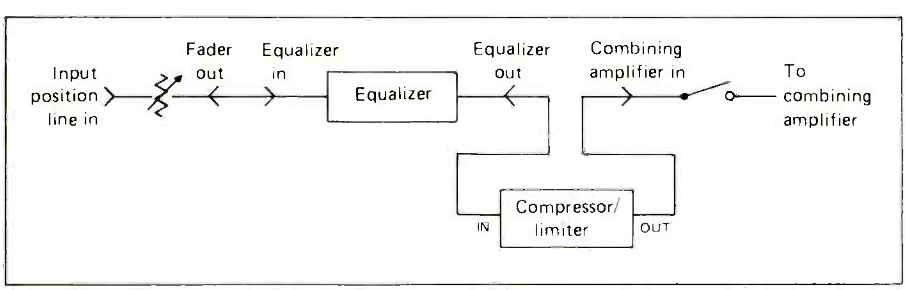

Fig. 9: Patching a compressor/limiter from EQ out to combining amp in.

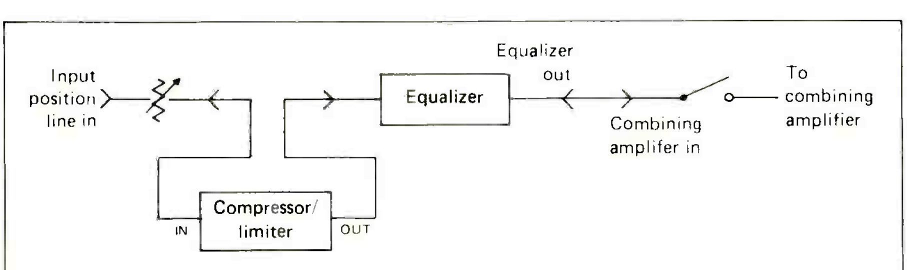

Fig. 10: Patching a compressor/limiter from fader out to EQ in.

I would suggest that the limiter be patched as shown in Figure 9—between the EQ out and combining amp in. However, if you desire to do your limiting before the equalizer (pre-equalizer) you can patch as shown in Figure 10. There is no hard and fast rule as to which way you should patch your compressor/limiter. If you have the patch points available to you, try both ways and see which gives you the results you desire. For one type of vocal or instrument you may find limiting pre-equalizer to be most effective while for yet another type of vocal or instrument you might find limiting post-equalizer to be most effective.

When to Use *Combining Amp Out and Line Amp In*

All of the patch points up to here have only affected the signal on a particular input position. This *usually* means auxiliary devices are used on single vocals or musical instruments. Usually a number of inputs are assigned to an output channel. In the example used earlier there was a vocal solo on one microphone and background vocals on four additional microphones. This requires five input positions and each microphone signal can be processed individually with inboard equalizers or auxiliary equipment via patch points. Oftentimes after a vocal solo and background vocals are combined together on a single output channel one may desire to add a little EQ or limiting to the output channel. You can indeed process the composite signal on the output channel rather than the individual signals. This can be done by patching after the combining amplifier as shown in Figure 11. If your console does not have patch points for combining amplifier out and line amplifier in you must connect the auxiliary device as shown in Figure 12.

Another common practice is to use a *stereo* compressor/limiter on a stereo output bus during a mixdown. This would be connected or patched as shown in either Figures 11 or 12. Of course the left channel would be patched to the left channel of the limiter and the right channel would be connected to the right channel of the limiter. It is important to note that only stereo limiters or single channel limiters that can be connected or strapped for stereo operation should be used for this purpose. If two single channel limiters are used for stereo limiting without being coupled there will most likely be an image shift in the

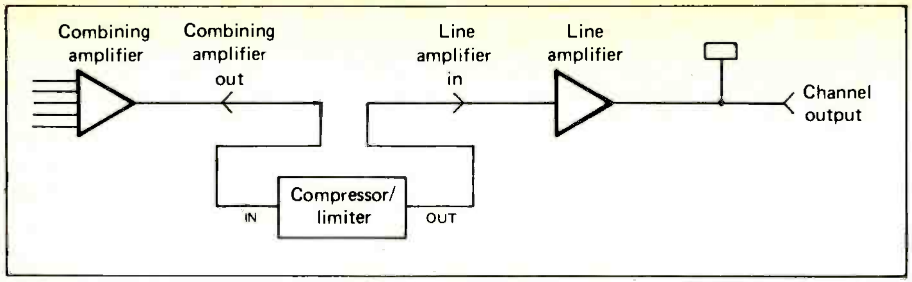

Fig. 11: Patching from combining amplifier out to line amplifier in.

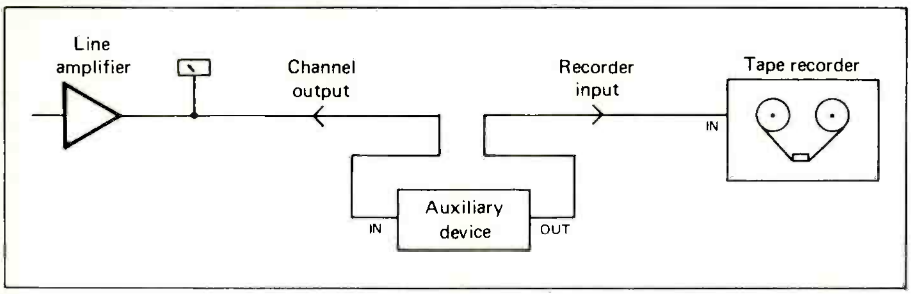

Fig. 12: Patching an auxiliary device from a channel output to a recorder input.

stereo perspective. In reference to Figure 1, there is a stereo output bus with access via patch points to the left and right combining amplifier outputs and access also to the left and right line amplifier inputs.

When to Use the *Echo Send Output*

A very common practice is to use equalization on the echo send signal. The equalizer is connected to the console echo send output and to the echo device input as shown in Figure 13. Delay lines or tape machine delay are also commonly used in the echo send signal path, and they would be patched or connected in the same manner as shown in Figure 13.

When to Use *Echo Return Inputs*

Sometimes equalization is used on the echo return instead of the echo send. Some engineers use expanders on the outputs of noisy echo devices to reduce the noise. However, the use of an expander on an echo return will also reduce the decay time of the echo device as well. If any auxiliary devices are to be connected in the echo return signal path the connections would be made as shown in Figure 14.

Wide Range of Equipment

There are many variations and combinations of auxiliary equipment hookups that can be done with patching and a good working knowledge of signal flow and the ability to read and understand recording console block diagrams or schematics.

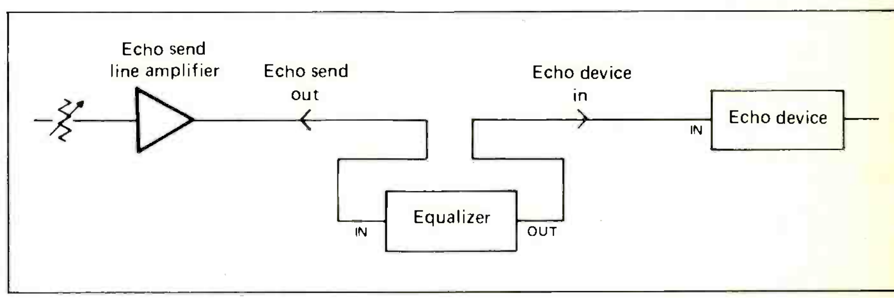

Fig. 13: Patching an equalizer from echo send out to echo device in.

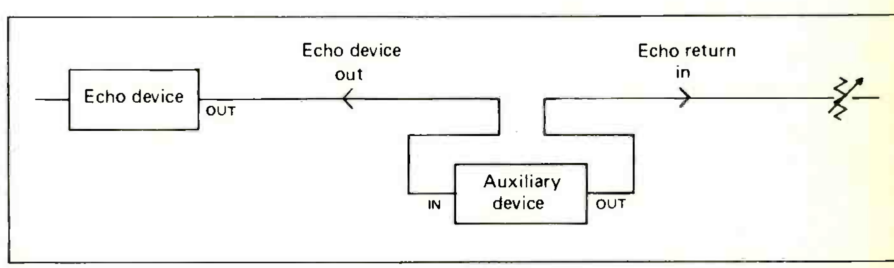

Fig. 14: Patching auxiliary equipment into the echo return signal path.

Throughout I have talked of compressor/limiters or equalizers for most of the examples of patching. If an equalizer is shown in one particular figure, I do not mean to imply that that is the only such auxiliary device that can be connected at that point. Any auxiliary device can be used at any patch point as long as the auxiliary device has line level inputs and outputs and has appropriate input and output impedances for your recording console. Some recording consoles will interface with auxiliary equipment with output impedances of typically 5,000 ohms. Check your console instruction manual for the recommended input and output impedances of auxiliary equipment. Most of the available auxiliary equipment will also be below 500 ohms output impedance and 10,000 ohms or greater input impedance. These input and output impedances will work with most any recording console. Most available auxiliary equipment is also line level in and out. If you have any questions regarding this, call the customer service department of the console manufacturer or of the auxiliary equipment manufacturer, or talk to your local recording equipment dealer.

A wide range of auxiliary equipment is available on the market. There are phasers, flangers, Harmonizers, filters, equalizers, delay devices, echo devices, compressor/limiters, expanders, etc., available to patch into your recording console. Each auxiliary device has some special effect it will perform or some special job that it will do.

It will be necessary for you to play with these different types of auxiliary devices personally to know and experience what they will do. The use of various auxiliary devices will enable you to add additional tonal colors, special effects and operating controls to your recordings.

The reading and thorough understanding of this two-part article will prove to be one of the most valuable and useful pieces of information you will ever obtain. If you don't understand something, go back and read that part again until you do understand. A whole new world of versatility and flexibility will now be open to you. Happy patching!


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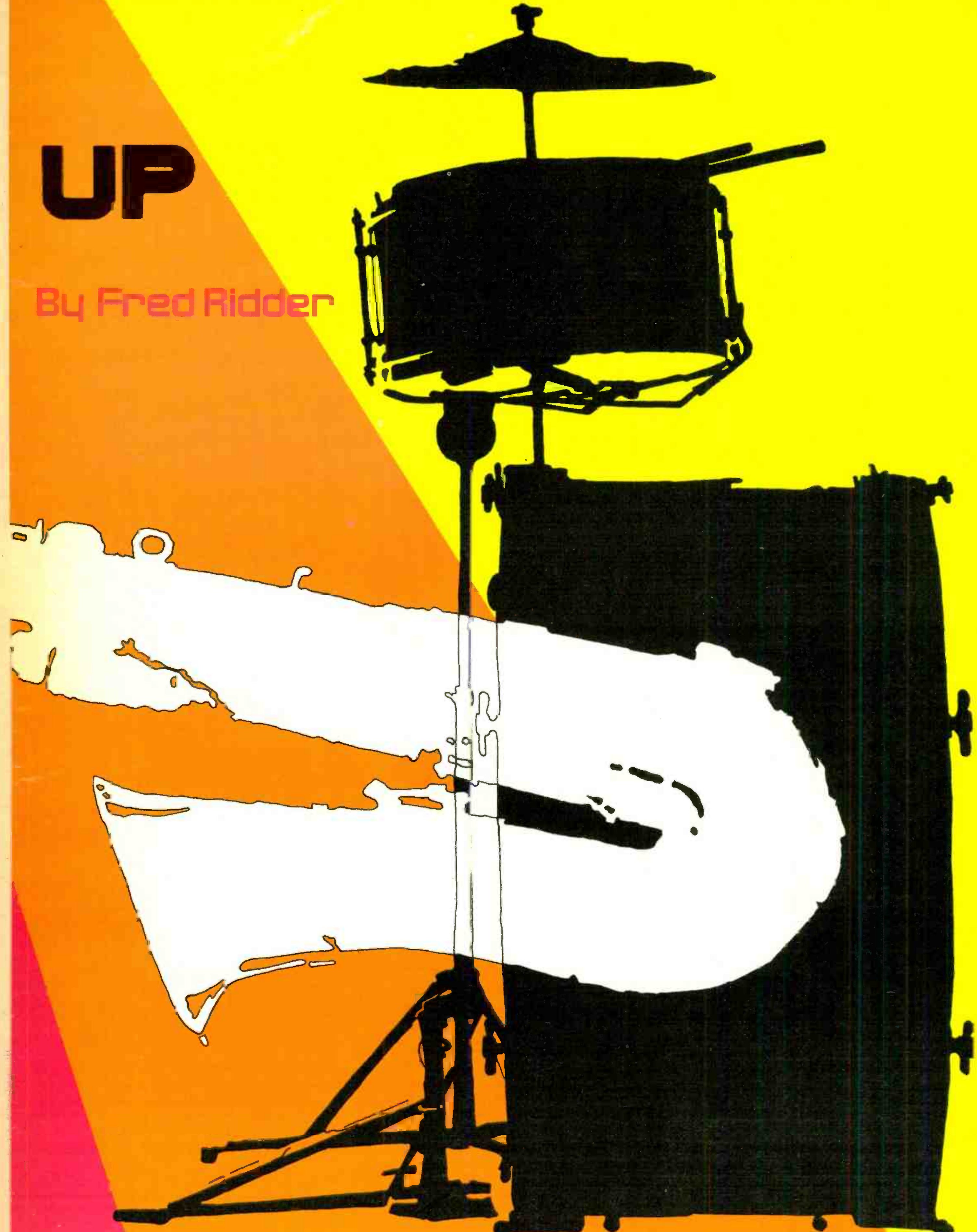
JAZZING PCM

Digital recording. These two words have generated a tremendous amount of interest in the world of audio engineering in the last few years. In this country, however, there was little general interest in digital recording until the autumn of 1977. All that changed at the 58th Convention of the Audio Engineering Society in New York that October where not one but two American-made digital recording systems made their debut; 3M showed a non-working prototype of their elaborate 40-channel digital recorder and demonstrated a 2-track digital machine, and digital pioneer Dr. Tom Stockham demonstrated his Soundstream digital recorder. Suddenly, digital recording was in the minds of thousands of recording engineers, studio owners and audio enthusiasts across the country. Yet in the ensuing year and a half, there has been little obvious progress toward making digital recording a reality in America. Dr. Stockham's Soundstream machine has been used

UP

By Fred Ridder



on a number of experimental classical recordings, a handful of which have been released commercially on the Telarc label, and as we go to press with this issue of *Modern Recording*, 3M has just delivered the first of their digital machines to two customers in Los Angeles.

In Japan, the status of digital recording is very much different than it is here in America. Most of the Japanese audio manufacturers have experimented with digital recording in some form or other, and many of them have quietly displayed their experimental machines or prototypes at recent Audio Engineering Society conventions. These devices never seemed to attract much attention from American engineers, partly because they [the devices] really were not much more than experiments and partly because America may just not have been ready for digital recording.

One of the foremost pioneers of digital recording in Japan has been Nippon Columbia Co., whose interest in and committment to digital recording has always seemed to stem from a desire to produce state-of-the-art recordings for release on their Denon label rather than from interest in manufacturing digital systems for sale. As early as 1972 Nippon Columbia had developed an operational but admittedly somewhat crude pulse code modulated digital tape recording system. (Pulse code modulation, or PCM for short, is the name given to the most classical process for digitizing an analog or audio signal.) Nippon Columbia did not keep their machine in the laboratory; their original system and a subsequent "portable" version developed in 1974 were put to work in the recording studio and on location in several concert halls and churches in "live" recording sessions. The output of these sessions was commercially released on the Denon label and amounts to some 150 LPs. These initial Denon PCM releases justly received critical acclaim for performance and sonic quality and were awarded several coveted prizes including the French Grand Prix du Disque and the Dutch Prix Edison. Yet Nippon Columbia was not satisfied with the results they were obtaining with their system—they knew PCM had more potential. In 1977, Nippon Columbia completed development of a new, updated 8-channel PCM tape recording system, designated the DN-034R which offered improved dynamic range and more sophisticated data handling for greater flexibility and convenience. It was this improved system which a team of Nippon Columbia engineers brought to Sound Ideas Studios in New York City in November of 1977 and again in November and December of 1978.

PCM Comes to New York

The reader might well wonder why a Japanese record company would travel half way around the world to use their new recording system. The answer to that question is fairly simple—they wanted to cut some sides of top-notch jazz and it is obviously better to come to the musicians than to try to bring large numbers of musicians to Japan.

But why go to that much trouble and expense to record some jazz LPs you ask? The answer to this question is a little more complicated and involves several factors. Foremost is the popularity of jazz in Japan. It has always seemed ironic that jazz, which is a characteristically American music despite its African roots, should be more popular in some foreign countries, most notably Japan and Germany, than here at home in America. It is not uncommon for a jazz record to sell as many copies in Japan or Germany as in America despite the much smaller total market for recorded music in these other countries. A second major factor is the musical taste of the audiophiles and audio purists who comprise the primary market for premium-priced digital or direct-to-disc recordings. Such dedicated listeners generally show a preference for classical and/or jazz records, which tend to be recordings of real-time performances, rather than rock and pop records which are most often "manufactured" in the studio. Both these factors relate to the size of the market for top quality jazz recordings which is large enough to make a PCM jazz recording project a realistic economic proposition. A third factor relates to conditions which make PCM jazz a technically feasible proposition. Very simply, jazz and classical recordings are much less likely to be constrained seriously by the technical limitations of Denon's PCM system which is only 8 channels and rather cumbersome to overdub with.

How did it happen that Nippon Columbia brought their latest and best PCM system to Sound Ideas, a fine, up-to-date but internationally unrenowned studio on West 46th Street in Manhattan? To organize their PCM jazz project Denon retained the services of Yoshio Ozawa as producer. Ozawa, an ardent jazz enthusiast, had produced a number of jazz recordings for other labels before his association with Denon, and one of the engineers he had worked with on some dozen LPs was a young but very experienced engineer named Jim McCurdy. McCurdy had subsequently joined Sound Ideas as a staff engineer, and it was his relationship with Ozawa that brought the Denon PCM project to Sound Ideas originally in November, 1977 and again a year later.

PCM Explained

Before going much further we should take some time to explore the basics of what PCM is, how it works, and how it differs from conventional analog tape recording. Our discussion is necessarily much simplified for the sake of easy understanding. For the technically minded who desire a more thorough, mathematical treatment of PCM techniques we would like to recommend Barry Blesser's excellent summary of the theory and current practice of digitalization techniques which appeared in the November, 1978 Journal of the AES ("Digitization of Audio: A Comprehensive Examination of Theory, Implementation, and Current Practice," *Journal of the Audio Engineering Society*, Volume 26, number 10, pages 739-771) as an excellent starting point.

The basic concept behind digital audio is fairly simple. There is a mathematical concept which says that any continuous function (that is, one which does not have any discontinuities) can be approximated to any desired degree of accuracy by sampling it one point at a time; as greater accuracy is desired, the points become closer and closer. If we take a sine wave function as an example, the coarsest usable approximation would result if we chose two points per cycle to correspond to the positive and negative peak values. (In fact, it can be shown that only these two points are necessary if we know that the function is a sine wave.) As we choose more and more sample points in addition to our initial two points per cycle, the shape of the curve becomes better defined whether it is a sine wave or not; eventually we will reach a point where the points are so dense that it may be virtually impossible to tell that our curve is actually made up of discrete points. The process of analyzing a continuous function in discrete time intervals is known as sampling.

If we take a very slow continuous function in electrical form, say a .1 Hz sinusoidal AC voltage, and measure its instantaneous voltage once every second we will have a series of voltage readings which can be used to generate an approximation of the original waveform; in this particular case we would have a ten point per cycle approximation. If we are dealing with an audio waveform, say a 1 kHz sine wave, we obviously cannot perform the sampling manually; we can, however, build a circuit which will sample the instantaneous voltage every so often and hold the sample until it takes a new one. Such a circuit is known as a "sample and hold" circuit for obvious reasons. To sample ten times per cycle of our 1 kHz signal we must take 10 samples every 1/1000th of a second; this is commonly expressed as a sampling frequency of 10 kHz.

If we use this same 10 kHz sampling frequency with a lower input frequency, say 100 Hz, we will obviously have more samples per cycle and will thus obtain a better approximation of the input waveform, but what happens if we take the input signal up in frequency? If we have an input of 5 kHz with our 10 kHz sampling frequency, we will have only two samples per cycle which is an adequate approximation only for a sine wave. Any input signal above 5 kHz (which would also include any distortion products of lower frequency inputs) will be sampled less than once per cycle. It can be shown mathematically that when a signal is sampled less than once per cycle, the output of the sample and hold will resemble the sampled output of some frequency other than the input signal. In other words, the sample and hold behaves as if it has generated a new, spurious signal; this phenomenon is known as "aliasing" and has two important consequences in the design of digital audio systems. First, it dictates that the sampling frequency must be at least two times the highest input frequency of interest, and second, it means that measures must be taken to prevent higher frequencies from reaching the sample and hold circuit. Consequently, a digital system with analog frequency response to 20 kHz must use a sampling frequency of *at least* 40 kHz, and must use an extremely sharp filter to attenuate everything above 20 kHz. In practical systems, the sampling frequency is usually above 45 kHz. (Denon uses 47.25 kHz, and preliminary proposals for a standardized sampling frequency have been in the 53 kHz area.)

Jazz composer and pianist Hilton Ruiz working out a part at Sound Ideas.

An astute reader might wonder at this point why considerably higher sampling frequencies aren't used to improve the sampling resolution of high frequency input signals. The most immediate consequence of a higher sampling rate is that more data storage capacity is needed to handle the larger number of samples, and this is bound to have a direct effect on system cost. The interesting thing is that the additional samples do not actually result in a significant improvement in audible quality. It can be proved that two samples per cycle is sufficient to reconstruct the input signal if we know that it is a sine wave, but what if the signal is not a pure sine wave? We know mathematically that any periodic wave form can be described a the sum of a fundamental sine wave and some particular combination of sinusoidal harmonic waveforms. If our system has a strict 20 kHz high frequency limit, it is obvious that harmonics cannot exist in our system for any fundamental 10 kHz or higher. Thus all waveforms above 10 kHz must be sine waves in a system with a 20 kHz band limit. Harmonic frequencies above 20 kHz do exist in naturally-occurring sounds, but it has been shown that bandwidth limiting a system to 20 kHz does not seriously degrade the sound quality. Raising the sampling frequency above 40 kHz will make it possible to sample frequencies above 20 kHz, but it will have no effect on the accuracy of the sampling of fundamental frequencies below 20 kHz since two samples are sufficient for a sine wave.

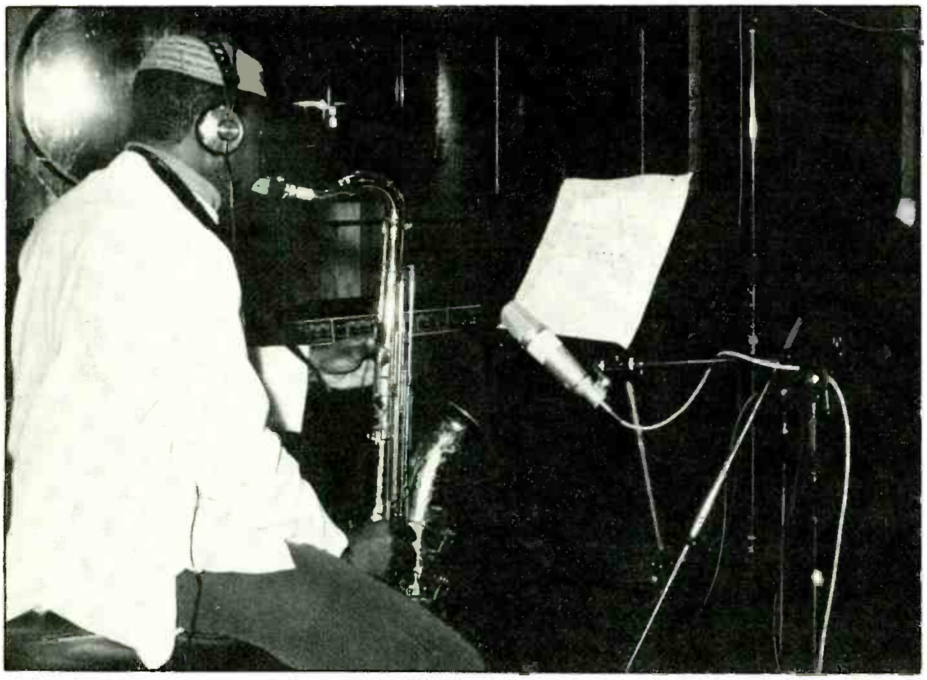
Now that we have sampled our input waveform, effectively changing a continuous function into thousands of discrete steps, the task is to store the value of each step so that we can later reconstruct the input waveform from the stored values, step by step. If we return to our example of a very slow sine wave and manual sampling, the obvious technique is to read the sampled instantaneous voltages on a voltmeter, converting the analog displacement of the meter movement into a digital value using the meter scale, and to write down the resulting number in a table. The process of reading a digital value from an analog meter is known as quantizing. We should stress that no matter how accurately we read the meter, we ultimately will always have to assign the same number to a small range of meter indications; in other words, we eventually must round off our reading, giving rise to a small error known as quantization error. The more accurately we can read the meter, the more decimal places we can read, the smaller will be the magnitude of the quantization error, but there will *always* be *some* amount of error in quantization. Quantization error is roughly comparable in effect to noise in an analog system; both quantization error and analog noise have the effect of masking small magnitude changes in the signal and thus determining the dynamic range of the system in question.

Quantizing a sampled audio signal is exactly analogous to the manual process we just described; to actually perform the operation we use an electronic device known as a quantizer or an analog-to-digital converter (A/D converter for short). Unlike our manually generated data which would most likely be recorded in decimal notation which we humans are most comfortable with, the output of the A/D converter would be in the form of a binary number, which is the only form that digital electronic devices are comfortable with. Each different binary number which can appear at the converter's output corresponds to a small range of possible sample voltages applied to the converter's input. The binary output of the converter will be in the form of a series of pulses or "bits" which correspond in time to the cycles of a carefully controlled external "clock," usually a crystal oscillator. At each clock pulse (what you might call a "tick") the output of the converter is either high or low in voltage to represent a binary 1 or 0. The particular sequence of highs and lows defines a binary number which in turn represents a unique small range of sample voltages at the converter's input. It is from this process that PCM derives its name; by sampling and quantizing the input signal, we have converted the amplitude modulation of the analog input signal into a sequence of binary numbers which are handled as a modulated pulse code.

For those not familiar with binary arithmetic, a brief introduction is in order. A binary number is expressed as a string of some quantity of binary digits (1s and 0s); the individual digits are known as "bits" and the length of the string of binary digits is known as the word length and is expressed as a number of bits. Digital electronic devices operate on a fixed word length; if it only takes five bits to express a particular number, those five bits will be preceded by enough 0s to make up the necessary word length. The last bit on the right of a written binary number corresponds to the last bit transmitted in a binary word; it is called the least significant bit (LSB) and has a unit value of 1. Each preceding bit in a binary number has a value two times that of the one which follows; the bit just before the LSB has value 2, the one before that has value 4, and so on up to the most significant bit which will have a value of 2^{n-1} in an n-bit number. The value of a binary number is the sum of the values of its bits, 0s carrying 0 value and 1s having a value corresponding to its position relative to the LSB as described above; for example, the digital number 101101 would equal:

$2^5+0+2^3+2^2+0+1=32+8+4+1=45.$

A 14-bit word, such as the Denon system uses, has a maximum value of 16,383 ($2^{14}-1$) if all its bits are 1s; this means that signal voltages over a 16,383 to 1 range can be uniquely expressed. In an audio system we deal with both positive and negative voltages, so one bit of our binary word must be used directly or indirectly to indicate the sign or polarity of the voltage. This means that our 14-bit system actually expresses positive or negative voltages over a range of slightly more than 8000 to 1, which is a voltage ratio of about 78 dB. This figure represents the dynamic range between the largest and smallest voltages which can be accurately quantized. The dynamic range of a PCM system is usually taken to be the ratio of maximum voltage to the quantization error, which is theoretically 3 dB below the level of the smallest accurately quantizable signal. This would yield a dynamic range of 81 dB for a 14-bit system which is what Denon specifies for their basic system. It can be seen that the most direct route to improved dynamic range in a PCM system is to use more bits in the quantizer to improve the resolution of small voltage changes and decrease the quantization error. According to theory, each additional bit doubles the resolution, halves the quantization error and thus improves the dynamic range by 6 dB. The 3M and Soundstream digital recorders use 16-bit words rather than the 14-bit words favored by the Japanese for exactly this reason. The disadvantage to using extra bits is that it signifcantly increases the cost of the system.



Veteran jazz saxophonist Pharoah Sanders recording a la PCM.

The only other process necessary to a digital recording system is the actual recording of the digital data. The task here is to magnetically record a stream of binary bits or pulses which occur at the frequency of the system clock which is usually in the 3 MHz to 10 MHz range. Fortunately this seemingly formidable task is commonplace in the world of instrumentation recorders, computer tape drives, and even video tape recorders. 3M's digital design team, for example, mostly came from backgrounds in instrumentation recorders and the machine shows this heritage. For economic reasons, most of the Japanese systems have opted to use video tape recorders. The Denon system multiplexes eight channels of digital information into one pulse stream compatible with a standard video format allowing them to use a standard 2-inch video tape recorder.

At the most recent New York AES convention Sony showed prototypes of PCM converters which will record a stereo PCM signal on either a U-Matic or a Betamax video cassette. Most of the systems using video formats have little more in their pulse stream than the binary words themselves and sync pulses to tell the machine where each word begins in playback, while systems designed from the ground up have no prior constraints on the data format and can thus embody sophisticated techniques of data interleaving and parity bits to allow the machine to continuously verify correct data and even to reconstruct invalid data in some cases. Denon's coding format is an example of the straightforward approach, while the coding scheme used by 3M and jointly developed by the BBC has very sophisticated self-verification built-in. The question of coding schemes promises to be a major stumbling block to compatibility between systems assuming that standards will eventually be set for sampling frequency, clock frequency and word length.

Much has been written about the advantages of digital recording over conventional analog recording systems. For those readers who have not followed the progress of digital audio, we will recap the qualities that make PCM potentially so superior.

In a sense, PCM's advantage stems more from the shortcomings of analog recording than from any advantage inherent to PCM. The basic problem in analog recording is that we rely on magnetic tape—which is inherently non-linear—to perform in a linear fashion over an extraordinary range of conditions. For audio purposes we require linear, undistorted operation over a 1000 to 1 frequency range and a dynamic range of some 70 dB (about 6000 to 1). In practice, the signal-to-noise ratio of the entire analog recording process is set by the tape itself; it is determined by tape saturation, or maximum undistorted output level on the one hand and the inherent noise of the tape on the other. To make things more difficult, tape saturation occurs sooner at high frequencies, and this effect becomes more pronounced as we increase the amount of AC bias we use to overcome the essential nonlinearity of the tape and thus reduce distortion to usable levels. The physics of magnetic recording dictate that the process depends on changing magnetic fields; a reproduce head will only produce an output if the magnetic flux of the tape passing the head is changing. As the recorded frequency goes down or as tape speed goes up, the rate of change of flux goes down and the head loses its ability to produce an output. The ultimate result is a limit to low frequency response.

Two additional shortcomings of analog recording which become problems all too often are wow-and-flutter and print-through. Careful mechanical design and improved tape formulations have minimized these problems, but there is no way to totally eliminate either problem in an analog system.

PCM recording overcomes the limitations of analog recording simply because it no longer relies on the magnetic recording process to be linear; magnetic tape is now merely a storage medium for data in the form of coded pulses. Since the pulse code only contains two pulse values corresponding to binary 1s and 0s, the dynamic range of the magnetic tape need only be good enough to produce reliable high and low output pulses. Tape noise and print-through effectively cease to have an effect on system performance, as does harmonic distortion. These are the factors that determine the degree of signal degradation in making an analog tape copy, and since none of them affect the performance of a PCM system there are essentially no degradation or generation losses in re-recording a PCM signal as long as the pulse timing is not affected. Wow and flutter of the PCM recorder does, of course, affect the pulse timing, but the sync pulses which are inserted between certain of the binary words allow the system to synchronize the data pulse and clock frequencies thousands of times a second.

In a PCM system, then, the tape is simply not the limiting factor. Virtually all of the normal performance criteria are determined by the architecture and performance of the conversion stages. Dynamic range or signal-to-noise ratio is set primarily by the word length. The high frequency response of the system is directly dependent on the sampling frequency, and the low frequency response truly extends to DC, since a PCM encoder treats the input signal as a sequence of discrete steady-state values rather than as a rate-of-change phenomenon.

The one tape phenomenon which does affect PCM recording (and analog recording as well) is the tape dropout. In analog recording we may hear dropouts as reductions in the output level or as total "holes" in the sound, or we may only hear them as a "graininess" in the sound if they are very short. In PCM, however, a dropout causes the machine to read a spurious data word because some of the bits are now incorrect, and we hear the result as a highly audible glitch. Most PCM systems include a "de-glitch" circuit following the digital-to-analog converter to minimize the audible effect of dropouts, and the more sophisticated systems use a variety of parity bit and data interleaving schemes to sense dropouts or incorrect data and either skip over the incorrect words or correct them using the interleaved data.

Lest we should think it perfect, there are a few shortcomings specific to digital recording we should not overlook. As we pointed out above, there is a quantization error which is inherent to the PCM process. Audibly, this error is heard as a kind of noise, but one which does not resemble the familiar hiss of analog devices. In most PCM systems the noise from the quantization error is so far below the signal that it is not a problem, but the sound of the noise is such that it may be objectionable out of proportion to its actual level. An operational shortcoming of PCM is the difficulty of performing edits in a data stream. Systems that use a simple, sequential format such as the Denon machine uses do allow physical editing, although they require that the edits be made precisely at sync pulses which may require extraordinary editing techniques. Most systems will require the use of electronic editing techniques which may prove cumbersome and time-consuming to engineers accustomed to fancy razor blade work. Fortunately, there is no loss of quality in electronic editing since there is no generation loss in PCM.

The Denon DN-034R System

In many ways the Denon PCM system is a simple and straightforward one. We must bear in mind that in designing this system, Nippon Columbia was looking to its own specific requirements rather than designing for some generalized set of requirements had it been intending to manufacture the system for sale. The system uses linear pulse code modulation with a 14-bit sign and magnitude binary word; this is the most classical and direct form of digital encoding of an analog signal. The 14-bit word length has sufficient resolution to yield an 81 dB dynamic range which is improved to 89 dB by using high frequency pre-emphasis/de-emphasis above 2 kHz. The sampling frequency is 47.25 kHz which is high enough to produce frequency response which is ±0.2 dB from DC to 19 kHz, falling to −1.0 dB at 20 kHz. Distortion is held to less than 0.1% at operating level. The transmission clock frequency of 7.1825 MHz and the data format are designed to be compatible with a standard video tape recorder. The Denon system encodes 8, 4 or 2 channels of audio information. The digital words for each sampling from each channel are assembled sequentially into a single data stream. Each stream of eight words is separated from the next by a data sync pulse, and each three samplings is separated by another sync pulse which corresponds to a video horizontal sync pulse for VTR compatibility.

The video tape recorder used with the Denon system is a standard 2-inch, four-head, low-band unit operating at a tape speed of 38.1 cm (15 inches) per second. In practice, Denon uses two VTRs running simultaneously and fed from the same PCM converter. This gives them a second master tape to use in case they should encounter a severe dropout problem or if the first master should in some way be damaged. Besides the two VTRs, their system comprises a PCM converter unit which contains all the encode and decode functions, a video and waveform monitor unit which is basically an oscilloscope display interface and an audio monitor which provides a headphone output of the audio program.

The Denon system does allow for overdubbing, although the process is somewhat cumbersome. To do an overdub, the existing data tracks are played from one VTR while the new track (or tracks) is encoded and added to the data stream which is then recorded on the second VTR. The process is very similar to the overdub technique that was used back in the four-track and eight-track days before sel-sync, except that there is no generation loss associated with "bouncing" the data from one machine to the other. The data transfer between machines is done at the data word level for best results; transferring data in audio form would increase quantization error noise because of the extra D/A and A/D conversions, and transferring transmission signals increases the noise and jitter of the signal.

Denon's PCM jazz projects have been nothing if not ambitious. In 1977 the project ran for five weeks in which time they completed eleven albums; this year's project was six weeks long and encompassed thirteen albums and a demo or two. These figures are almost astonishing to engineers used to working on rock projects which can and do drag on for months. McCurdy points out that it is really quite hard to take more than two days to record a jazz album when it is essentially being recorded "live." Ozawa and McCurdy wanted to discourage overdubbing on all the records, which they accomplished by telling the musicians up front that there would be *no* overdubbing, and they only relented in two or three cases. This made it possible to record an album's worth of music with alternate takes in one day or a day and a half and then mix it the following day. Occasionally things would go well enough the first day that it was possible to record and mix the whole album the same day and still get home in time to catch the 11:00 news. One truly remarkable session was the Bridgewater Brothers Band.

Cecil Bridgewater is a trumpet player who has been working with Ozawa as A&R Coordinator on the project. From his involvement in several phases of the project he became impressed with the immediacy of recording with PCM; to him it is the next best thing to direct-to-disc, and he decided that when he and his saxophone playing brother, Ron, did their album they would approach it in much the same way they would if they were recording direct-to-disc. On the appointed day they brought their band into the studio, did one run-through and then recorded the entire album, some 38 minutes of music straight through in two takes.

The artists recorded for the Denon projects span nearly the whole range of the music called jazz. Their scale ranged from a Kenny Byron-Tommy Flanigan piano duo (the two had never met before they got together in the studio that day), to the 25-piece Frank Foster Band, to an all-star quartet recording with Archie Shepp as front man. Some of the groups were existing units with extra players added to perform the parts that would otherwise be overdubbed, and some of them were combinations put together by Ozawa. For the Archie Shepp session, as an example, Ozawa booked Shepp on sax, Jaki Byard on piano, Roy Haynes on drums and Cecil MacBee on bass, an all-star combo who truly enjoyed the opportunity to play together.

Ozawa generally left the choice of material to be recorded up to the musicians. Existing groups normally would have a number of tunes rehearsed and ready to record when they came in to the studio. In many cases, the choice of tunes would have been influenced in advance by A&R coordinators Cecil Bridgewater and Reginald Workman who maintained contact with the bands prior to the sessions. In the case of sessions with combos put together for the date, the tunes were usually picked spontaneously in the finest jazz tradition. It is just this situation that is one of the greatest challenges a jazz producer will face, since it raises the question of how much structure and direction a producer should impose on what is ideally a spontaneous music. Ozawa's approach was to leave the players to themselves as long as the results were worthwhile, and to step in only when things failed to gel. According to McCurdy, on more than a few occasions Ozawa would go into the studio and say to the session leader something like: "Look, you recorded this tune back in 1964 with so-and-so on the such-and-such label, and it was super; if you don't think you can do better now, let's do another tune."

The wide range of music recorded was a considerable challenge for engineer McCurdy. "The first year we had this project here I learned that the secret to handling the situation effectively was to find out as much as I could about the session ahead of time. This year it seemed that I spent half the time I wasn't in-session on the phone trying to find out what the guys coming in next week had for instru-

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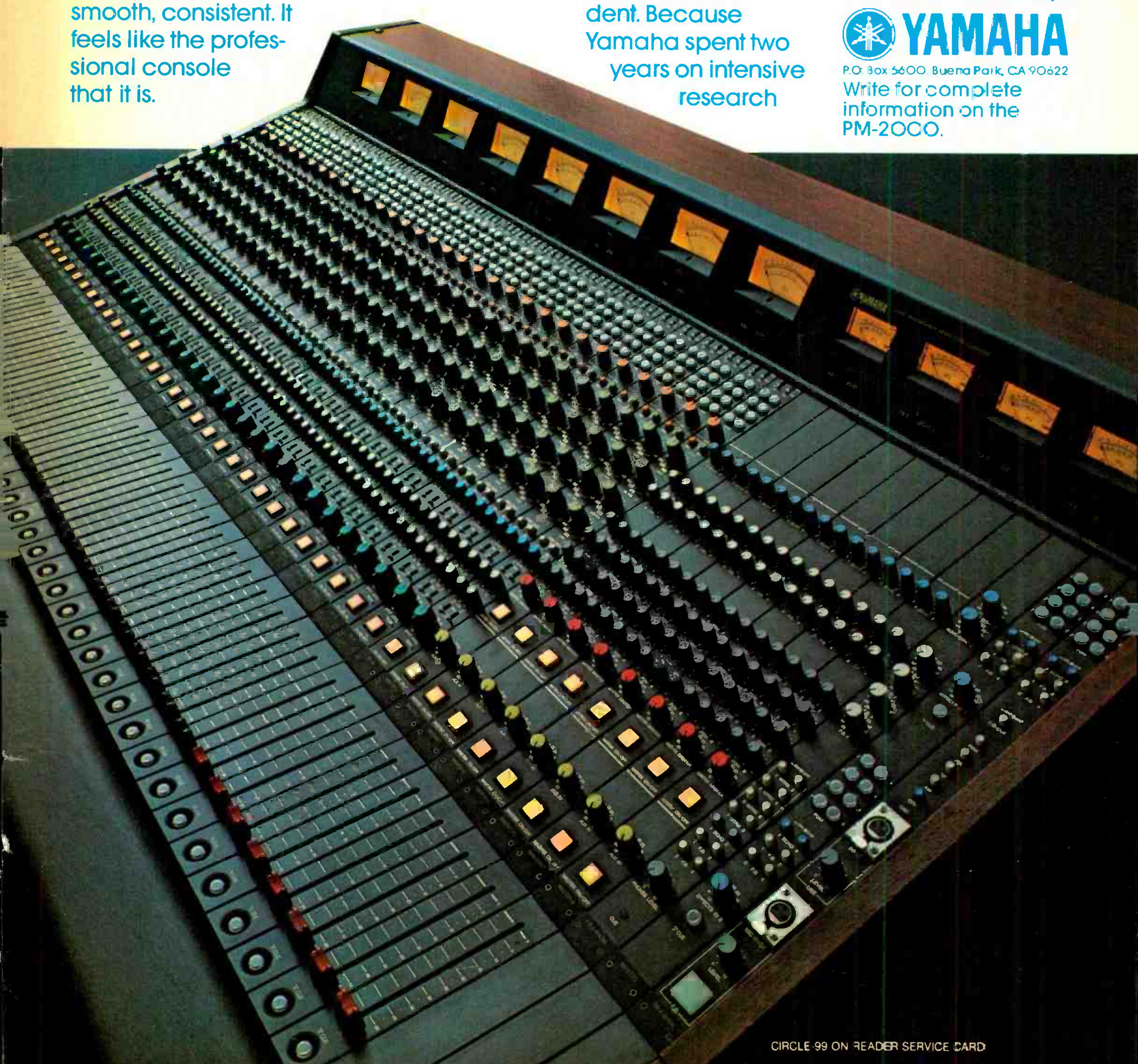

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ments. That way, I could have some idea of the set-up and have part of it ready when the musicians started to arrive. It really kept our assistant engineers hopping too, because it seemed like we were doing something different every day. One day we'd have the drums out in the middle of the room, the bass in the booth and the piano over here, and the next day we'd get a bass player who has to practically sit in the drummer's lap and a keyboard guy who has to see everybody in the band."

This year, McCurdy's task was made more complicated because the Denon project was booked into Sound Ideas' largest studio, Studio C, which is located down a flight of stairs from the other studios and the offices. "I was really pleased we got Studio C this year because it's such a great room. Last year we worked upstairs in Studio A, which is a good room, but it's much smaller and you just don't have as many choices about where you can put things. Studio C is fantastic because it's so flexible, but it meant that I really had to have a good idea of what was going to happen or else we would have wasted a lot of time."

The Studio Set-up

Studio C at Sound Ideas is a fairly large room, some 24 feet wide and about 35 feet deep with a high, 18 foot ceiling. In one corner of the room is a totally enclosed vocal booth with a semi-enclosed booth upstairs in a piggy-back arrangement. Along one side of the room is a topless drum enclosure and upstairs above one side of the control room area is a low-ceilinged isolation room measuring about 12′×15′. The acoustics are fairly uniform throughout the room, the overall sound being balanced but fairly dead.

Within this space, McCurdy would set up a given instrument where it made the most sense for the particular session rather than in a particular place all the time. It is easier for the engineer to work with everything in a customary place, but McCurdy feels this is not the route to the best sound. The one exception to this is the string bass which McCurdy prefers to put in the vocal booth whenever he has the choice. "String basses are such variable instruments that I like to put them in the booth where I can do whatever is necessary in miking to get a good sound without having to worry about leakage. Of course, some bassists just can't deal with playing in the booth; in those cases I just have to make do with them out in the room because it's the most important for the musicians to be comfortable with the set-up."

The Archie Shepp session was a good example of a small group setup: Shepp and Byard were in the main part of the room, Roy Haynes' drum ket was set up in the drum enclosure which was left partially open, while MacBee and his bass were isolated in the vocal booth. If the rhythm section had been larger or louder, McCurdy would likely have put Shepp in the upstairs booth above MacBee, but under the circumstances he opted for the more intimate set-up described. A large session like the Frank Foster date with its nineteen horns and six-piece rhythm section limits the options available; the horns obviously will dominate the set-up and occupy the bulk of the space available while the keyboards and amplifiers are pushed toward the walls. The most complicated set-up was for a John Stubblefield session which used both horns and a string section. "For that session

"... the music erupts from a silent background. It is an impressive moment..."

we left the horns in the main part of the studio and put the twelve strings upstairs in the little isolation room. Twelve strings in a 12′×15′ room was a little crowded, but they had a perfect view into the studio and it sounded surprisingly good. I was very pleased," says McCurdy of the date.

While McCurdy was very flexible on the physical locations of the instruments, he stayed with a fairly standardized mic selection. He finds it preferable to use a few types of high quality mics whose characteristics he is very familiar with and to spend some time finding ideal placements for them rather than using a whole variety of different mic types. His goal, particularly for the Denon project which has such high quality standards, is always to obtain a "real" sound for an instrument. In describing a four microphone set-up he experimented with on a tenor saxophone, he stresses that he was not trying for a gimmicky sound—he simply wanted to find a technique which would capture the whole sound of an instrument which produces sound from its entire body, a musical fact which many engineers are ignorant of or choose to ignore.

To mic a drum kit, McCurdy generally chooses an Electro-Voice 666 for the bass drum (this is an old favorite of jazz and R&B engineers), AKG 452 condensers with the accessory 20 dB pad for tom-toms and snare and Sennheiser MD421s for overhead mics. Asked about his choice of dynamics for overheads when most engineers would pick a condenser mic, he replied that condensers usually sounded too "sizzly" as overheads for a jazz drummer, who is likely to play a lot of ride cymbal. A piano will usually get three or sometimes four Neumann KM86s before being draped with moving blankets over the lid to reduce leakage if the other band pieces are loud. Beyer 160s are the customary choice for miking amplifiers, plus McCurdy tended to use direct boxes more frequently on the Denon project to eliminate noise from the amplifier which becomes very noticeable due to the much wider dynamic range of the PCM system. For saxes and horns McCurdy most often falls back on Neumann U-47s and U-87s. He explains that even though he really doesn't like the 87 that much he will use it on horn sections and string sections particularly for the simple reason that Sound Ideas owns some two dozen of them for use on the advertising agency dates which comprise the bulk of the studio's business. The string bass is the one instrument which will see a variety of microphones; for a particular instrument that is a touch dull the choice might be a KM84, while a thin-sounding bass may require an MD421 to fatten up the bottom a little, and a boomy instrument might get a Sony ECM50 and some windscreen foam attached to the bridge.

Recording with PCM

In the control room, things look much the same as for any recording session. The only clue that this is a PCM date is the 8-channel plug box sitting near the console's patchbay with its snake cables running out the control room door to a store room where PCM engineers Takeaki Anazawa and Kaoru Yamamoto hover over their machines. When the musicians are ready, McCurdy signals the PCM engineers and tells his assistant engineer to start the 2-inch analog multitrack machine which also runs as a safety. There is nothing at all

unusual about the session until it comes time for a PCM playback—McCurdy signals for a playback, and the play light on the special PCM status indicator panel to his right comes on but we don't hear anything at all until the music suddenly erupts from a silent background. It is an impressive moment.

McCurdy enjoys telling the story of his first PCM playback. It seems that he turned to one of the PCM engineers, both of whom spoke very little English, and said, "Playback." The engineer nodded and said something in Japanese to his associate in the other room. And nothing happened—no hiss, no anything. So McCurdy repeats, "Playback." The Japanese nods and says, "Playback." Figuring something is wrong, McCurdy turns to the console and reaches to turn up the monitor volume just in time for the start of the music. "It was *very* loud," is his recollection.

One's second impression of the sound of a PCM playback after recovering from the lack of the warning hiss we all know from analog recording is the solidity of the bass end of the spectrum. The PCM system's response extends to DC, and even though the monitors won't come close to that, there is tangibly more bass and more real, solid bass than one would think could come off a reel of tape. McCurdy has found that this improvement in low frequency sound quality has changed his approach to recording somewhat. The most concrete example of this is the importance the string bass has taken on. McCurdy finds that the PCM system is the only one he has ever worked with which can capture the full resonance of the instrument, and he is likely to spend some time getting the sound just right to take advantage of this potential.

The most difficult aspect of working with the Denon PCM system is how to allocate the eight tracks most effectively. Unlike a 16-track or 24-track date where he can record instruments and even individual mics in isolation, McCurdy is now forced to pre-mix several signals on each track just to get it on tape. "It's really not bad if you have a clear enough idea of what you're trying to come out with as a final product. You just have to approach recording the tracks with some of the same considerations you would have doing a normal mixdown. If you do it right, the two-track mixdown almost happens by itself."

In general, McCurdy would assign two tracks for a stereo drum kit (versus his normal six to eight tracks), one track for bass, two for stereo piano and the remainder for solo instruments. If tracks were tight, he would sacrifice the second piano track; beyond that he would have to combine tracks. There is nothing wrong with combining tracks, to be sure—remember that the Beatles' *Sgt. Pepper's* was recorded four-track, after all—but you lose flexibility every time you do so; you must have a good idea of where you are going with a tune because once you combine the tracks, you've gone.

For a session like the Frank Foster band, track allocation is a real problem. In this particular situation McCurdy retained separate tracks for bass and stereo drums but combined the piano and guitar into a single track. He then used one track each for the trumpets, saxes, and trombones, leaving one track left over for solos. This solo track was fed from a special console bus he had set aside that he could assign each soloing instrument to in their turn. This gave him the greatest flexibility to control the final mix that he could obtain within the eight-track format.

At the other end of the track assignment question was the Kenny Byron-Tommy Flanigan piano duo. McCurdy set up the two 9-foot Steinways facing one another "Ferrante & Teicher style," with four mics in each. "My first intention was to assign one mic per channel, but then I decided to make things easy and do it as a "live" two-track. I just spread the two instruments out with the lower strings of each instrument toward the middle and the higher strings toward the opposite sides. This gave plenty of separation for the melody lines while unifying the rhythm patterns each was playing. It sounded a lot better than if I had just panned them left and right, and probably made it easier for them to cut the disc, too."

McCurdy discovered early on that the PCM system is very demanding of good input signals. If you feed it a signal that is clean enough for an analog tape you'll hear the noise and distortion on PCM playback that would normally be hidden by tape hiss. "I never realized what a good mic the KM86 was till I heard it on a PCM playback, and I always thought Studio C's Aengus console looked like a turkey, but it *sounds* fine." For similar reasons, McCurdy used very little outboard signal processing devices, and little equalization, too, for that matter. "I tried to use a limiter a couple of times, but couldn't. The noise just stuck out like a sore thumb." For this reason, the mixes became rather simple affairs, mainly a matter of setting the balances and stereo perspective and adding a judicious amount of echo to the tracks which were all recorded dry.

There were only two or three of the thirteen sessions on which any overdubs were done. One of them, the 24th Street Band, posed an interesting problem. McCurdy knew going into the session that the group would need to do an Oberheim synthesizer overdub—fine, no problem. But once they got into the studio, they decided they needed to double their background vocals, and suddenly there weren't enough tracks. McCurdy originally figured that he would record the lead and background vocals on the same track and then do the double on a separate track, but thought that this would limit his flexibility in balancing the vocals and in placing the vocals in a stereo perspective. What he eventually did was to record the lead vocal on a separate track from the original backgrounds. Then, in performing the overdub from one machine to the other, he transferred all the tracks except the original backgrounds. That track he decoded, mixed with the doubled vocals as they were being done, and placed the full, doubled backgrounds back in place of the original backgrounds on the PCM copy, which then became the master.

McCurdy says that working with the Denon PCM system has been a real eye-opening experience, and that it has almost spoiled conventional recording for him. He relates that several times after the PCM project he called a maintenance man into the room because a track was "too noisy" which it was . . . by PCM standards. But ultimately McCurdy feels that "If the music doesn't come across, the medium doesn't matter." He sees his role as making it possible for the music to happen without a hassle for the musician, which might take away from his performance.

For anyone who makes the effort to seek out some of the Denon PCM pressings, the reward is some very fine jazz and some stupendous sound quality, courtesy of fine musicians, a good producer and engineer, and PCM digital recording.

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profile: The Brecker Brothers

By Joel Siegel

One could call the Brecker Brothers super-stars of the studios. Randy and Michael Brecker have appeared on dozens—possibly hundreds—of records in the past decade. Their discography reads like a Who's Who of popular recorded music. The list includes: James Taylor, Carly Simon, Stevie Wonder, Todd Rundgren, Aretha Franklin, Average White Band, Gladys Knight, Lou Reed, Bob James, Billy Cobham, Elton John, Freddie Hubbard, Parliament, Bruce Springsteen, Phoebe Snow, Johnny & Edgar Winter, Deodato, Aerosmith, Blue Oyster Cult and the Rolling Stones. Not bad.

Randy Brecker (age 33) is the elder, and probably the most well-known of the brothers. He was a founding member of Blood Sweat & Tears—the band most responsible for the commercialization of jazz-rock music. Randy has been playing trumpet since the age of eight.

Michael Brecker (age 30) plays saxophone, and has done so since he was thirteen. Michael was fortunate to have had an older brother in the business when he decided to pursue music as a career, in 1970. Both Randy and Michael have had extensive formal music training.

In 1975, Randy and Michael formed the Brecker Brothers Band. The band has recorded four albums on the Arista label, the most recent being Heavy Metal Be-Bop. *That record was recorded "live." The Brecker Brothers Band has been Randy and Michael's major interest, though session work provides the bulk of their income.*

Randy and Michael Brecker's schedule books are filled with studio dates that are the cream of the crop. In addition to record dates that span rock 'n' roll, jazz, funk, R&B and pop, they do hundreds of advertisement jingles, as well as television, radio and movies.

Recently, Randy and Michael became partners in a lower-Manhattan jazz club, Seventh Avenue South. Their club is an attractive and comfortable place to hear music. It is also the realization of a dream. The Brecker Brothers had always wanted a place, where their musician friends could meet and jam their hearts out. The musical menu consists primarily of swinging jazz, with hot and cool sounds raising the roof nightly. Randy and Michael appear there regularly with many of their famous friends.

With a recording band, countless sessions and a club to run, Randy and Michael are hard people to pin down. It proved to be impossible to interview them together. Both Randy and Michael agreed that to speak with them separately would be interesting. Frankly, this writer doubts that these two professionals are ever less than interesting to interview. Spell professional with a capital "P."

The first part of this Brecker Brothers interview was with Randy. I interviewed Randy at the office of the Brecker Brothers' publicist, on a Friday afternoon. He [Randy] was groggy from the past night's sessions. Nevertheless, our talk lasted for well over an hour. We covered the Blood Sweat & Tears days, Randy's beginnings in the business and many of the session dates he has played on. In addition, we discussed some of the technical aspects of his musical endeavors, the sound considerations for the club, Brecker Brothers records and the miking of his trumpet.

MODERN RECORDING: Let's start at the beginning. When did you start studying trumpet?

RANDY BRECKER: When I was about eight. I started studying classical. I was never a great sight-reader until I reached college, for some reason, just from doing it a lot. I can read pretty well now.

MR: Why did you study trumpet?

RB: I don't know. There's no logical explanation for it. I was too young at the time to make any sort of conscious decision. I was in third grade. It was either trumpet, trombone or clarinet, so I said trumpet.

I think this girl I liked was going to play trumpet too, so I figured I could sit next to her if I played trumpet. That's all I played ever since . . . a little drums, and piano.

MR: What made you leave Philadelphia (his home town)?

RB: Early after high school I went to college at the University of Indiana music school. They supposedly . . .they still *do* have the best music school, I think, in the country, as far as facilities. They had real good trumpet teachers. I studied with a guy named Bill Adam and a guy named Dave Baker, and Jerry Coker for improvisation and jazz.

They had a real good jazz big band and a good small group. Eventually we won a contest to go on a State Department tour of the Far East with the jazz big band. We toured all the Arab countries, and Ceylon, India, Pakistan, Cyprus, Afghanistan. Everywhere we went there were guns.

I never went back to school after that. I came to New York and transferred to New York University. After a semester there, I started getting gigs. I took a leave of absence from school and that's the last I ever saw of school.

MR: How did you become a working session musician?

RB: It was a long slow process, and it wasn't anything that I had really planned on. I guess it just had to do with meeting certain people that were doing sessions, people that called me up. It was mostly trumpet players that called me up to sub for them. People got to know me, like jingle people and record date people. I guess record people got to know me through the records we did on our own, like Dreams, and with Billy Cobham and Blood Sweat & Tears.

MR: This was all in New York?

RB: Yeah. Mostly it had to do with meeting other trumpet players that were doing sessions; they just helped get me into it. If they couldn't make a gig for like the first fifteen minutes, they would call me up and say, "Hey, can you cover me for fifteen minutes?"

So I'd play a little, and if the guy who wrote the chart heard me and later needed somebody, sometimes they'd call me. So that's the way I worked my way into the inner sanctums of the studios. It really took a while. It actually took a few years.

MR: What was your first session?

RB: My first session was with a band called Les & Larry Elgard, which I used to go out on the road with on weekends, when I first came to New York. I used to go out on the road and drive for ten hours to college proms in the Midwest. We'd do the gig and drive back the same night, because they wouldn't pay for hotels. I got to meet a lot of different musicians. There were always different musicians every week. That really helped to get my name around. That was my first record date.

MR: Who were some players that first called you up to sub for them?

RB: I don't want to leave anybody out. Marvin Stan helped a lot. Burt Collins, Joe Shepley . . . Clark Terry helped a lot. He had a big band. He gave me a job in the big band. Snooky Young helped a lot. Thad Jones helped. I played with him for a while. Alan Rubin helped.

MR: What events surrounded the Blood Sweat & Tears sessions?

RB: A friend of mine named Jerry Weiss called me. They were looking for a trumpet player. Like I say, you never know. I met Jerry on the road with Les & Larry Elgard. We did a Jackie Gleason television show together, years ago. They had a big band. It was mostly a pick-up band. They would fly people down from New York. Jerry was playing with Sammy Kaye and his band of something or others. It was one step away from Guy Lombardo. Jerry was playing with Sammy Kaye's band and so we met. He said, "I really like your sound, and blah, blah, blah."

He was going to Juilliard at the time. He got the job with B,S&T, and they needed two trumpet players, and the other trumpet player wanted to teach. So, he called me up. He said, "Hey, you want a gig with this band. They're part of the Blues Project, and it's going to be real big."

At the time, I was still just working on the road with Les & Larry Elgard, and playing for go-go girls at the Metropole with, like a soul band. I said, "Yeah man, sounds good." I would get a salary every week and get into the rock 'n' roll world. So that's how I started.

Like I say, you never know where you're going to get a gig from. From the outest gig you might get the most amazing gig. Who would have known that I would have got the B,S&T gig from playing on the Jackie Gleason show. And with Les & Larry Elgard, and a trumpet player from Sammy Kaye's band.

I left the Metropole where we had to do something like ten sets a night, and twelve on Sundays. "Soul Man," and "Hold On, I'm Coming" ten million times. Guys drooling over chicks. I'll never forget, there was this guy who would come in every day and stare at this one chick's feet. Foot-fetish maniac, no doubt.

I left B,S&T in 1968 to play with Horace Silver. I left the same night that Al Kooper left the group. I joined Horace Silver along with Billy Cobham, and Benny Maupin, who's now with Herbie Hancock. The bass player now plays on the *Tonight Show*, a guy named John Williamson.

MR: What were some of your most memorable sessions?

RB: That's a real hard one to do [answer], because I've done about ten million and most of them go in one ear and out the other. Let me think . . . probably some of the earlier ones. The James Taylor ones were kind of memorable, because we got really involved in helping him arrange the horn parts. Even though he knew what he wanted, he didn't know how to write it out for us. He wrote it out on graph paper. It must have taken him ten years, and we didn't know how to decipher it. Each little square was supposed to be a half-step. He made these amazing graphs, and we were supposed to somehow miraculously figure them out, which we didn't. He just sang us the parts. I couldn't believe that he went to all the trouble to make these graphs instead of just writing them out in regular notation. So that was pretty memorable. We did a couple with him. That was me, Mike and Barry Rogers. That was the old Dreams' horn section.

Paul Simon is always interesting to work with, because he's such a perfectionist. You can literally spend like six hours on two bars.

MR: How about Blue Oyster Cult?

RB: That was a last minute deal. The producer, who shall remain unnamed,

In a familiar pose: Trumpeter Randy Brecker (left) with brother Michael on sax.

called us the last minute to do something. They didn't even use anything that I did. I busted my ass to get paid for that date.

MR: Ron Carter?

RB: That was a good one. That was one of the few "live" dates where everyone was there. Usually it's overdubs, you know, sweetening sessions. Everything's there, you put on the earphones and play with the tape, just the horn section. That was a "live" record at Rudy Van Gelder's [studio], so that was like real music—everybody was there, and it was mostly improvised.

MR: Do you prefer "live" to tracked [overdubbed] records?

RB: Yeah, I do, but I understand why tracking is done. We do a lot of tracking on our records, but try not to overdub all of it. We try to do all the solos "live," so you get some interaction between the musicians. There are sound reasons for doing tracking, especially when you're playing at loud volumes—to isolate the sound. On a jazz record it's different. You can baffle the sound and still get the separation.

MR: Who are your favorite composers, people whose work you enjoy improvising to?

RB: There are ten million of them, 'cause I go back to listening to the Bird [Charlie Parker]. From Bird on, old be-bop and standards. Bird tunes, Miles tunes, Freddie Hubbard tunes, Coltrane tunes, McCoy Tyner. Currently, I like Weather Report, Herbie Hancock and Chick [Corea]. I like a lot of Brazilian music; Egberto Gismanti.

MR: Did you foresee your new club becoming a mecca for jazz musicians?

RB: I don't know about a mecca . . . I just hope it's a comfortable place to play, not like playing a bunch of dives. We still have to work on it a little.

MR: Who designed the sound system for the club?

RB: Two friends, Billy Shaw and John Nichols. If you want to ask me any technical questions about it, I know nothing. They custom-made it. I don't even know how you work it.

MR: Did you have to make any structural changes to the club?

RB: We're doing that now. I have a friend named Tony Bongiovi, who designed Media Sound [N.Y.C.], and who has his own studio called the Power Station. He's telling us what to do about the sound; we've had problems with the neighbors.

MR: Do you have any personally favorite studios?

RB: There are a few of them. My favorite as far as playing right now is the Power Station, I guess. There are a bunch of good studios. The Power Station is good both for rhythm sections and horn players. Horn players usually don't like a place that is real dead, and they designed it especially with wooden floors and pyramid-shaped ceilings, so that you can play and not have the sound just drop two feet in front of you. There are a lot of other good studios. Atlantic is good. A&R, Hit Factory, Media is good. I'm probably leaving some out.

MR: Are there any microphones that [you find] are particularly good for your instrument?

RB: It seems like old RCA microphones are good, or Neumann 87s or 47s. Those are good, especially for my brother. He likes the old ones, 47s with the tubes. I use the new Neumanns with the transistors.

MR: Do you get into the technical aspects of recording . . . engineering?

RB: I am trying, slowly but surely. I don't think that I am naturally into it, which is why I get *Modern Recording*.

MR: Do you produce?

RB: Yeah, I produced a couple of records. I learned something about it. I am more like a musical producer. As I say, I know very little about the engineering aspects. I can give people good ideas about the music, but I have to depend on the engineer to get the sounds that I want. I don't know the finite technical things, like what microphone sounds good on what, or where to place them. I just have to describe a sound to an engineer, and hopefully he can get it the way I described it.

MR: Are there engineers that you prefer to work with?

RB: Yeah, there are a few of them. The last one we used was real good, Bob Clearmountain. Engineers are like musicians, they get typecast.

MR: Are there any producers that you prefer to work with?

RB: That's a hard one. Not really that many. I would prefer to produce myself. I pretty much know what I want to do. I learned the hard way, that I, in the future, just want to produce myself because it eliminates a lot of problems. I know the way I want it to sound. Even though I am not objective about it, I know the way I want it to sound. I prefer not having someone else telling me how they think it should sound, even if they're right.

MR: Do you do any recording in con-

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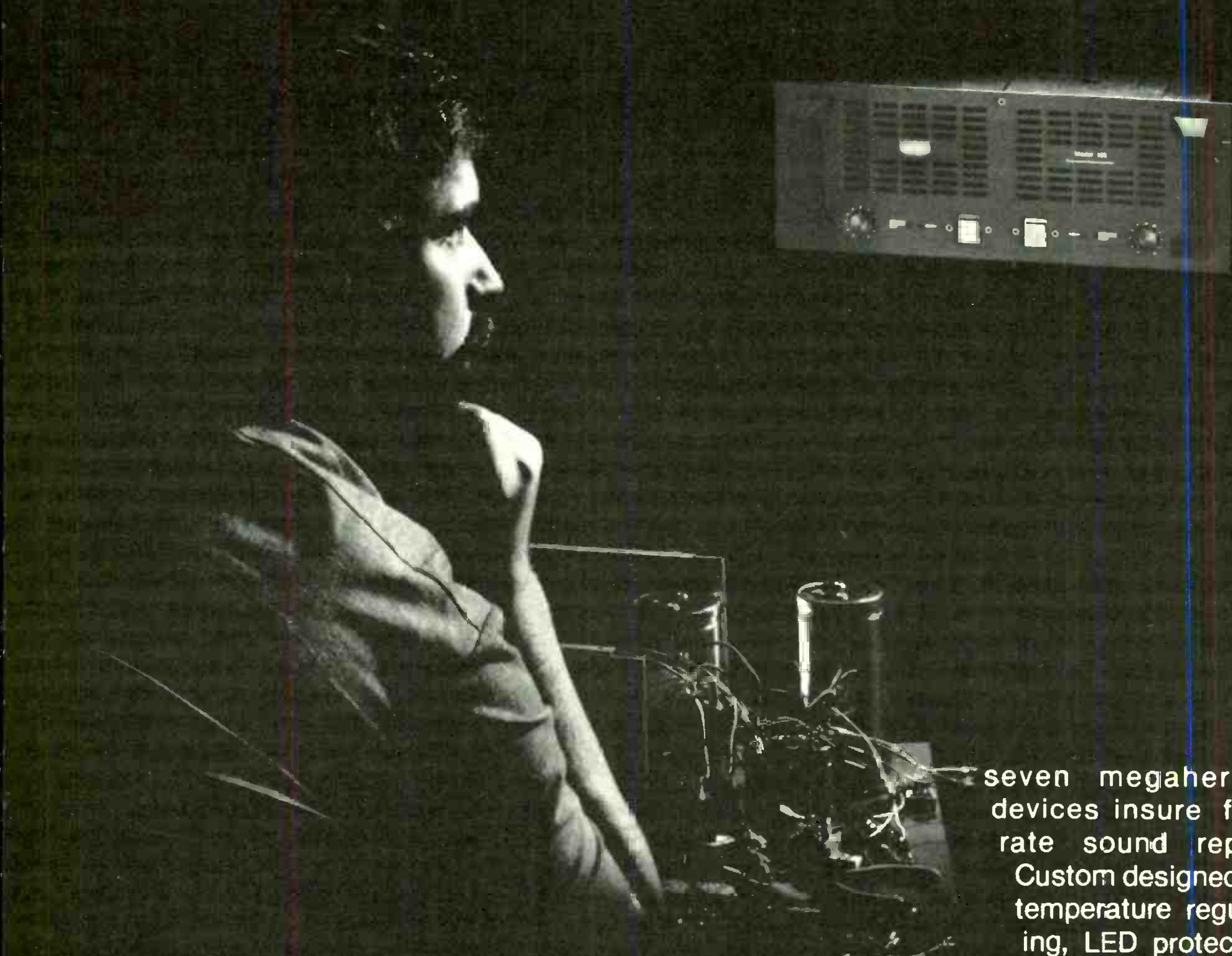

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junction with the club?

RB: Not yet. We might do some recording at the club. There's been some talk of that. It's a good place to record because the acoustics are good.

MR: What do you think is important [for a musician] to bring to a session in terms of equipment and in terms of mental framework?

RB: All I bring is my trumpet, flugelhorn, mouthpieces and my trumpet oil . . . my W-4 and W-2 form. As far as mental attitude, you just have to go in realizing that you're being hired, and [therefore should] do the best job that you can for whomever you're working; do whatever they want. Nine times out of ten it's not going to be any kind of creative bombshell; it's just a gig. It's fun in a certain way, if you're playing with good musicians, and there's a big horn section. It's fun blending with whomever you're playing with, and just trying to play in tune, and keeping your mind awake so you don't make stupid mistakes. Usually you just read charts. Whoever the arranger is, hopefully he is a good arranger so you can have fun reading the arrangements. It's good to sight-read. It's good to practice. Instead of sitting home and practicing, you get to earn while you learn.

MR: What's a difficult session that you remember?

RB: The most difficult ones, in terms of being on the spot, are the "live" jazz records. You don't have the chance to go back and repair them. Any jazz record like a Horace Silver, or Jack Wilkin, or Ron Carter record. There are a number of them where you just go in and play "live" solos. One take, two takes, *blam!* There's no overdubbing. If you goofed, too bad. So psychologically those are the probably the hardest sessions. I can't really single out any particular one.

MR: What are some of your favorite places to play?

RB: Without sounding prejudiced, probably my favorite is our club. It just feels really good to play in there. Generally I like playing places that aren't really big. I like playing at My Father's Place [Roslyn, Long Island]; that usually feels good. I used to like playing in Boston at Paul's Mall, but that closed. I used to like playing at Ratso's in Chicago. We like to do concerts if it's not too huge a hall, especially if we have the opportunity to do a good sound check.

MR: Have the Brecker Brothers toured to any degree?

RB: We've toured, but not extensively. We'll be going to Europe this summer, and Boston and surrounding areas in March. It boils down to finances; it costs a lot to tour.

MR: Did you record at any time with Stevie Wonder?

RB: I played with him on the road for about six months. I don't think I ever recorded with him.

MR: What was that [tour] like?

RB: It was amazing, just watching him work, and watching him perform every night. He's just amazingly consistent every night. It didn't matter what happened. One night before a gig at the Apollo, there was a fight in the dressing room between the guitar player and one of the roadies. We were all squashed in there—I was just trying to keep my trumpet from being smashed—and Stevie was in the middle of the thing. It happened right before we were supposed to play. They were really going at it—two big guys wasting themselves. Stevie went on stage and performed as if nothing had ever happened. It was like perfection every night.

> **"...Usually it's not going to be a creative bombshell; it's just a gig..."**

MR: What musical period [for Stevie Wonder] was that?

RB: It was around the Superstition time (*Talking Book* album). People say he's a genius and all that, but he works on his music perhaps twenty hours a day. Whenever he's not sleeping, he has earphones on listening to his music. In between sets, or when we're traveling on the road, he's got earphones on figuring out what to do on his records. He works at it.

MR: How about John Lennon?

RB: Yeah, he called me up, or rather his manager called up my old lady and said that he needed me for a John Lennon session. She thought somebody was kidding, and she hung up on him. Well, that was the end of that one. They got someone else. I forget which record that was, but Mike ended up on it.

MR: Charlie Mingus?

RB: I was lucky enough to do his last record . . . it just came out . . . I haven't heard it yet . . . He was an amazing influence on me; I really miss him. Since I was a kid, I've had all his records. Conceptually, he was a big influence on my life as far as writing, and the way he used horns, structurally keeping them loose. Tight and loose at the same time.

MR: What jingles [commercials] are you on these days?

RB: I do about ten a week. Burger King, Toyota, Kinney Shoes, Coke, Dodge, Maxwell House; I'll do any of them. Michelin Tires, Diet Pepsi, Chevy, NBC, Dodge Omni, TSS, whatever that is.

MR: Do they pay well?

RB: You never know. It's [a question of] residuals. Jingles pay $50 for an hour. If they run for a long time, you can make a couple thousand bucks, or you can make zilch. It's all a gamble. You get residuals. I don't know how it works, but I'm sure you're getting f - - - ed. It's easy to steal, especially if they don't play them in New York.

MR: What are some of the things that you hope to do in the future?

RB: We want to play with our own band more, maybe get into production, make our own records, and possibly some solo records in conjunction with the Brecker Brothers' records. Just get involved more in our own careers, rather than just doing record dates. We haven't been able to go out on the road as much as we'd like.

* * *

As with Randy, the conversation with Michael was lively and informative. I interviewed Michael on the following Sunday—Super-Bowl Sunday. Michael invited me to his loft in lower-Manhattan. The music room of the loft has a band set-up, including drums and a Fender Rhodes. Michael says that his neighbors don't mind the music, and that he has conducted all night sessions there, without a worry. This is a tall order for New York City.

MR: How did you first become a session musician?

MICHAEL BRECKER: Basically, I object to the question. I don't even know what a session musician is. The other phrase [I object to] is "studio-cat." I have played on a lot of albums, but I've always been kind of eclectic, in a weird way. It's hard to explain. The studio stuff that I've done has pretty

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much paid my bills, but I've always played "live." I've always been involved in a band for some project throughout my career, whether it's been Dreams or Brecker Brothers, Horace Silver, Billy Cobham, or whatever it might have been. I would always do dates when I was in town to pay for other things which I never made money doing. I look at it [playing sessions] as a necessity. A lot of times I like the stuff I am hired to do, but I am not a studio musician per se. Woodwind-wise, I don't double on a lot of instruments. A guy that you would consider a studio musician is someone that works day and night in the studios, and pretty much does just that. I am usually called [in] as a soloist, though sometimes I am asked to play in a section.

This whole past year has been a high percentage of studio stuff because we haven't been able to go out on the road. Going out on the road means losing money, so we've just had to wait it out here. If I had my way—and I don't like living on the road—I would be out there three-quarters of the year.

Through Dreams, which was a cooperative band years ago, I got into doing studio work. This was in late 1970-1971. The band had three horns and a real good rhythm section; it was an attempt to do something like Chicago, or B,S&T, but it was a lot freer, much more experimental. We improvised a lot, made up things spontaneously, kept it very loose. The horn section was me, my brother and Barry Rogers. As a result of those albums I started getting hired to do a lot of other things. The more you do, the more work you ultimately get. It's like a chain reaction.

MR: What was your first session?

MB: My first session was my brother's date, when I was right out of college. We did a record called *Score*. I think it was Randy, Chuck Rainey, [Bernard] Purdie, Larry Coryell and Hal Galper. That was the first session that I ever did.

MR: You mentioned that there are sessions that you enjoy as much as you enjoy playing "live."

MB: Sometimes, because of the exactness. It depends on what I have to do. I really enjoy putting solos on records. I did Art Garfunkel's [latest album] a few nights ago. There were just a few tunes that required real simple, melodic, just pretty-type solos, and I really enjoyed that.

MR: What are your favorite rhythm sections?

MB: That's a tough one because there are all different kinds of music. I love Elvin Jones, and Jimmy Garrison, who's no longer with us. Tony [Williams] and Ron [Carter].

MR: Have you ever played with Tony and Ron?

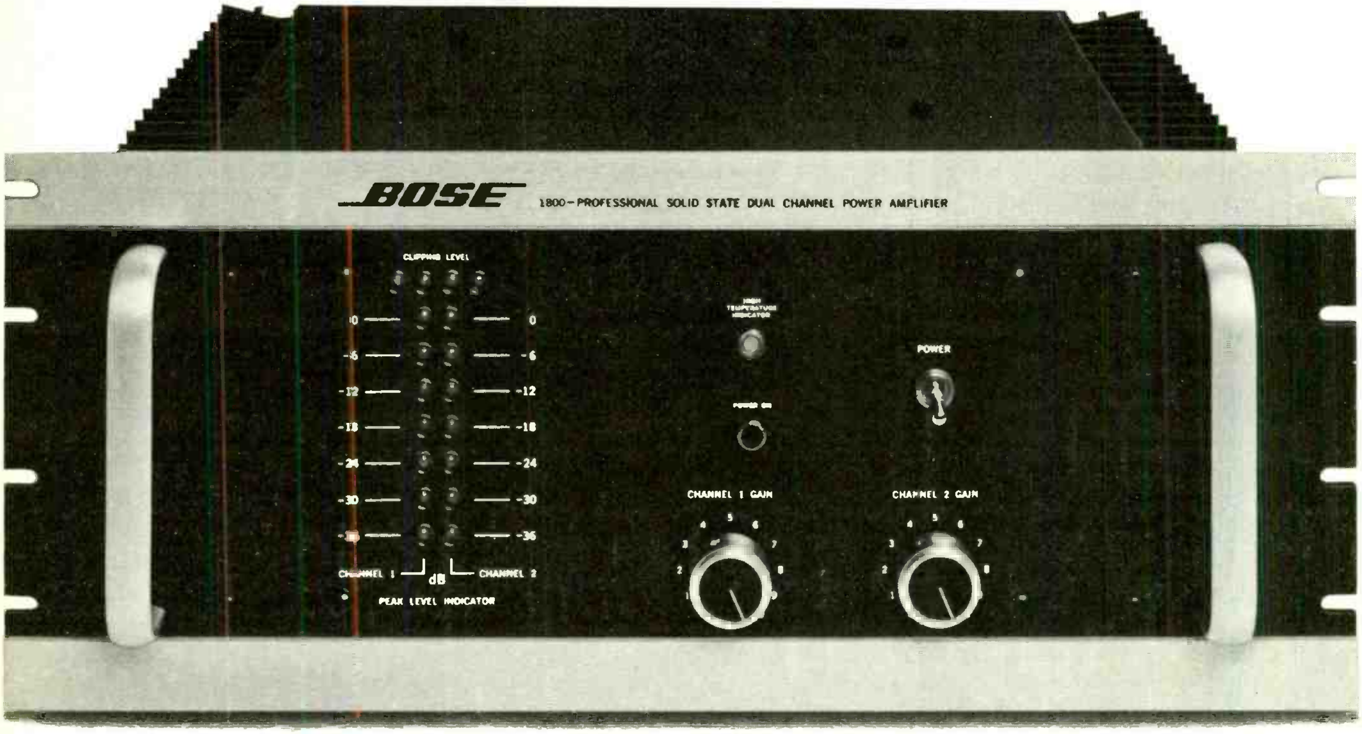
MB: Yeah, just on one album. I never played "live" with them. I think I did one album for Chet Baker with them [*You Can't Go Home Again*, Horizon-A&M]. I love Harvey Mason and Steve Gadd, Chris Parker, Stu Woods, Steve Jordan, Will Lee and Chuck Rainey. I don't want to leave anybody out.

MR: Favorite arrangers?

MB: My favorite arrangers right now are Charlie Callello, Dave Grusin, Bob James and Don Sebesky. There are a lot of cats. There's a guy in town who arranges great for horns, named Rod Mounsey. Tom Scott writes great. They know how to write for horns.

MR: Who are your musical influences?

MB: Everybody. There used to be a time when I would name Trane, Bird, Cannonball . . . I just can't limit it anymore. There have been so many in-



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fluences over the past ten years.

MR: Do you come home and listen to records?

MB: Not too often. Usually I listen to the radio, or I'll listen over someone else's house.

MR: Did you see your club as a place where your friends could come and wail?

MB: Hopefully, yeah! And not just limiting to jazz either, just good music—R&B, whatever it might be.

MR: Do you like the club's sound?

MB: We wanted to make sure that the sound was good and decent. It's generally better sounding than any of the jazz clubs around New York, but it still needs a little work.

MR: What microphones [do you feel] are good for the saxophone?

MB: It depends on the music, obviously. A lot of times I'll ask for an old U47 with the tubes in it, or 67s, occasionally 87s, perhaps an E-V RE-20. I've discovered a lot of times that from mic to mic, even though it's the same mic, there's a big variance, believe it or not. I'll never understand that. Sometimes I'll request a 47 and it will sound terrible. If I am soloing I'll usually change mics a few times to get the right sound. I get involved with placement. The big old RCA ones are good too. Those are great mics.

Generally, I think that horns were recorded much better, I hate to say it, ten, fifteen years ago than they are now. If you listen to any of the old bebop records, Rudy Van Gelder or old Art Blakley albums, the horns are just amazing, really warm and big, real sounding. Nowadays, the horns generally sound very condensed, and have a lot of limiting on them. A lot of times they sound very metallic. They don't get the real sound. It is possible to do it. All you have to do is work with the engineer. A lot of engineers aren't even used to working with horns, at this point, particularly the real rock 'n' roll engineers. You have to spend a little extra time with them for horns. I am always going for the sound that will benefit the kind of music being done.

MR: Do you think jazz artists crossing over to more commercial forms can become more successful?

MB: If your playing just happens to have more idiosyncracies, and is more esoteric like a lot of jazz musicians', and you try to make an obvious attempt at making your music commercial when it is naturally not meant to be, you're taking a big chance. It can work, but you can end up suffering a lot more in the long run if it doesn't work out. You can end up feeling like a complete idiot.

MR: Do you think artists like Coltrane could ever have large-scale commercial appeal?

MB: I think it would take a lot of education on a mass level over a long period of time to get people to understand what he was doing. I started listening to Coltrane when I was already a musician for a few years, and he was still alive. This was in 1963. I was playing the saxophone a lot then. I had a few of his albums for a year, and I really hated them. It took me a year to get into what he was doing, and this was during a period of time when I was already a musician. The music sounded so raucous to me, so frenzied, and so nuts.

A lot of jazz musicians say that the reason jazz records don't sell as well as disco records for instance, is purely marketing. Since it is not being played as much on AM stations, people don't get a chance to hear it or get used to it.

MR: Do you think that reasoning is partially right?

MB: No. I really don't.

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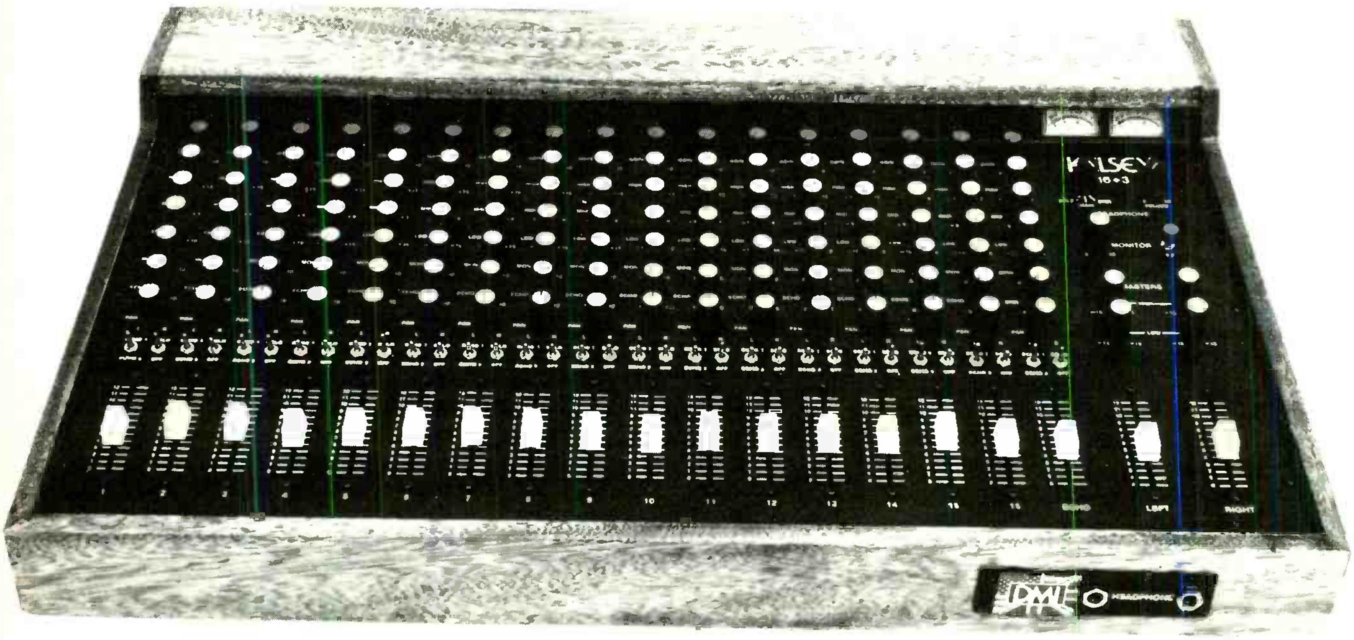

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Ambient Sound

BY LEN FELDMAN

Plugs 'n' Jacks 'n' Things

When I think of the number of times that I have had to interconnect a variety of cables and pieces of audio equipment, only to find that the terminating plug of cable "A" would not match the proper receptacle on amplifier "B," I often wish that I were in the connector/adaptor business. The types of plugs and jacks and multi-pin connectors and receptacles used in audio are almost without number and, if your experience is like mine, no matter how many kinds of "this-to-that" adaptors you keep in your junk drawer, sooner or later you will run up against a connection problem for which you are not equipped. Why so many different kinds of connectors? And which specific types are better for what individual applications?

Let's begin with home audio equipment. The output circuits of most amplifiers have an extremely low internal impedance—often almost a "short circuit." Even moderately priced power amps these days boast of damping factors in excess of 40. Since damping factor is nothing more than the load impedance (say, 8 ohms for a speaker) divided by the internal impedance of the amplifier, anytime you see a damping factor of 40, it means that the internal impedance of that amplifier is only 0.2 ohms. With that low an impedance and the high signal levels normally appearing at the speaker output terminals, it hardly matters if "open" unshielded wiring is used to get from the amplifier to the speaker terminals. And, with everything from common "lamp cord" to some of the newer, heavy-gauge, specially-braided cables now in use for amp-to-speaker wiring, there are no rules regarding the type of connectors used at these points in audio circuitry. You'll find everything from screw-terminals to spring loaded "push-and-insert-the-wire" connectors to three-quarter-inch spaced so-called "5-way binding posts" used as speaker output terminals and speaker connection terminals.

When we move back along the signal path however (from outputs towards inputs), the signal levels get lower and lower. Inputs to high-level circuits on amplifiers intended for home entertainment purposes are almost always phono-tip jacks. Frankly, of all the connector combinations ever conceived for the audio field, the phono-tip plug and jack is about the worst. Contact is maintained between the plug and the jack only by friction. Pull on a cable equipped with a non-molded pin plug in trying to remove it from its phono jack and chances are pretty good that you will tear the wire out of the plug. Still, the pin-plug, phono jack combination is about the simplest pair of connectors that can handle a shielded audio cable. And, when you get down to signal levels below 1.0 volts or so, shielded cabling is a must for interconnection of audio components. Often, at this point in the signal path, we are dealing with connections from relatively low impedances to fairly high input impedances. With even one end of the cable looking into a high impedance, magnetic fields in the vicinity of the cable can induce hum and noise currents in the system, and the simplest countermeasure is to use a single-center-conductor shielded audio cable. The only thing that the familiar pin-plug, phono jack combination has going for it is the fact that it is extremely easy to handle. It's easy to wire up. It does carry the shield or outer conductor all the way from the output of the previous component to the input of the next component. But it is, oh, so fragile and, in time, corrosion of the plated surfaces of such plugs and jacks makes for high-resistance connections and sooner or later you've got hum and noise pickup again caused by poor or lack of grounding.

One level above the phono-tip plug/jack combination in quality is the single-circuit phone-plug/jack combination. This plug and jack combination provides somewhat better contact between circuit points since the tip of the plug is retained in place by a spring-contact within the mating jack and a spring-loaded leaf of metal presses against the long ground-sleeve of the standard phono plug. In addition, modern embodiments of the phone plug have provision for strain relief and grab the outer insulation of the single-conductor shielded cable thereby preventing possible tearing of the conductors themselves. Inputs on amplifiers intended for professional application will generally use phone jacks rather than those flimsy phono-tip jacks.

A variation of the standard single-circuit phone plug/jack combination is the two-circuit phone plug and its matching two-circuit jack. With the advent of stereo nearly two decades ago, this plug/jack combination became extremely popular since it could handle any two circuits which shared a common ground. Requiring twin-conductor shielded cable (each inner conductor carries the "hot" signal of a single stereo channel, while the outer shield takes care of the common ground), the plug of this combination has its tip conventionally positioned up front, and insulated from the tip and the long sleeve is a "ring" of metal which carries the second of the two hot signals.

Meanwhile, Over in Europe

European manufacturers never did care much for the flimsy phono-tip plugs and jacks that we Americans used for hi-fi audio. For many years, these overseas manufacturers used a never-ending variety of plugs and jacks, none of which could be found in their neighboring countries of Europe, let alone in this country. After World War II, an organization known as the Deutsche Industrie Normen (basically, a German Standards Organization) adopted standards for audio interconnections. That gave rise to a whole new group of connectors known as DIN plugs and sockets. The DIN plugs and sockets combine more than one signal in a single connector. For example, the socket used as the TAPE DIN socket on a stereo tape deck will contain five terminals. Two of the contacts serve for left and right line-in connectors. An additional two serve as line-out, or tape-out signal points while the fifth contact serves as a common ground for all four signal circuits. Generally, DIN also specifies input and output level ranges for these various circuit points and, largely because so many signals are incorporated in a single multi-conductor cable and interconnected by plugs and sockets having very closely spaced terminals, signal levels per the DIN standards are lower than the nominal levels that we use in this country. There are DIN plug and socket arrangements for phono cartridge outputs and even for speaker to amplifier connections. One advantage of the DIN standard plug and socket arrangement, for example, is that if proper wiring is followed it is impossible to connect a pair of stereo speakers out-of-phase.

Professional Connections

As most readers probably know, in professional equipment many input and output circuits are "balanced" rather than unbalanced. In a balanced input circuit, both signal carrying conductors are "hot." That is, at the instant when there is a positive signal voltage on one of the two signal-carrying conductors of an audio cable, there is a negative but equal voltage on the other conductor. These two signal-carrying conductors usually end up at the ends of the primary winding of an input transformer. The center-tap of that winding is referenced to ground.

With balanced input or output circuits commonly encountered in professional equipment, then, it becomes necessary to use a three-terminal system of connectors and twin-conductor shielded audio cable. The ring-tip-sleeve type of two-circuit phone plug and its associated two-circuit phone jack can, of course, be used for such interconnections and very often is. But more often than not you will find that so-called XLR connectors are used. These are the most rugged plugs and sockets of all and are equipped with a locking system so that once a plug is inserted into a socket, the only way you can release the pair is by depressing a spring tab to unlock the mated plug and socket. XLR connectors are often called Cannon connectors (after the name of one of the manufacturers who produces them) but they are available from a number of manufacturers such as Amphenol, Switchcraft and others.

Most balanced input and output circuits have source and terminating impedances of 600 ohms. At least part of the reason for this choice of impedances has to do with the standard termination used for true VU level meters, whch is also 600 ohms. While 600 ohms would be considered a relatively low impedance, long runs of signal cables at an impedance level of 600 ohms might be subject to some hum and noise pickup if unbalanced lines were used. By using twin conductor shielded cable in which the signal conductors are twisted about each other, any hum induced in one lead is also induced in the other, but since these leads terminate at opposite ends of the primary winding of the matching input (or output) transformer, the induced voltages effectively cancel each other out.

Three-terminal connections are often also used even in balanced input/output situations. For example, most microphone connectors are of the three-terminal XLR type previously mentioned. Often, associated microphone cables may have an XLR connector at the microphone end, but a single-circuit phono-plug at the other end of the cable. When such arrangements are used, the two inner conductors of the cable are generally connected to the positive and negative signal outputs of the microphone element. One of the leads is effectively the "ground" side of the unbalanced signal, but this conductor is *not* connected to chassis ground at the microphone end of the line. Instead, it is joined with the shield only at the *plug* end of the cable. This technique prevents so-called ground loops which could develop if both ends of the shield were connected to both ends of the "cold" inner signal lead.

In cases where microphone outputs are "balanced" and of low impedance, the cable will usually be equipped with a 3-terminal XLR type connector at both ends. If such microphones are to be used with balanced inputs on mixers or preamplifiers, there is no problem and connection is made directly. In cases where such balanced lines must be connected to higher-impedance, unbalanced input circuits, it is necessary to interpose matching transformers. Such commonly available matching transformers have balanced primaries and unbalanced secondaries, which also step up the impedance (and signal levels) from the microphone. When using such step-up matching transformers, it is important that the long run of cable from microphone *precede* the transformer and that the transformer itself be located close to the actual input to the electronics of the system. In that way, the benefits of low impedance are retained but the microphone is readily coupled to higher-impedance circuits where and when this becomes necessary.

It would, of course, be nice if all audio equipment possessed only a couple of standard types of male and female connectors, but the audio industry, with all its diversity, has different needs for different situations and pretty much grew like Topsy. So, the best we can do is keep collecting all those adaptors whenever we find them and hope for good contact along the line.

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Denon DR-750 Cassette Recorder

General Description: The Denon DR-750 stereo cassette recorder is a two-head, two-motor model. The motor that drives the two capstans (one on either side of the tape path) is a DC coreless type associated with a servo system for detecting and controlling capstan speed in the interests of low wow and flutter and of accuracy of speed. Associated with the solenoid-control system is a built-in air damper designed to reduce "impact shock" on the tape heads when they are engaged (moved forward) for recording or playback. Nominally a front-loader and designed for upright or vertical placement, the deck also could be placed horizontally since there are four "feet" on its "back" as well as on its "bottom."

The cassette compartment is protected by a transparent door that slides up and into the chassis. There is no eject mechanism or button. To load a cassette, you position it and fit it into the area so that two small metal tabs at the top secure it in place. To remove, you take hold of the cassette and slip it out. When a cassette is in place (and the deck's power switch is turned on), it activates a small switch that illuminates the cassette compartment. A metal strip across the bottom of this area may be removed to facilitate access to the heads and capstans.

To the right of the cassette compartment are the tape index counter and reset button, a memory rewind switch and the power switch.

Below these items are the two meters, illuminated when power is turned on. The meters are calibrated from −40 to a little above +5. They may be switched to show either peak or VU levels.

Transport controls are grouped to the right of the deck below the meters. They provide for pause/mute;

record; forward (play); stop; rewind; fast-forward. The pause/mute button is actually a dual-function control: pressing it and holding it permits the tape to continue moving but with no signal recorded; releasing it stops the tape but leaves the deck in a "ready" state as per previous transport control settings. Pressing only the record button puts the deck in a stand-by condition for recording during which time input levels may be adjusted and viewed on the meters. Then to start recording, the next button (play) is pressed.

The transport buttons are "feather-touch" solenoid-operating controls that permit a "limited" kind of fast-buttoning. That is to say, you can go from either fast-wind mode to the other, and to play without pressing the stop button, and you can go from play to either of the fast-wind modes directly. You also can go into the record mode from either of the fast-wind modes by pressing both the record button and play button in rapid succession. To go into record mode from play mode, it is necessary to press stop, record and play. With a little practice this can be done quite swiftly.

The lower portion of the panel contains five knob-controls. One is for output level which handles both channels simultaneously on line and on headphone. Next is a control for fine bias-adjust during recording. Numbered from 1 to 10, it also has markings for LH (standard tape), FeCr (ferrichrome), and CrO_2 (chromium-dioxide). This last marking also has a detent.

The third knob is the main tape selector with four mixed positions—the same three as above plus a special stop for Co (cobalt-treated) tapes.

Line input levels are handled by the next control, a dual-concentric pair for adjusting either channel independently, or both at once. The last control provides similar options for microphone inputs. The arrangement here permits input mixing of line and mic signals.

Across the bottom of the panel are the headphone jack and pushbuttons for operation with an external timer, the meter peak-or-VU selector, a multiplex filter and the Dolby noise-reduction system. Farther right are the phone jacks for left and right channel mics.

The line in and out jacks, and the unit's AC power

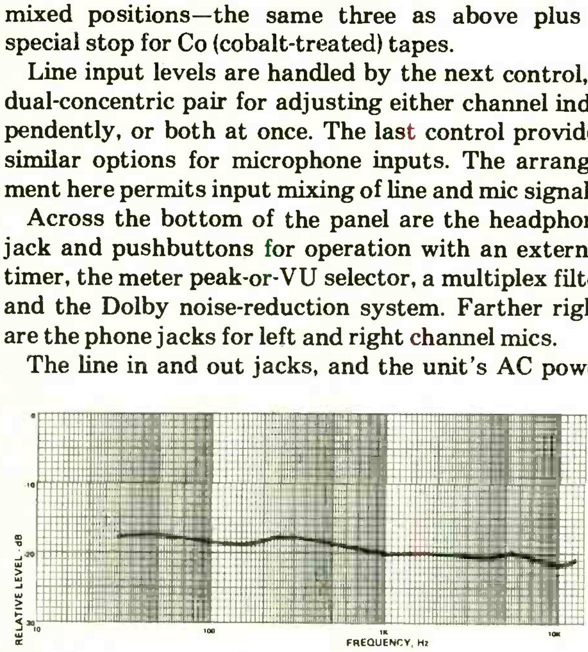


Fig. 1: Denon DR-750: Playback response using TDK AC-337 test tape.

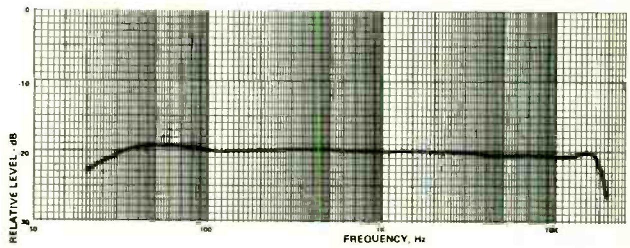


Fig. 2: Denon DR-750: Record/play response using Denon DX-3 C60 (LH) tape.

cord, are at the rear. The deck is styled in tones of gray and "grayish tan." All controls are clearly marked and layout permits relatively easy manipulation. The recorder came supplied with two pairs of stereo signal cables, a few cotton-tipped cleaning sticks, and two of Denon's own blank cassettes, a DX3 (normal or ferric oxide bias), and a DX5 (FeCr position).

Test Results: For a plot of record/playback frequency response, we used the Denon-supplied ferric-oxide tape. Its overall response on the DR-750 exceeded published specifications at both high and low ends of the frequency band, with the −3 dB rolloff point occurring at 20 Hz and at 18.5 kHz. To obtain this plot, we used the recommended setting of bias as noted in the owner's manual. Obviously, had we been willing to tolerate a little more harmonic distortion, we could have backed off a bit on the bias to obtain response all the way out to 20 kHz.

Using the suggested bias setting for TDK-SA tape (designated by Denon as "cobalt"), we plotted another record/play frequency response for this type of tape. Response this time extended a bit further at the high end (to 19.5 kHz), and was down only 1.5 dB at the 20-Hz low-frequency extreme.

In evaluating the headroom of this deck for both types of tapes tested, readers are reminded that "0 dB" on this recorder corresponds to a full 200 pWb/mm (Dolby level) rather than the lower levels of 165 pWb or 185 pWb/mm which some manufacturers set as their arbitrary "0 dB" point on the level meters. What did surprise us somewhat was the fact that although the signal-to-noise ratio obtained with the TDK-SA tape (referenced to the 3% THD point) was excellent at 57.5 dB without Dolby, it was just a shade poorer than the S/N figure obtained with the Denon ferric-oxide sample (58 dB).

In addition to these usual tests, we were able to run a few additional tests with the aid of a recently acquired series of prerecorded test tapes prepared by TDK. We now include a playback-only response curve, done with the aid of TDK test tape AC-337. This tape contains twelve spot frequencies, from 40 Hz to 12.5 kHz and is intended for playback with 120-microsecond EQ for the high end, and 3180 microseconds EQ for the low end (corresponding to the LH setting on the Denon's tape-selector switch). Reference level is −20 dB below 250 pWb/mm (315 Hz) for the test frequencies.

Using another TDK tape—AC-342 which is a 3 kHz-recording for wow and flutter and speed checks—we have, for the first time, been able to check speed accuracy of a deck by using a frequency counter to monitor the output during playback of this tape. Speed accuracy for the Denon DR-750 was an excellent 0.3 percent fast, exceeding the unit's spec. Wow and flutter, checked both in playback and in record/playback, was right on spec at 0.04% WRMS.

S/N with Dolby activated was among the best, showing 67 dB and 65 dB respectively for LH and for cobalt-biased tapes.

General Info: Dimensions (in vertical attitude) are 16$\frac{9}{16}$ inches wide; 11$\frac{7}{8}$ inches high; 9 inches deep. Weight is 27.5 pounds. Price is $1400.

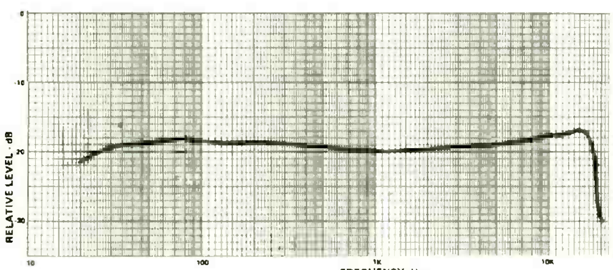


Fig. 3: Record/play response using TDK SA C-90 tape.

Individual Comment by L.F.: My first reaction to the Denon DR-750 was that it was very high-priced for a two-head cassette machine. Like many others, I have been conditioned to believe "three heads are better than two" and that therefore three-headed units necessarily cost more than even the best two-headed decks. I guess I wasn't really taking into consideration what can be done in a two-headed machine if one of those heads (the combination record/play head) is a Sendust-tip head as it is in this top-performing machine from Denon. The fact that it is possible to machine a super-narrow gap (between 1.0 and 1.2 micron) on this head (it is not as brittle as ferrite but it has almost the same long-wear characteristics), and that such a narrow gap—ideal as it is for playback—is nevertheless able to emit sufficient flux for proper recording negates the argument about "compromises" in record/play head designs.

There are advantages, too, in a two-head design which should not be overlooked. The problem of azimuth alignment, solved in various ways on three-head machines, is non-existent here since the record and play gap is one and the same. What is a bit tedious, of course, is optimization of bias level for a given tape. Although the owner's manual does list "optimum bias settings" of the continuously variable bias control for some fifty popular brands and types of tape, if you want to confirm optimization of the bias setting (and the owner's manual clearly suggests that this be done), the procedure—explained in detail—involves a trial-and-error sequence of record/play; record play that can be time-consuming. Still, I found the effect worthwhile, and I am of course all in favor of a front-panel bias adjustment.

Speaking of the owner's manual, I found it rather poorly laid out (especially when you consider the price of this deck), in that the front-panel diagrams are extremely small and difficult to read, and are located many pages removed from the verbal descriptions of the controls.

Individual Comment by N.E.: The Denon DR-750 has a lot going for it, and a few things that—in view of its high price—one might question.

In the deck's favor are its excellent audio response, its very nice S/N characteristics, its signal headroom, the built-in mixing facility, the option for watching either peak or VU levels on the meters and a tape selector with four (rather than the usual three) positions. The transport system also is commendable, being a smooth and—as far as we could determine—reliable mechanism with very low wow and flutter and a high order of speed accuracy.

Of course, it could be said that at its price, these characteristics are what one would expect. By the same logic, however, one would also expect a monitor head, albeit with the attendant need to align for azimuth or—as in the case of the physically unitized r/p heads—without that need. One also might expect unrestricted fast-buttoning for direct "run-in" recording, although this feature admittedly is hardly of paramount importance.

The absence of an eject mechanism and its associated button is a mixed blessing—there is nothing to get fouled up here but at the same time it does take a litte extra care to insert and remove a cassette. There is no Dolby/FM copy option, and no means of calibrating the Dolby circuit. Again, this can be debated both ways depending on your philosophy of cassette recorder design, and just how many controls and adjustments a cassette deck really needs, but not providing these features means they did not have to spend money on them.

Less debatable, it seems to me, is the rather low volume available from the headphone jack, even with the output level control turned to maximum.

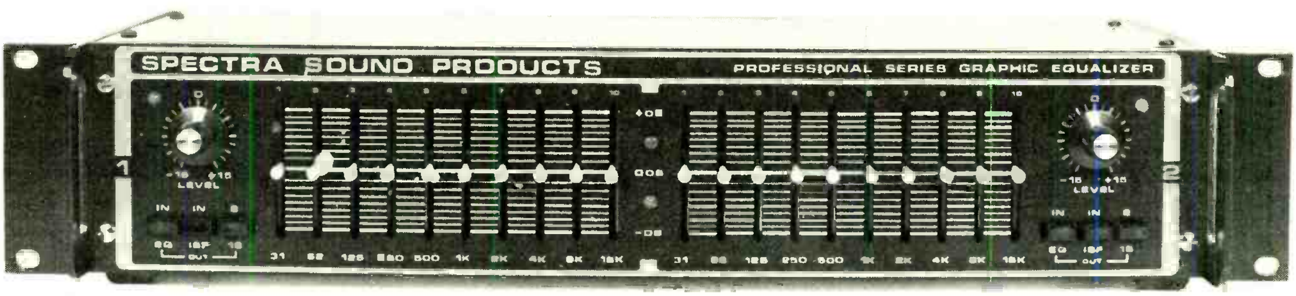
As for the owner's manual, it is unbelievable as the accompaniment for a recorder of this performance capability and price. It is awkwardly translated, "printed" in typewriter script, cheaply bound and inadequately illustrated.

DENON DR-750 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Frequency response,		
LH tape	± 3 dB, 35 Hz to 18 kHz	± 3 dB, 20 Hz to 18.5 kHz
Co tape	± 3 dB, 35 Hz to 18 kHz	± 3 dB, <20 Hz to 19.5 kHz
Signal-to-noise, LH tape		
Dolby off/on	NA/NA	58 dB/67 dB
Signal-to-noise, Co		
Dolby off/on	NA/65 dB	57.5 dB/65 dB
Wow and flutter (WRMS)	0.04%	0.04%
THD at 0 VU		
LH/Co	NA/NA	1.5%/2.5%
Record level for 3% THD		
LH/Co	NA/NA	+ 4/ + 1
Fast-wind time, C-60	70 seconds	67 seconds
Bias frequency	100 kHz	100 kHz
Mic input sensitivity	0.24 mV	0.29 mV
Line input sensitivity	77 mV	72 mV
Line output level	1000 mV	1000 mV
Headphone output level	90 mV (8 ohms)	82 mV (8 ohms)
Power consumption	38 watts	50 watts
Speed accuracy	± 0.5%	+ 0.3%
Playback response (120 μsec.)	NA	± 3 dB, <30 Hz to > 12.5 kHz

CIRCLE 19 ON READER SERVICE CARD

Spectra Sound Model 1000-B Graphic Equalizer



General Description: Ten bands of equalization are provided on each of the Model 1000-B's two channels. Nominal center frequencies are 31, 62, 125, 250, 500, 1k, 2k, 4k, 8k and 16 kHz. In addition to the sliders on each band, each channel has its own level control (marked from −15 through 0 to +15 dB), and a three-button switch group for EQ in/out, ISF in/out and 8/16.

The first of these buttons permits bypassing the EQ circuitry on either channel. The second button permits activating an infrasonic filter (cutoff is below 15 Hz at a rate of 12 dB/octave). The third button selects the total range for the sliders—either ±8 dB or ±16 dB.

Above each level control is an LED that comes on when the unit's AC power switch (located at the rear) is activated. The LED glows in green in normal operation. However, if overload is reached, the color changes to red. The slider scales have eight line-markings in each direction; they are not numbered since their working values are determined by the position of the "8/16" switch mentioned above.

At the rear, in addition to the AC switch, are a fuseholder, the AC line cord (fitted with a three-prong grounding plug), a grounding terminal and the unit's input and output signal connections. For input and output both unbalanced (1/4-inch phone jacks) and balanced (XLR) connectors are furnished. (An alternate version of the 1000-B with only unbalanced connections and priced $50 less, also is available).

Styled in black, with red markings, the Spectra Sound 1000-B is supplied with optional metal sidepieces to facilitate rack-mounting if desired.

Test Results: Published specs for the Spectra Sound Model 1000-B were either confirmed or exceeded in our tests. Especially noteworthy, the de-

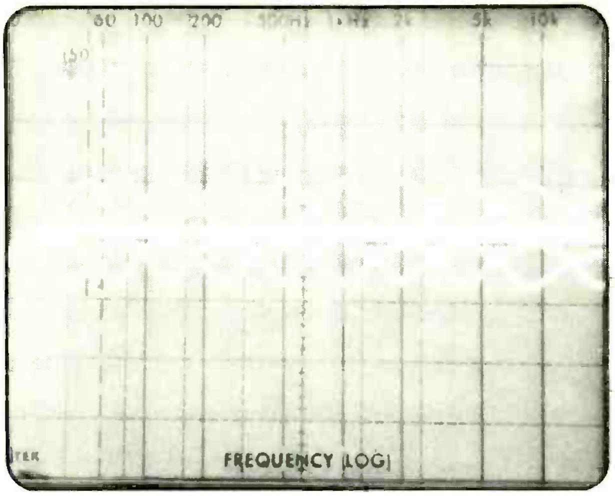


Fig. 1: Spectra Sound 1000-B: Boost and cut range of equalizer set to 8 dB range.

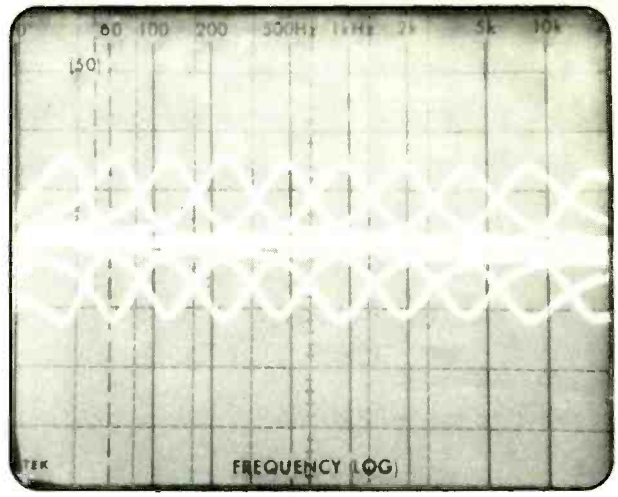


Fig. 2: Spectra Sound 1000-B: Boost and cut range of equalizer set to 16 dB range.

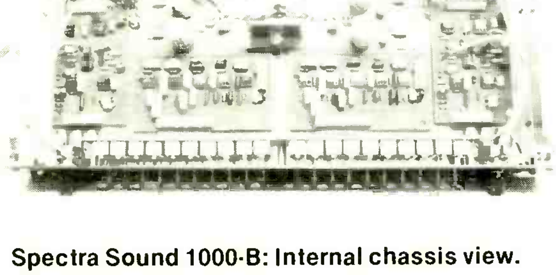
vice produced lower distortion than claimed (both harmonic and IM). Unweighted S/N was within 2 dB of spec, but the "A" weighted measurement at 100 dB was very impressive.

To study the range of control provided by each of the ten octave-levers, with the unit set to its ±8 dB range, a composite, 'scope photo was taken of the face of our spectrum analyzer (Fig. 1). This photo also shows the action of the subsonic filter (labeled on the unit ISF which stands for "infra-sonic" filter).

Our other 'scope photo (Fig. 2) shows the range of control of each center frequency when the ±16 dB range was selected. For both settings, we found center frequencies to be accurately positioned one octave apart. We also found that it was possible to set up complex response curves that probably exceed what might be needed if any real-life equalization situation.

The model we tested was fitted with both balanced and unbalanced inputs and outputs. If you don't need the 600-ohm balanced connections, you can save $50 by purchasing the alternate model in which the XLR connectors are replaced with plug buttons to fill the holes in the chassis that otherwise would be exposed.

We note that the unit's power off/on switch is on the back panel. This location makes sense since in normal use, one probably would prefer to leave the switch "on" at all times, energizing the unit from some master power switch elsewhere in the sound system. At that, even with the Model 1000-B energized continuously, its power consumption (a mere 8 watts) is low enough so as not to contribute significantly to the "energy crisis."



Spectra Sound 1000-B: Internal chassis view.

General Info: Dimensions are: width (with side pieces attached), 19 inches; height, 3½ inches; depth, 8 inches. Weight, 10 pounds. Price: $595. (Unbalanced only input/output version, $545).

Joint Comment by L.F. and N.E.: The number of graphic equalizers coming on the market (both for professional and for home hi-fi use) is of course increasing. Here is another one from what is basically a sound-reinforcement company located in Salt Lake City, Utah. The way they tell it, they decided to build their own octave equalizer after discovering that most of the units they had been buying off the shelf (at prices they thought could be justified) failed to meet their expectations insofar as their particular needs were concerned.

Accordingly, they designed and built their own, and this it it. Some of the things it has that we liked are its double-range boost and cut capability, its complete separation of circuitry for each channel, its input level control range, and its inclusion of a subsonic filter.

The double-range action of the sliders permits the user to achieve relatively close settings within the "throw" range of the slider. That is, where extreme values of boost or cut are not needed, the range can be set to ±8 dB and the actual degree of EQ can thus be more precisely accomplished.

The circuitry itself (see the "Internal View" photo) consists of separate and symmetrical channel layouts, right down to the power supply modules with their individual filtering circuits.

The wide-range gain controls (one on each channel) are at the input which means that you can install the equalizer just about anywhere in the signal path without encountering overload or poor S/N performance. The double-purpose LED (green for normal; red for overload) is a nice, handy touch.

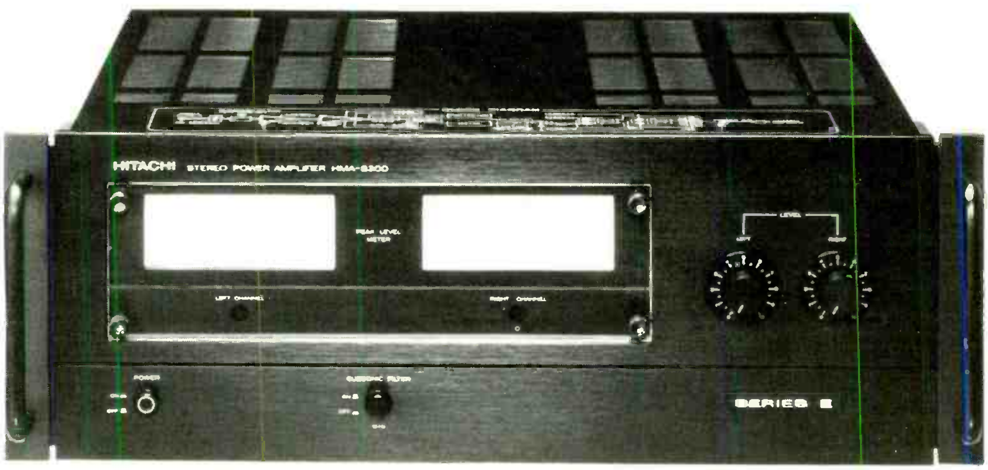
The subsonic (ISF) filter works very much as such a filter should, really lopping off signal crud below the audible range but achieving this without degrading the audible material above.

These characteristics, combined with the very low distortion and very high S/N ratio of the unit, not to mention the fact that its sliders do what they are supposed to do, add up to a very competent graphic equalizer—and at a very fair price on today's market.

SPECTRA SOUND 1000-B GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Rated output	+4 dBm	Reference level
Maximum output	+18 dBm	+19.5 dBm
Frequency response	NA	±3 dB, 10 Hz to 70 kHz
THD (at reference output)	0.008%	0.005% at 1 kHz 0.0035% at 20 Hz 0.0055% at 20 kHz
IM distortion (at reference output)	0.008%	0.006%
Signal-to-noise	95 dB (unwtd)	93 dB (unwtd) 100 dB ("A" wtd)
Input impedance, balanced	20 K or 600 ohms	confirmed
unbalanced	100 K ohms	confirmed
Output impedance, balanced	600 ohms	confirmed
unbalanced	100 ohms	confirmed
Subsonic filter cutoff	NA	−3 dB at 20 Hz
Power consumption	NA	8 watts

Hitachi Model HMA-8300 Power Amplifier



General Description: The Hitachi HMA-8300 is a stereo (two-channel) power amplifier designed for operation from a preamplified or line-level source. Its input sensitivity is 1 volt for a rated continuous "RMS" power output at 8 ohms of 200 watts (on each channel). It can, of course, hit peaks well above that level, and it can accommodate 4-ohm and 16-ohm loads.

The front panel contains the power off/on switch, a subsonic filter switch (effective below 15 Hz), a pair of level controls and two power meters. These show output power in watts and in dB values (from −40 to +5, with 0 dB calibrated for 200 watts).

The rear contains two "pin jacks" for signal input, two pairs of "press-to-connect" speaker outputs and the unit's AC line cord. The amplifier is housed in a black metal case fitted with four feet.

The circuitry employed in the HMA-8300 is known as "Class G" although it also is referred to in the owner's manual as "Series E" and again by the term "Dynaharmony." Basically, what it means is that in the amplifier's output circuit, two pairs of output transistors handle the output current. Under "low" signal conditions a lower-powered pair—supplied by the lower of two B+ voltages—handles the output current. When signal levels exceed this supply voltage, this pair of transistors shuts down and a more powerful pair—fed with a higher level of B+ supply voltage—takes over to complete the waveform amplification. In this way, each pair of transistors is intended to be operating over its most efficient range most of the time. The result is claimed to be higher overall efficiency and less heat generation under long-term musical listening conditions. No forced ventilation system is provided, and apparently none is needed.

Built-in protection includes circuitry to limit current and to shut down power amplification, and a DC-voltage detection system to safeguard speakers.

Test Results: Treated as a 200-watt per channel amplifier, the Hitachi HMA-8300 produced superb test measurements, easily exceeding its published specifications. The amplifier actually delivered 242 watts per channel at mid-frequencies, and a comfortable 220 watts at the frequency extremes before reaching its rated THD of 0.1 percent. Hitachi does not specify an IM distortion figure, but we measured it as a very low 0.01 percent for the 200-watt rated output, and we were able to push the amplifier to an output of 253 watts before reaching 0.1 percent IM.

The plot of distortion versus frequency for rated output of 200 watts (Fig. 2) shows that with both channels driven into 8-ohm loads, THD remained well below the 0.1-percent level at all frequencies tested.

Frequency response as such extended from 3.5 Hz to an incredible 250 kHz for the −1dB rolloff points with the subsonic filter out of the circuit. With this filter switched on, the −3 dB rolloff occurred right at 15 Hz.

Readers may note a discrepancy between Hitachi's signal-to-noise figure and ours. This difference arises from the fact that ours was measured with reference to a 0.5-volt input for level controls adjusted to a 1.0-watt output (the new IHF standard), while Hitachi's rating is referred to maximum rated output with gain controls wide open. In any event, the 88-dB figure we obtained is quite good as compared with S/N figures measured on other amplifiers.

During our tests, the amplifier ran relatively cool, with no need for forced ventilation as it might require were it of the "Class B" configuration and of this size and power rating. This observation, plus the obvious lack of "top-heavy" or massive heat-sinking, speaks well for the "Class G" approach.

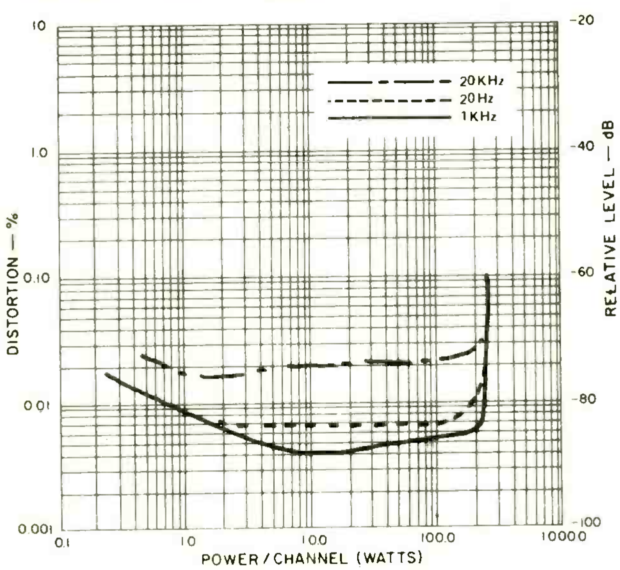


Fig. 1: Hitachi HMA-8300: Harmonic distortion vs. power output.

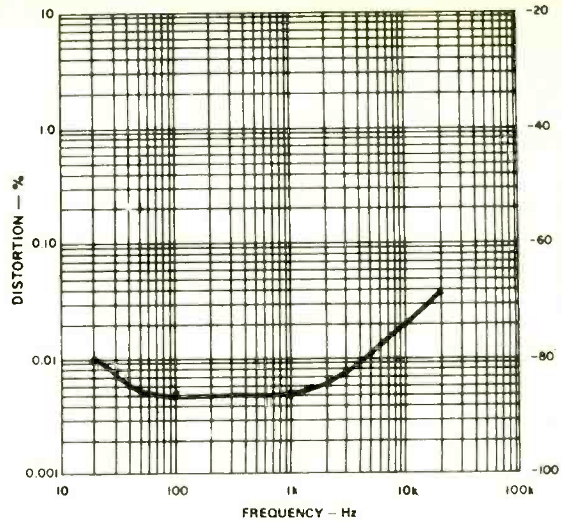


Fig. 2: Hitachi HMA-8300: Distortion vs. frequency at rated output, 8-ohm loads (200 w/ch.).

General Info: Dimensions are 17⅛ inches wide; 7³⁄₁₆ inches high; 16 inches deep. Weight is 53 pounds. Price is $800.

Individual Comment by L.F.: I am all in favor of amplifier circuits that improve efficiency without degrading audible performance, and Hitachi's HMA-8300 incorporates just such innovative circuitry, although apparently they have called it by three different names (Class G, Series E and Dynaharmony). All these terms of course mean the same thing: two pairs of output transistors to handle lower and higher output levels.

My first concern regarding this configuration was over possible additional notch distortion at the crossover points between the lower- and higher-powered transistor takeover. But this concern proved unfounded. Apparently Hitachi has gone to a lot of trouble and circuitry to prevent this form of crossover distortion.

I do have one serious gripe—not about the amplifier, but about the published specs in owner's manual. Would you believe that in this day and age, Hitachi is quoting "peak RMS power" (or music power) of 400 watts per channel for what is, in fact, a 200-watts-plus continuous power rating? For shame, fellows. Haven't you heard of the FTC Power Rule that has been in effect since 1974?

As it happens, the HMA-8300 does have a much higher short-term power output rating than do most conventional amplifiers. This is indicated (or should be, these days) by the new IHF measurement known as "dynamic headroom." As shown in our own table of measurements (Vital Statistics), this amplifier's dynamic headroom turned out to be 2.77 dB, or very nearly double the unit's nominal or continuous 200-watt rating, without encountering clipping. In sound reinforcement and in "live" performance situations, that kind of headroom is an advantage, since the performer or sound reinforcement person may not always be in a position to see the power output meters. Even if someone is riding gain, the meters probably won't be fast enough to catch those high-powered transients anyway. I suggest that Hitachi start using the dynamic headroom figure instead of that misleading 400-watt per channel number if they hope to avoid any clash with the Federal Trade Commission.

This point aside, sound reproduction from the HMA-8300 seemed to me flawless, and the transition from the lower-powered to the higher-powered transistors—even under high transient musical signal conditions—was inaudible. In fact, transient response seemed particularly good with this amp, despite the complexity of its output circuit and its required "switching" under high-level conditions. The price does not seem out of line for an amplifier of this ruggedness and performance capability. There are very few amplifiers today that deliver high power of this sort at "two dollars per watt." And since the days of "dollar-per-high-quality-watt" are past, I repeat that I see no reason for Hitachi's "inflated" music power rating of 400 watts per channel (especially when the amp's outstanding performance speaks for itself). Next thing you know, they may start combining the power of both channels to call it an "800-watt" amplifier!

Individual Comment by N.E.: We all know the expression about someone "being a good guy in spite of himself." It applies, sort of, to this amplifier. It is a great-sounding, and apparently very substantially built, amplifier in spite of some aspects that do not go down as neatly as one might wish.

For one thing, there's the inflated power figure which is based on the now discredited and abandoned "music power" rating. In the letter, if not the spirit, of the law, the more legitimate 200-watt rating is printed in a type size somewhat larger than the other, but that 400-watt figure does seem somewhat campy.

In trying to characterize this product as either an "advanced hi-fi" or a "professional" component, we once again face a problem that may be more semantic than technical. The Hitachi HMA-8300 certainly has the guts and the ruggedness to pump out vast amounts of clean, high audio power. Yet its "personality" suggests an appeal that is perhaps less than that customarily considered "professional." Its inputs are "hi fi" pin jacks; its speaker outputs are press-to-connect types you'd not expect to find on an $800 power amplifier. There is no provision for adding speakers, and none for combining both channels for a higher mono output. The unit is not styled for rack-mount.

The unit's chief claim to fame, of course, is its use of "Class G" circuitry which, simply put, uses different output circuits for "low" and for "high" power handling. We are not told what these values are, but Len and I have calculated that somewhere near the rated continuous power seems to be the dividing line between the two output circuits. This means, then, that at power levels up to, say, 190 to 200 watts, the amplifier is indeed operating more efficiently than a conventional amplifier would. Not sounding better necessarily, but drawing less voltage from the power line—which in itself is a good thing, especially these days. And apparently too, there is no heat problem with this piece of equipment.

At full rated power and above, however, there seems to be no significant difference in operating efficiency between this and a conventional amplifier (we did find that the HMA-8300 pulled 940 watts from the line to produce a continuous output of 220 watts per channel across the band). On the other hand, a more tangible benefit of this design is the nearly 3 dB more of signal headroom (about 400 watts per channel) it can provide when the program material and/or listening conditions call for it.

Whether the hi-fi system owner ever needs that kind of power is highly debatable. A pro or semi-pro user, of course, could need that much power and if so, the Hitachi will produce it. These same pro and semi-pro users, though, might wish the amplifier had more of a "pro personality" such as handles and professional input and output connectors.

Note: The power cord is fitted with a two-prong plug but it should not be connected to a "convenience outlet" on a preamp-control unit because the current drain may exceed the safety limits of the off/on switch of that preamp. Unless you are certain of the power ratings included, the HMA-8300 should be connected to its own AC line source and switched on and off by its own power switch.

HITACHI HMA-8300 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Continuous power/channel, 1 kHz	200 watts	242 watts
Continuous power/channel, 20 Hz to 20 kHz	200 watts	220 watts
Power bandwidth	20 Hz to 20 kHz	13 Hz to 25 kHz
Frequency response	+0, −1 dB, 5 Hz to 80 kHz	±1 dB, 3.5 Hz to 250 kHz
Damping factor	50	71
Rated THD	0.1%	0.0055%
Rated IM	NA	0.01%
Residual hum and noise	110 dB ("A")	88 dB (new IHF)
Input sensitivity	1.0 V	1.0 V
Dynamic headroom	NA	2.77 dB
Power consumption	850 watts	940 watts (at full output)

Neutrik AD4 Analog Delay Unit

By Jim Ford and John Murphy

Most of the audio time delay equipment currently available uses digital techniques to delay the input audio signal. However, the Neutrik AD4 uses strictly analog circuitry to provide up to 200 milliseconds (msec.) of signal delay. The unit was primarily designed to provide four discrete time delay outputs from a single input for improving the intelligibility of distributed sound systems. But it could be used in any application requiring single or multiple time delays. The input signal is delayed through the use of "bucket brigade" analog delay modules. The manufacturer's suggested price is $795.

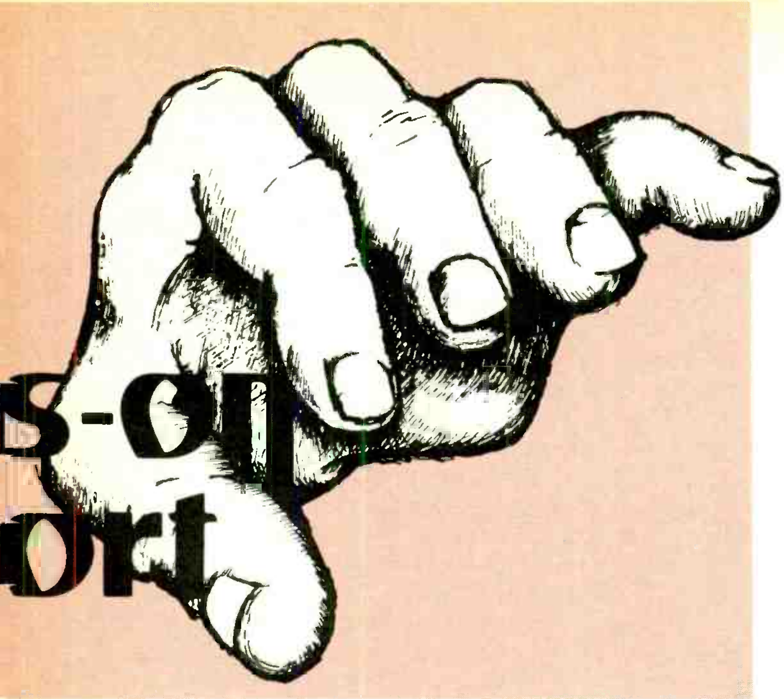
Background
What's an Audio Delay Unit?

The oldest and probably most familiar audio delay technique is the tape loop echo. By using a closed loop of magnetic tape and running it continuously past the separate record and play heads of a tape machine a delay system is created. The signal into the tape machine is recorded onto tape and played back an instant later as the tape passes the playback head. The output of the tape machine is then a replica of the input, the only difference (ideally) being that it is slightly delayed in time. The amount of time delay is controlled by the tape speed and the distance between the record and play heads. When the delayed signal is monitored along with the direct (non-delayed) signal the effect is perceived roughly as "echo." Until recent years this was the only practical method of obtaining a time delayed signal.

In the past few years, however, many purely electronic audio time delay systems have found their way to the marketplace. These devices perform exactly the same function as the tape loop echo. That is, the output is a copy of the input delayed in time by some fixed amount. But these units use electronic integrated circuits (ICs) to provide the signal delay, rather than a tape recorder. There are basically two forms of electronic time delay: digital and analog.

The digital technique employs analog-to-digital (A/D), and digital-to-analog (D/A) converters and a computer-like memory device. An analog audio signal (like the signal fed to a regular tape recorder) is input to the unit and undergoes A/D conversion. The result of the A/D conversion is an extremely rapid (approximately 50,000 per second) sequence of numbers (digits!) which represent the audio signal. The numbers are fed to the memory which simply stores them, waits the prescribed amount of time (whatever the user has selected for a delay time, say for example 50 milliseconds), and reads the numbers out in the same sequence that they were stored. The output stream of numbers then undergoes D/A conversion whereby the analog input signal is reconstructed. This signal is then sent to the unit's output. The result of all this is a digital delay unit.

Analog delays, on the other hand, employ neither signal converters nor microcomputers. Instead, they utilize special purpose integrated circuits known variously as analog shift registers, bucket brigade IC's, charge coupled delay lines, or simply, analog delay ICs. Basically, the input signal is fed to the IC and the delayed signal taken from the IC's output. What simplicity! The integrated circuit chip does all the work! Inside the IC, though, things are not so simple. The circuitry is rapidly (again, about 50,000 times per second) sampling the input signal level and storing the signal voltage levels in a long series of capacitors. The stored voltage levels are stepped along from one



capacitor to the next (much like buckets of water in a firefighters "bucket brigade") until the signal arrives at the end of the line and it outputs as a delayed copy of the input. The delay time depends on how many stages (capacitors) the signal passes through from input to output, and on how rapidly the signal is stepped from one stage to the next.

The analog and digital approaches to time delay each have their own advantages and disadvantages. Digital delay techniques, as a rule, offer the highest audio quality with regard to extended high-frequency response, dynamic range (signal to noise ratio) and distortion. However, analog delay units typically cost much less than digital-type units. There is a trade-off between high-frequency response and amount of time delay for both analog and digital types. With longer delay times, high-frequency response is usually compromised. When shorter delays are used, high-frequency response is widest. Both types also tend to have limited dynamic range, making it necessary to monitor signal levels through the unit. As with a tape recorder, the user tries to keep levels as high as possible without overloading the machine. Some units use internal noise reduction systems to improve overall dynamic range and make level settings less critical.

Applications
What Can I Do with It?

There are two general areas of application for audio time delay units. First is the application of time delay in sound reinforcement systems for the purpose of improving overall intelligibility. Second, time delay is widely used for sound effects during musical performances both on the stage and in the recording studio.

Consider the situation depicted in Figure 1(a) where we have shown a typical distributed sound reinforcement system. The system consists of one central loudspeaker cluster and four supplementary loudspeaker groups. Since there is no delay used here, sound leaves each speaker group simultaneously and the first sound to arrive at the listeners is from the supplementary groups. Sound from the central cluster arrives at some later time; overall system intelligibility will tend to be reduced if the time difference in the arrival of sound from the two sources is greater than about 30 milliseconds. Since the time difference is greatest at the back of the hall, that's where intelligibility will be poorest. In Figure 1(b) we have depicted the same situation, except that a time delay unit has been used to delay the sound going to the supplementary speakers. The delay time (T) is chosen so that sound from the supplementary speakers arrives at the listener at about the same time as the sound from the central cluster. The overall intelligibility of the system will be greatly improved as a result of synchronizing the arrival times of the sound from the different sources. An added benefit is that the apparent source of the sound can be maintained at the front of the hall if the sound from the central cluster is allowed to arrive at the listener slightly (a few milliseconds) before the sound from the supplementary groups. This is because the ear takes its localization cues from the first sound to arrive. In the system without delay the apparent source of the sound would be directly overhead rather than the front of the hall. That could be very distracting.

When used in conjunction with a musical performance, audio time delay units can provide a variety of effects. The most obvious use is in the same applications as tape loop delay. That is, slap back echo. By using a delay time of 35 msec. or more, the delayed sound is perceived as a distinct echo. If a delay time less than about 35 msec. is used, the effect is perceived less as an echo and more as a "doubling" of the original sound. In "live" performances this can give an effect similar to the "double tracking" available when recording in the studio. But the effect never seems to be quite as good as true "double tracking." This is probably because when the performer records the second track in the studio his performance will always vary slightly from the original, whereas the delay unit gives an *exact* copy of the original (non-delayed) performance. The variations in the two *recorded* tracks introduce a richness that is not found in the electronically "perfect" delay line doubling. Some manufacturers have taken the delay line a step further by introducing a means of modulating the amount of time delay by a small amount, thereby introducing some of the imperfections and variations that would be present in true "double tracking."

There is another useful application of time delay in conjunction with a reverberation system. By delaying the input signal to the reverb it is possible to improve the quality of almost any type of reverb. This makes sense when you consider that in a concert hall there is definite time lag between the arrival of the direct sound from the stage and the arrival of the sound reflected from the room surfaces. In most reverberation units there is no delay between the time a signal is input to the unit and the time the reverberated signal appears at the output (unless the reverb uses a delay line of its own). So delaying the reverb send signal can make a reverb sound more "natural." Another way of looking at it is to realize that the delay allows for some "space" between the direct and the reverbed sound. The delayed reverb doesn't mask the direct sound.

Description: The Neutrik AD4 is an extremely simple unit to operate. It has a single line level input (with a switchable 26 dB pad for attenuating hot input signals) and four delayed signal outputs. Each output has an individual output level control. There is a continuously variable input level control and an input signal limiter than can be switched in ("fast" or "slow") if desired. The amount of time delay through the unit is controlled by a single continuously variable control. This control is labeled 50 msec. at the minimum setting and 200 msec. at the maximum setting. These times refer only to the delay times for the fourth output. The first, second and third outputs provide 1/4, 1/2 and 3/4 the delay time of the fourth output (in that order). There is a single LED that lights up whenever the input signal exceeds the unit's overload threshold. When the limiter is switched in, this same LED lights to indicate that gain reduction is taking place in order to avoid overload. Outputs three and four are equipped with "active noise filters" to reduce the noise accumulated on the longest delays. (The delay stages are sequential with the signal of the fourth output having passed through all four stages of delay.)

The input and output connections are made on the back panel by way of XLR type connectors (with the signal on pin 2, not on pin 3). The input is at the far left with the switch for the 26 dB pad just to the right. Next are the four output connectors, one through four from left to right. Just to the left of each output connector is a level control for that output. There is a receptacle for the detachable line cord at the far right of the back panel, and the unit is equipped with an internal A.C. line fuse.

The power on/off switch is located to the far left of the front panel with a small green LED pilot light beside it to the right. The delay time control is to the right of these. Skipping to the right side of the front panel there is first the overload/gain reduction LED indicator, and next a three-position toggle switch for the limiter. With the switch all the way down the limiter is off. The first limiter On position is labeled "fast," the second On position is labeled "slow." These refer to fast or slow limiter response. The input level control is at the far right of the front panel. The unit is light weight and can be mounted in a single rack space of a standard 19-inch rack.

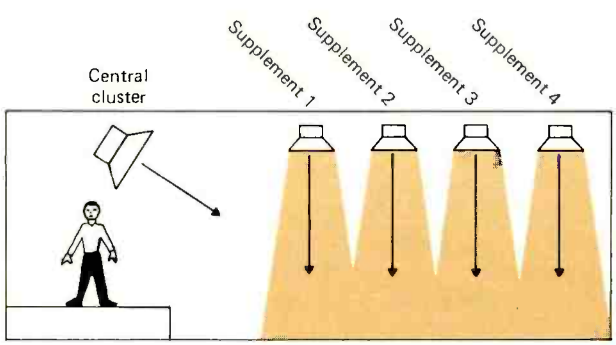


Fig. 1(a): With no delay, sound from supplementary loudspeakers arrives first.

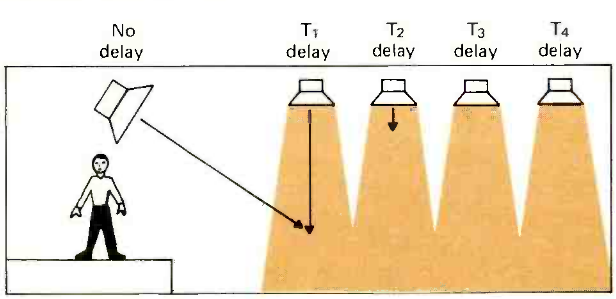
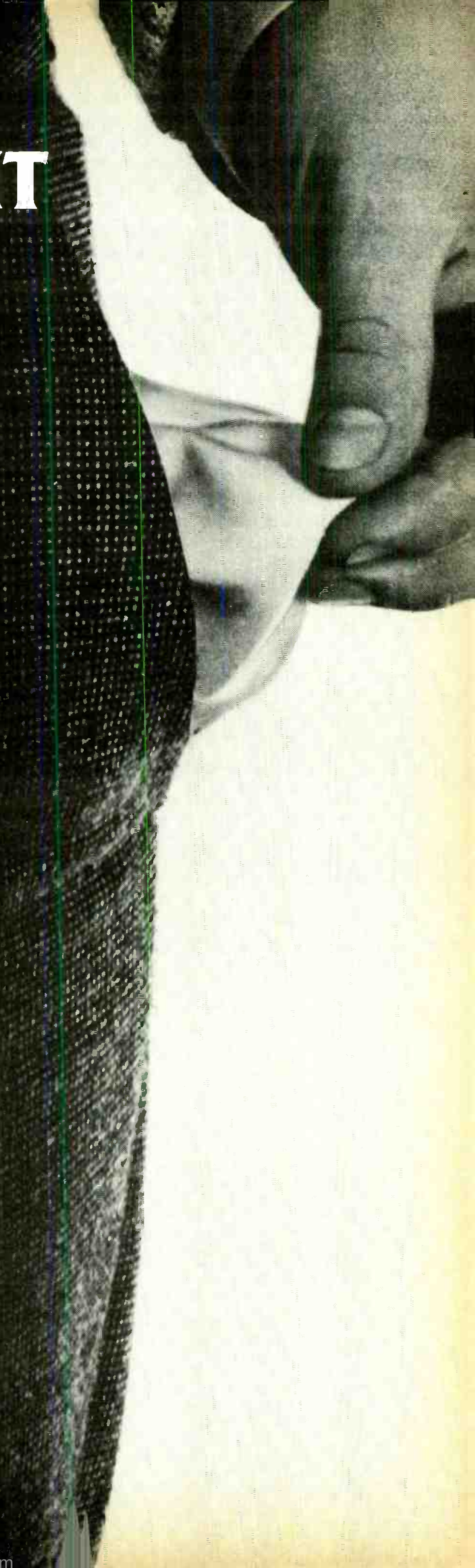


Fig. 1(b): With appropriate delay, sound from the central cluster arrives at the same time as sound from the supplementary loudspeakers.

Listening Test: We patched the AD4 into our reference system and selected some recorded music to check it out with. Then we adjusted the input level control so the overload indicator just flashed on occasional peaks, and set the output level control so that there was no level change when the unit was alternately switched in and bypassed at our preamp. First we listened to the fourth output with the delay time set at maximum, since from our lab tests we knew this would constitute the worst case for the unit's performance. Then we listened for any significant differences when we switched the delay in and out of the system. The most clearly audible difference was in the high-frequency range. With the unit switched in there was an audible loss of highs. For example, cymbals and other high-frequency sounds became weaker and lost some of their sparkle. Also, between cuts on albums there was audible hiss whereas with the unit switched out the noise virtually vanished. Another thing we noticed was a slight harshness in the sound when listening through the delay. With shorter delay settings and listening to the shorter delay outputs the audio quality of the unit improved somewhat, but there was still an audible difference in audio quality with the unit switched in.

However, keeping in mind the delay's intended application (intelligibility improvement in distributed sound systems) the shortcomings cited above become less important, especially with less than maximum

CLEAN UP YOUR TRACKS WITHOUT CLEANING OUT YOUR POCKETS.



The past few years have seen a proliferation of exciting, inexpensive new multi-track hardware. Unfortunately, this hardware retains two shortcomings which prevent the small studio from really competing with the "big boys"—noise and lack of headroom. These shortcomings become especially apparent when the tracks you're bouncing start sounding like a transmission from outer space.

dbx, maker of state-of-the-art tape noise reduction for the world's leading studios, also makes a line of products designed for the small studio: the 155, the RM-155 and the 158. All offer the same 30dB of tape noise reduction and 10dB of headroom improvement as our more expensive units, with which they are fully compatible.

The 155 offers four channels of tape noise reduction, switchable to record, play or bypass. Each channel is self-contained on a user-changeable modular circuit board. All this, for under $600. Also available rack-mounted (RM-155) for four-channel simultaneous or eight-channel switchable use.

The 158 offers eight channels of simultaneous tape noise reduction, rack-mounted in a compact chassis. It lets you monitor the noise-reduced signal while you record. Spare channel included.

A dbx tape noise reduction system is simple to install and use. It will give your demos the sound quality of master tapes. And when you're ready to expand, your dbx system grows with you, easily and inexpensively.

Above all, the product you produce will be very close to that of the "big boys," for a lot less bucks. That is why, for the small studio, dbx tape noise reduction is a necessity, not an accessory. dbx, Incorporated 71 Chapel Street, Newton, MA 02195 617-964-3210

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delay. The unit would probably have no trouble providing acceptable audio quality in a distributed sound system handling mainly speech material.

Lab Test: When we got the unit on the test bench we measured all the usual quantities: input/output levels, noise, distortion and frequency response. The results of these measurements are tabulated in our chart. The signal-to-noise ratio was in the neighborhood of 60 dB depending on the amount of delay used. Since this is about the same signal-to-noise ratio that's available from a quarter-track tape machine it's very important to closely monitor the input levels to get the best performance with respect to noise. All of the signal levels are referenced to the overload threshold = 0 dB.

Based on our observations of the input and output levels of the device interfacing with other line level gear should present no problem.

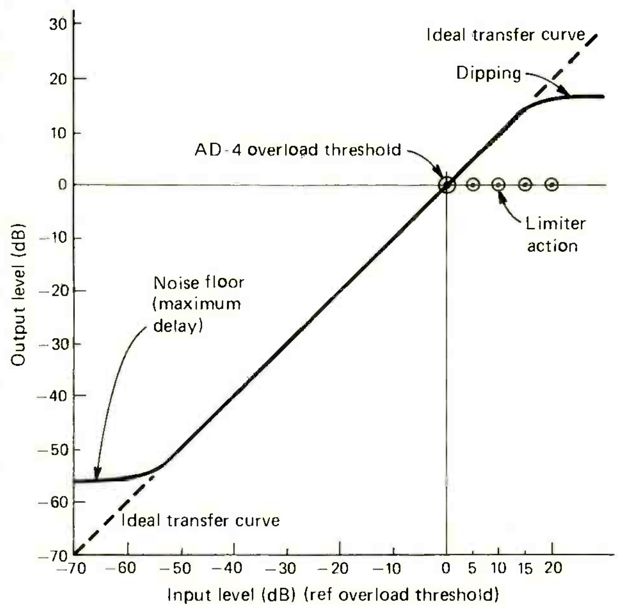

Fig. 2: Input/output transfer curve.

As expected, the signal distortion of the unit is dependent on the time delay used. The more delay you use, the more distortion you get. Our distortion and noise measurements were compounded by the unit's pickup of local radio stations, but no RF interference was observed during our listening test.

The frequency response of the unit was also delay dependent. With minimum delay, the high-frequency response was down 3 dB at about 10 kHz and cut off very sharply above that. Low frequency response extended well below 20 Hz. Response never varied more than about 1 dB between the frequency extremes.

When the limiter was switched in, the distortion was effectively held to the 0 dB levels. Even with a +20 dB input signal the distortion increased only slightly over what it was at 0 dB. The output level was of course held to the 0 dB level with the limiter switched in (it's not magic!) A +20 dB input level caused the output to increase less than 1 dB over the output with a 0 dB input level. That's pretty effective limiting. The input/output transfer curve in Figure 2 gives a graphic presentation of the limiter's action. The output clipping level without the limiter (about +15.5 dB) is also shown on this graph.

The AD4 is constructed on one large printed circuit board to which the various pots, switches and connectors are attached. There are eight tiny printed circuit boards mounted vertically off the main board. These contain the actual analog time delay ICs.

Conclusion: Neutrik does not supply an owner's manual with the AD4 because of its straightforward operation. But even so, it would seem appropriate to give the owner some applications information.

We recommend that you consider using the AD4 in distributed sound systems where absolute audio quality is not critical. The unit can do a good job of improving intelligibility in these systems at a modest cost compared to digital units.

LAB TEST SUMMARY

(Note: All dBV levels are referenced to .775 volts)

Input Levels

(level required to just light the overload indicator)

Input Sensitivity Control at Minimum:
.31 Vrms (−8 dBV) w/o pad
6.2 Vrms (+18 dBV) w/pad

Input Sensitivity Control at Maximum:
5.7 mV (−43 dBV)

Output Levels

(with input level at the overload threshold as above, and output level at maximum)

1.2 Vrms (+3.8 dBV)

Noise Levels

(referenced to the output level stated above, i.e., signal at the overload threshold)

Output No.	Minimum Delay	Maximum Delay
1	−66 dB	−57 dB
2	−62 dB	−54 dB
3	−62 dB	−60 dB
4	−60 dB	−57 dB

Frequency Response

(measured 10 dB below the overload threshold)

Output No.	Minimum Delay	Maximum Delay
1	12 Hz to 9.9 kHz +1, −3 dB	12 Hz to 8.7 kHz +1, −3 dB
2	12 Hz to 9.7 kHz +1, −3 dB	12 Hz to 7.8 kHz +1, −3 dB
3	8.5 to 9.8 kHz +1, −3 dB	8.9 Hz to 7.9 kHz +1, −3 dB
4	8.2 to 9.3 kHz +1, −3 dB	8.5 Hz to 6.5 kHz +1, −3 dB

Distortion

(total harmonic distortion)

Using a 1 kHz test tone with the signal level at the overload threshold.

Output No.	Minimum Delay	Maximum Delay
1	.18%	.25%
2	.23%	.45%
3	.41%	.74%
4	.73%	1.04%

Output No. 4, maximum delay, 1 kHz test tone

Test Signal Level	+5 dB	+10 dB	+20 dB
Limiter Out	2.25%	3.7%	17.2%
Limiter In	1.25%	1.3%	1.4%

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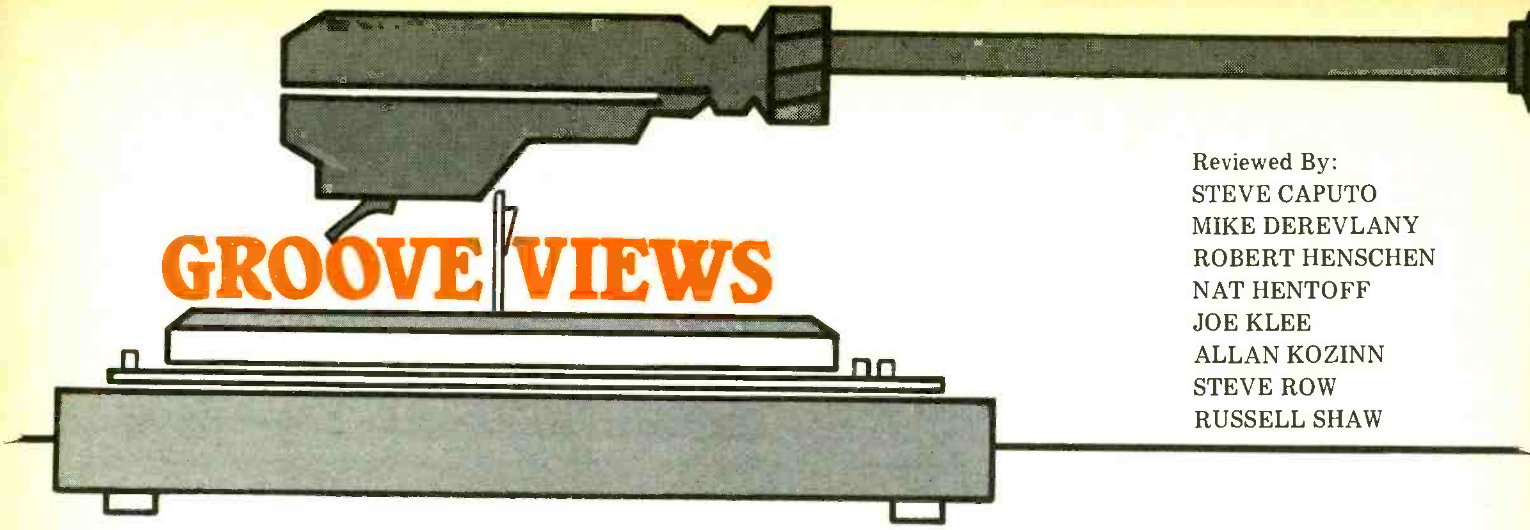

GROOVE VIEWS

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

POPULAR

THE WEREWOLVES: *Ship Of Fools (Summer Weekends And No More Blues).* [Andrew Loog Oldham, producer; Neal Teeman and Chris Andersen, engineers; recorded aboard "Xtort" by The Bearsville Land And Sea Sound Company.] RCA AF1-3079.

Performance: **Loosely tight**
Recording: **Unadorned accuracy**

Up until six months ago, this band only came out at nights—Texas nights filled with get-down, back-to-the-roots, loosely tight rock & roll. A successful western tour, and the production allegiance of Andrew Loog Oldham, gave The Werewolves a sudden burst of favorable exposure and an almost immediate 6-day recording session aboard the 62-foot, twin-engine vessel "Xtort," the band's second studio effort in six months. Oldham's first big group was capable of cranking out two or three rugged LPs per year too, ho-hum discs like *Out Of Our Heads* and *December's Children.* Ever hear of The Rolling Stones?

The spirit of raw, uncultured rock is integral to The Werewolves and *Ship Of Fools* is a rough mix of blues, boogie, country, rock and '50s-influenced piano banging. This band made their mark as a no-holds-barred "live" act in and around Dallas, so the barband vitality still infiltrates their wide-open, surprisingly organized style. Cuts like "Crazy Arms" and "Catch My Drift" are obviously geared for "live" delivery, but unpolished engineering retains much of the original excitement.

THE WEREWOLVES: A healthy transfusion for the American music scene

The Werewolves can definitely cook Southern style, but the big Phil Spector beat ("There We Were") and early British rock influences are often more pronounced. Lead singer Brian Papageorge sounds astonishingly like Mick Jagger on several tunes, particularly the very basic "Face On Wrong," and Andrew Loog Oldham does nothing to curtail other Stones parallels.

Lyrics are not a highpoint on *Ship Of Fools,* nor are technical gymnastics. But the album frequently draws incentive from the raunch-style tenor sax of Joey Stann, and the material is fairly diverse. "Days Of The Rest Of My Life" is a country ballad, "Waking Up Is Hard To Do" would be the skittishly perfect blues'n'boogie vehicle for Asleep At The Wheel, and the title track heaps upbeat horns atop happy, colorful Caribbean steel drums. Basically, however, The Werewolves play rock in the raw, a healthy transfusion for the American music scene. R.H.

IAN MATTHEWS: *Stealin' Home.* [Ian Matthews and Sandy Robertson, producers; Barry Hammond, engineer; Bryn Haworth and Ian Matthews, arrangers; recorded at Chipping Norton Studios, Oxfordshire, England.] Mushroom MRS 5012.

Performance: **Tasteful pop, good tenor**
Recording: **Slick**

For lo these many years, Ian Matthews has been looking for a groove, an identity, since he left first Fairport Convention and then Matthews Southern Comfort. His career has been summed up by Nick Logan and Bob Woffinden of the New Musical Express, in "The Il-

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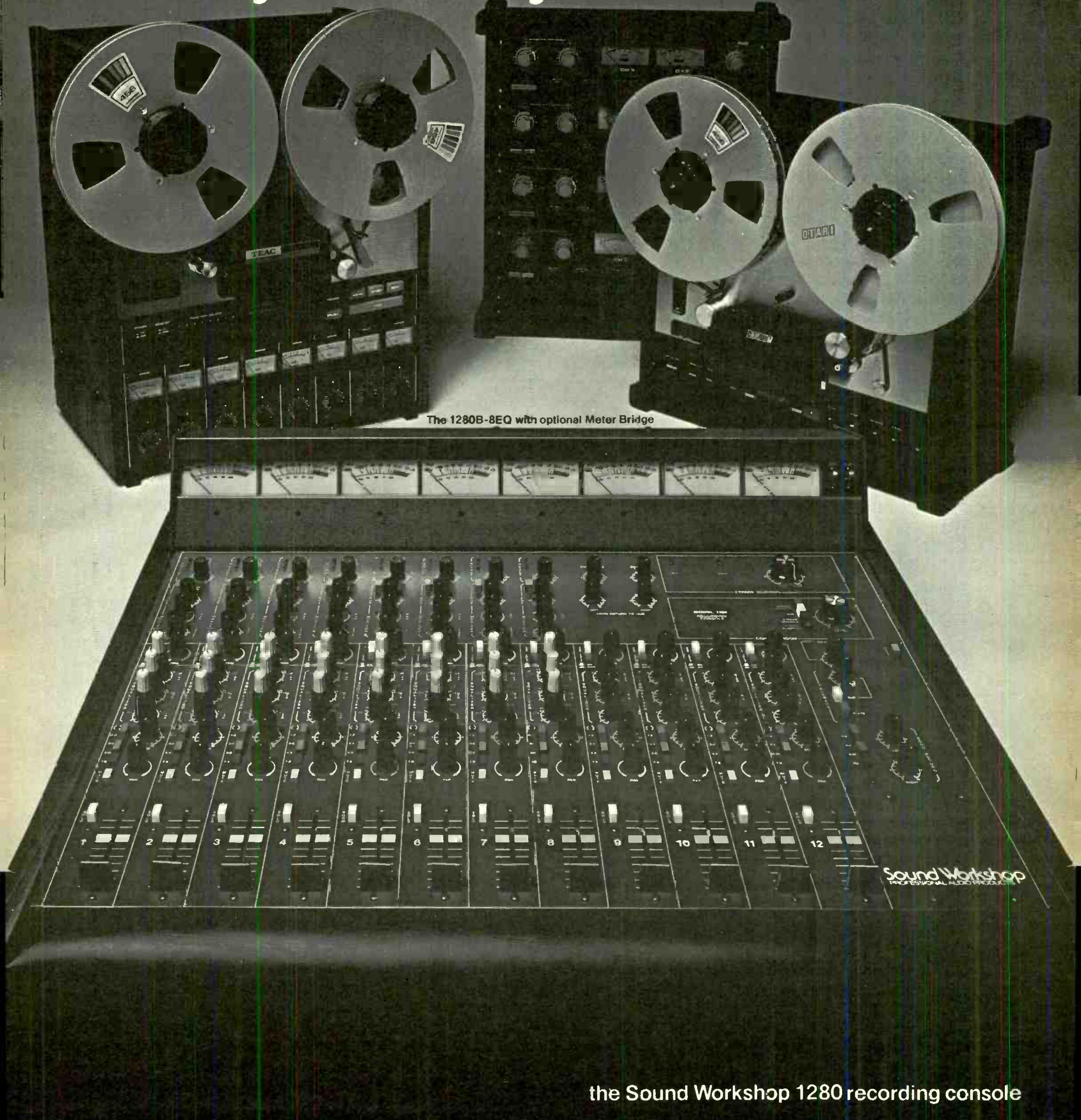

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lustrated Encyclopedia of Rock" in this way: "After being the man-most-likely-to for so long, he is now beginning to look like the man-who-never-did."

Despite what some say is a failure to fulfill a promise, Ian Matthews still has one of the finest tenor voices in all of pop music, and he invariably comes up with some of the most interesting selections of musical material of any solo artist going. His latest album, *Stealin' Home*, is no exception, and some (but, unfortunately, not all) of the album shows the same strengths as Gerry Rafferty's *City to City*.

Matthews' sources of material on this outing are Terence Boylan, John Martyn, Robert Palmer, Charlie Chaplin and Rodgers & Hammerstein, among others. He and some of his friends produce a powerful acappella reading of

IAN MATTHEWS: "E" for effort

"You've Got to Be Taught," from "The King and I," to open the "B" side, for example, and he has provided a stylish setting for Palmer's "Gimme an Inch" to open the "A" side of the album. He has found an interesting song by Richard Stehol entitled "Yank and Mary" that incorporates the first four lines of Chaplin's "Smile" as its chorus. His readings of Boylan's material ("Don't Hang Up Those Dancing Shoes" and "Shake It") are good but don't quite hide the similarities between Boylan's music and that of Jackson Browne. His cover of Martyn's "Man at the Station" has an instrumental break that sounds almost like the hook on the Rolling Stones' "Miss You."

His own material, however, tends to sound like country-rock tunes, perhaps showing that he is most comfortable in a setting designed by the Eagles or Poco. "Let There Be Blues," with nice mandolin work by Haworth, has single written all over it, and the title track is a fine new ballad that already is a single. Both are tinged with the essence of country-rock.

On both songs, Matthews shows that he not only can sing well, but he also can produce intelligent lyrics. His other two songs on the album are "Slip Away," and "Sail My Soul," both written with Bill Lamb. The former is a rather enigmatic piece, up-tempo, with a good chorus and good guitar break near the close. It comes as close as any song of the album to being a bona-fide rocker. The latter has a long instrumental introduction by the guitar and bass, and the keyboard playing is prominently displayed. More good lyrics here, too.

"King of the Night" by Jeffrey Comanor and "Shake It" are both songs done in Southern California tradition, although the good-natured rocking style of the latter contrasts with lyrics of the former, a poignant reminder of "Rusted out memories, bitter and proud" one who was "fearless and foolish/Forgotten somehow." (Copyright © Chicken Key Music, BMI).

Those who don't know Ian Matthews will find here some things that remind them of two more popular solo singers of this day—Jesse Colin Young and Gerry Rafferty. Young has pretty much maintained a comfortable popularity for several years, while Rafferty's musical talent is just now being rediscovered by a large audience who passed him over in the Stealer's Wheel days. Matthews still needs to be discovered, but I'm not sure "Stealin' Home" is the album that will bring about that discovery.

The voice is there, the material is interesting, the arrangement crisp and bright. He certainly gets an "E" for effort, even if he doesn't get an $ for sales. This is an album that should not be overlooked, for the plusses outweigh the minuses, and the version of "Carefully Taught" is a real gem. S.R.

JOAN ARMATRADING: ***To The Limit.*** [Glyn Johns, producer and engineer; recorded at Olympic Studios, London, England.] A&M SP-4732.

Performance: **For real**
Recording: **Earthy and honest**

JOAN ARMATRADING: Oh, so real

This remarkable woman is capable of turning a simple ballad into a personal triumph, imbuing each word with the kind of raw and propulsive emotionalism long forgotten by others. Armatrading's West Indies-to-London background provides her with a unique combination of gospel reggae/blues/roots plus British lyrical aplomb, and Joan molds it all into her own brand of folk-rock fusion. After fantastic albums like *Back To The Night*, *Joan Armatrading*, and *Show Some Emotion*, I'm ready to call Joan Armatrading one of the two or three most important singer-songwriters of the late seventies. That's the kind of reality she brings to contemporary music.

To The Limit attains that same basic spirit of liberated blues, and tunes like "Wishing," "Your Letter," and "You Rope You Tie Me" can certainly stand alongside Armatrading's best work. The album title is drawn from "Bottom To The Top," an infectious reggae that is ideal for Joan's funky sound, and "Let It Last" is the sort of gospel-blues lament that she is becoming famous for—you can only get this international woman ("Barefoot And Pregnant") down so low, then she's just *got* to get back up.

Glyn Johns has done a fine job of capturing the earthiness, honesty, and gentleness of *To The Limit.* Joan's voice, of course, must be the major instrument at all times, so over-production or cluttered instrumental arrangements would only damage the authentic feel of this music. On "What Do You Want," Armatrading's vocal approaches a soulful percussive quality, while she sustains longer, bluer notes on "Wishing." Philip Palmer (electric guitar) and Dick Simms (organ and accordion) are each given solos but Johns makes sure they don't intrude on the leader's intimacy. The engineering is supportive and warm.

Like other Joan Armatrading albums, *To The Limit* grows on the listener with repeated listenings. Apparent compositional lightness on "Baby I" and "Taking My Baby Up Town" are almost necessary respites between emotion-charged feature songs. This is twentieth-century inspirational music that reacts to this decade's blase pop music with a hot beat and gut-level humanism. Joan Armatrading means it when she says nothing is half as good as "touching me in real." On records like this, she offers herself body and soul.

R.H.

LOL CREME AND KEVIN GODLEY: ***L.*** [Lol Creme and Kevin Godley, producers; Nigel Gray and Chris Gray, engineers; recorded at Surrey Sound Studios, Leatherland, Surrey, England, March through June 1978.] Mercury SRM-1-3752.

Performance: **Nebulous**
Recording: **Top notch**

Who are Lol and Kev? And what is a Gizmo? And furthermore, what do they

LOL CREME AND KEVIN GODLEY: A dauntless duo, and Gizmo makes three

mean to each other?

Lol and Kev are, of course, Lol Creme and Kevin Godley, two former members of the original 10cc, and the Gizmo, now officially known as a Gizmotron, is their invention. After their departure from 10cc, Creme and Godley spent a great deal of time experimenting and recording with the Gizmotron, a small electro-mechanical bowing device for the guitar which allows the guitarist to duplicate the sound of almost any string instrument. The product of all this experimentation was a monstrously flawed three-record set that, with a list price of $18, was bound to be an irremediable failure if only because of economic reasons.

Apparently all that studio time was not a total waste for the dauntless duo; if anything, they learned the practical limits of the Gizmotron. Time and device aside, they didn't, however, develop a distinct musical identity but became a sort of nebulous reflection of 10cc, without the 10cc creativity or innovation; they are a redundancy with limited redeeming value and no true originality.

Much of the music on *L* is conceptual in style; individual pieces are around five or six minutes long or longer and are thematic though undeveloped. Most of the material gives the impression that Creme and Godley don't really know what they want to do, which is a shame since they not only have a great deal of talent but also an overwhelming amount of technical ability and knowledge that could result in an absolutely spectacular product.

Although these particular Creme and Godley compositions are basically aimless, they are all extravagantly produced and have some of the most thoroughly poignant and humorous lyrics of the year. These lyrics help

keep some of the cuts lighthearted even in their Spector-esque ponderousness. Creme and Godley undoubtedly had a good time making this record, since they got a lot off their chests, and there is good deal of fun in almost every song. Cuts like "Hit Factory" and "Business is Business" suggest that there is something that they distinctly dislike about contemporary music and its marketing.

L is a mixture of quality and it varies from nearly brilliant to downright mundane. Sandwiched between the witticisms of "Business is Business" ("M.O.R. is safe/M.O.R. is here . . . . And only the numb survive") and "The Sporting Life" ("Don't be hasty,/Why waste a life/Wait until there's a crowd down below") are cuts like the instrumental "Foreign Accents" that are just too contrived. "Foreign Accents" has a lot of electronically created and electronically altered sounds, some of which don't really fall together all that well. Cuts like this one show potential, but they also demonstrate a possible inability to live up to that potential. Most of the cuts, though, are rendered enjoyable by the lavish production; the effects are pleasing, even if the material isn't particularly so. All it means is that *L* proves that Creme and Godley are extremely adept at creating a lot of deceptive ear bending.

M.D.

CANO: ***Au Nord De Notre Vie.*** [Don Oriolo, producer; Ed Stasium, engineer; recorded at Phase One Studios, Toronto, Canada, June 20-27, 1977.] A&M SP 9028.

CANO: ***Eclipse.*** [Eugene Martynec, producer; Ken Friesen, engineer; Michael Jackson, assistant engineer; recorded at Eastern Sound Studios, Toronto, Canada, July-August, 1978.] A&M SP 9033.

Performances: **Fine but idiosyncratic**
Recordings: **Technically faultless, aesthetically aloof**

Cano is not your typical obscure musical group. For one thing, they are French-Canadian (undoubtably a major cause of their obscurity in this country) and they are on a major label. And because they're French—yep, you guessed it, most of their lyrics are in French and involve French traditions and lore. The result is occasionally quaint but more often a bit disconcerting, particularly to someone more accustomed to standard FM radio fare. Not that this makes a great deal of difference since it's usually impossible to understand the lyrics unless seen first in printed form, which Cano thoughtfully provides in both French and English.

Even with this unusual background Cano, while rising far above the drab, falls far short of the originality one might expect. Prefabricated might be a more apt description. In many ways they resemble Renaissance in their more mediocre incarnations. There is vocalist Rachel Paiement, who is clearly an Annie Haslam sound-alike (no, I take that back; she sounds better than Haslam). One of the cuts in which she demonstrates her vocal prowess is a ballad, of the same type that Renaissance does. Unlike Renaissance, however, Cano doesn't create a semi-

CANO: Out of Canada with remarkable vocals and technically flawless recording

symphony by filling it in with all sorts of exotic instrumental arrangements. The emphasis is on Paiement's voice, if only because the rather plain backing she is provided with couldn't stand on its own if it used crutches.

Cano's second album does have some songs with English lyrics, which makes sense if the group desires to achieve any sort of reasonable commercial success in the U.S. as well as in Canada (French-speaking Canadians only comprise about one-quarter of the entire Canadian population). You don't think Neil Young would have gone very far whining sad tales in French, do you? Or could you imagine Randy Bachman screaming "Tu ne vis pas encore"?

The production of both albums is fair; it doesn't have any sort of technical faults but for the most part doesn't particularly add to the material or help it come across more dynamically. An unusual, and somewhat annoying, feature of both albums is that they have the largest amount of dead air space between tracks of any album I have ever heard. Another note of interest is that Ed Stasium, who engineered the first album, also co-engineers the Ramones' albums. Stasium successfully captures the raucous energy of the Ramones; with Cano he is more limited, to a certain extent, by the excessive complexities of Cano's material.

Cano has a sound that ultimately proves itself uninspired despite its attempt at originality. And that's in two albums. Consistency like that should easily sustain their current obscurity.

M.D.

JULES AND THE POLAR BEARS: Flawless "Jules" forfeiting defiant distortion

JULES AND THE POLAR BEARS: *Got No Breeding.* [Larry Hirsch, Stephen Hague, Jules Shear, producers; Larry Hirsch, Stephen Hague, engineers; recorded at Paramount Recording Studios.] Columbia Records JC 35601.

Performance: **Impressive**
Recording: **Polished**

With the passing of punk and the cresting of new wave music, it is interesting to speculate on the next step in the progression centered around punk's original attempt to answer to disco and sugar rock. While that guess is not one to be ventured in this column, I do believe that Jules and the Polar Bears may hold some clues. *Got No Breeding,* though flawed, is as solid a debut album as one might expect from a group whose brand of music might be outdated tomorrow.

Jules and his arctic friends perform the kind of music that would easily appeal to fans of Elvis Costello and the Talking Heads. That is to say their sound is new wave. But, to their credit, *Got No Breeding,* (recorded at Paramount Recording Studios), has an even cleaner sound to it than albums by the others. This may very well open the music up to more listeners who are willing to give it a chance. The first cut, "You Just Don't Wanna Know," best tells the story of the rest of the album. All of the elements are there – pulsating drums, rhythm and lead guitars, jagged lyrics sung by the grating voice of Jules Shear, even keyboards. The success of the song's catchy hooks is

CHICAGO: Still mining the tried and true gold pile

owed primarily to a crisp recording that leads you right to the riffs, then back to the chorus. Next come the horns and background vocals a la Bruce Springsteen, or the Asbury Jukes. It gets better. The horns resurface in the title cut, on side two. A touch of synthesizer, and some more clean harmonies highlight "Shadow Break." My favorite cut is "Follow Every Finger," a pleasant tune that displays the group's ability to comfortably and competently diversify its performance. There's a nice, clean guitar riff or two in "Follow Every Finger," also.

Flawless 'Jules' are hard to come by, so when "Convict" suffers from a lack of musical conviction, or "Driftwood from Disaster" gets tangled in the seaweed of a relatively poor mix, we can't come down too hard.

I was pleasantly surprised by Jules and the Polar Bears. I found the music rather infectious, despite Mr. Shear's unique vocal style. I truly didn't expect the recording to be as refined as it is. Horns, harmonies and synthesizers are to be the transforming agents of the new wave, as melody transformed punk to new wave. What is happening is a compromise. In exchange for a broader cross-section of listeners, Jules and the Polar Bears have forfeited a bit of the defiant distortion so notoriously associated with their predecessors. They have even thrown in horns to give the jagged lyrics a different sort of brassiness. Perhaps next they'll be calling it fusion wave rock. S.C.

CHICAGO: ***Hot Streets.*** [Phil Ramone and Chicago, producers; Jim Boyer, engineer; Don Gebman, Lee DeCarlo, Dave Martone, Kevin Ryan, and Peter Lewis, assistant engineers; recorded at Criteria Studios, Miami, Fl.] Columbia FC 35512.

Performance: **Competent, relaxed**
Recording: **Bright and bold**

It's been nearly ten years since the release of *Chicago Transit Authority*, and the biggest question raised by that fact is whether any musical gold remains to be mined out of the jazz-rock style the group popularized in the past decade.

Hot Streets, the first Chicago album with a title, shows that the group can still come up with an infectious, engaging single that is stamped all over with the now-classic sound. "Alive Again" is bright, with some bold brass charts, hot guitar licks during the breaks and strong bass work. The vocal harmonies are among the group's best, and the instrumental accompaniment is energetic and entertaining.

That's one song; now, what about the other nine?

Without sacrificing the basic Chicago sound, *Hot Streets* represents today's urban jazz-rock sound very well. One can find a few disco shadings ("No Tell Lover") and a few more R&B devices than would be found five or ten years ago on a Chicago album, but the end result pretty much remains faithful to what has become the group's sound. Danny Seraphine's drumming continues to be among the best and most varied drum/percussion work in pop music today. The brass charts by James Pankow provide both musical and rhythmical underpinnings for the vocal line. Some good guitar licks can be heard, too, from Chicago's newest member, Donnie Dacus, who replaced the late Terry Kath.

The material is good but not great, however. In fact, most of the songs tend to pale by comparison with "Alive Again." "Greatest Love on Earth" and "Take a Chance" are both relatively low-keyed, bluesy tunes that have rather trite lyrics; "Gone Long Gone" features a George Harrison-like guitar introduction and a little country-and-western flavor in the vocal charts; "Show Me the Way" (not to be confused with Peter Frampton's song) has a brass arrangement that comes across almost like an oompah band, and a chorus arrangement that could have been written by George Martin or perhaps Roy Thomas Baker.

The Bee Gees show up on one of the cuts, "Little Miss Lovin'," and their presence doesn't detract that much from what is a bright number headed up by the instrumental work in the trombone and guitar. But the lyrics sound better suited to the Bee Gees than to Chicago.

There are better songs—the title cut has a nice shuffling rhythm in the drums and percussion, quick bass work and

JOHN HAMMOND: Energetic package

other aspects of the identifiable Chicago sound. "Ain't It Time" features a gutsy vocal by Dacus, using a shout and response in the chorus, and some nice guitar work. "Love Was Now" mixes the vocal sound of Steely Dan with the instrumental accompaniment of Chicago in a nicely blended, softly Latin-tempoed rhythm. The brass choir in back of the song is a nice touch.

Many members of the group contributed music to the album—Pankow, Seraphine, Peter Cetera, Robert Lamm, Lee Loughnane, Dacus—but the majority of the music still comes across as if it were the creation of a single mind. That musical mind has been cranking out material for a decade now and shows little sign of wearing down. So long as there are some real nuggets left to mine, such as the two or three on *Hot Streets*, then it will be worth waiting for Chicago yet another time. S.R.

JOHN HAMMOND: *Footwork.* [No producer listed; Charlie Repka, engineer; no studio listed.] Vanguard VSD 79400.

Performance: **Masterful**
Recording: **Functionally brilliant**

It is considerably less than amusing that John Hammond has not been given the overall credit that many of his inferiors have. Perhaps that is due to the paleness of his skin. Yet breathes there a living, acoustic blues artist with a more perceptive grasp not only of the technical requirements of the idiom, but the psychic subtleties of it as well? I think not.

In reality a content individual, Hammond does not project a personna of an emotionally active volcano. No, he doesn't scream, or coax wild, flattened fifth distortions out of his frets yet the feeling is there. On stage, however, he is a package of vibrant, seething energy, and it is therefore a supreme test to capture such a feeling in a four-walled studio as well as in a coffeehouse or concert hall packed with catharsis-seeking souls.

Tunes by Robert Johnson, jazz great Mose Allison, and legendary Georgia blues artist Blind Willie McTell highlight this latest offering. To the credit of the unnamed producer, lavish technical embellishments have been left out in favor of Hammond and guitar, and frequently dobro as well. We hear a persuasive reading of "Preaching Blues," a mean-as-an-alleycat harp guitar duet on "Who Do You Love," and a rocking adaptation of Walter Jacobs' "Go No Further."

Yet the palette is not John's alone, for celebrated New Orleans-based blues pianist Roosevelt Sykes collaborates on two tunes. "44 Rifle Blues," a song about bumping off your old lady because of her two-timin', is a Sykes composition. A thrilling work, it is, however, marred by frequent clutter of chordal roles between the two participants. Rather than a definite lead-rhythm script, both Roosevelt and John alternate between chordal maintenance and mean lead work. Their frequent lack of clarity is the album's only weak point.

R.S.

JAZZ

CHARLIE BYRD: ***Midnight Guitar.*** [Original session produced by Ozzie Cadena; reissue produced by Bob Porter; engineered by Rudy Van Gelder; recorded in Hackensack, N.J., August 4, 1957; released originally on Savoy.] Arista/Savoy 1121.

Performance: **Sublime**
Recording: **Surprisingly bright and clear**

KENNY BURRELL, WITH FRANK WESS: ***Monday Stroll.*** [Original session produced by Ozzie Cadena; reissue produced by Bob Porter; original recording and mastering by Rudy Van Gelder; recorded in Hackensack, N.J., Dec. 17, 1956, and Jan. 5, 1957; released originally on Savoy.] Arista/Savoy 1120.

Peformance: **Very good**
Recording: **A little fuzzy around the edges**

These two recent Arista/Savoy re-releases are good examples of two directions that have been taken by jazz guitarists over the past two decades. And after two decades, both recordings stand up as excellent examples of those two distinct styles.

Kenny Burrell's style was closer to both Django Rheinhardt and Charlie Christian, if for no other reason than his preference for the amplified guitar. Burrell also chose, on this record at least, to become as much a part of the ensemble sound (and even backup accompaniment) as to be the instrumental leader of the group. The first-time listener to this album may overlook much of Burrell's work because it

KENNY BURRELL: Relaxing

seems to be secondary to Frank Wess's fine flute work.

Byrd chose to keep the guitar right out front in his playing, and he also chose to stick to the unamplified (or classical, or folk) guitar. Some of his jazz pieces sound very much as if they were influenced by his training with Andres Segovia; his approach to the instrument and to the music is one of serious, personal intimacy, without being stodgy, however.

These two works bring to about a dozen the number of Savoy records from the 1950s that Arista has dusted off and repressed, and they are two of the best. Burrell and Wess, along with rhythm guitarist Freddie Green, bassist Eddie Jones and drummer Kenny Clark (or Gus Johnson on two of the seven cuts), produced a tight, small

The Italian Adventurer and The Graceful Bopper

By Nat Hentoff

It used to be said of the jazz on ECM that while it was often ingenious, it cast a cold, calculated light. The technique might be prodigious, but there was little, if any, of that earthiness which is the essence of jazz soul. But ECM is changing some, as is evidenced by the *Enrico Rava Quartet.* Rava, born in Trieste, started as a Dixieland trombonist, switched to trumpet after having been entranced by a Miles Davis performance, and became a renowned post-modernist in Italy and elsewhere in Europe. Currently, he alternates geographically between Italy, New York and Buenos Aires.

Rava is a fervently lyrical player, very much a part of the jazz tradition of "singing horns." His sound is vibrant, clear, often exultant, and he continually takes risks in time and in phrasing. His chief colleague in this quartet is Roswell Rudd, arguably the most personal and emotionally compelling of all so-called avant-garde trombonists. Like Rava, Rudd began as a Dixielander and has absorbed the total jazz tradition so that in the course of a solo, you can hear all sorts of unexpected echoes of the jazz past.

Though the forms they use are open and instantly flexible, there is a continual coherence in the way Rava and Rudd play off each other; and because they are essentially emotive players, their music is immediately accessible. Emphatic support comes from drummer Aldo Romano and the widely experienced bassist, J.F. Jenny-Clark. As is customary on ECM, the sound is spacious and vibrant—just right for these boldly soulful horns.

By contrast with Rava and Rudd, alto saxophonist Charles McPherson has no reputation as an experimenter. He is a largely unreconstructed bopper, his approach having been firmly formed by Charlie Parker. And yet, during those years when bop was considered to be as archaic as Dixieland by some younger players, McPherson continued honing and deepening his style. One particularly instructive influence was Charles Mingus in whose Jazz Workshop McPherson spent considerable time. With Mingus, he learned how to find and become more of himself so that while the spirit of Bird remained, the actual speaking voice on the horn emerged unmistakeably as that of McPherson.

Now, in *New Horizons* (Xanadu), McPherson is the very model of a mature bopper. Gracefully self-assured at all tempos, he remains a resilient swinger while so getting inside ballads that he has few peers at the demanding art of jazz romanticism. Simultaneously, he has so mastered dynamics that, unlike some of the boppers of yore, he can extract nuances from nuances.

The members of the rhythm section—pianist Mickey Tucker, bassist Cecil McBee, drummer Freddie Waits—are also at thorough, pleasurable ease in this idiom. Bop, like all other jazz genres, will never be outdated so long as there are musicians who, having been nourished by it, bring back life to the music. The recorded sound is clean, crisp, and vibrant.

ENRICO RAVA: *Enrico Rava Quartet.* [Manfred Eicher, producer; Martin Wieland, engineer.] ECM-1-1122.

CHARLES McPHERSON: *New Horizons.* [Don Schlitten, producer; Paul Goodman, engineer.] Xanadu 149.

ensemble jazz sound. The players all were used to the larger big band jazz sound, having been in Count Basie's band and since the time of the recording continued or are continuing in large ensemble jazz in workshops, records, concerts and television (Wess, for example, has been a member of the "Saturday Night Live" band).

The album contains a lot of unison playing between Wess and Burrell, solos by all the players and some interesting interplay among the players. Burrell plays rhythm accompaniment, lead chords and lead melodies at various times during the record. On "East Wind," for example, one barely notices Burrell's guitar behind Wess's flute lead, except for its rhythmic underpinning, until he moves into a picked solo midway through the song. In "Southern Exposure" he has a hot solo that features audible grunts not unlike those that are on the best Lionel Hampton records.

One senses a spirit of relaxation and fun on the album, particularly in the duets between Wess and Burrell. Wess's tenor sax in "Woolafunt's Lament" is matched quite nicely by Burrell's guitar, which both recalls Charlie Christian and forecasts Wes Montgomery. Listen for some extremely fast fingering in long and complicated runs here. They are not flawless, but what is lacking in perfection is compensated for in spontaneity.

One track will be familiar immediately to listeners, and the combo's version of "Over the Rainbow" opens with a statement in strummed chords and picking, before going through two key changes in the guitar and then a third to get to Wess's flute for one of the choruses. The whole sound is essentially very mellow.

The accompaniment by the other players is polished, but the strength of the album is the work together by Burrell and Wess. Byrd also has Gus Johnson on drums behind him and long time associate Keter Betts (spelled erroneously, I believe, as Keeter on this album) on bass, and the seven cuts on this album are showcases for Byrd's splendid guitar playing.

One identifiable jazz cut here is "Four O'Clock Funk," which Byrd now plays in concert and on record as "Something Like the Blues." A standard included here is "This Can't Be Love."

The first side is taken up for a suite of three songs assembled under the name "Blues for Night People," and for the most part the playing is low-key and, well, blusey. "First Show" features opening phrases that could be borrowed, adapted and transformed into quite a hot electric guitar hook by some enterprising rock group, while the body recalls some of Byrd's later explorations of 20th century Spanish music. He shifts in an out of chord playing (from chords to picking and back to chords again) with ease, and toward the end of this particular cut, he sounds almost like two separate players.

A classically-inspired introduction opens "2:00 A.M.," while "Four O'Clock Funk" in the two-decade-old version is much more mellow than his uptempo "Something Like the Blues" is now.

Delicacy is the hallmark of Byrd's playing throughout the album, but particularly on "Blues My Naughty Sweetie Gave to Me" and "Blue Prelude" he employs delicacy and feeling without becoming too trite. And the contrast with the bluesy side of the two songs is very good, too.

Byrd, like Burrell, sounds relaxed here, and his backup talent is just right. Betts, in particular, provides an interesting counterpart in the bass to Byrd's guitar playing, and Johnson's drumming keeps the proceedings moving where movement is dictated, bright where brightness is needed and shimmering where a hard sheen is needed.

Both albums, by the way, contain excellent program and historical notes—Herb Wong on the Burrell record and Bill Bennett of the Washington Star on the Byrd album. The sound of the albums generally is quite good, even excellent, with the edge given to the Byrd recording, a much cleaner, brighter, less cluttered sounding record than that by Burrell & Co. Burrell's album sounds something like an old jazz album, but Byrd's sounds fresh and new. S.R.

PHIL WOODS: ***Song For Sisyphus.*** [Norman Schwartz, producer; Keith Grant and Dick Baxter, engineers; recorded at RCA Studios, New York, N.Y., Nov. 9, 1977.] Gryphon G 782.

Performance: **Joy Philled**
Recording: **No complaints**

What's difficult about reviewing Phil Woods' records (one after the other) is that Phil—and his settings—change so little that the problem is finding new things to write. It seems like just about a year ago that I was reviewing a two-record package on RCA of this same Quintet recorded "live" at a club. The band hasn't changed. Phil hasn't changed. Even the repertoire is similar. The RCA album included a version of Django Reinhardt's "Manoir De Mes Reves;" this one has his "Nuage" performed as a solo vehicle by guitarist Harry Leahey. The RCA album had Phil Woods playing Irving Berlin's "Cheek to Cheek" which Fred Astaire introduced in the film, "Top Hat" in 1935; the Gryphon LP has Berlin's "Change Partners" which Astaire sang in the 1938 film "Carefree." There are other gems such as Harold Arlen's "Last Night When We Were Young" and Phil's pleasing original "Song For Sisyphus," but basically it's simply another LP of Phil Woods doing what he does best.

And why not? It's not a putdown to say that Phil Woods is still the same sparkling, scintillating player that he was last year at this time. If anything, it's a mark of consistency. This need, among some of the less secure jazz musicians, to feel that unless you're playing differently than you did last

year you're resting on past laurels often covers a cop-out mentality which, in reality, boils down to trying to get on board whatever is *in* this week. Sure a player wants to refine his art, become more adept at whatever he does, but there's no need to go different routes just to prove his versatility and cash in on all the fads. Phil Woods doesn't change—because he doesn't need to.

Phil Woods came out of Charlie Parker. A lot of alto saxophonists came out of Charlie Parker. There were those who blew Bird ver-notem, even the mistakes on the bad takes Parker never wanted issued but they came out anyway. There were those who played their own thing and yet clearly showed the influence Bird had exerted on their own thing. Few managed successfully a combination of an extension of Bird's style combined with their own works. Phil Woods certainly was one of the successful ones. You can hear this demonstrably in what Phil plays on Parker's classic "Shaw 'Nuff."

The recording is clear, crisp, quiet, everything one could want. The band is cooking and Phil is *on* (even Parker wasn't always on) and that makes for one fine recording. If you miss it—you have only yourself to blame. J.K.

HERBIE MANN: ***Sunbelt.*** [Herbie Mann, producer; Jimmy Douglas and Lew Hahn, engineers; Cengiz Yaltkaya, assistant engineer; recorded at the Atlantic Studios, New York, N.Y.] Atlantic SD 19204.

Performance: **Some fire, some ice**
Recording: **Excellent**

Herbie Mann's *Sunbelt* straddles the Latin-flavored music of his earlier years and the funky rhythms of his more recent years, and the result is a good, late 1970s jazz flute recording. In some respects, Mann's latest effort seems aimed at capturing some of the disco and R&B bucks, but below the surface and past the first impression the album more closely resembles what Mann has been doing all along.

A female chorus (called the "Girls of Bahia") that adds a little embellishment to the ends of phrases, a few strings thrown in here and there and some brash brass and reed work are all employed in the arrangements. Mann is not in the spotlight all the time during each cut; instead, he often waits until several bars of the song have passed before joining. The mix is just about right for each cut, too, with a light percussion prominent but never intruding, and the horns are quite bold but they are never overpowering.

"Watermelon Man" in the 1978 version is hotter and funkier than Mann's earlier versions, and Mann's flute is quite throaty and husky sounding. He blows a good solo during the song and is backed well by the horns. Before the cut is over, Claudio Roditi has a brief but good trumpet solo. Horns and strings carry the lead in "The Closer I Get to You," and this is one track where Mann doesn't join in until the third statement of the theme. The song sounds like something out of the Columbia/Tappan Zee jazz factory, by the way.

HERBIE MANN: Good, but not great

The title cut, which ends the album, and "Let's Stay Together" (the Al Green/Willie Mitchell/Al Jackson song) show Mann at his most mellow. The title cut, which is joined by the familiar "Weaver Woman" melody, begins as a sunrise, with the shimmering theme played on the alto flute and guitar. The tempo quickens, and soon the ensemble joins, with a muted trumpet by Roditi taking the lead. Amaury Tristao's guitar never lets up.

This album probably is not as completely satisfying as Mann's previous release, *Brazil—Once Again.* But there is something to like about this release, just as there is something to like about virtually everything Mann has done. *Sunbelt* shows off Mann's skills as a musician and arranger. He has surrounded himself with some talented folks, too—along with Tee, Tristao and Roditi, there are Roy Ayers on vibes, long-time associate Pat Rebillot on piano and electric keyboards, Steve Gadd on drums, Rafael Cruz on percussion and a host of others. Mann uses an echo device for two of the cuts, but the effect is so brief that it is hardly noticed.

Overall, this is good, if not great, jazz. S.R.

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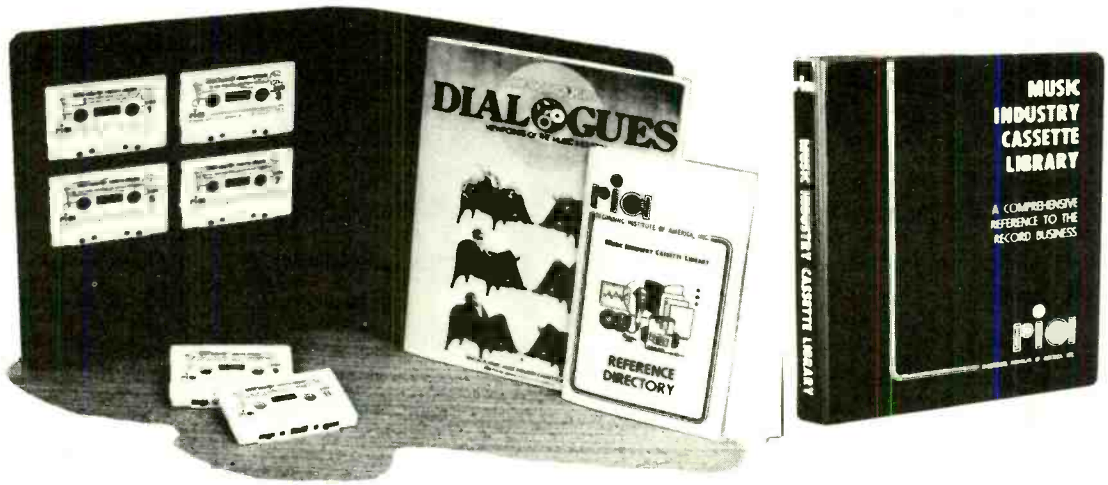

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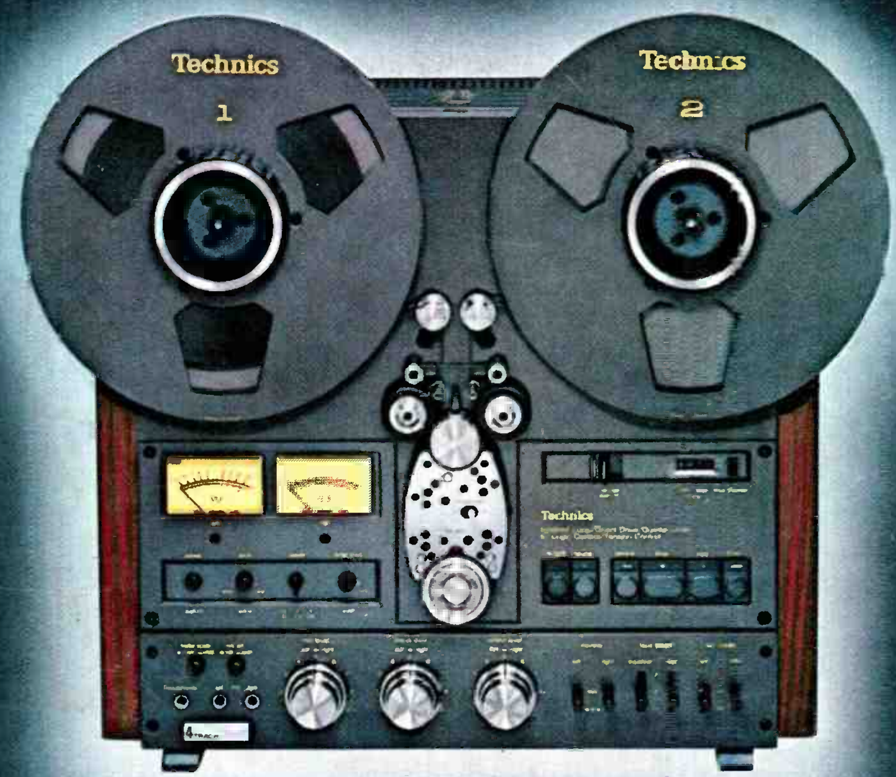

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