ECHO, REVERB & DELAY —PART II

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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VOL. 3 NO. 9
JUNE 1978

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Complete the official entry form at a participating TAPCO dealer and put it in the mail so it is postmarked no later than August 15, 1978. Winners will be selected in a random drawing September 15, 1978 by persons not employees of AKG, Anvil, Electro-Voice, or TAPCO. One entry per name. Odds of winning are determined by the number of entries received. You must be 15 years of age or older to enter. No purchase required. The results of the drawing will be final, and the winners notified by mail. If a winner has TAPCO products duplicated by winning a system, the winner will receive his or her choice of AKG, Anvil, E-V, or TAPCO products of equal value. State, Federal, and other taxes imposed on each prize winner will be the sole responsibility of that prize winner. Requests for winning names should be addressed to TAPCO, 3810 148th Ave. NE, Redmond, WA 98052. Employees of AKG, Anvil, Electro-Voice and TAPCO, affiliated companies, sales agents, and their families not eligible. Void where prohibited or restricted by law.

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CIRCLE 95 ON READER SERVICE CARD

MODERN RECORDING

JUNE 1978
VOL. 3 NO. 9

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
Eric Clapton "Live!"
Kris Kristofferson & Rita Coolidge
The Making of a Record—Part II

THE STAPLES

Cover photos by Herb Kossover

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MODERN RECORDING
SERVING TODAY'S MUSIC/RECORDING CONSCIOUS SOCIETY

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Letters to the Editor

Going Direct

I have just finished reading the Peter Weiss article on direct box construction ("Building a Direct Box," April 1978, pages 48-52) and found it to be very interesting. I have been associated with a Long Island-based direct box manufacturer and, based upon what I learned from this experience, I would like to offer some additional advice on direct box construction.

A direct box subjected to daily studio use should be made of die-cast aluminum to withstand nosedives off bass amps and Fender Rhodes pianos. In addition, the input connectors to the direct box transformers should be floating, (i.e. isolating washers on female tip sleeve connectors) above the chassis of the box. The chassis itself should be on ground potential (console shield or system ground) to provide additional hum shielding. All internal wiring should be twisted, except for hi-Z cable, and kept as close as possible to the inside of the box. I do not suggest the use of a cable run from the inside of the direct box to the instrument. Although convenient, it rarely holds up to the studio abuse that it receives. Besides recommending the use of high-quality parts and proper wiring, there is little more I can add to what has already been written on construction.

Moving to the slightly more technical aspect of direct boxes, I question your method of ground switching. In the Unit One diagram (page 50), you show the low side of the primary and the transformer's shield going to the ground switch, leaving the switch and being connected to pin two (C-2) of the XLR connector. Pin two in most balanced systems is audio low. The switching action should take the transformer's primary low side and shield to the console ground (generally pin one of the XLR). I suspect this was a misprint in the article.

Another point is the use of an external speaker jack as an input to the direct box. Although this technique produces a very nice "miked" sound, I would not attempt this practice without the use of a pad on the input to the direct box, seeing how the voltage developed across the speaker could burn out the primary of the transformer.

Last, but not least, keeping proper phase in all the boxes is very important, especially in using two boxes in dual output instruments, such as Fender Rhodes. A phase flip, the two signals subsequently cancelling each other out, could leave you wondering where the Rhodes went.

Having said my piece, I will conclude by saying you have a great publication and I hope it continues as long as recording does. Remember—"Go Direct!"

—Bruce Maddocks
Recording Technician
A&R Recording Studios
New York, N.Y.

[We forwarded Mr. Maddocks' letter to author Peter Weiss. Herewith is his response.]

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No other mixer delivers so many features for so little money . . .

Pan pots
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2 stereo RIAA phono inputs with EQ
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I will try to respond to the points you have raised as directly as possible, but I can't guarantee that they will necessarily be in the same order.

In both Units I and II, the aluminum mini-box is at the same potential as the "ground" side of the guitar amp chassis. This is because of the mechanical-electrical connection between the phone jack and the box itself, and the electrical connection between the input cable shield and the "ground" side of the phone jack. If it is required to have all of these at *console* ground potential, the grounding switch should be closed.

Let's next consider the question of the configuration of the output leads and pin numbers. There are six possible combinations of pin numbers and leads and many equipment users have modified original equipment wiring to conform to their own "standard." The wiring scheme shown in the article is compatible with many professional balanced systems using XLP ("large cannon") connectors. Most systems using XLR ("small cannon") type connectors, and most original equipment wiring schemes employ the method you've outlined (i.e., 1-ground, 2-Audio low, 3-Audio high). I agree that this point could have been made more clearly in the article.

As for durability, several versions of Units I and II have been in regular use (and abuse) in a large studio complex for at least seven years, cables and all. Also, I'd much rather be faced with a dent in a $5.00 mini-box than with a similar blemish in a more expensive die-cast enclosure.

Finally, although the power-delivering capability of an instrument amplifier is formidable, the maximum level developed at the direct box (Unit II)—assuming a 200 watt output signal to the main speaker is approximately +10 dBm—is well within the margin of safety for the transformer.

Thanks for your letter and especially the additional wiring "how-tos." Future construction articles are planned for publication in *Modern Recording* and an active correspondence with the readership is most welcome.

—Peter Weiss
Contributing Editor
Modern Recording Magazine

What to Charge

This may be a very stupid question, but how do you compute the hourly rate you would charge for an on-the-scene recording with a system consisting of a

The "better than" equalizer

crown EQ-2
octave equalizer,
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CIRCLE 93 ON READER SERVICE CARD

Tascam 80-8, a TEAC 3340-S, a TEAC A-7300, a dbx, a Tascam Model 5 board, etc.?

—John Castaldo
Eugene, Or.

The amount of money that you can charge for a session depends upon many variables–the number of artists, the complexity of the set-up, the length of time contracted for, as well as what other studios in your area are charging for their services. The wisest thing for you to do is to contact nearby studios, find out what they offer and what their going rates are and on the basis of this information compute what a reasonable cost would be.

A Reader's Viewpoint

My compliments once again on the excellence of your publication. I'd like to cast my vote for more articles concerning "live" sound reinforcement, although I do enjoy the recording studio "in session with" articles very much indeed.

I do wish to point out a small error made in the artist's profile of Frank Zappa (see Profile, March 1978, page 46). Some poor beginner might be confused by the mention on page 50 to the "8440" reference tone if he doesn't realize that it should have read "A 440 Hz."

Another subject of importance to me, and obviously to other readers as well judging by the letters I've seen printed, is sources of instructional materials for learning the technical arts of recording and "live" sound reinforcement. *All* the available magazines (*Audio, Stereo Review, Popular Electronics, db, Recording Engineer/Producer, Audio Amateur*, etc.) offer some sort of information, some of more value than others.

Many books have been published on related topics, but all seems to have shortcomings of one sort or another. Don Davis' *Sound System Engineering* is geared primarily to non-musical sound reinforcement and is extremely technical in nature. I found Bob Heil's *Practical Guide For Sound Reinforcement* (although containing quite a bit of information and ideas) to be poorly edited and somewhat biased in many of its statements. Of the many other fine recording books, I find John Woram's *Recording Studio Handbook* extremely well written. One book which seems to have been ignored to some degree, but which I found to be a real gem is Abraham B. Cohen's *Hi-Fi Loudspeakers and Enclosures*. It contains very thorough and practical explanations of all types of enclosures and in the process has much to say about acoustics and sound in general.

Hope you keep up the fine work.

—T. Young
New Hartford, Ct.

We can't figure out how that "erroneous tone" slipped past our usually sharp eyes, but you are absolutely right. Thank you for catching it.

Take Two

In April of 1977 I picked up my first copy of *Modern Recording*. Needless to say, that afternoon I mailed in a two-year subscription. Your fine magazine has been of great help to me in the area of sound reinforcement. I would like one question answered, though: where can I get a reprint of Jim Ford and Brian Roth's three-part "P.A. Primer?" The back issues of *MR* in which it originally appeared (June/July, Aug/Sept, Oct/Nov, all of 1976) are apparently out of print. When I tried to get it with the *MR* 1978 Buyer's Guide, I received a letter saying that because of the huge demand for the Guide you were temporarily out of stock. My check was returned. Please tell me where I can obtain a reprint of this article and, of course, a copy of the 1978 Buyer's Guide.

—Chris Christel
Greenfield, Wi.

We hope you still have $1.50 in that checking account because we have received a new shipment of Buyer's Guides and, if you act quickly, we will be able to satisfy your request this time. However, we cannot be sure how long we will have it in stock so please don't delay.

The Ultimate Tool Box

The following letter was prompted by a request made by Brian Roth and Jim Ford in their article "Inside A Soundman's Toolkit" that readers share their helpful hints on stocking up and taking along all the indispensables.

I found your article "Inside A Soundman's Toolkit" by Jim Ford and Brian Roth (February 1978, page 60) to be very interesting and I would like to offer a bit more on the subject.

Having a bit of a fetish for well-kept tools and being in the business of supplying technicians and engineers with the tools of their trade, we feel that we have at last come up with the ultimate tool box.

Looking at several of the more prominent suppliers of electronic tool cases (Jensen, Xcelite, Platt) we found that all offer very well layed out cases. The cases are generally available either with or without tools. The empty case seems to us to be the best choice for it allows for the most flexibility in the user's special requirements or preferences. However, none of these cases offer much in the way of roadability; that is to say, none will hold up to abuse—dropping, being stood on, etc.

Realizing that most, if not all, of the items for which tools are required (amps, speakers, instruments) have protective cases, we soon thought, why shouldn't the tools and test equipment be similarly protected? We had Anvil Cases build us cases modeled in part on the Xcelite TC100, which also was the source for the pallets. The pallets in the top of the case have fold out "wings" which provide plenty of pockets and the bottom pallet is suspended in the case allowing room beneath for meters, soldering irons and small parts. Supplied with a combination lock, and standard Anvil features, this case provides a well-organized, roomy and nearly indestructible portable workshop.

Advantages of this type of case with a separate pocket for each tool, include easy accessibility and easy inventory—it is simple to find out what's missing so we don't leave it behind.

We offer the case empty so that each person may choose those items he deems necessary to include. We are, however, willing to make some basic suggestions patterned very much along those mentioned in your article. More elaborate suggestions have included temperature-controlled soldering stations, portable vises (Panavise), parts bins, shrink-tubing assortments, heat guns, and some cordless tools. A recent purchaser plans to install removable legs on his!

I hope your readers find this material of interest.

—Joe Phillips
Vice-President
Pacific Audio Exchange
Hollywood, Ca.

Aroused By Aphex

I want to add my name to the list of enthusiastic readers who have taken advantage of this column space to offer their congratulations on a fine publication. I have found *MR* to be the most informative magazine around for both musicians and engineers.

My other reason for writing to you is the curiosity and excitement that you've aroused in me (and many other readers I'm sure) regarding the Aphex Aural Exciter. The authors of both articles you've printed thus far (Michael Gershman's "The Aphex Aural Exciter—A Psychoacoustic Phenomenon," October 1977, page 38 and Jim Ford and Brian Roth's Hands-On Report, March 1978, page 64) started off sounding a bit skeptical about the promises made by its creator, and wound up sounding, if not positively convinced of its capabilities, at least optimistic about its possibilities.

You've done an excellent job of covering its technical points, but as a follow-up, how about an article on its actual in-the-field performance, perhaps how different major artists are using it, such as James Taylor did on his recent success, *JT*?

—Jim Ittenbach
Crownsville, Md.

We have no plans at this time to search out a specific session at which the Aphex is to be used. However, we are fairly certain that in the course of covering a major recording session each month, we are bound to encounter this unique device at work sooner or later. When this happens, we will be sure to include all the details for those readers, like yourself, who are eager to learn more about its applications.

Two For The Dan

[*We received the two succeeding letters independently of one another, and we direct our response to both gentlemen as well as all those other "Dan Fans" who haven't written but that we just know are interested in the whys and wherefores of these fine musicians.*]

First off, I would like to congratulate you on a fine publication. I read every issue from cover to cover and love every word of it!

I would also like to know if you've ever done a feature story on Steely Dan. If so, is it still possible to receive a copy of that issue?

—Joe Bocchetti
Dorothy, N.J.

I would like to know if you have any plans to run a story on a band that we all really love—Steely Dan. I think that a superior band like the Dan would receive more attention from the media than they do now. Donald Fagen, Walter Becker and their producer Gary Katz are a team that have been writing and recording superior music for almost seven years now, but I never seem to see much about them. They are top-notch studio musicians and their albums prove their musical genius. How about an article on these wizards of today's music world?

By the way, did they cancel their tour scheduled for this year?

—Joseph E. Forthome
Maumee, Oh.

If there is a dearth of material written about Steely Dan, one must assume that it is through their own choice. Larry Solters of Front Line Management (the company that handles Steely Dan) told us that–while he's not ruling out the possibility of a tour sometime in the distant future–Becker and Fagen prefer to record and create in the studio. They've never toured and have no current plans to do so.

The Series IS, based on the world famous industry standard Series I. Unequalled features, technical sophistication and a modest price.

Input channels (12, 16 or 20)
Transformer balanced mic input with a 20dB pad. Variable gain mic amp. Insert send/return (line input). 120Hz high pass filter. Four band EQ, with the two mid band frequencies sweepable. Two monitor sends (post-EQ) and one echo send (post-fade). Automatic pre-fade Solo. LED peak indicator whose delay time indicates the relative size of the transient.

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Left and right main, monitors A and B and master echo, each with two band EQ, solo and insert. Each output may be balanced by a plug-in transformer.

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Two studio quality VU's and peak reading LED's display the main stereo output or any function soloed.

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There's both talkback and intercom. The talkback mic can speak into the main output, monitors A or B, or into a ClearCom (or compatible) intercom system.

Specifications
Excellent, ie incredibly quiet and distortion-free.

Finally
Two echo returns, conductive plastic potentiometers throughout, socket for Shure lamp and, of course, the Soundcraft comprehensive 2-year warranty.

Encore.

The new EX4S studio quality 2, 3 or 4-way stereo electronic crossover.

Internal switching
The facilities for changing the crossover points, and for converting the unit to a 2, 3 or 4-way are inside, to provide maximum protection for P.A. systems, by avoiding accidental switching.

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Eight band-attenuators, eight LED peak indicators, and LED's to indicate 2, 3 or 4-way mode.

Circuitry
Bessel function filters (superior to Butterworth filters in other crossovers) give an ultimate slope of 24dB/octave, the most linear phase response and the best transient response. The result is, quite simply, a better sound.

And the rest
EX4S is built into an all extruded black anodised 19" case, tough enough to stand up to all the wear and tear of the road. XLR and multipin connectors on the back. Inputs are electronically balanced while outputs may be balanced by plug-in transformers. Of course, it's also covered by Soundcraft's comprehensive 2-year warranty.

Début.

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CIRCLE 83 ON READER SERVICE CARD

While we haven't had the opportunity to feature a piece on the Dan, you'll be happy to know that Gil Podolinsky took a close look at their latest effort, Aja *(see March 1978 Groove Views, page 70), and confirms that their musical prowess is all it's cracked up to be.*

No Purchase Necessary

[We received the following letters in response to Brian Roth's answer to a reader's question concerning the feasibility of building his own mixing board (see "Plan On Buying It," Letters To The Editor, February 1978, page 8).]

First of all, thanks for a great magazine which fills most of my audio and recording needs. Keep it up!

Second, in regard to Neill C. Porter's Letter in the February issue, I can give you the source I used to build my 24-in/8-out mixer.

Most of the circuits were taken from a Howard W. Sams' publication *IC Op-Amp Cookbook* by Walter G. Jung. I had to tie the circuits tegether to build the mixer, but that was part of the fun (that and saving lots of bucks!).

With *well-matched* components, and no input transformers, I get a common-mode rejection ratio of equal to 105 dB, with an input noise figure of 1.5 dB uvolt and a flat frequency response to beyond audibility.

—John Nagle
Audio Consultant Specialist
Gales Ferry, Ct.

In regard to Neill C. Porter's request for information on building a mixer for his sound reinforcement system, I offer the following suggestions.

The magazine *The Audio Amateur* has run several articles on mixer construction with hints on expanding the mixer in building-block fashion. The address is P.O. Box 176, Peterborough, New Hampshire 03458. I understand that back issues are available.

Also, Craig Anderton's book, *Electronic Projects For Musicians* offers an 8-input basic mixer with instructions on expanding to 16-in and stereo operation with panpots. All of this is presented as an expansion of a basic high-level 8-in/1-out mixer which uses low-noise 4739 ICs as the summing amplifier. Craig's book is available from Godbout Electronics, P.O. Box 2355, Oakland Airport, California 94614. Godbout also offers kits for the projects in the book.

Hope this proves helpful to other lovers of *MR*.

—Bob Eiser
Toledo, Oh.

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you need a precision machine.

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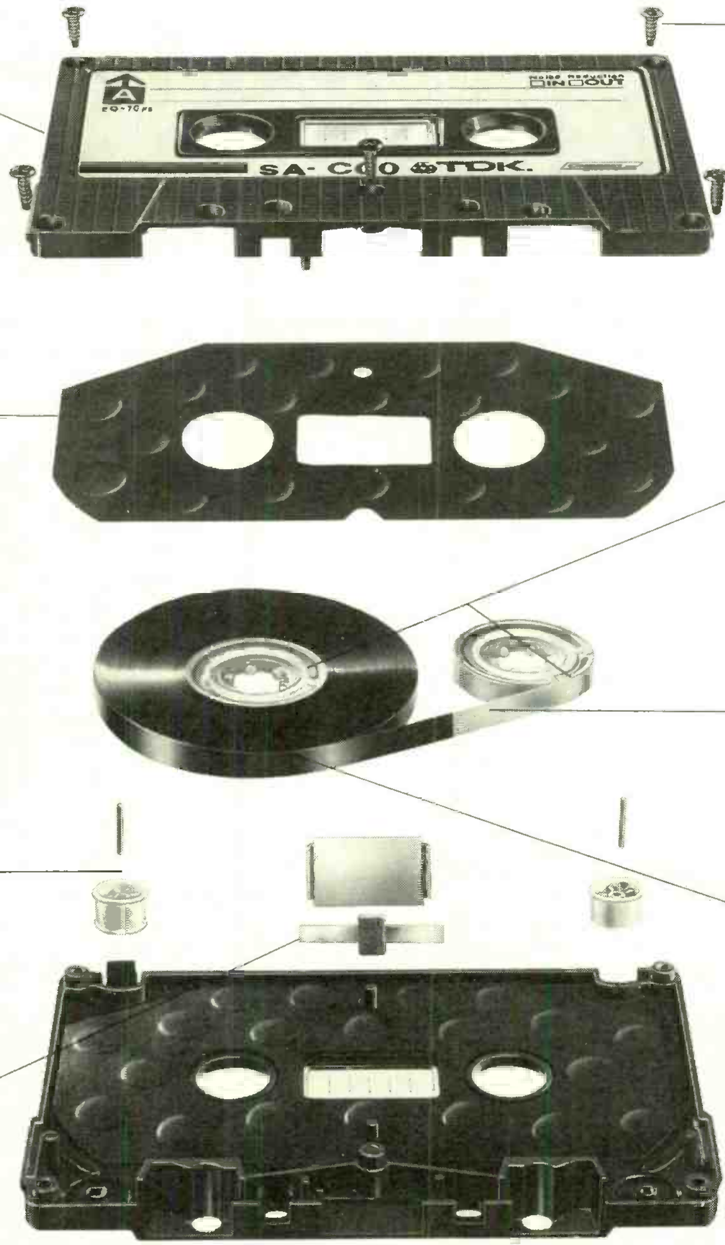
Tapered, Flanged Rollers—direct the tape from the hubs and program it against any up and down movement on its path towards the heads. Stainless steel pins minimize friction and avert wow and flutter, channel loss.

Resilient Pressure Pad and Holding System—spring-mounted felt helps maintain tape contact at dead center on the head gap. Elegant interlocking pins moor the spring to the shell, and resist lateral slipping.



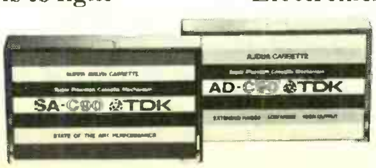
Five-Screw Assembly—for practically guaranteed warp-free mating of the cassette halves. Then nothing—no dust or tape snags—can come between the tape and a perfect performance.

Perfectly Circular Hubs and Double Clamp System—insures there is no deviation from circularity that could result in tape tension variation producing wow and flutter and dropouts. The clamp weds the tape to the hub with a curvature impeccably matched to the hub's perimeter.

Head Cleaning Leader Tape—knocks off foreign matter that might interfere with superior tape performance, and prepares the heads for...

Our famous SA and AD Tape Performance—two of the finest tapes money can procure are securely housed inside our cassette shells. SA (Super Avilyn) is the tape most deck manufacturers use as their reference for the High (CrO_2) bias position. And the new Normal bias AD, the tape with a hot high end, is perfect for any type of music, in any deck. And that extra lift is perfect for noise reduction tracking.

TDK Cassettes—despite all we put into them, we don't ask you to put out a lot for them. Visit your TDK dealer and discover how inexpensive it is to fight dropouts, level variation, channel loss, jamming, and other problems that interfere with musical enjoyment. Our full lifetime warranty* is your assurance that our machine is the machine for your machine. TDK Electronics Corp., Garden City, N.Y. 11530. Canada: Superior Electronics Ind., Ltd.



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 73 ON READER SERVICE CARD

...urselfer

...I am looking ...f speaker cab... ...musical instru... ...d you help me ...nd steer me to ...have this infor...

—Edward Sison
...t Palm Beach, Fl.

You should be able to find the information you need in the December 1977 issue on page 8 in the Talkback item entitled "Folded Horn 'How To'."

Help Is On Its Way

I wonder if you can help me out. I recently purchased an Ashly Audio SC-50 parametric peak compressor/limiter and, to my surprise, there was no instruction manual included. I understand the features found on a compressor/limiter, but I really need to know the parameters of this machine. I also thought you might feature this piece in an upcoming Lab Report. —Alain Benetis
Longueuil, Quebec, Canada

We called Dick Webber of Ashly Audio to tell him of your dilemma and he informed us that the instruction and usage manual for that piece is in the process of being printed. He has added your name to the list of those who are eagerly awaiting the manual and you can expect to receive it within three to four weeks. For more information on the piece, you can refer back to Norman Eisenberg's Product Scene column in the August, 1977 issue of Modern Recording *(page 19). We have no plans at this time to lab test this piece.*

Here's Herald!

Can you help me locate the address of Herald Electronics of Lincolnwood, Illinois? I need a part from them and only have their name and town.

—D. Olmstead
New Haven, Ct.

You now have name, town, et al. Herald Electronics is located at 6611 N. Lincoln Ave., Lincolnwood, Illinois 60645. Their telephone number is 312-675-1100. Hope this eases your route to replacement.

Mention Our Name

I am interested in Westlake Audio's Model 1200 Headphone Mult Box which was shown way back in the July 1977 Product Scene (page 30). However, I cannot find an address to write to for more information. Would you please tell me Westlake's address? I would appreciate it much!

—Brian Welty
Nappanee, In.

Westlake Audio, Inc. is located at 6311 Wilshire Blvd., Los Angeles, California 90048. You can write there for the info you desire or call Sales Manager C.J. Flynn at 213-655-0303. All we ask is that whatever you do, mention that you saw it in Modern Recording. *We would appreciate it much!*

Inadvertent Inversion

Somewhere along the line from Len Feldman's oscilloscope to the finished April 1978 Modern Recording *as it rolled off the presses, a photo was inverted. The figure on page 67 of the Lab Report on the Garrard MRM-101 Music Recovery Module would read correctly to a technician's eye if it were turned the other way.*

Sorry for any confusion this might have caused our readers, and our apologies to Len, who always gets them to us right side up.

—Ed.

His Music is Pure Magic...
His Sound Equipment is
Dynacord
"Mr. Magic"—
that's what they call Grover Washington, Jr. Washington and his group, Locksmith, make their most stunning musical magic with sound equipment from Dynacord.
"The Dynacord Sound is clean," Washington says. "The clarity of the sound is incredible."
We're proud that someone like Grover Washington, Jr. says nice things about our equipment. We know that our complete line of Dynacord Sound Equipment is durable, very portable and AFFORDABLE.
As "Mr. Magic" puts it, "Dynacord Sound Equipment ... use it to make your own magic."
EC 280
Analog Delay/Reverb System
MC 1640
16-Channel Mixer for P.A. Systems
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For free, complete catalogue of Dynacord Sound Equipment and Demonstration, call (215) 482-4992 or write, P.O. Box 26038, Phila., Pa. 19128.
MC 1230
Stereo Mixing Console

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.*

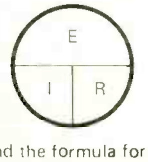
"O" Is For Ohm's Law

In one of your recent articles, a reference was made to the Ohm's Law. I tried to find out exactly what this law states, but to no avail. Can you give me a simple, complete explanation?

—Mark Allen
Salina, Ks.

Ohm's Law is a relationship that exists between three electrical quantities in a circuit. For D.C. (direct current) circuits the three properties of interest are E, voltage (in volts); I, current (in amperes) and R, resistance (in ohms). Ohm's Law written as a formula states: E divided by R equals I. This formula tells us that the amount of current flowing through a resistance is equal to the voltage applied to the resistance divided by the value of the resistance itself. Inspecting the formula, we notice that 1) a higher voltage can "push" more current through a given resistance and 2) a lower resistance will permit more current to flow in response to a given voltage. A simple circuit diagram consisting of a battery, resistors, voltmeter and ammeter will further serve to illustrate the relationships.

Ohm's law pie



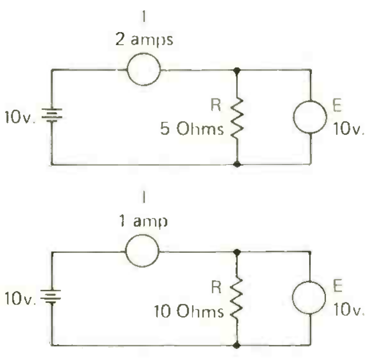
$E = I \times R$

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

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

To find the formula for a circuit property, cover the symbol for that property with a finger. The remaining symbols will be in the form desired.

Remember $\frac{E}{R}$ means $E \div R$ or $R\overline{)E}$.



Ohm's Law also applies to A.C. (alternating current) circuits in a similar but slightly more complex way. Basic circuit components—resistors, coils and capacitors—affect not only the amplitude of voltage and current but also the phase relationship between these two properties.

—Peter Weiss
Engineer
CBS Recording Studios
New York, N.Y.

A Perplexing Minimoog

In the process of creating and recording jingles, scores for about sixty TV shows a year and soundtracks for a dozen or so films, we have acquired quite a few very specialized instruments for use in our recording studio. The most perplexing have always been the recorders themselves, but I am writing this letter about something somewhat removed from recording—our Minimoog. I'm sure other studios have had this problem, so perhaps you could do us all a favor and come up with an explanation.

The instrument in question is the standard Moog Model D (ours is serial number 1211) which I bought new in 1972. I can't count on it ever being in tune—even when it's tuned right before a take. Either the separate oscillators go out or the whole thing does. If more than a few minutes have elapsed or if a Leslie is whirling anywhere in the vicinity, or if anything has happened to vary the temperature in the studio even a few degrees, it will not stay in tune.

I had it recalibrated a few yeras ago, but that did nothing to alleviate this problem. I have a friend in Italy with the same instrument who experienced the same difficulty and his was satisfactorily repaired over there.

Short of leaving it on all the time—even that doesn't always work—what can we do?

—Robert S. Gollihur
Vice President & General Manager
G.B. Faneh Productions
Pitman, N.J.

Your Model D Minimoog serieal number indicates it is an early production run. Several modifications and circuit updates have been implemented since its manufacture which increase the overall circuit stability. Some of these updates may affect your instrument, Please contact the factory for installation instructions or service.

Presently there are two versions of oscillator boards in the field. Earlier boards were designed with discrete matched transistor pairs in the current drive sections. These oscillator boards are easily identified as they contain ten trimpots as compared to the seven trimpots used on the newer, more stable, version.

Another modification affecting the later oscillator boards requires the installation of RC networks, along with some precision resistor and capacitor value changes to further improve oscillator stability.

One important circuit change concerns the interaction between front panel ranges. To determine the need for this change would require only a simple test.

With all modulations off, listen to oscillators no. 1 and no. 2 on a higher octave range (such as 4'). Depress any note on the keyboard and insure that the two oscillators zero beat. Without listening to oscillator no. 3, switch its octave selector through all ranges. If oscillators no. 1 and no. 2 begin to beat against each other, your instrument requires the additional buffer circuitry that is present on all current Minimoogs.

Pitch problems may also be due to some unstable integrated circuits. These ICs are factory selected for offset and stability and are located on the oscillator board. A change of 1 millivolt on the outputs of these ICs will cause a pitch change noticeable to the ear. Thus, it becomes evident how important these components are when analyzing the problem of changing pitch.

Due to the age of your instrument, it may be wise to clean the gold-plated printed circuit board contacts "pads" or "traces" which plug into the circuit board connectors. Smoke and dust deposit a tarnishing film over these areas which over an extended period of time may cause intermittent high resistance connections. Precision control voltages flowing through these connections become innaccurate and directly affect the system. Below the left hand controller are two other important connectors that should also be cleaned. A typewriter eraser would suffice in these areas to remove the tarnish.

Temperature variances will affect the tuning of the Minimoog to some extent, but drastic pitch changes would indicate that a circuit problem exists. A standard twenty-minute warm-up time should be observed before attempting to calibrate the instrument. This will insure the circuits have all temperature stabiliized.

If further technical assistance is required please contact the factory, at this address: Moog Music, Inc., 2500 Walden Avenue, Buffalo, New York 14225, telephone number716-681-7242.

—Donald J. Besecker
Field Service Specialist
Moog Music, Inc.
Buffalo, N.Y.

A Good Beginning

I'm in the process of buying a new P.A. system and I'd like your opinion on several things as well as the answers to a lot of questions. I am just getting into bi-amped or tri-amped systems, and to say the very least, it's all Greek to me.

So far, the system I am planning consists of four JBL 4560 type cabinets with Gauss 5842 speakers, two Peavey MF1-X radial horns, two Peavey T-12 tweeter banks, one Peavey CS-800 power amp, one Peavey CS-200 power amp and one Peavey 600 S mixer.

I guess the best way to start is to tell you how I plan to set the system up and let you advise me on my idea.

The four JBL cabinets are going to be powered by the Peavey CS-800. Two of these cabinets are going to be used for the extreme low end, and the other two cabinets are to be for the mid-bass or lower end of the mid-range frequencies.

The two Peavey MF1-X horns and the two T-12 tweeter banks are going to be powered by the Peavey CS-200. The MF1-X horns are going to be used for the upper mid-range and the T-12 tweeter banks for the high end.

The major part of my misunderstanding is in regard to ohms. The CS-800 is rated at 400 RMS per channel into 4 ohms. The JBL cabinets are 8 ohms each. So, if I connect two of these cabinets per channel, does it equal the required 4 ohm load?

My next question is a bit more complex, at least to me. The MF1-X horns are 8 ohms each. I cannot find any resistance rating in regard to the T-12 tweeter banks! Does a tweeter bank have any resistance?

To continue, the CS-200 mono amp is rated at 140 RMS into 8 ohms, 220 RMS into 4 ohms and 120 RMS into 2 ohms. The CS-200 is going to power the MF1-X radial horns and the T-12 tweeter banks, but I don't know what power rating I'm going to end up with, because I don't know the resistance of the T-12s! Please answer that question!

Now, as you have probably noticed, this system is still missing several important pieces of equipment, specifically, a crossover and an equalizer. This is quite simply because I don't know much about either one. I do know what they do and why they're necessary, but I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CIRCLE 96 ON READER SERVICE CARD

would want more advice before I invested in one. Several have been recommended to me. These include the Uni-Sync XO-4 Crossover, the Uni-Sync EQ-241 Equalizer, the Tapco 4400 Equalizer and the Tapco 2200 Reverb Unit. If you have any other recommendations I would appreciate you passing them on to me. Otherwise, I'd like your opinion of these.

Correctly setting up the system is another one of my concerns. The use of "sends" and "returns" is also new to me—an explanation of these terms please? Can you possibly tell me all the basics I need to know to properly set up the system?

I know this is a lot to ask, but you guys are the people that know. The dealers in the music stores down here, once you get right down to it, don't know that much more than I do. And the really heartbreaking thing is that they'll talk in circles about nothing if they think you'll spend more money. So if you'll help me out, I'll know what I need to know to get started and most of all I'll know it's right!

Thank you for your time and trouble.

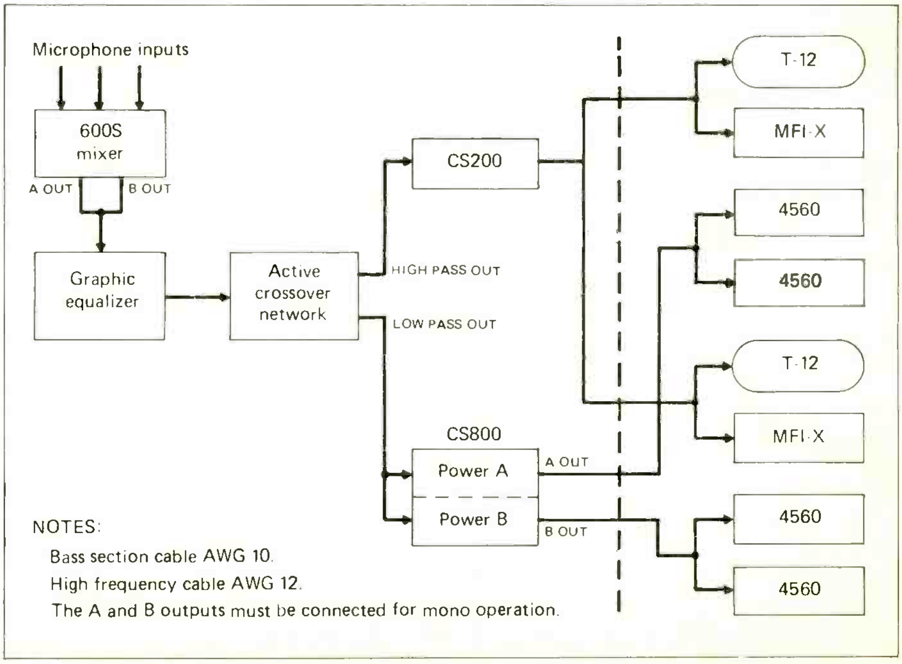
—Butch Fitzgerald
Moorestown, N.J.

As indicated by your letter, a great deal of information is in order and adequate answers to all of your questions would require writing a book on the subject. I plan to do this someday; however, even then I would not profess to be able to tell you everything you need to know. I hope, however, at this time, to be able to give you some interconnecting and operating guides for the equipment you have listed.

As you can tell from the block diagram, I have done a bit of redesign on your system. The most notable change is the departure from your original split bass cabinet approach. There would be no gain in performance by using two 4560s for low bass and two more 4560s for the mid bass. It will be much more efficient, easier to interface and most of all, less expensive to use all four bass cabinets for the low end of a two-way biamplified system.

A simplified explanation for this approach is that the 4560s are a combination horn/bass reflex type of enclosure that performs quite well up to approximately 800 Hz. They can be stacked for coupling if used in a two-way system, and you will lose the benefit of coupling if you try to use a split bass configuration. You will also run into phasing problems between your bass cabinets in a split bass system.

As far as bass system impedance matching is concerned, you will have a near optimum condition if you connect your components as indicated by the block diagram. Each side of the CS-800 will have a total load impedance of approximately 4.0 ohms which is optimum for this unit. You should also note that your power amplifier/speaker system is well matched. With everything connected as shown, each speaker will be required to handle an RMS power of 100 watts. This is a safe power



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Remember when recording was simple? The only one you had to satisfy was you. As tracks of information have grown from 4 to 16 and beyond, so have the demands placed on the board operator. Now, instead of satisfying just your mix, there's everyone else's to consider.

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for these units. The CS-800 power amplifier is fitted with LED peak overload indicators which indicate that the amplifier is at clipping. It is normal for these lamps to illuminate for short periods during extreme transient conditions. If the overload indicators remain illuminated for any length of time (greater than 20%), I will guarantee that something in your speaker system is going to quit. A modern high power amplifier can destroy almost any speaker made if it is run into clipping for any length of time!

In your system, I would recommend an 800Hz crossover frequency, although the use of 500 Hz can be considered. The reason I picked the 800 Hz point is that you will have a very good balance between your bass cabinets and high frequency horns without placing an undue amount of stress on your horn drivers. The lower you go with the crossover point, the more work your horn drivers will be called upon to deliver. With a quality system such as yours, there is no point in stressing any of the component parts.

The crossover network itself can be any professional quality unit as long as it has a 12.0 dB per octave or greater rolloff at the crossover point. For your application, I would recommend a Spectra Sonics crossover because it is extremely reliable, requires no adjustment and is compact. You can buy much more sophisticated units with all kinds of knobs and other frills but keep in mind that one incorrectly adjusted knob can destroy your horn drivers in short order. In your system, simplicity and reliability are essential.

The Peavey MF1-X radial horns are a good choice for your high frequency components. I have used and recommend JBL horns and drivers for specific applications such as whenever a very directive horn array is required. The basic difference between the MF1-X horn and a conventional radial horn design is that you have very wide, yet controlled, dispersion characteristics. This is essential in your system because you only have one horn/driver combination per side and you need a very smooth and controlled pattern at all frequencies within the passband of your horn/driver combinations.

As far as impedance matching is concerned, you will have no compatibility problems with your CS-200 and high frequency components. The total load on your CS-200 with two MF1-X systems and two T-12 tweeter arrays will be approximately 3.8 ohms. To answer your question on the tweeter banks regarding their impedance would take three pages of rather complex mathematical analysis. The tweeter banks have an impedance curve that looks like a roller coaster. They may be 500 ohms at one frequency and 5.0 ohms at a higher frequency. After using the things for a number of years, I can safely say Don't worry about it, they will work fine.

Because you are relatively inexperienced in electronics, I am going to recommend that you follow my advice on how to connect your total system. I can wave my undergraduate and graduate degrees all day long and still not establish any credibility with all of the "audio experts" that seem to exist everywhere. I can, however, say that after being on-the-road over a period of years as a professional sound contractor, I have learned quite a bit through the school of trial and error. Please follow my advice until you gain some practical knowledge.

All connections between your mixing console, equalizer, crossover network and power amplifiers must be made with good quality shielded cable. I would recommend that you purchase all of your shielded cables from a recognized company such as Switchcraft or Radio Shack. Pick up a few spares while you are there; I guarantee you will need them. Connect the equipment as indicated on the block diagram.

At this point, I am going to recommend something that will cause many "audio experts" to have a fit and go into controlled feedback. Locate your power amplifiers at the mixing console and run heavy cables to your speaker system. I have a very good reason for telling you to do this—*ground loops*. For someone who does not have a working knowledge of electronics, this is the safest way of assuring that you do not get a loud 60-cycle hum when you set up your equipment. For your size system, number 10 cable should be used for the bass section and number 12 cable should be used for the high frequency section. You will lose a few watts in the line; but take my word for it, this approach will work the first time you plug it in. The only precaution is to very clearly mark your low frequency cables to help prevent you from accidently plugging them into your high frequency horn/driver components. If you do reverse the leads, the result will be quite spectacular, loud, and expensive. When wiring your speaker system components (cables, jacks, bass cabinets, etc.),

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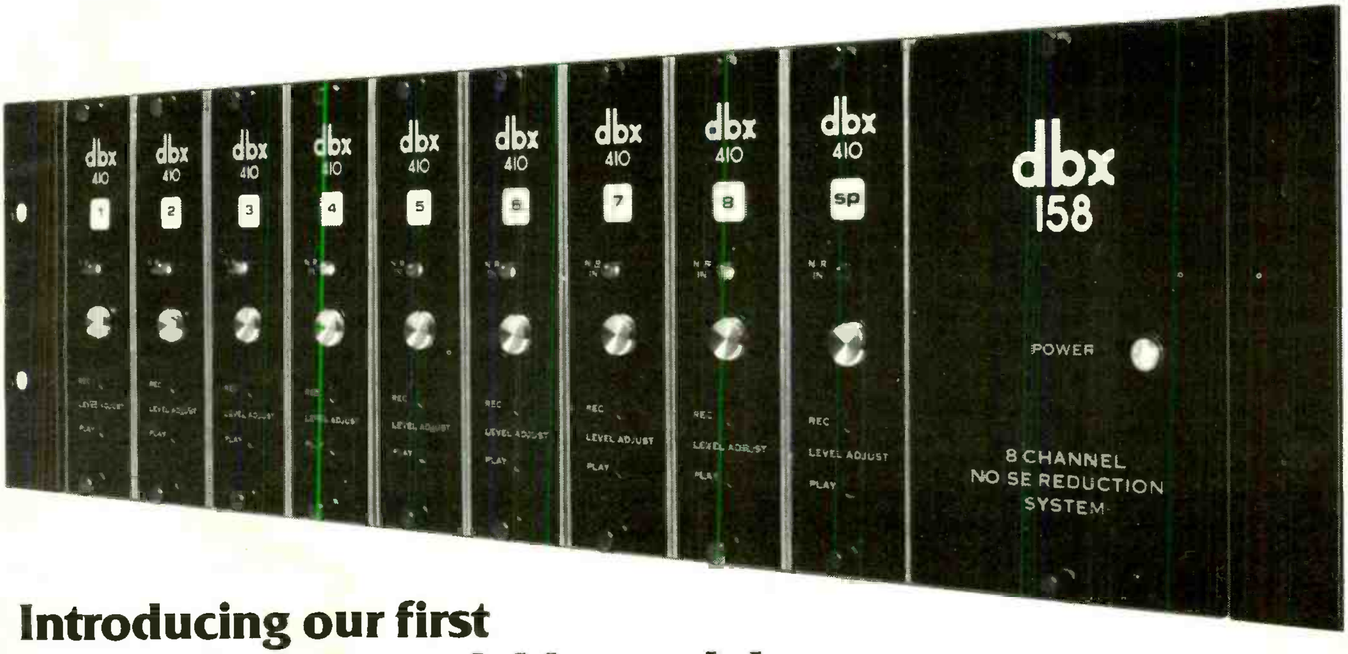

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Our new dbx 158 system can start life in your place with the 158 main frame and as few as two modules or as many as eight modules for its full eight channel capacity. It also has storage space for a ninth spare module in its compact chassis. The rear panel has phono and multi-pin connectors that will interface directly to your cables. Additional 158's can be used for 16 or 24 track recording.

The dbx 158 offers the semi-pro recordist or small studio all the advantages of dbx professional systems, including 30 dB of noise reduction, and 10 dB additional recorder headroom. It's a classic 2:1 mirror image compander which preserves the full dynamic range of program material without audible tape hiss. Each module contains separate record and playback noise reduction electronics. Its simultaneous record/playback capability permits the noise reduced, decoded tape to be monitored while recording without manual switching or remote control.

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You can add all the eq you want, and be heard as sweet or as gusty as you want, because what you put in is what comes out. There's no built-in coloration to irritate your audience or drive your sound man up the wall.

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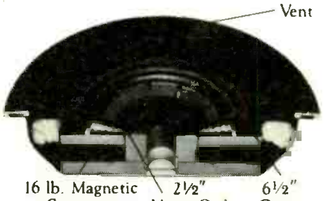
It used to be that if you wanted high sound pressure levels in the midranges you had to use a horn. But it took a *big* horn to keep the sound accurate. Trouble is, big horns just won't fit into small stage system enclosures. So you had to settle for the pinched, "honky" sound typical of small horn drivers. *No more.* The midrange speaker in the S15-3 is a cone. Not just any cone, but a cone in an integral *vented* enclosure coupled to a massive 16 lb. magnet structure. The vent and cone work together to deliver accurate midrange, without coloration, at high sound pressure levels heretofore associated with horns. The vented midrange speaker is an E-V first.



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IOWA
DES MOINES
Celestial Power & Light
DUBUQUE
Rondinelli Soundworks

KANSAS
HAYS
Sunshine Sound
MANHATTAN
Music Village
WICHITA
Superior Sound Rental & Service

make sure that everything is in phase. Have a qualified technician check your wiring and bass cabinets for proper phase relationships before actually using your system. Improper phasing will not destroy anything, however, the resulting sound will be terrible and the probability for feedback will be greatly increased.

I cannot begin to tell you how to operate your system, as every location will be different. I will, however, list a few of the most important operational considerations.

1. Use good quality microphones. I strongly recommend the Electro-Voice DS-35 as a general purpose microphone. Try to avoid the very sensitive (and fragile) condenser microphones until you gain some experience. *Do not* use a microphone with an on/off switch. They are always in the wrong position.

2. *Do not* over-equalize. Use the graphic equalizer for feedback control and the board equalization for the required tonal balance.

3. Stay with quality equipment produced by recognized manufacturers. A custom-made console may work fine in the shop but what do you do when it quits in Waco, Texas?

4. Use common sense and read everything you can on the subject. When you make a mistake, learn a lesson; this is the most valuable type of education.

—Lothar A. Krause, Jr.
Design Engineer
Peavey Electronics Corp.
Meridian, Ms.

To Gobo Or Not To Gobo?

I have recently read in *The Recording Studio Handbook* by John M. Woram his section on the use of acoustic baffles such as goboes. I am confused on the use and placement of such devices. He seems to avoid the things like the plague, yet I have seen them in studios. Woram gave me all the "don'ts" about the use of goboes, now can you please advise me on the "dos" on how and when to use goboes?

—Paul Kalris
Bellevue, Wa.

Well, you do have a point. Most studios do use lots of goboes, despite the fact that many microphone designers don't think much of them.

I've found that, more often than not, goboes cause more problems than they cure. However, if you *must* use them, consider the following few points before setting them up.

In order for a directional microphone to function as designed, the space surrounding it must be free from obstructions. This allows the sounds from the side and rear to enter the microphone (via its side- and rear-entry ports) and thereby cancel out. Prove this to yourself by covering these ports with your hand, and listen to the drastic deterioration in the microphone's performance—it's not subtle. The point is, *any* obstruction in the vicinity of the microphone is apt to have an undesirable effect on sound quality. The deterioration may be either slight, or quite obvious, depending on the nature of the obstruction.

The next point to ponder is that practically every construction material used for gobo-building (wood, masonite, fiberglass, etc.) has a very uneven "frequency response." This means that sounds reflected (or absorbed) by these materials have a drastically atlered sound quality. Also, unless the gobo is infinitely large, low frequencies are refracted (bent) around it.

Therefore, since no gobo offers total, or even uniform, absorption of all frequencies, the sounds that get by it have a severely distorted frequency response, and it is these sounds that are then picked up by the microphone. The result is usually an unpleasantly "muddy" sound.

If you don't mind this kind of sound "leakage" into your microphone, then feel free to use goboes wherever you like. As for me, I'd rather have "clean" leakage, and then try to minimize this by careful microphone placement.

A little trial-and-error will tell you if a gobo is helping or hurting your recording. Try listening (carefully!) to the leakage, with and without the gobo in place. If it's fouling up the sound, get rid of it.

Be prepared to make at least a few test takes, to convince the people you're working with that you are not really crazy. A lot of recording-types look at the gobo as a sort of acoustic security blanket and get very upset when there aren't any in sight. Unfortunately, the basic laws of physics have not yet been repealed in the recording studio, although it's often impossible to get this point across.

—John M. Woram
Audio Consultant/Author
Woram Audio Associates
Rockville Centre, N.Y.

Clearing Up Some "Live" Misconceptions

I have two questions concerning "live" sound reinforcement that I hope you can answer for me.

When setting up stage monitors, it seems to be common practice to cup one's hands over the microphone to induce any feedback, which is then (hopefully) EQd out. Why will a mic feed back when this is done? I know from experience that putting one hand against or near the mic can cause it to start and that during an actual performance the singer's face can also activate ringing. But what it seems is happening (especially with hands cupped around the diaphragm area) is that you are acoustically isolating the mic from the monitor speaker, therefore, why is there such severe feedback?

When employing real time analysis and equalization in "live" sound reinforcement, it has always seemed obvious to me that the end result is very seldom "flat." That is, once you obtain a flat response from your system, you continue to EQ by ear until a pleasant tonal balance is achieved. However, I've read numerous accounts by pros who seem to believe that a flat system is the

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CIRCLE 57 ON READER SERVICE CARD

end goal, and I have heard many concerts set up this way which sound flat and not very pleasing or natural. Does this indicate that pros who usually have access to advanced techniques and equipment are lacking aesthetic common sense or am I confused somewhere along the line?

—T. Young
Thomaston, Ct.

When you cup your hands around a microphone, you are *not* acoustically isolating the mic from the monitor speaker. Your hand is acting like a plate or ground to the diaphragm of the mic and it changes the pick-up pattern of the mic to an omni-directional or some other weird pattern that may be quite unpredictable. Thus, the mic will pick up the monitor speaker even stronger than before and this will cause excessive feedback. This changing of the mic's pattern can also occur if the mic is held too close to the singer's face or body.

As to why real-time equalization of "live" sound reinforcement doesn't always sound natural, this can be the result of any number of situations. Rather than go into great detail, I'll just briefly describe the various possible reasons.

The equalizing of the system overloads the amps in certain frequency bands causing distortion, or the equalizer used may not be the best suited for the application, causing much more distortion to be present in the system than without it.

The system may be set up for a "flat" response only in one spot in the room, therefore all other areas of the room may not sound as good.

When setting up the room the engineer may have set up the EQ for a "flat" response without people, then when the audience is there (bodies and clothing absorb the higher frequencies) the room will sound very dead and not pleasing or natural.

Also, most engineers know that even the most advanced real-time analysis equipment is not as good as the human ear and it can't tell you if the sound is pleasing or natural. Therefore, engineers should only use real-time analysis and equalization as a tool to obtain a good sound and rely on their ears and past experience when it comes to the final mixing and equalizing of the speakers.

—Clyde R. Green
Chief Engineer
Cookhouse Recording Studios
Minneapolis, Mn.

A Symphonic Affair

There is the possibility that I might have the opportunity to aid in the recording of an orchestra for mastering purposes. I now produce two orchestras for FM broadcast, but these are very simple mic layouts. I would like to know how producers set up full-scale symphonic recordings. For example, what types of mics are used for the various sections and how they are placed most effectively. I realize that every company probably has a different method for any given piece of music, but perhaps you could give me a basic format.

Also, could you please describe how all the mics are kept in phase during recording and how the panpots are set on the mixer during recording.

I'm in your eternal debt for these answers! Thank you!

—Patrick J. Suarez
Miami Valley Recordings
Dayton, Oh.

Your first inquiry as to whether our method of recording a symphony orchestra is very different from yours is hard to answer. This is for two reasons. First, I don't know what "yours" is, but more importantly, there is no "ours."

In any large commercial recording company, there are several producers on staff. Each one is allowed the freedom to develop and practice whatever techniques suit his aesthetic taste and working methods. Generally speaking these methods fall into two broad categories: 1) few mics and 2) many mics.

I suspect that the first system is similar to what you already do on your FM broadcasts. My own personal preferences favor the multi-mic technique, so I will describe that in detail.

To begin with, please note that I refer to "multi-mics" *not* multi-tracks. I feel that these are two distinct issues. Naturally, an advocate of the two-mic system is going to record on two tracks. But with many mics, one has two choices: record on two tracks or record on many tracks. I do not see these last two options as two different techniques. After all, it is all going to wind up on two tracks in the end either way. The difference is simply whether the producer wants to decide on final musical balances at the recording sessions, or if he is allowed to wait till later during the mixdown session. But the basic method remains the same—using many mics to pick up all parts of the orchestra.

Let's take it section by section. First, the strings. Normally, I use seven mics as follows: two for the first violins, two for the second violins and one each for the violas, cellos, and basses. Most of the time these are AKG C-12s (these are no longer available but the AKG 414s are equivalent). Neumann U-87s are also satisfactory, but to my ear, they produce a slightly strident edge to the high strings which I must subsequently remove with equalization. The mics are usually placed about three to four feet above the player's heads.

For the woodwinds, the number of microphones varies with the size of the section. We used to use one mic when we recorded the Philadelphia Orchestra or the Cleveland Orchestra. I use two in Louisville, but recently, in a recording of "The Rite of Spring," I used eight woodwind mics. Like the strings, they are placed a few feet above the player's heads.

The brass is more simple. Sometimes, they don't need a mic at all. Generally, I use one each for the trumpets and trombones, and two for the french horns. These last are placed a few feet behind the players, facing their bells. Sometimes, I utilize a mic up over the bell of the tuba, but this really depends on the kind of writing that has been done for the instrument.

The percussion is usually picked up on two or three mics but recently on a recording of Varese's "Ionisation" I used a total of fourteen percussion mics.

In addition to the above, I mic harps (sometimes as many as three, each with their own mic), celeste, piano, timpani, chorus (usually with three or four mics, but in "Carmine Burana" we used eight) and soloists (vocal or instrumental).

Your last two questions are somewhat puzzling. "Phase" is probably the most misunderstood word and concept in our business. All the mics are *not* kept in phase! Sure the polarities of the output transformers are wired so that a similar acoustic pressure impulse will produce similar voltage outputs in each mic—but this is nearly a meaningless convention. With one hundred plus players scattered all over a concert stage, the voltages that any one of them generate in twenty-five or thirty microphones can only be thought of as being randomly related in phase and the phase connections of any one microphone becomes truly insignificant. However, it should be stated that if a simple pair of mics is used for the entire orchestra, then the question of phase is critical—but then, the problem is easily solved with only two mics!

As far as panpots go, I hardly ever use them when I do a multi-track recording. The exact placement of instrument images is determined in the mixing sessions. If you wish to use several mics fed to only two tracks, you will have to make placement decisions (as well as musical balance decisions) right in the recording session. Then you *will* need to incorporate panpots into your thinking. How are they set? In whatever way produces the kind of stereo imagery you desire. Generally, it is wise to position the microphone signals to correspond with the actual seating of the orchestra.

Best of luck and I hope you do get to work on some orchestral recordings. I'd like to hear the results.

—Andrew Kazdin
Director of Masterworks, A&R Services
Columbia Records
New York, N.Y.

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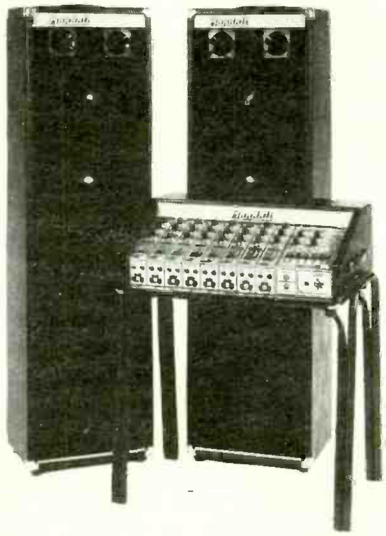
CIRCLE 76 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

RANDALL P.A. SYSTEM

Said to be a great favorite of many traveling groups, the Randall model RPA-300 public address system has eight channels capable of accepting either high- or low-Z mics. Each channel has EQ controls for highs, middles and lows, plus separate reverb and sliding pot volume controls. The master section has a five bar slide pot equalizer, master reverb and auxiliary input volume controls. Green and red LEDs show normal or overload power conditions. The system also has a gain boost and cut switch, plus a hi-F boost switch that provides an additional 10-dB boost at 10 kHz, and a low-F cut switch that reduces the 50-Hz region by 10 dB. Recommended columns have two 12-inch and two 10-inch speakers plus two piezo super horns. Self-contained, the RPA-300 system is rated to produce up to 300 watts and is claimed to be highly reliable and "almost totally free of failure due to open or short circuiting."



CIRCLE 13 ON READER SERVICE CARD

DBX OFFERS BOOM BOX

New from dbx is the model 100, colloquially named the Boom Box and more technically described as a sub-harmonic synthesizer. What it does, essentially, is generate low-frequency bass (which often has been removed deliberately from a recording). The range below 60 Hz is the one the device is concerned with, and especially the octave between 25 and 50 Hz. According to dbx, this range often is removed by mastering engineers in order to limit the depth and excursion of the record groove, a technique that can get more music on the same side at higher output levels. The Boom Box works to overcome this bass lack by using program material in the region above 60 Hz to synthesize signals an octave below. It then mixes them back into the program via the tape-monitor loop. This technique is claimed to restore the "missing bass information" without such undesirable side-effects as increased noise from turntable rumble, acoustic feedback or warped disc syndrome. The result, claims dbx, is heightened accuracy in the playback. The Boom Box also is said to permit the listener to a larger system to "physically experience the air motion created by the increased bass . . . tactile, as well as the aural sensation of being present at a live concert."

Priced at $199, the Boom Box has two controls, a bypass switch and an LED indicator.

CIRCLE 14 ON READER SERVICE CARD

AN APT FIRST PRODUCT

First product from the Apt Corporation of Cambridge, Ma. is the Holman preamplifier, said to reverse the "recent trends in preamplifier design" by being the "first unit to feature adaptablity to a wide range of system requirements without any sacrifice of sonic accuracy . . ." Its phono preamp and tone-control sections are based on new research. The design emphasizes freedom from detrimental interactions, defeatable infra- and ultrasonic filters, cross-talk-free program and recorder switching and "smart" muting of transients. The mode control is continuously variable to provide adjustment between mono, stereo and left and right mix for matching the "depth" dimension of true stereo recordings to the loudspeaker-room characteristic. Loudness compensation is claimed to be "psychoacoustically appropriate" with a special bass control that has two modes: loudness and program. Price is $447.

CIRCLE 16 ON READER SERVICE CARD

TILT THE DECK

Ruslang has introduced a deck-frame tilt feature available on its tape transport consoles. With the deck frame locked in tilt position, the recordist may view and work the deck while seated. The option permits operating a transport either flat or at some convenient angle. Other Ruslang console features include front-panel access in both horizontal and vertical positions, and a rear shelf for power supplies. The tilt feature adds $10 to the cost of any Ruslang console.

CIRCLE 17 ON READER SERVICE CARD

AUTOMATED MIXING CONSOLE

Sound Workshop of Hauppauge, N.Y. has announced its Series 1600, described as an automated mixing console and based on what the company calls "a new philosophy of console design." Using a fully modular mainframe, the console may be purchased in configurations from 12x8 up to 36x32, and any configuration may be expanded to full capability by adding sections. The automation package may be ordered with the console or it can be added later. The automation retro-fit is accomplished in two steps: one is the addition of VCA automation control cards to each input module, permitting VCA input subgrouping. Next is the addition of Sound Workshop's automation processor, which allows full level and mute automation, and which is compatible with MCI's automation system.

Instead of being designed around the standard I/O module, the new console has separate input and output modules which interface electronically and mechanically to form one unit. In addition, both the EQ and send assign matrix are separate interval subassemblies to allow ease of service as well as a choice of equalizers. Two EQs are now available—one is a 3-band, peak/dip type with four frequencies per band; the other is a 3-band parametric with a 20:1 frequency sweep and four "Q" positions per band.

Console interface is simplified by the unique design of the modular patch bay—all jacks associated with a given input/output channel are mounted on a removable PC board. Sound Workshop claims state-of-the-art circuitry throughout for "superior sonic qualities and specifications." Prices range from $10,000 to over $60,000.

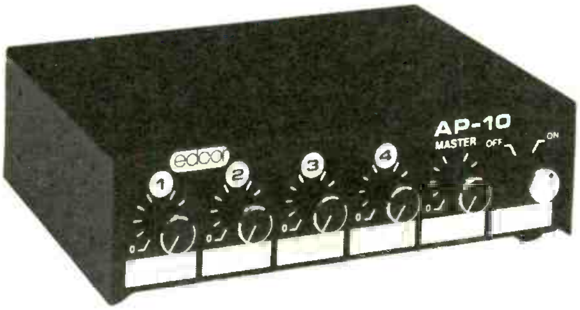
CIRCLE 19 ON READER SERVICE CARD

CROWN OFFERS FANCY RACK MOUNT

Originally intended as a demo and dealer display, the Crown 70R cabinet is now availabe for the general public. Free standing and finished in oiled walnut veneer, the 70R is 70 inches high and offers 28 vertical inches of rack mounting space including a special turntable shelf plus tape and disc storage area behind doors. The whole thing rests on precision bearing casters.

CIRCLE 18 ON READER SERVICE CARD

HEADPHONE AMPLIFIER



Designed to accept high or low level signal inputs and drive up to four sets of stereo headphones, the Edcor AP-10 headphone amplifier is rated to deliver up to 4 watts of power on each of its eight output channels. Outputs have separate low-noise amplifiers and individual gain controls, while a master gain control handles all channels. Inputs may be stereo or mono. Connectors are standard stereo phone jacks. An "info pad" provides the user with a "hands on" position to mark detail info such as gain setting, headphone assignment, etc. The AP-10 also may be used as a low-power amp to drive speakers. Self-powered, it measures 2½ by 6 by 9 inches. Two units will rack-mount side by side.

CIRCLE 8 ON READER SERVICE CARD

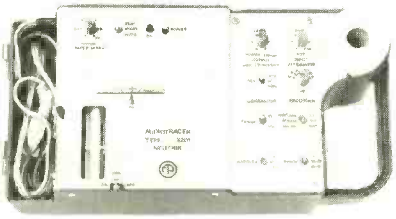
AUTOMATIC GRAPHIC EQ

Audio Developments International (ADI) of Palo Alto, California offers its Type 1500 Automatic Graphic Equalizer which features red and green LEDs above each of ten equalizer slider controls to indicate flat response when both LEDs light up. For excess energy in a given band only the red LED comes on; for too little energy, the green LED lights up. This technique is claimed to enable the user to achieve accurate sound shaping and flattening without any other test equipment and "in mere seconds." The Type 1500 is priced at $795.

CIRCLE 9 ON READER SERVICE CARD

NEW FROM NEUTRIK

From Philips Audio Video Systems Corp. comes word of two new professional products. One is the Neutrik AD-4 Analog Tapped Audio Delay Line. The AD-4 is designed for establishing a virtual sound source and for improving speech and/or music articulation in distributed-speaker sound-reinforcement installations, as well as for generating special effects and ambience (or enhancing reverb delay) in recording work. The AD-4's "bucket brigade" design employs charge-coupled devices and steep Butterworth filters that offer four discrete, time-incremented, delayed outputs. All are commonly and continuously adjustable over a 4:1 range (12.5-50 msec., 25-100 msec., 37.5-50 msec. and 50-200 msec.). Output level is independently adjustable on each. the unit's low-distortion, input-limiting amplifier has a three-position time-constant/defeat switch to suit music and speech characteristics, plus additional features including adjustable sensitivity. Dimensions are rack-mount.



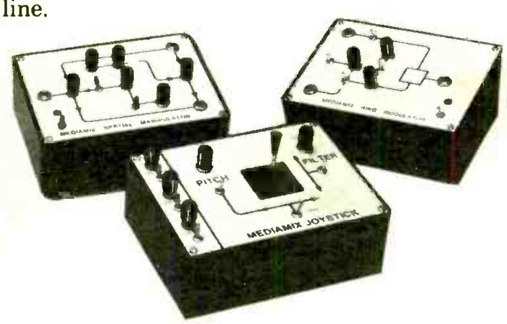
The other new item from Neutrik is their model 3201 Audiotracer. This device provides self-contained facilities for measuring and making permanent "hard copy" recordings of the level response of any audio system or device, electronic or electroacoustic. Basically, the 3201 measures and thermographically records audio frequency or time phenomena versus linear or log (dB) amplitude. Included is a voltage-controlled oscillator; a 5 Hz "warble" generator with switchable-width FM of VCO; switchable 1-kHz reference oscillator; output amplifier with RMS drive capability of 3 watts into 4.5 ohms; input amplifier with calibrated stepped and vernier attenuation; motional-feedback pen-drive amplifier and galvanometer movement with switchable range and writing speed; electronically-controlled paper drive mechanism; DC-heated pen.

Price of the delay line model AD-4 has been set at $795. Price of the Audiotracer was not available at presstime.

CIRCLE 10 ON READER SERVICE CARD

MEDIAMIX OFFERS RING MODULATOR

A new voice and instrument manipulation device called the Mediamix Ring Modulator is offered by the firm called Mediamix of Dallas, Texas. AC powered and totally self-contained, the unit features a built-in mic preamp, a variable symmetry audio oscillator (used in conjunction with mic to produce talking computers, androids, etc.), and LFO (for tremolo and stereo spatial effects—applicable, for instance, to a Rhodes stage piano), and Squaring function for synthesizer pitch doubling and stereo spatial effects and an external input enabling the user to sing along with a synthesizer, thus producing a melodic yet electronic-sounding singing or speaking voice. A kit version is priced at $85; the finished unit costs $120. A small company (so far), Mediamix also offers other specialized devices, including a stereo effects unit, a joystick for manual pitch bend on a synthesizer, and a series of add-on modifications for Oberheim, Arp and Moog synthesizers. For $2.50, the company will send you a 30-minute stereo demo tape illustrating its line.



CIRCLE 20 ON READER SERVICE CARD

SHURE ANNOUNCES "TOTAL DESIGN" PICKUP

"Trackability"—a term introduced into disc playback some years ago by Shure Brothers Inc.—receives renewed attention with the introduction of the latest of the V-15 pickups. The new model is the V-15 Type IV and it is based on new design techniques that are interrelated in a "total design" approach based on a lot of research and experimentation, which was explained to a group of invited press people (yours truly included) at a recent seminar held at Shure.

The basic concern is still the ability of the stylus to remain in contact with both walls of a record groove at the lightest possible vertical-tracking force (this defines "trackability" which Shure holds is the most important measure of overall cartridge performance). But the new techniques found in the V-15 Type IV that are integrated with each other and with the total product in an effort to further improve trackability are significant, in my view, and worth more than casual mention.

In general, disc playback is always subject to such problems as heavily modulated grooves, warpage and surface crud which can frustrate any pickup's ability to consistently track accurately. A major assault on these problems by Shure engineers is documented by a pile of engineering data and, more germane, by the new pickup itself which incorporates new design features. One is a new stylus assembly that uses a "hyperelliptical" nude diamond tip, a telescoped shank structure, a lightweight high-energy magnet and a new two-function bearing system that is independently optimized for low and high frequencies. Effective mass has been lowered, and the elongated tip-groove contact is credited with up to a 25-percent reduction in distortion vis-a-vis a conventional biradial (elliptical) stylus.

Also new is the "dynamic stabilizer"—a built in viscous-damped brush-like extension that is primarily designed to attenuate arm-cartridge resonance effects and, by resisting sudden warp-caused changes in motion, to maintain proper disc-to-pickup distance as well as vertical tracking angle and VTF. In addition to adding this kind of stability to the pickup performance, the stabilizer— consisting of over 10,000 electrically conductive fibers— also removes static electricity charges; it sweeps the groove ahead of the stylus to minimize dust buildup. Finally, it safeguards the stylus from damage since, in its engaged position, it will cushion the stylus from impact if carelessly dropped onto the turntable.

Now all this is very interesting in theory—but how does the new cartridge sound when playing records? In a word, great! The bass is solid, clean and well-defined. Middles and highs are smooth, with excellent inner detailing; transient response is forceful but not exaggerated. At a list price of $150 (which includes the option of also getting a free new Shure test record), the V-15 Type IV is hardly the cheapest phono pickup around. But to my ears it sure is (pun intended) one of the very best.

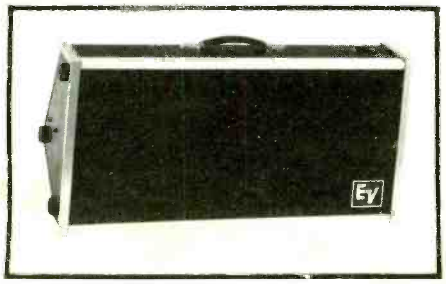
CIRCLE 12 ON READER SERVICE CARD

MUSICAL NEWSICALS

By Fred Ridder

MONITOR SPEAKERS

Electro-Voice has introduced a pair of new floor monitor designs. The Model FM12-2 is a two-way system using an EVM12L woofer and a T35 tweeter, while the FM12-3 is a three-way system which adds a Thiele-alignment-vented cone-type midrange driver to the EVM12L and T35. Both models are rated at 100 watts RMS, but to protect the T35 from receiving



that full amount of power and possibly blowing out, E-V came out with a special High-Frequency Auto-Limiting circuit which reduces the drive to the tweeter to safe levels in very high power situations. Both models are in wedge-shaped enclosures which allow 30-, 60- or 90-degree orientation for versatility.

CIRCLE 5 ON READER SERVICE CARD

From Soundcannon Industries comes an interesting solution to related problems of inaudibility and excessive leakage in stage monitoring systems. The Soundcannon SM112 looks rather like an oversized floodlight, but in reality it is a highly directional speaker system using a 12-inch extended-range speaker and a piezo-electric super horn in a special, hooded enclosure. The sound field from the Soundcannon is limited to about 20 degrees so that the sound can be focused exactly where needed, and the unit's small size and 360 degree swivel base allow for mounting the speaker much closer to the performer's ears and thereby reducing the amount of power ultimately needed for a given sound level. The net result of the design is said to be as much as 4 dB more sound at the performer's ears without feedback. The 12 incher is rated at 70 watts RMS, but the amplifier must be limited to 35 *volts* RMS to protect the piezo-electric tweeter.

CIRCLE 6 ON READER SERVICE CARD

MIXING CONSOLES

Biamp Systems, Inc. is now shipping their new six-, eight-, and twelve-input mixing consoles including the 12-channel Model 1282. The new Biamp mixers are notable for their use of the latest Bi-Fet operational amplifiers, which operate with lower noise and lower distortion than the more commonly used conventional op-amps, and which also feature very high slewing rates for reduced transient intermodulation (TIM) distortion. Input channels of the Biamp 1282 feature a variable 50 dB attenuator, a pre-EQ/pre-fader monitor send, three-band EQ, post-fader reverb/effects send (reverb is internal), slide fader and stereo panpot. The Master section includes master faders for left, right and monitor outputs each with its own variable-cutoff, 18 dB/octave bass filter, level control and panpot for the reverb and effects returns, and VU meters for the left and right main outputs. Other features of the Biamp series include input transformers on each mic input, transformerless balanced main outputs and direct channel outputs for multi-channel recording which can be taken before or after the channel equalizer. Several options are available including a 48-jack patchbay to facilitate submixing or interconnection of several Biamp Systems mixers.

CIRCLE 7 ON READER SERVICE CARD

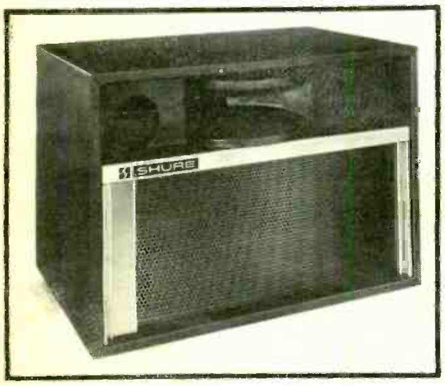
A totally modular audio mixing system is the new offering from Custom Audio Electronics. The XPC-16 series is totally modular and does not use a mainframe or mother board; the modules simply connect side-to-side, allowing the mixer to be expanded or the configuration to be changed at will. Two different input modules are available in the system. The XPC-16 input module features mic pad (20 dB), phase reverse switch, continuously variable mic gain (40 dB range), break switch for interrupting the signal path to patch external effects devices, LED level indicator, assign switches for eight stereo submaster buses, panpot, two echo/cue sends with selectable pickup points

(post-preamp, post-EQ or post-fader), low-frequency and high-frequency equalizers with four selectable frequencies each and a solo switch. The XPC-16P input module has all the same features as the XPC-16 but adds a third echo send and uses a three-band parametric equalizer. Also available as an option in the XPC-16P module is a limiter circuit built into the preamp; continuously variable controls are provided for threshold, compression ratio and release time. A variety of submaster, master and special function modules are available to allow a wide variety of overall configurations to suit particular applications.

CIRCLE 3 ON READER SERVICE CARD

SOUND REINFORCEMENT

Shure Brothers, Inc. has added two new speaker systems with identical performance characteristics to their line of sound reinforcement equipment.



The two models do, however, differ in packaging; the SR112 is designed for permanent installation while the SR116 is a portable model with a carrying handle and extra protection for the drivers. The units are very compact, measuring only 15¾″ high x 23″ wide x 15″ deep, yet they are designed to handle up to 100 watts of continuous power. Wide frequency response and maximum *broadband* efficiency were the primary objectives in the design of the new models, and the results are quite impressive for so compact a system: sound pressure output of 95.5 dB at 4 feet with a 1-watt input, and virtually flat response from 45 Hz to 16 kHz without a response-correcting equalizer. The systems each use a pair of heavy-duty 8-inch bass speakers in a bass reflex enclosure and a high frequency compression driver with a 120 degree radial horn, and either model weighs in at under forty pounds.

CIRCLE 2 ON READER SERVICE CARD

One of the most interesting new products presently available is the Schaffer-Vega Diversity System, a radio transmitter/receiver system designed for "wireless" musical instrument amplification systems. Wireless systems are becoming increasingly popular with performers because they offer virtually unlimited freedom of movement to musicians previously tethered by their cords, and also because the instrument is no longer electrically connected to a high-powered amplifier which under fault conditions could deliver lethal voltages to the musician. Wireless systems have traditionally suffered from several shortcomings, however —they are prone to fading in and out as the performer and his transmitter move in relation to the receiver; they are susceptible to picking up interfering signals from radio and TV transmissions (particularly from police and emergency communications bands and the now-overcrowded citizens band); and they tend to be noisy relative to normal instrument amplification systems. Designer Ken Schaffer based his system on the Vega Diversity system which was introduced in 1976 by Vega Division of Cetec Corp., who have long been leaders in the wireless microphone field. Their Diversity System uses two antennas separated from each other by at least two wavelengths so that it is virtually certain that at least one of them will be receiving an adequate radio signal at any given instant. Each antenna feeds its own tuner and demodulator in the Diversity Receiver, and a Diversity Switching circuit chooses between the two demodulated audio signals to give the strongest audio output at all times, virtually eliminating fade-outs. The switching circuit matches both the amplitude and phase of the two signals for undetectable switching and is designed to be "smart" enough to switch only when the alternate audio output is audibly better than the signal already in use rather than switching solely on the basis of RF signal strength. Vega's years of experience have led to a sophisticated, crystal-controlled receiver circuit with multiple helical resonaror filters to virtually eliminate frequency drift and interference signals. Previous wireless systems had a signal-to-noise ratio on the order of 60 dB, which is about the same as a good FM broadcast station and which was adequate for most applications; for use with a high-gain, high-power amplification set-up in a concert hall, however, 60 dB proved inadequate. Schaffer's solution to this was an effective compression/expansion system which stretches the overall signal-to-noise ratio to something in excess of 85 dB while maintaining frequency response to 15 kHz and total harmonic distortion under 1%. Oh, yes, one other specification you are undoubtedly interested in—the price of all this high technology. The base price for a single transmitter/diversity receiver system is $3300. The State of the Art is never cheap.

CIRCLE 4 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

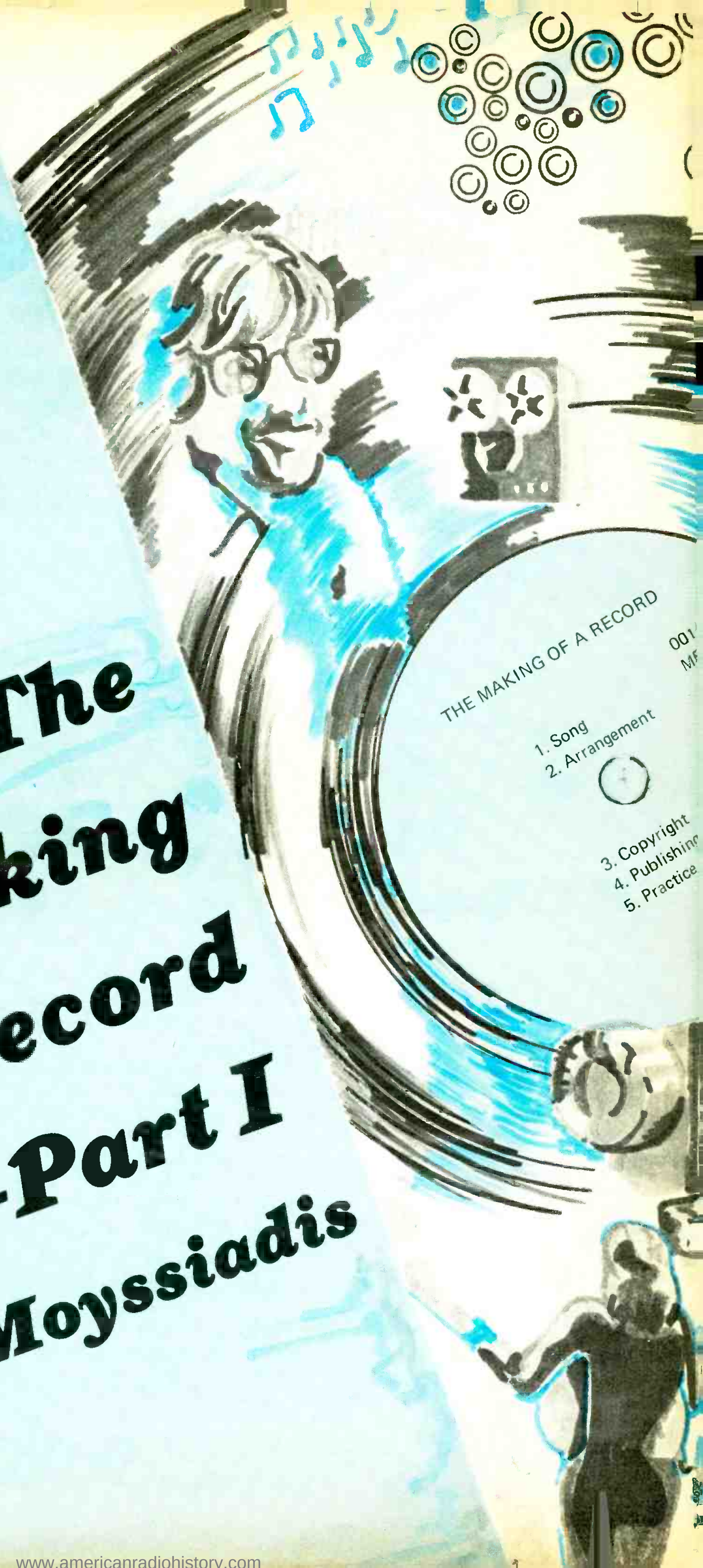
Multivox/Sorkin Music Co. has carried a line of high-quality, moderately priced amplifiers under the Premier label for many years, and they recently announced the new lineup of Premier amps which features several portable models using advanced "power-pak" circuitry. The Premier P-50 is a 20 watt RMS guitar amplifier with a special 10-inch speaker. Electronically, the P-50 features three inputs compen-

sated for normal, "bright," or microphone signals, volume, bass and treble controls, and a tremolo circuit with on/off switch, speed control and foot-switch jack. The same basic amp is available with a reverb unit as the model P-50R. The Model P-54B is a 20 watt RMS, 12-inch bass amplifier with three inputs, and volume, bass and treble controls, while the Model P-35 is a small guitar amp with normal and microphone inputs, volume and tone controls, and a tremolo circuit which delivers 7.5 watts RMS into its specially-designed 8-inch speaker.

CIRCLE 1 ON READER SERVICE CARD

The Making of a Record —Part I

By David Moyssiadis



After years of watching an incredible parade of musical life flow across that big glass window, one picks up on the better ways of marching along the route to a record. Probably the reason many fall down is because few realize just how complex it all is. And it has been getting more complex during the last decade.

There are several factors to be considered when making a record. First and most important is the song. Thought I was going to say the studio didn't you? But no, the song or whatever material you have is the most important. Without that you have no need for anything else, do you? Yet, you would be surprised at how many producers walk into a studio without the first idea of what they are going to record. Improvisation is great when it is called for, such as in a "live" situation, but not at an irreversible and unrelenting $2 per minute.

I'm not about to delve into the theoretical aspects of music since my index finger has spent most of its life lighting up record buttons, but common sense (which has never been all that common) will tell you that the recording studio is not the place to see if an arrangement works. At three or four cents a second you should *know* that it works.

Dollars and Steps

Unartistic as it may seem, you do have to consider the great god $. How much of him there is will determine what you do and how you go about it. Let's set up a situation as common to all situations as is possible and one that is easily adaptable. That is, producing your own record for your own (self-owned) record company. You go about it the same way you would if you were making a demo for a biggy. Whether you do it only with a self-contained group or with the "sweets" (strings & horns) depends on how big $ is. You will find musicians to be expensive. We will confine our discussions to a single, with just a few pertinent references to LPs. Perhaps this would be a good time to spread out the road map and explain where we are going with all of this. Here are the steps; you need:

1. A song. You know, lyrics & melody.
2. An arrangement (how to get one).
3. Musicians (how to get them).
4. A studio (how to find one).

5. A mastering house (how to find one).
6. A plating & pressing plant (how to find one).
7. Labels (how to get them).
8. Promotion (how to get it).
9. Distribution (how to get it).
10. $ (sorry).

Step 1. This is easy. You must already have a song or you wouldn't be planning on recording one, so on to Step 2.

If you have a group you probably have worked out the arrangement, or are working on it. Keep the following in mind. Playing "live" on stage in front of an audience and playing in a studio are two totally different animals with only a song title in common. What works on stage probably will not work on a record because very often, on stage you have a visual show to fill in the weak areas. While we're on this topic let's grope through the psychology of it. Just why are the two areas—studio and stage—rarely compatible? First, you can get away with almost anything on stage . . . short of exposing yourself. Mistakes, false starts, even wrong words to a song. Why? Because everyone is having a good time, even you, or you wouldn't be there and you know it. You have an audience and unless you are bombing miserably, they are with you. Empathy I think it's called. So if you have made them happy, made them laugh a little, they will forgive you for even a glaringly bad note or dropped line if you are smooth enough to turn it around. You can make a dozen minor musical bloopers and get away with it because in a "live" situation most of the errors probably aren't even heard. Even if the errors are heard they only last a fraction of a second and then are gone forever. On a recording they come back to haunt you.

Be Prepared

A good experienced arranger will know what works on a record and what doesn't. So if you can't get an arranger to do the job you will have to learn from experience. Don't ask! I don't know. Of course, when you get to the studio there will be minor changes especially if you are hiring studio cats with whom you have never worked before and therefore haven't rehearsed. The point is, don't ever walk into a studio unprepared. Have the charts completely written out, or, if it's a head arrangement, know exactly what your group will do. Expect a few changes in the studio but don't plan on a major rewrite or you've just blown the lid off your budget.

Even if your funds come from a bottomless hole in the backyard, it will cost you in other ways. For one thing, there is an electric feeling at the beginning of a session with all the last minute mic touch- ups, musicians tuning up or warming up, board set-ups and track assignments. The adrenaline flow helps get the session started on good vibes. But if someone looks at the chart in front of him and says it can't work (the bomb that sends everyone for the pencils and aspirin), and the arranger has to rewrite the horn parts on the spot, then that excitement turns to anxiety and eventually to boredom. Then the good vibes are as useful as a sonic boom, 'cause they just went bye-bye. And when there are bad vibes or hassles you just can't make a record.

These bad vibes also penetrate the glass window. There is no wetter blanket than for a studio crew all set to go, having to sit idly by while the music people get their act together. The fact that they will get paid is of little consolation. Eventually ennui turns to derisive commentary and you've not only knocked the enthusiasm out of your mixer, but lost his interest and sometimes his respect. At that point, no one can give his best no matter how much he would like to.

Session Planning

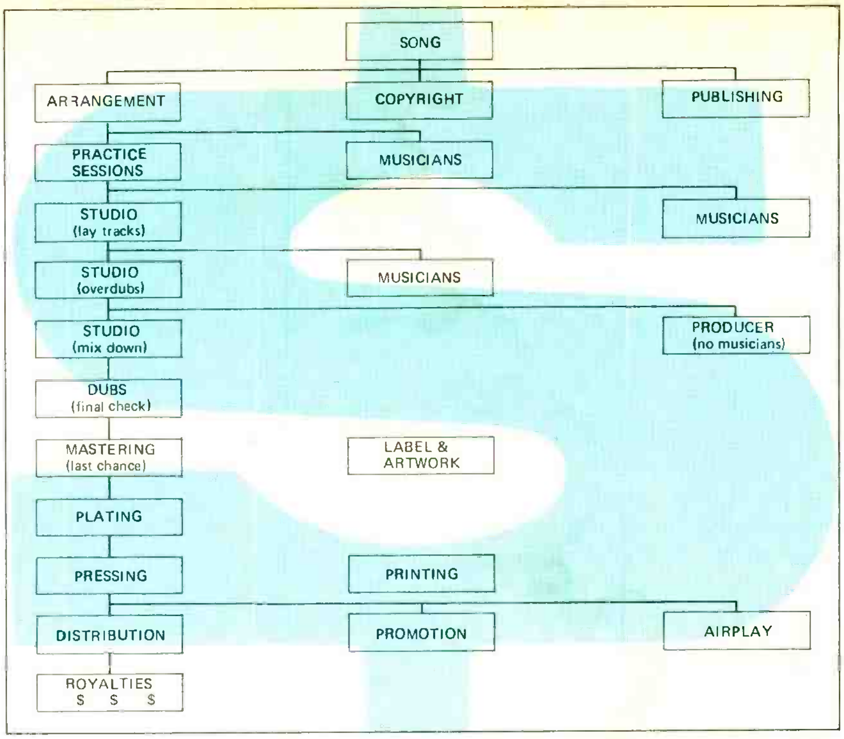
When you have your arrangement and know how many musicians you will need plan your studio time, but be flexible. The best laid plans . . . and all that. About the time you are getting your arrangement together you should also be applying for copyrights and publishing and all that legal rot.

It is suggested that sessions be limited to three or four hours each and that you go back to the studio several times on different days. This gives you time to think and get away from it. You can look at it [listen to the song] more objectively than you can during forty or fifty hour marathon sessions. I recall one time we went into our windowless cocoon and when we came out the city had gone through a snow storm, dug its way out and the snow had melted. We didn't even know it until we inquired as to why the parking lot was auctioning off our cars.

So, by now you have the music part

together on paper and should have it practiced to make sure it will work. If you've been in a studio before you have a good idea of what the differences are and what to compensate for with respect to the studio environment. If not, prepare for some surprises (next issue for that). Make sure of your basic plan, of what you want to record and of how to go about it. Make sure you've got yourself and your group together musically, *then* plan your attack. Know in advance *everything* that you want to record and that will be included in the music, down to the last tambourine, scratcher and jew's-harp. This is important; you might wind up without that essential "vibraslap" part you wanted.

Now once you have it down and are ready to go into the studio, you have to hire musicians (if you need them). You can call a bunch of guys yourself —if you know good experienced players—or you can call a contractor who will hire them for you and also handle the hassle of juggling the studio and the various musicians so that they all end up doing the same thing at the same time in the same place. For this he gets double pay.



Studio Tracking

By now you should also have picked out a studio to do your recording in. So let's see how we go about finding a studio to suit your needs. Usually you get what you pay for, but there are a few minor bargains around. Everyone dreams of recording in one of the big name studios that the superstars record in but most individuals have a hard time cracking the $200+/hr. nut. There is no doubt that you will get a technically superb recording at these places, but there are other considerations. Many super studios cater to the superstars, natch, and tend to look down on the little guys bumbling along. They don't want to waste their time on a guy who probably isn't going to make it (and the odds are against making it, if you want to be realistic). They want the big name to add to their trophy room. That is what they are geared for and one small session seems to throw the machine back into second gear.

Now on the other hand if you go to a studio suited more to your stature you will be greeted by the owner himself, treated more cordially and see people act as if they really want your business—because they do. This is not to imply that discourteous behaviour is the rule at the big places. On the contrary, the treatment will be first class, they're trained for it. But like the stewardess' smile it's painted on for the guy who can only spend a couple of thou' on a studio. They'd really rather cater to the group with a hundred kilodollars. And so would you. Anyway, if the size of the place doesn't put you off and you can hack the bread go to the big place, you can't do better. But if you want genuinely friendly service without having to leave your left arm and right leg in payment, go to a good small studio. Look for the owner-operated places. Here the owner sees what is needed equipment-wise (and everyother-wise) and does it without red tape and arguing with the accounting department about cost-effective crap. He has the biggest stake in making you happy.

Remember, some people are crazy and others don't know which end is up technically so a good small studio is hard to find. If you have friends who make good recordings and you know they don't own a printing press ask them. If not look around and go visit a few places; see if you can get the 5c tour. Then keep your eyes open. Is the receptionist efficient and not just a knockout; are last week's beer cans and pizza boxes still in the trash cans; is the console clean. Ask to see their test equipment, even if you don't know an oscillator from an oscilloscope. You may get a surprised look, but unless there is a proud display and a little *spontaneous* boasting about how well-maintained their equipment is, be wary. If there is total silence followed by excuses, apologies, and a lot of double talk about how great the equipment is and how it never needs maintenance and has stayed in spec since it came from the factory—without a clip lead in sight—run for the nearest exit. While you may not be able to hear the difference between a well-maintained recorder and one that is just barely on the outer edge of spec, it is an important consideration.

Today's professional equipment is of such high caliber that it is hard to distinguish a recorder that is in top condition from one miserably out of alignment by listening—especially with an untrained ear. But that is not the point. The reason for such high standards is not for that first generation recording, as almost any $800 amateur deck today can meet professional spec on that first recording. But, rather the overriding concern is the subsequent copies. That's where the expensive decks maintain quality above human ability to detect deterioration, and where the amateur stuff rapidly makes mincemeat of the program material. Otherwise a studio would have machines costing one-third the cost of their normal equipment outlay. Don't forget that by the time your initial recording gets down to a record, cassette or cartridge you will have gone through at *least* five or six generations!

Decisions, Decisions

After you've made as good a technical evaluation as you can then go to the standard areas of decision. Of course you want a decent rate that doesn't include a free ticket to the poorhouse. But if you have to pay more for a better place, choose that one over the $15/hr. place that looks as if it were just resurrected from the city dump, and has a mixer wearing a moth-eaten, coffee-stained T-shirt.

Also beware of the rate *structures.* Some places have favorable hourly rates, but charge to place the mic in front of you before the session and then charge to put the mic back in the corner after the session. All multiplied by how long it took the engineer to drive to work. Hang clear of a dive like that. All you should have to pay for is an hourly studio rate plus tape, plus rental on unusual musical instruments not commonly found in a studio (i.e., the Mormon Tabernacle organ, an electric bass flugelhorn or bass marimba, which the studio itself has to rent). Sometimes, use of an extraordinary piece of equipment beyond the normal studio control room contingent—such as a third 24-track—may bring a rise in the billing. But those are the exceptions. You *need* mics, recorders and equalizers, and they have to be set up before the session and torn down after the session, and a reputable studio will not charge extra for these normal and necessary tasks. *Read the rate card carefully.*

Another very important concern, is that you should have a good rapport with your mixer. You have to have a good line of communicaiton since the two of you have to function as one single unit. You're the brains and he's your hands. If right after he shakes your hand he says he's got to run out for popcorn or is "too busy to talk right now," you don't need him either.

Location is also a thing to think about. People for some reason prefer to fight city traffic, hassle with parking lot attendants and try to cram harps, vibes and Hammond organs into elevators which were obviously designed by a guy who hates music, rather than go to a quiet peaceful studio out in the suburbs, on ground level, with its own parking lot. Many attractive suburban studios have failed because of people's strange attraction to the city hustle-bustle, and I'll never know why.

There is of course the opposite extreme—the studio perched on a mountain top accessible only by helicopter, or the studio two hundred miles into the Everglades able to be found only with the aid of a jungle guide. Even that is preferable to the city, but it's awfully rough on the guy who has to haul in the special 13-foot grand piano for the channel on side B. But it's your choice, and if that's how your crew works best, then it's worth it.

Plan For Surprises

Once you find the studio for you and you're ready, it's time to book the room. Be ready with all pertinent information such as how much time you want, how may tunes you want to do, what instrumentation you are using, which mixer you want to work with and how the studio can get in touch with you (a working number or two that you *can* be reached at, not your estranged wife's old number).

It's best to plan about a month in advance so there is time to change things around without inconveniencing anyone should a snafu get loose. Remember, you, or your contractor, are trying to coordinate several musicians and a studio so that everything ends up in the same place at the same time. Have alternate plans (for the drummer who flew to the Coast the night before the session because he thought it was next week, and the conga player who woke up with a bad cold on the big day). When you book a studio for the first time figure on a little more time than you expect it will take (unless you are one of those meticulously well-organized people), because when the clock starts it seems that everything else slows down just like in that bad dream where you're running through molasses. You will also bump into a few surprises you didn't figure on.

After booking the time you will be expected to provide some sort of deposit to hold the time. Don't act like a pompous ass and get indignant over it. You've got to pay it anyway. You'd be surprised how many people book time all over the place for the same time and never show up. The deposit establishes your credibility and insures that you will arrive at the appointed time. You've got to admit that people in our business excel in flakiness. And the electric company just doesn't understand music people. Those meanies don't care that you really intended to show up but forgot. They want their money for supplying all that electricity, even if no one used it. They even get unreasonable about it and send out other mean people to shut off the electricity. So that's why you have to establish a line of credit before the studio will dispense with the deposit requirement. In this field studios have no power over a bad debt. Because chances are if you can't pay the bill the record is a bomb anyway, and no one, not even the owner, will pay to get his tapes back. So how can the studio get a bad debt back? It can't.

When the session is over pay your bill promptly. That will insure good service next time. If they have to worry about getting paid, you won't get all the amenities and enthusiastic service that Mr. Goodpay gets. Just remember that balking at bill-paying time or getting indignant over a deposit marks you as a deadbeat. Studios get stung a lot and have learned to spot quickly the losers. Don't get offended, try to understand their position.

Next issue we will talk about what happens when you finally get to the studio.

END—Part I of a three-part series

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So if you're intrigued by quality that speaks for itself, get down to your Sony dealer and check this new cassette deck.

Before they're all spoken for.

A SESSION WITH MARS TUCK BAN

BY LARRY REE

The only way to describe a session with the Marshall Tucker Band is to use the words engineer Kurt Kinzel uses—no frills and "live." The proverbial bottom line is: it works.

The distance between Macon, Georgia and Hollywood, California cannot even be measured in miles; they are *worlds* apart in many ways, but the guys in the Marshall Tucker Band have made the trip from Macon to Hollywood to work on their new album.

After six Gold and two Platinum albums, all of which were recorded in Macon at Capricorn Records, the band has decided to shake things up a little bit. They have a new producer, Stewart Levine, and are recording at Criteria in Miami, Fla. and Hollywood Sound, Los Angeles, Ca. Kurt Kinzel, who has engineered their previous four records, has handled the Miami sessions and is co-engineering the overdubs and mixing with Rik Pekkonen.

Marshall Tucker fans know that the band consists of: Toy Caldwell (lead guitar, steel guitar and vocals); Tommy Caldwell (bass and vocals); Doug Gray (lead vocals); George McCorkle (guitars); Jerry Eubanks (horns and vocals); and Paul Riddle (drums).

They met Kurt when he was working for the Record Plant recording studio in Sausalito, Ca. He had engineered a "live" KSAN broadcast and the band had been impressed with the sound. About two years later Kurt packed up and moved to Macon. His laid-back nature seems a contrast with that of producer Stewart Levine's.

High-energy Person

In the fifteen minutes I had to speak alone with Stewart, I found out more than I'd ever have to know to write a book about Tucker. Levine is a high-energy person, no doubt about it. He's constantly in motion. He also has impressive credits, most notably the Crusaders. His background seems to be that of one heavily into R&B and Jazz-oriented groups. Nevertheless, I came to realize that Levine is just right for the job. He has always been knocked out by the group's concerts, beginning with the early years when they opened the Allman Brothers' shows.

The relationship began one day when Stewart received a call from the group. Soon after he was off to Spartanburg, S.C., where the group lives. After spending three days with each member, trading ideas and getting to

know everyone, things began to gel. Levine with his "live" Jazz approach to recording was in alignment with Marshall Tucker's "live get-down-and-boogie" recording technique, and his background also would allow him to better communicate with Jerry [Eubanks, horns] and his jazzy horn lines.

Session Spontaneity

Now anyone who has ever been to a Marshall Tucker concert knows that this group can really put out on stage. The basis for their albums has always been play it and record it the way it goes down with a minimum of overdubbing. For the new album, the band is recording away from their homebase at Capricorn for the first time. They chose Studio C at Criteria in Miami.

The band has spent five or six days rehearsing the seven new songs; they do not rehearse the solos. The reason is simple. They don't want Toy, for example, to get locked in to a certain solo lick that would not add to the spontaneity of the session. Remember, everything is "live." All the guitar solos, all the horn solos, all the lead vocals are performed in the room. I frankly haven't heard of a band with the guts to try this approach in a long time, and I have to respect them for it. Stewart says that the new album will be about 80% "live," with the overdubbing mostly for adding emphasis wherever needed.

The seven songs were recorded in three working days of six-seven hours each. They did move into Studio A for the final day because the Bee Gees were booked in. For just such emergencies Stewart carries a tape of familiar material along with him, and he consequently is able to note the characteristics in each studio.

There is a certain irony in having these two groups recording in the same studio complex, because the Bee Gees rely heavily on a loop with a drum lick on it to set the tempo, and painstakingly add the other instruments, until they have a complete song. Tucker doesn't work that way; [they feel] the results would be too sterile.

Mics and Baffles

Kurt fills me in on the mic set-up at Criteria. Paul's drums have eleven microphones around them. The list is as follows:

Bass drum (batter side)—Beyer 101
Bass drum (inside)—Sony C500
Snare drum—Shure 57
Hi-Hat—KM-84
Shell Tom—U-87
Floor Tom—U-87
Floor Tom two—U-87
Overhead Left—AKG 414EB
Overhead Right—AKG 414EB
Ride cymbal—AKG 452EB
Swish cymbal—AKG 414EB

Paul is set in a corner near the back of the room. There is a hardwood floor in the corner and a canopy over the top. He has admired Kinzel's drum sound ever since the KSAN broadcasts. The philosophy remains straightahead.

"If you want a big drum sound," says Paul, "use BIG DRUMS! I can't see taking a big set and padding it down to sound like cardboard."

Kurt keeps track 1 & 2 for the two kick drum mics; 3 & 4 are for drums left and right; snare is on 5; and hi-hat stays separate on track 6.

For Tommy Caldwell's bass, the studio setup is exactly the same as his stage rig. Kurt positions Caldwell's amp in an isolation booth that is then sealed from floor to ceiling. The bass is taken direct, and also miked with a Sennheiser 441.

Lead guitarist Toy Caldwell's amp is baffled and picked up in stereo with side by side mics, a 421 [Sennheiser] and a Beyer 201. In an adjacent baffled area, one with a hardwood floor, George McCorkle's rhythm guitar is miked with a U-87. Jerry Eubanks is across the room, and he plays into a KM-86 [Neumann], through an 1176 limiter.

This arrangement pretty much leaves the room open for Doug Gray's vocals. Doug sings into an 86 with an 1176 on it. There is a room U-87 which gets recorded for some additional ambience if it is needed.

If there is an acoustic guitar, it is taken direct from a pickup and miked with a 414 with a Pultec MEQ on it.

The console is a 20-input desk built by MCI, but it has been highly modified. Kurt records Tucker 24-track at 30 ips. He uses Dolbys on all tracks except the drum, vocal and acoustic guitar tracks, because he feels you lose some "realism" if they are Dolbyed.

Hollywood Overdubbing

We are overdubbing in Studio A at Hollywood Sound. The finishing touches have to be added before mixing can begin. For this band, this simply means an additional rhythm instrument here or a harmony line there. George is out in the studio tuning his guitar for a rhythm overdub on "I'll Be Loving You," and the other members of the band and producer Levine are throwing a few ideas at him. He plugs into the Strobotuner, leans over to me and says, "I listen to what everybody tells me, and then I do what I want."

Co-engineer Rik Pekkonen wants to take the instrument direct, so George sits down behind the console with his guitar. Rik brings the fader up, George checks his tuning again, and the tape starts rolling. He adds a driving *chicka-chicka* part that gets Paul banging on the console and Toy jumping around waving his arms in the air. Soon the entire band is on its feet, supporting George's playing. There is applause when he's through. Stewart loves it. Doug yells, "Hey Kool George, KG. Maybe we ought to change your name to 'One Take George.' "

Happy about the strength he's added to the track, OTG heads out to put his guitar away and says, "People get too serious about makin' records. It oughta' be fun."

The playback is heard on a set of Auratones. An infectious opening riff sets you up for a locomotive of energy that stays up there until the last bar.

Next up is "Love Is A Mystery." George smiles and says, "This song has a real sleazy feel to it." Rik sets up the Caldwell brothers for bass and guitar ODs by putting an AKG C-12 on Toy's amp and a U-87 on Tommy's bass amp. Stewart is all smiles. "You guys are smellin' the end of this album," he says. Tommy and Toy are getting the sounds they want from their amps. Tommy adds some reverb on his Fender. They run through the song one time in order to work out their parts. Kurt patches in some 1176 limiters.

Stewart decides that all that is needed in certain sections is a little strength on the bottom. The two tracks are panned extreme left and right in the monitors, and the song starts. They lay out on the opening section and then add a six-note unison part. Stewart is thinking about saving this until the second chorus, but the guys are playing it straight through so that he will have it on tape and can add it when he needs it in the mix. The second take is the keeper. The band hears it back a few times, discussing when to bring it in. Tommy opens his notepad and compares notes with Stewart on what else has to be put on the songs before mixing starts tomorrow. There's not much left to do.

The plan for the mixing session follows the same pattern as the tracking and overdub sessions: straightahead.

Drag Racing Fanatics

It's a beautiful Sunday afternoon and some of the guys have headed over to Pasadena to race at a miniature Grand Prix track that is set up out there. The band has a couple of drag racing fanatics in it, and (I think) George holds some national speed record. (After all, South Carolina is the drag racing center of the world.)

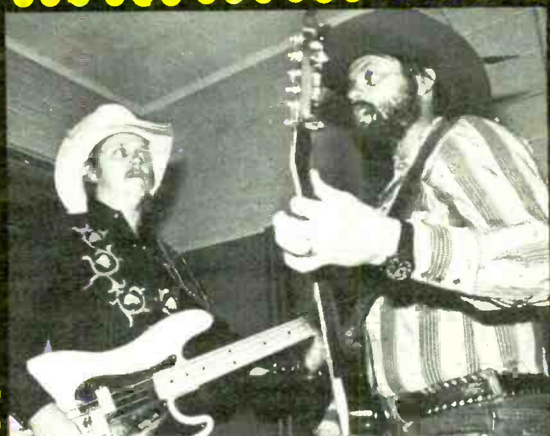
Toy tells a story about the time he was in the service and was hitching along a deserted road. He heard some crazy person chewing through his gears and then saw a '54 Ford tearing up the road in his (Toy's) direction. Toy stuck out his thumb, but was thinking, "I sure hope this nut doesn't stop." The "nut" did, and Toy got in, somewhat apprehensive. Well, this kid had a huge mother engine in the damn car and didn't even have a shift lever on the bare transmission. He was shifting with a pair of vise grips clamped on the side of the housing. Toy got home in record time. There are hundreds of these crazies down in S.C. Two of them are sitting here in Studio B as Rik begins to set up a mix on the board.

Simple Stuff

The console, like the one in A is very simple. The band likes it that way. George: "Yeah, it don't have none of that *weird* stuff on it." It has faders, API 550 EQ, track and echo assignment thumbwheel switches and echo send. Other outboard equipment that Kurt has at his shoulder includes a Cooper Time Cube, an Eventide phaser, an Altec Program Eq and a Lexicon digital delay unit. None of this is used on the Marshall Tucker album. A 3M 79 24-track and sister 2-track sit at the back of the control room. We are mixing on the Altec 604s but switching to a pair of bookshelf sized Mitsubishi speakers that are propped up on the console. Rik is handling most of the mixing today. He has worked at Hollywood Sound on many albums, and he and Stewart have recorded many tracks together.

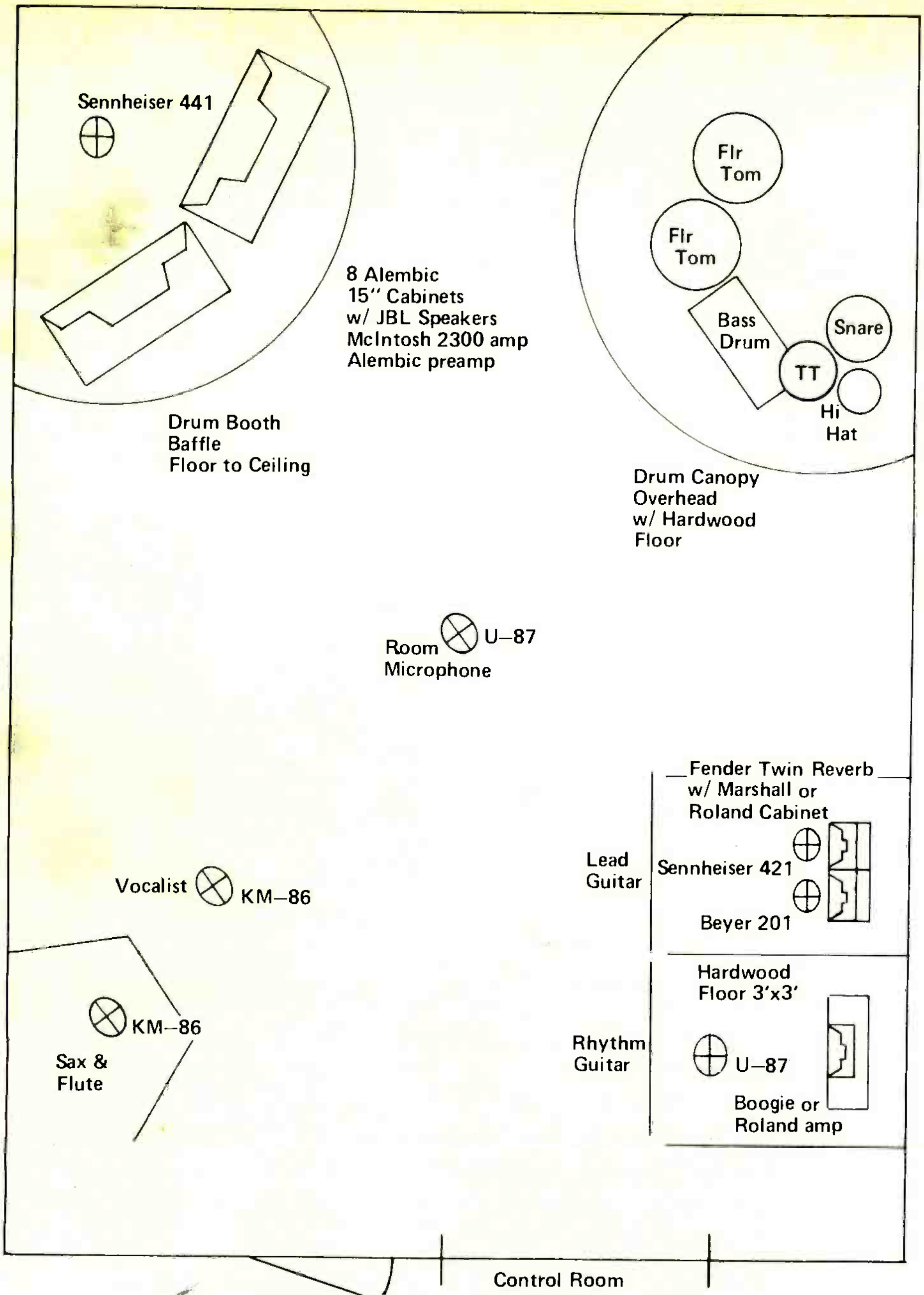
Tommy Caldwell (left) and George McCorkle taking a break.

Engineer Kurt Kinzel contemplating his next move at the board.



Tommy Caldwell (L) and Toy Caldwell working on a passage.

Photos by Herb Kossover



The set-up in Studio C of Criteria Recording Studios in Miami, Florida where Capricorn recording artists, The Marshall Tucker Band chose to record their latest group effort. It was the first time they had recorded away from their home base in Macon.

Love Is First

The schedule is: three songs mixed on Saturday, three on Sunday and one on Monday. First up is "Love Is A Mystery."

Pekkonen brings up the drum faders and supplies a bit of "10 K" to add a little crispness to the cymbals. He cranks in some "5 K" and "400 Hz" on the snare for bite and punch. Setting the echo return with the snare track (he has a "live" chamber at his disposal) he brings up the bass guitar and balances it to the kick drum. Once he is satisfied with those instruments on both sets of monitors, Toy's stereo guitar is blended with some echo. An acoustic guitar to the right, sax in the center, echo on the tamborine, and then, some decisions have to be made as far as building the track.

Stewart hears the song through a few times, and decides to keep the bass and guitar ODs out until the second chorus. Everyone gets in on the process. This song is unusual for Marshall Tucker in that it has about six tracks of overdubs. This mix will be the hardest one on the album. The saxophone solo is balanced so that it will slide right into the following guitar solo before anyone realizes that it is a new instrument.

The song opens with a bare guitar riff. There is an overdubbed line here because the original was really gritty sounding, but the band decides to keep it [the original] because there is something about the "feel" that they like. And that is what is important here.

With Stewart, Rik and Kurt on the board, the mix is finished after three or four tries. Everyone hears the entire song on both speaker systems before the final nod is given. Doug has a great vocal on this song and the harmonies are used very effectively in certain lines.

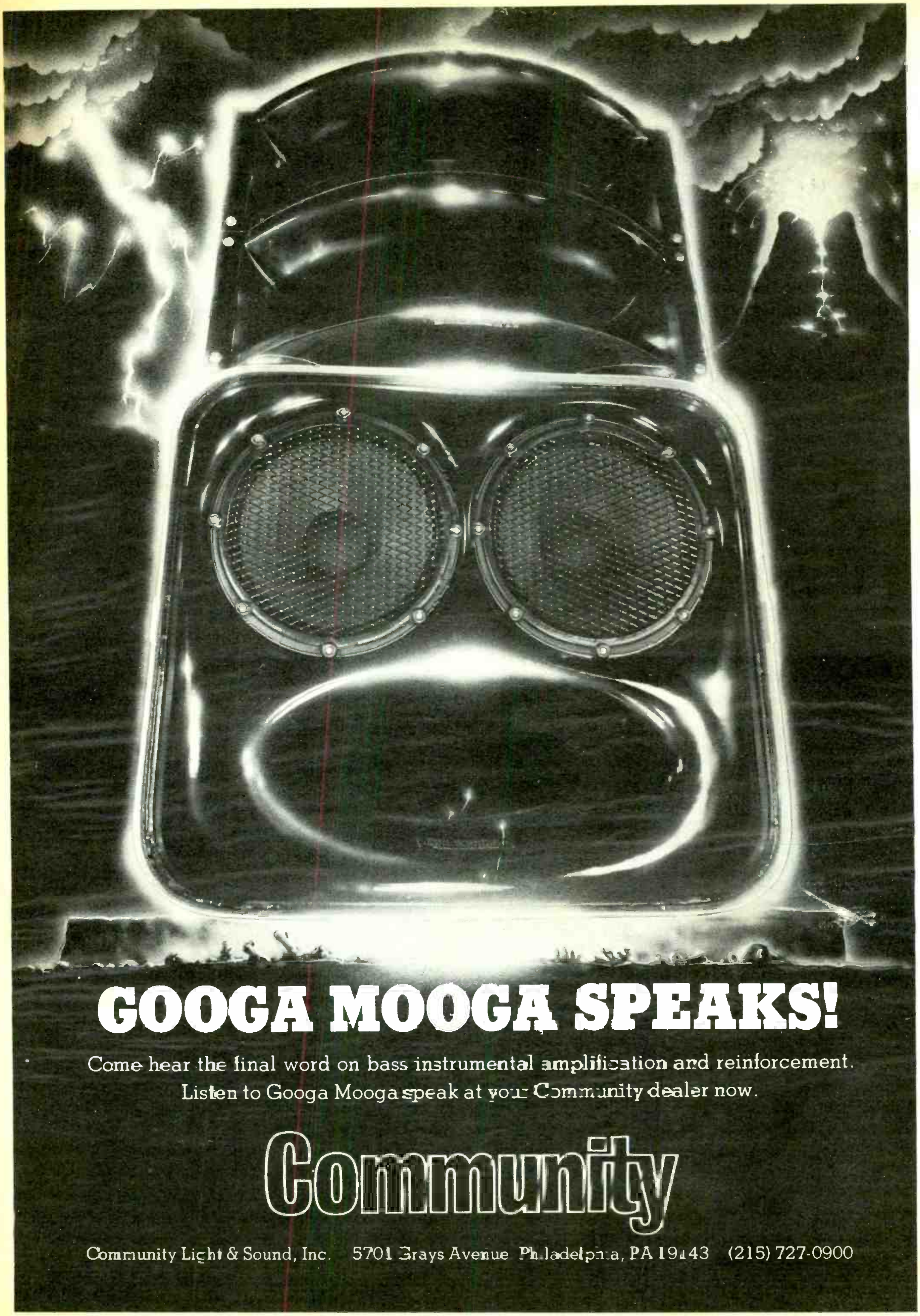
Everyone takes a break to clear his head out and grab something to eat. Rik tells a story about a well-known San Francisco acid-rock group that went to Mexico to do its album at a small 8-track studio that nobody was familiar with. When the mixes were completed, the group's engineer asked for the 8-track masters and was told that the tape they had used was the only reel the studio owned, and the studio needed it for the next group.

Playin' What They Like

If only one feeling comes across to the reader in this article, I hope it will be the sense of energy and pride that the Marshall Tucker Band has in its music. It is a good feeling to be playing what you enjoy and have the public start to pick up on it. It is a form of acknowledgement that you are doing something worthwhile.

These guys have a lot of guts to change producers and switch to two new studios during a string of albums that is all gold, but in a sense, the changes don't mean a thing. The Tucker band is the same; they just keep on playing what they like to play. No digital delays, phasers, flangers, Super Power Noobies or what have you. Straightahead, high-energy, Southern boogie travelin' music. They never went after that "single"-type record, although they did have a hit with "Heard It In A Love Song."

If their fans can pull a hit 45 off an album and expand the Tucker legions once, they can do it again. I think there are a lot of people out there who are going to be turned on to these guys when this album comes out.



CIRCLE 86 ON READER SERVICE CARD

REVERB ECHO DELAY

By Peter Weiss

part II

Last month, in Part I, we began to discuss the terminology and general applications of reverberation devices and effects. We left off with descriptions and simplified block diagrams of stereo echo systems. The block diagrams are simplified in that they show only basic signal paths for illustrative purposes, rather than actual electrical interconnections between units. Also, signal conditioning devices in the echo system, such as filters and equalizers, were omitted for clarity, and because these devices deserve separate attention. Now that we are familiar with basic artificial echo systems, we can proceed to discuss these additional features.

Recalling Definitions

In order to understand the need for filtering the echo send signal, let us consider what happens to sound waves in an actual room, and recall our definition of decay time. We defined decay time as the time it takes for a reverberating signal of a specific frequency to decrease in intensity to a level of 60 dB below that of the original signal. In general the decay time of a room (at a given frequency) depends on room size and the acoustical nature (absorptive or reflective) of the room surfaces.

Most room surface materials, even plaster, are more absorptive (less reflective) at increasing frequencies. This means that after each successive reflection of a complex sound wave (i.e., containing signals at many different frequencies, in varying proportions) from the surfaces of a large room, the high-frequency content of the reverberant sound will be smaller and smaller. Another way of saying this is that the high frequency portions of the reverberant signal are diminished in intensity to 60 dB below their original level much sooner than the low frequency portions. This shorter decay time for high frequencies creates an overall effect of "warming" or boosting the bass portion of the reverberant sound. Acoustical echo chambers reproduce these aural effects, as do most high-quality plate and spring units. Figure 1 shows the decay time vs. frequency characteristics of a high-quality spring echo chamber. Note that the difference in decay time between signals at 200 and 2000 cycles can be as great as two seconds.

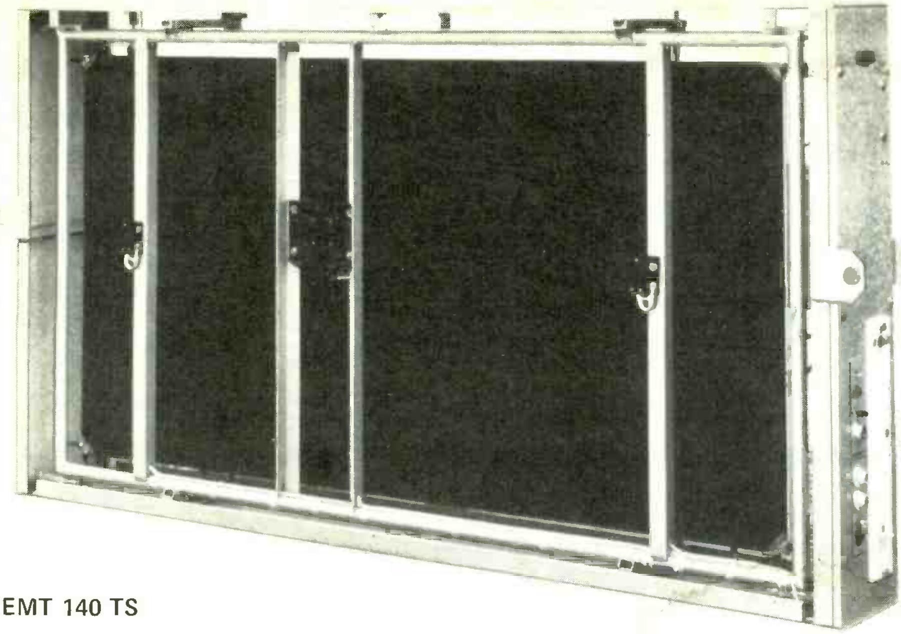
If, in all cases, echo chambers were used to recreate the ambience of a large room, then the extended, enhanced bass characteristics described would be necessary and welcome. There would be no need for any signal conditioning devices in the echo system. However, in actual multitrack remixing of many contemporary music productions echo is considered and used as a special effect, rather than as an aid to realism. ("Special effects" as used here refers to any means used to significantly alter the sound of a particular mix element. As applied to echo chambers, any use other than re-creating room ambience qualifies as a special effect.) In most multi-track applications enhanced bass response becomes a problem.

Bass Response Problems

In order to define this problem clearly, let us consider the amount of low-frequency signal present in a typical rock production—electric bass, mixed "up front," bass drum likewise, tom-toms, full-range keyboards, etc. Now let us imagine that we are engineering the remix of such a production, and we are asked by the producer to "add echo" to the snare drum track, or perhaps to the tracks containing the snare and other elements of the drum set. We turn off all the other tracks, open the echo returns to normal settings, and begin to crank open the echo send control on the drum track(s). If we are using a good quality echo chamber, we should hear a rich, reverberant, and generally very impressive sound. However, as we begin to mix in the remaining tracks, the reverberation seems less and less impressive, and finally almost disappears. The immedate impulse is to turn up the echo send control(s) on the drum track(s). Unfortunately, this causes the VU meter monitoring the echo send level to hit the pin with every snare drum beat, and we still do not hear enough echo. Okay, then let's turn up the echo *return*. We do this and discover that the bass portion of our mix is now indistinct and muddy. Also, the VU meters monitoring the stereo output level of the console are in the red a little more than we would like. We could turn down the stereo master fader to correct the level, but this would not help the bottom end.

The cause of all these problems is the bass response of the echo chamber.

The chamber is returning a more or less full-spectrum signal to the stereo mix, but we don't *need* the full spectrum. The only portion we need (and can hear distinctly) is the upper half (mid and high frequencies). The low frequencies coming from the chamber add to the stereo meter reading, but contribute very little to what ... r, except the muddiness ... before. What we do h... portion of our mi... the low-freque... return sig... two co... qu...



EMT 140 TS

of one is increased, the other is totally overpowered.)

Since the low-frequency portion of the echo signal (in our example, at least) is so troublesome, some means must be found to control or eliminate it. The best way to overcome these difficulties is to prevent the low frequencies from getting to the echo chamber at all. This is simply accomplished by placing a low- frequency cut-off filter in the echo send line.

Signal Conditioning

Very often conditioning of the echo return signal is required. The reason may be either a further need to overcome masking effects, or to emphasize a particular frequency range in order to produce special effects. For these applications a program equalizer of moderate flexibility can be inserted in the echo return line.

The various electro-mechanical echo chambers available on the market pro-ide different degrees of input and out-l conditioning. The very 140 TS plate units provide no integral filters or equalizers and require the user to supply these if deemed necessary. The Orban/Parasound spring echo chamber has low and midrange equalization (boost or cut) at the input to each of the unit's two independent echo channels. There is a "digital echo chamber" available from Quad/Eight that is reported to recreate room acoustics with amazing realism. It is assumed that this realism includes the extended bass response already discussed, since the front panel controls include a low-frequency cut-off filter. Several of the spring-type echo chambers available include a fixed-time delay circuit which performs the function of the external delay devices described in Part I. Using one of these units would free a delay device for other applications.

"When Do I Use It?"

Now that our understanding of echo chambers and their associated equipment is a bit more complete, the only real question remaining concerning their use is, "When?" As mentioned in the beginning of Part I, artificial echo is sometimes used to re-create the effect of an instrument, voice or entire group performing in a "live" (reverberant) environment. This use of echo chambers occurs most often in the recording and mixing of "serious" music. Whether or not this effect is desired in a given situation is a decision that rests with the person or persons ultimately responsible for the quality of the production. In most professional situations the responsible parties are the A & R person and the engineer. Often these two roles are assumed by a producer/engineer, and sometimes by performers.

The use of echo chambers to create special effects is also a production/engineering decision, and is probably the most commonly encountered use of these devices in contemporary music productions. Since echo chambers are designed for a more or less limited purpose the scope of special effects possible with them is also limited. Actually, by our definition, "echoing" only certain tracks in a mix is in itself a special effect, since in an actual room all elements of a performance would contribute to the reverberation, not just selected ones.

Other special effects involving echo chambers are usually created on the spot, and tend to be pretty subtle. For example, when a true stereo echo system, as described in Part I, is not available, but two or more single-channel echo chambers are on hand, it is possible to have a track and its returned echo at opposite sides or at the same location in the stereo spread. Any panning scheme between these extremes is also possible. With more than one single-channel echo chamber it is possible (assuming the chambers have variable decay times—the decay time of an echo chamber is usually rated at

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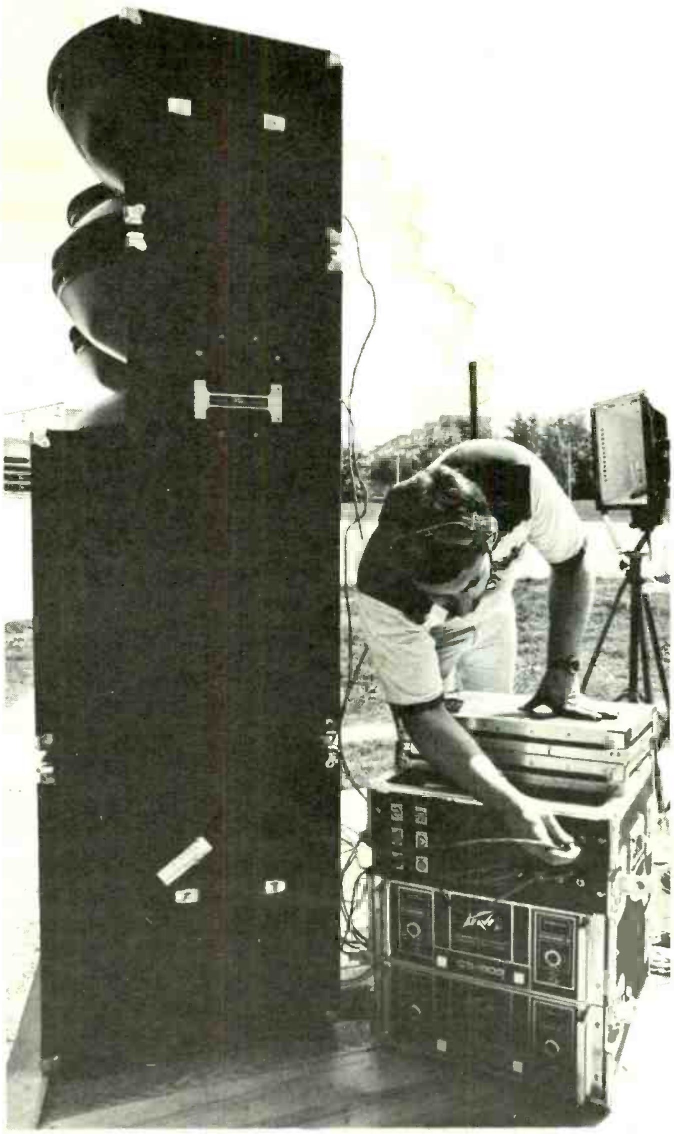
Dirt Band sound man Gary Mullen recalls, "One of the problems we faced was severe drops in voltage. At times we were running on voltages as low as 80 volts. I can't tell you how or why, but the equipment kept on working. Not only was it loud, but through the wonders of bi-amping, it was crystal clear. In the five shows at the bicycle track, the system was left on the stage each night and two nights brought enough rain to float a barge. Each time we uncovered it for a show it worked great,...the tour was a total success!"

The folks at Peavey appreciate the Dirt Band's confidence in our equipment. We're proud to have had a part in bringing a piece of the U.S.A. to the U.S.S.R.



"The system was set up with FH-1 bass cabinets stacked two high with two MF1-X horns on top of each stack and two stacks on each side of the stage. It looked pretty small but the system totally covered the area with no dead spots and enough acoustic power to make it loud enough to wake the dead!"

Gary Mullen
Dirt Band sound man

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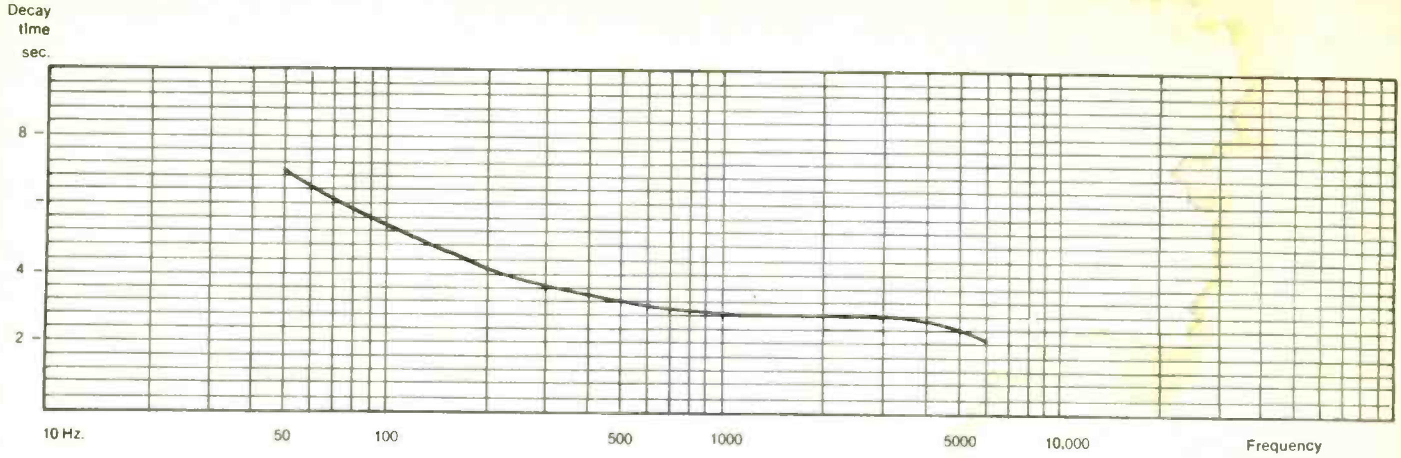


Fig. 1: The frequency vs. decay time characteristic of the AKG BX-10E spring reverb unit at a nominal decay time setting of 3.5 seconds.

a signal frequency of 1000 cycles. The actual decay time at other frequencies can be determined from graphs such as those in Figure 1) to have a long decay time for an electric organ, and a separate chamber with a shorter decay time for percussion tracks.

Sometimes echo chamber special effects can be emphasized to the point of becoming a major musical element of a mix. Examples of this kind of treatment can be found on Simon and Garfunkel's "Bridge Over Troubled Water," "The Boxer," and other of their recordings. The "exploding drum" effect heard on these productions was created by producer/ engineer Roy Halee, using only echo chambers and compression [*For more on Halee's techniques, see* MR*'s Oct. '77 issue—Ed.*] The resulting sound was then placed prominently in the mix.

Delay

In Part I we discussed the use of delay devices (tape & digital) in conjunction with echo chambers as an aid to achieving natural-sounding echo. Now we will re-examine the different types of delay devices and explore their uses in creating various effects. We will divide our discussion into two parts, based on the kinds of effects produced by delay devices and the types of interconnection schemes required to produce these effects.

Delay Without Feedback

(See Figure 2) Delay without feedback can be achieved with either tape or digital delay. The signal from the track to which the effect is to be applied is split, generally at a point after [illegible] equalizer and before the associated fader. Part of the signal travels the normal signal path and the other portion is sent to the delay device. If a delay device with only one output is used this output is patched to an available mixing fader and placed in the desired left-to-right perspective in the stereo spread. Other outputs, if available and needed, are treated separately. The stereo placement of the delayed signals may coincide with that of the original signal, or the signals may be panned to different stereo locations. Each of the signals, original and delayed, is available for any processing which may be required, such as further equalization, limiting, or adding echo to some or all of the signals.

The precise nature of the effects created in the above set-up will depend on the actual delay time available. For tape delay on a 7½/15 ips deck, the times are (approximately) 260 and 130 milliseconds, respectively. These times are a bit long for many purposes. For example, a vocal track mixed with a 260 ms.-delayed version of itself will become unintelligible if the delayed signal is mixed loud enough to be clearly heard. The original and delayed signals overlap and mask each other. Even a 130 ms. delay may be too long to use effectively for vocals, but may

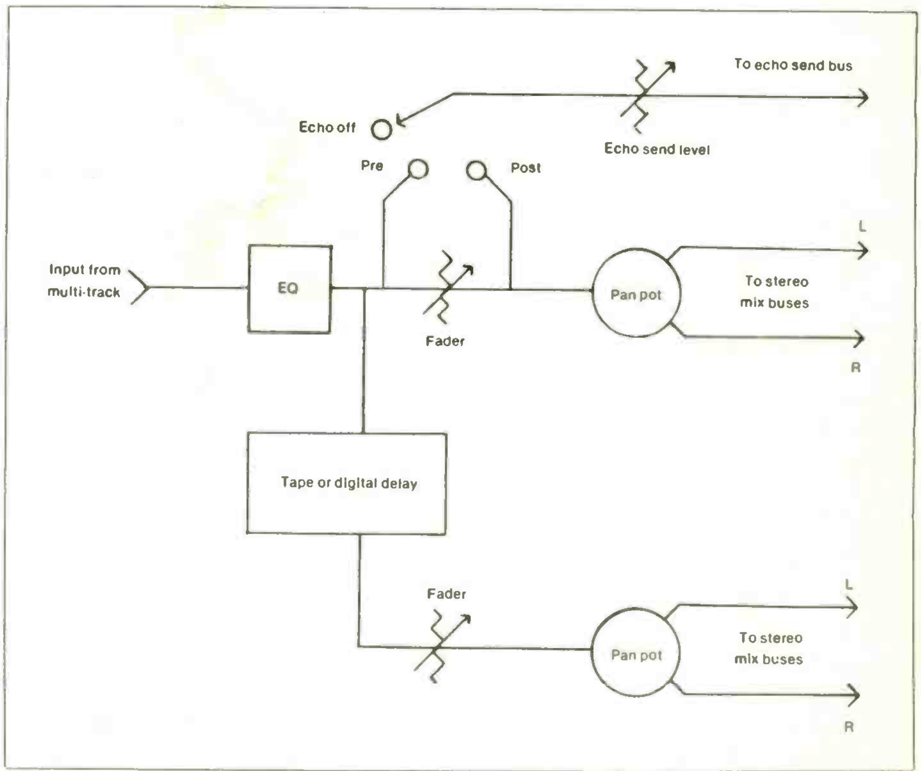
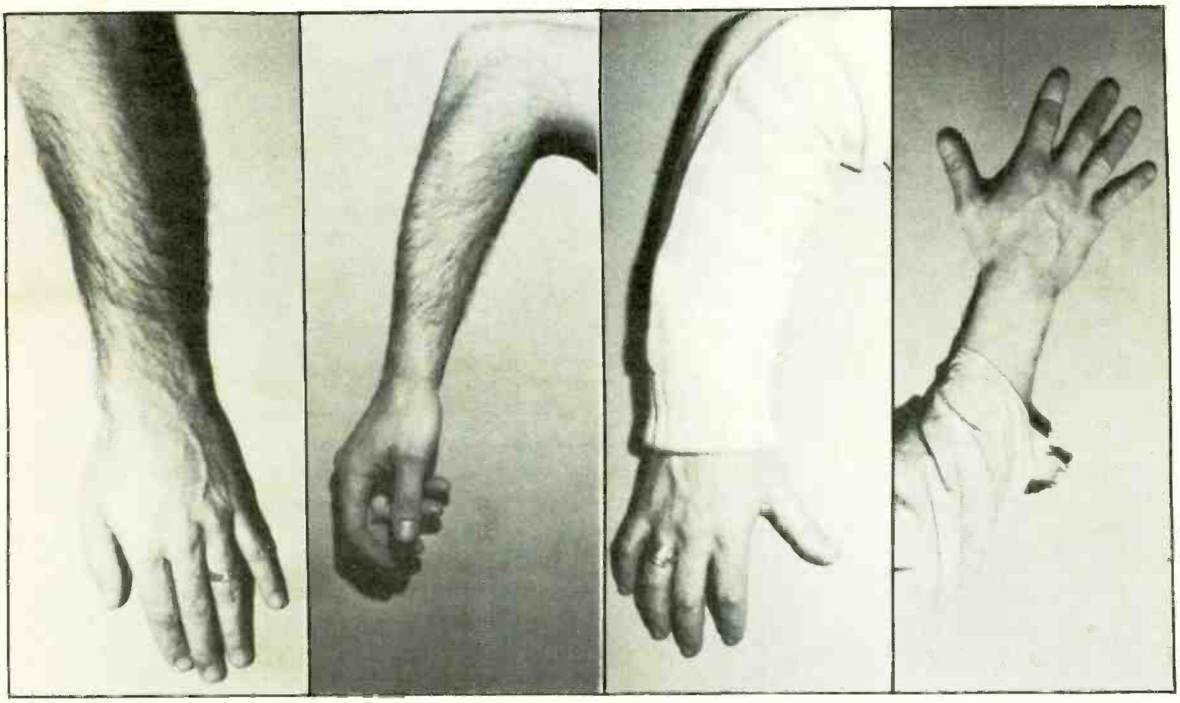


Fig. 2: Delay without feedback.

Sound Workshop introduces its arms.



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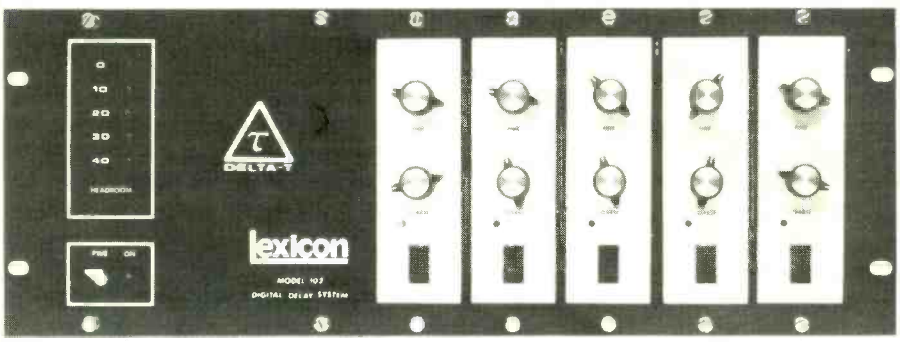
be suitable for other mix elements. A tape deck running at 30 ips, producing a delay of 65 ms. is probably more generally useful for tape delay without feedback.

Digital delay devices offer much more versatillity than tape delays. Most digital delay devices have more than one output, with variable delay times for each output. The following is a partial but representative list of currently available digital delay devices, with the delay range and number of outputs available with each.

MXR Digital Delay—single channel input and output .08 to 320 ms. delay range.

Lexicon delta-T—single channel input, up to 5 outputs 5 to 40 ms. delay per output.

Eventide 1745M—single channel input, two independently adjustable outputs. .02 to 320 ms. delay range, expandable to 640 ms. with narrowed frequency response.



Selecting Delay Times

The selection of delay time for a no-feedback delay effect should be based on the aesthetic needs of the production, but perhaps we can offer some guidance. Delay times in the 10-50 ms. range produce a "doubling" effect, which sounds almost as if the delayed signal were a separately *performed* version of the original track. Delay times in the 20-150 ms. range are used when the delay device is to be incorporated into a complete echo system as described in Part I. Delay devices with variable delay times and multiple outputs also find wide application in sound reinforcement work, where they can be used to synchronize speakers that are at different distances from the stage with the direct sound and action from the stage.

Story Time

If the reader is willing to suffer a short digression, some insight may be gained into how and why certain special effects are created in the control room. If we think about it, even on our favorite album there may be a cut that is our least favorite, one that is a musical outcast for any number of reasons. Well, performers and producers often have "least favorite" cuts on albums they create. These are the cuts that don't really move anybody but end up on the album anyway.

This writer, while engaged in the remix of just such a gem, was asked to "come up with something that'll help that guitar solo stand out." The solo was tonally and dynamically pretty tame, so the only recourse was to use some sort of special effect. On this particular album, delay and other reverberation effects had already been applied to other solo instruments, and when applied to the guitar solo in question, didn't help. As a result (probably) of a combination of sleep deprivation and caffeine overdose a variation on straight 15 ips tape delay without feedback was suggested by this writer. A small lump of masking tape was fixed to the delay tape machine capstan, causing it to flutter the tape speed at a high rate. This delayed, warbling, growling guitar, when mixed with the original guitar track during the solo, certainly was attention-getting, and a mix containing this effect ended up on the album. Unfortunately, as is the case with many effects, indeed whole mixes on occasion, that are created under conditions as described above, that particular album cut and effect are *still* held in low esteem by the people responsible for their creation.

Delay With Feedback

When a portion of the delayed output signal from either a tape or digital delay is fed back to the input of the same device, the effect created is one we have defined (in Part I) as "reverb,"

or slap echo. This effect can be heard on many rock productions of 1950's vintage and is characterized by a series of low-level, closely-spaced repetitions of the last note or word of a phrase. The effect is most noticeable (as is the case with practically all reverberation effects) in relatively sparse places in the music, where the repetitions can be heard. Moderate amounts of "reverb," sometimes in conjunction with echo, adds extra fullness and resonance to a mix element.

In order to understand how this effect is created, let us trace a signal through a reverb set-up. The signal from the track we wish to enhance with this effect is split, as before, with a portion following the normal signal path, and another portion sent to the delay device. The delayed output is then assigned to a mixing fader. This time, however, a portion of the delayed output signal is split off after this fader and sent back to the input of the delay device. (Some delay devices feature an integral feedback loop with a front panel mix control. If such a unit is used for reverb, no external feedback arrangement is required.)

The reverb effect is created when the delayed signal is sent back to the delay device to be delayed over and over. Now, instead of a single repetition, as when using delay without feedback, we have multiple repetitions, evenly spaced in time. If we wish to maintain the volume level of the repetitions we must have a feedback loop with unity gain, or no loss in level. (Gain is the ratio of output to input level. Unity gain means that the output and input levels are identical.) The fader to which the output of the delay device is connected in the description above serves to reduce the feedback loop gain below unity, resulting in a fall-off in volume for each successive repetition. If we were to have a gain greater than unity in the feedback loop, the system would begin to run away and overload, since each repetition would now be louder than its predecessor. These effects can be verified by experimentation with a tape deck with separate record and playback heads, and a mixing facility.

In the most widely used version of the slap-echo effect, the repetitions diminish in volume and are equally spaced in time. The time interval between repetitions depends on, and is equal to, the delay time selected on the delay device. Once again, it is worth mentioning that tape machines provide a very limited choice of times, corresponding to the tape speeds available. With variable delay time digital devices it becomes possible to vary the time interval between repetitions to complement or coincide with rhythmic patterns in the music. Also, some units provide a means of sweeping the delay time between preset values.

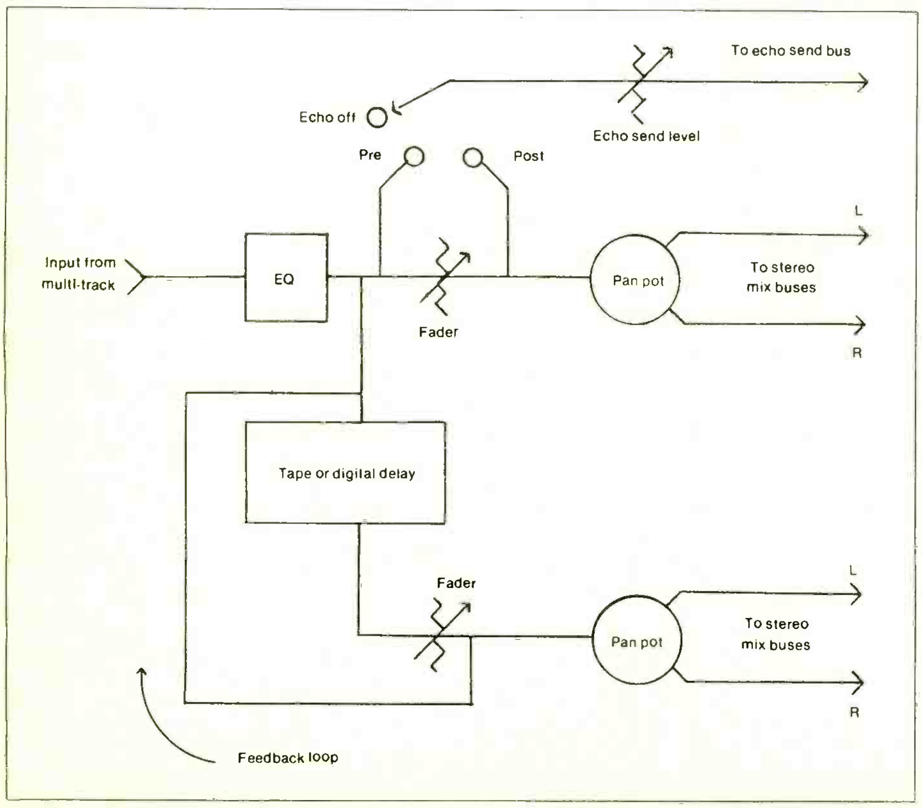


Fig. 3: Delay with feedback or "reverb."

Unique Device

A discussion of delay devices and reverb would not be complete without mention of a unique device first encountered by this writer at the studio complex of a major record company. The unit has no specific name, but it is or was manufactured by the Audio Instrument Company of New York. The machine consists of a deck plate, ¼-inch tape guides, capstan, puck, erase head, record head and three moveable playback heads that slide in a track. The outputs of the playback heads are combined in a single line-level output channel, and there is a feedback mixing control. A loop of ¼-inch tape, moving at 30 ips is used as the storage medium. The moveable playback heads provide the means for varying the time between repetitions. In this case the first three repetitions occur as the recorded signal reaches each of the playback heads for the first time. The aural illusions attainable with this device were intriguing enough to cause the studio staff to do some modification work on one of them. A variable speed capstan motor was installed, and a fourth playback head was added. Also, rather than leave the outputs of the playback heads to be combined within the unit, each of the four heads was fed to an individual playback preamplifier. This provided four individually controllable line-level outputs, which could then be fed back or not, as desired. Some engineers tried placing the four outputs at the "corners" of a quad remix and achieved very interesting results. The original unit, modified in 1972, is still in use.

Although our discussion of reverberation effects ("Echo," "Reverb," "Delay") has dealt with these effects separately, in many working situations all of them are put to use, sometimes all in the same mix. The choice of effects and the extent to which they are used is a decision that is unique to a mixing situation, and should be governed by the participants' collective musical and auditory tastes and judgement. It is the intent of this and the previous article to provide guidelines, but there is no substitute for hands-on and ears-on experimentation.

End of a Two-Part Series

Ambient Sound

BY LEN FELDMAN

Just for the Record

Ever since this publication first saw the light of day, the emphasis has been on tape recording. Taping is what the majority of our readers is into, whether they be professionals who earn their livings in a recording studio, musicians who use their tape equipment and its peripheral hardware as a tool to further their artistic pursuits, or serious audio hobbyists who have progressed beyond the point of passive enjoyment of their sound reproducing systems. But, when you stop to think of it, most of the music we hear originates from discs, not tape. The end goal of the recording engineer is to have his or her carefully created master tape mix become a commercially successful record. The amateur or semi-pro music group with its multi- track semi-pro tape equipment hopes that some day one of its tapes will make the transition to vinyl—and thenceforth to gold-plated fame. So, I thought that this month it might be a good idea to explore some of the problems that still beset record playing in the real world of recorded sound.

Most of the earlier treatises I have read concerning record playing deal with the laboratory, or ideal situation. Records are presumed to be perfectly flat—a presumption which anyone who regularly purchases discs from the current crop of offerings knows is far from true. Records are presumed to be free of dirt and dust and devoid of electrostatic charge or build-up—again, hardly ever the real-world case. Phono cartridges are often analyzed without regard to the interface between them and the tone arm in which they are mounted. Recommended downward tracking forces of phono cartridges are often quoted without regard to some of the external influences which might significantly alter the trackability of the stylus tip in the record groove.

The importance of these problems was highlighted in a recent technical seminar sponsored by Shure Brothers Inc., the well-known makers of phono cartridges, microphones and related electronic products. The climax of this seminar was, as you might have guessed, the introduction of a brand new cartridge (V-15 Type IV) which had, according to Shure, taken many of these real-world record playing problems into account and gone a long way towards alleviating them. Our purpose here is not to evaluate the relative success of the new pickup, but rather to focus on some of the problems themselves.

Tone Arm/Cartridge Resonance

Any tone arm and cartridge combination will be resonant at some frequency. This resonance exists because the arm and pickup assembly behaves like an effective mass that is coupled to the record groove by means of a stylus assembly with its own mass, compliance and mechanical resistance. Just as a weight hanging from a spring has a natural resonant frequency, so do the compliance of the stylus and the effective mass of the pickup/arm assembly. If you have ever read the specification sheets relating to tone arms you will know that, in most cases (and for most cartridges mounted in modern tonearms), this resonance frequency is well into the sub-audible range—usually in the range between 5 Hz and 10 Hz or so. You might at first suppose that such resonances are unimportant, since there is no musical signal content at such low frequencies (and even if there were you could not, by definition, hear such low frequencies). It can be argued, therefore, that there is nothing in the record groove which will excite the arm/cartridge combination at its resonant frequency. Well, here is where those "real world" imperfections in the record come into play.

Warped Records

The most common source of resonance excitation is that of warped records. It has been found that warps occur in a broad low-frequency spectrum extending from 0.5 to about 10 Hz. This form of excitation of resonance operates mostly in the vertical direction.

One major problem caused by the excitation of the stylus/tone arm resonance is that of mis-tracking. Assume that a tone arm has a low-frequency resonance of 5 Hz, a frequency well below the range of human hearing. If that resonance should be excited by a warped record, for example, the arm will move vertically up and down, magnified considerably by the resonance effect. This motion effectively increases and decreases the tracking force in an oscillating manner at a rate of 5 cycles per second. At the points of reduced tracking force, mistracking is more likely to occur than would be the case if no resonance existed. At points of increased tracking force, record and tip wear are accelerated.

A second problem that results from the excitation of stylus/tone arm resonance is known as the "scrubbing" effect. With the stylus effectively moving up and down with respect to the record surface, the vertical tracking angle alternates about its correct angle and this causes a significant change of *relative* speed between the stylus and the record groove with an attendant frequency modulation, or "wow" of the program material. A change of speed of stylus in the groove of as little as 0.6 cm per second (not unusual when resonance is excited) calculates out to be a pitch change of around 1.5% and is quite audible.

Another problem which can result from the low-frequency stylus/tone arm resonance is the creation of high-amplitude low-frequency signals which can overload amplifiers. This is especially significant now that so many amplifier manufacturers have extended low-frequency response of their products down to "DC" or at least well into the sub-sonic region. Such signals may also produce low-frequency high-amplitude motions in loudspeakers, which will result in Doppler distortion and possibly overload distortion in those speakers.

Static Charges on Records

To the audiophile who regularly deals with high-quality record playing equipment, static electricity charges are a never-ending nuisance. The problems are usually worse in the winter, when humidity is low, but some of the effects can be felt even in mid-summer. These effects show up principally as crackling or popping noises during record playback, brief popping sounds during arm set-down, excess stylus force due to electrostatic attraction of the cartridge to the record, dust and dirt attracted to the playing surface (which produces wear and playback noise) and, in the case of record changers, attraction of the arm to the unplayed stack which can interfere with setdown and even reduce tracking force to induce mis-tracking.

A research project conducted by Shure Brothers disclosed several startling facts concerning electrostatic charges on records. It was discovered for example that electrified records which had been rubbed with cat fur while on a turntable had negative charges of between 2000 and 4000 volts. The act of lifting such records off the turntable increased the charge to as high as 30,000 volts! Replacing these records on the turntable decreased the measured charge back down to between 2000 and 4000 volts because the electrostatic field is then concentrated between the turntable and the underside of the record. Static charges will not be discharged by pickups having a grounded metal shield and grounded stylus assemblies until a threshold voltage of around 4000 to 5000 is reached. Thus, charges below this voltage will not cause a discharge but will exert an electrostatic force on the pickup. When a record is charged above this threshold voltage, a playing will discharge it down to the threshold voltage of the pickup but not to zero. Investigations showed that a charge of 4200 volts on the surface of a record adds an extra 3/8 of a gram to the stylus force! This additional tracking force will, of course, increase record wear. And even if this added wear is not considered to be significant, it will change intended optimum tracking conditions significantly. In addition, a natural charge on most records will not be uniformly distributed over the surface of the disc so that cylical "bumps" or variations in tracking force will give rise to somewhat the same sort of "scrubbing" motion already described in connection with the arm/cartridge resonance effect.

Further experiments with household dust showed that a charge voltage of as little as 1000 to 2000 volts was enough to make fine particles adhere to the record and resist brushing or blowing, especially out of the bottom of grooves where ordinary bristles usually cannot reach.

Trackability Versus Tracking Force

It is well known that mis-tracking of heavily recorded grooves in a record can be partly or wholly eliminated by increasing tracking force. Most audiophiles are reluctant to increase tracking force to the high side of a given cartridge's tracking force range. Intuitively, they deduce that such increased tracking force will accelerate record wear and even reduce the life of the diamond tip itself. Little data has been available, however, to quantify just what the influence of increased tracking force is on the life of the diamond tip in a cartridge. In this connection, Shure Brothers presented some definitive data at last. Given a stylus tip exerting a downward tracking force of three grams, and assigning that condition a relative life of 100%, reduction of the tracking force to 1.5 grams will increase the life of the tip by 20%. A further small reduction in tracking force to one gram will, however, increase the life (relative to the 3 gram setting) by a full 70%, while further reducing the tracking force to 0.75 grams (assuming the cartridge can continue to properly track groove modulations) will increase the life of the tip by 120%—more than double its expected life using a 3 gram tracking force.

As you can see from even the few studies described here, record playing involves a good deal more than a platter spinning at uniform speed and a stylus riding in the grooves of a disc. As we said at the outset, we don't propose to spell out solutions to all these problems in this brief overview of the real world of record playing. It is clear, however, that anything you can do to reduce the effects of the tone arm/pickup resonance effects or to reduce the degrading effects of static build-up on records will bring you that much closer to the ideal record playing scheme that is often theorized in the laboratory and in technical papers but hardly ever realized in the real world of sound reproduction.

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

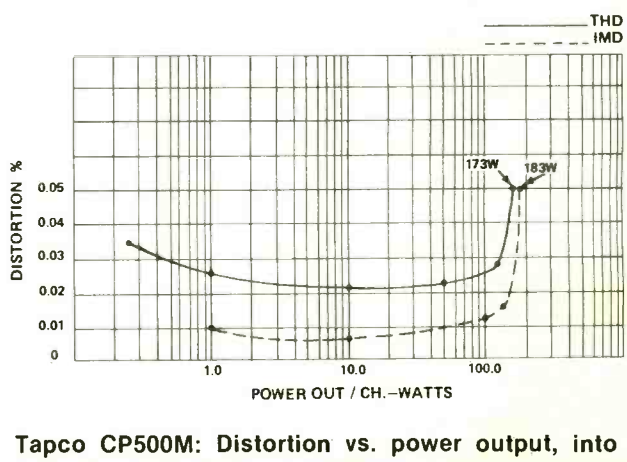
Tapco Model CP500M Power Amplifier

General Description: The Tapco CP500M is a two-channel power amplifier rated for up to 250 watts per channel into 4 ohms or 150 watts per channel into 8 ohms. It is of rack-mount dimensions and is fitted with handles and a built-in cooling fan at the rear. The front panel sports a pair of output meters calibrated in power ratings and in decibels, plus a "PowerLock" feature by means of which the operator can set the upper power limit furnished by the amplifier, separately on each channel.

The power off/on switch is a separate control at the left. Next to it is a power-on indicator. The two PowerLock controls have four positions: out; 250 watts (−0 dB); 125 watts (−3 dB); 62.5 watts (−6 dB). Next to each of these controls is an indicator that comes on when the upper limit selected is reached in use. Below each PowerLock control is a gain control. Two additional indicators show conditions of a blown fuse and of thermal protection.

Input connectors at the rear are standard ¼-inch phone jacks. Each channel has two bridged inputs, 20 K unbalanced. Below these jacks is a recessed slide switch for converting the amplifier to single-channel operation if desired. Speaker output terminals are standard binding posts, color-coded for polarity and arranged so as to provide for two-channel (stereo) output or single-channel (mono) output. The cooling fan is centered on the rear panel. Completing the picture here are a fuse-holder with a 15-amp rating and the unit's AC power cord which is fitted with a three-prong (grounding) plug.

The PowerLock circuit, which may be used to limit output power, uses output-voltage sensing which is referred to a fixed reference voltage which in turn is a function of the supply voltage itself, so that regardless

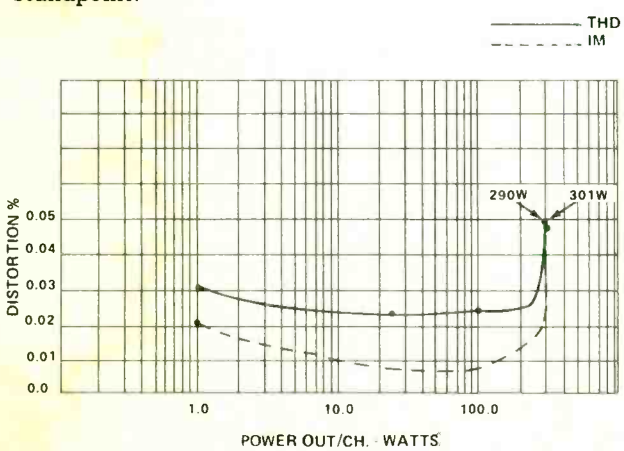


Tapco CP500M: Distortion vs. power output, into 8-ohm loads, both channels driven.

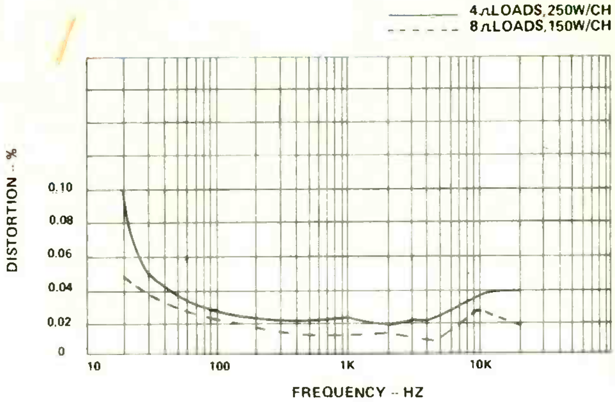
of voltage fluctuations caused by external sources, the unit simply cannot go into clipping when this circuit is employed.

The basic amplifier circuitry itself is fairly conventional in that full complementary outputs are used. The output stages do provide gain, however, unlike many other complementary designs. An input differential pair feeds signals to an emitter follower and then to a class A stage which drives the bias string. Bias sensing is accomplished at the drivers and predrivers, and it also is based on ambient temperatures sensed at the output stages. Driver and output stages constitute a Darlington configuration which has a collector-loaded output. The input stages are powered from a zener-regulated ±18 volt supply, while the output stages operate at ±62 volts filtered by a pair of 11,000 mFd capacitors.

Test Results: Ruggedly built, the Tapco CP500M performed in our tests better than its published specs would suggest, and in general shaped up as a first-rate powerhouse that can be recommended for demanding professional applications. In fact, performance in general (except of course for the audible fan noise) rivaled that normally expected of "hi fi" amplifiers. Tapco, with understandable pride, advised that they precondition their amplifiers (as per FTC requirements) with 4-ohm loads, often even using a high-frequency test signal at one-third rated power (in this case, around 85 watts) for the test. They invited us to do the same, and the results of our static measurements represent readings taken after one full hour of such preconditioning. Note that the amp just missed making its 0.05 percent rated distortion at the 20-Hz extreme for 4-ohm loads, but we would hardly fault the unit for that since the difference between 0.05 percent and 0.1 percent is rather academic from an audibility standpoint.



Tapco CP500M: Distortion vs. power output, into 4-ohm loads, both channels driven.



Tapco CP500M: Distortion vs. frequency at rated output.

General Info: Dimensions: 19 by 5¼ by 17 inches. Weight: 35 pounds. Price: $779. Also, as model CP500, less meter and error indicators, $649.

Individual Comment by L.F.: Since I test both "hi fi" amplifiers and those intended strictly for "pro" use, I usually know what differences to expect between the performance of one type and the other. While I recognize the need for extreme ruggedness in a professional power amp, I have never believed that the incorporation of such ruggedness and fail-safe features necessarily meant that the pro unit had to sacrifice performance specs that are normally expected of an "audiophile" product. Apparently Tapco agrees with me. No audio purist would ever tolerate the cooling fan noise generated by the CP500M in a home listening environment, but mounted in a rack and subjected to the kinds of environments that high power amps of this type encounter "in performance" and in sound reinforcement applications, the presence of the constantly running fan just adds that much more to the safety and long life of this ruggedly built powerhouse.

Since the amp is a wide-range unit that may find use with almost any speaker array, perhaps the smartest circuitry incorporated in it is Tapco's "PowerLock" feature, a form of precise limiter which can be preset to 62.5, 125 or 250 watts or can be turned off completely. I can't begin to guess at the number of speaker systems that are likely to be saved if this feature is used correctly.

The most impressive thing about the CP500M is its ability to deliver high orders of power output for long periods of time without thermally cycling and with no evidence of strain. Since I am an inveterate hi-fi buff, I could not resist hooking up the amp as part of a component hi-fi setup and judging it as I would judge a high-powered audiophile amp. I must confess it sounded great, exhibiting no harshness or transient distortion. Aside from the fan noise (which I didn't bother to mask or eliminate by remote placement), had I not known that its primary applications are professional, I would have been perfectly content to live with it on a more permanent basis.

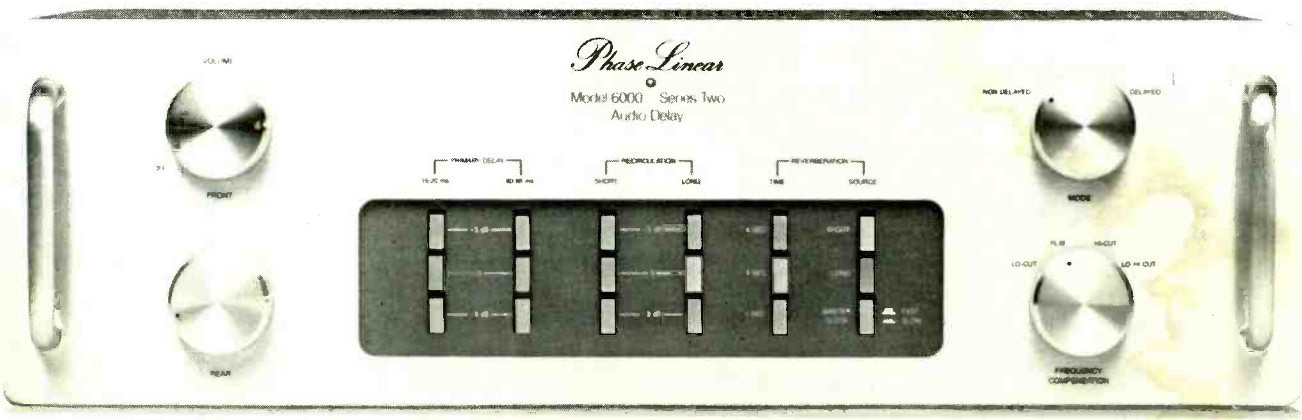
Individual Comment by N.E.: Aside from the possible annoyance of the fan noise, there is nothing about this amplifier to criticize adversely or even to question. It is a sturdy, robust piece of professional equipment and the PowerLock feature is both handy and effective. Lab measurements and listening tests confirm that the "listening quality" of the unit rivals that of "home hi-fi" units, while ruggedness is definitely in the "pro class."

It is interesting to note that Tapco also has a somewhat smaller amplifier, the model CP120 which costs $389. The CP120 has the PowerLock feature but it lacks the output meters. It also lacks the cooling fan which means no kind of noise at all. We hooked this amp into a high-quality listening system and can confirm that it provides excellent drive for monitor-quality speakers in a good-size sound room.

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power per channel, 1 kHz	250 watts into 4 ohms 150 watts into 8 ohms	290 watts into 4 ohms 173 watts into 8 ohms
Continuous power per channel 20 Hz to 20 kHz	250 watts into 4 ohms 150 watts into 8 ohms	240 watts into 4 ohms 150 watts into 8 ohms (see text)
Power bandwidth	20 Hz to 20 kHz	20 Hz to 39 kHz
Frequency response	20 Hz to 20 kHz, 0.2 dB 5 Hz to 100 kHz, −3 dB	8 Hz to 60 kHz, −1 dB 5 Hz to 110 kHz, −3 dB
Damping factor	N/A	99.2 at 8 ohms
Rated THD	0.05%	0.025%, 4 ohms 0.027%, 8 ohms
Rated IM	0.05%	0.023%, 4 ohms 0.015%, 8 ohms
Residual hum and noise	−95 dB	−98 dB
Input sensitivity	+4 dBm	+4 dBm
Power consumption	165 watts, idle 1150 watts, max	confirmed

CIRCLE 11 ON READER SERVICE CARD

Phase Linear 6000 Audio Delay



General Description: Phase Linear's model 6000 is a "bucket brigade" (analog) delay unit designed to add controlled amounts of reverb to a stereo playback system in conjunction with a second amplifier and two added speaker systems for "rear channels." It is intended for interconnection between preamp outputs and power amp inputs—it should not be used in a tape-monitor loop due to incompatibility of signal levels.

According to the instruction manual, a signal entering the 6000 "is compressed, fed through low-pass filters, entered into bucket brigade delays, low-pass filtered and made available to summing amplifiers . . ." which "can pass the signal to an output expander and frequency-contouring buffer amplifiers to the main outputs as well as set up recirculation paths. The reverberation circuits pick up primary delays, frequency-contour the signal, pass it through an electromechanical transducer, recontour the signal and make it available for output summing."

In listening terms, various effects can be created, including "enlargement" of the listening space to suit the size of the ensemble performing.

The device is dimensioned for rack mounting. Front panel controls include four large knobs and six banks

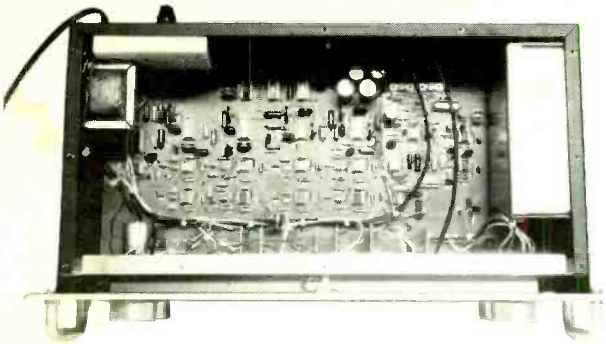


Fig 1: Internal view.

of pushbuttons. The knobs handle power off/on and the volume level of the front pair of speaker systems; rear speaker volume; mode selection (delayed or non-delayed); and frequency compensation (a filter with positions for low-cut, high-cut; high and low cut; flat).

The first group of pushbuttons control "primary delay" and select 15-20 msec. or 60-90 msec., each in terms of a 3-dB step (+3, 0, −3). The second group handles the recirculation delay, selecting short or long, also over a 6-dB range. The third group of buttons controls the reverb sound field decay. Three buttons handle time in steps of 1, 2 and 4 seconds. Three other buttons are marked "source"—of these the top two are marked short and long, and control the rate of decay of the reverb signals, The third button in this group, labeled "master clock," controls the delay time on all delay paths (primary and recirculation) in terms of "fast" or "slow." Detailed instructions for using the controls, as well as diagrams and hints on speaker location and on selecting the added speakers and amplifier are all included in the owner's manual.

The rear of the 6000 contains standard phone jacks for signal connection, including stereo pairs for delayed and front outputs, and another stereo pair for hookup to the system preamp. Also at the rear are the unit's AC line cord and a fuse holder.

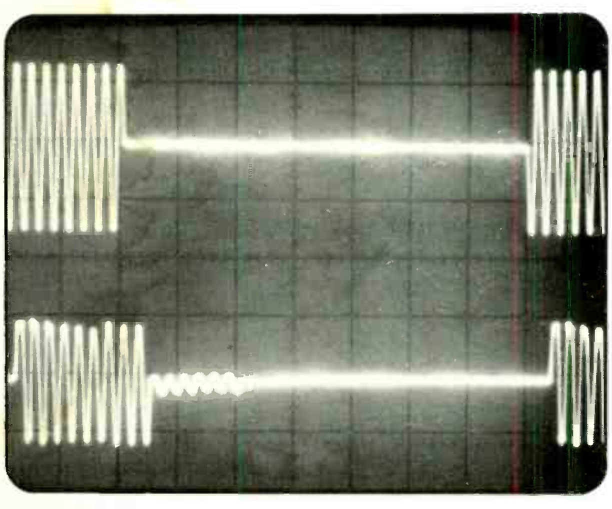


Fig. 2: Phase Linear 6000: Frequency response of delayed output using short (upper trace) or long (lower trace) primary delay settings.

While the manufacturer's avowed or main intent in offering the model 6000 is that it can be used by audiophiles who want to "increase" the size of a listening room to "concert hall proportions," MR's experience with the unit suggests that it could also be used by performers, particularly small performing groups working in small surroundings who might like to sound as though they were performing in a larger and more acoustically dramatic environment.

Test Results: Published specs for the model 6000 generally were confirmed or exceeded in MR's lab tests, and the device's action confirmed as effective in listening tests. An idea of the bandwidth of the delayed sound in both the short and long delay modes can be had by examining the photo of frequency response sweep as stored on our spectrum analyzer (Fig. 2) which shows the (short) response down by −3 dB at 2 kHz. The manufacturer does not specify dB

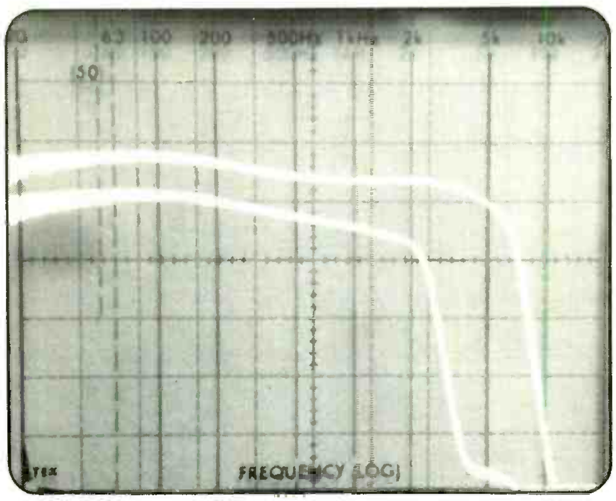


Fig. 3: Phase Linear 6000: Input burst signal (upper trace) used to produce short and long primary delay outputs (lower trace).

tolerances for either of these but the relative effects are valid enough.

Fig. 3 represents our attempt to show what happens when only a short-term delay is introduced—the upper trace is the input tone burst while the lower trace shows what comes out of the delayed output terminals. For Fig. 4 we let loose with all barrels, choosing maximum delay, recirculation and reverberation effects. Some of this is clearly evident in the 'scope photo but much of the overall "sonic event" is lost because of superimposition of the different components upon each other, since we were using a discrete sinusoidal frequency in the tone burst. The effect would have looked more impressive had we been able to photograph an actual complex musical signal on the face of the oscilloscope.

The model 6000 is truly a "second generation" device compared with earlier units we tested, and as such it merits serious consideration by audiophiles and professionals alike. Because of its extremely wide

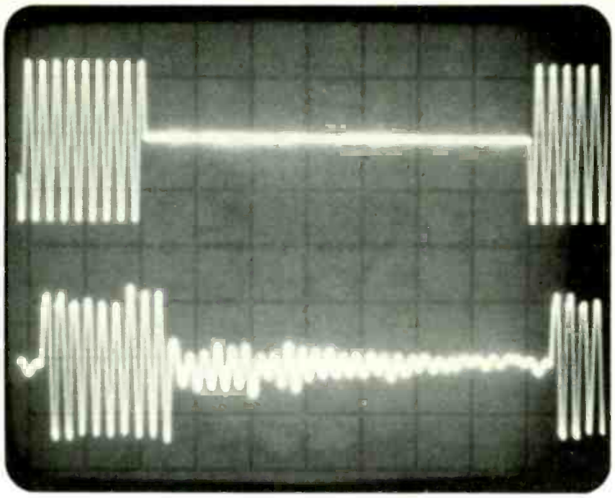


Fig. 4: Phase Linear 6000: More complex delayed signal (lower trace) is achieved by adding recirculation and reverb circuits via front panel buttons.

range of adjustment it is possible to overdo the effects, particularly when dealing with vocal selections or the spoken word. The key here is to adjust the unit so that you are not conscious of the rear speakers at all.

General Info: Dimensions are 19 inches wide; 5½ inches high; 10 inches deep. Weight is 20 pounds.

Individual Comment by N.E.: Other than a subtle vagueness in the way the pushbuttons go in and out, there seems nothing to fault in this unit which is obviously well designed and crafted for its intended uses. As to those uses—whether a device of this kind is more suited for the sound-reinforced performing situation or for the serious listener—I pass.

Individual Comment by L.F.: While checking out this handsomely styled and good performing delay unit it occurred to me that in audio it may not always pay to be "first." My earliest experience with audio delay units intended for customer hi-fi systems was with early models from Audio Pulse and Sound Concepts. At the time of their introduction they were wondrous units indeed, offering a good alternative to those who wanted to create "concert hall ambience" in their homes but who were not overly attracted by quadriphonic methods. As it happens, the Audio Pulse unit used digital technology, while the Sound Concepts model employed analog (or bucket brigade) delay.

Phase Linear's ultra-flexible 6000 went the bucket-brigade route, and the variations in control it affords are almost without number. The front panel is a model of good "human engineering"—what could turn out to be confusing controls for the uninitiated are here very clearly marked and grouped. You can introduce two degrees of initial or first delay, selecting their amplitudes over a 6-dB range. In addition the overall master control affects all delayed information, while another control may be used to adjust the undelayed or front-speaker signals. Pushbuttons introduce varying degrees of recirculated delayed signals, while two or more sets of buttons handle reverb time. We tried all the recommended settings in the manual, and they all worked to perfection. Although the delay principle employed here does in and of itself reduce bandwidth (or frequency response) of the delayed signals as it should (reflected sound in the real world of concert halls always is restricted in bandwidth relative to the primary sound), we found that when short delays were used we were aware of highs from the rear speakers. Happily, Phase Linear took that possibility into account too, and provided a high-cut filter control for just that purpose.

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Initial delay times	15 msec. and 60 msec., adjustable by "clock" control to 20 msec. and 90 msec.	Confirmed
Reverberation decay time	200 msec. to 4 seconds	Confirmed
THD, direct	< 0.1%	0.04% at 1 kHz
THD, delayed	< 0.5%	0.17% at 1 kHz; 0.6 % at 100 Hz; 0.18% at 4 kHz
Input impedance	47 K ohms	Confirmed
Maximum input level, delayed	2.5 V	2.3 V
direct	5.0 V	5.5V
Output impedance	< 5 K ohms, direct or delayed	Confirmed
Output level (max), delayed	4.0 volts	Confirmed
direct	8.0 volts	Confirmed
Output noise level, delayed	< 80 microvolts	−87 dB re 1.0 V in
direct	< 100 microvolts	−82 dB re 1.0 V in
Frequency response		
Delayed, short	40 Hz to 6 kHz	−3 dB at 6 kHz
Delayed, long	40 Hz to 2.5 kHz	−3 dB at 2 kHz
Direct	± 0.1 dB, 5 Hz to 20 kHz	Confirmed

CIRCLE 21 ON READER SERVICE CARD

Aiwa AD-6800 Cassette Recorder

General Description: The AD-6800 is a top-of-the-line cassette recorder from Aiwa (distributed in the U.S. by Meriton) that incorporates some unusual features. A front-loader, it accepts the cassette on a platform that slides automatically into a rather deep recess. Near the front of this area is an adjustment for azimuth alignment. Lifting this section permits access to the heads for cleaning. The cassette area is covered by a hinged glass-panel lid and the interior is illuminated during use.

The AD-6800 has three heads, but the "third head" is not a play or monitor head; rather it serves as a "test head" that is used for picking off test signals (400 Hz and 8 kHz) which may be recorded onto a tape by the machine as part of setting variable bias to suit different tapes. This is called by Aiwa its "flat response tuning system" (FRTS) and the procedure—detailed in the owner's manual—involves moving the input selector switch to the "test" position, using the tape selector switches to choose the general class of tape (normal/LH; FeCr; or $Cr0_2$); adjusting the azimuth control for a maximum reading on the right VU meter; adjusting a "bias fine adjustment" for a matching reading on the left VU meter.

The VU meters themselves are unusual in that they have two pointers. One shows average VU levels; the other shows peaks. The former pointer always is operative, but the peak pointer comes into play only if you switch it on via a separate button. In addition, another button permits the peak pointer to "hold" at maximum levels. On each meter the peak scale runs from −40 to +10; the VU (average) scale runs from −20 to +5. Some valuable hints on relating the two scales, and how to get the benefit of each are explained in the owner's manual.

The fast-forward button has the option of being used with the tape in contact with the head so that some sound can be heard in fast wind. Aiwa calls this a "cue" feature. The rewind button has a similar option, termed by Aiwa a "review" feature.

All controls and operating features are on the front panel. To the left of the cassette compartment are the power off/on switch, left- and right-channel microphone jacks and a stereo headphone output jack. Below the cassette compartment are the transport controls: open (the cassette door); record; rewind/review; forward (record or play); fast forward/cue; stop/eject; and pause. The pause control permits setting levels before actually recording, and this control also has a position for use with an external timer to start the machine.

A three-digit tape counter and its reset button are to the right of the cassette compartment, and just to their right is a button for memory-rewind. Below the counter is a light display; six amber LEDs that come on successively from left to right when the tape is moving at normal speed, and which flicker steadily in fast wind. To the right of this group are the two meters, with a recording indicator light between them.

The bias fine adjust knobs (one each for three major classes of tape) are located under the left meter, while the peak indicator buttons, plus a limiter button are grouped under the right meter.

Additional features are found near the bottom of the front panel. There's a DIN socket which is covered by a snap-on lid; the input selector with positions for the test signals mentioned earlier, line/DIN and microphone. The Dolby system switch has three positions: off, on and on with mpx filter. The main tape selectors come next—one each for bias and EQ. The labelling here indicates the classes of tape as well as the corresponding bias and EQ values in percentages and microseconds respectively. The output level controls are a dual concentric pair, and the record level controls are also dual concentric and larger.

The rear of the AD-6800 contains stereo pairs of pin-jacks for line input and output, another DIN socket, a fuse-holder, the power cord and a convenience AC outlet—unswitched and with a maximum rating of 500 watts. There's also a grounding terminal, and a

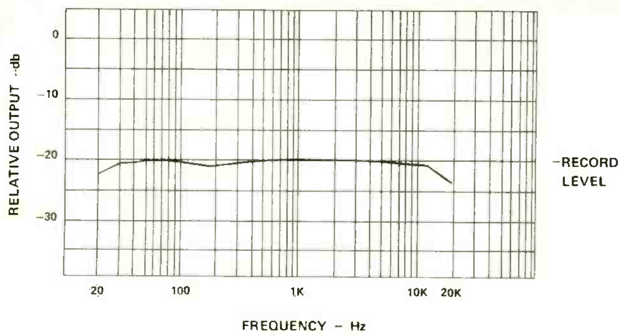


Fig. 1: Aiwa/Meriton AD-6800: Record/play response, using TDK AD tape.

"player sync" jack for connecting to a turntable that has sync output for disc dubbing.

The AD-6800 does not have built-in mixing provisions. It does have the automatic sensing for switching to CrO_2 adjustment when a CrO_2 cassette with the special groove is used. It comes with walnut sides and four "feet."

Test Results: Record/play frequency response was measured using three tape samples: TDK type AD for standard bias and 120-msec. EQ; TDK type SA for the high bias and the 70-msec. EQ settings (equivalent to CrO_2); and 3M Scotch Master III for the FeCr settings (medium bias and 70-msec. EQ). In each instance the machine's built-in test procedure was first utilized to fine-tune the bias for the tape sample. Results were plotted in Figs. 1, 2 and 3—best results were obtained with the FeCr sample tape, with response extending out to 17 kHz for the −3 dB rolloff point. This easily confirmed Aiwa's spec for FeCr tape. Our measured response for CrO_2 settings was 1 kHz shy of the 16 kHz top spec'd, but our results with standard tape exceeded the spec by 0.5 kHz. S/N figures were very good for all tapes, with the best showing made with FeCr tape. Highest headroom was achieved (relative to 0 dB readings on the meters) with the TDK-AD tape. Distortion reached 3 percent with a +9 dB record level using this tape, while with the other tapes, only a +3 or +3.5 dB recording level was possible for three percent THD.

Evidently, Aiwa's 38-pulse servo motor succeeds in

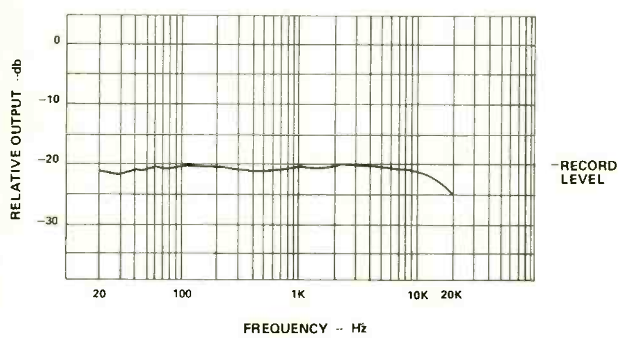


Fig. 2: Aiwa/Meriton AD-6800: Record/play response, using TDK SA tape (CrO_2 setting).

holding wow and flutter down to an impressively low 0.03 percent WRMS (as against 0.05 percent claimed); even unweighted, the wow and flutter measurement was a very low 0.08 percent. Transport controls worked smoothly and reliably. The novel cassette loading system here literally takes the cassette from your hand and positions it precisely inside the compartment. The protective door may be left open for making the azimuth adjustment prior to adjusting bias, and then closed when the tape is in motion.

The dual pointer arrangement on the meters is noteworthy. Aiwa has addressed itself to the old problem of "peak factor"—the difference between average VU readings and true peaks that occur during musical transients. The red pointer reads in VU units while the black pointer follows peaks (if you opt for this via the switching)—truly a neat solution and one that can prove effective in actual use.

General Info: 18⅞ inches wide; 6⅜ inches high; 13$\frac{13}{16}$ inches deep. Weight: 22 pounds. Price: $650.

Individual Comment by L.F.: I've lost count of the number of times I have stressed the importance of using the "right" cassette tape with a stereo cassette deck. Determining which brand and type of tape that a given manufacturer has used to calibrate his machines is not always an easy task, since most manufacturers (no doubt not wishing to offend *any* tape supplier) will

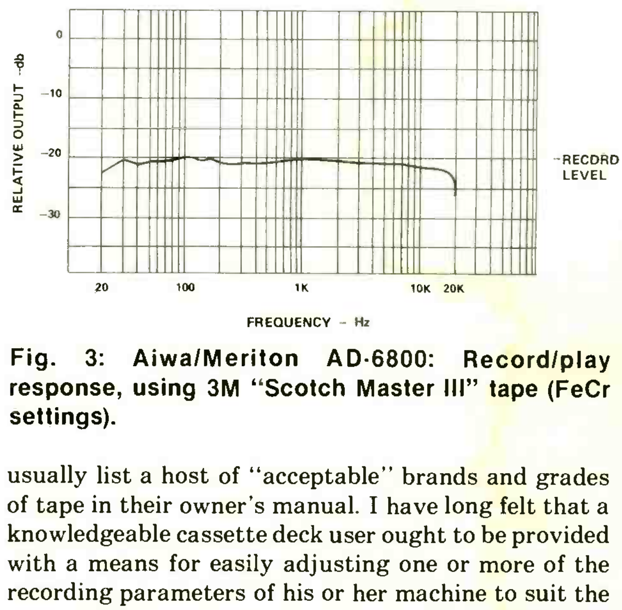


Fig. 3: Aiwa/Meriton AD-6800: Record/play response, using 3M "Scotch Master III" tape (FeCr settings).

usually list a host of "acceptable" brands and grades of tape in their owner's manual. I have long felt that a knowledgeable cassette deck user ought to be provided with a means for easily adjusting one or more of the recording parameters of his or her machine to suit the tape, rather than the other way around. Evidently Aiwa agrees with me and has come up with an elegant (if obvious) solution. The usual three switch settings for bias (standard, CrO_2 and FeCr) are augmented by three fine-tune adjustments, one for each bias setting, each of which allows the user to adjust bias by a couple of dB. The procedure, explained in the "General Description" section of this report, makes use of that third head. This head prompted Aiwa to call the deck a "three head" machine—not a falsehood in itself, but an appellation that might lead unsuspecting prospects to believe that the AD-6800 has record-monitoring capability (like some other three-headed machines) which, of course, it does not.

Nonetheless, its performance—in just about every measured specification—rivalled that of many true three-head machines, and several of the AD-6800's other noteworthy features particularly appealed to me, such as the two pointers on the meters.

Evidently, in concentrating on such excellent performance and tape transport facilities, Aiwa could not afford to provide mic/line mixing in what already is a fairly high-priced cassette deck. For those users who employ a separate multi-microphone mixer and thereby would enter the deck at line levels anyway, this should pose no serious problem. Others will have to settle for either "live" mic or line recording from other sources—but not both at the same time.

Particularly for readers of *Modern Recording*, it seems to me that the Aiwa AD-6800 answers many of the objections that semi-pro and pro recordists have voiced against the cassette format. Here is a deck that acknowledges the critical and limited dynamic range capabilities of slow-speed, narrow-tape performance, but which provides the needed metering and adjustment capabilities to let you get the very last available dB out of this tape format.

Individual Comment by N.E.: Not being privy to the economics of Aiwa's manufacturing processes, I am in no position to state whether or not they could have afforded to include mic/line mixing in the AD-6800, but it seems to me that this machine has a few flourishes that do add to its cost and which I for one could live without. One is the automated slide-in platform for cassette insertion. I have encountered no difficulties in numerous other models in which the cassette is placed manually. Moreover, here is a mechanism which could conceivably go sour after a time. I am reminded of the old engineering adage about the simpler that things can be made, the better. I also see no need for the series of lights that tell you the tape is moving. Cosmetics like this add to a machine's cost—and doesn't the regular tape counter tell you the same thing? I see no need for two DIN sockets, especially for U.S. users who probably won't even have the occasion to make use of one such socket.

The "cue" appendage and the "review" label added to the fast-forward and rewind functions are questionable—all they did in my sample was allow the tape to run at fast speeds in contact with the record/play head, so that you could hear an indistinguishable hash of squeals. This is the kind of high-frequency garbage that has been known to send nasty transients through a system. I still do not understand its real use since you cannot really distinguish actual signals this way.

On the plus side, the azimuth and fine bias adjustments are to the good, although I feel that once you have set bias via the main selector switch, the additional tweaking via the knobs is of marginal value and not the sort of thing that will make a really big difference between work recorded on this, versus recorded work done on another, competent machine. At that, I had trouble making the adjustments as spelled out in the manual; I could not always get both meters to agree. The system seemed to work best, in my tests, for standard/LH tape with which I was able to get good conformation on both meters. In any case, even without this adjustment, I found that tapes made on the AD-6800 did sound very good, but no better than tapes made on other similarly priced cassette decks. The best thing I liked about the AD-6800 was its twin pointers on the meters.

AIWA AD-6800 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, std	30 Hz to 15 kHz, –3 dB	20 Hz to 15.5 kHz, –3 dB
CrO_2	30 Hz to 16 kHz, –3 dB	20 Hz to 15 kHz, –3 dB
FeCr	30 Hz to 17 kHz, –3 dB	20 Hz to 17 kHz, –3 dB
S/N (without Dolby)	N/A	std, 54.5 dB CrO_2, 55 dB FeCr, 57 dB
S/N (with Dolby)	FeCr, 65 dB	Std, 63 dB CrO_2, 64 dB FeCr, 65 dB
THD at 0 VU	Std, 0.9%	Std, 0.9% CrO_2, 1.6% FeCr, 1.9%
Record level for 3% THD	N/A	Std, +9 CrO_2, +3.5 FeCr, +3
Line input sensitivity	50 mV	46 mV
Mic input sensitivity	0.25 mV (200 to 10 K ohms)	0.3 mV
Line output level (for 0 dB)	775 mV	730 mV
Headphone output level (at 0 dB)	2mW/8 ohms	1.25 mW (100 mV across 8 ohms)
Wow and flutter (WRMS)	0.05%	0.03%
Rewind time (C-60)	90 seconds	80 seconds

On Choosing a Mixer

By Jim Ford
and Brian Roth

A mixing console is at the heart of any recording or sound reinforcement system. Consoles come in all sizes, shapes and colors, and range in price from about $100 for the simplest four-input microphone mixer to over $100,000 for a recording console that will handle a 24-track recorder and is ready for automation. Every band that is just getting started and every small studio that is just opening its doors for business are faced with the problems of purchasing a mixer. Of course everyone is looking for a "good deal" on the price and doesn't want to buy more than he needs, but also one must beware of spending most of his money on a console that will not do the job that needs to be done. Oh, what a problem! There are so many knobs and dials and buttons and switches and meters and plugs and transistors and transformers and flashing lights and buzzers and bells and will this list never end? I think I've gotten a headache!

Well, it is truly a problem so in the next few pages we will list some of the more important points to consider when comparing mixers. The discussion that follows will be directed at mixers that can be used for small 2, 4 and 8 track recording and at mixers for sound system mixing in small clubs and auditoriums. The price range of this group will start at approximately $150 and will go to about $4,000. However, there are some units that incorporate more professional features that can cost up to $15,000. (To keep a good perspective on what we are talking about, a professional 16 input/out recording console with the minimum acceptable features and audio quality should start at $20,000 and go up from there.) Most of the mixers that are available and that fall into our category of discussion are manufactured by the following companies: Tascam, Yamaha, Uni-Sync Trouper, Sound Workshop, biamp, Shure, Tangent, Soundcraft, Tapco, Quantum, Stevenson, Dallas Music, Carvin, Peavey, RSD, and many others not included here.

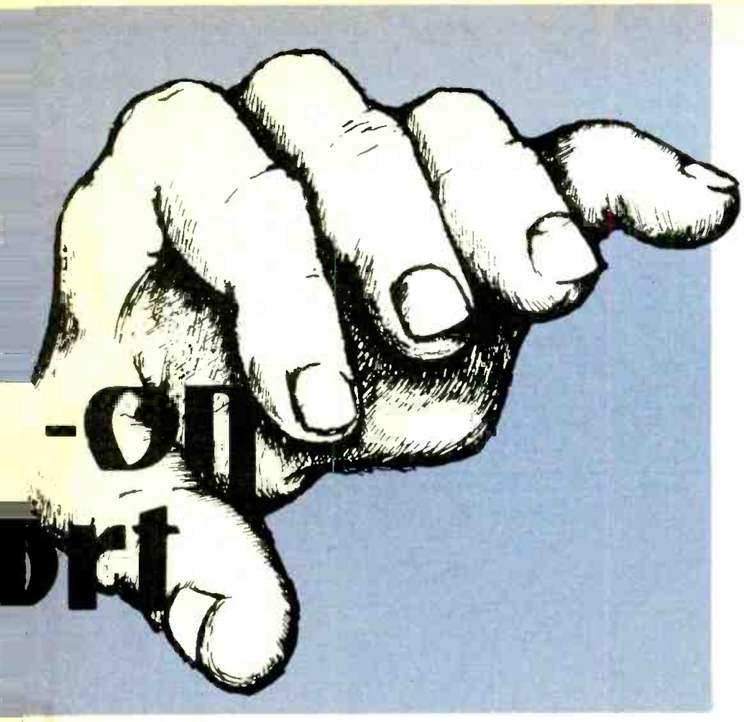
Inputs

The first item that seems to come up is "how many mics can we plug in?" The real question is, "How many inputs are there and what type are they?" The number of inputs needed is determined by the number of microphones that are to be used or by the number of tracks of a multi-track recorder that will be connected. If the band is going to use twelve mics, then twelve inputs are necessary. If an 8-track recorder is going to be connected, then eight inputs are required. In both cases it is wise to have more inputs than is absolutely needed. For example, when recording and mixing-down the engineer may want to use a piece of special effects equipment (like a digital delay unit), and at that time an additional input would be necessary to return the signal into the mix. Also, in case of an electronic malfunction it would be nice to be able to switch to an extra input and then solve the problem *after* the job, when there is plenty of time and a minimum of pressure.

The type of input should be examined next. The purchaser needs to know what is required to connect his mics or equipment into the mixer. The standard inputs are: a) low-impedance microphone; b) high-impedance microphone; c) semi-pro level; d) professional line level; and e) phono. In general the line inputs are for tape recorders, reverb chambers, limiters/compressors, phasors, digital delay units, etc.

As for the microphone inputs the purchaser should know if it is high or low impedance; if it is balanced or unbalanced; and if it is a transformer or differential input. We advise that low-impedance mics and consequently low-impedence inputs be used. High-impedance mics are lower in cost, but they should be used with mic cables that are shorter than twenty-five feet. This severely limits the size of the set up. Also, high-impedance mics are more susceptible to noise, C.B. and RFI pickup. For these reasons the majority of all semi-pro mixers are built to be used with low-impedance mics (and that's the way it should be).

The microphone inputs should be balanced because unbalanced inputs are more susceptible to noise, C.B. and RFI pickup. If the sound or recording is important, then you don't want any interference. One example case would be: You just bought a new 4-track recorder and mixing console, and the local church called up and asked you to come record their special Sunday service. So you did, and in the middle of this very important recording, when the preacher is at the high point of his sermon and he says, "The Lord said"—"Breaker, Breaker, Nineteen. Ya got a copy on me good buddy." Well, we all laugh at the C.B. that came



in at the wrong time, but it is very embarassing to the person doing the recording. Also, it could be very costly if it happens in the middle of an important recording session with the hottest group in town, and it ruins the best "take" of the day. Overall, balanced microphone inputs will provide the best results, and for a recording mixer we advise only the use of balanced inputs. Now, for a P.A. mixer where the sound is at high volumes and the voltages produced by the microphones are high, the need for balanced inputs is not as great. In these cases unbalanced inputs may be acceptable. Unbalanced inputs cost less money to build, and this may be a factor to consider.

The use of transformers in audio always stirs up a big debate. They have several weak performance areas, and good design engineers are always trying to get around these problems. Basically, a transformer is not the lowest distortion, most linear device that audio engineers have to work with. Well, why are transformers used? In the case of a mic input they offer voltage gain which improves the signal-to-noise ratio of the mic preamp. If the mixer is to be used for quality recording then the inputs should have transformers. If the mixer is used for "rock and roll" P.A. where the volumes are high, then the signal-to-noise ratio may not be as important and transformerless inputs may be acceptable. Another reason design engineers attempt to eliminate transformers is their high cost—$10 to $40 each. It is easy to see that an 8-input mixer with transformers will probably cost $160 more than a mixer without the transformers, and that's a lot of money!

To summarize, it is best that microphone inputs are low impedance, balanced and have transformers.

Line inputs are designed to accept a signal voltage that has been preamplified. A microphone output voltage goes into a mic input and is amplified up to a line level by the mic preamp.

The average output voltage from a microphone is about .0025 volts (of course this varies from mic to mic and also depends on how loud you scream into the mic). The standard line-level voltage for most professional equipment is about 1.23 volts, and the standard line-level voltage for most semi-professional equipment is about .25 volts. Some semi-pro equipment has line-level voltages around .775 volts. Before any equipment is purchased, it is best to know the line-level voltages (and impedances— we will talk about this in the months to come), so that you are sure that all of the equipment will connect together and operate properly. For example, a tape recorder that has a standard operating line level of .775 volts should operate best when an input voltage of .775 volts is supplied to it, and its output should provide .775 volts to drive the piece of equipment that follows the tape recorder.

Now, line-level inputs (and outputs) can be high or low impedance, and balanced or unbalanced. Also, they may or may not have a transformer. Due to the high voltage at standard line levels, usually they are unbalanced. Most inputs are high impedance and most outputs are low impedance. Transformers are used mainly for ground isolation. In small systems where there are short wire runs, a minimum of wires and a minimum of pieces of electronic equipment, most often unbalanced operation without transformers will work satisfactorily. In large systems or where electrical noise and interference is a severe problem it may be necessary to use transformers and balance the lines.

Phono inputs are not really a standard input for most small mixers, but with the recent disco boom they are appearing on some small pieces of equipment. It is nice to be able to connect a turntable to a mixer and listen to the latest disco hit. If you want to be capable of handling a disco-type job, the built-in phono preamp will make it much easier.

Outputs

The next important category to discuss is the number of outputs. Once again this should be determined by the job that needs to be accomplished. If the mixer is to be used for P.A., count up the number of mixes that is desired. First, is the sound system mono or stereo? (or quad?). For mono—one main output is needed; for stereo—two main outputs are needed, and so forth. If there is to be a stage monitor system, then there should be a separate mix and output for this purpose. Some of the larger sound systems (and rock groups) have need for several stage monitor mixes. If this is the case then there should be a separate mix and output for each stage monitor system desired. If you want four independent stage monitor systems, then you need four independent outputs.

Another type of output class would be for use with special effects equipment (limiters/compressors, digital delay units, equalizers, reverb chambers, etc.). Usually for P.A. one or two additional separate mixes and outputs are satisfactory for this. In most situations the reverb is attached to one output, and the other equipment is patched in and out of the other output. The majority of all mid-priced mixers offer these outputs for special-effects equipment.

If the mixer is to be used for recording, then the number of outputs to be used simultaneously must be determined. First, how many tracks does the tape recorder have? If it is a stereo recorder, then it has two tracks and two outputs are needed. If it is a 4-track recorder, then four outputs are necessary. Generally this method can be continued to determine the number of tracks and outputs, but there is another point to consider. If the tape recorder is an 8-track, do you need eight outputs? Well, not necessarily. Only count the number of tracks that you want to record on at the same time. If you can record four tracks at one time and record the other four tracks later, then a 4-output mixer is satisfactory. The process of recording several tracks at different times is called overdubbing. A smart recording engineer plans ahead and uses a minimum of outputs and tracks. Some 24-track recording consoles only have eight main outputs.

After the number of main outputs has been decided upon, the next problem is getting a mix to an earphone system so that the musicians can hear what is on the tape when they are recording. This headphone system is called a "cue system," and it is basically the same thing as a stage monitor system for a "live" P.A. except that it is usually amplified through headphones. For each cue mix desired there should be one separate mix and output on the mixer. Most small recording mixers provide one output for this purpose, however some large studio consoles have four to six individual cue mixes.

Another output that is needed on a recording mixer is one for special effects. This output is the same as the one described above in the P.A. special effects discussion. Usually a small recording mixer will have one mix and output for a reverb chamber. During the recording of the tracks the cue mix and its output will be used for the musicians' headphone system, but during mixdown the cue mix will be used as a special effects output.

All of the outputs in a mixer are identical electronically (or should be). A good engineer will soon be patching different outputs into all types of equipment in order to accomplish the job he wants done. For example, if the mixer has four outputs, and it is connected to an 8-track recorder, what do you do when it is time to mixdown to a stereo recorder? Well, you unplug two of the outputs into the 8-track, and plug them into a 2-track.

Monitor Section

Up to now inputs and outputs have been discussed, and for the average P.A. mixer this is all that is required. However, time after time a P.A. mixer is purchased, a 4- or 8-track recorder is purchased and then multi-track recording is attempted. This is where a big problem arises and the end result is usually a very discouraged and broke group of musicians. For simple mono or 2-track recording most P.A. mixers will do a good job, but for multi-track recording the mixer needs another set of controls called a "monitor section." The monitor section is what is missing on most low-priced mixers, and it is this group of controls that makes multi-track recording a success or a frustrating experience.

The monitor section is a separate stereo mixer that is connected to the outputs of the tape recorder (sometimes to the main outputs of the mixer). The outputs from this stereo mixer are connected to stereo headphones or monitor speakers and the engineer listens to the tracks of the tape. Each track of the tape has a separate volume control in the monitor section so that the engineer may individually adjust the volume of the instruments to get the mix and sound he wants. Because each track of the tape needs a separate volume control, the monitor section must match the tape recorder. This means if it is a 4-track recorder then it must be a 4-input monitor section. If it is an 8-track recorder then it must be an 8-input monitor section. Now, the monitor section may be larger than the tape recorder which means an 8-input monitor section will work fine with a 4-track recorder. The purpose of the mic inputs is to amplify and modify the sound of the mics. The purpose of the main outputs is to get the signal to the tape recorder properly. The tape recorder's job is to record and store the sound with the lowest noise and distortion. What goes into the tape recorder should come out. The job of the monitor section is to let the engineer and musicians hear the tracks of the tape and achieve the proper instrument and sound balance during the recording session.

The monitor section should also have a reverb mix and output for the reverb chamber and a cue mix and output for the musicians' headphone system. The cue and reverb mixes are usually connected to the regular cue and reverb mixes that were discussed earlier. The monitor section may also have a "talk back" output so that the engineer can talk to the musicians in the studio. The stereo output that the engineer listens to is called the "control room" output, and most often there is an identical output that goes to the studio speakers so the musicians can listen to tape playbacks.

To summarize the important facts, a P.A. mixer needs enough mic inputs to handle the number of mics, and it needs enough outputs to drive the P.A. and the stage monitors. A recording mixer needs enough mic inputs to handle the number of mics; enough line inputs to handle the largest multi-track recorder; enough outputs to record on the multi-track tape recorder, send a cue mix to the musicians and send a reverb mix to the reverb chamber; a monitor section with enough inputs to handle the largest multi-track recorder that will be used.

We hope that this information will provide a foundation for choosing a mixer and that it will result in good investments for the musician and small recording studio. We did not discuss equalization, pre- and post-cue, reverb, and monitor sends, solo buses, direct outputs, patch points, reverb returns, etc. It would require a book to discuss all the factors, but we will try to cover some of these other important points in the months to come.

"The Sansui AU-717 is a superb amplifier. We like it with no ifs, ands, or buts." (Julian Hirsch) It offers "as much circuitry sophistication and control flexibility as any two-piece amplifying system."

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Julian D. Hirsch, Contributing Editor Stereo Review

"The Sansui AU-717 is a superb amplifier. We like it with no ifs, ands, or buts." [Reprinted in part from Julian Hirsch's test report in **Stereo Review**, February, 1978.]

"One clear advantage of DC design is apparent. Even at the low 20Hz extreme, the amplifier delivers a full 92 watts — the same value obtained for midfrequency power — compared with its 85 watt rating into 8 ohms....

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"... worth serious consideration — even by those who prefer separate amplifiers and preamplifiers." [Reprinted in part from Len Feldman's test report in **Radio-Electronics**, January, 1978.]

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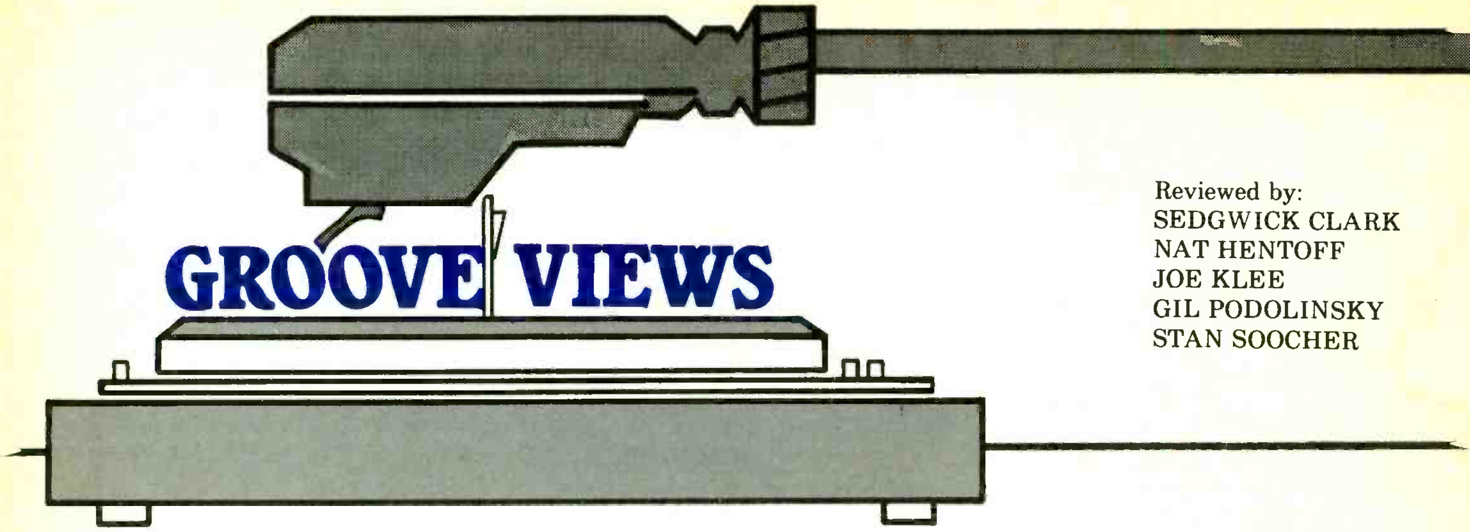

Reviewed by:
SEDGWICK CLARK
NAT HENTOFF
JOE KLEE
GIL PODOLINSKY
STAN SOOCHER

POPULAR

BOOMTOWN RATS: *The Boomtown Rats.* [Robert John Lange, producer; Steve Brown, engineer; recorded at Dieter Dierks's (sic) Studio, Stommein Koln, Germany.] Ensign/Mercury SRM 1-1188.

Performance: **Up & down**
Recording: **So-so**

This group sounds like a cross between David Bowie, Bruce Springsteen and the typical punk rocker. The music is fairly basic, mostly chordal and nothing new. What makes this group different from the rest of the new wave of wash-ups is the above average production and the clarity of the recording.

Going beyond the basic punk line-up of guitar-bass-drum-vocal, The Rats use acoustic piano, guitar and dual electric guitars. One track in point is "I Can Make It If You Can," a basic rip-off of Springsteen's "Thunder Road." A Fender Stratocaster guitar, scrapped of its bottom, churns out two chords in a stereo intro, leading into drums and a scratching Hammond organ playing the same two chords. Next an acoustic piano falls into line, followed by another electric guitar which plays the major notes of the two chords, only in a different octave. As soon as the vocal comes on the scene, mixed center stage, the organ drops out until its re-introduction after the chorus, which then gives way to a bad imitation of a Clarence Cleamons sax solo. While the orchestration of instruments is intelligent, it is hardly original. The bass is recorded 'way in the background and the snare given that deep, solid sound.

If you've never heard Springsteen or Bowie, you'll think this album is great. If you have, you'll pass this one by. G.P.

BOOMTOWN RATS: **Pass them by**

MICHAEL JOHNSON: *Ain't Dis Da Life.* [Michael Johnson, producer; Steve Wiese, engineer; recorded at Creation Audio Recording, Blomington, Minnesota.] Sanskrit SR 0774.

Performance: **Pleasant**
Recording: **Good**

This isn't half bad for a first effort at recording, producing and running a small, independent record company. Although he doesn't write much of his own material, the obscure songwriters from whom he draws turn a few good phrases and his voice is pleasing—sounding something like a cross between James Taylor and Kenny Rankin. Actually, all this album lacks is proper distribution, for it's much better than most of what's being released in this vein. Also, a second listening by a major label may benefit all concerned.

MICHAEL JOHNSON: **Quite good**

"Ain't Dis Da Life," a song in the James Taylor vein, begins with an acoustic guitar center, giving way to the vocal, also center, backed by bass and drums. These instruments are joined in the middle of the song by marimba, which enters right. Next is a two-part call and response vocal, left and right, which is accented through the use of clarinets. An ascending B-flat clarinet is left while a descending bass clarinet counters right, making for a perfect ragtime piece. Overall, it's a quite commendable recording. G.P.

LEONARD COHEN: *Death Of A Ladies Man.* [Phil Spector, producer; Larry Levine, engineer; recorded at Whitney Recording Studios, Gold Star Recording Studios, Devonshire Sound Studios, Los Angeles, Ca.] Warner Brothers BS3125.

Performance: **Strained**
Recording: **Textbook Spector**

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MANUFACTURED BY OTARI ELECTRIC CO. TOKYO, JAPAN ard Cohen, the poet, with Phil Spector, the painter of sounds, seems like an unusual idea, if not a complete mismatch. After all, Cohen's work has always been more subtle while Spector often borders on the manic. Yet, for all its faults, *Death Of A Ladies Man* is a unique achievement because Phil Spector comes close to working his production style around that of a singer/lyricist instead of simply adding the vocalist as part of the overall production. For Leonard Cohen, the experience must have been frustrating because his voice has to fight the arrangements to obtain any real presence. In "True Love Leaves No Traces," for instance, Cohen's vocal is overshadowed by back-up singer Ronee Blakely.

Producer Phil Spector first became known for his massive mono mixes that peaked with the Righteous Brothers' classics "You've Lost That Lovin' Feeling" and "Soul and Inspiration," greatly influencing contemporary producers like Jon Landau (Bruce Springsteen) and Brian Wilson (Beach Boys). On those mid-sixties Spector records and with his earlier productions of female vocal groups like the Ronettes, Phil managed to include the most

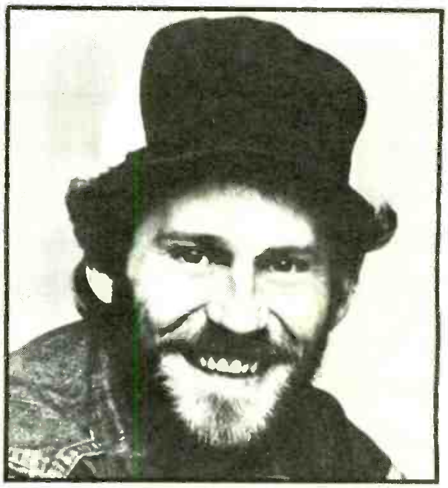
LEONARD COHEN: A battle of styles

musical elements in the smallest possible space—and succeed in the process. Working in the stereo medium on *Death Of A Ladies Man*—which is foreboding in mono as well—Spector is redefining space where tracks overlap in the mix and give the illusion of a larger field of sound than seems possible. What Spector does is build from the bottom up in layers, filling each hole, then "widening" the presence of each instrument.

Spector also wrote the music here for Leonard Cohen's lyrical poems. Spector does pay attention to the cadences of Cohen's exploration of failing relationships between men and women, but when the melodies are translated through Cohen's folk persona and set against Spector's "wall of sound," they become flawed. Among the exceptions are the up-tempo numbers "Don't Go Home With Your Hard On" with guest vocalist Bob Dylan and "Fingerprints," a country-flavored tune complete with fiddle. If Cohen's vocals are overtaken by Spector's production on most of the songs, Cohen's lyrics are nevertheless brilliant and biting. Perhaps this album would be better if Spector's production was panned to one side in mono while Cohen's vocals were panned to the other. S.S.

LEVON HELM: *Levon Helm and The RCO All-Stars.* [Levon Helm and the RCO All-Stars, producers; Eddie Offord, engineer; recorded at RCO Studios, Woodstock, N.Y. and Shangri-La Studios, Malibu, Ca.] ABC AA1017.

Performance: **Democratic; everything but the kitchen sink**
Recording: **Full of little surprises**



LEVON HELM: Lacking importance

With the release of his first solo album, drummer Levon Helm does much to recreate the loose improvisational sense of community that permeated most of the Band's work. Yet, just as many of the Beatles' solo albums prove that the sum is sometimes more than equal to its parts, Helm's album does not stand up well next to Band albums like *Music From Big Pink* and *Cahoots.* What Helm has given us is an enjoyable, goodtime collection of songs that lack the importance of the Band's statements—perhaps because Helm borrows tunes from

so many varied songwriting sources, including himself and the RCO All-Stars—making his musical personality closer to that of the Beatles' Ringo Starr.

That said, this album offers quite a few interesting moments. Primarily, the considerable talents of the notable session players add authority to the spontaneity of the songs. The RCO All-Stars include Steve Cropper on guitar, Booker T. Jones on keyboards and percussion, Mac Rebennack (Dr. John) on keyboards, guitar and percussion, Duck Dunn on bass, Fred Carter, Jr. on guitar and Howard Johnson on baritone sax and tuba. The entire horn section—trumpets, saxophones, and trombones—is as unpredictably arranged throughout as they were on Band songs like "Life Is A Carnival," yet intricately enough so that they never overwhelm the rest of the instruments on each song's tracks.

In the same way, the nature of the mix adds greatly to the album's fluid feel. Apparently the producers and engineer Offord are not from the "fix it in the mix" school because they aren't afraid to leave in some of the loose ends such as the stray electric piano at the end of "A Mood I Was In." It is Levon Helm's work that ultimately ties the album together, though, but not necessarily for the strength of Levon's voice because he was not the only good singer in the Band. Still, his overall vocal performance is nearly as good as his performance on the Band's classic "Rag, Mama, Rag."

Given a good bottle of whiskey and a few close friends then, this album is worth the price if only for the fun of its festive atmosphere. S.S.

JAZZ

LARRY CORYELL AND STEVE KAHN: *Two For The Road.* [J. Vince Cirrincione, Tom Paine, executive producers; Steve Kahn, producer; Rich Okon, production assistant; recording dates and places and names of engineers unlisted.] Arista AB 4156.

Performance: **Nothing less than dazzling**
Recording: **Nothing less than honest**

LARRY CORYELL: Tasteful variations

I first heard Steve Kahn and Larry Coryell working more or less together at Max's Kansas City in the old days. Steve was playing in "Count's Rock Band" which was alternating with Larry's Eleventh House group. I was at once struck by the similarities in vocabulary and frame of reference between these two players, as well as their differences. Steve played top volume at all times, but Larry (remembering what Jelly Roll Morton once said about 'if you start with a full glass of water you can't add anything more to it') varied his bursts of top decibel playing with acoustic and nearly acoustic moments of gloriously subdued, unashamedly tasteful playing. Either Steve wasn't showing us everything he could do or he's learned a lot since then because there's not a jarring electric note on this album. I suspect that it's totally acoustic but if any electronics were used either in performance or post-production sweetening, they're used so tastefully that they're not abrasive—and I can't quarrel with that.

One thing I can take issue with is the lack of recording dates and data. The entire itinerary of the tour is listed and several dates (in Montreaux, Switzerland and in Miami—they don't specify whether that's Miami, Florida, or Miami, Ohio) are listed as "live" recordings. But nowhere does it state whether these recordings came from either or both of these dates. While Steve Kahn's personal impressions of the tour are a delight to read, I do think that vital information was overlooked to make room for them.

It is especially interesting to me that in this day and age of ego-tripping bands that play only their tunes that Kahn and Coryell feel so secure in their own abilities that they include only one of Larry's originals and one joint effort. The rest of the tunes come from such accepted standard sources as Wayne Shorter and Chick Corea and Bobby Hutcherson with a revival of Steve Swallow's

CIRCLE 50 ON READER SERVICE CARD

The Jazz Band Sound: From the Far-Flung Present to Its Roots

By Nat Hentoff

Toshiko Akiyoshi—long a crisply swinging, melodically inventive pianist—has evolved in recent years into the leader of one of the most resilient, invigorating big bands in all of jazz. Actually, she's the co-leader, the other being her husband Lew Tabackin, a skilled tenor player and a much more compelling flutist. But it is Toshiko who gives the band its distinctive identity because practically all the writing is by her. And her spirit—thoughtful, sometimes jubilant, and always lyrical—pervades the music.

Insights (RCA) is a landmark album in the band's odyssey. Not the first side, which combines a buoyant swinger, a subtly colored reflective piece, and "Sumie"—a graceful fusion of jazz and intimations of Toshiko's Japanese roots. This is all performed with precise but relaxed ensemble brilliance and a series of warm, incisive soloists. And as is characteristic of any Toshiko group, there is unusual sensitivity to dynamics by the entire cast. This is a band that can swing hard, but it can do many other things as well.

It is the second side, "Minimata"—a work by Toshiko that lasts almost twenty-two minutes—which insures this set an historic place in the jazz canon. Oddly, the liner notes, while extolling the music, do not say a word about the origin of the piece. This is Toshiko's way of putting into music the horrifyingly true story of a Japanese village invaded by industrial pollution, the results of which led to the transmogrification of many of the inhabitants into grotesquely misshapen witnesses to corporate criminal greed. (For the pictorial record, see the book about Minamata by W. Eugene Smith.) Yet this is not agit-prop music. Toshiko, with the empathic aid of her musicians, has created a panorama of impressions and indeed insights—occasionally intertwining Japanese with jazz shadings—that stands as music. With, by the way, effective echoes of Duke Ellington. The recorded sound is somewhat tighter than I like for a big band, but the expressive power of all comes through clearly and climactically.

Even in "Minamata," there are exuberant shouting, even strutting sections; and that is part of the jazz band legacy. In New Orleans, after all, the band keened on the way to the grave, but marched back in jaunty celebration of the continuum of life. And in New Orleans, there are still such high-spirited marching units, one of the best of them being Dejan's Olympia Brass Band.

I have delighted in these New Orleans phenomena for forty years and if you've never experienced this root-force of collective jazz improvisation, Dejan's is a set to start with. I doubt if there is any way to keep still, or seated, while listening to the Olympia players whose music dances in the ear. The label is Biograph (16 River Street, Chatham, N.Y. 12037) and copies can also be purchased from the latter-day source of vintage New Orleans sounds, Preservation Hall (726 St. Peter St., New Orleans, Louisiana 70116).

The recording is bright, spacious, and gives you a sense of being right out on the street, testifying to the glory of these sounds.

TOSHIKO AKYOSHI: *Insights.* [Hiroshi Asaki, producer; Grover Helsey, engineer.] RCA AFL1-2678.

HAROLD DEJAN: *Dejan's Olympia Brass Band.* [No recording informations given.] Biograph VPS-4.

"General Mojo's Well Laid Plan" thrown in as a reminder of the famous Coryell/Swallow/Gary Burton/Roy Haynes quartet. It's a delightful record. I hope there'll be more of the same—but next time please give us the dates and places as well. J.K.

KUSTBANDET AND CHARLIE HOLMES: *Star Dust.* [Gosta Hagglof, producer; Gert Palmcrantz (Stockholm), Fred Miller (New York, engineers; recorded at Europa Film Studio, Stockholm, Sweden (1974-75) and Downtown Sound, New York City, N.Y. (4/10/75).] Kenneth KS 2039.

Performance: **Enthusiastic and hotter than. . .**
Recording: **A transatlantic gimmick, but one that worked**

On paper it looked kind of hokey. This Swedish band that plays arrangements transcribed from records of classic early jazz bands such as Fletcher Henderson, McKinney's Cotton Pickers, Luis Russell and Duke Ellington made an LP's worth of tapes and left some solo space for Luis Russell's featured alto sax star Charlie Holmes to fill. Holmes, here in New York, overdubbed onto the band tapes in a session recorded by Fred Miller and supervised by Al Volmer. It's easy to say it won't work . . . but it did. Just listen to "Saratoga Shout" or "Star Dust." Then listen to the astounding things they do with Ellington's "Rockin' In Rhythm" with its wild fugue for saxophones and Doc Vollmer shouting over the final chorus—stay with it until the end.

Kustbandet has some flaws to be sure. They lack the kind of dynamic soloists in each department that made the black bands of the '20s the powerhouse units they were. Players like clarinetist Erik Persson and pianist Ake Edenstand can get off some very impressive hot choruses but the trumpets lack a hot man who can solo with the consistency of an Armstrong, Rex Stewart or a Red Allen. But what's more important with a band like this is the enthusiasm and excitement and drive it generates and even that the rhythm section swings a lot harder than most of the '20s bands did (even though they do cheat a bit and use a string bass as well as a tuba).

Regardless of how one feels about Kustbandet (and I myself find them completely charming and wonderfully exciting) this record is important as what may well be the last recorded evidence of the saxophone playing of Charlie Holmes. Charlie was practically in retirement at the time this recording was made and he doesn't play at all any more, so I doubt that we'll have another chance to hear this giant—and he is a giant indeed. I have never heard a more splendid account of Hoagy Carmichael's classic "Star Dust" than Holmes gives it.

While at this writing Kenneth Records have to be ordered by mail from Gosta Hagglof, Ramgrand 1, 17547, Jarfalla, Sweden, plans are under way for distribution by a U.S. distributor who will have Kustbandet available either on his own label or as imports on Kenneth Records. However you are able to come across them, they are well worth the trouble. J.K.

CHARLIE PARKER: *One Night In Birdland.* [Gary Giddins, producer; Don Young, reissue engineer, from airchecks recorded by Boris Rose 6/30/50.] Columbia 34808.

CHARLIE PARKER: *Bird With Strings.* [Gary Giddins, producer; Larry Hiller, reissue engineer, from airchecks recorded by Boris Rose 1950, 1951 and 1952.] Columbia 34832.

CHARLIE PARKER: *Summit Meeting At Birdland.* [Gary Giddins, producer; Don Young, reissue engineer; from airchecks recorded by Boris Rose 3/31/51, 3/23/53 and 5/9/53.] Columbia 34831.

Performances: **Still contemporary, still masterpieces**
Recording: **Terrible, but it's all we have**

These recordings are part of a series Columbia calls the "contemporary masters series." I don't know just how a musician who died in 1955 can be considered as "contemporary," unless it's that even two decades after his passing his influence still shows up daily in those of his contemporaries who are still with us and the kids who grew up listening to him and were intimidated into imitating him. Certainly imitation is not the innovative way for a musician to go but a case can be made. How much more could any saxophone player hope to say on the changes of Cole Porter's "Easy To Love" than Bird did? And not only did Bird fairly exhaust the possibilities of the tune but he did it three times (the Apollo Theatre on 8/23/50, Carnegie Hall on 11/14/52 and Birdland on 4/7/51), all on the *Bird With Strings* ɔum. And each time he said it differ-ıtly. As a contemporary master ıarlie Parker certainly lives up to the ›finitions. The sides with strings are, ' necessity, pretty well charted leaving ıly brief holes for Parker to improvise but what Bird could do with just a ·ief hole in the arrangement! The :her airchecks show Charlie Parker ith the famous (Dizzy Gillespie and ud Powell) and the not so famous but]ually fine (Fats Navarro and Milt uckner) displaying his virtuosity and nprovisational fertility on standards of ebop ("Round Midnight" and "Night ı Tunisia") and prebop ("Embraceable 'ou" and "Star Eyes"). Bird's ability ɔ find musicality in just about anything e picked up was one of his most ndearing features. They're not here but ; should take only a little digging to urn up Charlie Parker's airshots of unes like "On A Slow Boat To China" or a lesson in the manufacture of silk ›urses from sows' ears. That's if you ıappen to be a genius.

As for the recording it was done by ı man named Boris Rose who did the ɔest he could with the primitive home ·ecording equipment of the 1950s and :he wretched sound of those pre-FM radio remote broadcasts to begin with. But this is history. This is the way it was and if we were to find a crude recording of Beethoven conducting his Ninth symphony or Bach improvising on the organ, who would complain that it wasn't stereo, hi-fi or quad? J.K.

CLASSICAL

VAUGHAN WILLIAMS: ***Fantasia on a Theme by Tallis;*** **DVORAK:** ***Serenade for Strings, Op. 22;*** **PURCELL:** ***Dido's Lament.*** Royal Philharmonic Orchestra, Leopold Stokowski cond. [Antony Hodgson, producer; Neville Boyling, engineer.] Desmar DMS 1011.

Performances: **Large-scale**
Recording: **Church acoustic**

STOKOWSKI/WAGNER: ***Gotterdammerung:*** Orchestral Highlights. London Symphony Orchestra, Leopold Stokowski cond. [Richard Mohr, producer; Christopher Parker, engineer.] RCA ARL 1-1317.

WAGNER: ***Rienzi:*** Overture; ***Die Walkure:*** Magic Fire Music; ***Die Meistersinger:*** Prelude to Act III, Dance of the Apprentices and Entrance of the Meistersinger; ***Tristan and Isolde:*** Prelude and Liebestod. Royal Philharmonic Orchestra, Leopold Stokowski cond. [Richard Mohr, producer; Robert Auger, engineer.] RCA ARL 1-0498.

Performances: **Succulent**
Recordings: **Less resonant**

TCHAIKOVSKY: Aurora's Wedding Ballet Music from ***The Sleeping Beauty.*** National Philharmonic Orchestra, Leopold Stokowski. [Paul Myers and Roy Emerson, producers; Robert Auger and Mike Ross-Trevor, engineers.] Columbia M 34560

SIBELIUS: ***Symphony No. 1; Swan of Tuonela.*** Same credits as Aurora's Wedding, except no producers listed. Columbia M 34548.

STOKOWSKI: Great Transcriptions for Orchestra. Same credits as Aurora's Wedding, except that Robert Auger is listed as the only engineer. Columbia M 34543.

BIZET: ***Carmen*** and ***L'Arlesienne*** Suites. Same credits as Stokowski Transcriptions disc. Columbia M 34503.

Performances: **Romantic**
Recordings: **Hugely resonant**

Leopold Stokowski's conducting career spanned over 70 years—60 of them in the recording studio, where he was the leading experimenter with nearly every new technique developed in this field. Active to the end, the 95-year-old maestro had been scheduled to record Rachmaninoff's Second Symphony the day following his death on September 13, 1977.

Nowadays it is *de rigeur* for musicians to familiarize themselves with the studio, but Stokowski's concern with the quality of sound led him to be more active in the recording process than any of his contemporaries. He recorded the music with The Philadelphia Orchestra for Disney's *Fantasia* (1940) in six-channel stereo long before the invention of the stereo disc; still in the days of mono, he experimented with mixing separately miked choirs of the orchestra in the studio to achieve greater clarity; his Vanguard recording of Stravinsky's *L'Histoire du Soldat* was the first American recording to utilize the Dolby System; and, not surprisingly, four-channel techno ogy found one of its earliest enthus asts in the ever-adventurous Leopol Stokowski. "But remember," he cau tioned in an interview given in 1967, " recording can give you nothing but th illusion of a performance."

The records listed above are all char acteristic of the famous "Stokowsk Sound"—sensuous, organ-like richnes of texture, with a seemingly endles variety of tone color—and they al receive interpretations of invigorating romantic warmth and sweep. As usua with this most imaginative and willfu of conductors, there are details that one might criticize. And, as always, he can't resist personalizing and height ening the drama of the music with alterations of scoring, dynamics and tempos. Yet the final impression re mains of an infinitely lively musica mind, deeply committed to his art. "The love of music is a continuous life of enjoying beauty and sound," Stokowski stated in an interview with Robert Jacobson. "It has been a continual effort to make music more alive, so that it is not a mechanical reproduction of what is on a piece of paper—but a real expression as it always was with the greatest artists."

The Desmar release has an interesting background. The company made its debut two years ago with proclamations of high quality, but founder Marcos Klorman soon discovered how easily things go awry. After exasperating attempts to achieve those chosen standards in American pressing plants, Desmar records are now pressed by Telefunken in West Germany. This particular recording waited over a year-and-a-half for its release while Klorman rejected over a dozen separately mastered test pressings. It was worth the wait. Stokowski's performances of the Vaughan Williams' Tallis Fantasia, the short arrangement of Dido's Lament from Purcell's opera *Dido and Aeneas* and Dvorak's String Serenade are unique for their passion and large-scale extroversion. While one may prefer more intimate accounts of the Williams and Dvorak works (especially the latter in a superb version by The Czech Chamber Orchestra, conducted by Josef Vlach, formerly on Crossroads in the U.S. and presently available on a Supraphon import), the multi-hued richness and Brahms-like heft conjured up by Stokowski is interpretively valid. Incredibly, the Dvorak recording was apparently the first time he had ever conducted the work!

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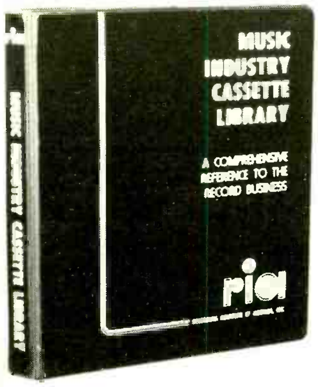

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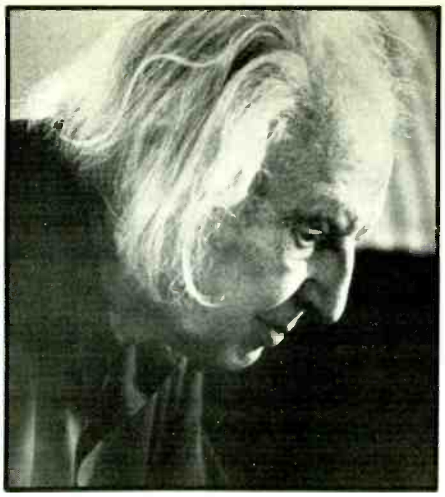

LEOPOLD STOKOWSKI: Inspiring

Recorded in a very resonant acoustic, the spacious sonics may seem to diffuse when played in rooms with high ceilings, but focus is quite satisfactory in more modest settings. Surfaces and processing are excellent.

The two Wagner discs on RCA receive much tighter sound—presumably due more to a markedly smaller recording venue than to closer miking. Instrumental lines and textures are thus clearer, complemented by excellent playing from the respective orchestras. Indeed, the disc containing excerpts from *Rienzi, Tristan* and *Meistersinger*, displays some of the finest playing on a Stokowski record in recent years: The Royal Philharmonic is truly playing *for* him, sensitive to every tempo change and *rubato*, and especially impressive in the propulsive *Tristan* Prelude and Liebestod, where rarely a bar goes by without some tempo adjustment or caressing of a detail. The *Gotterdamerung* disc is one of the conductor's "symphonic synthesis" reworkings of opera, the music patched together and rewritten with solo instruments substituting for the vocal lines. Mastering is fine, but surfaces were an occasional distraction.

The four Columbia releases apparently emanate from the same production team (no producer is listed for the Sibelius disc, the least sonically successful of the four). Like much of engineer Robert Auger's work, the sound has plenty of impact within a huge acoustical perspective; the result is quite different from his work on the second Wagner disc for RCA, however, reflecting the final decision by the producers in each case. Obviously after a big effect, their success is admirable. The Farandole which concludes the *L'Arlesienne* excerpts is a particularly exciting demonstration piece.

Tchaikovsky's "Aurora's We ding," consisting of music taken mos ly from the third act of *Sleeping Bea ty*, represents the best playing of Tl National Philharmonic, a London pic up band of the finest players from tl city's five major orchestras. Unde standably, the conductor's contr diminished with age, and lapses ensemble that he never would have tc erated in earlier years had to stand order to conserve his energy for cor pletion of the sessions. Minor imprec sions apart, the Bizet disc is wholl recommendable for its unabashed e: hilaration. Stokowski has judiciousl selected the best-known numbers froi the pair of suites for each worl Sonically, the *Carmen* side seem slightly more intimate in perspectiv than the *L'Arlesienne* excerpts, whic have a bit more depth and vibrancy The Transcriptions disc is enjoyabl light listening, and the sound is up t the best on these Columbia releases Highlights are the conductor's set tings of Rimsky-Korsakov's *Flight o the Bumblebee* and Albeniz's *Fet Dieu a Seville.*

Listeners attuned to Colin Davis classical approach to Sibelius (*MR* Oct. 1977) are apt to find their eye brows converging with their hairline over Stokowski's ideas about the Fin nish composer's *Symphony No. 1.* Always best in music where structura considerations did not interfere with his ever-active imagination, the ageless maestro not only emphasizes Sibelius' debt to Tchaikovsky but contributes many individual touches of his own. This symphony does not lend itself to the "showpiece" route, however, and the cavernous sound sweeps all sorts of inner detail (and, one suspects, considerable sloppy playing) under the rug. Still, Stokowski's love of the music conquers most of the objections (not, however, the crass playing and recording of the timpani), and the *Swan of Tuonela* which completes the disc is evocatively rendered with all the mystery and sensitivity one could wish.

Still in the Columbia can are recordings of Brahms' *Symphony No. 2,* Mendelssohn's "Italian" Symphony and the Bizet *Symphony in C.* It is difficult to believe that we must content ourselves in the future with *reissues* of recordings made by the most youthful and inquiring of musicians. S.C.

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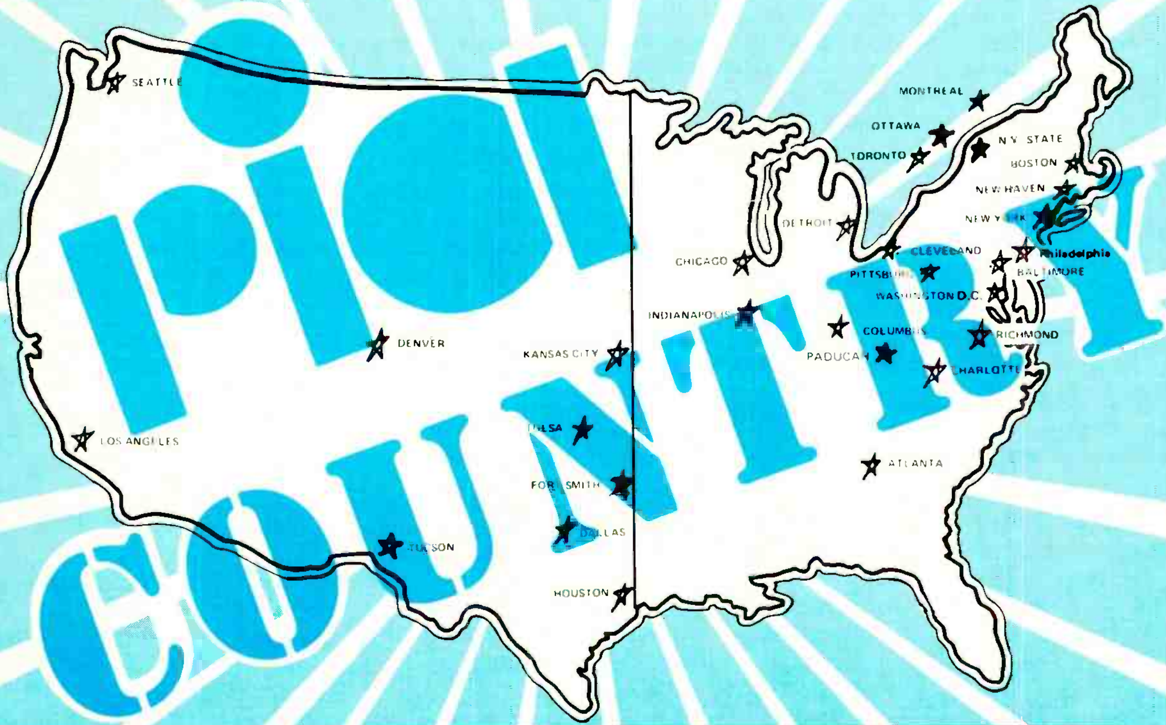

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A 140-page comprehensive directory listi names, addresses, and phone numbers every major record company, publisher booking agents, managers and independe record producers. Also, sample contra forms for each. All for $4.95. R.I.A., 15 Colur bus Circle, New York, N.Y. 10023.

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If you have an 8 or 16 track studio, and are interested in becoming a licensed representative for R.I.A.'s Modern Recording Techniques courses, call or write: Mr. P. Gallo R.I.A., 15 Columbus Circle, New York, N.Y 10023 (212) 582-3680. A Large profit potential with low operating costs.

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