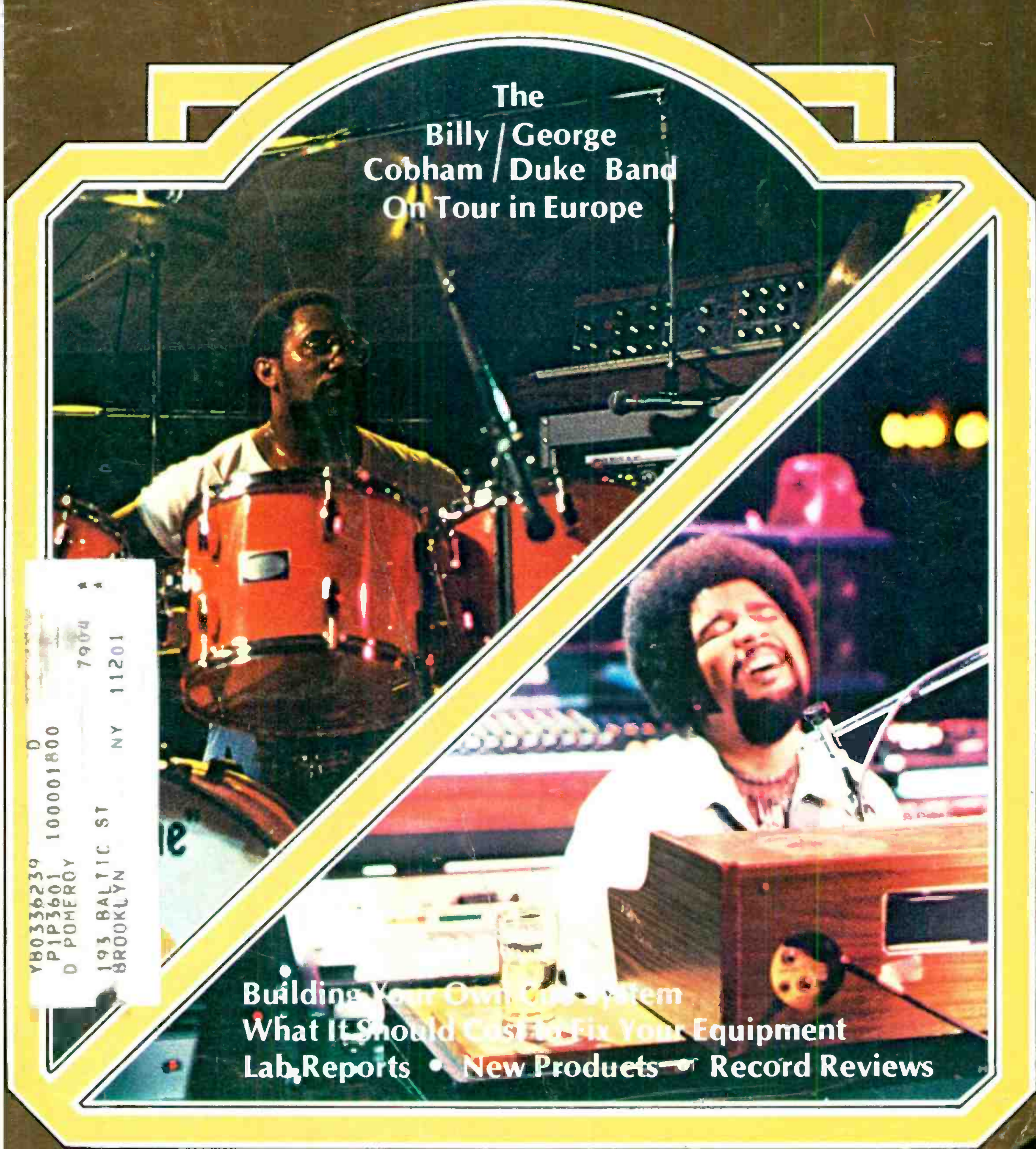
APRIL 1977

VOL. 2 NO. 4
$1.50

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY



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CIRCLE 25 ON READER SERVICE CARD

BUILD YOUR OWN SYNTHESIZER STUDIO WITH ROLAND'S SYSTEM 100

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Roland, long a leader in electronic music technology has finally developed low cost, modular synthesizer components. All the elements of a basic synthesizer plus accessories which have before now been either too complex or too expensive are included in the System 100. You can purchase them all at once for under $2000 or one component at a time. Mixing and monitoring is all self-contained so that you can begin at once to discover all the possibilities electronic music offers today's musician. Many major recording artists and studios have already purchased the System 100. Send in the coupon from this ad and we'll send you a detailed brochure of the unique features this new synthesizer has to offer.

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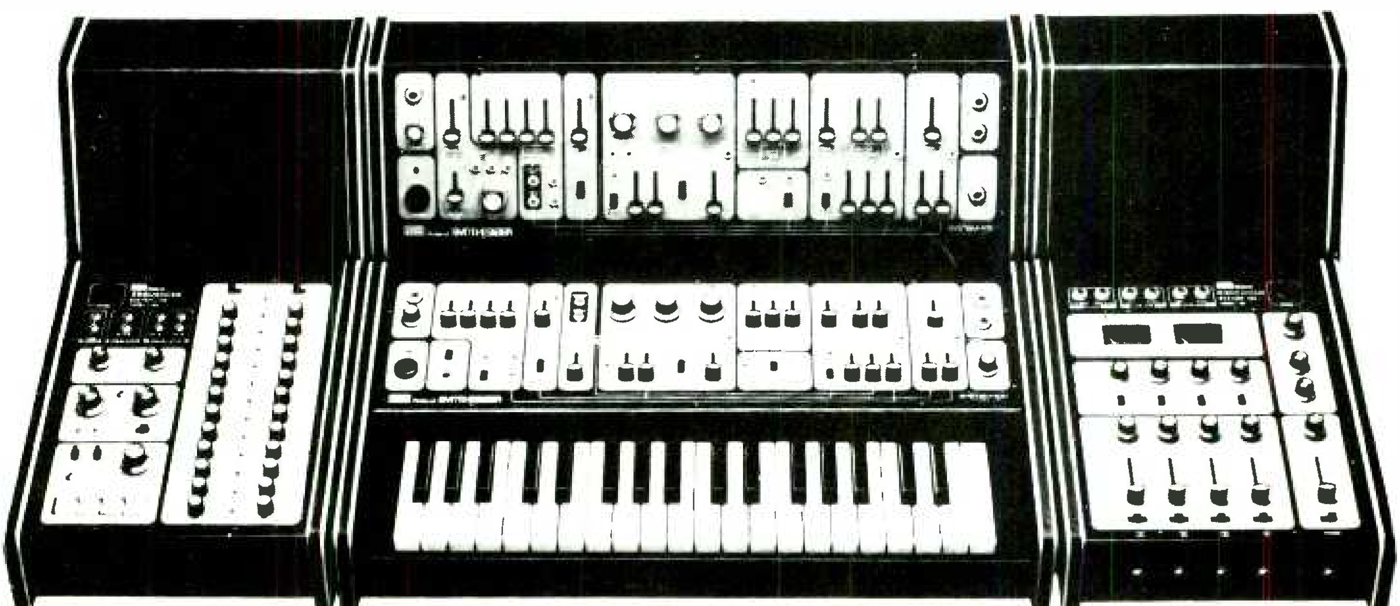
MONITOR SPEAKER 109 — EXPANDER 102 — MONITOR SPEAKER 109

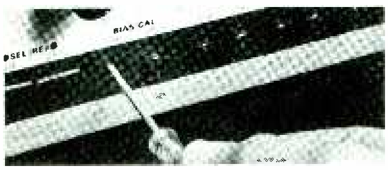


SEQUENCER 104 — BASIC UNIT 101 — MIXER 103

CIRCLE 68 ON READER SERVICE CARD

OTARI MX-5050 the original (and still the best) compact professional recorder

Just over two years ago, Otari introduced a unique new product —the first truly professional recorder in a compact package— the MX-5050. Since then, the performance and reliability of this innovative new machine have been tested and proven in over a thousand critical professional applications—by broadcasters, recording studios, A/V departments, musicians, and semipro recordists worldwide. Universal acceptance and repeat orders by these satisfied customers tell this remarkable recorder's success story better than we can.



Bias can be re-optimized in seconds.

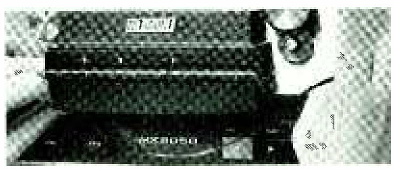
As you compare the MX-5050 with other recorders, keep this in mind. The MX-5050 is not a hi-fi machine with a few professional features added later as an afterthought. It was designed from the ground up based on Otari's 10 year experience as Japan's leading manufacturer of professional recorders and high speed duplicators. It is a full professional machine with the performance, features, and field proven reliability that you expect to find only in the larger professional recorders.

Here are some of the key reasons why the MX-5050 is the best compact recorder available today.

Production Features: Creative production is simplified with: Front panel edit to spill tape. Lift-up head cover to mark splices and clean heads. Built-in splicing block on head cover. Adjustable cue to defeat head lifters. Selective reproduce to add new tracks in perfect time synchronization. Two speed operation, 15 and 7½ or 7½ and 3¾ ips (field changeable in dc servo versions).

Performance Features: Headroom is 19 dBm, a full 15 dBm over the switch selectable fixed output of +4 dBm. This standard reference level output can be rear panel switched to −10 dBm to drive a PA system or power amplifier.
S/N ratio is NAB weighted 69 dB full track, 68 dB half track, and 65 dB quarter track. Crosstalk is greater than 60 dB half track. Outputs are 600 ohm balanced (standard on half track) or unbalanced. Line input and output connectors are XLR.

Operating Features: Bias is front-panel continuously adjustable (not limited to fixed positions). With built-in test oscillator (not available on other compact professional recorders) bias can be optimized in seconds when changing tape. Record EQ and standard reference level are also front adjustable. Straight-line tape path simplifies threading. Capstan is located on back side of tape for improved tape life. An extra reproduce head is standard on all versions to allow playback of tapes in different formats.
For pitch control and freedom from power line variations, an optional dc capstan servo is available with ±10% correction range.



Easy threading; capstan on back side.

Versatility: Available in full-track (with half-track reproduce capability standard), two-track, and quarter-track versions. Walnut case (standard), rugged portable road case, rack mounting adaptor, or floor console. Universal power supply standard. Low impedance input and output transformers and remote control also optional accessories.

See your nearest Otari dealer for the full story or contact Otari. And, if it's multichannel you need, ask about the standard-setting four and eight channel versions of the MX-5050.

OTARI

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

Otari Electric Co., Ltd.
4-29-18 Minami Ogikubo
Suginami-ku, Tokyo 167, Japan
(03) 333-9631 Telex: J26604

CIRCLE 76 ON READER SERVICE CARD

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

APRIL 1977
VOL. 2 NO. 4

THE FEATURES

COMING NEXT ISSUE!
Microphones by John Woram
Noise Reduction: Part 1

THE STAPLES

Cover photo by Hojones

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LETTERS TO THE EDITOR

Women Engineers Speak Out

In answer to engineers who wish to know of other "Ladies" in the field of engineering, I am a woman engineer. I've worked as a staff assistant engineer at Record Plant, Los Angeles and Davlen Studios, Universal City. I'm currently working as a Trainer for TEAC Corporation of America, Montebello, Ca., teaching people to record from using the simplest cassette machine all the way up to record making, eight-channel recording equipment and mixers.

Last year, I traveled throughout the United States and Canada with a show entitled, "The Care and Feeding of Your Tape Recorder." It's a four projector multi-media show that covers the art of tape recording. Presented in a "Monty Python" style, the show covers such topics as cleaning, demagnetizing, editing, etc. After the show, I go into such topics as Equalization, Dolby vs. dbx, etc., with objectives of clearing myths, giving information and just how to make "The Best Recordings."

The best advice I can give anyone wishing more information on recording is what I call "Hands On." If you have any friends with recording equipment, borrow it and record, record and record *anything*. Start training your ears. Keep reading such magazines as *Modern Recording*. Also several good books have now become available such as *Modern Recording Techniques*, by Robert E. Runstein.

I will be traveling the United States and Canada again in 1977 with the show "Care and Feeding of Your Tape Recorder." When you hear or see it advertised in your area, please stop by. I'd love to meet any of you out there, and share ideas.

—Linda D. Feldman
TEAC Corporation of America
Montebello, Ca.

Your magazine is so excellent that it is a pity that your issues numbers 1, 2, 3 and 4 are not available. Women in engineering is a topic that greatly interests me because I'm a woman engineer. I thought it would be of interest to your readers that my job is doing audio for a remote television truck.

I mix "live" and recorded sports events for Marvin H. Sugarman Productions—producers of "The Champions" and portions of "The CBS Sports Spectacular." There's a wonderful woman on our crew who is the technical director on many shows. Although she switches (video), she is quite capable of doing any job on the truck from trouble-shooting to running the tape machines. Her name is Susan Knoll.

We also have women assistant directors as well as producers. Engineering is simply not a man's world anymore.

At Indiana University where I was a music major, I did reinforcement for jazz concerts on a 16-track Altec board. At Indiana there were several other capable women on the crew who set up mics, mixed, knew how to run the tape machines, edited and made the constant artistic and technical decisions that are necessary for this kind of work.

My advice to women who want to get into recording engineering is: don't be intimidated, neither by men nor by the vast amount of knowledge that there is to learn. When men see women are serious about working they usually get on with the job at hand. As far as the vastness of knowledge, (i.e. the controls, etc.,) it may make one squeamish at first, but I defy any man to say it didn't strike the same response in him the first time he set foot in a recording studio. If there is an incurable curiosity and fascination along with the squeamishness, the former will usually prevail and one will find that all of the controls can be broken down and conquered one at a time.

My next statement may draw a lot of angry comments. I find that many of the men I've worked with have a very limited concept of the artistic side of engineering. It seems that the mechanical end is stressed to the point of short-changing spontaneity, instinct and plain good listening. This is why I feel that a thorough study of music cannot be over estimated in the training of a serious music mixer. One can learn what the controls do and why in a rather methodical way, however, knowing exactly *when* to use them to create special effects or achieve the desired balance with clarity and originality takes a good degree of artistic skill. In order to use this approach the engineer ideally should have a good understanding of the music and a strong identification with the composer.

Once again, I'd like to say that your magazine is excellent. It would be nice to see an article on a classical recording session.

—Carol Jay
No address

For a classical recording session in a unique environment, see our Feb/Mar 1977 issue, "A Session with the Boston Pops" and also Aug/Sept 1976, "Behind the Scenes at Tanglewood."

Teac and Otari—Take Two

The following joint response was elicited by a Talkback question. The previous answers had created considerable inter-company (Teac and Otari) controversy.—Ed.

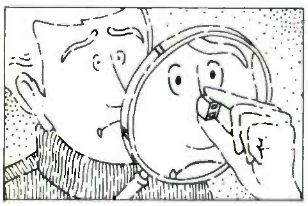
This letter concerns the Dec/Jan 1977 issue's Talkback answer entitled "Weighty Spec Problems." If you will recall, the question concerned a published s/n difference between the Otari MX5050-8D and the Teac Tascam series 80-8.

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

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

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

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



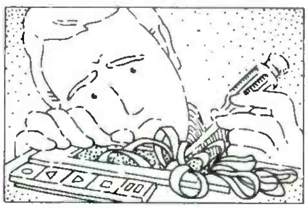
HEAD WEAR IS CAUSED BY YOUR RECORDER. OR IS IT?

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

Maxell is the only tape that has one.

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

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



JAMMING IS CAUSED BY YOUR RECORDER. OR IS IT?

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

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

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

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

DROPOUTS ARE CAUSED BY YOUR RECORDER. OR ARE THEY?

Maxell tape is made of only the finest polyesters. And every inch of it is checked for even the slightest inconsistencies.

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

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

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

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

Maxell Corporation of America, 130 West Commercial Ave., Moonachie, N.J. 07074

CIRCLE 42 ON READER SERVICE CARD

Bruce Gold, who originally asked the question was looking at a preliminary spec sheet on the MX5050-8D and was curious as to why there was a difference between the published specs of Otari and Tascam. Well, to resolve the question, MR sent both Teac and Otari a copy of Mr. Gold's letter and we each answered separately. We felt that it was a good idea to now jointly address ourselves to the question.

Specifications are a numerical representation of performance and unfortunately they can be quite confusing because of the many variables involved. We are in agreement that the most apparent difference between the specs was due to the kind of tape that was used to test each machine. Both Otari and Teac are now measured using the "hotter" level of Ampex 456, with the result eliminating the numerical difference. This is reflected in the new Otari brochure.

It is important to note that there are no technological secrets in the tape recorder industry. The main difference between products is the marketing and design philosophies of the companies making them. Each company believes in their decision as strongly as its competitor. You could put the designers and marketers of two companies together in a room for 12 hours and discuss such issues as: number of heads, tape speeds, types of equalization, etc. It is doubtful that after the 12 hours either of them would have changed their minds on these issues. As they say, "That's what makes horse races."

As to the 80-8 and the MX5050-8D, given equal situations, you can make as quiet a recording on one as on the other. Both are fine machines and both will give you excellent results. Both do the jobs they were intended to do and do them well.

The 1/4-track multichannel industry is a young one and we represent the chief manufacturers who build recorders for it. In that regard, we sincerely hope that the meaningless "specification wars" of the hi-fi industry will not be dragged into our industry. The result would only serve to confuse you, the customer, and cloud an otherwise bright and exciting field.

Keep on recording.

—Theo Mayer
Teac Training Dept.
TEAC Corp. of America
Montebello, Ca.

—Lou Barrett
Otari Corp.
San Carlos, Ca.

Unsnarling the Tangle

The facts, as I reported them, regarding the use of a delay line at Tanglewood ("Behind the Scenes at Tanglewood," Aug/Sept 1976 issue, p. 38) pertain to the relation of "live" sound from the orchestra on stage vis-a-vis the reinforced sound from the column speakers covering the lawn. Surely Mr. Schaffer (Letters to the Editor, MR Dec/Jan) must know that an electronic signal which travels at the speed of light will reach a given spot sooner than "live" sound when both sounds emanate from the same general spot. In a sense, as it was explained to me, it's a "race" between the sound of the orchestra and the amplified/mixed sound put into the wires. The former, by definition, has to take longer to travel the same geographical distance. It is for this reason primarily that the delay line is used at Tanglewood.

This explanation, of course, applies to classical concerts which as a rule do not use stage speakers. When it comes to other types of music, which often use stage speakers as part of the "live" presentation, other factors enter the picture of course. If Schaffer thinks they're doing something wrong at Tanglewood, maybe he ought to contact the crew there and enlighten them.

There was a slight error in my copy which no one has yet called attention to except the Tanglewood people and that is, simply, that their recording schedule does not include the pop-artist programs.

—Norman Eisenberg
Audio Editorial Board
Modern Recording Magazine

Valuable Primers

I am just beginning to learn the business of music recording, and I read the three parts of your "P.A. Primer." I not only found them totally helpful but also very enjoyable to read. This is why I have written to you. I have a problem. Due to unfortunate circumstances, I have lost the copies of *Modern Recording* that had contained the "P.A. Primer." I would be very grateful to you if you could tell me where I could get copies of the three parts of the Primer.

—K.M. Potvin
Walled Lake, Mich.

The "P.A. Primer" parts 1—3 appeared in the June/July 1976, Aug/Sept 1976, and Oct/Nov 1976 issues of MR. They are still available. Simply write to our Subscription Department and request them ($1.75 per issue).

Home Cookin'!

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

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

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

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

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

TASCAM SERIES BY TEAC

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

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

TEAC Corporation of America, 7735 Telegraph Rd., Montebello, CA 90640 ©TEAC 1977

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.

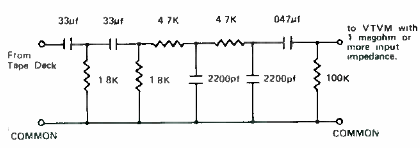
We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel we are skirting any issues, fire off a letter to the editor right away. "Talkback" is the Modern Recording *reader's technical forum.*

Recorder Levels, Part 2

Note: The following is an additional response to Thaddeus P. Floryan's question that appeared in the Feb/Mar 1977 issue of MR. —Ed.

Since you are correct in the procedure for testing frequency response, I will confine my answer to the latter three questions. First of all, a professional 7.5 ips test tape is recorded 10 dB below the normal operating level, because high frequencies will overload the recorded tape at that speed. By setting the output level to play back the test tape at 0 VU, proper recording will occur when you feed in a -10 record tone, and therefore you won't be saturating the tape.

In question 2, I think you are asking how one measures the s/n ratio of a particular *brand* of tape. You simply compare the level of signal on the tape that produces 3% THD to the weighted noise level of the tape. To test your *deck's* s/n ratio you'll need a distortion analyzer, oscillator and a VTVM capable of reading down to 1 millivolt full scale. It is now standard practice for manufacturers of sound equipment to state the s/n ratio in accordance with the USASI "A" weighting curve, which has a frequency characteristic similar to that of the human ear. The "A" weighting network shown is modified to prevent bias from affecting the readings.



A.

Next, feed the tone to the deck and bring up the level to 3% distortion at the machine's output, without the weighting network. Note this level. Now disconnect the oscillator and short the deck's inputs. Continue recording while measuring the noise level through the "A" weighting network. S/N ratio will be the *difference* between the two levels. *Note:* your deck must have less than 1000 ohms output impedance.

To answer your last question, the change in sound pressure or electrical power that produces a change of one VU, on a standard VU meter, is equal to one dB.

—Shimon Ron
Chief Engineer
Electric Lady Studios
N.Y.C., N.Y.

Control Room Monitors

First let me congratulate the staff of *Modern Recording* for the fine job you are doing. I am especially thankful for the Talkback section of MR! It is a very informative section!

Here is my question: Is it desirable to have your control room speakers set flat to the room (thru the use of a real-time analyzer) before adding any equalization?

This is the problem that I am currently confronted with: recordings that are mixed in my control room (which is rather small) sound muddy and distant when played on other systems outside the studio. It is my belief that due to the small control room, low-frequency signals do not receive proper dispersion, thus resulting in boosted lows which muddy up the mix when played elsewhere. Could that be the problem? Also, if all studios set their control room speakers flat to the room, a mix done in studio "A" would sound basically the same when played back at studio "B." Is this a correct assumption?

—Steve Morgan
Savannah, Ga.

In order to answer your questions I will attack them in reverse order. Your assumption that material played on speakers "set flat" in room A would sound the same in room B, is only partially correct. A number of other factors such as reverberation time, floor, ceiling and wall surfaces, and room volume and dimensions, all affect the sound quality. To produce two rooms with exactly the same response characteristics is an extremely difficult, if not impossible, task when one considers all of the variations permitted within the tolerances of equipment and material specifications.

Presence or absence of bass response in a control room is more a function of the acoustic design (or lack of it) of the room than of the actual size. I have seen some very large control rooms with erratic, inaccurate bass response and have also seen some small rooms with a good, clean, tight bass sound. I would guess that your small control room probably has a good coefficient of absorption in frequencies from 250 Hz and up due to the soft absorptive cloth, rugs or fiberglas on the surfaces, while the bottom suffers from an almost total lack of absorption. Properly designed bass traps, slot resonators or panels might be brought into play to relieve the problem you discuss.

And finally, to your first question. Here a discussion of the function of the analyzer and equalizer is in order. One analyzes the acoustic response of the room by using a real-time analyzer, a soni-pulse or other measuring device. Thus, the curve or the frequency response of the room is developed, and following study of this curve, corrections are initiated. These corrective measures include addition or removal of absorptive or reflective material, changes in wall angles or other dimensional changes and finally, equalizers. The use of 1/3-octave graphic equalizers offering cut-and-boost or cut-only capabilities in twenty-seven frequencies has become standard in control rooms. The desired goal is accuracy in the monitoring environment so that it is possible to hear material being produced through the monitor speakers without additional coloration from the room itself. Assuming that your monitor speakers are capable of reproducing the standard audio spectrum of 40 Hz to 16 kHz, the utilization of high quality equalizers should enable you to tailor the room response to a curve of your liking. Note, however, that an analyzer is a measuring device and does not set speakers to any response.

Good luck with your project.

—Hamilton H. Brosius
President
Audiotechniques, Inc.
Stamford, Ct.

Is It Working?

I've heard of sound men who've hooked up LED indicators to show that their individual speakers are functioning and/or when one is blown or malfunctioning. Would you describe a possible circuit for this device?

—T. Young
Thomaston, Ct.

Hooking up LED indicators to show speaker condition is possible, however, it will change the dampening factor of the amplifier drastically. For lower levels and levels far below the amplifier's rated output, the idea would be practical, but it would not be recommended for high-level applications.

It would be best to simply check the speakers before each concert and make sure that you have not allowed too much power to drive each particular speaker. The system can be designed so that when all amplifiers are running at their full output, there is adequate speaker load to handle that amount of

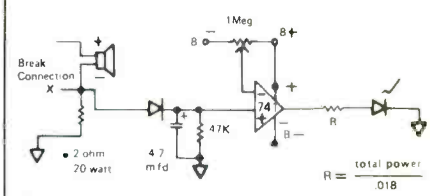


Fig. 1

power. Shown here (Figure 1) is a circuit that will indicate the current flowing in the speaker line, and key the comparator/LED indicator.

—Don Hartwig
Engineer
Heil Sound, Ltd.
Marissa, Ill.

dbx Noise

How much does dbx noise reduction limit the signal on tape?

—G. Crowley
Chicago, Ill.

The dbx tape noise reduction system achieves noise reduction by utilizing compression while recording and expansion during playback.

Compression is done at a factor of 2:1 over the entire dynamic range of the incoming signal. For example, an input signal of 100 dB dynamic range compressed by 2:1 would be reduced to 50 dB dynamic range. This compressed signal can be easily placed on tape above the noise and below the level of tape saturation. When the compressed tape is expanded during playback, the original 100 dB of dynamic range would be restored. The dbx system will provide in excess of 30 dB tape noise reduction (inaudibility) and 10 dB additional headroom.

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

Different Sounds

Why do European consoles sound "different" from American made consoles?

—Thomas Straks
San Francisco, Ca.

In an effort to deliver a complete answer to this question, I posed it to several of my industry associates. It seems most appropriate to begin with a quote from "designer extraordinaire," Dave Harrison of Harrison Systems.

"Any console with a sound of its own has something wrong with it. Control rooms have sounds, monitors have sounds, microphones have sounds, but consoles and tape decks should merely transmit and store those sounds."

When receiving the view of an English sound engineer, he agreed giving the credit of "different" sounds to the English engineers who "aren't as afraid of equalization as American engineers. Our attitude in the treatment of sound translation and building is different from yours, i.e. the Beatle drum mix and the type of mixing done by Ken Scott and Alan Parsons."

Nonetheless, and watching the thin ice upon which I tread, I shall transmit some observations of a meticulous and respected American engineer who points out that the "different" sound characteristics may originate from the use of single-ended power supplies of the British consoles which don't take full advantage of transistors and require the use of output coupling capacitors introducing phase shift in the output and resonances between the capacitor and the output transformers.

Also, the European consoles tend to overuse transformers which can cause phase rotation and various anomolies of distortion.

I believe this may begin to answer this question, but I have one. What do you mean by sounds "different?"

—Charles J. Flynn
Westlake Audio, Inc.
Los Angeles, Ca.

Your Own Studio

I am currently a communications student and practicing musician who is more interested in recording than any other phase of the industry. Since reading your magazine, you have really opened my mind on that subject.

I would like some information on a subject that I haven't seen in your magazine; independent recording studios. I would like to build my own and want to know what to look for and look out for.

(1) I will be the principal owner of the studio — will I need a technician's or engineer's license?

(2) What other licenses will I need to operate such a studio?

(3) What are some of the common pitfalls involved in such a venture?

Any additional information on the

subject (i.e. books, publications) would be greatly appreciated.

—Joseph Walker
Springfield Gardens, New York

Those of us in other areas of the industry share your enthusiasm for the informative role that MR serves.

In dealing with clients such as yourself, the questions you pose are amongst those which always arise early on in our discussions. Herewith, we'll attempt some incisive and pithy suggestions to your questions (in order):

(1) The only "engineer's or technician's license" we are aware of is the one possessed by all good recording engineers—guts. That's about the best word we can think of to describe that attitude which seems to possess most successful recording engineers. Associated descriptive terms might be inventive, courageous, tactful, intuitive, tenacious, artful and arty.

(2) If you are contemplating operating a commercial enterprise, you will of course, be subject to local and state requirements which regulate commerce. You will have to acquire a business license and other license which relate to such a business (tax bond, health and safety certificates, public utilities). We suggest you check with your local Chamber of Commerce for additional information.

(3) The most important step you can take is initiate and maintain a close relationship with a reputable company which specializes in designing studios and providing equipment and training. Too often we are contacted by clients who have attempted a do-it-yourself approach only to find themselves more confounded than they were at the onset of the project. Also, avoid the so-called stereo shops which "can get" studio-type equipment but can't tell you how or why to use it. Secondly, read, read and then read some more before you make any commitments to buy anything (see no. 4 below). Thirdly, the location you pick for your studio will be of extreme importance. Solicit as much information as you can relating to acoustics and studio design. Your books and magazines (no. 4 below) will help here. Other studio owners are obviously a great source of knowledge on this subject ("Don't do what I did"). A little homework on your site location will save you much time and money and will allow you to determine what ratio of your total budget must be spent for room treatment and isolation. Also, make certain you've done some work on

"...The Sansui TU-9900 qualifies as a true 'super-tuner'..."

Stereo Review, March 1977

" 'Super' FM tuners are usually priced from $1000 up. Sansui's new model TU-9900 tuner, at (under) $450,* matches their performance..., at least in the most important respects".

Popular Electronics, January 1977

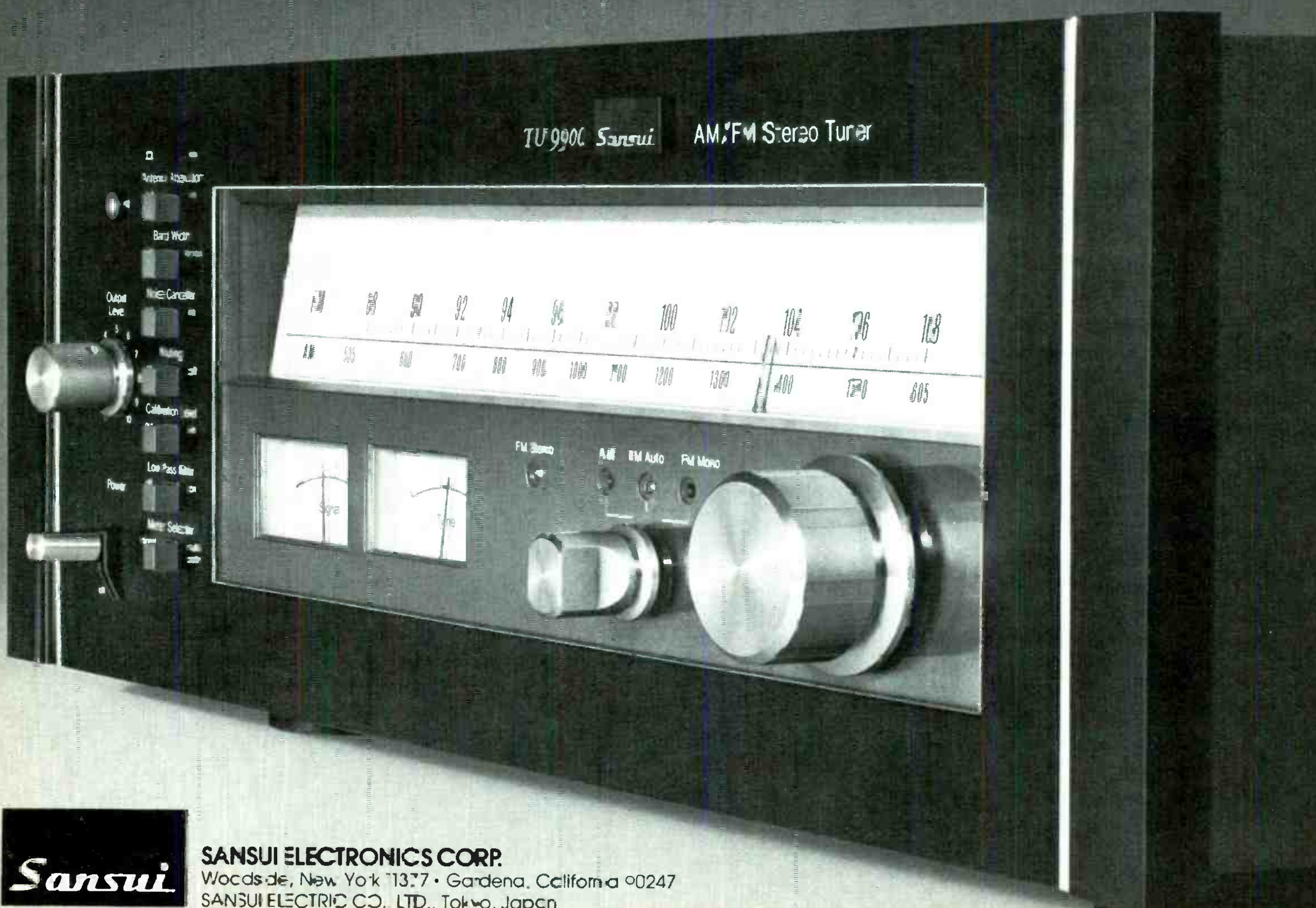
"... The TU-9900 is highly selective ..... In the wide-band mode, the IHF sensitivity was ... 1.7μV in mono ... indicative of very high performance ..... S/N at 65dBf (1000μV) input was one of the best we have measured ....The distortion at 65dBf was also lower than we have ever measured on an FM tuner ..... Mid-range separation was not only the greatest we have ever measured, but far exceeded the guaranteed rating of our Sound Technology 1000A FM generator ..... Overall, we would consider the TU-9900 to rank with the other 'state of the art' FM tuners ..." Stereo Review

"The Model TU-9900 ... is an ideal mate for the highest quality amplifiers and speaker systems. ... Image rejection was unmeasurable, exceeding the 100dB range of our test equipment. ... Stereo channel separation was almost as unbelievable as the distortion figures, exceeding 60dB from 60 to 600 Hz ..... Clearly, the Sansui Model TU-9900 tuner is a very superior performer ... [and] any untoward sounds heard via this tuner originate from the FM station ..... It's a top value unit." Popular Electronics

Sansui offers a complete line of highest quality amplifiers and matched tuners, the AU/TU series. Visit your nearest franchised Sansui dealer today for a top value—in the price category right for you.

*Approximate nationally advertised value. The actual retail price will be set by the individual dealer at his option.

A whole new world of beautiful music.



Sansui

SANSUI ELECTRONICS CORP.
Woodside, New York 11377 • Gardena, California 90247
SANSUI ELECTRIC CO., LTD., Tokyo, Japan
SANSUI AUDIO EUROPE S.A., Antwerp, Belgium • In Canada: Electronic Distributors

CIRCLE 54 ON READER SERVICE CARD

the marketing end of things. Does your location allow you sufficient access to enough potential clients for a supportable bottom line? What competition will you have to confront and how will that competition affect your pricing policies? Lastly, don't underestimate the time required to start and maintain a successful studio. Initially, you can expect to spend every available hour on your new business. You can also expect to "never have enough time." Remember, in the studio the old adage applies two-fold; *Time is money!*

(4) Fortunately, there are several books and magazines available which expound on the music business as it specifically relates to recording. A listing follows:

db Magazine, Sagamore Publishing Co, Inc., 1120 Old Country Road, Plainview, L.I., N.Y. 11803.

Recording, Engineer, Producer, Recording and Broadcasting Publications, 1850 Whitley Ave., Hollywood, Ca. 90028.

Modern Recording Magazine, Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050.

Modern Recording Techniques by Robert E. Runstein, Howard W. Sams Co., Inc., Indianapolis, Indiana 46268.

Studio Sound, Link House, Dingwall Avenue, Croyden, England CR9 2TA.

Recording Studio Handbook by John Woram, Sagamore Publishing Co., Plainview, N.Y. 11803.

—John J. Boyle
The Express Sound Company, Inc.
Costa Mesa, Ca.

Resurfacing, Anyone?

I am the operator of a small studio near Charlottesville, Va., where we do jingles, dubs and soundtracks for educational television. Through the selection of a number of pieces of equipment of relatively low cost, and their careful use, we have been able to put out a consistently good product, and most important, have been able to meet those inevitable "deadlines." We do all our master recording on ¼-inch tape, using an Ampex two-track and a TEAC 3340 quarter-track. My questions to you revolve around the recording and playback heads for these machines.

As the recording time builds up, (sometimes as much as 18 hours per day) so does the oxide on the heads, especially after the first thousand hours or so when wear becomes apparant, so we change them. (We demagnetize and clean several times each day). We are now using Nortronics replacements, and find

MIX-MASTER

When you're onstage trying to whip up your own personal blend of sound, you want a mixing system that gives you maximum control. That's where the Acoustic 860 really delivers. □ The better you can monitor your performance, the better it's going to sound. That's why each of the 860's eight input channels has two level controls, so that you can submix the monitor and the main independently. And both the monitor and the main have 9 band graphic equalizers to protect against feedback. □ The 860 gives you big board flexibility on each channel. LED overload indicators, low impedance balance inputs, 16 dB cut and boost on channel equalization and line level effects inputs add up to the kind of control onstage that you usually expect to find only in a studio. □ The 860 Mix-Master, backed by our unique Protection Department service, is on its way to your authorized Acoustic dealer. We think it's going to whip up a lot of excitement.

For more information about the 860 please write to
Department MR, Acoustic Control Corp., 7949 Woodley Ave., Van Nuys, CA 91406

CIRCLE 64 ON READER SERVICE CARD

ourselves with a box of used heads of various brands on hand.

(1) Can modern heads be resurfaced? I remember years ago doing a session at a studio where the head engineer was speaking of "refacing."

(2) Is it feasible cost-wise, remembering that the heads in question cost about $100 each to resurface.

(3) Who, if anyone provides such a service? We would appreciate any suggestions you might have on the subject, as we have appreciated almost everything you have printed since issue no. 1. We are loyal fans hoping for a monthly.

—Paul Brier
Creative Music & Advertising
Trevilians, Va.

The answer to the first part of your question is yes, this process is available. Resurfacing or lapping, as it is sometimes called, is a process whereby the worn heads are restored to new condition. A properly resurfaced head, when put back in use, will perform as a new head and will have as much wear as a brand new head. In essence, a resurfaced head is equivalent to it s new counterpart in all respects.

As to the second part of your question, the answer is also yes. While the cost of resurfacing will vary somewhat, you can expect to pay between $25.00 and $35.00, plus tax and shipping, for a ¼-inch head stack. As you can see, this makes using this service feasible, cost-wise, since you end up with a head in new condition for quite a bit less money.

There are many companies that perform this service whose names and addresses are available from local parts distributors. One such company that I am familiar with is Grandy Inc., 1275 Bloomfield Avenue, Fairfield, New Jersey 07006. Their telephone number is (201) 575-1433 and they charge $25.00 per ¼-inch head stack.

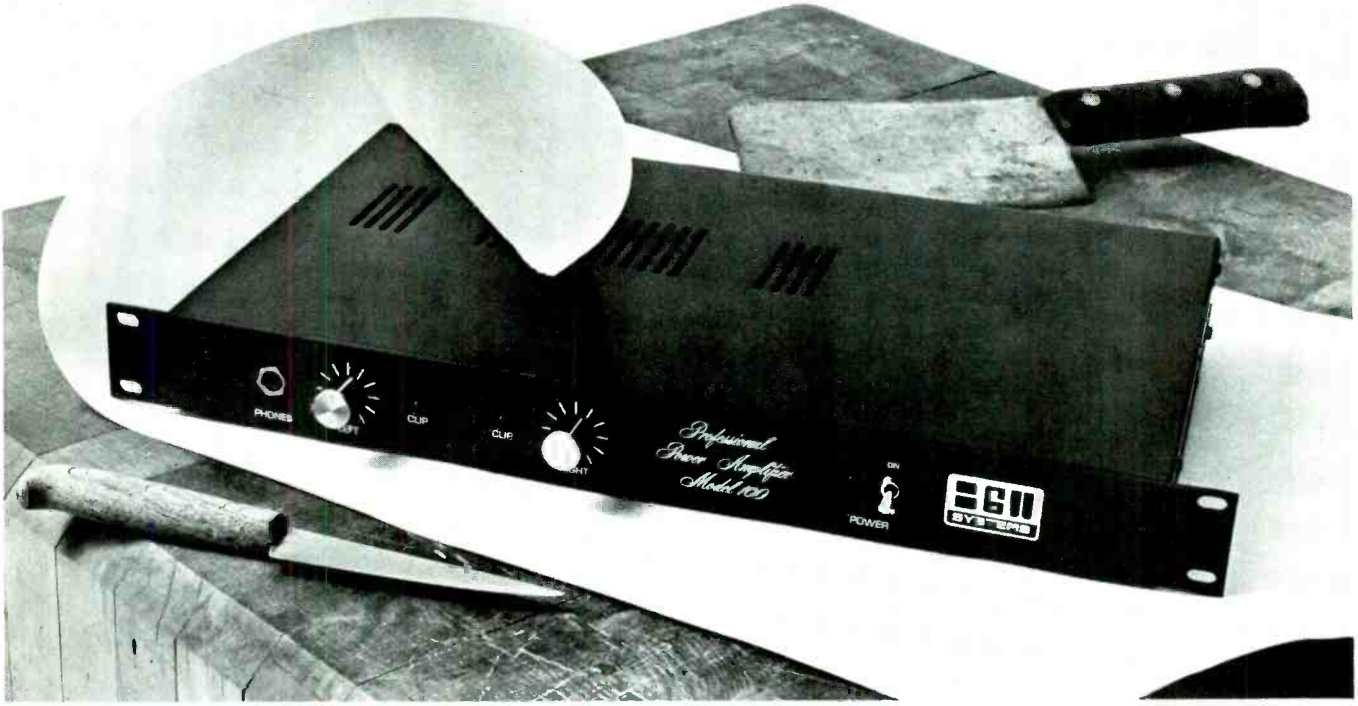
I must also mention that heads which have been excessively worn can not be resurfaced. It is advised that the heads be inspected prior to resurfacing to see if this process is worthwhile. This is usually done by the resurfacers as they have the necessary measuring equipment. You should make sure of this inspection.

—Joe Gennarelli
Generation Sound Studios
New York City, N.Y.

Homemade with Teac

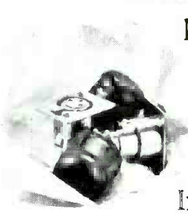
In August 1975, I purchased a Teac 3340S for my home studio. Since none of our dealers would put one of these

BGW PRIME



BGW is about to whet your appetite with their new Model 100 Stereo Power Amp.

The Model 100 will drive the most difficult loads you can throw at it. Electrostatic headphones and reactive speakers are driven flawlessly due to the 100's unique design. No form of current-limiting is used whatsoever. A precision monitor amp for any application, the modular construction of the 100 means one integrated amplifier circuit board with the biggest heat-sink we could package in 1-3/4" of vertical rack space: 340 square inches of efficient sink. No more thermal shutdowns.



There's a sophisticated "loss-of-feedback" circuit with front panel L.E.D's. Also unique to the 100, the L.E.D's are driven by a one-shot circuit for precise clipping indication. Instantaneous peaks, which get by most amps, are stretched out in our L.E.D. circuit so you can see them.

On the inputs, you can get optional Cannon XLR's and plug-in transformers...and into 8 ohms you get 33 watts per channel, or 80 watts mono. Into 4 you get a meaty 44. The I M distortion is an incredibly low .01%.

It's about time somebody like BGW would mete with the competition. Check out the Model 100. It's just one of six professional prime cuts...of course, all above the rest.

BGW SYSTEMS

BGW Systems 13130 South Yukon Ave., Hawthorne, Ca 90250 (213) 973-8090
In Canada: Omnimedia Corp., 10245 Cote de Liesse, Dorval, Quebec H9P 1A3 (514) 636-9971

CIRCLE 80 ON READER SERVICE CARD

machines on the display shelf because of the cost of doing so—meaning they were too expensive to stock—I decided to buy the machine solely on the literature Teac makes available along with the record, *Homemade with Teac*. The quality that the group "The Hello People" got from their 3340 is fantastic and I for one, would like to know what type of equipment other than the 3340 was used to make this record; such as mics, mixer, delay and reverb units. I am ordering a four-channel Dolby for my machine which I need desperately. I also would like your opinion on a good quality tape for multi-track recording. I understand that Scotch 207 with the backcoating is as good if not better than TDK, Maxell and others, and since it is available in 10-inch pancakes, that it would be cheaper to buy in bulk paks. My reel-to-reels consist of the Teac 3340, Sony TL650, and Pioneer RT-1050.

—Ken Karns
Tar Heel Publishing Co.
High Point, N.C.

The *Homemade with TEAC* album was produced almost three years ago—and it *does* sound very good. The album was produced in two stages, the first stage was the music. The "Hello People," recorded with a 3340 (the old one with the lever) and a 3340S. The Model 2 was in its final development stages, but not on the market yet. So they used a comparable (though not as flexible) unit. They had an SAE graphic equalizer, Dolby "B" and a hi-speed two-track. As for the echo in "Creego" (Side 2, band 2), they used their tape recorder and the spring from a P.A. head.

The *big secret* to hi-quality sound, (even for a budget studio) is hi-quality microphones—"Garbage in, garbage out" as they say. The "Hello People" had three mics in the "several hundred dollar category. One Neumann for their vocals and two Sennheiser 405's for overheads on the drums. They did have separate rooms for control room and recording room and used JBL 4310's for monitors.

Stage two of the album was the assembly of the album itself. Combining the music portions with the announce voices, effects etc... In places we wanted to break the music down to four parts plus an announcer and effects. This takes six tracks—so we took the music from the 3340 and transferred all four tracks onto a TEAC Tascam Series 70H8 and added the announcer.

Then the final mix of the album was

To be professional a speaker should be:

Flat,[1] Wide Range,[2] Rugged,[3] Loud,[4] Uniform, Wide-Dispersion.[5]

What else could you possibly want?

Handsome,[6] Small,[7] Low-Cost.[8]

1 & 2. 45 to 18,000 Hz ± just 3dB. Or 32 to 18,000 Hz with optional equalizer in step-down mode. A result of sophisticated Thiele/Small vented speaker technology. You choose either 4th or 6th order Butterworth filter response.

3. Withstands 30 watts long-term or up to 300 watts short-term (10 milliseconds). Tweeter protector included. Professional enclosure construction.

4. Delivers 111 dB with 300 watt peaks. 96 dB SPL at 1 meter with 1 watt.

5. 126° horizontal dispersion ± only 31° from 400 to 16,000 Hz. 66° vertical dispersion ± just 16° from 4,000 to 16,000 Hz.

6. Satin finish oak and matte black with protective black edging is quietly contemporary.

7. Less than 4 cubic feet. Just 20″ wide, 28½″ high, and 11¾″ deep. Mounts readily to wall or ceiling. Tweeter rotates for same room coverage whether vertical or horizontal.

8. At $291.00 a remarkable achievement. Optional SEQ Equalizer $60.00. Prices suggested net, slightly higher in the West.

The new E-V Sentry V.

Send today for a fact-filled data folder on the New Sentry V. It will thoroughly prove that even your smallest space can enjoy good sound. Or better yet, ask your E-V sound specialist to arrange an audition. We promise a remarkable experience.

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Buchanan, Michigan 49107

EV a gulton company

All of the above.

CIRCLE 40 ON READER SERVICE CARD

mastered on a TEAC 3300 SST through our mixer. The exact same machine that mastered Rick Ruskin's album on Tokoma.

I have talked to several 3340 owners who ask—"How do I make it sound that good." Which is a little like asking how do you get to play in Carnegie Hall. Parts of the *Homemade with TEAC* album went four generations plus through disc mastering and pressing (which is pretty brutal). If those tracks still sound a lot better than yours right off the machine, take a hard look at your microphone locker and keep working on it. To quote the album: "Tape recorders don't record just because you stick the plug in the wall." A lot of the quality of that album is plain ol' fashion good engineering and good production.

—Theo Mayer
Teac Corp.
Montebello, Ca.

Question the Primer

I read, with much interest parts 2 and 3 of "P.A. Primer" (Vol. 1, No. 6 and Vol. 2, No. 1). In part 3 in the hypothetical sound set-up, they were using two mixing boards, one at the mixing tier and one at stageside. Is it necessary to use two mixing boards in concert sound reinforcement and why?

Also why do some mixing boards have several submaster modules in addition to a master module?

I am also interested in locating Vol. 1, numbers 1, 2 and 3 of *Modern Recording*.

—R.R. Yeager
Shreveport, La.

Thanks for the kind comments on the P.A. Primer! As for your first question, I guess we did not fully cover why two mixers are used (we could have written a book on this). The split mixer approach is always expected in a pro system for this reason: The stage monitor mix is controlled at the stage and is totally independent of the house mix. This way, the performers can have their own balance and can have good communications with the monitor balance engineer who is hearing the same sound mix as the musicians. If the stage monitor mix was being controlled from the audience position, the house mix engineer would not be able to hear the exact balance that the musicians on stage hear. Additionally, the performers would have a difficult time conveying any changes they wanted in their monitors ("Test-Test is this mike on? Joe, turn up my vocals, please") due to the fact that the man controlling their mix is maybe 100 feet away in the audience.

As for your submaster question, I'm not sure what mixer you are referring to, but in many mixers the submasters allow "sub grouping" of inputs. This means that in a four submaster mixer, submaster no. 1 could control vocals, no. 2 could control drums, no. 3 could control guitars, and no. 4 could control keyboards. Thus if it is decided that more vocals are required in the mix, the no. 1 submaster would be turned up accordingly instead of having to turn up all the individual vocal input volume controls.

Submasters also allow additional control in other specialized areas, but each application tends to become unique. If you have further questions concerning this, contact *Modern Recording* with specific details.

—Brian A. Roth
Ford Audio & Acoustics, Inc.
Oklahoma City, Ok.

Note: Sorry, but MR issues 1–4 are no longer available –Ed.

At $795, it's a great little synthesizer for beginners...

Sure, Chick and Herbie use a Micromoog. They're hardly beginners. So with lots of heavy electronics to choose from, why do they play a Micro?

Not because it's easy to play, though it is. And not because it's easy to buy, though it costs less than many acoustic guitars and conventional keyboard instruments.

The pros play Micromoog because for all its simplicity and economy, it's packed with all the great Moog features they need to shine on stage. Including some of the latest synthesizer innovations.

The new Moog ribbon controller is one of them. With a slide of your finger, you can glide up and down and bend pitch as bluesy as B.B. King. Release, and you snap back to the key you're playing instantaneously.

The sound at your command? Though Micromoog is light enough (20 pounds) and compact enough (24" x 15" x 15½") to go just about anywhere, it gives you a full eight octave range and loads of electronics. Like a modulator wheel controlling vibrato and trills. Reverse contouring. Sample and hold. Variable wave shaping. Rapid filter modulation for haunting bell sounds. Blendable suboctave doubling. There's even a noise generator for a thunderstorm of imaginative and realistic sound effects.

Pitch drift, incidentally, can't happen with Micromoog. No matter how much heat you generate on stage (and you will, you will), Micromoog's special oscillator control circuit maintains itself precisely at 130° F. And that's cool.

So if you haven't gotten into synthesizers because you've been put off by price or complicated electronics, put it off no more. Micromoog's an easy to play, easy to pay way for beginners to get into synthesizers.

Just ask Chick and Herbie.

like Chick Corea and Herbie Hancock.

Another quality product from Norlin

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CIRCLE 50 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

EIGHT-TRACK RECORDER

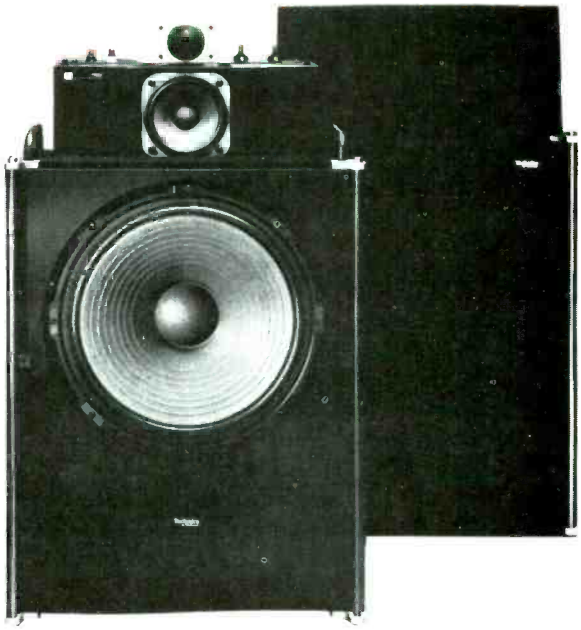
New in Teac's "Tascam" series is the model 80-8 tape recorder, a 15 ips, 1/2-track model with eight tracks and eight channels for record and play. Aimed at the professional who wants to create a master tape, the 80-8 also has other applications in audio/visual production. According to Teac, using the half-inch format at just the one speed has enabled them to hold down the price to "less than $3,000."

The unit has three heads—erase, record/reproduce, and monitor—which are controlled by three buttons in the output select: input, for source calibration; normal, for most modes including sync and reproduce; monitor, for off-the-tape monitoring. An optional dbx module (encode/decode processor) allows up to 30 dB of noise reduction. Peak levels are shown on LEDs. The head cover is hinged

to facilitate editing and cueing, and the eight-meter section may be opened to provide access to calibration controls. Designed to handle the 10½-inch (NAB) reel, the model 80-8 weighs 76 pounds and measures 21 inches high, 17½ inches wide and 12 inches deep.

CIRCLE 10 ON READER SERVICE CARD

LINEAR PHASE STRESSED IN NEW SPEAKERS



Literature describing a new line of speaker systems from Technics by Panasonic emphasizes phase linearity for "precise waveform duplication." Three models have been announced. The SB-7000A is a 3-way system using bass-reflex loading and can handle up to 150 watts of peak power. It stands 33¼ inches high and weighs 72.8 pounds. Price is $399.95. The model SB-6000A is a 2-way bass-reflex system of about the same height, but weighing 55 pounds and rated for 100 watts of peak power input. Its price is $299.95. The SB-5000A is another 2-way bass-reflex system 28¼ inches high and weighing 35.2 pounds. Rated for peak power input of 75 watts, it costs $159.95. Special consideration in all three models has been given to crossover networks and to alignment of drivers as part of the phase linearity design effort.

CIRCLE 12 ON READER SERVICE CARD

TIME MODULATOR

Said by its manufacturer to be the "first voltage controlled, time sweepable analogue delay line," the Marshall Time Modulator (TM) employs a non-quantizing time delay design that allows the user to dial the exact delay or phase cancellation desired, without the error of a step system. One knob controls delay, with no patch cords or switches. Delay can be changed noiselessly while in use. It is possible to program delay or phasing swings from any remote source, allowing drive interfacing with most synthesizers, joysticks and foot pedals. If no control voltage is present, an internal low-frequency oscillator is normalled to the control jack. This unit was designed to consolidate and handle all fast time base processing being done today, and also make available a new spectrum of effects said to be previously not available, such as true automatic double tracking; discrete triple tracking; time shift flanging or phasing; doppler panning and shift of any signals, exceeding two octaves; true vibrato; Leslie effects; cardboard tube echo.

CIRCLE 19 ON READER SERVICE CARD

CLASS G POWER AMPLIFIER

Among new product offerings from Hitachi is its "Series E" power amplifier, model HMA-8300, rated at 200 watts per channel. Listing for $750, the new amplifier incorporates Hitachi's patented Class G circuit design which features a unique secondary power output stage said to be able to provide up to 400 watts per channel of transient music power within rated distoriton. According to a company spokesman this design permits the HMA-8300 to offer the performance capabilities associated with

power amps rated at 400 watts per channel while maintaining the lower heat sink requirements and higher efficiency of a 200-watt per channel power amp. Companion piece to the HMA-8300 is the $350 model HCA-8300 preamp.

CIRCLE 5 ON READER SERVICE CARD

NEW DEVICE LOCATES TAPE CUES

For use with its professional Series 79 multi-track recorders, the 3M Company's Mincom Division has announced its Selectake II unit which is a tape-position locator for fast, accurate cueing. Described as a powerful microprocessor, the device is self-contained in a compact calculator-style case for remote operation. Programming a cue during a session involves merely entering a "store" command and the digitally-displayed time on a keyboard. Up to nine separate cues can be thus stored. No information is put onto the tape itself. Cue recall is accomplished by touching the recall key, number key (corresponding to store position) and locate key

which moves the tape drive at high speed, stopping within one count of the cue point to eliminate overshoot. Full tape-control functions are provided on the control panel. Tape motion, digital time and locate readouts cover minutes from 0.00 through 99.99. The display is "frozen" upon tape run-out to permit re-threading without loss of location. The unit also features automatic "hi-low" speed switching with manual select for 7½-15 and 15-30 ips ranges. The Selectake II is priced at $1,750.

CIRCLE 17 ON READER SERVICE CARD

DYNAMIC MIC

New addition to the series of professional microphones from Unicord is the model DM 138, a dynamic uni-directional cardioid type. Rated response is 50 Hz to 15 kHz. The ball-shaped microphone is supplied with a 15-foot two-conductor cable, a 3-pin cannon-type female connector, and a male-type phone connector, plus a stand adapter and vinyl carrying case. List price for the DM 138 is $89.95.

CIRCLE 3 ON READER SERVICE CARD

NEW SPEAKER LINE IN U.S.

Visonik brand loudspeakers—made in Berlin, Germany by Heco-Hennel & Co., K.G.—are now being handled in the U.S. by Visonik of America, Inc., based in Oakland, Ca. There are two lines. One is the David series of five compact speaker systems priced from $95 to $259 each. The other is the VL series of four systems priced from $69 to $167 each.

Also available—for the David models 30, 50 and 60 systems—is a sub-woofer or "super-woofer" designed to enhance the bass response below 80 Hz. Known as "The Bottom End," this unit is sold to mate with a stereo pair of the other systems. Price is about $300.

CIRCLE 6 ON READER SERVICE CARD

NEW NOISE FILTER

From Burwen Research, Inc., a subsidiary of KLH, there's a new dynamic noise filter. Dubbed the model DNF 1201A, the device features a bandwidth controller circuit which measures the high-frequency content of the sum of left and right inputs from the source material and adjusts the bandwidth in accordance with level and with frequency. Dynamic filtering is accomplished as the bandwidth controller generates DC control voltages to constantly regulate cut-off frequency of the filter. The variable low-pass filter has a cut-off frequency from 500 Hz to 30 kHz with an attenuation rate of 9

dB/octave. At 10 kHz, up to 38 dB of noise attenuation is available. Tape-hiss reduction is rated from 5 to 14 dB. Unit price is $379. Full specifications and details are available from the manufacturer.

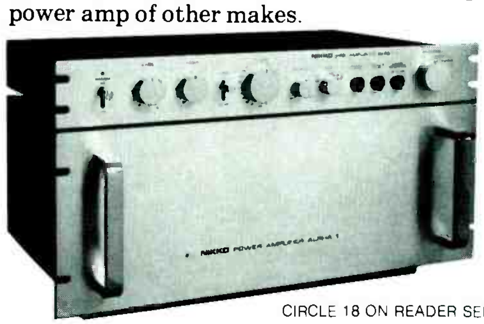
CIRCLE 5 ON READER SERVICE CARD

NEW AMP AND PREAMP

Alpha 1 and Beta 1 are the names, respectively, of a new power amp and preamp-control being offered as part of Nikko's new professional product line. Both units are designed for rack-mounting. Preamp inputs include two phono sockets, auxiliary and tuner. A tape monitor feature (play 1, play 2, dub 1 to 2, dub 2 to 1) also is included. Some published specs include: S/N on phono inputs, 72 dB; S/N on high-level inputs, 100 dB; phono 1 and 2 frequency response, 30 Hz to 15 kHz (within 0.01 dB) for high-level inputs. Preamp price is $299.95.

The power amp is rated for continuous output of 220 watts per channel (minimum RMS) into 8 ohms, from 20 Hz to 20 kHz with no more than 0.08% THD. IM distortion is listed as less than 0.08% for rated output; frequency response is given as 10 Hz to 100 kHz within −1 dB. Retail price is $599.95.

While both units are offered as "companion" units, either may be used with suitable preamp or power amp of other makes.



CIRCLE 18 ON READER SERVICE CARD

TAPE CLINIC EXPANDS

A traveling seminar-and-demonstration on recording tape, given by manufacturer TDK Electronics, is scheduled for expansion in the coming months, following a "highly successful cassette tape clinic program" in 1976, according to a company spokesman. The program features a testing system for tapes in which consumers can see graphic representations of the differences in cassette tape performance. Touring nationally and set up at trade and consumer shows as well as in TDK dealer stores, the program uses a Nakamichi 1000 cassette deck, a Hewlett-Packard 181A oscilloscope and H-P 3300A function generator to present displays of response and noise curves for two different tapes at a time.

This year the traveling clinic will add college campuses to its itinerary as part of the "Great American Music Machine" coordinate by Communications Resources, Inc. and sponsored by Pioneer and MCA records.

MIXING CONSOLES

Soundcraft Electronics Ltd., the British firm, recently showed some of its extensive series of professional studio equipment, including mixing consoles of varying capabilities, options, and features, ranging from a 12/4 type priced at about $500 to a 24/8 model priced at $9,682. Details are too varied and numerous to go into in this space, but literature may be obtained from the manufacturer at 5-8 Gt. Sutton Street (4th floor), London EC1VOBX, England.

CIRCLE 14 ON READER SERVICE CARD

QUAD SOUND TODAY, OR MAYBE TOMORROW

Some of my contemporaries are insisting that quadraphonic sound is not "dead." That may be, but if it is "alive," it still has to prove itself as a healthy, viable format. I suspect that much of the current anti-anti-quad sentiment stems from wishful thinking by many who are beguiled by the promise, rather than convinced by the performance, of four-channel sound.

It is significant, I believe, that one of the major promoters of CD-4 admits, in a recent issue of the journal *CD-4 Forum,* that research and development "remains the most active area in quad, be it discrete or matrix. Laboratories have been fairly bristling with new developments which make quad performance of just a few years ago quite obsolete."

Take special note of the latter part of the second sentence which is an astonishing admission from a group that has been vigorously promoting just such quad performance as if it were the answer to everyone's musical and sonic quest. Ironically, this admission comes very closely in the wake of another booklet, issued late last year, in which this statement—which now seems outrageous in light of the recent statement—appears: "Now quadraphonic sound has truly come of age." For something to come of age and become obsolete in the space of two or three months is truly a modern miracle. Unfortunately, the miracle that was hoped for—that hundreds of thousands of listeners would rush out and support this achievement—did not come to pass. And so back to the drawing board.

Apropros of which, it is debatable as to whether the laboratories are "fairly bristling" or whether a few engineers in a corner are struggling to come up with something that will not be openly admitted to as being obsolete two years from now. At that, we are cautioned (in this same statement) "that breakthroughs in the laboratory are at least a year and a half ahead of any production in the field." Since this material was issued with a "Dec/Jan 1977" dateline, we can assume then that maybe, sometime in 1978, the new age of four-channel sound may finally dawn.

With these grains of salt digested, we can look at some of the reported developments. For one, JVC has shown new demodulation circuits "destined to make the conventional phase lock loop (PLL) obsolete in future designs." What we now hear about is a phase tracking loop said to make for cleaner carrier recovery from CD-4 discs while also permitting higher modulation levels to be applied to them. The other technique is a double phase lock loop which is claimed to effectively place high-frequency interference signals outside the demodulation range.

While you ponder what this means to those who already have invested in "conventional" or "obsolete" demodulators, be heartened by the news of yet another evolutionary step in the history of the SQ matrix decoder—this one described by its manufacturer as a "super-performance parametric" unit. Designed by Peter Scheiber, it is said to bring to "SQ disc and broadcast reproduction a higher level of separation and lower distortion than any other quadraphonic disc system."

Both the new CD-4 demodulator, and the new SQ decoder may indeed be great improvements in their respective areas, but what their simultaneous appearance suggests is simply this: That the whole subject of quad sound still has no definite identity except to partisans of one system or another; and while these matters are of undeniable interest to insiders and professional sound men, they have yet to jell into anything of genuine or lasting importance to the mass of consumers.

Maybe next year?

MUSICAL NEWSICALS

The National Association of Music Merchants held their Western Market trade show at the Disneyland Hotel in late January. Since many manufacturers time the introduction of their new products to coincide with the NAMM trade shows, we've been so inundated with new product information that it will take several issues to tell you about everything we saw in California. This month we'll concentrate on products from new and lesser-known companies, and in the next issue or two will bring you up to date on new products from the more established manufacturers.

SOUND REINFORCEMENT... Tangent Musical Engineering (2810 South 24th Street, Phoenix, Arizona 85034) showed five new pieces of sound reinforcement equipment. Models 801 and 1201 are mixing consoles of eight and twelve inputs respectively. Each input channel includes a preamp gain control with LED peak overload indicator, bass and treble equalization with center detents for flat response, pre-fader Monitor send and post-fader Effects and Reverb sends, plus a rotary main fader. Inputs are low-impedance and balanced, connected via XLR connectors, and high-impedance connected via ¼-inch phone jacks. The main output channel has controls for Reverb Return from the built-in spring reverb (fed by the reverb send), effects return for external effects devices fed by the effects send and an auxiliary high-level input in addition to the master fader. The monitor output channel has bass, mid, and treble equalization and a rotary master fader. Both main and monitor outputs are low-impedance and balanced, and use XLR connectors. Metering of the two output channels is via peak-reading LED displays.

Models 1200 and 1200eq are power amplifiers rated at 200 watts RMS into 4 ohms, and capable of delivering some 350 watts RMS into 2-ohm loads with less than 0.25% THD. Inputs are balanced, via XLR connectors, or unbalanced, via ¼-inch phone jacks. The amp is fan-cooled and uses a fast-acting resettable circuit-breaker rather than conventional current-limiting circuits. The front panel features a level control and peak-reading LED meter plus LED indicators for AC power and overload. In addition, the 1200eq features a three-band parametric equalizer which should be very useful in reducing or eliminating feedback modes.

The bp6030 is a self-contained, bi-amped, three-way speaker system. An electronic crossover and power amps for 60 watts RMS of bass power and 30 watts RMS of treble power are built-in so that the only connections necessary are AC power and a line-level signal feed from a mixing console.

Among the various speaker systems offered by Gollehon Industries (210 Front Street NW, Grand Rapids, Mich. 49504), the self-contained two-way systems are

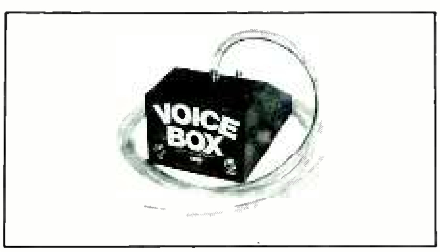
of particular interest. The Penetrater 1 is a compact loudspeaker specifically designed for use on speaker stands for portable use, or ceiling-hung in a distributed array for permanent installations, although it should also be excellent for use as a stage monitor. The Penetrater 1 uses an extended-range 12-inch speaker for bass and a pair of Gollehon's T-200 diffraction horns for the high end, is rated at 100 watts RMS power handling, and weighs in at 60 pounds. Models 8201 and 8202 are relatively compact front-loaded horn enclosures which provide more substantial bass output and higher power handling than the Penetrater 1. The 8201 uses an extended range 15-inch woofer and a single T-200 treble horn; the unit weighs 75 pounds and is rated at 125 watts RMS power handling. Model 8202 uses two extended range 12-inch bass speakers and a T-200 treble horn resulting in a 200-watt RMS power handling capability and a 90 pound weight. All Gollehon professional series loudspeakers feature tongue-and-groove construction and super-rugged black fiberglas resin finish and are furnished complete with carrying handles.

Mitchell Speaker Cabinet Co. (Riverside, Ca.) offers a very wide range of musical instrument and sound reinforcement speaker systems. Of special interest are the company's sand-filled guitar speaker cabinets and extra compact bass guitar cabinets. Each of the sand-filled models has some 12 to 15 pounds of sand hand-packed in the walls of the box to damp out cabinet resonances and are finished on the inside with fiberglas resin. This construction, combined with the acoustic design of the enclosure, is said to result in high efficiency and smooth response in a very compact physical package. Mitchell Sand Cabinets are available in 1x12" ($150), 2x10" ($200), and 2x12" ($260) configurations. The latter two versions are also available with a built-in amplifier at $400 and $460 respectively. All these models are covered on the outside with Tolex vinyl fabric, and include a kick-proof metal grille and latch-on cover for safe transport. The Mini Bass speaker ($250) is a 12-inch driver in a compact folded horn enclosure measuring only 30"x18"x18". The Model R-25 ($300) is somewhat larger at a still compact 26" x24"x20" and houses a 15-inch speaker in a half-horn enclosure, and the model R-50 is a similar design for an 18-inch driver. All three models have a fiberglas resin interior finish and a vinyl-covered exterior, but are available in all-fiberglas finish at slightly higher cost. All Mitchell models are available at extra cost with Altec, Gauss or JBL speakers instead of the standard Mitchell Lineal Transducers.

The Edcor MP-7V is an 8-input stereo mixer utilizing VCA technology for maximum versatility (Edcor, 3030 Red Hill Ave, Costa Mesa, Ca. 92628). Each input channel has the following conventional features: low impedance, balanced microphone input via XLR connector; balanced, line-level input via ¼-inch jack; five-position attenuator switch (3 mic positions, 2 line level positions); pre-EQ, pre-fader Foldback level control; low, mid and high frequency EQ with 20 dB of cut or boost; post-EQ, post-fader Echo send control; panpot; and cue (solo) button. The output channels feature foldback master level, echo send master level, monitor/cue volume, separate echo return controls for left and right channels, left and right master faders and cue pushbuttons for foldback, echo send, and tape. The unusual features of the mixer are the line level access jack for each input which allows multi-track recording or insertion of accessory equipment (such as a limiter) on any individual input, and the voltage-controlled level-set access jack which allows the voltage-controlled amplifier in each input channel to be controlled individually or in groups by an external control voltage. This latter feature allows for extremely versatile operation with multi-channel submastering, remotely controlled mixing, or even interfacing with an electronic music system.

Rapidly gaining popularity on the West Coast, particularly among country-rock musicians, are the Risson amplifiers (Risson Musical Instruments, 2108 S. Wright Street, Santa Clara, Ca. 92705). The Risson line ranges from the SSG50V-210, a self-contained amp delivering 50 watts RMS into two 10-inch speakers, up to monstrous modular systems delivering up to 450 watts RMS into multiple 6x12" cabinets. Particularly versatile is the SSG100-210 Studio-Stage Stack. This model delivers 35 watts to its self-contained pair of 10-inch speakers, but when the 6x10" slave cabinet is connected, the power output increases to 125 watts RMS. The SSG 100-212, which comes in a "standard" version for clean sound at high volume or a "rock" version for overdrive at low volume, works much the same way, delivering 40 watts into its own pair of 12-inchers or up to 125 watts when properly loaded with external speakers.

ACCESSORIES. . . A/DA Electronics (26034 Eden Landing Road, Suite 4, Hayward, Ca. 94545) offers three interesting electronic sound modifiers. First is the Auto-Flanger ($189.95), which is a variable analog delay line used to simulate the tape-flanging effect. The Auto-Flanger has both automatic and manual sweep modes with a "width" control which both limits the width of the sweep and shifts actual control of the sweep between the automatic and manual modes. Other features include an "Enhance" control which regenerates the delayed signal for more intense effects, an "Even / Odd Harmonics" switch, and a high/low sensitivity switch for optimum results with normal or lower-level input signals. The Flanger Pedal ($199.95) is basically the same circuitry housed in a die-cast aluminum pedal designed to give musicians "hands-off" control of the most used functions. A switch at the forward end of the pedal travel activates or bypasses the flanger circuitry, and a switch at the back end of the pedal travel switches the function of the pedal from manual sweep to a rate control for the automatic sweep. Both A/DA flangers have a .4 milisecond to 10 milisecond delay range, and are supplied with AC adaptor. More unusual is the Harmonic Synthesizer ($469.95), which produces a harmony note for every input note. The harmony interval can be adjusted any-



where from two octaves below to 1½ octaves above the input note. The unit is said to use a micro-computer and digital memory system which allows use on chords as well as single notes, unlike conventional octave dividers.

"Talk boxes" are hardly a new idea—the "talking steel guitar" records of the 1950's were done with them—but they are certainly a hot item right now after the success of Peter Frampton and his use of a talk box. Dean Markley Strings (2333 El Camino Real, Santa Clara, Ca. 95051) offers their version, the Voice

Box, in three different models with varying efficiencies and power-handling capabilities; the Model 50 ($120), Model 100 ($129), and Model 200 ($150) are rated at 50, 100 and 200 watts RMS respectively. All three models use Electro-Voice compression drivers and come with 6½ feet of non-toxic vinyl tubing. Also manufactured by Dean Markley Strings is the Mighty Mouth ($149), a similar product using a University driver, which is sold through Coast Wholesale Music, a West Coast distributor.

Another "talk box" device is made by the Mitchell Speaker Cabinet Co. (Riverside, Ca.). Mitchell's unit is called The Bagman ($150), and is rated at 150 watts RMS and has a built-in protection device. Frequency response is given as 80 Hz-12 kHz, which is quite remarkable since most talk boxes use treble-type compression drivers which have very limited low-frequency response.

Paraclete Productions (P.O. Box 5013, Sherman Oaks, CA 91413) has a complete line of electronic sound modifiers including the usual assortment of wah-volume, fuzz-wah, fuzz-sustain, tone modifying and phasing unity. More unusual are the Auto-Wa ($139.95), an oscillator-controlled wah-wah, the Deluxe Foot Phaser ($127.95), a pedal-controlled phaser with LED rate indicator, and the Wa-Sustain ($130.26), which combines a wah-wah with a compressor which is said to have a 90 dB range for unequalled sustain. Unique to the Paraclete range is the Chopper ($69.95). This unit pulses or chops the input signal and features a Duty-Cycle control to vary the effect from staccato to sustained, a foot-pedal to control the rate of pulses and an LED rate indicator. All Paraclete pedals are available in a variety of finishes including bright chrome plating.

EQUIPMENT: what should it cost TO FIX?

Let's face it—no matter how good your tape deck or mixing panel, power amp or digital time delay device, the day's going to come when something goes wrong in your audio system. When it does, how much is it going to cost to fix, and what else should you know about audio servicing?

First, it's not going to be cheap. The major cost in all electronics repairs is labor. No matter where you live in the United States or what's wrong with your tape recorder, it's going to cost you a minimum of $22.50 to get it fixed. That's the price charged by May Electronics of Jackson, Miss., and even that doesn't include parts. If the machine is in bad shape, the cost could rise to $32.50 (or even more)—that's the starting level for service technicians in Los Angeles and New York.

If your equipment is under warranty, of course, there's nothing to worry about. Either drag it down to a "factory-authorized service center" or bundle it off to the manufacturer and let him worry about it. You're stuck for the shipping charge—sometimes one way, sometimes both—but the manufacturer will make the repairs as long as the warranty is in force and you have complied with its provisions.

But warranties have a nasty way of running out, usually before the trouble starts. Then it's up to you and your bankroll. Let's suppose that your problem is in a tape deck which has lost a channel, a common malfunction. You bundle it up and lug it down to the service station.

But which service station? The phone book is full of them. Perhaps even there's a guy up the street who operates an electronics repair business in his garage. How about him? Altogether, there are some 11,000 people in the United States who claim to be electronic technicians. In some states, they must pass some sort of test before they begin operating on your equipment. In most states, they don't.

Whenever possible, you're better off dealing with a service station that's been authorized by the manufacturer of your tape deck, mixer or power amp. The prices tend to seem higher than those of your friendly neighborhood TV repairman, but there are benefits. In the first place, the factory-authorized warranty station is likely to have a service manual for your unit on the premises. That means that the service technician doesn't have to waste a lot of time (at your expense) learning the circuit. Chances are he's serviced dozens of similar units, and knows where to look for the most likely trouble spots. While his time per hour may be worth twice that of the general practitioner, it may take him less time to find the trouble and cure it, thus saving you money in the long run. Even more important, he's likely to have the parts for your unit in stock, instead of having to order them and wait. The proliferation of brands and models in recent years has made it impossible for any service technicianor any parts wholesaler–to stock parts for every single piece of electronic gear. But factory-authorized warranty stations do try to stockparts for the equipment theyservice. And if they do have to order a part, they usually can counton getting what they need more quickly than can the independent serviceman.

Okay, so you've found out who your local factory-authorized serviceman is. You put your unit up on his counter. What are you letting yourself in for?

In those states and cities with aggressive consumer protionlaws and agencies, you're likely to find a schedule of rates hung on the wall, so that you can see for yourself. Some years ago, most servicemen and service agencies charged so much an hour, plus parts. Nowadays, most factory-authorized service agencies charge a flat rate, or a minimum and maximum rate, depending on the type of equipment. A tape deck in New York costs a minimum of $35 to service at Component Service Corp., whose rates are typical. In San Francisco, the minimum rate is $30 to $32.50, while it's down to $22.50 in Mississippi. However, the minimum rate in San Francisco and New York is also the maximum rate in the vast majority of cases, while in Mississippi, it's designed to cover the first hour on the bench. If the recorder is really in bad shape, prices in Mississippi rise to $32.50 plus parts; in southern California to $65 or so.

These days most established service businesses will give you an estimate before they begin work. Prices for estimates range from $4 in northern New England and Mississippi to a median of $7.50 in New York, Chicago and Los Angeles. Some technicians charge as much as $15. If you decide to have the work done, the service agency usually absorbs the estimate fee. If you decide to take your trade elsewhere, you pay. In some cases, you've got to make your decision on the spot. May Electronics' four-dollar estimate charge jumps to $11.50 if you wait overnight to accept the estimate.

As we've seen, the cost of servicing your unit depends on several variables—where you live, the kind of service technician you choose, the complexity of the circuit you're having fixed and what kind of condition it's in.

"You're presenting me with so many variables that it's impossible to give you a specific answer," said Len Gaynor, Harman-Kardon's customer service manager when I asked him.

Four-channel equipment is the most complex, according to most service managers. In New York, you won't get your four-channel amplifier or receiver back for less than $45. Next come tape decks and stereo receivers. The going rate in New York is $35. A tuner or power amp will cost you a minimum of $30. None of the shops surveyed for this report post prices for repairs to equalizers, mixing panels or signal processors—some because they don't do the work, others because they've had very few calls for such work compared to the volume of work they perform on conventional audio equipment. Those who do handle noise reduction units, digital delay units, mixers and other similar equipment said that the minimum flat rate would depend on the complexity of the equipment and its condition. While New York service agencies can charge $45 for four-channel gear, servicemen in parts of Pennsylvania, the upper midwest, and Alabama, Georgia and Mississippi don't even list it. "Sure, I'd service a four-channel unit if it came in," a technician in suburban Louisville, Ky. said, "But there weren't that many sold around here. I haven't seen one yet." Asked what his minimum flat rate would be, he replied, "I dunno, about the same as an open-reel tape deck, I guess." That was $27.50. Asked what he'd charge for a control panel, he replied that he'd never serviced one.

Getting service for some of the more exotic audio equipment now available is more of a problem than paying for it. While there are thousands of service organizations qualified to take your stereo tape deck apart, there are perhaps a tenth as many qualified to work on your mixer or synthesizer. Unless you live in an urban area near one of them, that usually means one of two alternatives—packing it up and sending it back to the factory, where experienced technicians equipped with the necessary parts can work on it; or handing it over to a technician who may understand electronics, but doesn't know your unit. Then you're in the same boat with the audiophile who's paying a TV technician to learn the insides of his Crown amplifier or Sony tape deck. Since these servicemen don't have enough experience with semi-pro gear to establish a flat rate, they usually charge an hourly rate which may range anywhere from $10 to $22. Yamaha's audio division's service department.

By Robert Angus

one of the few "authorized" agencies which does charge an hourly rate, gets $20, while independent serviceman John Snyder of Selinsgrove, Pa., gets $10 an hour.

"Sometimes the customer comes out ahead with our hourly rate, and sometimes it's cheaper for him to pay the flat rate of an authorized service agency," says Yamaha's Greg Class. "It all depends on how long it takes us to find the trouble and fix it."

That's where you run into trouble with the hourly rate, particularly if it's inclined to be low. The meter starts ticking from the moment the serviceman puts your unit on his workbench. Each technician has his own check-and-eliminate procedure to trace the trouble in a circuit he doesn't know well. If you're lucky, the first thing he checks will be the cause of the trouble, and your bill will be pleasingly low. It rarely works out that way. The trial-and-error procedure can take two or three hours in some sophisticated audio equipment just to discover what's wrong. Then the technician usually can fix it fairly quickly—provided he's got the parts, and provided that the problem he solves isn't caused by a malfunction somewhere else in the circuit. In that case, you can blow $10 or $15 worth of parts and an hour or two of time without accomplishing anything.

Time and Turnaround

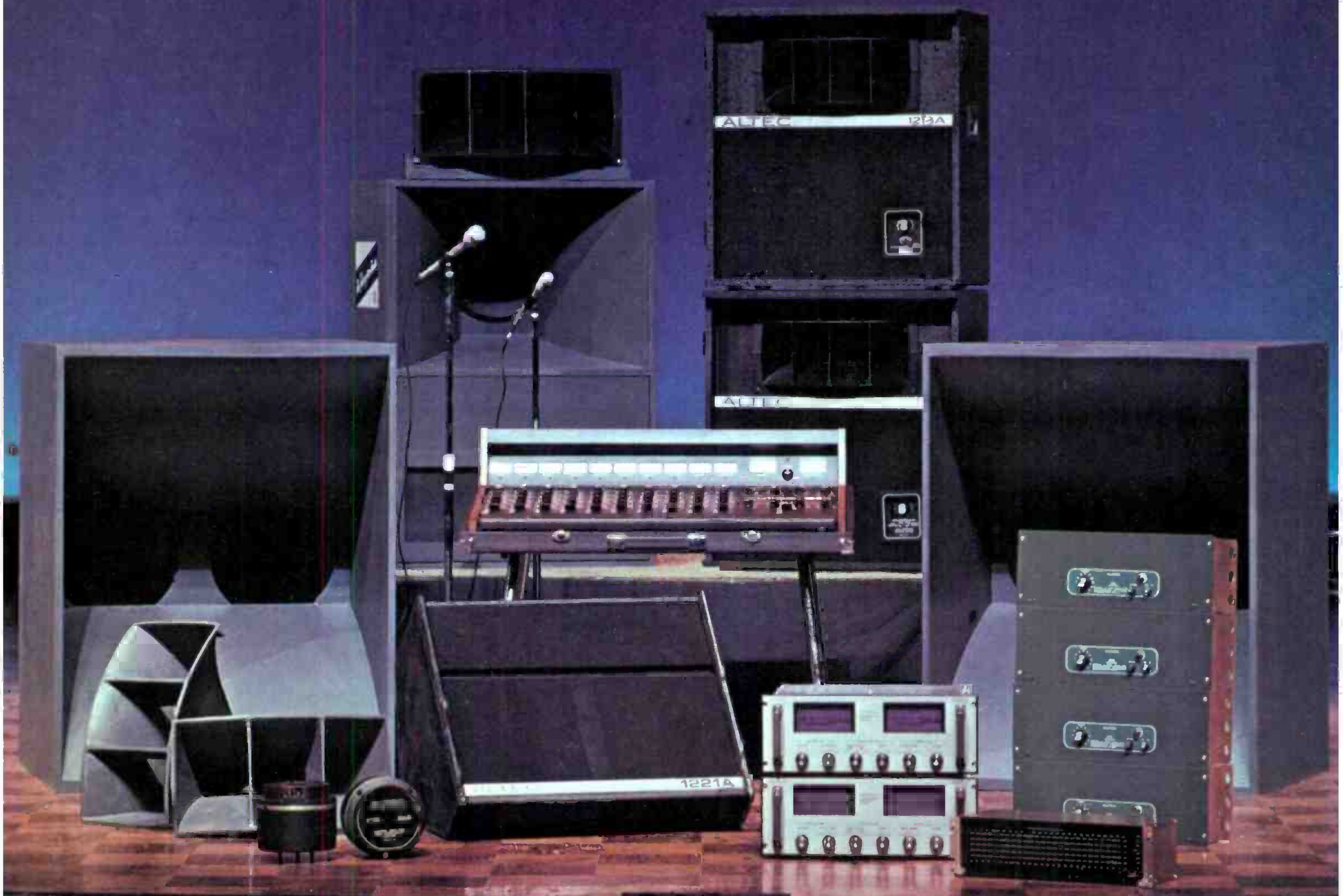
The moral of all this is that, in any given community, the hourly rate really doesn't tell you anything, and you can't usefully compare it to a minimum flat rate. What may appear to be the lowest price may end up being the highest if the serviceman charging it isn't lucky or doesn't know your equipment. The prices charged by servicemen and service agencies do reflect something—the value of the technician's time, the cost of overhead such as rent and advertising, the cost of special training or maintaining a large inventory of parts, and much more. While the man working out of his house can charge less per hour than the manager of a store on the main street in town employing twelve other technicians, he can end up costing you as much or more because he doesn't have the diagnostic tools or skills of his bigger competitor.

There's another important difference. The turnaround time in most manufacturers' service departments is three to five days. That's the amount of time which elapses between the arrival of a piece of equipment at the receiving dock and its exit out the door on its way back to its owner. To that, you must add elapsed shipping time, which can vary from a few days to a few weeks, depending on the time of year and the shipper you use.

Time of year has lots to do with turnaround time at your favorite service station, too. If it maintains a reasonable supply of parts, you can expect a turnaround of anywhere from four to forty days. At press time, New York area stations were reporting turnaround in the seven to twelve day range. In Minnesota, two agencies said turnaround was more like a month because some crucial parts were out of stock, while in Mississippi, one agency blamed a three-week turnaround on the loss of one technician.

If the serviceman you select doesn't have the necessary parts in stock, he's got to get them. That means, in urban areas like Dallas or Chicago or New York trying to buy them locally or, for the average serviceman, ordering from a distant wholesaler or perhaps even the manufacturer. Smaller service companies don't order parts every

THE DEPENDABLE PERFORMERS



No glory. No fame. No glitter. But every bit a star.

The pros. They're on stage night after night. Performance after performance. The business is in their blood.

But there are other pros too. Like Altec sound systems. Most of the time they're not even seen—but you know they're there.

The Altec "Voice of the Theatre" heritage offers the dependability, clarity and versatility that performers of all kinds recognize the world over.

The dependability acquired through 40 years of experience. The clarity for which the "Voice of the Theatre" has long been famous. The versatility to meet the varied demands of professional musicians. Altec sound systems—the dependable performers.

Location photography courtesy of Knott's Berry Farm

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ALTEC CORPORATION

CIRCLE 30 ON READER SERVICE CARD

day; the technician may set aside one morning a week to go shopping at his distributor's. That means delay, and in some cases it can take forty days or longer to get equipment back.

Turnaround time usually is shorter with the larger service agencies because they're geared to getting equipment in and out quickly, they have more servicemen to work on the equipment, and they want to build repeat business where possible. One-man service agencies are more likely to be swamped with work at some times of year and stalled by unavailability of parts of a service manual on specialized equipment at other times.

Helping Yourself

Are there things you can do to cut down on the service charge? Well, if you're dealing with a technician who charges an hourly rate, you can save him some time by narrowing the problem down as much as possible. That recorder we started out with, the one with the defective channel. Which channel? Is it completely dead, or is there an input signal on the rear-panel input but not on the mic input? If so, then you've narrowed the trouble to the mic input, or perhaps the microphone itself. You can rule out that possibility by switching the mic to the other channel. If you're now getting a signal on that channel, the trouble is indeed with the alternate input or amplifier section. By reversing inputs and outputs, it's possible to narrow any audio trouble to the offending component, and frequently to determine where in that component the malfunction is occurring.

A technician who charges a flat rate may find all of this interesting, but usually will perform his own diagnosis, just to be sure. It doesn't cost you anything, and it may turn up a problem you didn't know existed. But if you're on the hourly rate, every minute counts, and any time you save the serviceman is money in your pocket.

You can also save money on tape deck repairs by keeping your machine clean and demagnetized. Dirt is the main cause of under-warranty service complaints, according to several company service managers. A regular program of cleaning and lubricating the tape guides and rollers, and of keeping dust and grit out of the mechanical parts of your recorder will pay off in fewer visits to the repair shop. Demagnetization of recorder heads at regular intervals will keep the recorder performing according to manufacturer's specifications.

Finally, you can operate the equipment within the guidelines set forth in the owner's manual. Stacking power amps without providing enough ventilation can impair their performance and shorten their life. Using a tape deck improperly can produce a satisfactory recording, but may result in premature aging of the transport. And so on with all equipment.

Aside from these common-sense procedures, there's really very little you can do to eliminate service costs. They happen to the best equipment, with time. When it does happen, don't go shopping for a bargain-basement solution. There isn't one that's satisfactory. Instead, find the best, most experienced service technician you can—one who's got a parts inventory and an expensive overhead. One who knows your equipment. It'll certainly pay in the long run.

What's Cookin'?

Why it's none other than Fabulous Felix and the Flamethrowers, the hottest band this side of Dante's Inferno.

But while Felix is burning up the stage with all his visual pyrotechnics, the Flamethrowers' sound isn't exactly setting the world on fire. There's more synthesizer and lead guitar in the bass monitor than there is bass. And all those instruments cooking together are cremating the vocals. What Fab Felix and the boys need at this point is a little less incineration and a lot more separation. And that's where a Tascam Series mixing console comes in.

If they'd simply install a Model Three or Model Five between their songs and their sound system, they'd have the same precise control over their sound during a live performance as they have at a recording session.

And after the gig, they could take their trusty Tascam equipment back home, connect it to any one of Tascam's multitrack recorders and turn out studio quality tapes. Which makes the Tascam at least twice as good as any single-purpose mixer.

The Tascam mixing consoles. Created to help you sound as hot as you look.

TASCAM SERIES BY TEAC

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

Take a spin on our new fully automatic turntable. And leave the direct driving to us.

With Sony's new PS-4300, you just sit back and enjoy the ride. Wherever the record takes you.

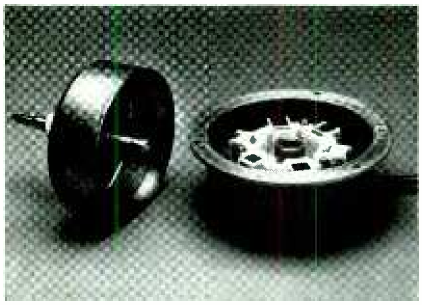
That's the blissful simplicity of a fully automatic turntable.

But the PS-4300 is more than purely practical. We like to think of it as a model union: combining the convenient and the complex.

It is a profoundly engineered machine, with intelligent design slashing through down to the smallest detail.

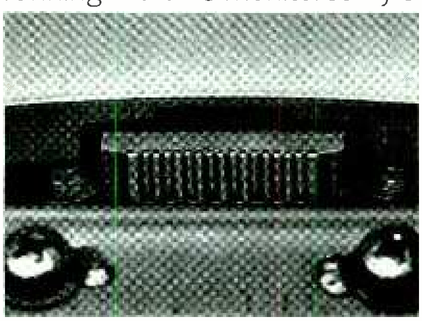
We gave brushes the brush.

The motor that powers the PS-4300 is brushless and slotless. Direct drive, if you will.



This deceptively simple construction makes for a smooth-running motor with less friction and noise than traditional DC motors. And it eliminates cogging.

What's more, this smooth-running motor is monitored by a



smoothly-engineered 8-pole magnetic pick-up head. And our magnetic speed sensor works through an intricate electronic feedback system; driving the platter directly—without a jumble of belts and pulleys getting in the way.

So our torque is not a turkey, and we've got low wow and flutter and high speed stability to boot.

An electric eye. For your ear.

Hands off the PS-4300!

Our optical sensing system automatically returns the arm when your record is over.

Optical sensing is light years ahead of the conventional mechanical linkage. Eliminating the pressure and distortion you'd ordinarily get at the end of a record.

A tone-arm that's a strong arm.

Now we're not calling anyone clumsy. But there *is* the chance you might make a mistake and grab hold of the tone-arm while it's in motion.

That's why the PS-4300 has a tone-arm that's more than just statically balanced. It comes with a protective clutch device. (The only clutch you'll find on our fully automatic turntable.)

This latching set-up protects your arm against too much strain.

Moving from arms to feet, ours are designed to cut feedback. They're rubber-soled: suspended by cup-shaped rubber shock absorbers.

And they're adjustable, letting you level the turntable. So you might say our feet come with elevator shoes.

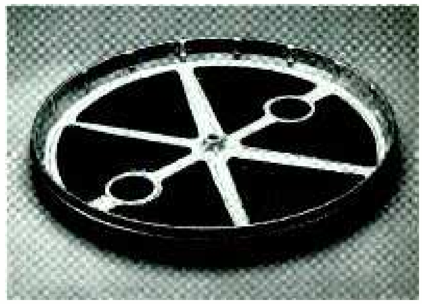
Our vibration-reducers are great shakes.

Sometimes the cabinet itself can vibrate—distorting what comes out of it.

Not so with the PS-4300.

Our cabinet is built out of a material with a low Q. Low Q materials hardly vibrate, and nobody watches their P's and Q's like Sony.

Even our platter has been undercoated with a damping material.



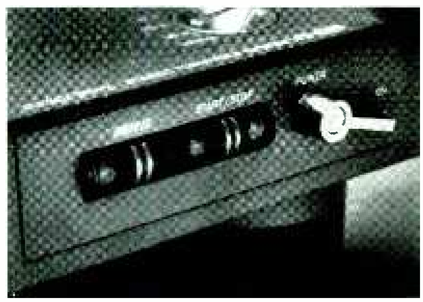
And what looks like a bad case of acne on our record mat is a series of bumps that provide an air cushion and absorb vibration.

An exercise in self-control.

You can see that we've covered just about everything when we created the PS-4300.

Even the cover.

Our dust cover is ingeniously simple. When closed, it leaves the controls accessible.



And what controls they are! One-touch, LED-indicated switches for start/stop and repeat.

One light tap starts everything going, while your record, under the dust cover, is in splendid isolation.

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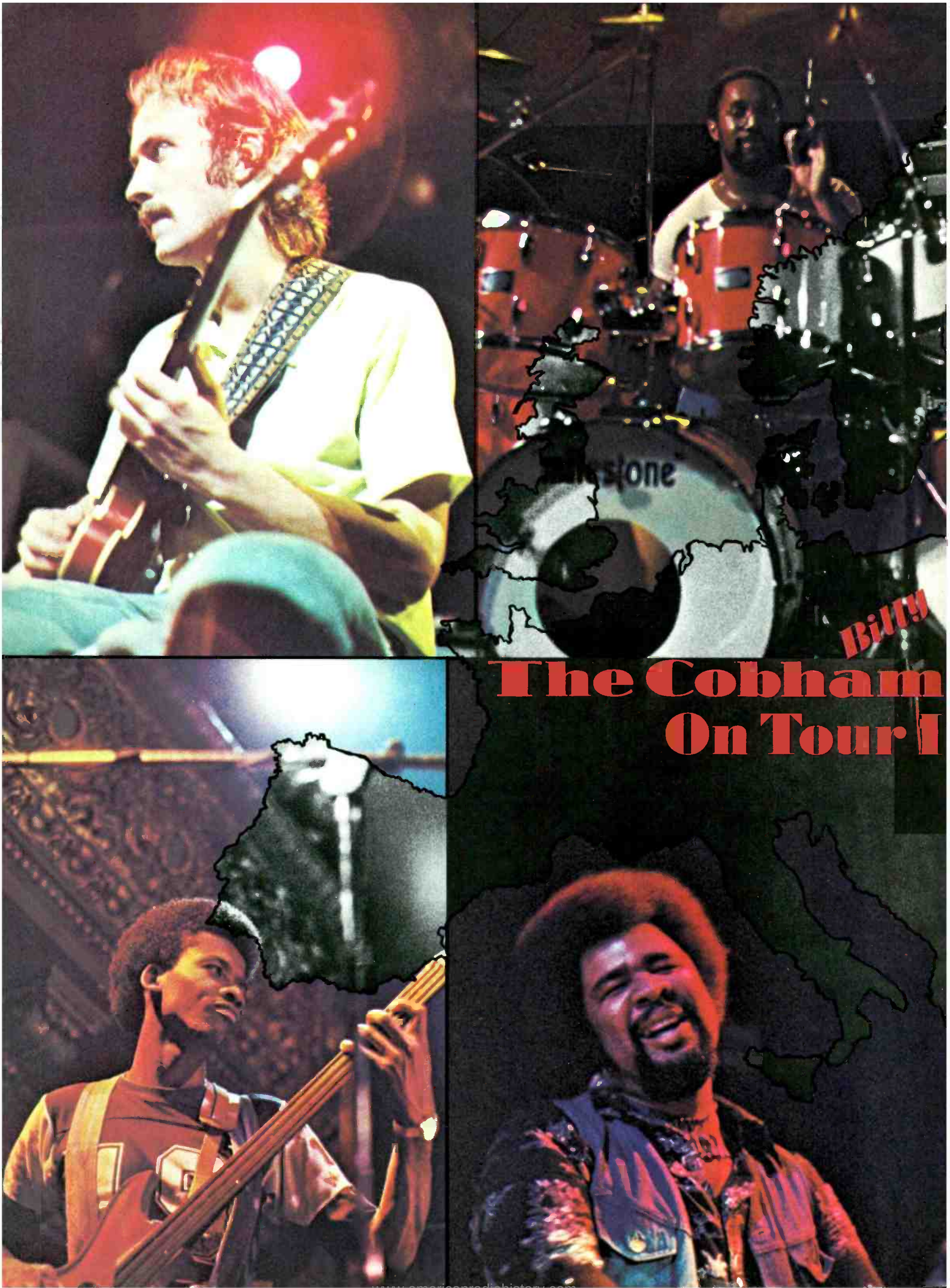
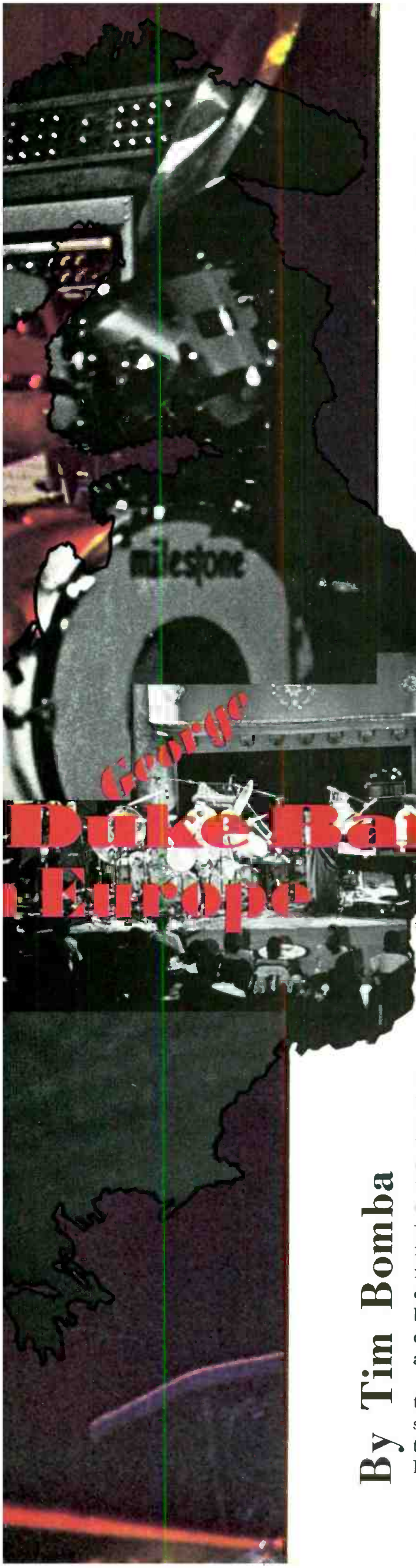

Billy
The Cobham
On Tour!



George Duke Band Europe

By Tim Bomba

Acid Jazz—it's a term with which the majority of the musical public is not yet familiar. The term, like the music, is new—self-coined to describe the new, electric, high-level Billy Cobham/George Duke Band. Cobham, known to many not only as a great drummer and percussionist, but also as a pioneer in the "new" jazz, has teamed up with pianist George Duke, whose keyboard talent is surpassed only by his zany stage show. Joining Cobham and Duke are Alphonso Johnson, bass and John Scofield, guitar. It's a band; there are no on-stage superstars, just four great musicians who *do it.* And there are four others who do it: Ken Graham, production manager, lighting designer and drum set-up coordinator (and the reason that most shows turn out to *be* shows); Warren Wallace, road manager (responsible for *everything*); Bruce Heigh, keyboard technician and monitor mixer; and me, sound engineer, responsible for guitar and bass set-up, technical maintenance and most audio concerns. Take this most intricate—musically and technically—band, add forty-nine road cases, mix in a thirty day European tour, complete with language barriers, unstable political situations and no time off, and you have my tour with the Billy Cobham/George Duke Band.

The Tour

Scheduling is the most important part of a tour. Where will the largest audiences be? Will the equipment make it from one show to the next in time (i.e., long mileage, poor road conditions), will the border officials be at the crossings at 4 a.m., or must the drivers—driving three full semi's—wait until opening at 8 a.m. and lose four hours driving time? Will all facilities at the concert location be adequate? These are all considerations to be taken into account. And you wonder at times if the agency [booking] and management realize these points.

When we didn't fly, the crew took a touring bus, complete with bunks, seats and tables, in order to arrive at the location for scheduled "load in." Finishing at 4 a.m., bus to the hotel, take a shower, back into the bus, sleep for six hours, up again and back on another damn stage. Fourteen of us were included in this particular schedule. We pull into the gig. It's 9 a.m. I push open my curtains to see that a few have been awake for a while, and others are still asleep. Ola, our driver, has just driven fourteen hours straight, taken us across two borders with no problems and managed to drive a ten-foot-wide bus through what seem to be eight-foot-wide streets. We all start waking up, and I wonder who will get sick today. A member of the tour, Leon, drank the water one day and got intestinal flu the next day. Another member, Klaus, was even harder stricken and had to stay in one place for two days to recuperate. I fell over into a water canal and had four stitches put into my chin during the early morning hours in an emergency room in Switzerland (partial exhaustion, not intoxication).

We are all hungry, but realize that most promoters don't feel an obligation to give a crew breakfast. A crew that works from 10 a.m. to 3 a.m. to put on a show so that this promoter can make his money. No breakfast, though we often are the ones who must figure out how to produce a show despite the problems and lack of organization that afflict some promoters. *Man,* a good promoter you always remember . . . but a bad one you *never* forget.

People don't realize that the crew, commonly known as "roadies," (a word that should be stricken from the English language) often has to speak English, French, German and Spanish. Along with having to make sure that the stage crew is there to help "load in," change sets and "load out." And, of course, if the stage crew is not there, we have to do it all. But, we do it. Frequently with no sleep and on one semi-nutritious meal a day served up by the illustrious, aforementioned promoter.

And you wonder if some promoters read the Rider—the information sent to each city in advance of each show to explain such things as the stage size, electrical requirements and dressing room conditions. All of which have specifications to be met before a smooth flow can exist. Then you have to guess if your equipment will work on a ten-percent (plus or minus) variation of line voltage. The hall electrician (if there is one) doesn't understand English, let alone electricity. At times like this, I'm glad we have a sound and

Photos by Hojones & Tim Bomba

light crew of seven people who can handle the power problem, speak the necessary languages and figure out why we have seventy-two volts between power ground and microphone ground. No, not *all* performances plagued us with such matters. But, then again, where was the interpreter we requested, where was our eight-man stage crew, where was our food? So, what happens? We all become interpreters, we all become a stage crew, and we all don't stop hounding until we see our one meal guaranteed (?) to us in the Rider. In reality, the *only* guaranteed thing is that as long as the promoter pays his money for the show, it goes on, somehow. In one country we even had a political riot one night during the opening act. Soldiers with rifles were on hand the next night. Whether they were on hand to insure that we *did* the show or to insure that a riot would not *stop* the show, we did not know. And you can't bitch! You have to work under these conditions. If you get uptight, it makes the rest of the crew uneasy and aggravates the promotion people. It's the rules of the road. Play by the rules—hope for the best, expect the worst.

The Technique

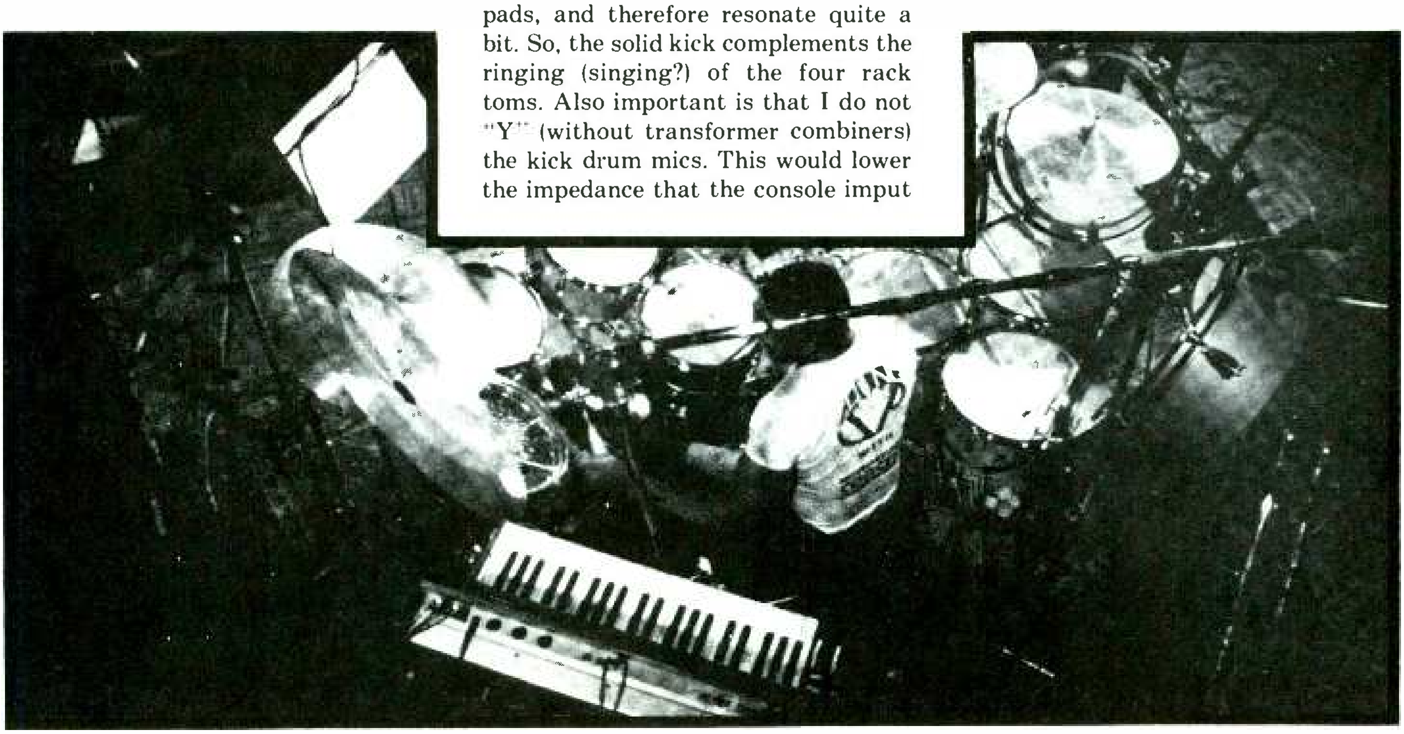
As I previously stated, this is one of the more technically intricate bands on the road today. It utilizes sixteen drum mics, two drum-effect direct lines, seven keyboard direct lines, three bass direct lines, two guitar direct lines and three vocal mics. Because of the nature of the music (ninety percent of the numbers are instrumentals), each instrument must be able to "vocalize." Each must reach out as a solo instrument—an extension of the artist's feelings—and communicate much as a singer's voice, but without the benefit of words. The most difficult part of this, for me, to present is, of course, the drums.

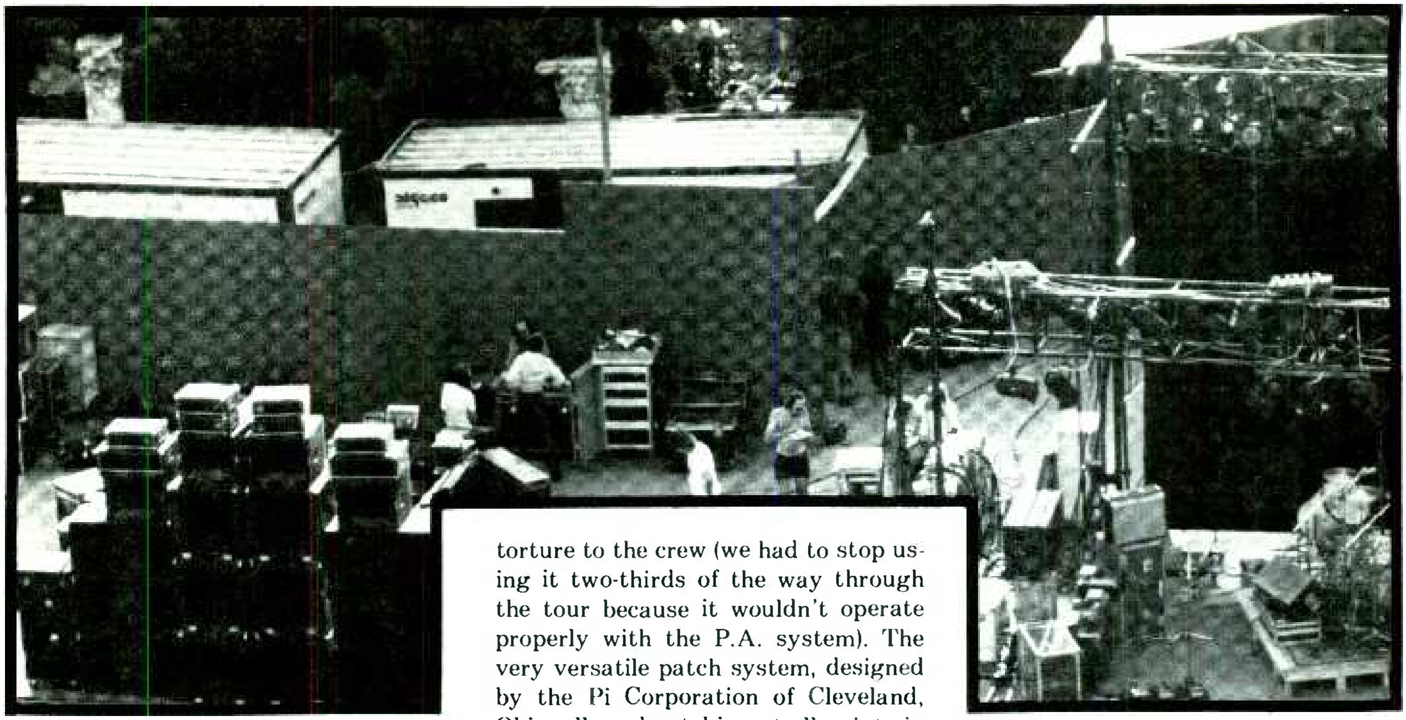
The many sound engineers with whom I've talked all have their special ways of setting up and miking drums and they, like me, are very serious about mic selection and placement. We all would like special mics; I have been fortunate in that Billy Cobham has purchased a selection of Neumann, Electro-Voice, Sony and Sennheiser mics solely for the purpose of creating his drum sound. A sound he achieved in the studio with Ken Scott, the noted engineer and producer, and would like to recreate on the road.

I turn primary attention to the kick drums to establish a solid foundation on which to build a drum sound. EV RE-20's, placed inside the kick drums, bring out the extreme lows of the drums, yet have the frequency response to allow me to boost the 2 kHz-3 kHz range for attack emphasis. The front heads each have a ten-inch hole cut away (the heads do not flutter), and road blankets are inside the drums. I need a dead sound because all of the toms have both the top and bottom heads on, have no dampening pads, and therefore resonate quite a bit. So, the solid kick complements the ringing (singing?) of the four rack toms. Also important is that I do not "Y" (without transformer combiners) the kick drum mics. This would lower the impedance that the console imput would "see" (250//250=125 ohm) and confine the low-end response of the microphones.

The second order of importance is the snare drum. I look for a good, full sound topped with some high end. Here I need a mic that will furnish enough level to feed the effects rack when the snare is being played softly, and that can take possible abuse (e.g., being clubbed with drum sticks.) The mic, a Sony ECM-22P, gets placed two inches below the top rim in order to mic the shell of the drum. The end result is a nice, full, rock sound which has no snare "head resonance." Also, the mic then does not get clobbered.

Next in order are the overhead mics. We use two Neumann U-87 condensers. These not only have a superb "sound," but are more durable than most good condensers and *most importantly*, have good pick up capability so I always have enough level to feed our effects system. The emphasis on that last point comes from my belief that a sound man has to take more into consideration than his infatuation with a brand name and model number. I've worked with numerous sound systems that didn't have enough high end even to warrant setting up the Neumanns, but because of the pick up capability I needed for the effects rack, they went up. Audiences cannot tell most of the time whether there are U-87's or Shure SM-57's on the overheads. Just equalize it "right," and it sounds like cymbals. (I can hear the boos from the "perfect"





engineers, musicians and audiophiles. You know, the ones that come up to you in the middle of a show and tell you that a one dB boost is needed at 13.5 kHz. No comment.)

Lastly, we look at the toms—eleven to be exact. I would, if I had the option, use a Sennheiser 421 mic on each tom. The 421's are pretty flat, and you need flatness when miking to avoid coloration of sound (e.g., the "boom" on some of Billy's toms). We own three 421's and must rely on the sound company for the remaining tom mics. I finally used Shure SM-57's on the four rack toms "Y'ed" into two lines. Leaving the 421's to mic the North bass drum, the front two North drums (optional), and the Roto Tom—a drum that rotates, changes head tension and thus, changes pitch. I only will "Y" mics if I have identical models, because mics made by different manufacturers may be out of phase or may have distinct pick up characteristics. Rack toms have a similar enough sound characteristic to allow "Y-ing," but only with identical mics. The same theory applies to the two floor-tom mics.

And now a word about the electronics. Along with the drum moog—a Mini-moog which is triggered by a small bongo-like drum with a pressure sensitive transducer under the head—we have the "Rack." This device consists of an Eventide digital delay, phaser, flanger and Omnipresser, and a Roland Space Echo, and like Edgar Allan Poe's rack, it did inflict mental torture to the crew (we had to stop using it two-thirds of the way through the tour because it wouldn't operate properly with the P.A. system). The very versatile patch system, designed by the Pi Corporation of Cleveland, Ohio, allowed patching at all points in the rack electronics and was compatible with all input and output impedances, connections and levels. The rack was set up to Billy's left with a series of foot switches to allow Billy hands-free operation of all effects. The input was supplied by me via a pre-fader cue bus, a mix of various drum mics, on the front console. Because the "effected" signal also went to Billy's monitor, we ran into level vs feedback problems, and many nights could not get enough effects level in either the main or the monitor system. The more I fed into the rack from the drum mics, the more feedback we got in the drum monitor, which, of course, cut the level of definition of effects I could get out front. After we discontinued using the rack on stage, I started using the effects out front and got much better results. The reason? I could put much more level into the rack, as there was no need to put the effects into the monitors. Coincidently, I could then also use these effects on the rest of the musicians in the band.

Next in order of technical impossibility is the keyboard system. Each of the various keyboards listed in my layout feeds a Clover 714-B equalizer, MXR noise gates, and MU-tron Phasors (all into individual channels of a Kustom 12-input mixer). Mixed down, the signal feeds a Crown VFX-2 crossover and Crown DC-300A and D-60 amplifiers into a bi-amped Kustom speaker system. (The system has since been modified to include a Yamaha PM-1000 sixteen-input mixer, Yamaha electric grand piano, two Alembic pre-amps for the stereo Rhodes, AKG BXE-20 reverb unit, Crumar organ, Eventide Harmonizer, a JBL-Gauss bi-amped speaker network and, because of impedance problems, the Clover EQ system has been eliminated.)

Because George Duke is the main vocalist in the band, monitors are that much more important to him. Therefore, we split his Sennheiser 441 vocal mike to allow him independent control of his stage monitoring. His split is fed to his console, sent out of a cue bus to an MXR equalizer and a Crown DC-300A amplifier, and then on to a two-way JBL monitor positioned near George. All instruments are taken direct via a split system on the back of George's rack. And if not for Bruce's [Bruce Heigh—keyboard technician] meticulous attention, it would be impossible to try assembling this system and yet stay sane. (Speaking of sanity, check out George's dolls, mannequins and toys sometime. We even miked a toy train once.) Rounding out the band are the bass and guitar systems. First, the bass. Along with a modified triple-pickup Fender bass and a custom, Chuck LeBeau, fretless bass, Al Johnson has an Emmet Chap "Stick"—a 12-stringed instrument allowing bass, rhythm and lead parts to be played simultaneously. Each instrument is plugged into a series of Morley Power-Wahs, phasers and mechanical echoes, MXR Distortion+ boxes and Line Drivers, and Echoplexes (tape echo). This "mess" feeds a Wood amp, modified to be used as amp or pre-amp, and the pre-amp circuit feeds a Phase Linear 400 amplifier to a pair of Bruce Heigh dual 15-inch speaker cabinets. Both basses and the low and high end of the "Stick" feed a mixer (independent stage hookup) and mix down to a single output for use with the P.A. system.

The guitar is the least complicated unit in the band. Guitarist John Scofield plays his Les Paul through a wah-wah and Echoplex attached to an Ampeg VT-22 amplifier head with a Hi-Watt speaker system. A Marshall amplifier head and VT-22 speakers are used as back-up. For softer numbers, we use an Ovation acoustic guitar plugged into a direct box.

All of this equipment is positioned on stage under the close scrutiny of Ken, the production manager. Risers, amplifiers and all other necessary equipment have their special place, with the layout thought out well in advance of the first show. Taken into consideration are: the performers' need to constantly view each other, the audiences' need to see the performers at all times and the audio-related arrangement. The entire band is positioned so that all amplifiers face in toward center stage. This gives the band the full capability of hearing themselves, saves the front row audience from being blasted by a guitar or keyboard amp stack and allows much more control over the house mix.

The Sound System

You *must* hire a separate P.A. company—the European sound system standard is one that would make any engineer uncomfortable. Also, take plenty of spare parts, unless you can describe transistors, fuses, etc., in Italian, French and Spanish to a salesman who, even if he *did* understand you, wouldn't know what the hell you were talking about!

The P.A. company which traveled with us throughout Europe was Orbec Sound of Copenhagen. Their system included a 24-channel Tychobrahe console on both the main and monitor, with White 1/3-octave equalization being used for the stereo mains and the four channels of stage monitoring. The amps, all Crown DC-300A's and D-60's, fed a four-way Cerwin Vega/JBL speaker system. Robert Hansen, crew chief for Orbec Sound and Lights—and the one saving grace on the tour—had plenty of spare parts on hand.

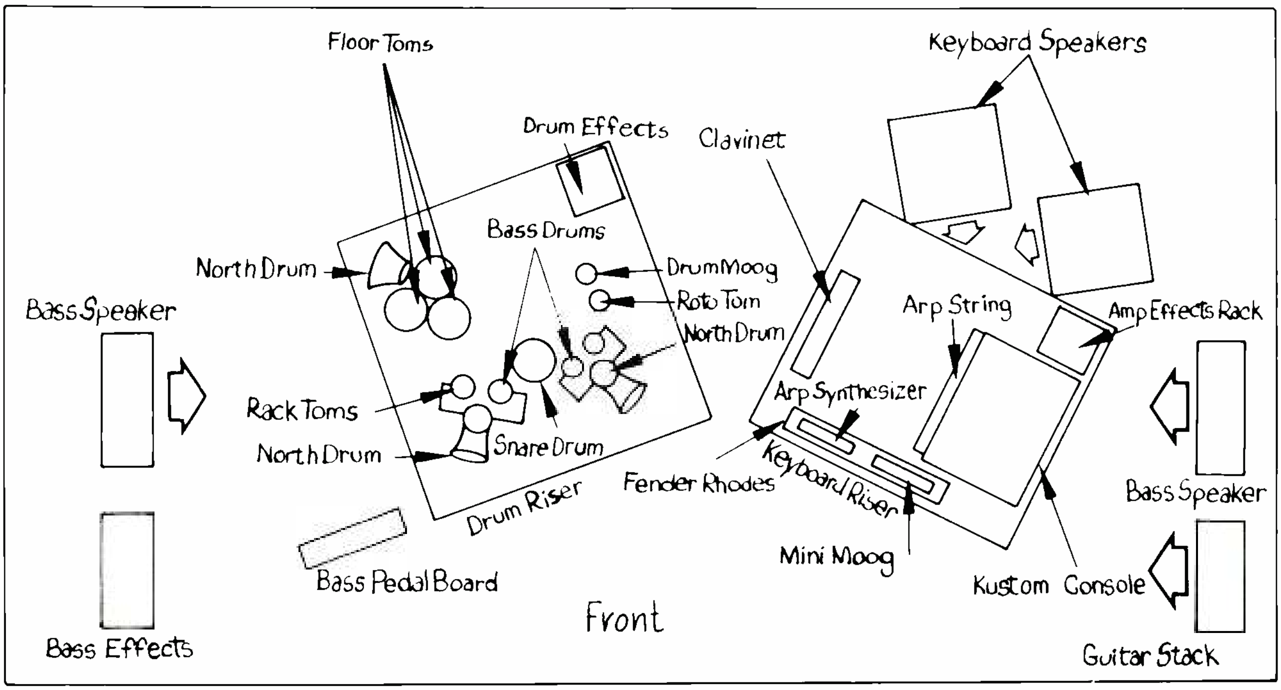
One night, feeling that the system just wasn't what it should have been, Robert checked it prior to a show and found the output down eighty percent. He promptly displayed a spare, plugged it in, and we had Rock'n'Roll! (How *do* you say, "Tychobrahe output circuit board" in Spanish?)

The "Live" Recording

I had been looking forward to the

MIC LAYOUT

DRUMS:	
Kick (2)	2 EV RE-20's
Floor Toms (3)	2 AKG D-12's
Rack Toms (4)	4 Shure SM-57's
North Bass Drum	Sennheiser 421
North Tenor Drums (2)	2 Sennheiser 421's
Roto Tom	Sennheiser 421
Snare	Sony ECM-22P
Hi-Hat	Sony ECM-22P
Overhead	2 Neumann U-87's
Drum Moog	direct
Eventide Effects Rack	direct
Vocal	Shure SM-58
KEYBOARDS:	
Fender Rhodes	stereo "Y'ed" to P.A. (direct)
Clavinet	direct
Arp Odyssey	direct
Mini Moog	direct
Arp String	direct
Taped Effects	direct
Vocal	transformer split—Sennheiser 441
BASS:	
Basses	direct
"Stick"	low and high—direct
Vocal	Sennheiser 441
GUITAR:	
Amplifier	SM-57
Acoustic	direct



STAGE & EQUIPMENT LAYOUT

performances in London, England for a variety of reasons:

(1) All three nights were to be recorded for a "live" album, and this, to me, presented a greater challenge and interest than had previous shows.

(2) We had a full week in London, but were playing for only three nights, all at the Hammersmith Odeon. This meant no equipment moving and a lot of time off.

(3) It would be the end of the tour.

Upon arriving in London, I called Dennis McKay (the engineer for the recording session), to set up a meeting. We met the night before the first show to lay down tentative arrangements—mic lists, set lists, stage set up, sound check times, etc. We knew problems would be facing us on the first day of a concert bill including Cobham/Duke, Weather Report and Shakti, featuring John McLaughlin. Especially now, with sound checks for the recording truck required. I also had contacted Manor Recording Company, the organization supplying the remote recording truck and three extremely capable engineers, to set up a tentative schedule.

Stage call for band crews, stage hands and promotion people (complete with breakfast) was set for 8:30 a.m. The main event that morning was to hoist the P.A. onto the multi-tiered tower where it would properly disperse sound to all points of the hall. At noon the Manor truck began unloading, running the various cables and doing preliminary checks. I sat down with the Manor engineers and drew up several copies of both the 24-channel P.A. and the 40-input recording list. Every engineer received a copy of each list so that in the event of an emergency, any engineer would know which numbered input corresponded to a particular numbered snake (two for P.A., two for recording). This eventually came in handy more times than I had thought.

We started setting up the band, and were ready for a check to the recording truck at 3:30 p.m. The direct inputs to the keyboards, bass and guitar were really no trouble. No hums or buzzes. The only "sound" that we had to create was from the drums. Billy came in at 5 p.m. as scheduled and proceeded to record some drum tracks to check out sound vs mic placement. The remainder of the band came in for sound checks about 6 p.m. It all worked. Now the most important event was a monitor check. We couldn't have any feedback; but then Bruce had been able to run the monitors for the entire tour at levels that satisfied the band, *and* with no feedback. The other monitor concern was that of the monitors possibly bleeding [leaking] into mics and destroying any instrument isolation already achieved. This, also, proved to be no problem. It was all going right!

Because of my involvement with the recording check, I had no time for a house sound check. Instead, Brian Risner, the engineer with Weather Report, did a channel check with our band. All channels were working. Actually, I very seldom do more than a channel check because we rarely have the time, and because I believe that sound checks are for observing monitor levels and in most cases do very little for building a proper house mix. It was 7:55 p.m. as I climbed through the spaghetti wiring that allowed each mic to plug to the main, monitor and recording consoles. Also in my path were the lines which supplied me with the mixdown from the truck. It still was going okay. Of course, the realizations that this mess had to be converted back to a P.A.-only system, and that the stage had to be changed to the next act in only fifteen minutes were always there. But, it all flowed.

At three minutes to eight, I was running to the sound console for an eight o'clock show. At eight o'clock, one of the sound crew was running out to tell me that the intercom didn't work and that the show was going to start. Meanwhile, the curtain was going up. As it rose, the intercom began working. With the exception of one loose connection the first night, we had three excellent nights of performing and recording.

The Final Curtain

On the road, you not only are a soundman, but also a technical repairman, carpenter, and someone capable of speaking all languages with his hands. European tours (or tours in general) are a lot like audio—there are no absolute, correct things you must do, only a lot of things you *don't* do. And you learn by experience. If your band is planning a European tour, get down off your cloud. It's a chore! Don't pack your basket yet, 'cause it ain't exactly a picnic.

Build Your Own Cue System

By Robert E. Runstein

A good headphone or "cue" system is a necessity for multitrack recording. The phones enable the musicians to hear each other in "proper" musical balance when their (the musician's) placement in the studio and the baffles used to achieve the desired acoustic separation between microphones would prevent this. The cue system permits musicians to hear their own instruments when the loudness of the other instruments would mask them, and conversely permits musicians to hear the other instruments when the loudness of their own instruments would prevent this. Phones also enable musicians to hear previously recorded tracks while overdubbing, and prevent these same tracks from being picked up by the mic along with the new overdubbed performance.

A good cue system should have the following characteristics:

(1) It should allow a large number of phones to be used simultaneously without interaction effects when one set is connected or disconnected, and it should permit the system's output to be distributed throughout the studio by simple parallel connections.

(2) The system should use phones capable of producing adequate volume with low distortion and good tonal balance, and should protect the phones against accidental burnout from being overdriven by program material or from ultrasonic signals which may accidentally be fed to them.

(3) The system should provide each musician with a control box which permits individual control of the phones volume as well as individual selection of different mixes at the flick of a switch.

A cue system consists of a mixer with at least as great a number of inputs as the number of tracks available on the tape machine it is being used with. The inputs must be individually switchable to accept either *program* (console output) or *tape playback* for use in recording and overdubbing. Some cue mixers can also be fed the signal from individual mics for additional flexibility in achieving the proper balance in the phones. The mixer may have more than one output so that different mixes can be provided for different musicians. The most economical and useful configuration is a two output mixer which provides a choice of either a stereo mix or two different mono mixes. A stereo phones mix enables individual instruments to be heard more clearly because instruments with similar timbres can be placed in opposite channels, thus reducing their mutual masking effect.

Each output of the mixer is fed to a separate channel of a power amplifier which in turn drives the phones. Unless otherwise stated, this article assumes that the system uses stereo headphones with the left and right earpieces being driven by separate power amplifier channels. Any statement regarding one channel of the system will apply equally to the other.

When more than one set of phones is to be used at once, they are connected in parallel with each other across the power amplifier's speaker output terminals, as shown in figure 1. Since modern power amps have a low output impedance and use large amounts of negative feedback, they behave like *constant voltage sources* as long as they are not clipping. This means that for a particular input signal level, the voltage at the speaker output terminals will remain the same regardless of any changes in the load connected, as long as the value of the load impedance is not less than the amplifier's minimum rated load impedance (for this could cause clipping). For example, if the amp is rated for a 4-ohm minimum load, its output *voltage* will remain the same regardless of whether the load impedance is 4 ohms, 8 ohms, 400 ohms or infinite (open circuit). A 2-ohm load, however, might cause the amp to clip under certain conditions, producing distortion as well as allowing the output voltage to vary as the load impedance changes. While amplifiers are often rated for higher output power at lower load impedances, this is due to the greater amount of current drawn by the lower impedance, not an increase in the output voltage (power = voltage × current). Some power amps provide a headphones output jack which is connected to the speaker outputs through series resistors. This jack should not be used with a cue system because the resistors prevent it from acting as a constant voltage source and also attenuate the signal, producing less volume from phones connected to this jack than from those connected to the speaker outputs.

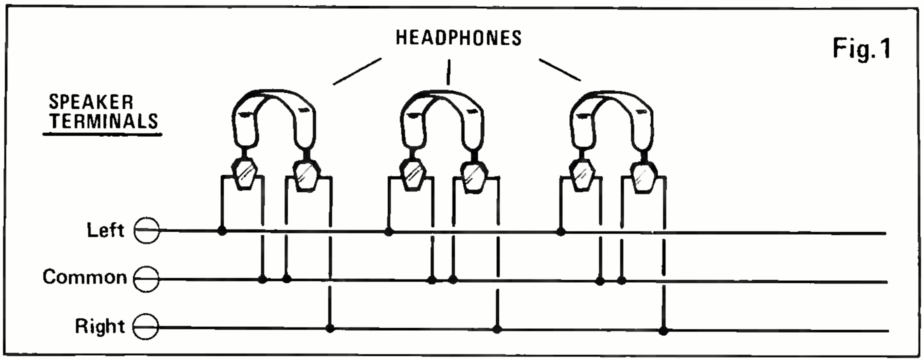


Fig.1

As long as the combined parallel impedance of the phones does not fall below the amp's minimum rated load impedance, we can connect as many or as few sets as we like, and adding or removing phones will have no effect on the volume produced by the other sets. If all the phones have the same impedance, the number of phones which can be connected is computed by the formula: $n = Z_P/Z_L$, where n is the maximum number of phones that can be connected to the amplifier's output terminals at the same time; Z_P is the impedance, in ohms, of one earpiece of a set of stereo phones; and Z_L is the specified minimum load impedance, in ohms, of the power amp. The formula assures that the total impedance presented by the phones will not be less than the amplifier's minimum load impedance, and shows that it is advantageous to use high-impedance phones so that many sets can be used simultaneously. In determining the number of phones being used, a set of stereo phones strapped for mono operation so that both ears are driven by the same power amp channel must count as two sets of stereo phones on that channel.

If phones with different impedances are to be used simultaneously, the total phones impedance—Z_T, can be computed from the formula:

$$Z_T = \frac{1}{1/Z_1 + 1/Z_2 + \ldots + 1/Z_n},$$

where $Z_1, Z_2 \ldots Z_n$ are the impedances of the various sets of phones connected. In this case, the number of phones will be limited to that which permits Z_T to be greater than or equal to Z_L. Using different types of phones at the same time is not recommended because different models may not have the same efficiencies. This will lead to complaints from some musicians that the phones are too soft, while others will insist they are too loud. In addition, different impedances will draw different amounts of power—resulting in volume differences even if efficiencies are equal. While this problem can be avoided by providing a volume control for each set of phones as will be described later, the frequency response of different models of phones varies as much as it does between different models of loudspeakers. This will cause the musical balance heard by each musician to differ even though they are receiving the same mix. This problem is avoided if the studio has enough sets of the same model phones to supply everyone.

To be effective as a cue system, the sound reaching the musicians' ears from the phones must be low in distortion and its sound pressure level (SPL) must exceed that of the sound in the studio by at least 10 dB (and preferably more) for clarity. This can be achieved by using phones with high SPL generating capability and ear cushions that attenuate the level of the studio sound reaching the ears. The phones must provide some isolation since sound levels in the studio can reach as high as 115 dB or so on rock sessions. Generating 125 dB or more for clarity would put the level of the phones above the threshold of feeling (118 dB), producing discomfort and rapid listening fatigue in the musicians. Non-isolating or "open-air" phones can be used when the surrounding studio levels are low enough that the phones can produce the required 10 dB greater SPL without causing discomfort. This situation can occur during overdub sessions. When using non-isolating phones, care must be taken that their output is not picked up by the mics, for this would reduce track separation and result in a loose, muddy sounding recording.

Other considerations in choosing phones are that they are comfortable enough to be worn for long periods of time; that they will not fall off if the musicians move around while performing; that they can withstand being dropped and stepped on as will invariably happen in studio use; and that they can be repaired easily both electrically and mechanically. Phones with modular parts simplify repairs, and the use of metal jacketed plugs will reduce the likelihood of broken connectors. Well balanced frequency response is extremely important so that, for example, a bass player being recorded direct—who must rely totally on the phones to hear himself—can hear his instrument clearly without the engineer resorting to a phones mix that blots out the rest of the instruments. It is also important that the subjective response of the phones be similar to that of the monitor speakers in the control room so that musicians will have some idea of the quality and tone of the sound being recorded without having to return to the control room for a playback. This will also free them from having to rely totally on the judgment of the engineer and producer. It can be very disturbing for a musician to learn that the engineer and producer in the control room think his instrument sounds great when it sounds terrible to him in the phones.

In choosing the phones to be used in the cue system, we must take into account the maximum SPL they can produce with low distortion and without damage to their drivers. This can be determined from the phones' distortion, maximum signal handling capability and sensitivity figures on the manufacturer's spec sheet. The distortion figure states the percentage harmonic distortion produced at a particular SPL. The maximum signal handling capability is given in terms of the maximum instantaneous *peak* power or voltage that can be applied to the phones without damaging them. Since the ear's perception of loudness is proportional to the *continuous* signal producing capability of the phones, we must convert the peak value to the appropriate continuous value. Since most musical material has a peak to average ratio of about 10 dB, we must deduct 10 dB from the manufacturer's peak rating to arrive at the continuous signal rating. The sensitivity of the phones is given in terms of the SPL generated for a particular continuous power or voltage at a given frequency. We can determine the maximum safe SPL which the phones can produce by computing the difference in dB between the continuous signal handling rating and the level of signal used for the sensitivity specification, and adding this number to the SPL quoted in the sensitivity spec. For example, the Beyer DT100 phones will produce a continuous 120 dB SPL with 0.2% distortion and have a maximum peak voltage handling capability of 20 volts. The maximum continuous voltage that can be safely applied is 10 dB lower or 6.32 volts, to allow 10 dB of headroom for musical peaks. This lower voltage is computed by dividing the peak voltage by 3.16, so $20/3.16 = 6.32$ volts ($20 \log 1/3.16 = -10$ dB). The sensitivity of the DT100s is such that a continuous 0.632 volts produces 110 dB SPL at 400 Hz. Since 6.32 volts is ten times 0.632, a 20 dB difference, we add 20 dB to the 110 dB figure, and arrive at a maximum continuous SPL of 130 dB which is more than enough for a cue system. If the phones cannot produce a continuous level of at least 118 dB with low distortion, they will be burned out rapidly in attempts to provide sufficient level to the musicians.

If the manufacturer's specifications are given in terms of power rather than voltage, the values can be converted to volts by the formula $E = \sqrt{P \times Z_p}$, where E is the voltage in volts, P is power in watts and Z_p is the impedance of the phones in ohms. Again using the Beyer DT100s with their 400 ohm impedance, their power handling capability is stated as 1 watt peak, so the peak voltage capability is $E = \sqrt{1 \times 400} = 20$ volts, peak. Their sensitivity is also specified in terms of power with 1 milliwatt continuous power producing 110 dB SPL, so $E = \sqrt{0.001 \times 400} = 0.632$ volts continuous for 110 dB SPL. From these voltages we can compute the maximum SPL as above.

Having chosen phones capable of producing sufficient SPL, we can avoid accidentally destroying them by inserting a level control between the cue mixer output and the power amp input, and calibrating the system so that the signal reaching the phones cannot exceed the maximum continuous voltage previously calculated. If the power amp has built in level controls these can be used. If not, the simple potentiometer circuit shown in figure 2 can be used. A 10 k ohm linear potentiometer will suit almost all situations, and should be located where it cannot be adjusted by unauthorized personnel.

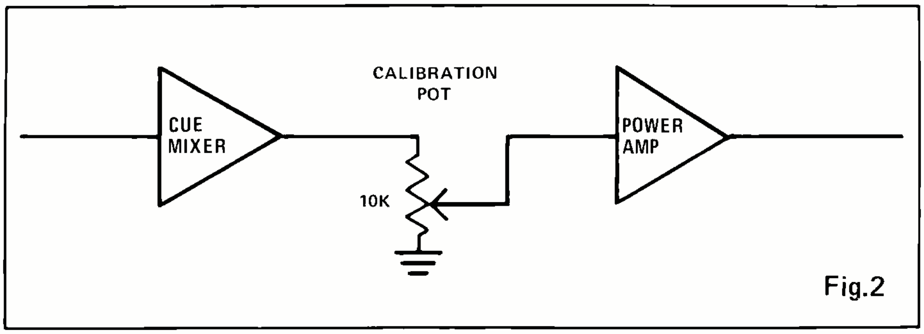


Fig.2

To calibrate the system, we must first determine the maximum continuous output of the cue mixer under typical operating conditions. To do this, disconnect all headphones and connect an AC voltmeter across the output of the cue mixer. Next, play a multitrack tape through the mixer with all individual track cue faders and cue master faders set for full level. Note the highest voltage read on the meter during a short period of time such as one minute. The tape used for this should have different instruments on each track, and each track should playback at normal operating level, reading near 0 VU at its loudest point. Do not use a test tape because it is important that the signal on each track have a random phase relationship with the other tracks. The tracks on a test tape would all be in phase and would produce a voltmeter reading higher than would be obtained in any real recording or overdubbing situation.

Stop the tape and feed a sine wave oscillator set to 1 kHz to one of the cue mixer inputs and adjust the oscillator level so that the voltmeter reads the same level noted when the tape was running. It is a good idea to connect an oscilloscope across the cue mixer output to make sure that the oscillator does not drive the cue mixer's input stage into clipping. If no oscilloscope is available, listen to the cue mixer output as the level of the sine wave is raised to hear if there is a change of tone. A change from a pure tone to a harsher one indicates clipping. If the cue mixer cannot be driven to the desired level without clipping, connect the oscillator directly to the power amp input and proceed in the manner described above.

Now, connect the AC voltmeter

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across the power amp's speaker terminals and set the level control between the cue mixer and power amp so that the amp's open circuit output voltage reads the maximum continuous voltage the phones can handle, as computed earlier. While this could have been done directly with the signal from the multitrack tape, the use of the sine wave signal permits the adjustment to be made much quicker and with more precision than would be possible with a varying musical signal.

The phones are now protected against being burned out by driving them beyond their signal handling capabilities, but there are two other potential causes of damage. The first is leaving the phones connected when lining up the console or tape machine with sine waves, or when playing test tapes. As mentioned earlier, random phase signals add differently than in-phase signals. The signal level increases 3 dB each time the number of random phase signals mixed together at the same level doubles, while with in-phase signals the level increases 6 dB each time the number of signals doubles. If we feed randomly phased signals at 0 dB to each input of an eight input mixer with each input set for unity gain, the output level would be +9 dB. If we feed in-phase signals to the same mixer at the same level, the output would be +18 dB. Thus, the potential exists to drive the phones to a continuous level 9 dB higher than the maximum continuous level we have calibrated the system for. With a sixteen-input mixer the difference in level between random and in-phase signals would be 12 dB. Continuous sine waves at this level would rapidly heat the phones elements and lead to damage. The solution to this problem is to make sure the cue mixer master level controls are turned all the way down when lining up the equipment.

The other potential source of damage is from high-level, supersonic signals such as console oscillations due to ground loops or improper patch connections and the high-frequency chatter caused by tape passing the playback heads at high speed in fast-forward or rewind. Both of these types of signals cause rapid heating of the phones elements. Damage from this source can be avoided by rolling off the high-frequency response of the cue system above the audio range. A simple low pass circuit which accomplishes this is shown in figure 3. This circuit is adapted from one shown

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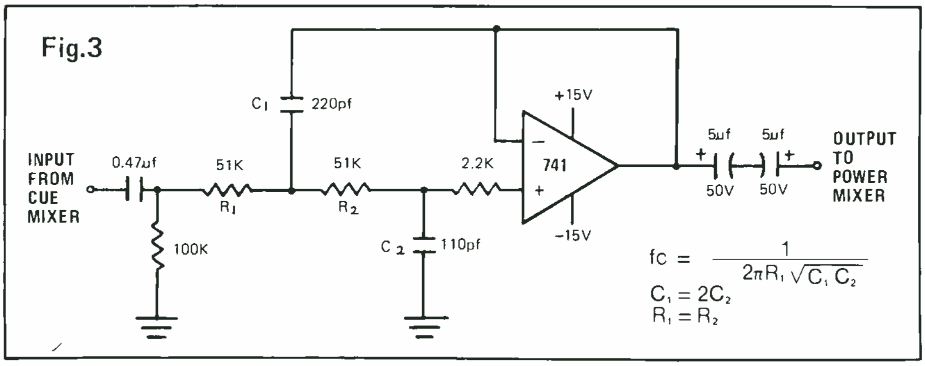

Fig.3

in Walter G. Jung's *IC Op-Amp Cookbook* and has its −3 dB high-frequency cutoff at 20 kHz. Response rolls off at the rate of 12 dB per octave above 20 kHz. The circuit has unity gain in the passband and its high input impedance permits it to be driven by virtually any cue mixer. The op amp's low output impedance can easily drive the cue system's 10 K ohm calibration pot. This circuit should be installed before calibrating the system.

The flexibility of the cue system can be increased by providing the musicians with control boxes that permit them to adjust the volume of their phones as well as to select which mix they will hear. Thus, musicians who require loud phones can have the volume they need without deafening the others. The circuit of figure 4 is built taper. The use of single pots permits one side of the phones to be shut off for vocal overdubs when the singer prefers to use only one earphone in order to hear himself directly as well as through the phones. Shutting off the unused side of the phones reduces leakage into the mic. A dual pot, however, is less confusing for the musicians to operate.

When control boxes of this type are included in the cue system, the combined impedance of the potentiometer and headphones determines the maximum number of control boxes that can be used according to the formula:

$$n = \frac{\dfrac{Z_{phones} \times R_{pot}}{Z_{phones} + R_{pot}}}{Z_O}$$

A 100-ohm pot works well with phones

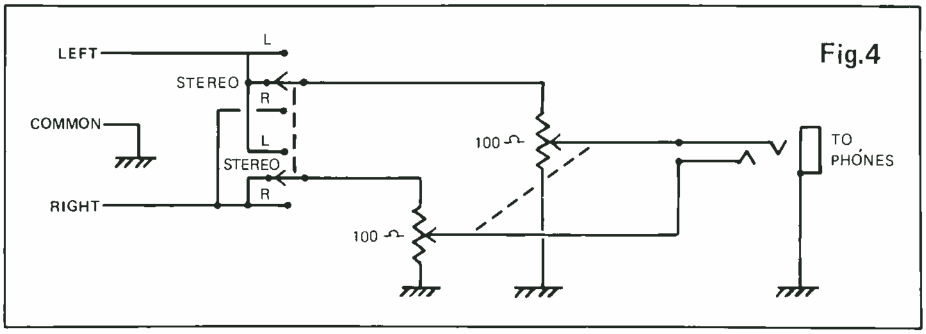

Fig.4

into a minibox which can be clipped either onto a mic stand or onto the musician's belt. The switch lets each musician choose between a stereo mix, the left mix in mono to both ears or the right mix in mono to both ears. A long cable attached to the box eliminates the need for additional extension cords. Cue systems with more than two channels can be accommodated by using switches with additional positions and by using cable with the appropriate number of conductors to connect the control boxes to the cue system output jacks.

The volume control can be either a dual pot or two single pots (one for each ear), and should have a log (audio) of 100 ohms or greater impedance. With an amp capable of driving a 4-ohm load, and 100-ohm phones, twelve control boxes set for stereo or six set for the same mono channel could be used. With 400-ohm phones such as the Beyer DT100, twenty control boxes set for stereo or ten set for the same mono channel could be used.

References:

1. Feldman, Leonard. "Headphones Around The House." *Audio.* Vol. 58, No. 5. May 1974. p. 24-26.
2. Jung, Walter G., *IC Op-Amp Cookbook*, Howard W. Sams & Co., Inc. 1974. p. 332.

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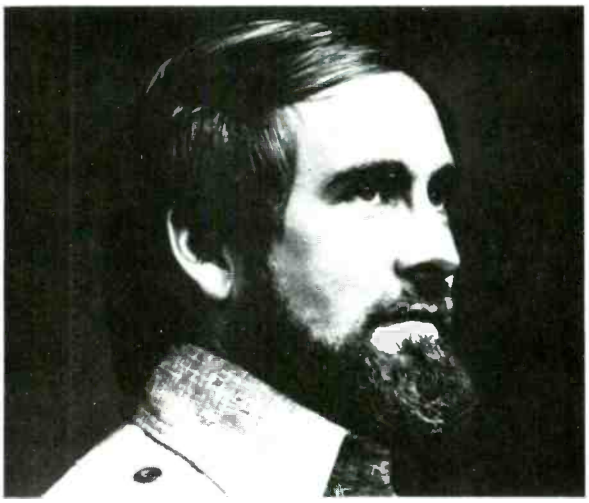

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Ambient Sound

BY LEN FELDMAN

The Tape Alphabet

Back in the days of the Roosevelt administration (Franklin D., that is), I am told that people used to despair over the proliferation of "initialized" bureaucracy in government. We had the NRA, the CCC and dozens of other abbreviations which required the use of a handy guide-book to keep track of the agencies. Anyone involved with tape recording (and with the periodic selection of raw recording tape) can sympathize with the folks who had to learn what all those initials meant, for when it comes to tape types, we too are being besieged by initials and abbreviations from every quarter. Is FX tape inherently different or better than MRX_2 tape? Or, how about FeO, FeCr and CrO_2? Are we all going to have to become chemistry majors to be able to select a tape that's right for our cassette decks or open-reel machines? Does the fact that some manufacturers are beginning to use Roman numerals in their tape designations (UDXL-I and UDXL-II) mean that we have run out of alphabet?

What used to be called "standard" tape consisted of simple, ferric oxide particles which were suspended in the binder solution. Ferric oxide particles, being related to iron ore, are what give most tapes their reddish-brown appearance and, of course, all iron ore and its derivatives are subject to magnetization. But, as time went on, manufacturers of raw tape discovered various techniques whereby they could improve the performance of tape. One of these techniques involved the packing of particles onto the tape with greater density. Other techniques involved treating or "doping" the ferric particles with additional chemicals such as cobalt. Tapes were developed which used magnetic particles other than ferric oxide, such as chromium dioxide and, finally, some tapes actually were fabricated using combinations, or layers of both ferric and chromium dioxide particles. Unfortunately (or, fortunately, depending upon your point of view), the operating parameters of magnetic recording tape differ appreciably from one manufacturer to another as well as between different types of tape from the same manufacturer.

As far as your tape deck is concerned, there are three variables that you must be concerned with when considering the use of any given tape. One of these, recording level, is controllable by you, the operator. The other two, equalization and bias, are usually selectable in fixed increments on home-type machines (though in professional machines, bias, at least, is continuously adjustable).

Some Magnetic Properties Defined

If you have ever examined the specification sheets of a given tape manufacturer, you will probably have encountered some unfamiliar terms that come neither from the physical nor the electronic world but are specifically applicable to the world of magnetism. Let's define a few of them.

Coercivity of a tape is measured in Oersteds and is an indication of its sensitivity to an applied magnetic field. The published specification usually lists the strength of a magnetic field needed to completely erase a saturated tape. Coercivity of audio tapes may range from around 250 to 400 Oersteds.

Retentivity is a measure of the tape's flux density after a saturation producing magnetic field has been withdrawn from the vicinity of the tape. It is expressed in gauss, or magnetic flux lines per cross-sectional square centimeters of tape. In home recording tape, it may vary from around 1000 gauss to over 1500 gauss. *Remanence* describes pretty much the same thing, except that it is expressed in lines of flux per linear quarter-inch of tape width. A tape playback head's output level is a direct function of the tape's remanence.

Sensitivity is an indication of a tape's output level as compared with some specific reference tape. As a result of recent advances in oxide formulation techniques, tapes with high sensitivities (so-called "high-output" tapes) have become readily available. The sensitivity increase or improvement is a direct result of greater retentivity. For a given oxide coating thickness, greater flux density results in a higher remanence value and therefore in a higher output level.

Separating Trade Names From Generic Abbreviations

Certain manufacturers, perhaps enamored by those alphabet abbreviations, have taken to using letters of the alphabet to denote their tape grades. These abbreviations should not be confused with the generic abbreviations (such as FeCr, CrO_2, FeO etc.) which describe basic tape-coating formulations. While we cannot list all the brand names which choose to use letters of the alphabet to denote their different tape grades, here are just a few, with brief descriptions designed to take some of the mystery out of the alphabet notations:

Fuji FX cassettes are low-noise, high-output, ferric tapes. Their FL tapes are a low-noise variety a grade below the FX, while tapes labeled FC are actually that company's brand of CrO_2 (chromium) tapes.

Maxell's new UD-XL I tapes are ferric compound varieties that require normal biasing for low-noise, high-output tapes, while their new UD-XL II is a cobalt-doped, high-output, low-noise, ferric tape that requires CrO_2 (70 microsecond) equalization and bias.

Memorex's MRX_2 is that company's low-noise, high-output variety, but is still basically a ferric tape that operates properly with standard (low-noise, high-output) settings of bias and EQ.

Nakamichi's EX and EX-II tapes are ferric and ferricobalt formulations that require "standard" or low-noise/high-output settings, while their new SX tapes are single-coated formulations of ionized cobalt and ferric oxide that require the higher bias and 70 microsecond equalization normally associated with CrO_2 tapes. Nakamichi, like several other companies, has discontinued the manufacturing and distribution of actual CrO_2 tapes now that cobalt-doped, ferric tapes are able to provide the same performance as chromium without chrome's inherent disadvantages (higher abrasion of tape heads and poorer maximum output level or headroom).

TDK (an alphabetized or initialized company already) calls its best ferric, cobalt-doped tape SA, which really stands for Super-Avilyn (their own trade name for their particular particle formulation) and it requires bias and EQ settings normally associated with chrome tapes. Their best grade of standard-biased tape is called Audua and calls for the "standard" or "normal" EQ setting on cassette decks. A lesser grade of low-noise tape from the same company which requires the same standard settings is called SD (for Super-Dynamic), while their general purpose cassettes are called D-Cassettes (for Dynamic) and feature their standard gamma-ferric coating.

Equalization

Those tape selection switches on your cassette deck also change the record equalization characteristics of your machine to suit the needs of the family of tapes with which the machine is used. Generally, playback equalization is fixed on most machines and equalization changes based upon tape formulation are confined to recording, though there are some machines that alter equalization for both modes of operation. The equalization characteristic for ferric, low-noise, high-output tapes these days is 120 microseconds (the time constant of a simple R-C network used to provide the necessary response curve required during the record process). CrO_2 tapes and their equivalent varieties utilize a 70 microsecond equalization time constant, which amounts to less high-frequency boost (compared to the 120 microsecond equalization constant) and therefore permits a higher recording level (more "headroom") and attendant improved signal-to-noise ratio.

Back to the Alphabet

Differences in the parameters defined above will result in different recording characteristics for different tapes. Chromium dioxide tapes, for example, require considerably higher bias levels compared with ferric oxide tapes and, when you switch to the CrO_2 tape selection position on your cassette deck what you are doing is increasing the amplitude (but not the high frequency) of the self-contained bias oscillator voltage in your tape deck. Increasing bias for any tape results in several inter-related effects. With no high-frequency bias applied, all tapes would produce a highly distorted recording during playback because tape is inherently a very non-linear storage medium. As bias begins to be applied, sensitivity of the tape increases (up to a point), harmonic distortion decreases (again, up to a point) and maximum recording level increases. But, as bias is increased beyond the optimum point, a condition is reached where high-frequency response begins to roll off. Thus, each type of tape formulation has an optimum bias point and, if CrO_2 (higher) bias were to be applied when using ferric-oxide tapes, the high frequency response of the resultant recording would be worse than that which would have been achieved if you had left the switch in the "normal" or "standard" position. It is for this reason that owners of tape decks (cassette or open-reel) who do not have a special CrO_2 switch position are warned not to attempt to use chromium dioxide (or equivalent) tapes, since the reputed virtues of such tapes (better frequency response and improved signal-to-noise) will simply not be realized unless the recording parameters are changed to suit that kind of tape.

Back to Basics

While most recording enthusiasts (professional as well as amateur) quickly familiarize themselves with all the controls and functions of their favorite tape decks, the characteristics of the tapes they use in conjunction with those decks remain something of a mystery. Perhaps a review of tape fundamentals would be in order and would be helpful in understanding that melange of abbreviations we referred to earlier. All tape consists of a base material (usually mylar, or acetate) upon which has been placed a magnetic coating. The coating consists of a binder solution in which tiny magnetizable particles have been suspended.

We have not included in this summary those manufacturers who use descriptive (rather than letter abbreviation) terms for their products, such as Scotch "Classic" tape (which is a double-layered, ferri-chrome formulation) simply because we wanted to differentiate between trade-name abbreviations and abbreviations that tell something basic about a generic type of tape (FeCr, CrO_2 etc.). About the only way to tell the differences between those tapes which bear descriptive names rather than letters (BASF's "studio" cassettes are higher grade than their "performance" series, while Sony's "Ultra-High-Fidelity" tapes are a bit better in grade than their "Low-Noise" variety) is to check the suggested retail prices of the different grades available.

If there's one message that can be garnered from all of this, it is that oft repeated tip that there is one (or perhaps a couple of) tape type that will work best for your tape deck. If your deck maker doesn't tell you what that tape is (or, in the interest of not offending anyone, lists a half dozen or more tapes that are compatible) the only thing you can do is experiment until you find the brand and type of tape that works best for you and your equipment.

So, in closing, SYNT (See You Next Time. . . . .).


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MODERN RECORDING **LAB REPORT**

NORMAN EISENBERG AND LEN FELDMAN

Nakamichi Model 610 Control Preamplifier

General Description: The Nakamichi 610 is a unique product, literally in a class of its own from a functional standpoint. It is, essentially, a preamp-control unit (though some traditional preamp functions are lacking, more of which presently) for interfacing various audio components in an elaborate stereo system, and with more than usual applications for recording. It also serves as a device to help make simple checkouts of some performance areas. Finally, it is a high-quality, low-distortion mixer, allowing a stereo mix of any five of the 19 inputs that may be connected to it.

Most obviously, the Nakamichi 610 is a style-mate for the Nakamichi 600 cassette recorder (reviewed here, October/November 1976 issue), adding to the 600 microphone inputs, the mixing facility, and a stereo headphone output. Dimensionally, it is a twin to the cassette deck, and like the recorder it too is available in a choice of brushed aluminum or matte black finish. But beyond its obvious mating with the 600 recorder, the new preamp has many added uses that would be of interest to the general audiophile as well as to the semi-pro recordist seeking a compact and versatile "mini-studio" setup.

As a system preamp, the model 610 accepts all standard input signals, and provides correct pickup equalization and gain for feeding power amplifiers and/or recorders. It does not have tone-controls or noise filters, the design philosophy here being that if one is serious enough to own the 610, he would logically obtain tone adjustments in terms of a separate equalizer device. Dubbing from one tape deck to another is of course possible.

As a performance checker, the 610 has a built-in sine-wave generator capable of producing seven discrete frequencies by a combination of three buttons. The buttons each select 1 kHz, 3.16 kHz, and 10 kHz and since they may be operated together for additive effects, additional frequencies of 4.16 kHz, 11 kHz, 13.16 kHz, and 14.16 kHz also are available. Included too in the 610 is a pink-noise generator which produces three mics and one stereo tape deck or turntable; or mixing two tape decks and a blend mic; or a turntable, tape deck, and blend mic; or two tape decks; etc.

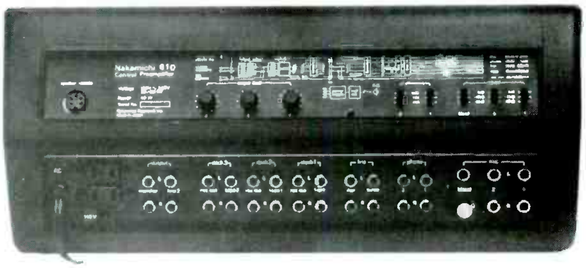
The control panel, like that of the model 600 cassette deck, slopes forward. At the top are the two meters, peak-level reading and calibrated from below −40 to +10 dB. To their left are three buttons for channeling the monitor output through one of three level-matching controls. At the right of the meters is the a single, broad-spectrum tone covering the 50 Hz to 15 kHz range ±2 dB (1/3 octave). These signals, which can be monitored on the 610's own meters, may be run through the playback system and may also be recorded for tape recorder tests. The performance checker facility also permits checking the phase of components in terms of correct stereo hookup. It also provides for A-B comparison tests. The last function includes phono pickup, and tape-source comparisons, and—with the addition of Nakamichi's remote control unit, the RM-610—the A-B testing of loudspeakers becomes quite simple too.

In its role as a mixer, the model 610 facilitates "live" or prerecorded source mixing in many combinations, such as five-microphone "live" recording; or mixing power off/on button. The function and signal input buttons are worked out to provide for line A, line B stereo, mixing mode, or test-tone mode in conjunction with the test-tone buttons below or the rows of signal selectors at the right which include, separately on lines A and B, phono 1, phono 2; mic 1, mic 2; tuner; aux; and tape 1, 2 and 3. To the right of this group are the tape-monitor buttons (source, tape 1, 2 and 3). Below them are four phase-check buttons. Below the line A and B selector controls are phase-inverter buttons which apply only to the mixer inputs and operate in the mixing mode.

In addition to all these buttons there are nine control knobs for signal level adjustments, including separate knobs for test-tone, master volume, left channel and right channel on line A, ditto for line B, a blend (L plus R) control, a channel balance, and monitor level.

A stereo headphone jack is at the lower left. The panel has handles for lifting.

Signal jacks at the rear include twelve stereo pairs of phono pin-jacks marked for monitor, output; line output; tape inputs and outputs 1, 2 and 3; two line inputs (nominally for tuner and aux); and two phono inputs. Additionally there are five standard mic jacks—two stereo pairs and the blend input. The rear also has the socket for the optional remote-control hookup plus three level matching controls for use with various amplifier/speaker combinations, a pair of phono input impedance selector switches, and three mic attenuator



Nakamichi 610: Rear panel view.

switches. A grounding terminal, two convenience outlets (both switched by the front panel power button), an input line voltage selector, and the power cord complete the picture at the rear—except for a block diagram that shows the various signal paths through the device.

Test Results: The Nakamichi 610 is supplied with a complete roster of performance specifications, all of which were either met or bettered in MR's tests. Especially noteworthy were the extremely good figures for input sensitivity, the very high signal-to-noise ratio, the very low distortion (which MR is willing to admit could well be the residual distortion of its test instruments), and the accuracy of the test-tones supplied by the device's built-in generator. The meters on the 610 are excellent, accurately calibrated, and provide a true dynamic range of better than 50 dB.

With nineteen available inputs on the 610, it is almost impossible to detail the variety of signal combinations and functional combinations possible, but MR did try some. Three mic inputs were mixed with a tape and then with a phono source; several combinations were tried of two stereo programs (phono and tape; two phonos; tuner plus tape deck, etc.). A "live" recording was made using five microphones. Throughout, everything worked as claimed. MR also compared a direct feed of the best FM signal available in its area (connecting it directly to a top-quality power amp) with the sound heard when feeding that same signal through the 610 and thence to the power amp. As far as could be determined there was no detectable difference in sound quality. In this regard, MR's testers feel that the Nakamichi 610 comes as close as anything yet experienced to the oft-cited objective of a high-quality preamp—namely, to produce the equivalent of a "straight wire with gain."

A word of caution: The control and pushbutton arrangement on the model 610, though functionally logical and aesthetically pleasing, does take some getting used to. We strongly urge buyers of the 610 to read the instruction manual carefully—perhaps more than once at that—before attempting to use the device in all its possible modes.

General Info: Dimensions are: 15¾ inches wide; 6.7 inches high; 9⅓ inches deep. Weight is 15.5 lbs. Price, in matte black, $570; in brushed aluminum, $550. Supplied with lift-off plastic cover, two pairs of stereo signal cables, polishing cloth and owner's manual. The owner's manual is excellent—very detailed and amply illustrated.

Individual Comment by L.F.: I am not sure whether Nakamichi engineers set out to design a versatile home-mixing console and then realized that the unit also could serve as the center of a stereo system,

Nakamichi 610: Remote control unit.

or the other way 'round. But to me, at least, the emphasis is on recording—and home recording at that. Which is as it should be since Nakamichi's first claim to fame (in this country at least) was with its top-performing cassette decks.

With this emphasis, it is understandable that in the model 610 such conventional preamp features as tone controls and noise filters are omitted. Should you want to add pre- or post-recording equalization you will have to do so with a separate graphic equalizer. But I'll settle for that arrangement any time when it means distortion levels (in both phono and high-level input modes) that comes as close as they did to the residual distortion of our test instruments.

No matter what cassette deck you now own (unless it's one of Nakamichi's top units) you'll be better off using the peak-reading, wide-range level meters on the 610 preamp, ignoring the meters on your own deck (except for initially establishing a continuous-tone correlation between the two sets of readings). Imagine being able to read a dynamic range of better than 50 dB while recording on a cassette deck. That facility alone should do away with over-recording or recording so that low levels are buried in the noise floor of the deck or the tape used with it.

In my view, the 610 is an ideal instrument for "live" recording efforts by amateurs. The serious audiophile also should note the built-in test tones and the pink-noise signal source, useful for room equalization with a graphic equalizer.

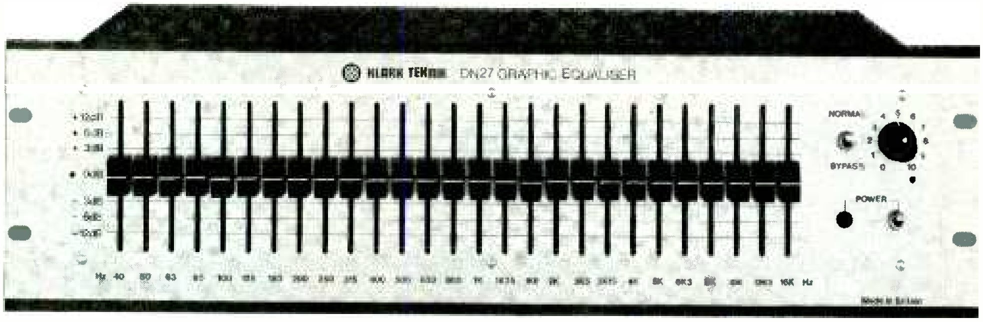
The only function of the 610 I could not check out was the series of three pushbuttons for making A-B comparison tests of three sets of speakers, or between different power amplifiers driving the same pair of speakers. To do so requires using the optional RM-610 remote control unit.

Individual Comment by N.E.: The model 610 has all the attraction of a combination-product (not unlike an electric tool that serves as a drill and a cutter, etc.), and some of its limitations too. That is to say, it is a system preamp but it lacks tone controls and noise filters. It has a performance-checkout facility but the test tones available go no lower in frequency than 1 kHz (which of course is adequate for tape tests and calibrations but not for complete testing of other audio areas). And to use the test facility to its full potential requires buying an additional unit, the $75 RM-610 accessory. As a mixer, however, for applications just below the studio requirements, it is fine. And as a recording control center for the advanced amateur or semi-pro it seems unimpeachable. It is a unique product; there is nothing like it on the market to my knowledge, and it may well prove to be the kind of device that bridges the gap between the rank amateur recordist and the professional.

NAKAMICHI 610 CONTROL PREAMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTICS	LAB MEASUREMENT
Frequency response (line level)	±1 dB, 8 Hz to 120 kHz
RIAA Equalization	accurate to within 0.2 dB, 50 Hz to 15 kHz
Input sensitivity/impedance:	
Mic/1 K (attenuators, −15, −30 dB)	0.2 mV
Phono/200, 50 K, 100 K	1.0 mV
Aux; tuner/25 K	75 mV
Tape playback/75 K	235 mV
Tape monitor/75 K	320 mV
Maximum input levels:	
Mic (att: −30 dB)	1.3 V
Phono	280 mV
Aux, tuner, tape playback	50 V
Maximum output at clipping, monitor out/1 K	6.0 V
Signal-to-noise ratio (IHF A, phono)	92.5 dB
Equivalent input noise	−142.5 dB
Distortion (rec. master at −20 dB; monitor vol. at max.; line out at 2 V):	
mic	0.007%
phono	0.0025%
Power consumption	12 watts

Klark-Teknik DN27 Graphic Equalizer



General Description: The Klark-Teknik DN27 Graphic Equalizer is a British-made product distributed in the U.S. by Hammond Industries of Syosett, N.Y. It is a single-channel device with 27 individually-adjustable frequency bands. Nominal center frequencies are 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1 K, 1.25 K, 1.6 K, 2 K, 2.5 K, 3.15 K, 4 K, 5 K, 6.3 K, 8 K, 10 K, 12.5 K, and 16 K Hz. Each slider has a range from −12 to +12 dB; markings show settings for the 12, 6, 3 and 0 dB values each way. In addition to these controls, the front panel has a power off/on switch and pilot lamp, a normal/bypass switch and a level control marked from 0 to 10. The rear of the device has the AC power socket, plus a pair of XLR connectors for input and output signal hookups. The front panel is suited for standard rack-mounting, or, since it is supplied with small hard-rubber "feet," it also may be placed on a shelf or cabinet top.

Intended primarily for the professional user, the DN27 has various applications in many areas of sound including compensation for inadequacies of signal sources and/or of preamps, to emphasize certain frequency bands for desired effects, for use in recording, or as a speaker or "room" equalizer in playback. The unit is rated to handle signal levels of up to +21 dBm before clipping, and the sliders and their associated circuitry are designed to permit changes during "live" programs without introducing instability or added noise. The use of actively coupled LCR filters in a special configuration is said to provide optimal band pass/stop characteristics.

Since the input is directly coupled to the level control, a constant impedance of 10 K ohms is maintained for all input sources. An input buffer amplifier can be adjusted internally to provide up to 20 dB gain for low-level operation with no increase in noise. A hybrid power operational amplifier, with low output impedance and high slewing rate, helps keep the output level constant over a wide range of loads, without increasing distortion.

Test Results: With its unusually large number of individual band adjustments, and the accuracy of those adjustments, the Klark-Teknik DN27 is an impressive device that should be of genuine interest to sound-professionals or even, possibly, the very affluent and all-out home sound-buff. Its equalization action is precise and fine, and the audio segments into which it divides the total spectrum are narrow enough to effectively reduce hall peaks, decrease P.A. feedback, and fill in sound-system "valleys" no matter how narrow they may be.

For all practical purposes, the unit can be operated with outputs running as high as +21.5 dBm, or up to +18 dBm if you want to keep THD levels well below 0.1%. Since residual noise is at a −90 dBm level, the unit has a total dynamic range of at least 108 dB, which is excellent.

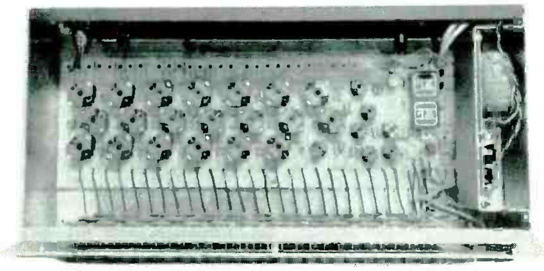
In lab tests the device met or exceeded its published specifications, and in the view of MR's testing staff—when it is patched into a recording or P.A. chain—the DN27 is not likely to add any distortion or frequency aberrations other than those you deliberately choose to introduce yourself.

Special note: On the test samples we examined, the power switch had to be moved down for the "on" position instead of up, as is customary on electronic units. In addition, the XLR connectors were wired "backwards" with respect to ground and hot sides. According to the U.S. distributor, Hammond Industries, these points are being corrected in England for future shipments.

Klark-Teknik DN27: Rear panel view.

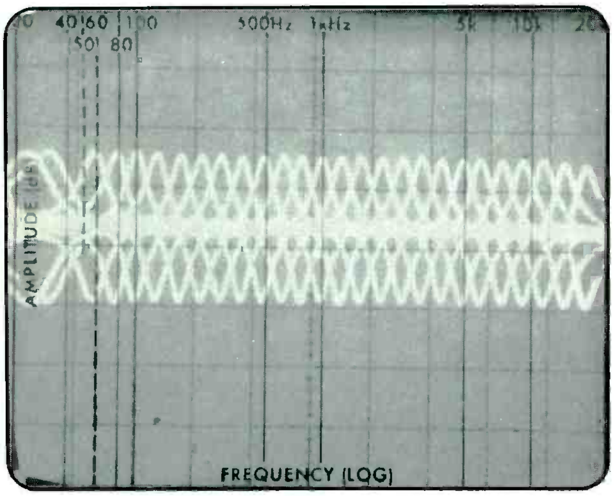
General Info: Dimensions are: 19 inches wide; 5¼ inches high; 8 11/32 inches deep. Suggested retail price, $649.

Individual Comment by L.F.: The operation of the DN27 points up dramatically the difference between "home type" graphic equalizers and those truly intended for professional recording and "room-



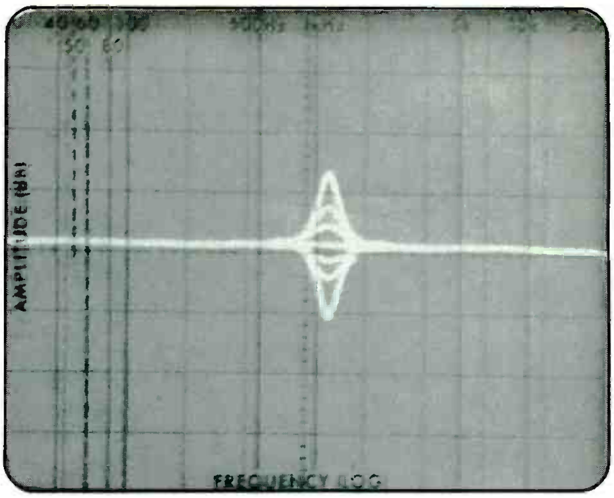
Klark-Teknik DN27: Internal view.

voicing" applications. The sheer number of adjustable frequency bands, organized at approximately one-third octave intervals, combined with their extremely accurate and precise action, make for an equalizer which permits fine adjustments beyond anything I have yet encountered. I even found it possible to adjust adjacent bands in opposite directions and encountered little or no interaction. Examination of the circuitry discloses the lengths to which the designers have gone to make such fine adjustments possible. Fourteen of the twenty-seven filter networks are switched into the circuit via one feedback path, while the remaining interleaved thirteen networks employ a separate feedback network, so that adjacent bands are

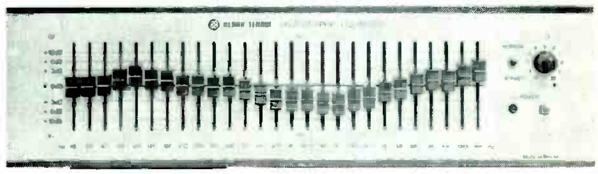


Klark-Teknik DN27: Response range, all 27 bands.

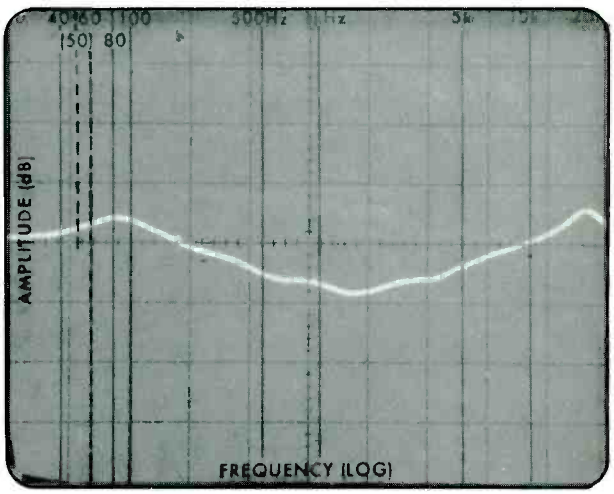
effectively isolated from each other. Discrete L and C components are used for each band (none of your fancy new "gyrator" inductors here), but the L components used are of the finest quality, and I saw no evidence of inductance saturation at any operating level at which the unit was tested.



Klark-Teknik DN27: Intermediate setting response of 1.25 kHz band slide control.



Klark-Teknik DN27: Front panel view.



Klark-Teknik DN27: Overall response obtained with controls set as shown in front panel view.

Individual Comment by N.E.: Tests indicate that the DN27 equalizer can "fine-trim" an audio system much more precisely and in greater frequency-band detail than most. It also is comforting to know that when the slider on the front panel is set to the indicated marking of +3 or +6 dB, etc., the actual degree or boost or cut is going to be just that. In the lab, oscilloscope photos were made for control ranges that confirmed the accuracy of slider positions. Each band has exactly the same range from top to bottom (you could draw a straight line across the upper and lower edges of the twenty-seven response waveform "envelopes") and, with the exception of the first two bands, there is uniform spacing between all frequency centers.

With this kind of performance and at its announced price, it seems obvious that the DN27 is not intended for everyday audiophile use. It is a professional device in every sense of the word. My only question concerns the possible desirability of dB markings more specific than the 3 dB gradations indicated.

KLARK-TEKNIK DN27 EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTICS	LAB MEASUREMENT
Number of bands	27
Input impedance	10 K ohms
Output impedance	4 ohms
Operating level	−20 dBm
Center frequency accuracy	±2%
Calibration accuracy	±0.3 dB
Frequency response	20 Hz to 20 kHz, ±0.4 dB
Output clipping	+21.5 dBm into 600 ohms
THD at 1 kHz, +4 dBm, 600 ohms	0.008% (mostly noise content)
THD, 20 Hz to 20 kHz, +18 dBm, 600 ohms	0.0055% at 20 Hz 0.0065% at 1 kHz 0.15% at 20 kHz
Equivalent input noise	−90 dBm

dbx Model 3BX Dynamic Range Expander

General Description: The dbx model 3BX is a sophisticated volume expander for stereo signals. Each of its two signal channels provides adjustable expansion in three frequency bands, not specified as to actual frequencies but designated as high, middle and low. Each band is displayed by its own row of ten LEDs marked from −20 dB to +12 dB on the front panel. The lefthand group of LEDs, colored yellow, indicate downward expansion (volume decrease), and the righthand group, colored red, indicate upward expansion (volume increase).

The avowed aim of this device is to restore to program material (from broadcasts, discs or tapes) the dynamic range (difference between loudest and softest passage) that is restricted for one reason or another in order to suit the program material to a given sound medium or format by means of signal-compression or "limiting." A simplified way of thinking of volume expansion is that it makes the louds louder and the softs softer. As such it is primarily a device for use in a playback system. However, the 3BX also could serve as an ancillary device during tape-recording of sources which have been previously compressed. However, expanding a program and copying it onto tape may produce a dynamic range that exceeds that of the recorder (except for highly compressed program sources); to overcome this possible problem, dbx recommends in this application the use of another device—a tape-noise reduction unit—patched into the recording system as per instructions.

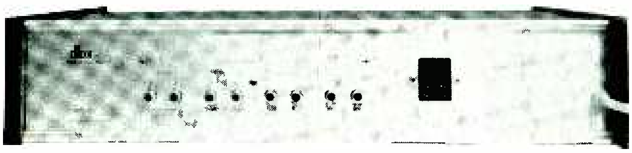
In any expansion system, the circuit requires some form of sensing in order to "know" when to act. The dbx unit employs "rms detection" (the letters stand for root-mean-square) which is said to be superior to other techniques in that it responds precisely and accurately to all changes in input level, but without over-reacting on transients or "noise spikes." The change in signal level is accomplished by a voltage-controlled amplifier circuit which, according to dbx, varies the program level not by a fixed amount but rather on a "linear decibel" basis—i.e., the dynamic ranges of output and input are linearly related by the expansion ratio (as chosen by the user via a control). Attack and release times, instead of being predetermined or fixed, are allowed to vary so that they follow the "envelope" (the changing level of a waveform with respect to time) of the program. The use of three frequency bands, as opposed to a single overall frequency band, is claimed to eliminate audible "breathing" and the possibility of a strong tone in one frequency region causing audible effects in another frequency area.

A detailed explanation of these design factors, plus a generous amount of information on the subject of volume expansion, compression, noise-limiting, etc., is included in the owner's instruction booklet for the dbx 3BX range extender.

The unit's front panel contains a power off/on switch and, next to it, a switch for use with an optional remote-control unit. A horizontal slider controls the degree of expansion and is marked in increments of 0.1 from 1.0 to 1.5. The amount of gain change in each of three frequency bands is shown on the LED display mentioned. To the right of this display is another horizontal slider, the "transition level" control, which sets the level at which expansion occurs; the general idea here is to set this slider so that the yellow LEDs glow during quiet portions of the music while the red LEDs glow for loud passages. Four more buttons are marked "source," "tape," "pre," and "post." The "source" button selects programs coming into the expander other than from a tape recorder. The "pre" button is pushed in to expand a program before recording it. The "tape" button, pressed together with the "post" button, permits expanding material played back from a tape recording. Pressing the "pre" and "post" buttons simultaneously bypasses the expander functions.

The rear of the device contains signal connections. These are standard phono pin-jacks for stereo signals from and to preamp connections, plus tape-recorder outputs and inputs. The remote-control socket and the unit's power cord also are at the rear.

Test Results: Lab measurements generally confirmed or exceeded published specifications for the dbx 3BX. Distortion and noise level were both better than claimed. MR measured 5 mV higher than stated for the bottom of the transition level range (35 mv as compared to 30 mV specified), but got a full volt better at the top of the range (4.0 V as compared to the 3.0 V

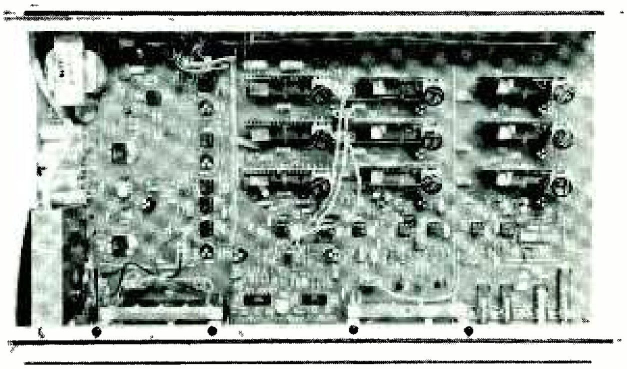


dbx 3BX: Rear panel view.

specified). Maximum output level in MR's tests was 6.8 V as compared to the 7 V specified.

Photos of an oscilloscope pattern monitoring the circuit action of the device confirm that the 3BX was working "as claimed." Since this is better understood by seeing the actual 'scope traces, the reader is referred to the accompanying photos and explanation.

In listening tests, there was some mixed opinion as to the desirability of the device (see individual comments below). In general it was agreed that the expansion ratio setting is the most critical adjustment on the unit, and that incorrectly setting it could exaggerate the expansion effects.



dbx 3BX: Internal view.

General Info: Supplied in metal case with wood side panels and small rubberized "feet." Dimensions: 17¾ inches wide; 3¾ inches high; 10¼ inches deep. Weight: 8 lbs., 10 oz. Suggested retail price: $650. Owner's manual for the unit is excellent, very informative, amply illustrated.

Individual Comment by L.F.: The ability of the 3BX to determine which bands of frequencies require expansion at any given instant is what makes this product so superior to earlier expanders from this company. Proof that it does work is shown in the accompanying 'scope photos. dbx is deliberately secretive about the crossover frequencies they use in dividing the expander into three bands. Apparently, the device's effectiveness depends, in part, on a judicious choice of those frequencies, and dbx isn't about to tell the competition what those frequencies are, or what the rate of slope is from one band to the other. As a curious tester this leaves me a bit miffed. But as a critical listener I couldn't care less. The fact is, the dbx 3BX is absolutely inaudible in its action—and I tried all kinds of music from hard rock to chamber music and symphonies to try to "trick it" into audible breathing or pumping. All I got from the 3BX is a restoration of full dynamic range with no "side effects." Adjusting the expansion ratio control is critical, but with a little practice you can find the setting that is just right for every record, tape or FM program you care to play. In discussing this product with one of the representatives of dbx, I was tempted to ask whether the 3BX will some day be incorporated with elements of their 120 series compander in a single product. Obviously, the people at dbx had thought of the same thing, but at the moment they are holding off, feeling that the price and complexity might limit the marketability of such an all-in-one product. I'm not so sure! But, in the meanwhile, if you want the benefits of a superb 3-band expander and increased dynamic range capability when making your own tape recordings, you'll still need both the new 3BX expander and dbx's earlier Model 122 or 124 encode/decode compander to do the job.

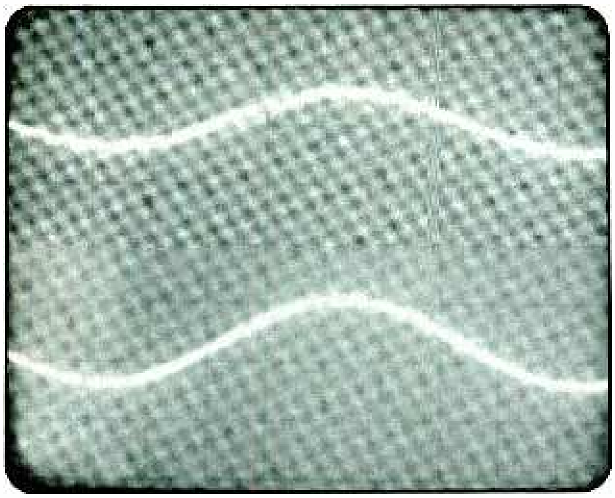


Fig. A

dbx 3BX: In Photo A, upper trace is input signal, a low frequency whose level is just about at the threshold. Therefore, lower trace (output signal) is almost equal in amplitude.

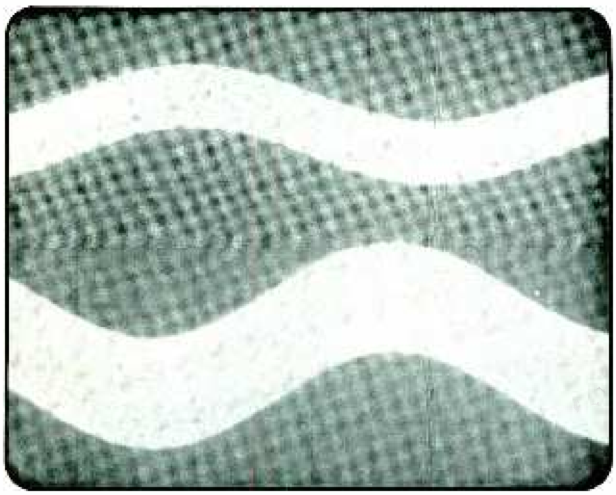


Fig. B

dbx 3BX: Without changing gain settings on either the oscilloscope or on the 3BX, we superimposed a higher-amplitude high-frequency signal, mixed in with the original low-frequency signal. This is shown as the upper trace in photo B. Since the amplitude of the high-frequency signal was above the threshold level, the output (lower trace, photo B) shows that the high-frequency content has been expanded considerably more than the low-frequency portion. In a one-band conventional expander, all elements of the composite signal would have been expanded equally. These photos attest to the quality of the 3BX to determine which bands of frequencies require expansion at any instant.

Individual Comment by N.E.: I have never been fully convinced of the fittingness of a volume expander used in a high-quality playback system. To explain why would entail too long (for this space) a discussion of recording philosophy, recording technology and the dynamic interrelationships of audio components. A hint to this viewpoint is the question of whether a hi-fi system is supposed to render a faithful replica of the input signal or to try to "improve" on it. If the latter, we get into a kind of "open-ended" discussion in which many viewpoints—no less valid for their differences—are possible. The question of dynamic range seems to me the most difficult to pin down. Not only are highly subjective factors involved from the perception standpoint, but so too are many other variables—from specific recordings to the power capability of a particular amplifier/speaker combination and the effect of room acoustics.

Be that as it may, I will allow that the dbx 3BX seems a more sophisticated and advanced kind of volume expander than others I have experienced, but only when its expansion control is carefully adjusted for a given program source. Incorrect adjustment of this control can introduce an unnaturally heavy bass into the signal which not only is objectionable from a listener's standpoint but which also can drive a power amplifier to near clipping levels. It is not enough to say simply that one can set this control correctly for FM, or tape, or disc as such—one may need to adjust it for each station tuned in on FM, if not for different selections played on the same station. On the user's own equipment, each record and each tape will also require readjustment for best results. With these reservations stated, I would say that the 3BX might be your audio cup of tea if you favor volume expansion on playback or, more likely, if you are into "creative recording" in which case you also will want a noise-reduction device such as those made by dbx.

dbx 3BX DYNAMIC RANGE EXTENDER: Vital Statistics

PERFORMANCE CHARACTERISTICS	LAB MEASUREMENT		
Expansion	1:1.5 maximum, each band		
Dynamic range	110 dB		
Transition level range:	35 mV to 4.0 V		
Frequency response (at 1:1 expand)	±0.5 dB, 16 Hz to 20 kHz		
Harmonic distortion, 20 Hz to 20 kHz:	20 Hz	1 kHz	20 kHz
Max. expand, 1 V out	0.1%	0.04%	0.06%
Min. expand, 1 V out	0.02%	0.035%	0.09%
IM distortion	0.013% at maximum expansion, 5 V out		
Input noise below 1 volt, "A" weighted	92 dB down		
Input impedance	50 K ohms		
Output imedpance	Okay for 5 K ohms or higher		
Max. output level, 1 kHz, 5 K load	6.8 V		
Power consumption	30 watts maximum		

By Jim Ford and Brian Roth

In this issue, our first in MR*'s new monthly schedule, we are extremely pleased to present an equally new staple—the "Hands-On Report." The section will be authored by the gentlemen who received much acclaim for their "P.A. Primer" series printed in previous issues—Jim Ford and Brian Roth.*

The Hands-On Report will review equipment which, by its nature, lends itself to lab and *field testing, in order to establish a proper critical evaluation.*

We are certain that this section will make MR *of even greater informative value to you and, as usual, we would like for you to send us your comments.*

General Description: The Yamaha PM-1000 mixing console has become a very popular sound reinforcement mixer due to its professional features and moderate price. The PM-1000 is basically a 16-input, 4-output mixer of modular construction; Yamaha also offers 24- and 32-input versions. Each input module contains the following:

A. Slide fader volume control
B. A rotary switch to vary input gain
C. A two-frequency bass roll-off filter
D. A three section equalizer with:
  (1) Low-frequency boost or cut
  (2) Mid-frequency boost or cut (choice of three frequencies)
  (3) High-frequency boost or cut
E. Two echo send "pots" (both are connected pre-input fader)
F. A cue (solo) pushbutton
G. A phase reversal switch
H. Four output selector push buttons
I. A stereo pan pot that operates in conjunction with the output selector pushbuttons.

Yamaha PM-1000 Mixing Console

Each of the four output modules contains a slide fader volume control and a portion of a "4 x 4" monitor mixer. This section allows the four main outputs to be remixed into four monitor outputs. The console also includes a headphone output, a talkback section and a two-frequency oscillator.

With one exception, all inputs and outputs utilize "cannon" 3-pin connectors and are balanced and transformer isolated. The only connections not using the "cannon" connectors are the four patch points that are wired into the submaster modules; they use quarter-inch phone jacks. Also, the signal level at these jacks is about 20 dB below the main outputs which are calibrated for +4 dBm (1.23 volts) at 0 VU.

Field Test: Since the mixer is the main control center in a PA system, total reliability is crucial. The PM-1000 performed flawlessly at more than twenty different jobs over a period of several months.

To our ears, the mixer was very quiet—which indicates a superior signal-to-noise ratio. In fact, the PM-1000 seemed to be the quietest (in its price category) mixer we have used. Additionally, no overload distortion was heard at any time if the mixer was properly operated. Both the phase reversal switch and the low-frequency roll-off filter were most useful in several difficult auditoriums.

On the minus side, we were not totally pleased with the sound of the equalizer. In particular, we feel that

the mid-range section is somewhat "peaky" and was of limited usefulness in a "live" situation. Also, we would have preferred that the echo sends be post-fader instead of pre-fader, and that the console had echo returns (although an input module can be used for a really flexible return). The headphone circuit had insufficient volume, particularly when attempting to use the cue (solo) pushbuttons. Finally, the patch points on the submasters caused problems when used to drive a limiter or equalizer; the level at these jacks should have been the industry standard +4dBm as are the main outputs. *Note:* We were very pleased to find that the console could drive some of the new generation compressors such as the UREI LA-4A and dbx 160. It was necessary to install attenuator pads at the outputs of these compressors to "match" their levels back into the console.

Although we do not usually operate a PA in stereo, we nevertheless checked the operation of the panpots. We found them to be "touchy" and to cause volume "jumps" when panned near the center of their range.

Basically, all of the major functions of the mixer performed admirably and we were impressed favorably with the abilities of this mixing console.

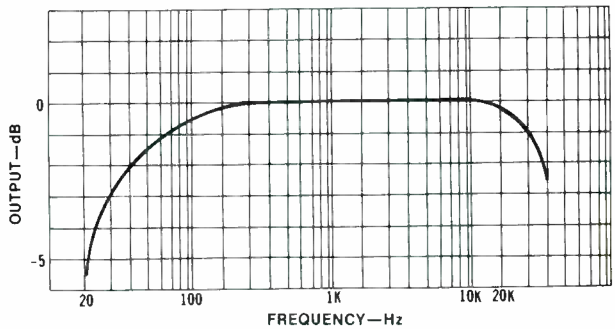
Lab Test: The tables and graphs tell the story about the PM-1000. Noise levels are low, especially considering that this is a sixteen input mixer. The section labeled "typical mix" was a fairly random setting of input levels and equalization similar to those we used during a typical show. Although this measurement is not particularly "scientific," we feel this figure demonstrates the noise level for actual situations.

Frequency response was adequate with the response extremes being intentionally rolled off. The outputs are capable of producing in excess of +22 dBm into 600 ohms, and even more output voltage into higher impedance loads. The distortion figures are very good, and the harmonic distortion residual consisted of lower ordered products.

One thing we encountered during testing was that the console inputs could not accept unbalanced sources. This is absolutely no problem with microphones, but will cause high distortion and low-end roll-off if a tape recorder or similar equipment with unbalanced outputs is fed into the mixer.

Mechanically, the unit is very sturdy, and accessibility for servicing appears excellent. The power supplies are quite robust for a mixing console. This is reassuring since if the supply blows, the show goes!

Conclusion: We feel that the PM-1000 is one of the very best mixers we have used, even with some of its peripheral shortcomings. It behaved in an excellent manner and produced some of the cleanest (particularly in the noise department) sound we have obtained with our PA system. The input signal handling ability was superlative for any level, and with the exception of the pan pots the controls were smooth and precise. Having transformer isolated inputs and outputs was a big plus in eliminating grounding problems. All in all, the PM-1000 is a most remarkable performer.



Yamaha PM-1000: Frequency response.

Performance Specifications

Output noise (20 Hz-20kHz unweighted) referenced to 0 VU (+4 dBm)

Master fader normal, all inputs off	−77 dB
Master fader normal, 1 input set at +4 position	−77 dB
Master fader normal, 1 input set at −44 position	−76 dB
Master fader normal, 1 input set at −60 position	−68 dB
Equivalent input noise	−124 dBm
"Typical mix" (see text)	−75.5 dB

Harmonic Distortion at 0 VU

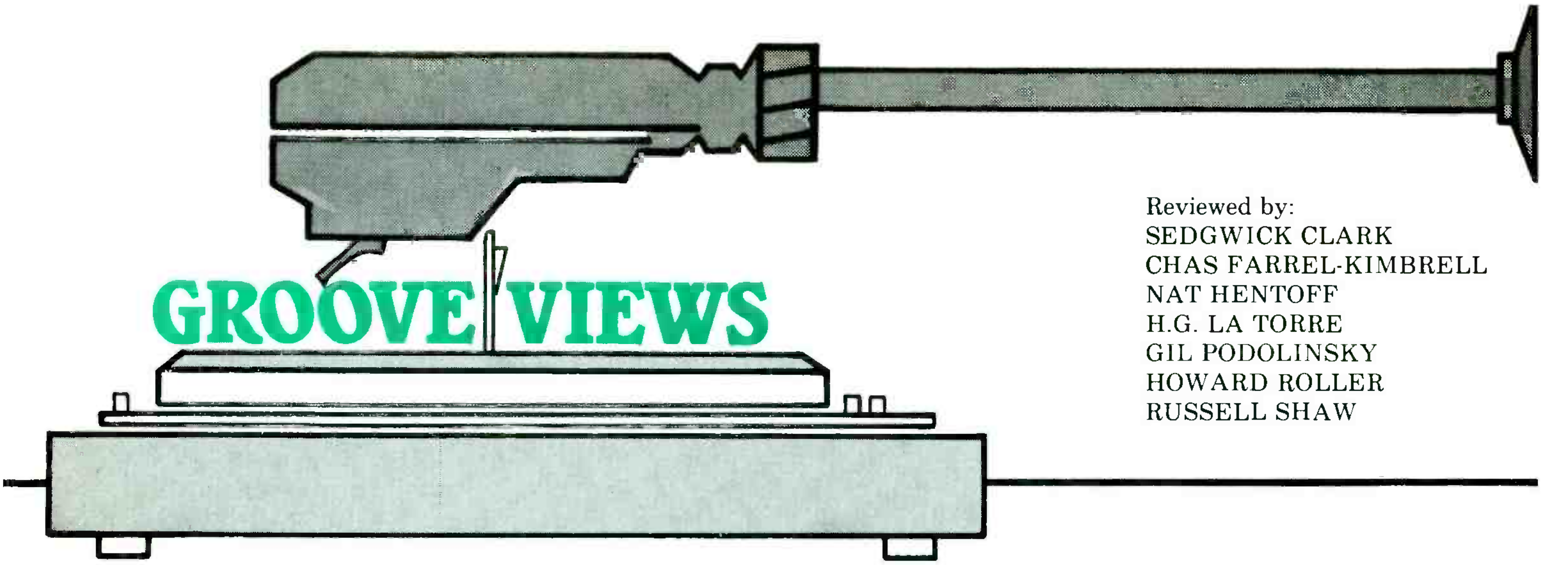
Frequency	*Distortion*
50 Hz	.1%
1 kHz	.07%
10 kHz	.09%

Harmonic Distortion at +20 dBm Output

Frequency	*Distortion*
50 Hz	.5%
1 kHz	.04%
10 kHz	.1%

Intermodulation Distortion—60 Hz and 7000 Hz mixed 4:1

at 0VU:	.12%
at +20 dBm:	.45%



GROOVE VIEWS

Reviewed by:
SEDGWICK CLARK
CHAS FARREL-KIMBRELL
NAT HENTOFF
H.G. LA TORRE
GIL PODOLINSKY
HOWARD ROLLER
RUSSELL SHAW

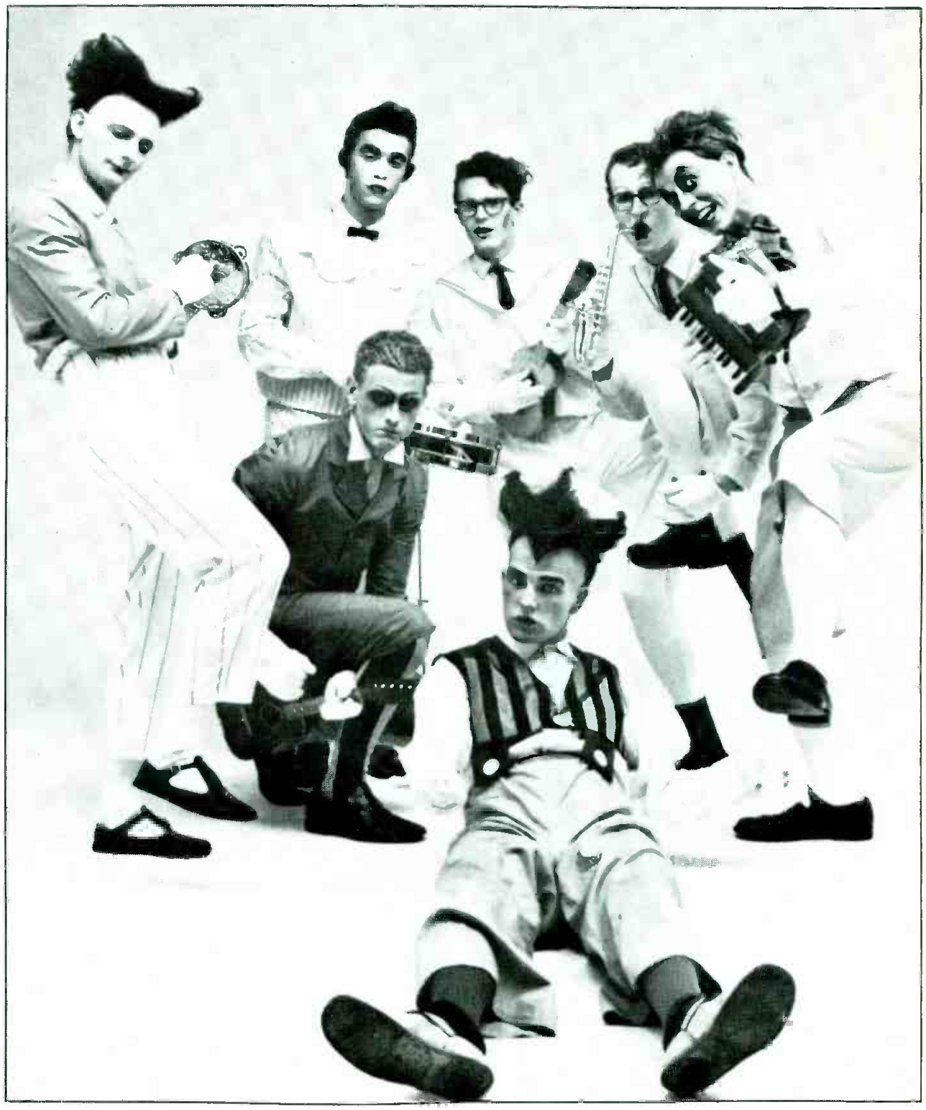
POPULAR

SPLIT ENZ: *Mental Notes.* [Phil Manzanera, producer; Rhett Davis, Guy Bidmead, engineers. Recorded at Basing St. Studios, London, England.] Chrysalis Records CHR 1131.

Performance: **Genuine**
Recording: **Superior**

Split Enz, another Australian entity, is difficult to categorize. I hesitate to label them theatrical, bizarre or experimental, but their use of various musical forms makes for an uncommonly demanding engineering mix. This seven member group utilizes percussion, acoustic, electric, and synthesized instruments as well as various horns and woodwinds. Definitely not a horn band, each instrument is used to *accent* more than to accompany, requiring great patience and expertise in balancing. Typically, the acoustic piano achieves sharpness as the sole instrument behind the vocal without forcing an unnatural edge to the vocal track. Separations are extremely exact, careful not to infringe upon the group's use of musical space while at the same time not making the separations too wide and sound totally unrelated.

The selection of Phil Manzanera as producer is an excellent one, for his own recordings share a common avenue of conceptual concerns as those exhibited by Split Enz. The use of space in this recording is quite remarkable for as a completed work it does not affect areas of continuity, clarity or enjoyment as is so often found in experimental jazz. Rather, it enhances the weight of the production, and enlivens the recording by arranging the instruments to create natural points of emphasis rather than the commonplace surge of electrical current. Further, this meticulous use of space enables the group to combine influences of the 20's and 30's with modern day style and vision. Happily, the producer and engineers don't rely



SPLIT ENZ: Making uncommon demands.

upon tired sound effects tricks or gimmicks to achieve their end. For a first effort, this album ranks, recording-wise, as one of the best I've yet heard.

G.P.

JEFF BECK: *Wired.* [George Martin, producer; Pete Henderson, Dennis McKay, John Mills, John Arrias, Jan Hammer, Geoff Emerick, Mark Guercio, engineers; recorded at Air Studios and Trident Studios, London, England, Cherokee Studios and Sound Labs, Hollywood, Ca., Caribou Studios, Nederland, Co., Red Gate Studios, New York, N.Y.] Epic PE 33849.

Performance: **Beck-tech**
Recording: **Straight ahead**

Wired is probably some of the best technical work Beck has done since the *Truth* and *Beck Ola* albums. The guitar in his hands is like a marionette. Beck pulls the strings and the guitar moves, talks and takes on a personality. The musicians on this album defintely didn't hold Jeff back. Max Middleton and Jan Hammer handled keyboards and synthesizer. Narada Michael Walden carried most of the drumming chores (with the exception of two cuts played by Richard Bailey, Beck's *Blow By Blow* drummer, and one cut by Jan Hammer) and Wilbur Bascomb played bass guitar.

The album is more Chick Corea flavored than Beck's previous album *Blow By Blow,* probably due to the choice of musicians (less rock, more jazz). Still the numbers retain that unmistakable Beck quality. Beck seems to be developing more of a jazz interest these days. For instance, can you imagine Beck playing Mingus? Well, he does! (I don't know how Mingus fans feel about it, but Beck fans should love it.) The number is basically a blues tune and even though Beck isn't especially noted as a blues guitarist, by the time he finishes the song it is unimportant that it was blues or that it was Mingus.

Most of the other material flows in the vein of a Jazz/Funk fusion. All the material is extremely listenable and is played with freshness and real enthusiasm. One track of special interest is Jan Hammer's "Blue Wind" which is written, produced, engineered, and remixed by Hammer. The only musicians on the track are Beck on guitars and Hammer on drums and synthesizer (that's right, drums). The song has

JEFF BECK: Refreshing enthusiasm.

more drive than any other track on the album and Hammer blends beautifully with Beck. Narada Michael Walden makes his piano debut on a track he wrote which was recorded with *no* drums. Sound interesting enough?

Recording-wise, the album is clear and clean. Beck's guitar is really up-front with a lot of bite. But for some reason all the keyboards seem a bit too far back in the mix causing them to be occasionally lost. The drums sound a little too muffled and flat (this of course is a matter of taste). The snare has that cardboard box sound (still, very clean, not to be confused with muddy, lest anyone should feel I'm contradicting myself).

Overall, the album is very well planned and flows evenly from one track to the next. There's no complaint about the production but that should figure when the producer is George Martin. With no more problems than exist on this album, who can complain?

C.F.-K.

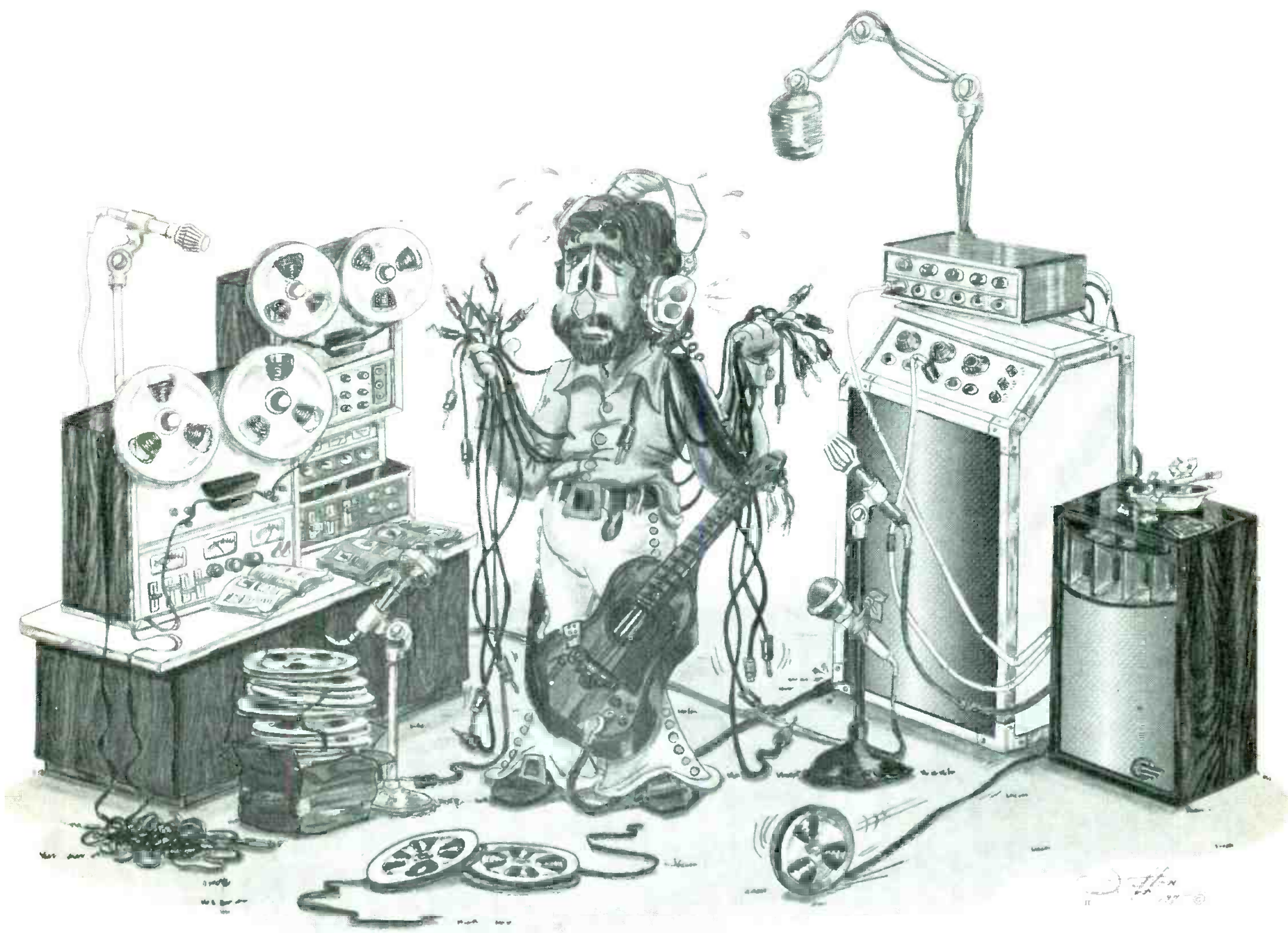
SANTANA: *Festival.* [David Rubinson, producer; Fred Catero and David Rubinson, engineers; recorded at Wally Heider Recording, San Francisco, Ca.] Columbia PC 34423.

Performance: **Tiresome rehashes**
Recording: **Functional**

At long last, Carlos Santana seems to have shed his penchant for endlessly repeated chord solos, invocations to an ever-changing deity, and hackneyed rehashes of the McLaughlin guru drone. Opting for a return to his Latin roots, Carlos seems to be recycling

SANTANA: Not taking creative risks.

GET SMART!



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some abandoned riffs rather than pursue a policy of musical exploration.

The result of Santana's latest quest for the lost chord is a tiresome cooks tour of all the Latin-funk cliches that have signified the boring aspects of much of his earlier work; single-layered, pointedly uncomplex rhythmic textures, choruses of banal lyrics, and lifeless pseudo-spontaneity. And while the playing on *Festival* is quite flawless, the scope and range of subjects tackled is quite limited. Boredom is an overbearing visitor.

A more imaginative mix might have alleviated some of the tonal doldrums; track after track finds the interesting keyboards of Tom Coster buried in favor of several relentlessly monotonous congas. What's more, Carlos' solo acoustic contribution, the potentially interesting "Verao Vermelho," finds its introductory mellowness aborted midstream by the sudden arrival of some extraneous percussive artifacts. The wooden intrument is buried once more.

If Carlos and company wish to rival the Salsoul Orchestra in the metronome department, then Rubinson has provided the perfect laboratory setting; innovative rides are muted, and antiseptic cleanliness is substituted for the obligatory risk-taking which characterizes true creative efforts. R.S.

BLUE OYSTER CULT: *Agents of Fortune.* [Murray Krugman, Sandy Pearlman, David Lucas, producers; Shelley Yakus, Andy Abrams, engineers; recorded at the Record Plant, N.Y.C., N.Y.] Columbia PC 34164.

Performance: **A calculated blitz**
Recording: **Excellent**

Onstage, BOC comes off as another contender in the high-decibel/low-energy, heavymetal sweepstakes. The redundant wall of sound/noise lacks dynamics and the vocals—an important element of the BOC mystique—certainly lack clarity.

But it can't be denied that BOC has a rapidly expanding sphere of loyalists—in the last year, they've reached headlining status at medium-to-large size halls. Last year's commercial success of the "Reaper" single from this album may offer a clue, but a careful perusal of BOC's vinyl performances best explains the band's large concert draws.

Agents, BOC's fifth album release, is brilliant. The highs, midrange and lows are clearly defined. The performances, especially those by lead guitarist Donald (Buck Dharma) Roeser, are thoughtfully inspired. The vocals alternate between hauntingly beautiful ("Don't Fear The Reaper") and warbled dementia ("Tatoo Vampire" and "This Ain't the Summer of Love"), but most importantly, said vocals are *intelligible*.

The listener will have to decide if BOC's quasi-Aryan themes of contemporary kinkiness please or disgust. In either case, the album offers a great deal more than your basic rock and roll production. The tracks are loaded with imaginative little guitar and keyboard riffs, vocal wig-outs and phasing effects that are mixed so subtly that an inferior turntable won't pick them up. There's a good deal of lead guitar and vocal transfer from channel to channel, and the delay used on Buck's lead in "Tenderloin" is quite effective.

In short, one doesn't have to be sandwiched between Quaalude and Coors freaks to enjoy BOC's lyric outrageousness and accomplished

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musicianship. All that's needed is a good stereo at home, where, if I may add, it's certainly safer. S.P.

CLASSICAL

GRIEG: *Works for Orchestra, Vol. 1.* Utah Symphony Orchestra, Maurice Abravanel cond. [Marc Aubort & Joanna Nickrenz, musical supervision; recorded in the Mormon Tabernacle, Salt Lake City, Utah, in Feb. and Mar. 1975.] Vox QSVBX 5140 (three records).

Performances: **Good**
Recording: **Warm and resonant**

Here is another worthwhile Vox Box. This one is a follow-up to Vox's 1970 box of Grieg's complete piano music. Maurice Abravanel and his fine Utah Symphony are their usual solid selves, contributing over two hours of delightful listening. Grieg may not have been the headiest of composers, but he surely penned some of the most beautiful and relaxing music to be heard.

This first volume, of which another will presumably follow shortly, contains: *In Autumn, Op. 11*; *The Bridal Procession Passes By, Op. 19; Peer Gynt Suites 1 & 2, Opp. 46 & 55*; *Two Elegiac Melodies for String Orchestra, Op. 34*; *Of Holberg's Days, Op. 40*; *Old Norwegian Romance with Variations for Large Orchestra, Op. 51*; *Lyric Suite, Op. 54*; *Sigurd Jorsalfar, Op. 56*; *Symphonic Dances, Op. 64*; *Evening in the Mountains*; and *At the Cradle* (Lyric Piano Pieces No. 4 and No. 5 of Op. 68 set for orchestra).

The *Peer Gynt Suite, Lyric Suite* and *Symphonic Dances* are well known, especially the first-named, and perhaps more characterful performances could be found in the catalogue. But anyone desiring this music should hesitate no further, for the smaller pieces in this selection are no less melodically inspired.

The sound is typical of this recording team in the hugely resonant warmth of the Mormon Tabernacle. The pick-up is distant, adding to the romantic aura of the music. One problem with such an acoustic, however, is that tape splices are more apparent. Good as the editing is, every so often the reverb from an orchestral passage will suddenly be cut off by a solo line that would have matched more easily in a drier ambience. But this is merely a technical observation which will hardly interfere with one's enjoyment of the music. Surfaces were variable.

S.C.

MAHLER: *Das Lied von der Erde.* Janet Baker, mezzo-soprano; James King, tenor; Concertgebouw Orchestra, Amsterdam, Bernard Haitink cond. [Volker Straus, producer.] Philips 6500.831.

Performance: **Autumnal**
Recording: **Warm and spacious**

DVORAK: *Symphony No. 7.* Concertgebouw Orchestram, Amsterdam, Colin Davis cond. [Vittorio Negri, producer.] Philips 9500.132.

Performance: **Energetic**
Recording: **Gutsy**

TCHAIKOVSKY: *The Nutcracker* (Complete Ballet). Concertgebouw Orchestra, Amsterdam, Antal Dorati cond. [No producer listed.] Philips 6747.257 (two records).

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A Jazz Natural and Jazz Futures

by Nat Hentoff

Zoot Sims is a natural. A natural swinger, a natural melodic improviser. And always, his tone and spirit are naturally warm, more often hot. But Zoot has not moved forward for a long time, some have said. He plays as if there had been no late Coltrane or late Miles. On the other hand, on what tablet of immutable law is it written that everybody has to be on the cutting-edge of the avant-garde? If Lester Young were still alive, he would still be Prez, and only Prez; and we would all be most grateful. Or, as arranger Bill Holman says, "People have wondered why Zoot doesn't progress. I figured it out—it's simply that guys like him don't need to progress; they just mature. With his talent, what else do you need?"

Now, in *Hawthorne Nights* (Pablo), the overwhelming authority of the mature Zoot Sims is placed in the most persistently stimulating context of his entire recorded career. The charts are by Bill Holman who, among his many other big band credits, was one of the very few arrangers who could make Stan Kenton's orchestra perform more like a sports car than a cross-country moving van. Here, scoring for ten pieces, Holman has created exceptionally lithe, crisp and resilient settings that propel Zoot into brilliantly swinging, emotionally powerful and ceaselessly inventive improvisations on standards, originals and two triumphantly realized Ellington numbers ["Main Stem" and "I Got It Bad (And That Ain't Good")]. The set is about as flawless a jazz recording as has been released in a long time. And the recorded sound is equal to the music—very "live," open, and, as befits the protagonist, natural.

Meanwhile, of course, there are other musicians who are naturally experimental, who *need* to keep on stretching the limits of their ideas, their concepts of sound, and their very intruments. Three of them—vibist Bobby Naughton, trumpeter Leo Smith and clarinetist Perry Robinson—combine in *The Haunt* on Naughton's own Otic label. The result is, first of all, one of the most immediately accessible of all avant-garde jazz sets because it is pervasively lyrical. A continually floating, melodically curving series of conversations which subtly keep changing colors and timbral intensities while a mysterious implied pulse makes it all cohere in a rather dream-like way. But this is a most vivid dream, the kind that lingers in the mind long after the recording has ended.

Naughton, in my view, is the most creatively originist vibist now in jazz. Leo Smith (an alumnus of Chicago's Association for the Advancement of Creative Music) is perhaps the most thoughtful of the newest wave of trumpeters, and Perry Robinson is by far the leading post-modern-jazz clarinetist, having found quite new ways of sounding and talking on that instrument. The recording, by my taste, could have a closer and fuller presence, but I expect the intent was to create a certain distance, as in a dream. In any case, *The Haunt*, in its way, is as permanent and dateless an addition to jazz discography as is Zoot Sims' *Hawthorne Nights*.

ZOOT SIMS: *Hawthorne Nights.* [Norman Granz, producer; Grover Helsey, engineer; recorded at RCA Studios, Los Angeles, Ca.] Pablo 2310-783.

BOBBY NAUGHTON, LEO SMITH, PERRY ROBINSON: *The Haunt.* [Bobby Naughton, producer; Eddie Korvin, engineer; recorded at Blue Rock Studio, N.Y., N.Y.] Otic 1005. Otic Records, Southbury, Ct. 06488; also distributed through JCOA New Music Distribution Service, 6 W. 95th Street, N.Y., N.Y. 10025.

Performance: **Thrilling**
Recording: **Microscopically detailed**

These releases, each presumably the result of a different Philips production team recording the same orchestra in Amsterdam's famed Concertgebouw (literally, "concert hall"), are perhaps the finest-sounding discs I have yet heard from this source. But, damn it, why is Philips unwilling to give proper credit on the record jackets to their producers and engineers? The sound quality on these records is quite different, and I want to know who is responsible for such sonic splendor. I also would like the recording site and date to be listed, but since it took years for Philips to break down and provide timings on the record labels, I shouldn't be too greedy.

Many of the Philips Concertgebouw recordings of the last 15 years have sounded fuzzy—the aural equivalent of a grainy photograph. The Abravanel Grieg set reviewed above is distantly miked, but the sound is in focus. Actually, the Concertgebouw recordings were praised for the sound, especially in Europe where listeners appear to prefer a more conservative perspective. Reviewers asserted that the Philips recordings sounded just like the orchestra in performance. But what, then, do these recordings sound like? To my ears, the ambience on these three new recordings is even more beautifully captured, because instrumental sound is emanating from a *direct* perspective, rather than becoming cluttered with other resonances in the upper reaches of the hall.

Haitink prefers a blended, rather than differentiated orchestral sound. The recorded sound on his Bruckner

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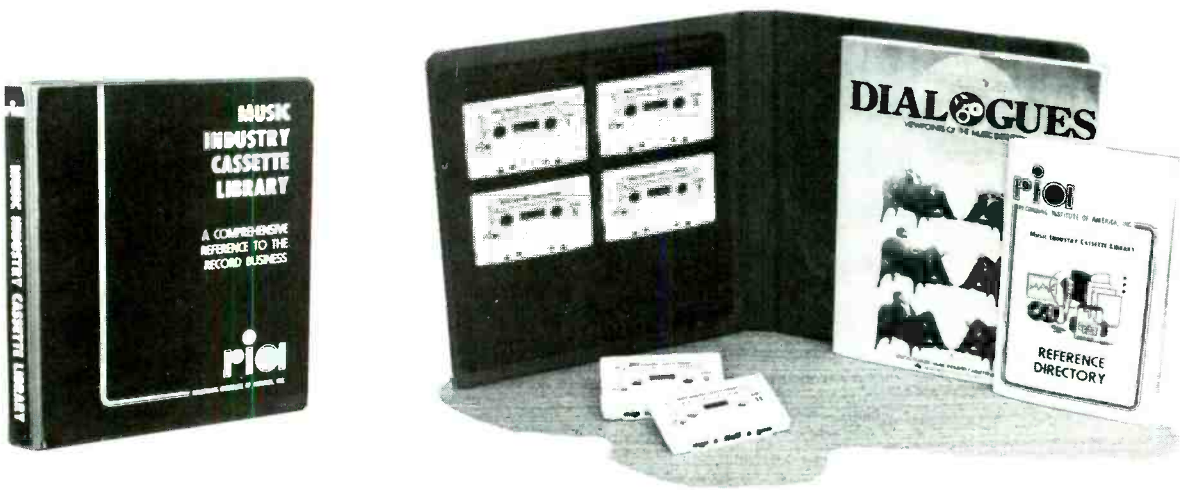

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Seventh and Ninth and Mahler's Third, Seventh and Ninth were commendable, as was his Strauss *Ein Heldenleben*. But compare even those fine recordings with the new *Das Lied*, which boasts crystal-clear instrumental lines, unobscured solo virtuosity, and yet sound true to Haitink's style. His personal producer, Volker Straus, has been miking closer and closer since he began recording Haitink and the Concertgebouw in 1973, and I feel that he has now found an ideal perspective.

The sheer sonic beauty of Mahler's score has never before been revealed on record with such glory. Haitink's control is second to none and the orchestral contribution is ravishing. I wish the vocalists were of the same exalted standard. In the tenor songs, James King sings with insight, but the strain in his voice is not pleasant, and Janet Baker, for all her intelligence and vocal beauty, fails to project the text with clarity. For the ultimate in Mahlerian *angst* the Bernstein recording on London remains my favorite, but Haitink is supreme for a more autumnal approach.

Colin Davis' *Dvorak Seventh*, the first in a cycle of the nine symphonies, is dramatic, tightly symphonic rather than broadly singing, and marvelously performed. Strings have gutsy presence and the woodwinds are a bit more closely balanced. The sound befits Davis' energetic manner. He wants more separation and solo brilliance of instrumental sound than Haitink, and Vittorio Negri's production is perfectly complementary.

Antal Dorati has always been a stickler for clarity of orchestral textures, and his new recording of Tchaikovsky's complete *Nutcracker* has been one of the most exciting reviewing sessions I have ever had. The sound is phenomenally detailed and clean, with the entire orchestra spreading out before you as if you were on the podium itself. Yet there is plenty of depth perspective and a wide dynamic range, unlike many recordings so closely miked.

The diffuse sound on many of Haitink's recordings of yore frequently made me wonder just how accomplished the various sections of the orchestra really were. The harps and percussion were only rarely audible in the past; the winds were frequently blanketed and without color; the strings, often matted and lacking either guts or sheen; and the brass, usually fine but occasionally was also too far in the background. All of these choirs distinguish themselves with continuously astonishing virtuousity.

Perhaps this is not the most "natural" sound on disc—I doubt if even the conductor could hear an orchestra with such startling immediacy—but it certainly is a monumental example of the recording art. These will be my number one demonstration discs from this time on. Now who-in-hell produced and engineered them, Philips??? S.C.

SHOWS and SOUNDTRACKS

RAKSIN: *David Raksin Conducts His Great Film Scores.* New Philharmonic Orchestra, David Raksin cond. [Charles Gerhardt, producer; Philip Wade, engineer.] RCA ARL 1-1490.

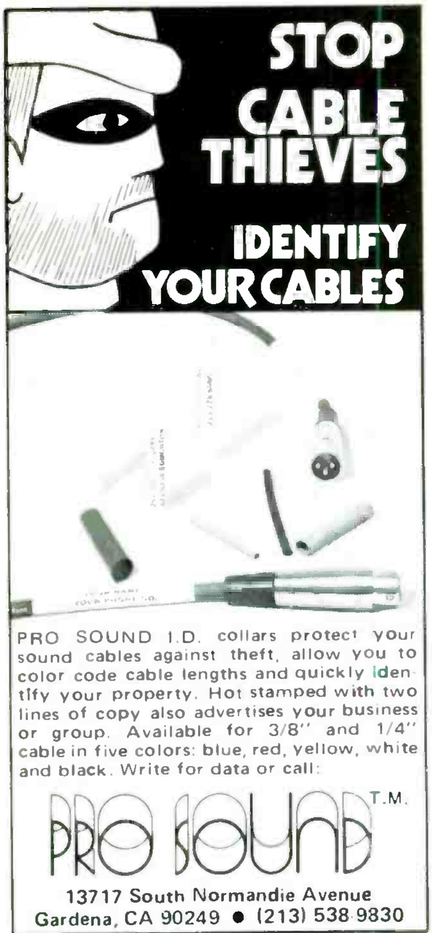
TIOMKIN: *The Classic Film Scores of Dimitri Tiomkin.* National Philharmonic Orchestra, Charles Gerhardt cond. [George Korngold, producer; K.E. Wilkinson, engineer.] RCA ARL 1-1669.

Performances: **Fine**
Recordings: **Excellent, as usual**

With two of the newer entries of RCA's "Classic Film Scores" series before me and ready for review, it seems like a good opportunity to talk about the series itself as well.

The two newer discs, *David Raksin Conducts His Great Film Scores*, and *Lost Horizon—The Classic Film Scores of Dimitri Tiomkin*, both maintain the general standards of excellence found in the other recordings of this set. All the discs are beautifully recorded with accomplished orchestras (usually England's National Philharmonic), and handsomely packaged with photographs from the films involved and well written informative notes written sometimes by the composer hinself (as in the case of David Raksin). Virtually all the records in this group are worthy additions for the film music enthusiast. Gerhardt and Korngold, whose brainchild this series seems to have been, deserve much credit for maintaining the integrity of these discs by consulting with the composers whenever possible, seeking out the best sources of the music, and for making this series an obvious labor of love.

However, there is one growing



CIRCLE 32 ON READER SERVICE CARD

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dissatisfaction with these recordings that I feel compelled to express. Who chooses the selections and why? Consistently, the choices of music on these discs is baffling. While this is less true of the Raksin album, still one has to wonder why the bulk of the disc—one whole side—is devoted to "Forever Amber." This being the least interesting of the three scores included. This trend can be found throughout the series. The Bernard Herrmann album entitled *Citizen Kane* contains roughly 13 and a half minutes of music from this score, which well deserves a complete recording. Meanwhile, almost 25 minutes are devoted to "White Witch Doctor" and "Beneath the Twelve Mile Reef," which are both among Herrmann's lesser efforts. Every film composer had to eventually write pap like this. However, must we buy it? The initial disc in this series, *The Music of Erich Korngold*, entitled "Sea Hawk," at least devotes a fair amount of the recording (seven minutes) to its title score, although "King's Row," generally considered Korngold's best, gets a scant minute and a half or so. Once in a while it might be explainable. Perhaps certain scores are longer than others, for instance, or availability of the music could be another factor. But consider the newest one, the Dimitri Tiomkin record. Tiomkin's three best known unavailable scores are "High Noon," "High and the Mighty," and "Old Man and the Sea." Is there a note of any of them on the record? Instead we get such nonentities as "Search for Paradise" (do you remember that one? I don't), "The Fourposter" (I didn't even remember there was music in this film!), and "The Big Sky." But the *piece de resistance* which fills a full side of this disc is the score for "Lost Horizon." I didn't remember how bad this music was from merely viewing the film, but here it is in all its awful glory—twenty-three minutes of slush. Tiomkin is not a favorite of mine, but surely what is popular and is of interest to listeners should be selected for an album.

I, for one, am tired of movie-score records on which less than a third of the music is worth hearing. Here's hoping this series, outstanding in so many ways, will be as concerned with *what* it records as *how* it records it in the future. H.R.

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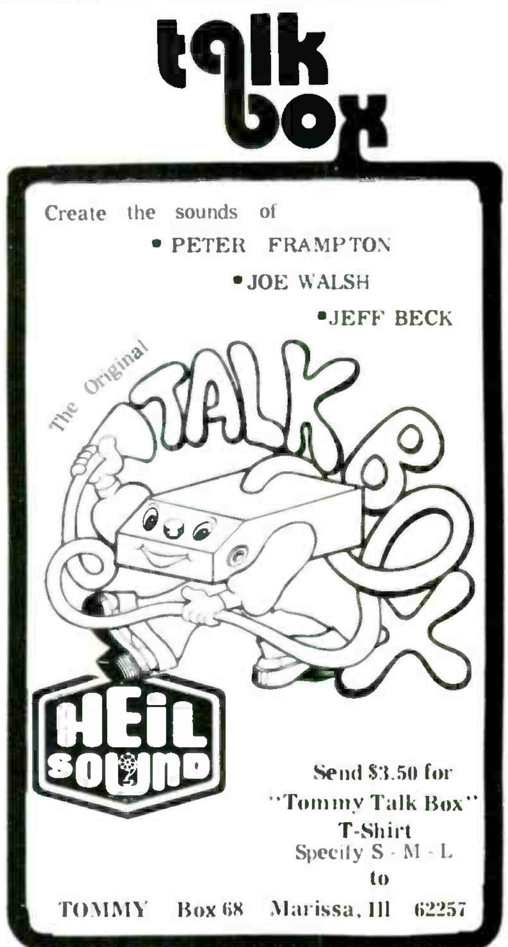

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