AUG/SEPT 1976

VOL. 1 NO. 6
$1.50

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

Recording Outdoors with Fleetwood Mac

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History of Recording, PART 6
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Lab Reports • New Products • Record Reviews

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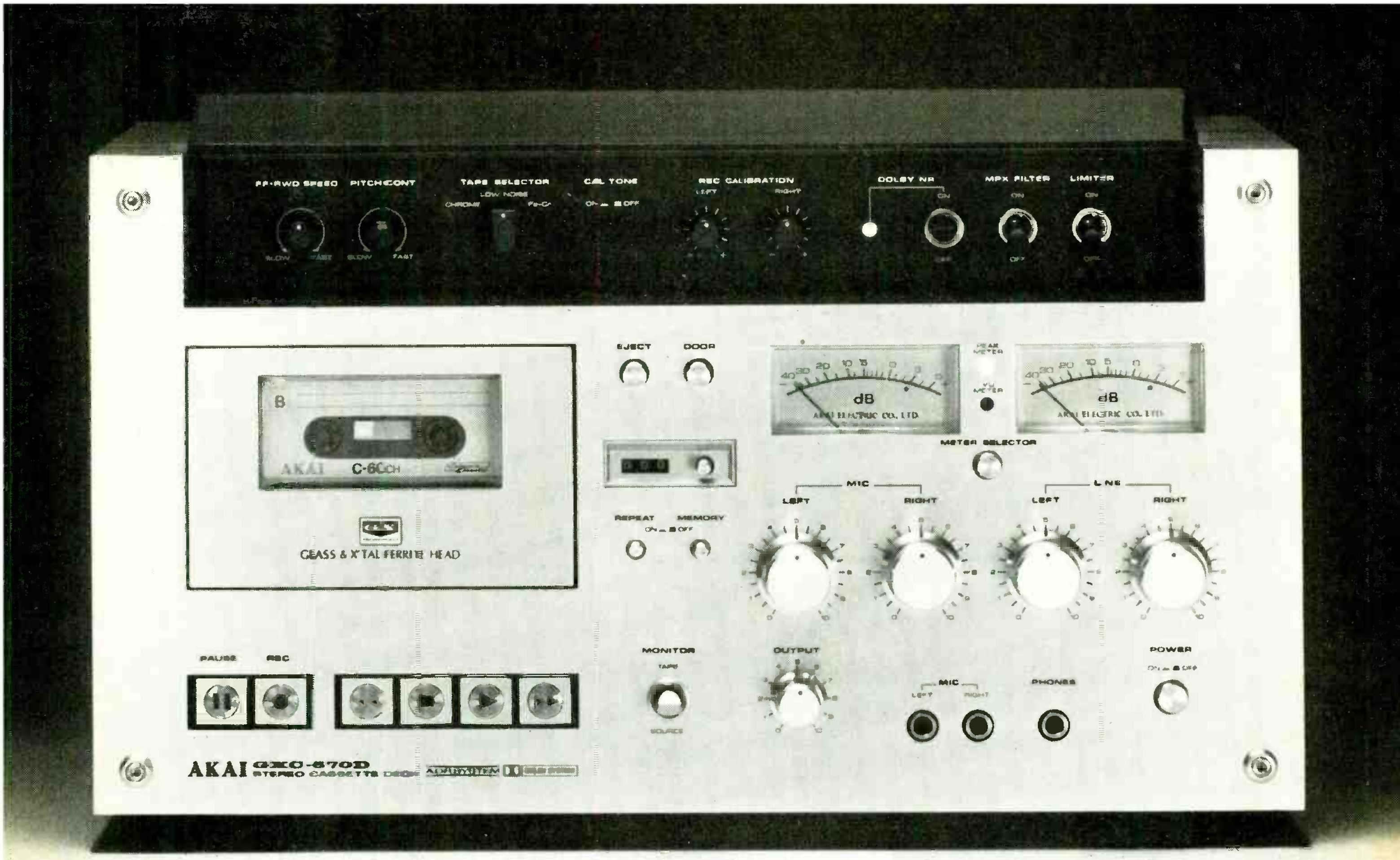
CIRCLE 94 ON READER SERVICE CARD



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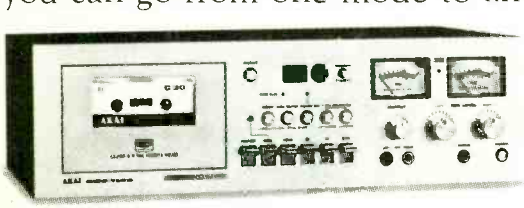

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Manufacturer	Brand	MAGAZINE A		MAGAZINE B	
		S/N Ratio Weighted in dB	Output @ 3% THD	S/N in dB (re: 3% THD)	THD at O dB (%)
TDK	SA	66.5	+4.2	66.0	0.9
AMPEX	20:20+	56.4	+1.9	—	—
FUJI	FX	60.0	+2.3	—	—
MAXELL	UD	—	—	58.5	1.1
MAXELL	UDXL	62.5	+2.7	—	—
NAKAMICHI	EX	60.0	+2.3	55.0	1.1
SCOTCH	CHROME	—	—	64.0	1.3
SCOTCH	CLASSIC	62.5	+2.0	—	—
SONY	FERRICHROME	64.0	+2.1	64.0	1.8

Decks used for tests: Magazine A-Pioneer CT-F9191 (cross-checked on DUAL 901, TEAC 450); Magazine B-NAKAMICHI 1000.

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AUG/SEPT 1976
VOL. 1 NO.6

MODERN RECORDING

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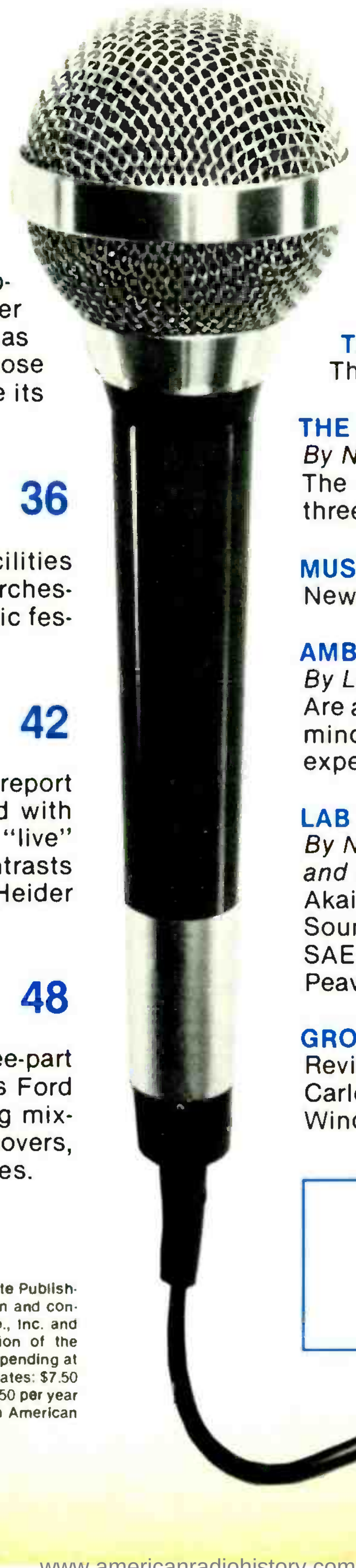
THE FEATURES



Cover Photo by R. A. Neilson

THE STAPLES

COMING NEXT ISSUE!
A Session with Judy Collins
The Future of the Videodisc
P.A. Primer, Part 3

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READER'S FORUM

Quality Control: It's Just a Slide Away

I would like to suggest that all "live" mixing engineers be required to take a course of instruction in amplifier, speaker and ear performance. Unfortunately, I have been to one too many "live" performances where the power amplifiers and speakers have been driven beyond the manufacturer's intended performance capabilities. The result: an ear(s) ache and disappointment in the performance/entertainment. For example, I attended a Nils Lofgren/Hall & Oats concert several weeks ago where my head and ears were brutalized to the point where I had to actually hold my fingers in my ears because of the high level of distortion—and that's distortion with a capital D, caused by over-driven amps. Honestly gentlemen, I get off like crazy on a full-sounding level, but there's nothing worse than clipped wave forms at 100 dB-120 dB. It's pointless. It makes the performer look bad (Nils actually got booed—not because of his playing, but because of his P.A. levels), it makes the equipment look bad (a Phase Linear isn't all *that* bad if you keep it below rated output), it makes the mixing engineers look bad (but not bad enough), it shortens the life of the equipment, and it's a burn for the spectators.

With the quality of equipment available today, it's possible to obtain full-sounding audio levels without the distortion and resulting pain often encountered in many "live" rock performances. I'm not deaf (yet), and I doubt that most concertgoers are. I'd much rather listen at a slightly lower level and hear more music than distortion at a higher level.

I realize that Nils is a rowdy act, but I like his guitar work, and I really get off on the "live" sound of a concert hall. Nils's musicianship got butchered several weeks ago by a heavy-handed mixing technician—[Nils and the group] were unlistenable. Too bad for Nils, too bad for those of us who paid to see it.

Using McIntosh electronics and JBL loudspeakers on a daily basis, I have become accustomed to a certain level of sound quality which I believe is possible to reproduce in a "live" situation. In most cases, it's simply a matter of properly controlled levels, compatible with the equipment's capabilities. A large amplifier when driven to clipping can deliver large amounts of power, but probably with about 40% or more THD. The extra energy content of the clipped signal will damage most loudspeakers and most human ears. Long live rock, but let's let distortion die!

—Michael Nathans
Audio Consultant
Glick's Audio, Inc.
Lancaster, Pa.

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CIRCLE 19 ON READER SERVICE CARD

Letters to the Editor

An Aware Soundman

I am an amateur soundman for a local band in my area. We don't do Disco or Copy music. We write our own and I believe that if we do make it, it will be a bizarre show. About two or three months ago we started getting into multi-media performances. So, when I read the article on the Tubes, I was totally amazed. I didn't believe a band like this could be accepted in a 75% Disco society. The cover story in *Modern Recording* was fantastic! My compliments to H.G. La Torre. His interview with Don Wood was phenomenal. So when I was finished with the Tubes article, I decided to see this "multi-media" band. Fortunately, they were at the Tower in Philadelphia on May 10. I went and, to my regrets, Mr. Wood wasn't with them on this tour.

The band was great. I had never heard of their albums or seen them before, so this was an experience for all. The act was very much together. And the P.A. *system sounded* good, but the engineer didn't have the slightest idea what he was doing, in my opinion. When a band member went for his or her mic, it wasn't functional; it was as if a fader wasn't up for that person at the mixing console. Or when the harmonica break came in, it wasn't there. Only people five feet away could possibly hear it. Not mentioning that the clavinet was only heard once that night when the keyboard player played it through most of the concert.

I look at the Tubes and then I look at my fast-growing multi-media band and wonder if I could have done a better job than what Mr. X did at the controls that night.

So this is an open letter to Don Wood (wherever he may be). I don't know if there were inner hassles with you and the band [if youwere just] offered a better opportunity somewhere else, but a band depends on a good, well-built, sound system and a very aware soundman. As we all know, the sound system could make or break the band and I would really like to see Disco-madness drop out and the Tubes media come in. So I feel that Don Wood is that aware soundman for the Tubes.

—Eugene P. Ramble
York, Pa.

A Drastic Solution

In *Modern Recording*, Apr/May 1976 issue, on page 9, Mike Purrel inquires about unmatched speakers. Mr. Jack Fassel replies that ideally you should get four matched speakers . . . quite a drastic solution when a simple $3.95 stereo L-Pad would do the trick.

—Dave Nulman
Brooklyn, N.Y.

Song Genesis

I am interested in articles about producing and mixing sessions. Also, I'd really get off on reading about the development of a song from writing through recording, mixing and placement on an album. Maybe you can catch one of those songs that are created in a recording session.

—Jim Fox
Beaver Falls, Pa.

Video Disc Scrutiny

The magazine so far is quite good. I hope you plan on getting into the video disc some. This might prove to [be] a popular affair. Also some information on the operations of the recording business, i.e., fees to producers, etc., contracts, and so forth. Nothing in close scrutiny of this, but an article or two of enlightenment.

Also, I have two favorites for your artist selection: Frank Zappa & the Mothers; Little Feat.

—Jeffrey Kanter
Syracuse, N.Y.

Women & Engineering

Here's something that might be of interest to you. It's something that at first I thought I was probably imagining, but every day becomes more evident . . . that is, the lack of women interested in practically any facet of stereo knowledge—much less recording.

I find that I can rap for hours with people at work—providing they are male—about equipment, developments in engineering—practically anything, and be fully entertained—and yet, if I mention the subject to any of my women friends—I either draw a complete blank or a yawn.

Maybe it's only the Detroit area, but I think it's a shame that such an interesting field should be overlooked by most women—if it's not in *Glamour* or *Seventeen* it's just not important to them.

—Helen Michals
Detroit, Mich.

For those MR readers who missed it, a similar concern was voiced in the second issue (Dec/Jan 1976) by a woman recording enthusiast. Utilizing MR's "Reader's Forum" column—where any reader may submit an editorial on any subject in the recording field—she asked why the recording industry was ignoring women.

Autocorrelating RFI

In regard to Bill Concevitch's letter in Talkback, last issue [Apr/May 1976], I thought I'd give another opinion to question three in that letter.

I'll concede that the radio interference, once recorded, is on the tape to stay. The clicks are another matter, however. We've had some very good results by processing just such a marred tape through a Phase Linear Autocorrelator. The result is actually quite surprising, and if the tape can't be re-recorded as such, a try with the aforementioned device is certainly worthwhile.

—M.C. Rose
Part I of II Studios
No. Dartmouth, Mass.

Open-reel Edit Cuing

In reference to Len Feldman's Ambient Sound article in Vol. 1 No. 4, *MR*, there are two top-quality reel-to-reel machines in addition to the Otari MX5050 that offer edit cuing. These machines are the Tandberg models 9200XD and 10XD.

—Michael G. Nathans
Glick's Audio, Inc.
Lancaster, Penn.

The Tandberg 10XD was lab-tested by Messrs. Feldman and Eisenberg in MR, *Vol. 1 No. 5 (June/July 1976).*

More Technical Slant

We really enjoy your magazine a great deal. Mostly, we enjoy your technical articles about recording techniques and equipment. Personally, I would like to see you go into more technical detail for people like us at Pyramid Sound who want to build mobile and stationary studios and our own console, etc. What about doing a primer on "live" and studio recording techniques and how about schematic or at least block diagrams on the how's of recording, i.e., mic placement, console, mixing, hi vs. lo impedance mics, etc.?

—Kenneth F. Sauer
Pyramid Sound
LaSalle, Colo.

P.S.—In answer to the person who criticized your including a record review section, I would point out that your reviews are not only more technically oriented than those found in other magazines, but also seem to exhibit a greater perception and understanding of the musicians and their performances. I like it.

Record Reviews Revisited

In reference to a comment by one of your readers—in last month's issue [Apr/May 1976]—concerning the pages wasted on record reviews, I would have to agree—but what do I know? Do think one "nice," well-written and informative review on such subjects as production, miking, mix, etc. of an *outstanding* lp. If one is to take the time to sit and read something, why not make it time well spent?

Seems clear to me .... Like your mag very much!!

—Wally Watson
Tampa, Fla.

I enjoyed the profile of Andrew Kazdin in your Apr/May issue very much. In response to the question of [whether or not] *MR* should review records, I don't really care; but I do think Bruce Mallion's letter about *MR* doing stories on disc mastering, quality pressing, disc technology and basic rules for easily transferring a master tape to a disc was right on the money.

—Armand Caputi
San Francisco, Cal.

Correction

The distributors for the Furman Sound PQ-3 parametric equalizer, featured in MR's Product Scene in the June/July issue, have advised us that the price is \$300, not \$250.

Also, the photographs accompanying the Beach Boys article on pp. 46 (bottom photo), 48 and 50 should have been credited to Gary Nichamin. Our apologies.—Ed.

TALK BACK

Mobile Unit Info

I am trying to put together a mobile unit and would like some information on top-of-the-line portable equipment. I would especially like your opinion on the Stramp 16-input mixer and whether most two-track studio machines can be made portable, such as the Ampex or Scully.

If you could also inform me on where I could purchase such state-of-the-art equipment at reasonable prices, I would greatly appreciate it.

Lately I've noticed that "live" albums, per se, are becoming quite popular and I would appreciate your telling me what kind of equipment is being used as far as two- and four-track machines and mixers go.

I am also aware that many of these new albums are being recorded 16-track "live," and remixed, but that's too much of an expense for me to undertake.

Thank you for your time.

—Frank Laino Jr.
Feasterville, Pa.

I am not familiar with the Stramp 16-input mixer to which you referred. However, I am familiar with and would recommend the State of the Art boards both for their dependability and versatility in mobile recording; Yamaha, Quantam, Tascam, also; the Scully, Ampex, and MCI tape machines are portable and dependable in remote situations. This equipment is very popular in most professional mobile recording control rooms.

Viking Studios Ltd. is located in Denver and we are able to purchase our equipment through Colorado Nashville, in Colorado Springs, Colorado. You may find a professional audio dealer in your area by checking your local directory.

In considering the larger multi-track mobile control rooms, the Audiotronics 27-input board and Ampex machine are the most popular. But it should be pointed out that most "live" albums today are recorded in either 8-, 16- or 24-track modes. Also, for very important projects, second back-up machines will roll tape along with the number one machine as a safety. The boards and machines that I mentioned earlier in this reply are widely used in smaller two- and four-track remote sessions.

—Wade R. Williams
Chief Engineer/President
Viking Studios Ltd.
Denver, Colo.

Studio Reverbs

I think your magazine is fantastic! It's what I've been looking for. I do have a question to which I would very much appreciate an answer: Is there another kind of reverb other than the spring kind? What kind is used in today's studios?

—Mike Perry
Cincinnati, Ohio

Generally, the spring units that you've mentioned are the easiest to obtain and the least expensive. Although some very good chambers such as the AKG series used springs in their systems, some special circuitry is used to get out the "springy-boingy" sound. And, motional feedback is used to control reverb-length times.

The next type is the acoustic chamber, consisting of a large room, sometimes built with wet plaster and uneven non-parallel walls. The walls are covered with many coats of shellac or epoxy enamel to make it acoustically reflective. Installed at the far end of the room are selected speakers and microphones. This gives the most natural echo effect that I know of.

The third type is a compromise on room vs. quality. It is a suspended, hardened steel plate, approximately 3/16″ thick and four by eight feet, on which a send and return transducer is installed. This is known as an EMT chamber and is very natural-sounding without taking up a lot of room. With the proper tension on the plate and careful adjustment of the "send" and "return" levels as well as equalization in the system, it will, in many instances, approach the sound of a very good acoustic "room" chamber.

—Ken Sands
United Sound Systems, Inc.
Detroit, Mich.

Should a Noise Reduction Circuit Be Monitored?

When using a noise-reduction unit, such as a Dolby or dbx unit, what are the advantages of using additional circuits to monitor the material as one is recording? What are the disadvantages of not monitoring the material through the noise-reduction circuits? For example, when using a dbx 154 (containing four dbx processing circuits), would one achieve as high a quality of recording by using the four circuits to recording in 4-track quad than he/she would if they recorded in 2-track stereo, using two circuits to record and the additional two for monitoring? Also, how does the Phase Linear 1000 Noise Reduction unit com-

pare to a dbx unit in the amount of noise which is eliminated?

—Bill Concevitch
Bethlehem, Pa.

Dbx and Dolby "A" or "B" are compander-type noise-reduction systems. Noise reduction is achieved by compressing the signal before recording and expanding the signal during playback.

A switchable system has a processor for each channel that is switched to either record or play. While you are recording, the noise-reduction unit must remain in record (compress). If the noise-reduction unit were switched to play during recording, the signal sent to the recorder would not be compressed, thereby defeating the noise-reduction process.

A simultaneous system has two processors per channel, one for record (compress) and one for play (expand). This prevents switching each channel from record to play, as on a switchable noise-reduction system. A simultaneous system also allows monitoring of tape while recording.

The question asks if monitoring of tape is necessary when using compression/expansion noise reduction. The advantage is that you can listen to the expanded signal at any time to insure that everything is recording properly and that the same sound you are monitoring is in fact what you are getting back off tape. Is it absolutely necessary? The answer is no. But with a switchable noise-reduction system, if off-tape monitoring is done you will hear the compressed signal. If there are any subtle problems with what you are recording, you will more than likely not hear them when monitoring the compressed signal. The simultaneous noise-reduction system is advisable for more flexibility and insurance. You can know exactly what is on tape at any time you wish.

The Phase Linear Auto-correlator is not a compression/expansion noise-reduction system. The Auto-correlator is an "after the fact" noise-reducer that breaks the frequency spectrum into a number of bands that open and close according to the signal-level present. Systems of this type achieve noise reduction by subtraction of frequency response, which often causes the loss of harmonics and overtones.

—Larry Blakely
dbx, Inc.
Waltham, Mass.

A Compression Precis

I'm a subscriber to your mag and think it's great. With the information you provide, you surely fill a void for people like myself who are interested in recording "... as self-expression—as a creative act" in your words. I would like to ask some questions for your "Talkback" column:

(1) What are some situations where the use of a compressor and/or limiter might be advisable?

(2) When using EQ, are there any guidelines as to what specific frequencies, when boosted or cut, will produce a certain result?

(3) How can EQ be used to make instruments blend better and prevent them from masking each other?

—Michael Loomis
Lebanon, Ohio

(1) Limiters or compressors are devices which are normally used to "limit" the gain of a signal. In today's recording, there exists a limited dynamic range which can be captured on tape and which might be defined as the "space" between the loudest level, where a pre-determined percentage of distortion begins to affect the sound, and the softest level—which is the point at which the "noise floor" (thermal noise) becomes apparent or in any way annoying. This limited range is around 64-66 dB of one track of a properly aligned NAB tape recorder with no noise reduction. The dynamic range of many sounds is far greater than the tape's limited range. The need to contain the sounds within this limited range was helped by the invention of the limiter, which reduced the level of the signal applied to it. Nowadays, the limiter has surpassed its original gain-limiting function to take its place among the many electronic "tools" in the studio.

Limiters or compressors act on sounds by reducing the level of the signal applied to the input in a ratio of anywhere from 1:1 to infinity:1. In simple terms this means that a 20:1 ratio will allow an increase of only 1 dB for every 20 dB applied above the pre-selected threshold, whereas a 4:1 ratio would allow an increase of 1 dB for every 4 dB applied. Compressors operate on lower ratios than limiters, and are useful where there is not as great a need for level control, where the sound generated does not have very large peaks or sudden changes in level.

The best kind of limiter, in my opinion, is one which has a variable ratio, from 1:1 to infinity:1, such as the dbx limiter. This allows the user most flexibility, and "flexibility" is one of the key words in sound recording. Since any kind of compression or limiting affects the sound to some extent, care must be taken in choosing the best ratio and amount of limiting needed for a given sound. Higher ratios of limiting are needed on instruments such as vibes or human voices which have very large sudden changes in level. Certain sounds have very "fast" peaks or transients and may require a different kind of limiting for best results. Generally limiters work on the RMS value of a signal, which often is an average level rather than an absolute, true "peak." This type of limiter would allow certain very fast transients to "escape" and not be limited. The best limiter for those peaks is a peak-action device such as the Allison Gain-Brain. This device also allows one to use a combination of peak and RMS limiting, by setting the threshold for the peak-limiting action several dB above the RMS threshold. This dual limiting can be an ideal solution to certain sounds, among them the human voice.

Uses for limiters:

(a) The main use is to reduce the dynamic range of a sound to "fit" the tape's dynamic range. Reducing the very loudest part of the sound allows a louder "average" level on the tape since one can ride the overall level up higher without fear of distortion. This technique, commonly used in almost every studio, gives better signal-to-noise ratios on the tracks, as well as easing the engineer's task in the mix.

(b) To *change* the sound of an instrument. Many limiters have slow release times, meaning they "hold" the limiting action for a period of time beyond which the loudest sound has been made. This feature can be used to advantage on piano, for instance, to produce a "breathy" singing sound by limiting the high-end mic and making the decay of the strings on the piano hold out longer and louder, resulting in what some people might call a "grandiose" sound. Others may hate it, since it is unnatural. But we are not dealing with truly natural sound in pop recording, but rather the creation of whatever illusion suits us at a given time.

(c) Inversion of sound envelopes is a relatively new technique. Certain limiters, such as the Eventide Omnipressor, allow one to take a sound such as a cymbal crash and "invert" it by using severe limiting combined with a very fast attack and slow release time. The fast attack "grabs" the signal and, with severe limiting set, pushes the attack down to a low level, then gradually releases it, causing the impression of the end of the sound growing gradually louder. The Allison Gain-Brain can also achieve this effect.

(d) Limiting an echo-chamber return can cause the decay time to change, as well as the entire echoed sound. Limiting can be used to intentionally "thin" out a sound, since limiting has a tendency to reduce low-frequency information to the extent of the degree of limiting.

(e) Frequency-dependent limiting has been used for years, under the name of "de-essing," in which certain high frequencies only are limited, leaving lower ones untouched. This idea has been developed recently, and in the near future dbx will present a limiter with an adjustable high- and low-frequency filter incorporated in the unit, allowing the user to select any one bandwidth and limit it, leaving the other frequencies untouched.

Since space does not allow more detail, I advise experimentation with a good flexible limiter to learn first-hand all the changes and effects which may be produced.

(2) I feel that there are few guidelines as to specific frequencies to use when equalizing sounds. Any guidelines that are useful would be in frequency "ranges." There are so many types of equalizers on the market, and each can add a different "sound" due to the variables of bandwidth, curves, slopes and level, that all I can advise again is experimentation. Basic guidelines I can offer would be rather obvious. For example, if one wants a "brighter" sound, one should make changes in the midrange and high end—say from five-thousand cycles up—boosting those frequencies. One can also remove some lower frequencies which may be too loud, thus "masking" the higher frequencies.

Rather than go into detail on this subject, which would really require a lot of space, I advise experimentation with a very flexible equalizer, such as the Orban/Parasound Parametric, which has four separate overlapping frequency ranges, a lot of boost or cut level on each frequency area, and continuously variable bandwidth controls. This equalizer is an important tool in any studio as well as being a very thorough, flexible teaching device for beginners.

Remember that a well-balanced monitoring environment is a must for proper use of equalization. If the monitor sounds favor any frequency areas, your EQ will not be consistent when the tape is played on other monitor systems, which, if properly balanced, will tend to defeat or change your equalization. Remember also that we can only EQ sounds for the "average" listening environment, since many home systems have deficiencies in their monitor environment.

(3) In simple terms, instruments can be made to blend together by removing harsh peaks in certain frequencies, or reducing them, or accenting weak areas. This work is done by the equalizer. For example, if two trumpeters are playing the same notes together, but each has a different sound on his instrument, there may be a blend problem. If one man's sound is more piercing, thin or strident, while the other's is fuller, warmer and more rounded, the producer or engineer may find the combination displeasing—possibly because the more strident horn tends to mask the warmer sound of the other horn. Reducing the harsher horn level is not the answer, since then the effect of the two horns is lessened. Therefore, the engineer may attempt to remove the harshness by equalization. In this case, he may remove some high frequencies—say around 7000 cycles—and add some low frequency—say in the 400-1000 cycle area. Thus, the harsher horn will become fuller, richer and less strident, and the two horns will blend better.

There is no formula for making sounds blend. Again, experience and use of flexible equalizers is the answer.

—George Klabin
Sound Ideas Studios
New York, N.Y.

Dokorder Discovery & Pseudo-stereo Tricks

(1) Having an 8140 Dokorder four-track deck, I've been messing around with different dubbing and mixing techniques. I've found I can record on three tracks; then, with the help of a few patch cords, I can transfer those three to the remaining one, after which I have three more tracks to reord other voices or instrumentation. And I can do that until I have accumulated up to 12 tracks. Why didn't they put that in the manual? Will it harm the recorder? (Note: When I transfer, I don't use the sync

THE TAPE THAT'S TOO GOOD FOR MOST EQUIPMENT.

Maxell tapes are not cheap.

In fact, a single reel of our most expensive tape costs more than many inexpensive tape recorders.

Our tape is expensive because it's designed specifically to get the most out of good high fidelity components. And unfortunately, there's not much to get out of most inexpensive tape recorders.

So it makes no sense to invest in Maxell unless you have equipment that can put it to good use.

No other tape starts off by cleaning off your tape recorder.

No other tape sounds as good as ours because no other tape is made as carefully as ours.

For example, every batch of magnetic oxide we use gets run through an electron microscope before we use it. This reveals the exact size and proportions of individual particles of oxide. Because if they're not perfect, the tape won't sound perfect.

Every employee, vacuumed.

And since even a little speck of dust can put a dropout in tape, no one gets into our manufacturing area until he's been washed, dressed in a special dust-free uniform and vacuumed.

(Yes, vacuumed.)

Unlike most tapemakers, we don't test our tape every now and then. We test every inch of every Maxell tape.

Which is why every Maxell tape you buy sounds exactly the same. From end to end. Tape to tape. Year to year. Wherever you buy it.

We clean off the crud other tapes leave behind.

After all the work we put into our tape, we're not about to let it go to waste on a dirty tape recorder head. So we put non-abrasive head cleaning leader on all our cassettes and reel-to-reel tapes. Which is something no other tape company bothers to do.

Our cassettes are put together as carefully as our tape.

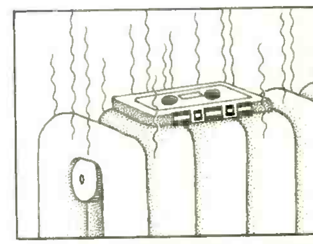
Other companies are willing to use wax paper and plastic rollers in their cassettes. We're not. We use carbon-impregnated material. And Delrin rollers. Because nothing sticks to them.

A lot of companies weld their cassettes together. We use screws. Screws are more expensive. But they also make for a stronger cassette.

Our tape comes with a better guarantee than your tape recorder.

Nothing is guaranteed to last forever. Nothing we know of, except our tape.

So our guarantee is simplicity itself: anytime you ever have a problem with any Maxell cassette, 8-track or reel-to-reel tape, you can send it back and get a new one.



Our guarantee even covers acts of negligence.

Give our tape a fair hearing.

You can hear just how good Maxell tape sounds at your nearby audio dealer.

(Chances are, it's what he uses to demonstrate his best tape decks.)

You may be surprised to hear how much more music good equipment can produce when it's equipped with good tape.

Maxell Corporation of America, 130 West Commercial Ave., Moonachie, New Jersey 07074

switches, but when recording on other tracks [after mix-down to one channel] I use the sync.)

(2) Also, what is the technique of recording a single voice or instrument, and splitting the signal into two separate channels on the board, but applying more bass and less treble on one channel, and less bass and more treble on the other? It really makes a dynamic sound when the two channels are played back in stereo.

Are there anymore little tricks like that that us amateurs could use on our small-to-medium-sized decks and mixers?

You've got a great magazine. Keep the presses rollin'.

—George R. Sutton
Lithonia, Ga.

(1) The method which you describe is basically quite acceptable, and in fact is an intended application for the 8140, albeit not mentioned in the product's manual. This function is often referred to as "ping-pong," since it involves bouncing the recordings around from track to track. But there are a few things to be considered in the interest of fidelity:

First, there's the problem of generation loss, which degrades response, increases noise and distortion. This is inevitable anytime a signal is processed through any circuit. When you re-record a signal as many as 12 times, generation losses can effectively multiply the little quirks present in all equipment to the point where it becomes distressingly audible. To keep this to a minimum, you should:

(a) Keep the heads and tape path clean and demagnetized.

(b) Pay particular attention to your recording levels with each transfer.

(c) Carefully select the programs to be ping-ponged relative to their musical significance; that is, record first, if possible, material that is relatively superfluous or background, that will be masked by later harmony and lead material.

Also, remember that a bounced set of tracks will be a mono mix. Don't ping-pong material that you'll wish had separation or pan-positioning later.

All these things taken into consideration, a home four-channel machine can often substitute as a "studio sixteen."

The final caution involves the method of patching. While ping-ponging is usually done with the assistance of a mixer, patch cords can be juggled around to do the job. But you shouldn't parallel outputs directly together because their low impedances will load each other, causing distortion (transformerless low-impedance outputs are usually meant to "look into" high-impedance inputs). The solution is simple, however. Connect a 5600-ohm "build-out" resistor in series with the "hot" (center) conductor of each patch cord that is to be paralleled. If you haven't yet done this, don't worry; you haven't harmed the recorder. If you don't feel at home with resistors and soldering irons, try your local service technician. He can assemble the "mixing patch" and whatever he charges ought to be a very small fraction of the cost of even the most austere mixer. If you're up to getting electrons all over your hands, however, but you need more details, write us a note and we'll send you a photocopy of some diagrams from another manual.

Which brings us to the embarrassing question as to why this data was not included with the 8140. This information was left out of the manual through the kind of simple oversight that frustrates managements, and delights competitors (who fortunately are not immune to the same thing).

—Arne Berg
Dokorder, Inc.
Lawndale, Cal.

(2) I am not really sure of an exact definition of the technique you mention, if there is one at all, but I suppose you could call it channel- or input-multing. The procedure you mention is usually used during the mixdown of a tape when there are more console inputs than tape channels. You can create pseudo-stereo effect by using multiple inputs of the same signal, using equalization to emphasize different frequency ranges in each input; through the use of stereo panning, the mono signal will appear from more than one point in the stereo spectrum. The use of different amounts of the same effect, such as reverb, echo, etc. or different effects on each input channel will also tend to increase the apparent "size" in stereo of the original mono tape channel.

In track-limited multi-channel recording, it is sometimes necessary to record different instruments on the same channel at different times during a song. Once again, the use of multiple inputs of one tape channel allows you to pre-set different variables such as level, stereo position, equalization, and effects for each instrument. The total treatment for each instrument can then be achieved by turning on and off the proper input channels preset for that particular sound.

—Bob Stoughton
Recording & Maintenance Engineer
Intermedia Sound
Boston, Mass.

Magnetic Fields and Remote Recording Trucks

With all the new mobile studios coming out, I am led to wonder whether they encounter problems with magnetism? For this reason I address this letter to you, hoping that you may answer my question.

Question—I am interested in knowing whether there are any magnetic fields set off in a remote truck due to magnetized parts that turn?

Can any of these magnetic fields (if any) be measurably strong to the point of affecting tape or any other instruments inside the truck as it is being driven (rotating parts may set off magnetic flux)?

—Jesus Alfredo Rivas
Arlington, Va.

I assume you refer to parts of the truck's drive-train, which rotate while the truck is moving. There are three reasons why any magnetized part of the truck doesn't affect the equipment. First, the distance from the drive-train to the tape machines is usually more than five feet, meaning that only an extremely high magnetic field could penetrate such a distance to the electronic equipment. Trucks just don't have high magnetic fields. Second, all parts of a recorder that contact are either made of non-magnetic material or are magnetically shielded. Third, other parts are either non-magnetic or, such as transformers, do not retain any magnetization. Of course, when the electronics are operating at a recording site, any AC magnetic fields will induce currents in transformers and heads, which is the reason for shielding all parts (such as input transformers) which operate at low levels.

—Jeff Eustis
Fred Ehrhardt
Fedco Audio Labs
Providence, R.I.

Record Defect Origins

I recently purchased two copies of the same album and found both to contain a scratchy defect. I returned them and was given two others; again, both had a very annoying surface noise in certain sections.

What causes this? Is it the recording studio's fault? What is the solution other than playing every album the record shop has before taking it home?

—Marilyn Demara
Amherst, Mass.

The answer to your problem, I can almost definitely say, is not the recording studio. It sounds like one of three things:

First, in some cases, the mastering lab (the place that does the original master tape-to-disc transfer) can put too high a level on the disc. At times this could cause the defect you state. But before this disc is cut, a reference disc is usually made, using the same specs that the master disc is going to have. This disc is played and checked for this type of defect. If the disc is good the master record is then cut. If distortion is heard, the problem is corrected and then the master record is made. If the record is made on a tight budget, the step of making and checking the reference disc is omitted. In this case this trouble may happen, but it's not very common.

Second, and most likely, the trouble could be the pressing plant. Since you didn't state which record or record company you are talking about, it's hard to know which plant they use to press their records, but some plants are notorious for using a cheap grade of vinyl. This type of vinyl is very rough and noisy and non-homogeneous, but because of its relative cheapness in cost, some pressing plants and record companies will go for this to save some money. Their thinking is that 90% of the listening public wouldn't care about or ever hear the distortion.

The third cause may be your own equipment. You didn't tell me what type of equipment you are using—more importantly, what kind of turntable and cartridge you are using. If your turntable is mistracking, or if your cartridge is worn, not all but only some of your records will exhibit this kind of noise. It depends on the type of music and the level on the record. The better the cartridge you use, the more resistant it will be to these defects and the better the sound will be even with these defects.

As you can see, any of these things, or any combination of these things, can cause the problem you state. Unless the record is really in bad shape, playing the album at the shop might not prove anything. If the shop uses a very good cartridge, and a well-tracking turntable, the defect still might only show up when you get the record home. Since all of the records released in a given geographical area are usually pressed from the same master record or at the same pressing plant, any defect will appear on all the discs, so buying a second or third record of the same title will not usually yield any better results.

—Stewart Jay Romain
Chief Engineer
Frankford/Wayne Mastering Labs
New York, N.Y.

Audio Workshop Queries

My question comes as a result of an effort in organizing an audio workshop. I build as much of my own equipment as possible or try to modify existing equipment to suit my tastes.

Advise me on two separate areas:

I. (a) What are the reasons and differences in making low-impedance and high-impedance microphones?

(b) In what situations are the different types of microphones applied: (1) from the recorder; (2) in a P.A. unit?

(c) When I use a small mixer(s) powered by a ±9v and ±12v, do I change the original impedance?

(d) What is the scale in ohms for determining high and low impedances (at what value do these terms change)?

II. (a) When I use two or more speakers within a cabinet, do I wire in series or parallel?

(b) How do I determine the number of speakers to carry the maximum load with minimum distortion?

(c) Describe and explain the different functions of a speaker pertaining to: (1) size in inches, such as 6″ vs. 18″; (2) design differences in magnets and cast of frame; (3) cone design differences in paper gauge, rings in the paper and the effect on the depth of the cone.

—R.O. Hornak
Weirton, W. Va.

I. (a) High-impedance microphones are designed for use within 6 to 10 feet from the tape recorder or mixer. They are usually of the crystal-type diaphragm and are very inexpensive to make. The frequency response is usually only good to 6,000 Hz and they

CIRCLE 50 ON READER SERVICE CARD

usually come with most low-cost consumer tape recorders.

On the other hand, low-impedance microphones are designed for use up to 600 feet from the tape recorder or mixer with little or no loss in frequency response. They are usually two to ten times more expensive than high-impedance mics and usually don't come with most tape recorders. Low-impedance mics also have a much better frequency response than high-impedance mics, and are usually used for recording music.

(b) The application on the different mics depend on the input impedance of the mixer, tape machine, or P.A. However, you can use low-impedance microphones with high-impedance inputs if you use a matching transformer within six to ten feet from the mixer or tape machine. In this way, you get the frequency response and the use of a long cable length of a low-impedance mic, but you use it in a high-impedance input.

(c) You do not change the original impedance.

(d) Low-impedance mics usually have impedances of 50 to 600 ohms; high-impedance mics range from 15K to 50K ohms.

II. (a) When hooking up two or more speakers in a cabinet, you should wire them in parallel. However, it is best to use a crossover which prevents wide overlapping of the frequencies between the two speakers, causing intermodulation distortion.

(b) The power from the amplifier is split equally between the speakers. So, you use speakers with a crossover network that is rated just a little lower than the RMS output of the amplifier. Again, the better your crossover, the less distortion you will hear from your speakers.

(c) Tweeters range in size from one-half to five inches in diameter and are usually used to reproduce the higher frequencies (8,000 to 20,000 Hz); midrange speakers range in size from six to ten inches and reproduce frequencies in the range of 600 to 10,000 Hz; and woofers are ten inches in size or bigger and reproduce frequencies from 1,000 down to 1 Hz.

The main difference in magnets in speakers are that some use permanent magnets and others use electromagnets. The electromagnet design makes for a lighter speaker but requires a strong power supply before it can function properly. Although the "W" magnet may be the most expensive magnet to buy for a speaker, the strength of the magnet is the key factor in determining how powerful the speaker will be.

The differences in paper gauge determines what frequencies the speaker will reproduce. The thinner the paper gauge, the more high frequencies the speaker will reproduce; the thicker, the more lower frequencies will be reproduced. Also, the softer the material used, the less higher frequencies will be reproduced. The depth of the cone tends to make the whole speaker resonate at one frequency as the cone gets deeper. Therefore most woofers have a deep construction to them.

The rings or ribs of a speaker are designed to prevent cone break-up, which is when the cone tends to separate into individual vibrating sections that are not harmonically related to the main driving frequency, thus causing distortion. These structural ribs supply a stiffening factor to those sections which would normally vibrate independently of the main cone.

—Clyde R. Green
Cookhouse Recording Studios
Minneapolis, Minn.

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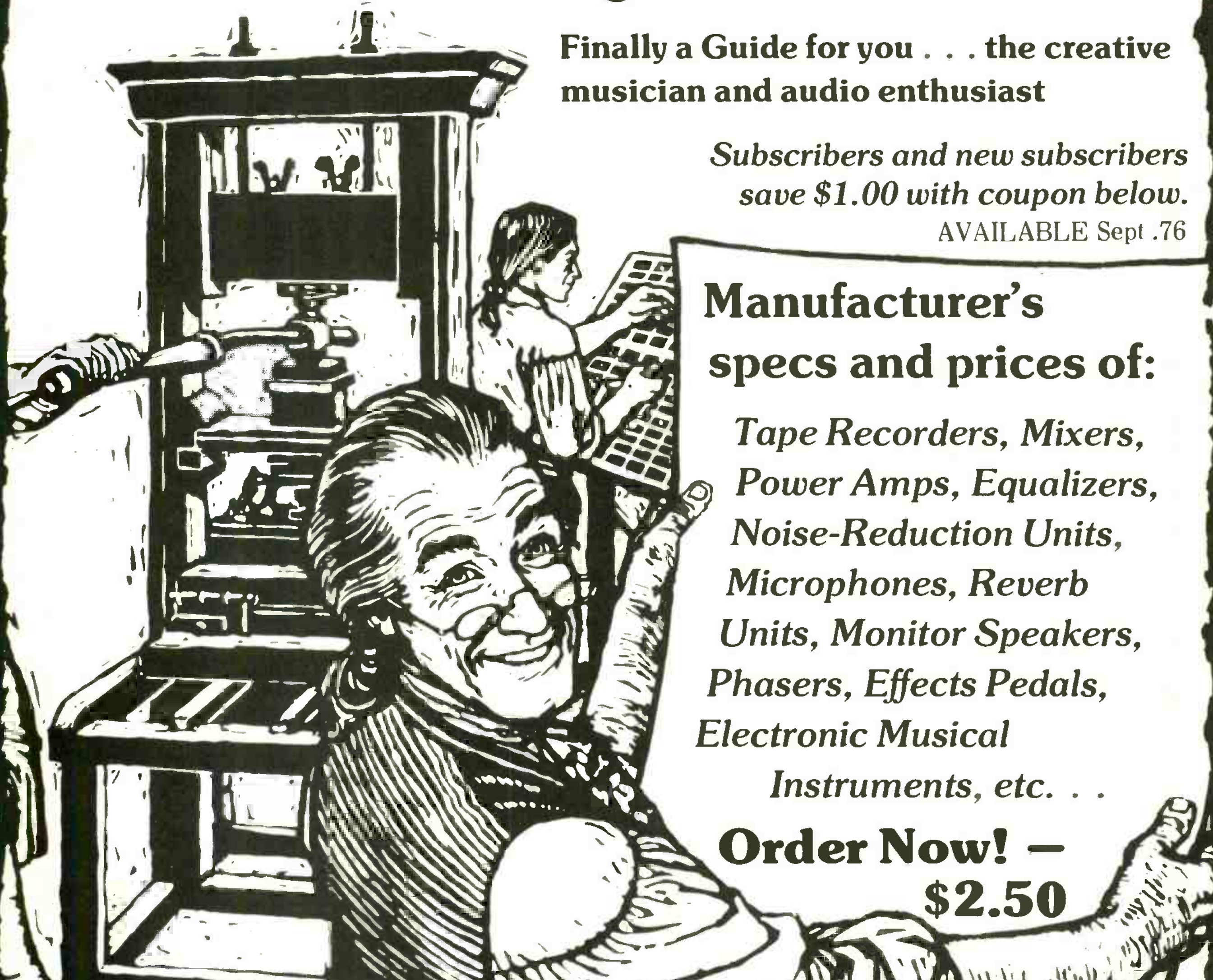

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MR6-1

PRODUCT SCENE

By Norman Eisenberg

CROWN UPDATES SYSTEM PREAMP-CONTROL

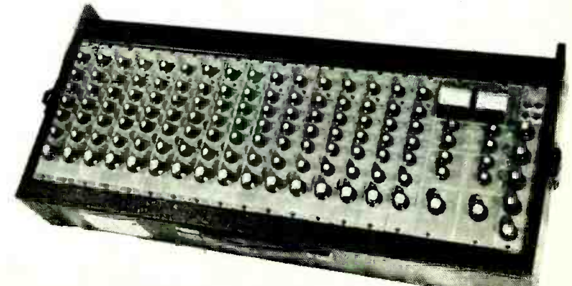
Aimed at the professional user and the sophisticated audiophile is the new model IC-150A preamp from Crown International, which replaces the company's former model IC-150. The new version has been completely redesigned and provides new features as well as new circuitry. Unweighted hum and noise figures are 95 dB below the rated output of 2.5 volts in the high-level section, and 85 dB below 10 mV input for the phono section. The unit's maximum output of 12 volts RMS provides headroom to eliminate preamp clipping distortion. A precision-tracked step attenuator provides level attenuation. A 31-position switch covers a range of 58 dB in 2 dB steps, accurate to within ±0.2 dB. Tracking between channels is accurate to within ±0.2 dB. The IC-150A by itself can control and route signals among two turntables, two tape decks, a tuner, and three auxiliary (high-level) components. Two tape inputs are provided with front-panel monitor buttons, and one of the aux inputs has front panel phone jacks. A stereo phone-jack monitor output also is found on the front panel. The unit weighs 10 pounds and its 19-inch width is suitable for rack-mount. Retail price is $399.

CIRCLE 1 ON READER SERVICE CARD

TWO HEIL UNITS

The Ohmega series of stereo power amplifiers from Heil Sound features modular construction which makes, says the company, for 60-second repair while on the road. Three models have been introduced with power output ratings of 85, 150 and 250 watts per channel, respectively. Each has balanced inputs, meters, sensitivity controls, cooling fan, and is mounted in a rugged case with removable lid. List prices are $490, $675 and $885 respectively.

Heil also has a new mixing console, the model HM-1600, which features 16 balanced and 16 unbalanced low-Z inputs with four full-range outputs, a stereo two-way electronic crossover (12 dB at 800 Hz), a low-noise outboard power supply, individual overload L.E.D. and pan-pots. The console, in carrying case with lid, also features an optional four-channel mix, a sub-master module and a phaser module. List price is $1,830.



CIRCLE 2 ON READER SERVICE CARD

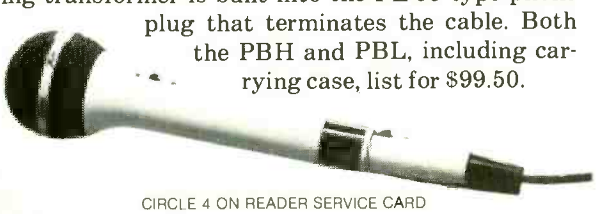
STEREO VERSION OF DBX SIGNAL PROCESSOR

The new model 162 from dbx is a stereo version of the firm's model 160 compressor/limiter. It offers true RMS level detection with the threshold variable from 10 mV to 3 volts. Compression ratios are variable from 1-to-1 up to infinity, with low distortion maintained even at the high ratios. For proper stereo image location, the sum of the channel signals is used as the control voltage. Four or more channels of operation are achieved through strapping so that a single voltage proportional to the sum of the channels controls all the outputs when two or more model 162's are used. Meters may be switched to read input or output level or gain change over a 60-dB range; zero level may be set anywhere from −10 to +10 dBm. The model 162 accepts input levels up to +26 dBm, and it will drive a 600-ohm load at +24 dBm. List price is $600.

CIRCLE 3 ON READER SERVICE CARD

NEW MICS FOR P.A. WORK

The PBL/PBH microphones from Peavey Electronics are specifically designed for extremely high-quality sound-reinforcement work. Based on a newly developed dynamic element using a Mylar diaphragm, the mics are said to be stable, long-lived, and boast high-impact strength and immunity to moisture, temperature extremes, and most acids, alkalis and solvents. Metal bodies are coated for durability and appearance. The magnetic assembly is claimed to be highly efficient and with extremely high sensitivity and clarity of reproduction. The PBL mics come with a 20-foot, two-conductor shielded cable terminated with a three-pin "cannon-type" connector. The PBH is similar to the PBL except that its low-to-high impedance matching transformer is built into the PL-55 type phone plug that terminates the cable. Both the PBH and PBL, including carrying case, list for $99.50.



CIRCLE 4 ON READER SERVICE CARD

TECHNICS BY PROFESSIO

From Technics by Panasonic comes word of a new series of professional units intended for use in broadcasting and recording studios. The model SP-10 Mk 2 is a quartz-controlled, direct-drive turntable (less arm). The quartz control and the use of a phase-locked servo circuit are designed to maintain constant, drift-free speed. High start-up torque means the platter comes up to 33⅓ rpm speed in 0.25-second. The platter also can be braked to halt within 0.3 second. The turntable offers three speeds (33⅓, 45 and 78 rpm). Suggested list price is $700.

The Model SU-9600P is a preamp/control center with an abundance of signal switching, connection, and control functions including two phono inputs with sensitivity/impedance adjustment, 18 dB/octave high/low filters with selectable turnover frequency; separate and stepped right/left bass and treble controls with defeat and frequency turnover selectors; two tape monitors with dubbing feature; and more. Price is $630. Companion to the preamp is the SE-9600P amp (110 watts per channel; $800).

Final item in the series is the SH-9090P Universal Frequency Equalizer, which can tune precisely to any frequency from 5 Hz to 64 kHz. Twelve active filters cover approximately 10 octaves from 10 Hz to 32 kHz, and each of these has a frequency "swing" of plus or minus a full octave around the nominal center frequency, and full control of the filter curve from narrowband to broadband—in addition to 12 dB of boost or cut. Price was not announced at press time.

CIRCLE 5 ON READER SERVICE CARD

TEAC OFFERS NEW LINES

Several new product lines, grouped under various names and scheduled for August-October release, have been announced by Teac. To begin with, there are several regular Teac models in both open-reel and cassette formats. New open-reel models include the A-2340SX, a quarter-track, four-channel deck (7½ and 3¾ ips; 7-inch reel capacity) that offers multi-track recording with simul-sync and costs "less than $850."

In cassette decks, Teac's new front-loading A-650 uses a phase-locked loop, DC servo-controlled capstan motor plus a DC reel motor with full logic circuits for controlling solenoid micro-switch push-buttons. Bias and EQ selectors both have three positions, and the Dolby system has FM/copy and MPX functions. Input mixing is included. Teac lists "nationally advertised value" at "less than $550."

The Tascam series from Teac includes the Model 1—an 8-in/2-out line-level mixer, intended for cue and monitor mixes. A built-in, 1-watt amp powers headphones. Price is "less than $150."

Tascam Model 3, listing for "less than $900," is a basic 8-in/4-out mixer with many features normally found on costlier units—three-position mic switches, two bands of selectable EQ, pan, straight-line fader, a submix section, etc.

CIRCLE 6 ON READER SERVICE CARD

ESOTERIC CASSETTE DECKS

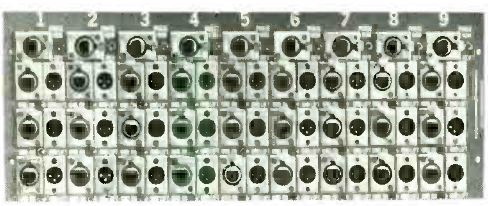
Also new from Teac is its Esoteric Series, which includes the model 860 cassette deck, claimed to be the first in the world that offers an integral dbx noise-reduction system for recording and for decoding any external dbx II-encoded signal. The model 860 features dual-capstan drive, three motors, and a newly designed dual-gap record and play head that is electrically isolated so that the deck provides three-head functioning, such as instant tape/source comparisons. IC logic circuits govern transport functions. The 860 has 4-in/2-out mixing; three-position selectors for bias and EQ; a pitch control to vary tape speed by ±4%; built-in Dolby circuit with external-decoding versatility; meter selector to read average or peak levels; and lots more—for "less than $1600."

The other new deck in the Esoteric series is the model PC-10 which can double on the road or in the field as a portable stereo cassette recorder. Special attention has been paid to operational stability under extreme temperatures (32° to 140° F). The PC-10 weighs 11 pounds and comes with a shoulder strap. Meters read average levels, and there's also an LED for peak level indication. Dolby noise reduction is built in, plus switches for bias and EQ. The PC-10, listing for "less than $500," also has a built-in monitor speaker and other features.

CIRCLE 7 ON READER SERVICE CARD

9 x 3 MIC SPLITTER

From Sescom there's news of a new 9 x 3 Mic Splitter, designed to split up to nine microphones three different ways. The unit has three outputs for every input, which enables it to handle multiple-feed requirements as, for instance, during a "live" performance when three simultaneous outputs may be necessary. In a typical set-up, one output could feed the P.A. system, a second might be used for monitor or foldback sound, and the third could accommodate TV or recording. Other applications include remixing in recording studios, or TV overdub-



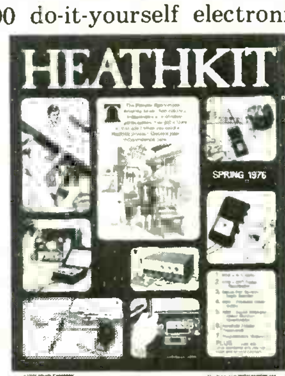
bing and remixing. Originally developed on a "customized basis" for studio and network use in microphone split-bridging, the device is now being offered for general use.

Design features include male and female XLR-type connectors, Sescom transformers with isolation resistors, phase reversal and ground lifter switches. Its rated primary and secondary impedance is 150-250 ohms, but the unit will bridge a mic at over 1,200-ohms impedance. Input level is −10 dBm at 30 Hz with less than 0.2% THD. Price is $1,045 with carrying case. The unit measures 7 by 19 inches and weighs 8.5 pounds.

CIRCLE 8 ON READER SERVICE CARD

SPECIAL FOR KIT BUILDERS

Heath Company, world's largest electronic kit producer, has published its new catalogue containing descriptions of over 400 do-it-yourself electronic projects covering a wide range of equipment from color TV to ham radio. Included are new low-cost test instruments and other first-time products. The catalogue will be mailed free on request to the Heath Co., Dept. 350-02, Benton Harbor, Michigan 49022.



CIRCLE 9 ON READER SERVICE CARD

THREE-C

The failure of four-channel sou... consumer market, not to mention its ... quadraphonic adaptation for fancy mixing and overdubbing by recording activists, has slowed down what seemed not too long ago to be a mad rush by equipment manufacturers to put a lot of four-channel hardware onto the market. Most of what is now coming along is two-channel stereo.

And yet the lure and excitement of multi-channel sound remain. Indeed, the idea of using more than the two channels minimally needed for stereo is as old as stereo itself. The first stereo experiments (back in the days of movies like *Fantasia*) actually employed five or seven channels. And when two-channel stereo became a commercial reality in the late 1950's, a number of audio enthusiasts began experimenting with added channels regardless of how the source material had been recorded. The literature, from the late 1950's and on through the 1960's, is full of material dealing with center-fill speakers, surround speakers, phantom channels, debates over "A plus B" versus "A minus B" mixes, acoustic fill versus electronic fill, out-of-phase rear speakers, and so on. All this, mind you, was based on using plain old-fashioned two-channel stereo as the source material. The only hint of an actual added channel in the recording was a remark casually dropped by Peter Goldmark (then head of CBS Labs and co-inventor of the LP disc) that it might be a good idea to record the hall ambience on a separate track along with the normal two-signal channels, and then reproduce it on a third channel to enhance the playback experience in the home.

The more I listen to quadraphonic sound, the more I am convinced of the essential wisdom and correctness of that idea. For perhaps 99 percent of the world's music (in whatever style or form), there may be no real need for four separate channels. Well-made stereo recordings played on really good equipment, using some kind of extra-speaker array, can give you an awful lot of convincing sound. The egg in the beer would be a rear channel reproducing ambience. Personally, I feel that not only is this all that music in the home needs, but I also feel it is about all the industry can expect consumers to go for in terms of added equipment and cost.

So I propose that someone out there start beating the drum for "triphonic" sound. This idea strikes me as a marvelous compromise that would add something worthwhile to existing music playback while also being in the realm of the possible and viable from the standpoints of technical and economic realities.

SICAL NEWSICALS

NEWS . . . *Polyphony* is a new electronic-music quarterly published by PAIA Electronics. Unlike most electronic-music publications, *Polyphony* is primarily directed at the user rather than the designer of synthesizers. The contents include staff articles and reader contributions (modifications, patches, questions, etc.), and the magazine concentrates, not unexpectedly, on PAIA's own synthesizers and electronic-music module kits. Subscriptions ($2/year) and free copy requests (*Polyphony*, c/o PAIA Electronics Inc., 1020 W. Wilshire Blvd., Oklahoma City, Okla. 73116).

GUITARS . . . The Electra MPC guitar ($500-575, depending on color; you must also buy the case for $65) features interchangeable electronic-effects modules which may be inserted, two at a time, into the back of the guitar. Modular-powered circuits presently available include built-in phaser, built-in overdrive, built-in treble-bass boost, built-in tank tone and built-in fuzz; more modules are on the way. The guitar also has tone-spectrum circuitry which throws the humbucking pick-ups in phase, out of phase, in series or in parallel (Electra Guitars, St. Louis Music Supply Co., 1400 Ferguson Ave., St. Louis, Mo. 63133).

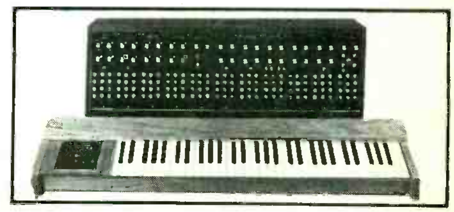
The B.C. Rich Mockingbird ($899) is a solid-body electric guitar with split switch for enhancing highs, phase switch for a hollow, funky sound, six-position varitone control, and a built-in preamp (B.C. Rich, 4770 Valley Blvd., Stall 119-120, Los Angeles, Cal. 90032).

AMPS . . . The RMI Modamp System 700 is a high-power sound system that enables the user to choose and position input and processor modules within the mainframe, according to his/her needs at the time. The mainframe power amp ($1095) produces from 175 watts continuous power (for one

Modamp MC-175 speaker cabinet) up to 700 watts for a maximum of four MC-175's. The speaker cabinets feature two 15-inch woofers, six midrange speakers and six tweeters. With the addition of the power-supply module ($90) and interface module ($88), the Modamp input and processor modules can be added. The KBD (keyboard) module ($121) features controls for input level, preamp gain, channel assignment, hi and lo EQ, reverb/effect mixing, and hi and lo level inputs. Guitar and microphone input modules will be available soon. Processor modules currently available include Graphic EQ ($182) and Filter ($208) with pedal ($68). All modules feature an extractor handle for easy removal and insertion from front without use of tools (Rocky Mount Instruments, Macungie, Pa. 18062).

Polytone's line of studio and stage amplifiers for guitar and keyboards ($395-$1195) features electronic crossover in the preamp at 400 Hz. The split signal can be bi-amped when the amplifiers are used in conjunction with Polytone Side Kick Kabinets ($260-$625) which contain a separate power amp and speakers. Bi-amping can give a monophonic signal a stereo effect. Polytone amps also feature a three-way brite/dark switch, push/pull dials with built-in distortion and sustain, vibrato, reverb, and room EQ switch. "Works in a carrying case"—which is what Polytone calls their amp/preamp—and speaker cabinets are available separately (Polytone Musical Instruments, 1261 N. Vine St., Hollywood, Cal. 90038).

MISC. INSTRUMENTS . . . The Aries System 300 is an attractively priced modular synthesizer available in kit form or factory-wired. The modular system gives the musician the opportunity to start out with a few modules, to choose the configuration of the synthesizer, and to expand or modify the system easily. The complete Aries system ($1195, kit/$1895, wired) consists of keyboard with interface and case, two envelope generators, voltage-controlled (VC) filter,



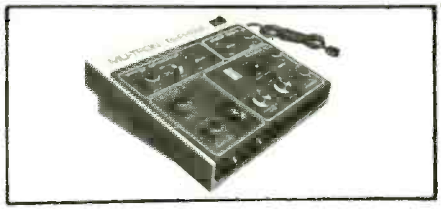
balanced modulator and alternator, VC amplifier, two VC oscillators, VC clock/noise-generator/sample and hold, dual-mixer, dual 10-frequency oscillator/lag/inverter, power supply, output and power module, and case with back-plane connectors (Aries Inc., 119 Foster St., Peabody, Mass. 01960).

ACCESSORIES . . . Electro-Harmonix has come out with a pair of new effects pedals and boxes for the musician. Their Octave multiplexer extends the range of a musical instrument down an octave. One useful application is to enable a guitarist to play especially rapid bass runs. The unit comes in pedal ($199.95) and floor ($89.95) models. The Queen-Triggered Wah-Wah Pedal ($229) has voltage-controlled filters and a function generator to eliminate pot scratch and microphone noise. One of four factory pre-

set frequency ranges can be selected and the automatic trigger allows extra fast wah-wah. Other features include bass and treble boost controls, resonance control, and low- or band-pass outputs (Electro-Harmonix, 27 W. 23rd St., New York, N.Y. 10011).

Rotoverb ($298.50) is Logic Research Lab's electronic simulation of the doppler characteristics of mechanically rotating speakers. Frequency, amplitude and phase modulation are blended to produce the effect plus a wide range of vibrato and reverb effects. The Rotoverb P-100 is a floor unit with power and delay on/off switches; hi and lo input jacks; gain, variable, slow-fast and expand dials; and foot switches for start, fast, slow and choral modes. Logic Research also offers a number of speaker cabinets with self-contoured Rotoverb (Logic Research Laboratories Inc., 2478 E. Fender, Fullerton, Cal. 92631).

The Mu-Tron Bi-Phase ($279.95) consists of two independent six-stage phaser circuits, two sweep generators (oscillators), and additional controls for generating a variety of phasing effects. Other features include LED indicators, mic-stand socket on bottom for mounting, and AC cord storage on back panel. The unit is furnished with a dual foot switch, optional photoelectric foot pedal and provision for



synthesizer interface for external sweep control. Effects include exaggerated phasing, stereo phasing, and synchronized phasing of separate instruments (Musitronics Corp., Rosemont, N.J. 08556).

The new Rowe-De Armond Pan Pedal ($49.95) allows the performer to mix the effects of two sound-effects devices, or go from one to another smoothly. In use with two amplifiers, the pedal also allows the player to move the clean or effect sound from speaker to speaker (Rowe-De Armond Inc., 1702 Airport Highway, Toledo, Ohio 43609).

Wundt Audio has a line of audio accessories for the musician. Their DB-1 direct box ($59.95) can serve as an interface between high-powered amp and mic-level mixer input, or between a high-impedance pick-up and low-impedance mixer input. The box features an automatic grounding system, a switchable filter that simulates most speakers' natural hi-frequency roll off when recording direct, and it accepts levels from a 1000-watt amp to a low-level pick-up. Their CT-1 cable tester ($29.95) is fitted with phone jacks and X-L connectors. By plugging both ends of a cable into the box and rotating the six-position dial, the LED indicates any shorts or breaks in the cable. Wundt also offers a wide variety of cables and snakes (Wundt Audio Engineering, 13026 Saticoy #4, N. Hollywood, Cal. 91605).

Nashville Straights ($5.95-$9.95/set) are the first guitar strings to be packaged straight, instead of coiled in a square envelope. Hermetically sealed in plastic to eliminate oxidation, the nickel- and bronze-alloy strings are available in various gauges for acoustic 6- and 12-strings. Both flat- and round-wound "straights" are available for electric guitars (Nashville Straights, 101 W. Prospect Ave., Mt. Prospect, Ill. 60056).

THE HISTORY OF RECORDING

By Robert Angus

PART 6: The Stereo Years—1950-1976

By the time stereo came to the tape recorder, much of the fun and adventure had gone out of the business. As the tape recorder became broadly accepted, first by professional users in broadcast and recording studios, and much more slowly by the public at large, it became more an instrument from which money was to be made, and less an object of glamour, mystery or wonderment. The story which follows, then, is largely one of men making money and of an instrument evolving from the recording studio to the teenager's bedroom.

The year was 1950, and the stereo tape recorder had only just become available. Developed by Magnecord the year before as a result of a request from General Motors for a twin-track tape recorder which would enable the company's engineers to analyze auto noise, it was being used by a company employee, Bert Whyte, to record music stereophonically. Whyte recorded such groups as the U.S. Navy Band, Lionel Hampton, Leopold Stokowski and Benny Goodman for Magnecord, merely to prove that stereo recording was both feasible and aesthetically pleasing.


Stereo pioneer Emory Cook and his Magnecorder.

His efforts weren't lost on two Easterners. In Stamford, Conn. and Livingston, N.J. lived two young men with parallel interest in audio and stereophonics. They were, respectively, Emory Cook, who would shortly become the major supplier of high fidelity demonstration discs during audio's early years, and Ched Smiley, who would become the first man to produce stereo tapes for sale to the public. Before that, however, he and Cook would develop the binaural record and a system for playing it back.

The first step for both men was to acquire one of the stereo Magnecords. Cook packed his off to the New York Central's railyards at Harmon, N.Y., to the seashore, to Mexico, to wherever there were dramatic and unusual sounds. In 1951, Smiley took his to Florence, Italy, where the May Festival was under way. There he recorded nearly three hours worth of symphonic favorites with the Florence May Festival Orchestra.

The Cook binaural record, for which these recordings were destined, consisted of two bands on the record—an outer band representing the left channel; an inner band for the right. To play it back, Smiley's Livingston Audio Products Corp. developed a Y-shaped arm with two cartridges. Theoretically, when the arm dropped onto one of Cook's records, the two channels would be synchronized. But the synchronization was practically never perfect because it was well-nigh impossible to insure that the dual styli would drop exactly into matching grooves. So Smiley abandoned records and turned his attention to producing stereo recordings on tape.

Rush on Mono Recorders

Meanwhile, tape enthusiasts, blissfully unaware of the behind-the-scenes stereo activity, rushed out to buy monaural recorders made by Wilcox-Gay, Crestwood, Bell Sound Systems, Revere, Webcor and others. Recording Associates beat Smiley to the punch by issuing the very first commercially-recorded tapes—eight half-hour programs of organ music. However, these weren't stereo recordings. Nor were the Magnecordings by Vox, assembled by Whyte for Magnecord from monaural European recordings made by George Mendelssohn of Vox Productions. To the hopeful sales managers of Scotch, Ampex, Berlant-Concertone and Magnecord, it looked as if the popularly-priced monaural tape recorder might do in the home what the professional recorders of Ampex and Magnecord were doing in broadcast and recording studios—eliminate the disc altogether.

Alas, it was not to be. Tape recorders were edging downward toward the $100 level. And while those first recorded tapes cost $8.95, you could buy records for about half that price featuring big-name orchestras, bands and soloists which contained 50 percent more music.

What made the difference was Smiley's stereo tapes, finally released in 1954 when he felt there were enough players available to make the project commercially feasible. In addition to the Magnecords, Smiley developed and manufactured his own low-cost stereo tape deck and stereo amplifier. Suddenly RCA sat up and took notice. So did Mercury Records, a Chicago-based label for whom Whyte by now was working. Both RCA and Mercury began making stereo recordings of their major symphonic recording sessions—the latter with Whyte in charge. RCA's initial blockbuster, Strauss' *Also Sprach Zarathustra*, recorded by Fritz Reiner and the Chicago Symphony [still available on Victrola VICS-1265], added impetus to the stereo recorded-tape movement, and by 1957, half a dozen classical record labels and several pop labels were producing their own stereo tapes or leasing masters to independent producers like Smiley. Then something happened.

The Stereo Record

What happened was that Decca Records in England and Westrex, a subsidiary of Western Electric in the United States, had found a way of doing what Smiley and Cook could not—putting both stereo channels into a single record groove. At about the same time, RCA started talking about a tape cartridge it would shortly introduce—a cartridge which could be changed automatically like a record, which contained the tape so that no threading would be necessary or tape spill possible. Best of all, the new cartridge would cost about the same as a phonograph record, thus eliminating the need for open-reel tape. It took RCA more than a year to get its product onto the market. But in that time, sales of recorded open-reel tape plummeted from about $2.4 million in fiscal 1957 to less than $100,000 the following year.

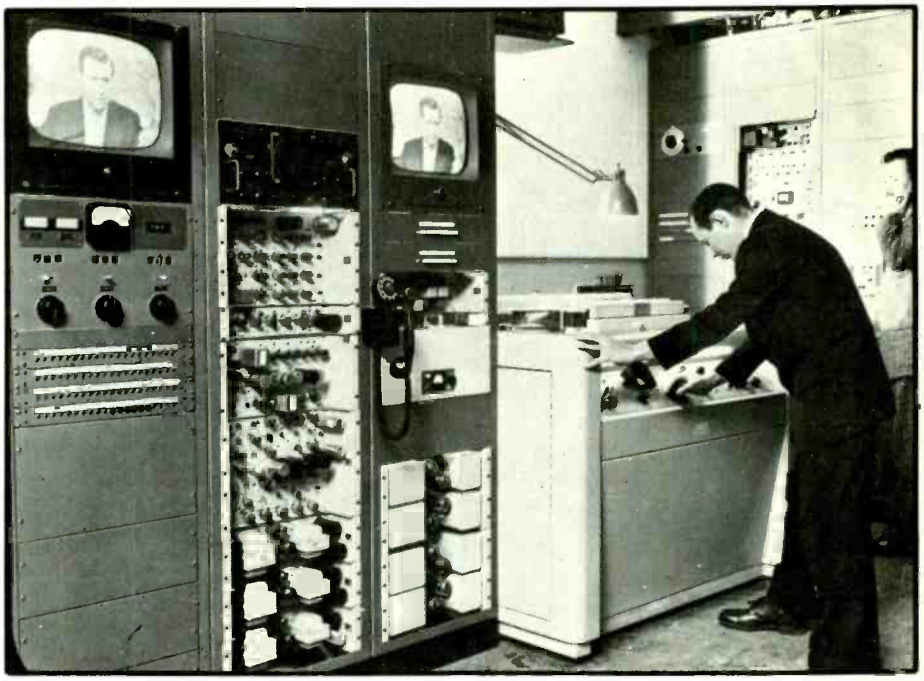
When the RCA cartridge eventually did appear, the price was about midway between that of a stereo record and an open-reel tape. The cartridge actually was the forerunner of today's cassette, containing two hubs, quarter-inch tape and an overall size larger than that of a paperback novel. The worst thing about the system, however, was that RCA seemed to be incapable of making a changer that worked. The tape jammed or sounded bad. By the time RCA, Mercury Records, Bell Sound and a few other manufacturers gave it up as a bad job in 1963, open-reel tape was making a comeback.

That was due primarily to some forthright action by Ampex early in 1959. Not only had the 1958 slump hurt the sale of music on tape, it had been equally devastating to recorder manufacturers. As one of the largest and best-financed, Ampex decided to do something about it. What Ampex did was to create the quarter-track format—two pairs of stereo tracks where once there had been one on conventional quarter-inch tape. First the company made the machines, urging every other recorder manufacturer to do the same. Then it created a subsidiary to release music on tape—not the organ musings of Hack Swain, or Lenny Herman's Mightiest Little Band in the Land, but the biggest names it could find. To get them, Ampex went first to London Records, English Decca's American subsidiary. It locked up tape rights to the mammoth London classical and pop catalogues, then went on to sign up as many smaller labels as it could. To head the operation, Ampex hired Edward R. Wallerstein, a former chief executive at RCA Victor, and president of Columbia Records at the time that company introduced the long-playing record.

Early Videotape Experiments

Before we leave the 1950's, there's one more important development to examine. In November, 1951, Bing Crosby Enterprises demonstrated the first videotape recorder. Produced by Ampex, it went into network use some 2½ years later on CBS's evening newscast with Douglas Edwards. Just as audio tape had, in seven years, eliminated the practice of repeating programs "live" for the different time zones, videotape almost overnight swept the television industry, making delayed programming possible. Even before the Edwards tape was aired (in black and white, of course), RCA was announcing plans for home videotapes in color. The year was 1953, and by that time, Bing Crosby Enterprises was already demonstrating an Ampex color videocorder, designed for professional use.

The RCA cartridge, while it may not have worked, managed to disrupt the tape market for nearly three years. Before it vanished from the scene, there was yet another attempt at tape automation. This one had auspicious sponsors. It was the brainchild of Marvin Camras, the man who had done so much pioneering work on wire recording in the early days of World War II, and the man who held the basic patent on red iron oxide formulation. The tape cartridges—three-inch squares of plastic containing a single hub with the tape anchored firmly at the end and a heavy polyester leader which the player mechanism could grab, to pull the tape out of the shell—would be made by Minnesota Mining. A 3M subsidiary, Revere-Wollensak, would make



The historic first broadcast via tape was CBS' Nov. 30, 1956 airing of "Douglas Edwards and the News" from New York City.

the cartridge changers, and Columbia Records would provide the music. Bell Sound Systems and its recorded music subsidiary, Bel Canto Tapes, still involved with RCA's unit, rushed to provide tapes and hardware to accommodate the new system. Unfortunately, the 3M and Bell changers suffered the same problems as the RCA units. They were highly successful at pulling the tape out of the shells, not nearly so efficient at getting it back in again. The changers jammed, and the entire system collapsed.

During the summer of 1964, the open-reel tape business was humming along nicely, thanks in no small part to Ampex's aggressive merchandising of recorded music. With the RCA and 3M disasters fresh in mind, it hardly seemed the most opportune time to launch a new tape system. Nevertheless, that's exactly what Philips of the Netherlands did—not as a music playback system, but as a lightweight portable dictating system.

The Norelco 150 Carry-Corder didn't look like the kind of machine that could revolutionize anything. Sure, it was compact, small enough to fit in an overcoat pocket. At 2¾ pounds it was lighter than any conventional recorder. But the sound simply wasn't good enough for music reproduction, and the initial price—$149.50—put it in the class with some AC stereo decks. For more than a year, it seemed that officials at Philips and Norelco didn't realize what they had in the Carry-Corder and the Compact Cassettes it used. These tiny plastic wafers looked like miniature versions of the ill-fated RCA cartridge. Yet, despite their size and the slow speed with which the tape moved—1⅞ inches per second—they worked. And, if you weren't too critical, you could even record and play back music on them with not-too-disappointing results.

Automotive Tape Players

Still, to the people who invented it, the Carry-Corder seemed to be nothing more than a high-priced dictating machine or electronic toy. But something was to happen which would make them change their minds. What happened was that a West Coast marketing genius named Earl Muntz had run across a 1954 invention by George Eash, and decided to build a business.

Sporadically during the postwar years, manufacturers of record-playing equipment and automobiles flirted with the idea of putting record players into automobiles. The problems were many. How could you keep the tone-arm in the record groove on a bumpy road? Unless the records were of a special type, how could you make a player small enough to fit into the dashboard of a car? Even if you achieved these goals, how could you make records which would last more than a couple of playings?

Eash, a garage tinkerer, was working for a man who said he'd like to have a music system in his car. Instead of monkeying around with records, Eash concentrated on tape, coming up with a compact (by 1954 standards) player which utilized an endless loop cartridge that didn't have to be threaded or turned over. The tape unwound from the center of the tape pack, fed across an open face which permitted contact with the player head, and rewound on the outside of the tape pack. Eash patented it. Some six years later, he mentioned it to his friend Earl Muntz. Muntz had built a fortune in the early postwar years in used cars. One of the first used car dealers to use television to advertise, he became known throughout southern California as Madman Muntz. Then, in 1950, he decided that the television business looked even better than used cars, and began making Muntz television sets. Several years later he sold out, profitably.

When Muntz saw what Eash had been working on, he saw new business vistas opening before him. Before long, Muntz was manufacturing car tape players based on Eash's invention, modified to utilize the four-track standard developed by Ampex. The public didn't actually become aware of the four-track cartridge until just about the time the cassette made its appearance. In 1964, however, Muntz not only was selling players and building a library of music to go along with them, but setting up his own Muntz car stereo centers to sell and install the equipment.

Earl "Madman" Muntz (left), legendary supersalesman and pioneer of car stereo tape equipment, oversees installation of a new system.

Business, in fact, was so good for Muntz that all sorts of people were hopping on the bandwagon. One of those who heard it rolling by was Bill Lear, an inventor and electronics executive best known for building the Lear Jet executive airplane. Just how Lear's Stereo 8 cartridge (containing eight tracks instead of Muntz's four, but similar in external size and shape) came into existence isn't entirely clear. Muntz has said that it was a blatant attempt to get around the Eash patents, and Bill Lear hasn't effectively denied that. In any event, by 1965, Lear had a product good enough to interest the executives at RCA, Motorola and the Ford Motor Company. A plan was announced whereby RCA would provide music exclusively for the new system, Ford would sell cars with eight-track players built into them (and would urge its dealers to sell players for older cars), and Motorola would make the players. With the advantage of a heavy advertising barrage, plus RCA's leverage in the record

business, eight-track overshadowed Muntz's cartridge and the lowly cassette almost from the start. After all, in 1965, there was no such thing as a stereo cassette.

One man who was upset by all of this was Wybo Semmelink of North American Philips. Semmelink was in charge of selling the 150 Carry-Corder and the blank cassettes which went with it. A tape and audio fan from 'way back, Semmelink was convinced that the cassette could do a much better job of reproducing music—and doing it in the car—than either of the cartridges. It took him a year of heated arguing with the powers that be at Philips in Holland to convince them. In 1966, Norelco introduced an assortment of cassette products which included not only a stereo deck for the home and a car conversion kit for the Carry-Corder, but a cassette changer system and a car tape player. But Semmelink went further. He persuaded Ampex, which was already making music for both the Muntz and Stereo 8 systems in addition to its open-reel tapes, to produce recorded cassettes as well. A three-way race was on.

The cassette quickly became a home music and portable music playback system. As improvements tumbled one on top of another, the cassette made short work of the less expensive open-reel recorders, while eight-track chased the Muntz cartridge off the scene in about three years. By 1976, the cassette had eliminated the open-reel portable recorder (with the exception of a few professional units), and had made open-reel home stereo recorders priced under $400 virtually obsolete.

Dolby Noise Reduction

Perhaps no development played a greater part in the cassette's success than the development of a low-cost noise-reduction circuit which could be built into the recorders. Because of the slow tape speed and ultra-narrow tape tracks, noise had been a problem with the cassette from the beginning.

The man who supplied the idea was Ray Dolby, an electronics engineer who had worked for Ampex during his high school vacations, then won scholarships to Stanford University and Cambridge. Dolby was a music lover who was intrigued by the problem of noise which seemed to be inherent in tape recording. During a stay in India with the Peace Corps, he hit on the idea of breaking up the frequency spectrum during recording so that certain frequencies and certain levels of signals are recorded at higher levels than others. Then, during playback, these signals are suppressed by exactly the same amount, at the same time suppressing any background noise or hiss.

By November 1965, he had built a full-frequency unit, designed for professional recording work. Decca Records was so impressed by his demonstration that they bought every unit he could produce during the next five months. Within a year of delivery of his first working machine to Decca, the Dolby A noise reduction unit had become a fixture at broadcasting and recording studios in Britain.

Wybo Semmelink and the first Philips cassette players. (Inset) 3M's early stereo cartridge player.

The following year, Dolby received a phone call in his London office from Henry Kloss, then president of KLH. Kloss wanted to know why it wouldn't be possible to manufacture a low-cost version of Dolby A suitable for building into a home recording tape deck like the one being marketed by KLH. By late summer 1968, Dolby did just that, enabling Kloss to market the first two home recorders with Dolby B noise reduction. Both were open-reel recorders.

During this period, Dolby took a good look both at cassettes and cartridges with an eye toward incorporating his noise-reduction system. He was unimpressed with what he saw and heard. Nevertheless, during a visit to the United States, he told the Audio Engineering Society that there was no reason why a Dolby B integrated circuit chip mightn't be developed which could be incorporated into a cassette recorder.

One of the people who heard him was Richard Ekstract, publisher of the trade newspaper *Audio Times.* Ekstract mentioned Dolby's comment to his friend Walter Goodman, then chief executive officer of Harman-Kardon. As it happened, H-K had just come out with a cassette deck which was drawing rave reviews from audio critics. But the company was all too aware of its shortcomings and was looking for a way of upgrading the new model which was already on the drawing boards. Dolby noise reduction seemed like a heaven-sent answer.

Meanwhile, Dolby had taken a Wollensack 4700 cassette deck back to England with him. For the first time he saw a cassette unit with what he considered high fidelity possibilities, and started to work on the necessary IC. And at just about the same time, Henry Kloss, having left KLH, was setting up a new company, Advent, to market a television projection system he'd been working on.

In order to pay the bills, Kloss needed a product to sell right away. The product he chose was a cassette recorder, and to increase its high fidelity capabilities, Kloss fell back on his

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CIRCLE 92 ON READER SERVICE CARD

idea of Dolby noise reduction, not knowing that H-K and perhaps other manufacturers were thinking along the same lines. Kloss's idea this time was that the cassette had the potential to emerge not only as the ultimate home music system by combining all of the technical ingredients available, but might even be the ideal picture-and-sound medium. So he designed a deck which featured the best mechanical tape transport he could get, Dolby noise reduction and the ability to handle chromium dioxide, then being touted as the ultimate tape for both audio and video. Kloss moved to become the first manufacturer to offer not only a deck capable of using chrome tape, but the tape itself. Further, he planned to offer recorded chromium-dioxide cassettes, to complete the music system. The latter were not to appear for another three or four years, due to difficulties in getting really good tape masters, the proper duplicating equipment, and ironing out all the bugs.

Those first Dolbyized cassette decks were instantly successful. By the early 1970's, you simply couldn't sell a cassette deck priced above $300 if it didn't include Dolby. Several Japanese manufacturers who tried to buck Dolby with their own noise-reduction systems found it very difficult, indeed.

Developments in Japan

Until the late 1950's, virtually all of the tape recorders sold in the United States were made either domestically or in Europe. But with the coming of stereo in 1958, Japanese manufacturers began to show an interest. The first attempts were low-cost, low-fi battery portables. But in 1957, a Hollywood executive named Joseph Tushinsky was in Tokyo on business. As a result of his activities, he found himself with some extra yen which would be very difficult to convert to dollars when he got home.

"So," he recalls today, "since I had all that money, I went around looking for things to buy." In this pursuit, he traveled to the Shinagawa section of the city to visit a small new company named Sony. "There were only a handful of employees, and they were working on a dirt floor inside a converted quonset hut. They took me into a corner of the hut where there were some wooden tables and a couple of signs. On one of the tables was a tape recorder. Now, I was interested in tape

Ray Dolby, whose noise reduction inventions played perhaps the greatest part in the cassette's success.

recorders, so I asked about it. They told me they'd just developed it. They claimed it was the first stereo tape recorder anywhere in the world with built-in power amplifiers.

"So I asked how many they had. They told me there were seven. I asked how much each one would cost. I think they replied a hundred dollars each. 'Okay,' I said, dipping into my pocketful of yen, 'I'll take all seven.' They looked shocked and asked why I wanted them all. I replied that I wanted to take them back to the United States to see whether there was a market for them. If so, I explained, I might be interested in distributing them in the U.S. The real reason was to keep anybody else from seeing them before I'd finalized a deal."

Tushinsky's Superscope Inc. became Sony's distributor in the U.S. in 1958, the first company to offer a high fidelity recorder from Japan. Another man responsible for the Japanese invasion was Robert Metzner, who had been buying motors from Akai Electric Company for his Metzner Starlight turntable. At the time (roughly 1960), the Ampex 601 was the envy of every amateur recordist who couldn't afford one. When the officials at Akai saw it, the idea of producing a copy which would look suspiciously like the 601, would record and play back stereophonically and which would sell for a fraction of Ampex's price seemed almost irresistible. The next thing Metzner knew, he was heading a company called Roberts Electronics, the American distributor for the Akai recorder. Metzner, an engineer, insisted on taking an active part in the design and engineering of the Roberts. The success of these two encouraged other manufacturers to develop and market their own tape recorders.

During the 1960's, the percentage of recorders made in the United States dropped from about 65 percent at the beginning of the decade to less than five percent at the end, thanks initially to lower labor costs in Japan; later to superior technology and performance.

The first Japanese tape recorders were unimpressive devices, cheap rim-drive portables which generally sold for less than $100. One, in fact, was offered by mail in 1958 for the incredibly low price of $29.95. The manufacturer, Wakataka Bussan, got its hands on the American Medical Association mailing list, and wrote doctors proposing that they become distributors in their towns for the company's tape recorder. As distributors, they could buy the machines at the distributor price of $29.95, half the price the recorders were supposed to sell for. The letters suggested that by selling the recorders, the doctors could supplement their incomes. The National Better Business Bureau regarded this idea as preposterous and suggested that all Wakataka Bussan was trying to do was to give the doctors the idea that they were getting a bargain on the recorders.

Until Sony and Roberts came along, then, Japanese tape recorders had a poor reputation in the United States. However, the success of Tushinsky and Metzner encouraged other American businessmen and Japanese manufacturers to get in on the act. One of the first was Hosho of America, a company formed in 1966 with actor William Holden as a principal, which bought recorders from the giant Matsushita Electric Company. Unfortunately, component-quality tape recorders were new to Matsushita, and the number of lemons delivered to Hosho was very high. The company changed its name to Concord, and spoke sharply to Matsushita. The percentage of lemons dropped, but they still appeared with disconcerting regularity.

Meanwhile, other Japanese companies—Mitsubishi, Hitachi, Japan Victor and others—published elaborate catalogues showing, among other things, a range of professional tape

It's all in your hands now.

These essential music-making tools let you get in, around, and behind your music. One part, one track at a time. Then when all the elements are the best they can be, you blend them together and generate the finished product. A process that involves your innermost sensitivities and perceptions. A personal statement.

The 3340 has undoubtedly helped more people make more demos and masters than any other tape recorder. Our new model, A-3340S, has extended high end frequency response in the sync mode, plus manual cueing. The Model 2 is about the best value going in small mixers—6-in, 4-out with full panning and multiple patch points. Plug in a few high quality TEAC ME-Series electret condenser mics, and you could be quite literally in business.

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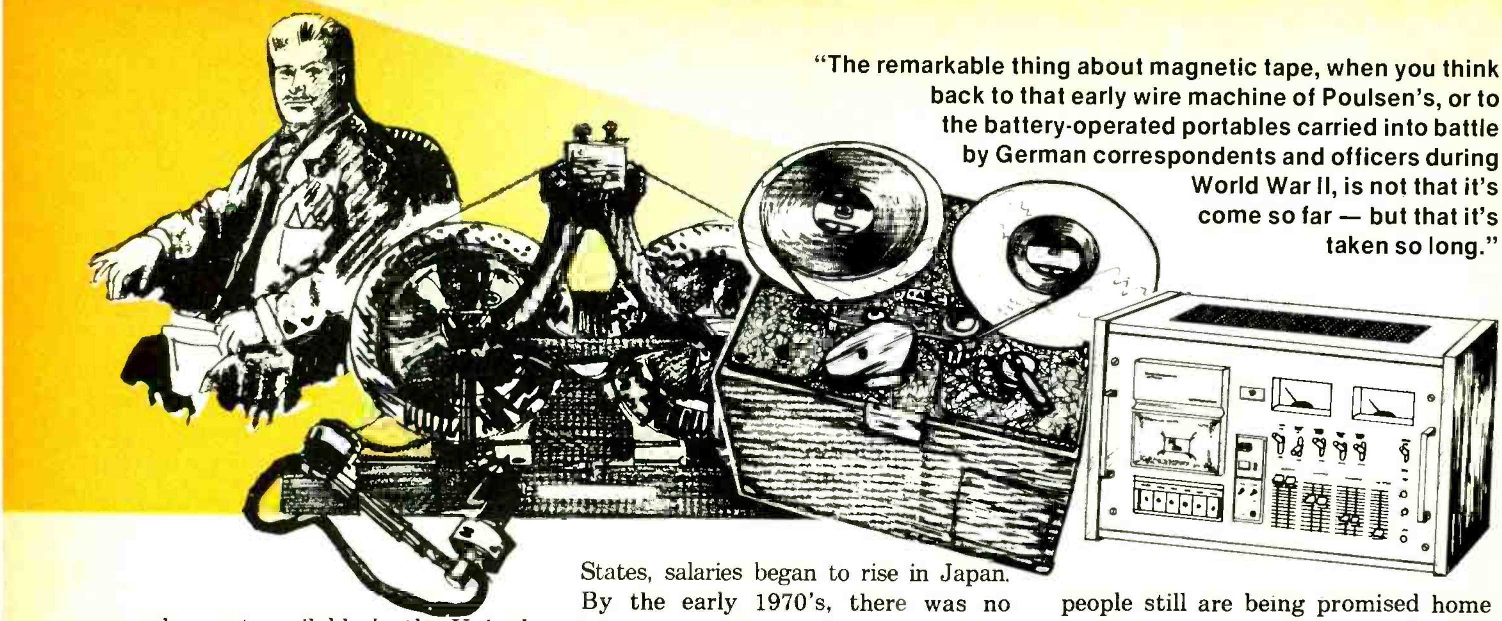
"The remarkable thing about magnetic tape, when you think back to that early wire machine of Poulsen's, or to the battery-operated portables carried into battle by German correspondents and officers during World War II, is not that it's come so far — but that it's taken so long."



recorders not available in the United States. These studio-type units looked better than anything available in the United States at the time, but the companies never delivered. That in turn led some Americans to suspect that the units didn't really exist.

Another suspicion was that the lemons being delivered to Concord, Roberts and others were the result of Japanese manufacturers using the brand names of others to experiment, make mistakes, and learn the tape recorder business—without tarnishing the reputations of their own brand names. That suspicion seemed to be confirmed in the mid-1960's when first Matsushita and later Akai and others introduced new tape equipment designed to compete with the recorders they were supplying to importers—and the failure rate on their own was much lower than on the American brands.

Perhaps as a result of their earlier experience, or as a result of improved technology, one Japanese manufacturer after another introduced tape recorders which not only represented value and possessed audiophile or professional features, but which performed up to specification. As they did, American manufacturers like Ampex, Concertone and others first began buying recorders from Japan, then shut down their domestic factories, and eventually got out of the tape recorder business altogether. The cassette and cartridge, introduced in the mid-1960's, accelerated the process.

What gave the Japanese a competitive edge initially was the low cost of labor in the Far East. But as Japanese-made tape recorders increased their share of sales in the United States, salaries began to rise in Japan. By the early 1970's, there was no longer the differential in wages which had existed a decade earlier—but by that time, American tape recorder manufacturing capability had all but disappeared. The result was more expensive—albeit much more reliable and much higher-fidelity—tape recorders for home and studio.

Home Videotape Recorders in the Future?

In 1964, the British Information Service startled the tape world with news that a British manufacturer, Telcan Ltd., had produced a videotape recorder which used ordinary audio tape and cost a mere $94. The black-and-white one-of-a-kind recorder made a belated appearance in New York, where it drew unfavorable notices, but it spurred Sony, Ampex, RCA, Concord and others to race for a video recorder which would be suitable for home use. By 1967, Sony was running television commercials for its $1000 black-and-white system, while Ampex actually was marketing a similar system on a limited basis. Telcan never went on sale in the United States (or anywhere else), although an importer later tried to sell a kit version manufactured in Japan for $164. That never got anywhere, either.

Whether it was their high cost, the fact that color was just becoming popular, or merely that they were ahead of their time, the Sony, Ampex, Concord, GBC and other "home" video recorders never made it. Nor did color versions introduced in the early 1970's—partly because of cost, partly because they were difficult to set up and operate, and partly because there was no recorded material to play on them. At this writing, the American people still are being promised home videotape (or magnetic videodisc) at some time in the foreseeable future.

The remarkable thing about magnetic tape, when you think back to that early wire machine of Poulsen's, or to the battery-operated portables carried into battle by German correspondents and officers during World War II, is not that it's come so far—but that it's taken so long. Charles Dexter Rood had hundreds of small stockholders in 1912 who were convinced that his recorders were superior to anything obtainable from the phonograph. During World War II, the Germans demonstrated that their tapes could fool experts into believing they were listening to "live" broadcasts. Yet it's taken until the 1970's for the average American to accept magnetic tape as a music medium equal to the gramophone record. Even now, we lag behind our British and European cousins, who spend much more of their francs or pounds or lira for recorded music on tape than we do.

The tape recorder in the studio may look like a remarkably unglamorous item. But lurking behind its brushed aluminum template is a remarkable history peopled by the likes of inventors Valdemar Poulsen and Marvin Camras; Charles Dexter Rood, the man who may have been a visionary or a scoundrel, or may simply have been senile; S.J. Begun and a Signal Corpsman named Jack Mullin; men like Bert Whyte and Ched Smiley, who had definite ideas about what the recorder could be used for; and men like Joe Tushinsky and Earl Muntz who knew how to make a buck.

End of Series

Think of them as your musical instruments.

The audience can't see you. But they can sure hear you.

They don't know it, but they're depending on just one person to get the music to them. And that guy is you.

It's not something an amateur can do. It's an art. And that's why Yamaha has designed 3 superb mixing consoles with the qualities and range of controls that the professional sound reinforcement artist needs.

For instance, our exclusive 4x4 matrix with level controls gives you more exacting mastery over your sound than the conventional method of driving speaker amps directly from the bus outputs.

Features like that are years away except on the most expensive mixers. On the Yamahas, it's standard equipment. And so are transformer isolated inputs and outputs, dual echo send busses, an input level attenuator that takes +4 dB line level to −60 dB mike level in 11 steps, and 5-frequency equalization.

Whether you choose the PM-1000-16, the PM-1000-24 or the PM-1000-32, Yamaha gives you the flexibility you need to turn your job into an art. And because they're designed from the ground up to perform on the road, more and more professional sound men around the United States and the world are depending on Yamaha, night after night, gig after gig.

If you've never thought of your mixing console as a musical instrument, we'd like to invite you to stop by your Yamaha dealer. Once you've checked out the operation manual and tested for yourself what the PM Series can do, we think you'll come away a believer.

YAMAHA

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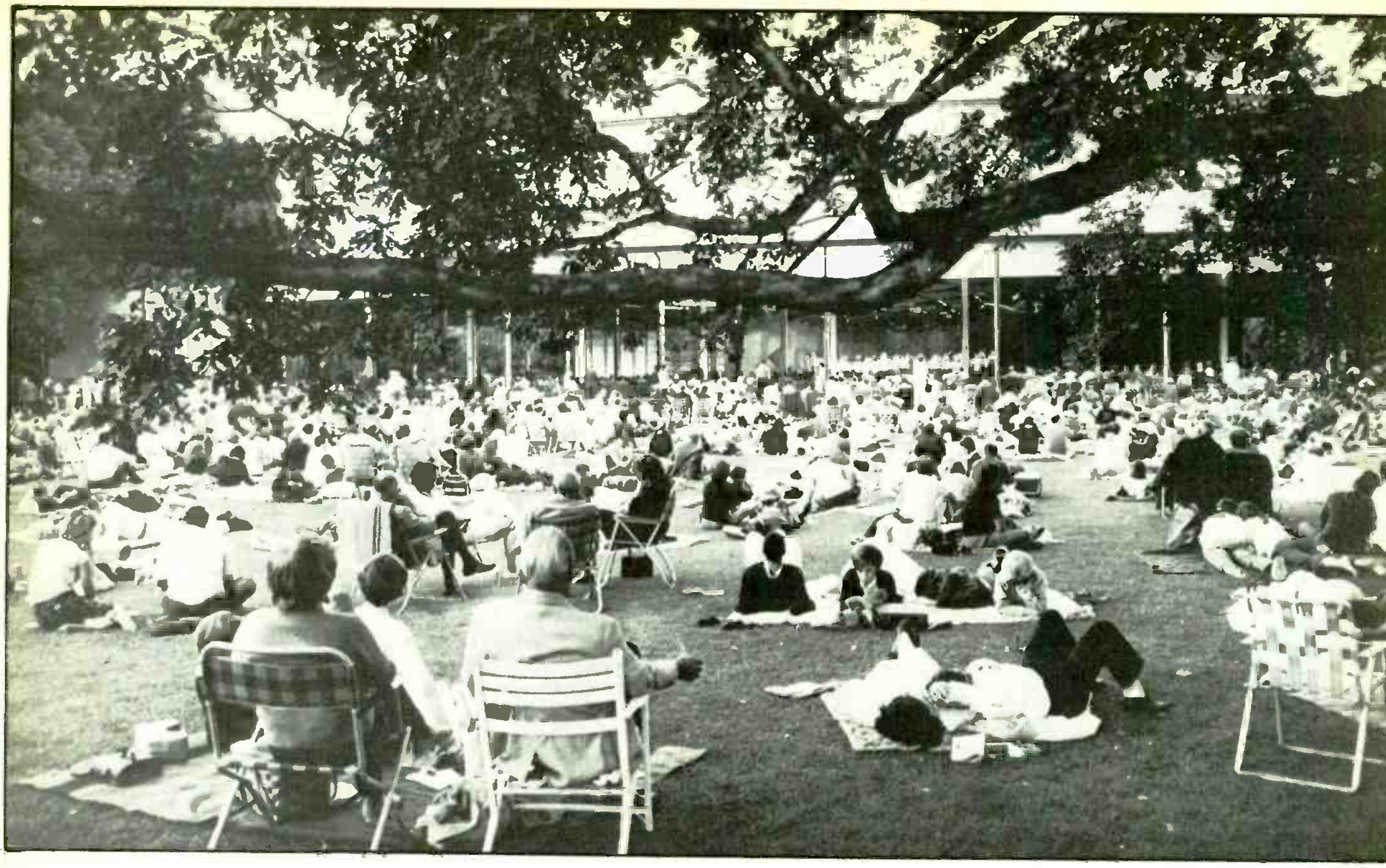
CIRCLE 25 ON READER SERVICE CARD



Whitestone Photo

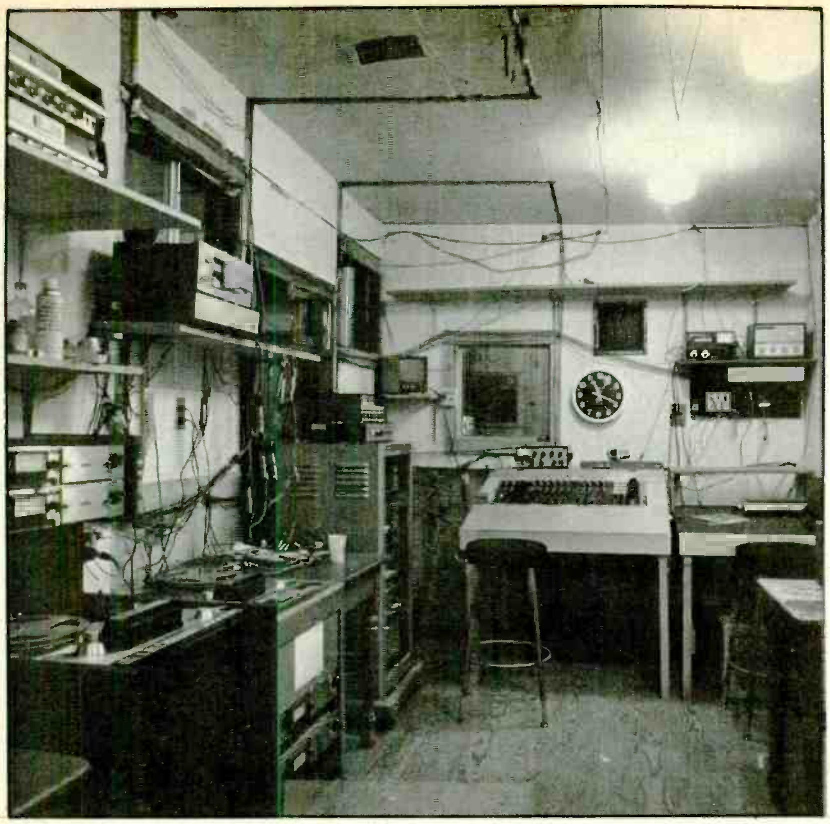
Behind the Scenes at TANGLEWOOD

By Norman Eisenberg

Photos by Walter H. Scott

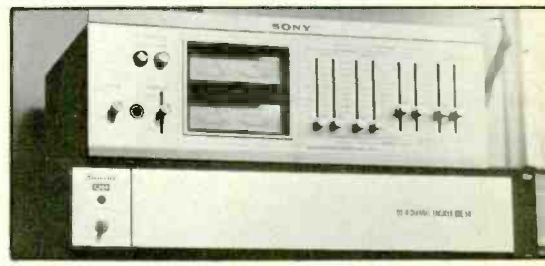
Tanglewood is the name of the oldest summer music festival in America. The name is the same as that used by Nathaniel Hawthorne in his *Tanglewood Tales,* and denotes a particularly idyllic section of the Berkshire Hills between the towns of Lenox and Stockbridge, Massachusetts. Here on 210 acres of lawn and shrubbery, with lakes and hills as a backdrop, the Boston Symphony has made its summer home since 1934, attracting an ever-growing audience. More recently, the Festival has added rock and pop concerts to the normally classical offerings and in a real sense Tanglewood is becoming all things to all kinds of music lovers.

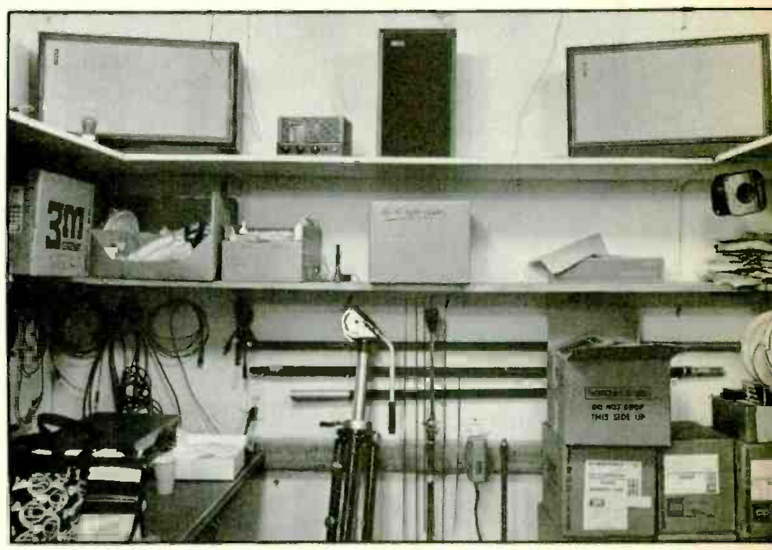
It also has become all things to some 130 radio stations here and abroad (including Canada, Mexico, Britain, and Australia), thanks to an audio electronics center behind the performing stage where an imposing array of modern signal-processing, mixing, recording and monitoring equipment is crammed into a room which itself is so behind the times that plastic covers must be placed over the precious equipment during heavy rains to protect it from ceiling leaks. What the equipment is used for—in addition to "live" broadcasting—is to produce recordings for re-use by stations transmitting in every possible mode: mono, stereo with Dolby processing, matrixed quadraphonic (both SQ and QS), and discrete quadraphonic. In addition, the room is the nerve-center and control-station for a time-delayed sound-reinforcement set-up for listeners on the lawn (the sound reaching them from the "live" players takes longer to travel than the amplified sound and so the latter must be delayed by a critically precise split-second); for a sophisticated P.A. system in the shed that is used only for special voice-narration parts with the musical performance; for a special P.A. rig that has to be used to reinforce the tones of the harpsichord when it is



(left) General view of audio control and signal-processing room behind the stage at Tanglewood. Far right is producer's spot, fitted with telephone, squawk box and timers. On wall is P.A. equipment that connects to stage, press-room, and other strategic locations around the grounds. Window over mixing console looks out to window in announcer's booth. TV set at far left provides control team with view of stage, since it was impossible to arrange this room to have a direct view of the performers. Shelf at far wall usually holds two speaker systems for in-room monitoring (these were out for repair when picture was shot); in addition, there are two more speakers at viewer's end to comprise full four-channel playback.

(below) Tapes made of concerts include quadraphonic versions using both encoding systems preferred by different radio stations. Sony encoder handles SQ matrix; Sansui decoder does the job for QS matrix. Stations getting these tapes need do no further processing; tapes may be broadcast "as is."



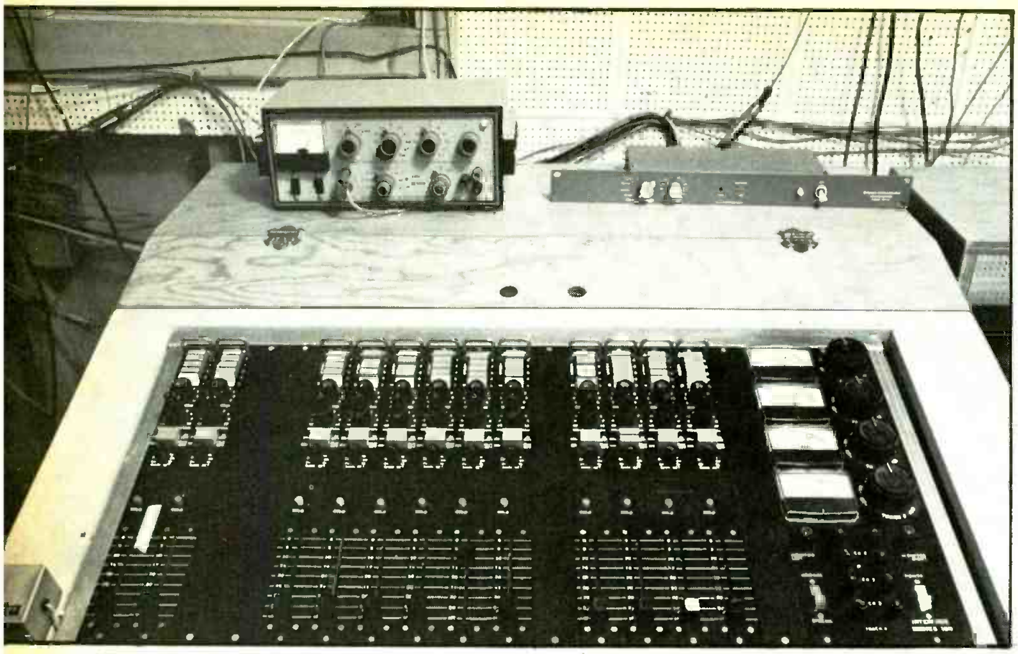


played (this was installed at the prompting of one-time music director Erich Leinsdorf); for a separate series of tapes made for experimental and demonstration purposes; and to add the voice of the radio announcer who, from his own small anechoic booth, announces each program for the radio audiences. Superimposed on the whole conglomerate is a private communications system that links strategic parts of the scene, such as the press office in another building and a special monitor position in one of the box-seats.

Mastermind and general factotum of the set-up is Richard L. Kaye, manager of the Boston Symphony Transcription Trust, and executive vice-president of radio station WCRB, Waltham, Mass. Producer of the recordings is Jordan M. Whitelaw who writes the radio program scripts and helps determine the levels for the mixes. Resident engineer is Steven R. Colby who, with Whitelaw, faithfully attends rehearsals to get an idea, ahead of tape-time, of what the levels are likely to be. The announcer, whose well-modulated voice is familiar to millions of radio listeners, is William Pierce. The newest member of the team, and its first female, is Beth Park.

The concerts originate on stage at one end of an expanding shed that contains 5000 seats. Beyond the shed are

(left) Pair of Ampex model 440B recorders handle tapes made in straight stereo (fed from mixer via Dolby-A units) and in SQ-encoded quadraphonic sound. Each is a quarter-inch, half-track model. Discrete four-channel tapes are made on a Tascam half-inch recorder; QS matrix quadraphonic tapes are made on a ReVox; cassettes are made on a Sony. Cassettes are not sent to broadcasters but serve as demonstration samples to "show our wares." Cassette unit also is used for interviewing.
(above) At opposite end of room, shelves hold rear-channel speakers plus usual assortment of spare parts, cables, hardware and so on. Note cartons of tape at right. Schedule this year calls for recording 25 full classical concerts and two full-length pop artist programs.



(below) In the rack are the audio delay unit and the Crown amplifier for sound reinforcement. Below Crown is AR amplifier and Technics by Panasonic receiver. These two units are used solely for in-room monitoring. Receiver, rather than simple amplifier, is used in order to get cue signals from local FM station WAMC. Units on top of rack are spares—one is a University amplifier; the other, an extra QS matrix processor. Panel below receiver houses special relay to signal announcer.

(right) Engineer Steven R. Colby adjusts module output level controls while checking VU meters on mixing board.

(above) Mixing console at Tanglewood is Interface Electronics series 100, modified by recording team to suit their own needs. Unit is sold as a 16-on-4; it is used here as a 12-on-4. Twelve inputs may be connected through modules to any combination (or none) of four outputs. Each module can handle high-level or mic signals; each module has four delegate buttons, four output channels, four meters, four gain controls. Any input can be fed and mixed into any channel. In normal set-up four-channel sound uses "around-the-clock" arrangement: left rear, left front, right front, right rear. For normal stereo dubbing, two front signals are used. Other modules on board handle main microphones, left solo mic, right solo mic, left and right chorus mics, and rear solo mic. Test equipment above board is used to generate audio tones to normalize modules. A 440-Hz tone is used for the Boston Symphony; when recording New York Philharmonic, a 500-Hz tone is used to avoid possible mix-up between the two orchestras. Board also has echo-send facility (first knobs just below delegate buttons) used for sound reinforcement set-up on lawn beyond shed. Mics pick up sound from stage; separate mix is made and sent through delay system (Gotham model 101) which feeds Crown DC-300 amplifier which in turn drives six Bozak columns installed in rear structure of shed facing lawn. Without this delay, "live" sound would reach lawn listeners ⅕-second before reinforced sound, causing choppy echo effect. Delay system introduces 190-millisecond delay in amplified sound. Result is that both sounds are perfectly matched, and psychoacoustic effect takes over so that although amplified sound is actually monophonic, "live" directional effects from performers on stage are preserved for lawn audience. This processing is completely independent of the mixes made for the broadcasting and the tape-recording.

vast lawns where thousands more listen, on folding chairs or blankets. To help the sound get to them, a special delay-and-amplification system is used. For most classical concerts this system handles from six to 12 watts of audio power; for pop concerts the power climbs to 200 watts. This sound is synchronized with the "live" sound so that no one is aware of the reinforced element in it. In the shed itself no reinforcement is used except for a spoken-word part in an orchestral work (such as the narrator in *Peter and the Wolf*) and, oddly enough, for a harpsichord part. The former sound comes from a bank of horn drivers high above the audience; the latter via a baffled speaker implanted in the front apron of the stage. Few in the audience are aware of the horn units; virtually no one knows about the harpsichord speaker. Levels for these special set-ups are constantly monitored during performances and the message "more delay" or "less delay" comes into the control room from one of the team who listens from the lawn area and gives his instructions via any of several strategically placed telephones. When the "spoken-word" horn or harpsichord speakers are used, two seats in Box 14 in the shed are preempted; one for the monitoring person, the other for the equipment.

In addition to the equipment shown here, Kaye plans this summer to add a Sennheiser shotgun mic (highly directional) strictly for harpsichord pick-up. A gadget the whole team is especially fond of is the Wahl cordless soldering iron which they use quite often.

The $900 Sony Turntable.

Study this page, because we don't want the price to suck you in.

It would be a shame.

People responding to something because it costs $900*. Not because it's worth $900.

People captivated by price, not performance.

We at Sony don't want anyone spending good money for a great turntable for a bad reason like an impressive price tag.

Especially because there's so much technology in the PS-8750 for you to fall back on. After you spring for the $900.

So before you spend a lot of money on us, spend at least a little time with us.

Total speed accuracy is our speed.

Speed accuracy can be a problem for turntables because the stylus continually puts pressure on the record (and, in turn, on our engineers.)

In fact, as little as one gram of stylus pressure can cause a slow-down in record speed. A slow-down that is particularly noticeable in loud passages.

Up till now, most good turntables achieved accuracy with a direct drive motor and a servo-system to control speed variations.

It was fine for most people. And it still is.

But for those with more elegantly attuned hearing, it's just not good enough.

That's because the servo-system will not serve when it comes to small, low-frequency speed variations. It is not sensitive enough, and the result is there to be heard—if you have the discernment to hear it.

To get around this, Sony took the conventional servo-system and 'evolutionized it by adding a quartz ference and a phase lock cuitry.

That mouthful is really easy to st. The stable quartz generator s a constant frequency. Any tions in speed monitored by the netic head are converted to ges in the phase of the signal. is then compared against the rtz generator's phase signal.

If they do not match, our Xtal-Lock corrects the speed variation instantly.

A conventional servo-system has to wait for the error to appear as a change in frequency, and then it takes time to correct it.

Sony can make the corrections 10 times faster. And within one cycle. All because Sony uses the phase difference as a source of information on speed error, rather than using the angular velocity.

Chart A dramatically illustrates the dramatic difference.

*Cartridge sold separately.

Why our tone-arm costs an arm and a leg.

After conquering the drive system, Sony sped along to the tone-arm. The problem: constructing a light, strong tone-arm that has a low resonance quality.

A high resonance quality means the tone-arm vibrates—performing a duet with whatever record is playing.

Sony wrestled with the arm problem and came up with a different material: a carbon fiber of enormous strength and equally enormous lightness. Moreover, it has a much smaller resonance peak than the aluminum alloy commonly used. (See Chart B, where the difference is demonstrated.)

The carbon fiber worked so well that it was even incorporated into the head shell of the PS-8750. But Sony didn't stop at the tone-arm's construction. Next came the actual operation of it.

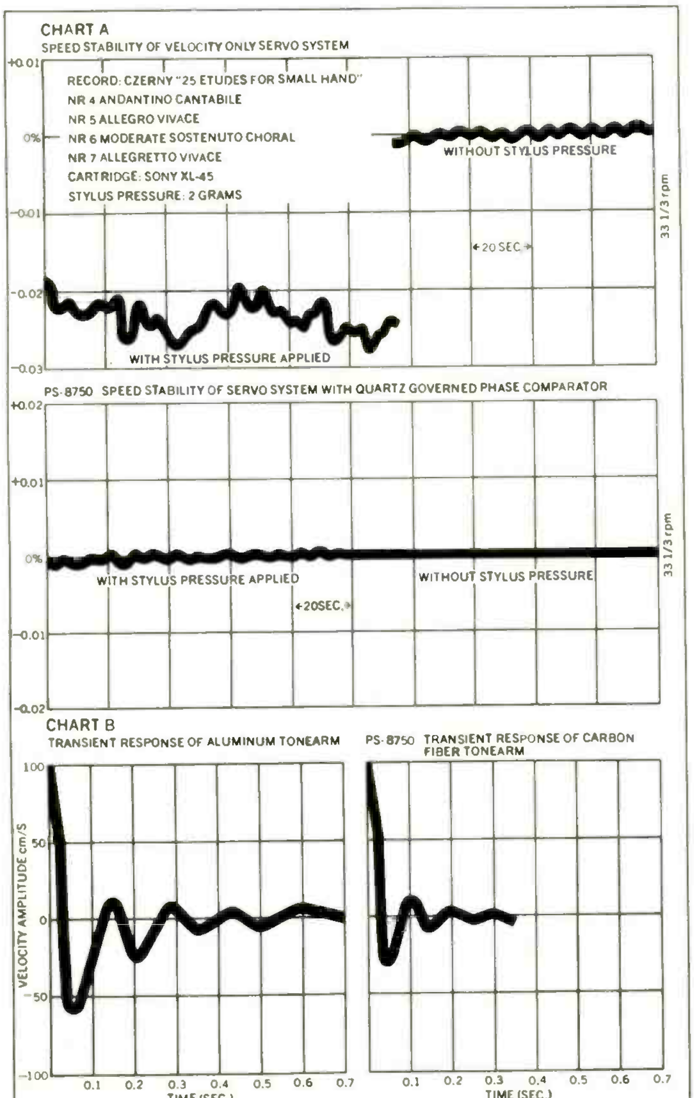
Most turntables have one motor, operating both the drive system and the return mechanism. Meaning that the turntable is linked to the tone-arm. And very often, this linkage produces a drag on the arm.

The PS-8750, however, proves that two motors are better than one. The motor that runs the tone-arm is totally isolated from the other motor that runs the turntable.

This eliminates the drag, particularly the drag at the very end of the record. This drag is really a drag, because the return mechanism is preparing to activate itself, and the friction is therefore increased.

Sony further innovates by designing pick-up and return cues that are optically activated. Like the doors in a supermarket, if you will.

With the PS-8750, you get the best of the direct drive manual and the best of the semi-automatic. With none of the worst of either.



Does your turntable give you bad vibrations?

The same sound waves that travel from your speakers to your ears also travel to your turntable.

This transference excites the equipment. Becoming acoustic feedback, or IM distortion. And the louder you play your record, the more of it you get. There's cabinet resonance. Caused by sound waves.

And there's something called record resonance. Caused by the friction of the stylus in the groove of a warped record.

Sony, however, deals resonance a resounding blow.

We have built the PS-8750's turntable base of an inorganic material that is acoustically dead.

We have also undercoated the platter with an absorbing material that prevents it from transferring any bad vibrations to the good vibrations on the record.

And we cut down on record resonance by pumping a silicone damping material into the record mat itself. By having contact with the entire record surface, it offers more support.

Not for people who want the latest. But the greatest.

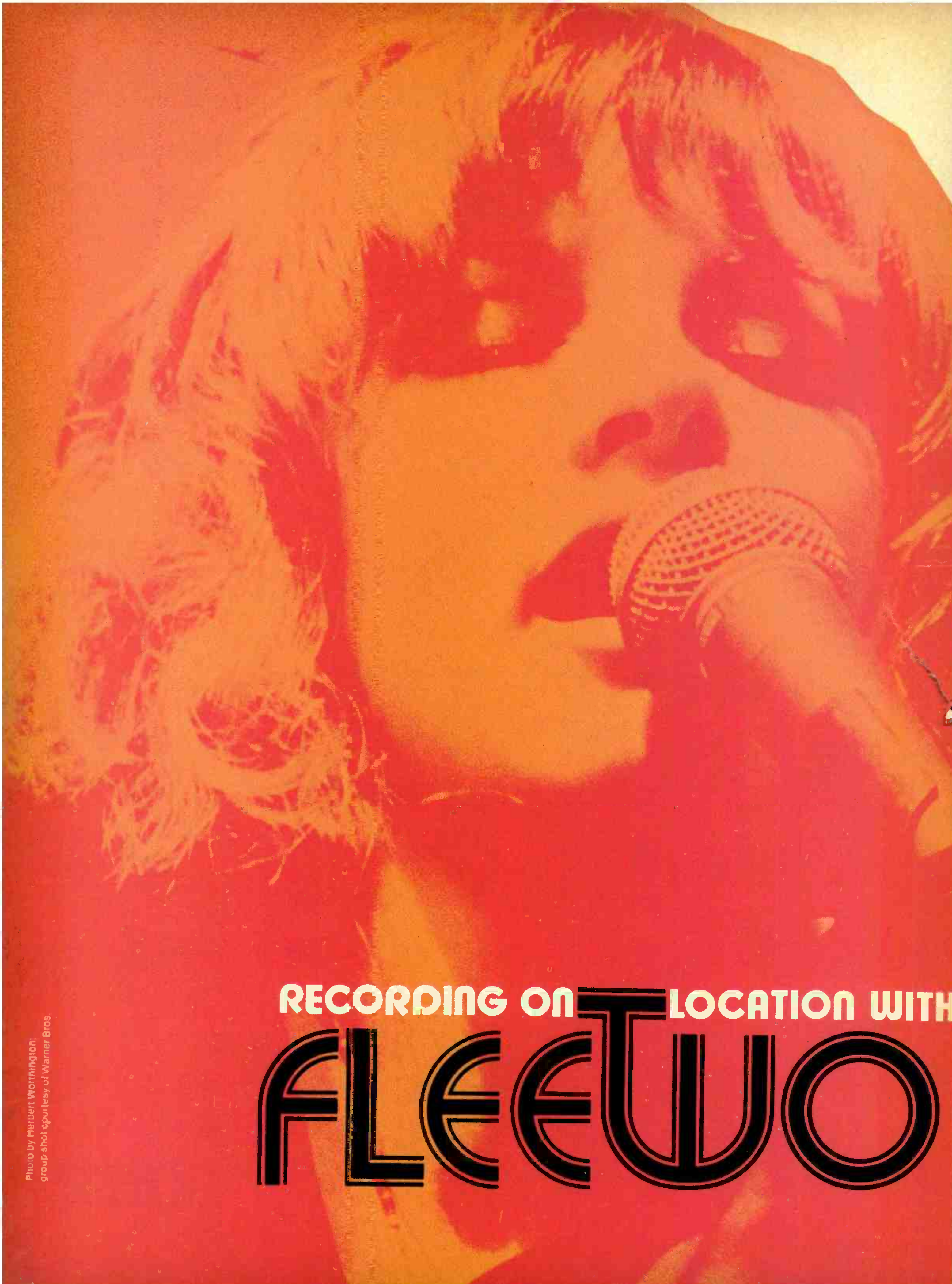
The PS-8750 represents a tonnage of innovation and a couple of real breakthroughs.

It is not for those who want to spend $900 so they can say they spent it.

It is for those who want to spend $900 so they can hear they spent it.

SONY®

CIRCLE 81 ON READER SERVICE CARD



RECORDING ON LOCATION WITH FLEETWO

Photo by Herbert Worthington; group shot courtesy of Warner Bros.

Gridiron Mac —or, two ways to record a concert in a football stadium

I can recall very few things having caused me to wake up before 6:00 A.M. Birds even sleep in late in California. Oh yeah, the Apollo moon walk and Watergate got me up early. And this assignment: A remote concert recording of Fleetwood Mac with the inimitable Heider bunch.

I'm now terra-planing to Santa Barbara on the freeway at 5:30 A.M. In the back seat, my Nakamichi 550, two AKG condenser mics, a stage pass; in my lap an Egg McMuffin and a styrofoam cup of hot, horrible coffee.

Voted by the recent Billboard Talent Forum as "Breakout Group of the Year," Fleetwood Mac was going to perform the last gig in a series of successful concerts. Earlier in the week they had played in front of a Bill Graham event in Oakland for a crowd of 100,000. The commissioning of the Heider Remote Recording group was to document this last show for remote recorded material and sound tracks for a promotional film. In the football stadium of the University of California, Santa Barbara, to an audience of 15,000 loyal fans, Mick Fleetwood, Lindsey Buckingham, Miss Stevie Nicks and Chris/John McVie would reaffirm to everyone why their recent album has been on the charts 48 weeks . . . and I was to check out this interesting remote recording event and report on what happened.

Intended for taking verbal notes of the professional recording set-up, my Nakamichi 550 unexpectedly got its first acid-test for recording music on this trip.

After pacing back and forth several hundred times between the stage, the sound reinforcement tower, and the Heider truck to take verbally recorded notes of the set-up and equipment, I decided to relax and enjoy the concert and occasionally check in on the remote truck where Ken Caillat would tackle the pro-recording job. (More on Ken and that later.)

Stage left was about as good a position as any to open up my mics, and so I sat comfortably atop an empty drum case ready to record my own mini-version of what was going down in Heider's "Studio on Wheels."

The Nakamichi 550 I used has three inputs: L, R and "Blend," the latter feeding a composite mono feed to both recording channels. Since I only had two mics and no recording/mixer flexibility except the input controls to ride gain, I either had to hold two microphones (one in each hand) or opt for a single mic fed into the "Blend" input. Needing one hand free and after fumbling around with various combinations of clutching both mics in the other hand, I decided to hand-hold just one mic and go for the stereo-blend mono input. There were no mic stands

OD MAC

By
R.A. Neilson

with me and all others were consumed in the stage set-up. Cables weren't long enough, and a myriad of other very practical reasons told me to shoot for it in the simplest possible manner.

The L and R gain pots on the Nakamichi are ganged to work quite well together with a thumb and forefinger controlling motion, so I could have mixed stereo levels to some degree (in lieu of a missing mixer). But, again, common sense of the moment dictated the practicality of recording mono.

Close to the Action

As close as I was to the "action" and yet unable to get useful stereo separation by mic placement, my final rationalization was that a good mono recording is just as good as a stereo one. Don't know if I really believe that, but it fit the occasion.

Being ten to fifteen feet from the onstage monitors for the performers, a wall of 120 plus-dB sound would necessitate the use of the built-in limiter and an accessory mic pad—which I had left at home—for my AKG 451. As it turned out, the incredible sound pressure levels in my onstage position required a setting of the gain control to nearly an "off" position. Although the peak-reading needles almost pegged a couple of times on heavy bass transients and piercing steel heavy-metal, I was able to catch most level variations and come up with a very decent, and very clean last-minute recording of the concert.

Lacking the headroom of a professional recorder, the Nakamichi in combination with the AKG mic and Maxell UD-XL 90 cassette did a phenomenally good job of recording good music in a less-than-optimum environment. But that's what remote recording is all about. In a limited way, I discovered some of the problems the remote crews face routinely—getting those studio-quality tracks in the middle of an acoustically compromised, hectic maelstrom.

Some additional notes on this recording experience:

The built-in peak meters are great—yet I wish they were switchable from RMS (VU) to peak reading. Although a little heavy when packed on your shoulder for six hours, the wide, beefy strap assembly is ergonomically superb. The "Nak" has a nice, quiet headphone amp (I also had left my headphones at home), but no monitor speaker. There's plenty of room inside to mount one, however, as I found out when removing three hundred screws and the labyrinth packaging to get to the head azimuth adjustment.

My Nakamichi is the third semi-pro machine I've used that suffered from factory-fresh, imprecise head alignment. Maybe German and Japanese ears hear things differently? At least their checkout quality control equipment doesn't agree with my ears. At least 3 dB of improved high-end response was gained by tweeking the head adjustment screw. Blame it on shipping vibration. By the way, the Heider engineers spend at least one hour on every remote aligning their big machines—even the Ampexes go out after bumping along rhythmic tar-strip road oscillations.

Now a word about the AKG microphones I used for my impromptu recording of this football-stadium concert-in-the-round. The 451 is an excellent remote condenser. I found out that not only is it a good idea to have the accessory windscreen, the 10- or 20-dB mic pads would have come in handy, too. The output of these mics is so hot that you'll overload and get diaphragm distortion if you don't use the pads judiciously.

The 451 had a built-in, high-pass roll-off circuit and switch that is necessary to utilize on remotes. Wind-noise in particular necessitates either the 75-Hz or 150-Hz setting (as opposed to the "flat" normal studio position for miking such things as acoustic guitars).

With the Maxell UD-XL tape in combination with the integrated Dolby circuitry (if it's aligned properly, a built-in 400-Hz oscillator takes care of this), good recording friends will be hard pressed to differentiate between a Revox at non-Dolby 15 ips.

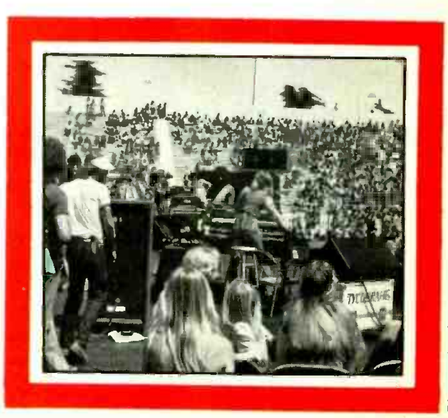
Needing headphones (for appearance, if not for necessity), I borrowed a pair of Sennheiser HD414's from Andy Bloch, who is manager of Heider's crack remote team. Supreme irony it was: the headphones had an annoying intermittent connection. Seemingly not designed for the rigors of remote sessions, the 414 "cans" are the kind you place gently on the music stand after the string ensemble ends the overdub take.

This is starting to sound like a product review. But why not? The gear that you use on a remote is easily half of the story on a dynamite track—or the whole reason for an abysmal and sometimes costly fiasco.

The Pro Side

That's the amateur side of things. Now let's get into *How the Pros Handle It.*

Tycobrahe, a large West Coast P.A. company, handled the sound reinforcement. Someplace near where the quarterback would call a huddle, a skeletal and precarious (although solid)-looking multi-tiered scaffolding stood. Above the sound-mixing tier, a 16mm Arriflex was loaded and ready to shoot the whole panorama. Clear-Com headset packs provided communication to the T.D. and roving



Photos by R.A. Neilson

camera—its Angenieux zoom lens poking into flying fingers and catching aura sun halations of rock stars' heads. Back down to the soundman, he had a Tycobrahe 24-input mixing console with all inputs plugged, a stereo tape machine for some background Rock Muzak and other things, an analog delay line (CCD's versus Digital shift registers), a power-line monitor, and a can of beer.

Strapped securely to the tower were two of the four ambience audience mics used by Heider for screaming fans. Shure SM56's. Good all-purpose dynamics. I've heard of a demonstration that Lou Burroughs does with Electro-Voice mics where he drives nails with an EV and then proceeds to show that the mic still works. I've never seen it with an SM56—but then, again, who says the recording team has to erect the stage? These mics, along with a stereo P.A., feeding to the Heider truck parked behind the stage, were neatly cabled under a rubber runner mat stretched to the stage for protection of these critical lines.

Two of Heider's crew, Mike Carver and Andy Bloch, crawled atop speaker stacks onstage to position the remaining two audience mics. Neumann U-87 condensers. These guys don't fool around. Standard operating procedure is to use four audience mics. These were hooked up to multed inputs on Heider's 40-input API board as "Audience L & R."

I had one mic to record my version of the event. The Heider mic plot I reviewed had 35. That's at least one fundamental difference between a semi-pro do-it-yourself and the pro arrangement. Also, with often double the number of mics used in a studio date, the potential problems with which the remote engineer has to contend are even greater.

The full 40-input API studio board in the remote truck, with three-band equalization, quadraphonic pan-positioning, and 36 meters to watch, is a necessity for achieving the fine degree of artistic control that puts down valuable tracks on that two-inch master tape.

Positioned onstage a Sony CCTV was Big Brother to the man in the booth: Chief Engineer on this job, Ken Caillat. Ken has been with Heider six years and did the superb engineering on Joni Mitchell's *Miles of Aisles*, to mention just one. Eyes glued to this video monitor in the truck, and ears attuned to the 604E Altec monitors, Ken would watch for the subtle nods of musicians' heads and accordingly adjust the levels and equalization throughout the entire day's recording work. In addition to the visual contact the video monitor would provide, Mike Carver was to hang out on stage and be in direct contact with Ken in case a mic went, or Ken couldn't see what was happening.

Prior to the day's concert, Ken had met with the group to review the order of songs and take special notes of any last-minute changes that he would have to watch for during the day's recording. His memo pad was filled with the stage layouts, mic complements and plotting, channel assignments and little notes to himself.

This pre-show preparation is generic to the remote engineer's job. Too many things can happen during the course of the normal, real-time recording work. Not to have reviewed and "rehearsed" the equip-

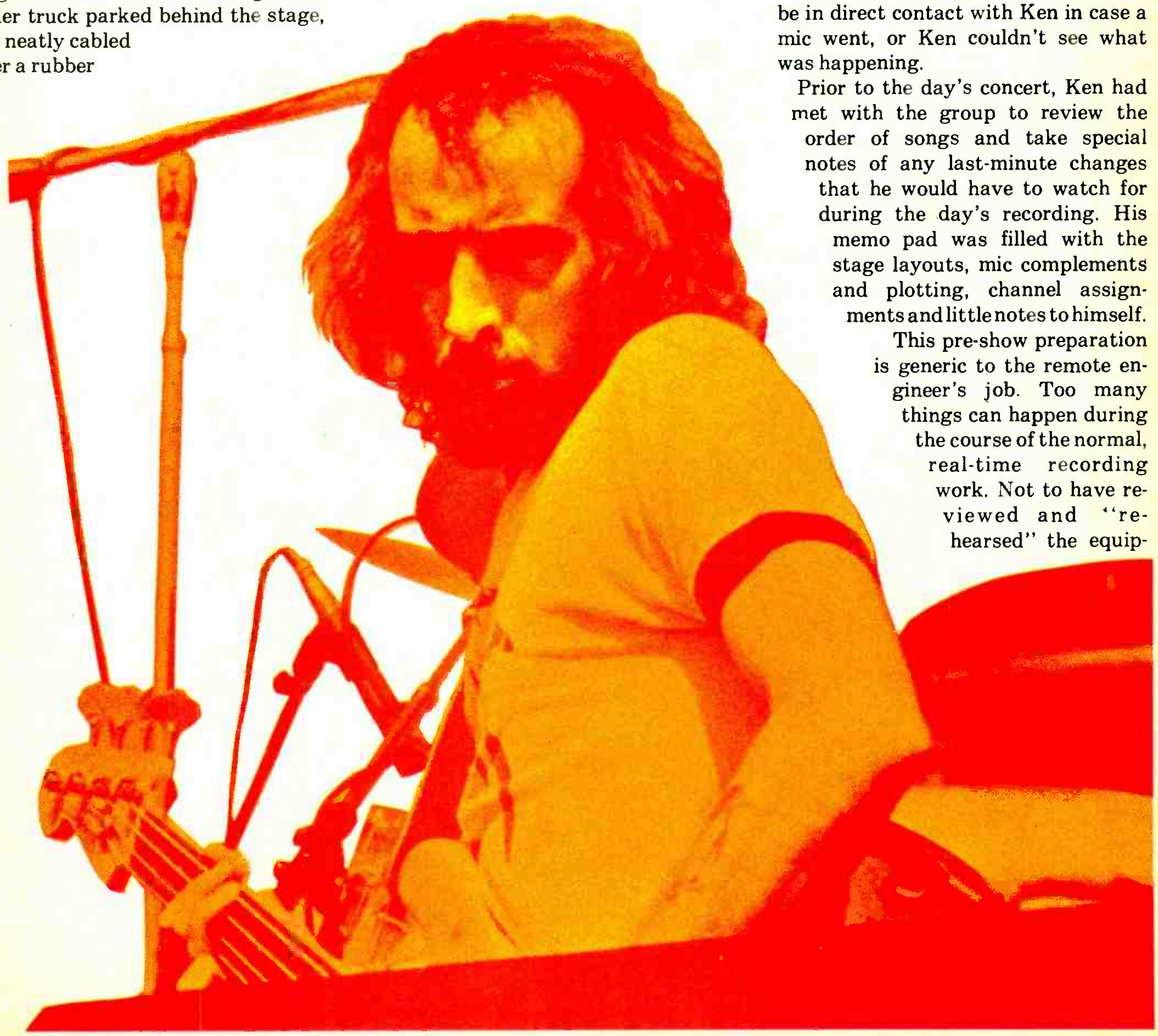


Photo by Herbert Worthington

ment and channel/mic logistics maps would be akin to General Patton sending out the troops without helmets, boots and walkie-talkies.

Now that we've made it to the stage, see the box on this page for the complement of mics and direct inputs used for recording the group's encore set.

All of the inputs terminated onstage into a "Mult-Box" where P.A. and recording lines would meet and simultaneous feeds could be achieved. From this backstage box, a long line would snake its way across the football field to find the Heider truck.

Tape's rolling! The Ampex 24-track machines had been readied and were now gobbling up 1400-foot reels of tape. Two machines were used—with the second one ready to begin as the first one ran out of tape.

"Have mercy baby on a poor girl like me,
You know I'm falling, I'm falling, I'm falling
At your feet . . ."

ON KEYBOARDS:

Arp 2600	direct input
Hohner Piano	direct input
Fender Rhodes Piano	direct input
Acoustic Piano	2 U-87 Neumann condensers & Helpinstill direct pick-up
Leslie Speakers	2 Shure SM56's for top & bottom speakers
Keyboard amplifier	1 Shure SM56

ON GUITARS:

Electric	Amplifier/ 1 Shure SM56 & 1 AKG 451
Ovation Acoustic	direct

ON BASS GUITAR:

Amplifier	1 Shure SM56
Direct	(high & low mixed) stereo
Alembic bass	

ON DRUMS:

Kick	1 Shure SM57
Toms	4 Shure SM57's (2 left & 2 right)
Snare	1 AKG 451
HiHat	1 Neumann KM84
Overhead	2 AKG 451's
Gong	1 Shure SM56
Conga	2 Shure SM56's (multed)

ON VOCALS:
All Shure SM57 mics

Photo by Herbert Worthington

Fleetwood's on. Pretty exciting after the warm-up act and nearly eight hours of anticipation for the Big One. Dwarfed by the 24-track machines, the miniscule Nagra stereo recorder which was recording the sync-film sound stereo feed, almost looked like a toy. Some toy at $3000!

Some rock acts, quite understandably feel a little intimidated and nervous about all the extra people, gear and preparation that's necessary for a remote session. Judging from their performance, Fleetwood appeared at ease regardless of the differences between "live" and studio recording. When under the additional pressure of the red light and unforgiving tape, a "live" audience can be an extremely effective vehicle for transporting the last ounce of artistic juice into the air—likewise, the concertgoer is hearing his favorite entertainers in a well-tempered mode. Moneysworth. Both know a good performance is in order.

Solar Guitar Tuning

Meanwhile, back at the board, Ken has his hands full: 35 inputs all cooking. And the challenge begins on a set with increasing problems. Everything goes okay in the first number until near the end, when the instruments, quite noticeably, start going severely out of tune. Seems this incredibly beautiful, bright, sunny day is too good to be true. Great for the 10,000 tanned, half-naked, frisbee-throwing crowd, but not so cool for the instruments. The sun is beating down unforgivingly on resilient drum heads and sensitive wooden guitars and causing havoc—enough for Fleetwood Mac's Lindsey Buckingham to nervously comment on it at the end of the first tune.

Reflecting on that problem after the gig, Ken commented, "I was spending most of my time keeping the EQ tight. Next time, those instruments are going to sit on stage in the sun for an hour before the show." Ken had worked out everything with Fleetwood and the recording layout. But not this situation.

Up on the stage, the monitor amplifiers had carefully taped expanses of Reynolds Wrap foil covering their sun-exposed flanks—a clever, improvised detail that was attended to early in the equipment set-up. It was obvious that the amps would bake in the hot sun if not covered; unfortunately, there wasn't much that could be done for the instruments.

Watching from the rear of the truck at times, I saw Ken making wide EQ adjustments to keep the sound even. "In a 'live' mix, a 2-dB cut or boost can change the whole mix." In that everything was tonally changing quite drastically, it seemed a moot point. However, Ken's professionalism and quiet manner took the challenge and came through. The soundtrack will be used by Warner Bros. on behalf of international Fleetwood Mac promotion.

"Anything you can do in the studio you can do 'live'. . . . You can lose a lot of feel in the studio . . . plenty of time . . . sometimes too much. The musicians refine and re-refine until the spontaneity and magic goes. . . ."

Couldn't agree more with you Ken. The luxury of creative hours in the studio aren't there on a "live" remote concert recording. You get it while it's hot or you drop back thirty yards and punt.

Long echo delays, short variable delays, double voicing, short slap-back echoes, hard reverberation, flanging, true vibrato, FM modulation. All this in the MXR digital delay system. Who else could do it?

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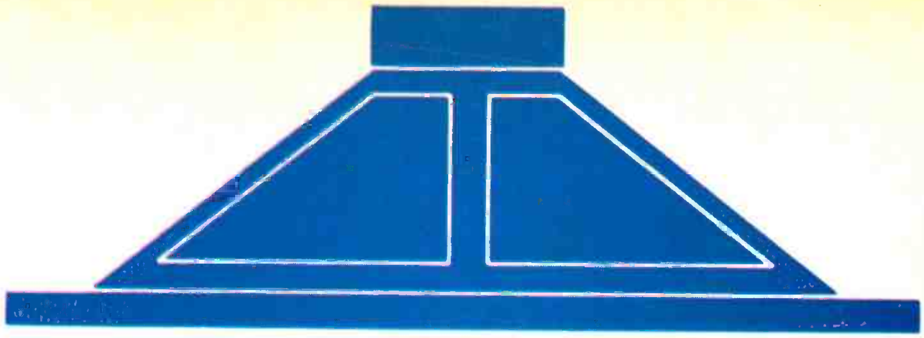


P.A. PRIMER

A GUIDE TO SOUND REINFORCEMENT

PART II

Introduction

Part 2: *Part 1, appearing in the previous issue (June/July), consisted of basic information and rules about sound, speakers, horns, bass boxes and crossovers. Part 2 was originally conceived to complete the Primer with a presentation of practical answers to choosing the components that make up a sound system. But we soon realized that the subject required more space than we had anticipated. A hasty consultation with the editors determined that space limitations should not restrict coverage of all the necessary topics. Consequently, this part will be followed by a concluding part in the Oct/Nov issue.*

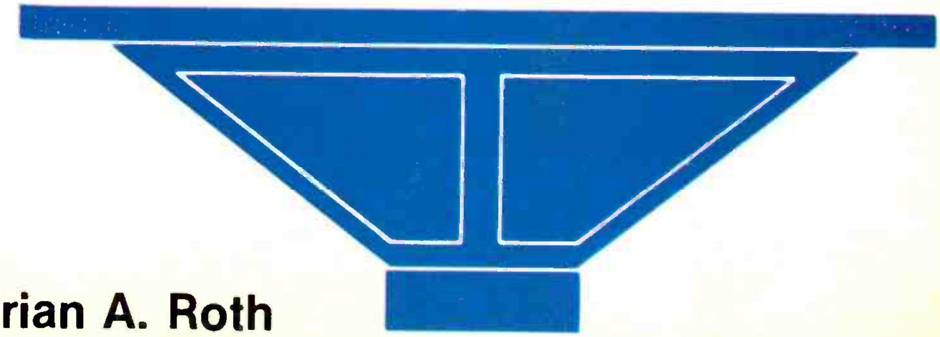
MR's P.A. Primer covers many areas and is intended to give general explanations and guidelines. The business of concert sound reinforcement is gigantic and complex; however, with a basic understanding of the problems, the reader can become involved in a very exciting aspect of the music business.

The photographs of equipment illustrating this article have been chosen by the authors as representative of some of the high-performing gear available on the market today.

How to Choose a Mixer

Every year there seem to be at least 15 new models of sound-system mixers available to the public. They are covered with knobs, flashing lights, buttons and numbers, and come in any color desired. Although the designs differ, the mixer has a certain job to do, and it must be evaluated on how well it performs.

Basically, the mixer takes a group of inputs (microphone-level or line-level), processes them (changes tone, reverb), and then mixes them and subdivides them into outputs to be sent on to the main system and monitor system. A simple mixer would have six or eight input channels and each channel would have (a) a volume control, (b) a bass and treble boost and cut control, (c) a reverb-send control, and (d) a monitor-send control. The mixer may also have a master volume control, master reverb-send and -return control, and a master monitor-send control. This mixer would have three outputs which are the main P.A. output, the reverb-send output, and the monitor-send output. If the board were stereo it would have on each input a stereo pan-pot (a control that moves the apparent sound source back and forth between the left



By Jim Ford and Brian A. Roth

Yamaha PM-1000/32 Console

and right speakers) which would mean there would be two main P.A. outputs.

Now there are numerous boards available with more inputs, outputs and gadgets. For example, standard P.A. mixers are equipped with from four to 32 inputs, up to eight separate outputs for the main system, two to four reverb-send outputs, two to four monitor-send outputs, mono, stereo or quad mixing ability, and separate recording outputs. The tone-control section (equalizer) may be divided into four bands and can be adjusted as to boost, cut, frequency, and shape of boost and cut curve. Other mixers offer built-in graphic equalizers, crossovers, limiters, hi-low filters, solo buttons and tone oscillators.

Now all that this proves is that some company somewhere makes a mixer that would appear to do "everything" that is possibly needed! However, the quantity of knobs and outputs does not mean that the mixer is well-built and will do the job correctly. It is a difficult problem to determine which product is the best buy (lowest cost, largest number of functions, highest-quality construction, greatest reliability, and lowest cost to repair).

A P.A. mixer should be checked and evaluated by the following:

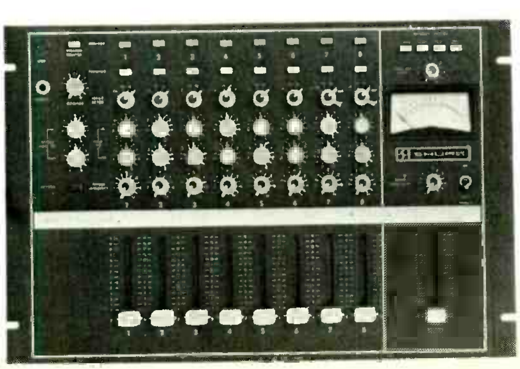
(1) **Number and Type of Microphone Inputs:**

(a) **High- or Low-impedance Mic Inputs?** High-impedance mic inputs are unacceptable in almost any type of quality or professional concert sound work. High-impedance mic lines should not be longer than 20 feet, at which time there is a noticeable loss of high frequency. High-impedance mic lines are also noisy and receptive to picking up radio frequency interference (radio stations, CB's, etc.). Low-impedance mic lines should be used whenever possible. They can be run for hundreds of feet without any degradation of the signal, are low-noise, and are immune to most outside interference.

(b) **Do the Mic Inputs Have Transformers?** A quality mixer will probably have low-impedance mic input transformers. Although transformers are one of the lower-quality elements in the audio mixer, in the mic-input position, it serves a very good purpose. Used in a "balanced" configuration it insures that the mic lines will have the best noise rejection. The transformer also has a voltage step-up, which means that the signal-to-noise ratio of the mixer input amplifier is much better. These two advantages usually outweigh other shortcomings of the transformer, and, therefore, a good mixer will have low-impedance, balanced-transformer, isolated-microphone inputs. Unfortunately, transformers are expensive, so beware of cheaper-quality mixers that advertise low-impedance mic inputs that are missing the transformers. In a big system with a lot of long mic lines and complicated connections, this type of mixer could present big problems.

(2) **Does Each Mic Input Channel Have a Mic-preamp Gain-trim and/or Mic-input Attenuator?** One problem that exists with trying to reproduce "live" music is the extreme dynamic range of the sound. A mixer may have to amplify a low-level violin or a high-level rock guitar amplifier, and this requires a very large range of mic-preamplifier gain. The very low acoustical volume of the violin means that the mic preamplifier must have a lot of gain and must be very low-noise. The very high acoustical volume of the guitar amplifiers means that the mic preamplifier must be protected against being overdriven and distorted. This large dynamic range cannot be handled by one mic-preamp gain-setting, so consequently, there needs to be a method of adjustment.

A mic-preamp gain-trim changes the mic-input sensitivity, and this works very well for adjusting the gain exactly to the input signal strength. However, this gain adjustment is usually limited to about 20 dB. For controlling larger dynamic ranges the mic gain-trim should be used in conjunction with a mic-input attenuator (called an "input pad"). An input pad is a resistive network that reduces the incoming voltage and protects the mic preamp from overload. Usual amounts of attenuation are 10, 20, 30, and 40 dB. Also, if the mixer is designed properly, the input pad will be a balanced network that precedes the mic input transformer. In this position, it will



Shure SR101 Audio Console

Tapco 6000P Six-Channel Mixer and Pre-amplifier with Reverb

protect the transformer and mic preamp.

(3) **Does the Mixer Have Slide Pots or Rotary Pots?** In general, rotary pots are electrically far superior to slide pots. They offer lower noise levels and longer life. However, slide pots are more convenient, look better, and have more "sales appeal." A slide pot that is technically equal to a good $2.50 rotary pot will cost a minimum of $25, and slide pots on the better audio boards cost about $100 each. There are several mixers that have slide pots that actually turn a rotary pot, and these are excellent. This arrangement gives good looks, ease of operation, low noise, long reliability, low repair cost, and low initial cost. So, take a good look at the pots and do not waste money on cheap sliders!

(4) **What Kind of Equalizer Is Needed?** First of all, an equalizer is a fancy name for a tone control, and the main function of the unit is to contour or shape the level and relationship of the low, mid and high frequencies to a desired sound. Simple equalization may be done by just a bass and treble control, or complex equalization may be done by very narrow frequency bandwidth controls that act on only small segments of the sound spectrum. Although some mixers have complex equalizers, it is best to have simple equalizers with broad frequency bandwidths on the input channels. Complex third-octave and narrow-notch EQ should be left as outboard equipment. On each input, an equalizer with three sections is adequate (low, mid, high). The low section should operate in the 50 to 200 Hz region. The high section should operate in the 3,000 to 15,000 Hz region. The mid section should be centered between 500 to 3,000 Hz, and its area of operation should be broad and overlap into the low and high areas. Equalizers with peaky or narrow bandwidths are not desirable for tonal changes because they act on too small of a frequency range to be effective. The result of the use of narrow bandwidth equalizers is to cause peaks in the response that start feedback and place high demands on amplifier and speaker power requirements.

One of the major mistakes made by young sound engineers is the excessive use of equalizers—especially third-octave graphic equalizers. Remember, a 3-dB boost requires that the power amplifiers and speakers must be capable of twice the power, and a 6-dB boost means four times the power. Extreme boost usually results in distortion and blown-up speakers and horn drivers. Consequently, all equalizers must be used with knowledge and care if good results are to be obtained.

(5) **Is There a Separate Echo-Send Mix?** Some mixers have a built-in spring reverb with a push-button to activate it, and one volume control to set the reverb level. This is not very flexible, and if the mixer is low-priced, the reverb unit is probably of the worst kind. Most good mixers will have a separate set of controls (one on each input) to get a reverb mix. Usually, there will not be a reverb unit in the mixer, and the actual reverb chamber will be an outboard unit of good quality. The set of reverb-send controls will allow an exact mix to be made of the amount of reverb on each mic. This mix is then sent out of the board (reverb-send output) to the reverb chamber where it is "reverbed," and then back to the board (reverb-return input) to be mixed in with the final P.A. output. There is a reverb-return pot that mixes the right amount of the reverb signal in with the main P.A. mix. If reverb is to be used, this is the correct method, although in most big systems the room is so large and reverberant that reverb is rarely used. In a small, "dead" club, reverb may be a great advantage.

(6) **Are There One or More Separate Monitor Mixes?** Technically, the monitor mix looks identical to the reverb mix. There should be one pot on each input. These pots combine together to provide an independent mix to be sent to the stage monitor system so that the performers may hear what they wish. As far as many performers are concerned, this is the most important part, and for that reason nearly all big systems have a separate monitor-mic mixer on stage. The stage mixer will usually be just as big and complex as the main P.A. mixer. Some stage mixers may provide four or more independent stage mixes for the individual performers. Of course this is very dif-

Teac Tascam Series 5 Mixer

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Mu-tron Micro V is operated from a single 9-volt battery. A jack is supplied for operation from a standard 9-volt battery eliminator.

The unit is physically very compact and is every bit as rugged and reliable as you would expect a Mu-tron product to be. And, of course, Mu-tron Micro V is covered by the same three-year warranty.

So if only the best will do, it's Mu-tron III — the original.

But if you'll settle for second best, check out Mu-tron Micro V.

8304

CIRCLE 68 ON READER SERVICE CARD

Interface Model 104-16x4A Mixer

ficult to control, and knowing where feedback is coming from is sometimes difficult to determine.

The monitor and reverb-mix pots should be checked to see where they get their signal from. The monitor pot is usually connected after the mic preamp (and sometimes EQ) prior to the input volume pot. This means that monitor signal strength is completely independent of the setting of each input volume control. By this method, changes in the main P.A. mix do not effect the stage monitor mix or volume. On the other hand, the reverb pot should get its signal after the input volume pot and equalizer. Thus, the reverb pot volume and EQ will follow any change made in the channel volume pot or EQ. After the basic reverb level is set on a channel, then any change made in the channel volume pot or EQ will make a proportional change in the reverb-send level. This is the proper and easiest method to use.

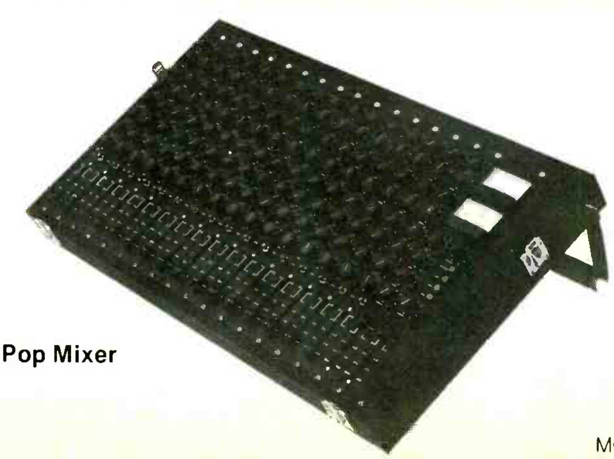
This part may not make much sense if the reader has not had an opportunity to follow a schematic diagram of a mixer, but this is necessary in order to be a successful sound engineer.

(7) **Does the Mixer Have VU Meters or Peak Indicators?** Any good mixer should have some method of checking the voltage levels in the unit from mic preamp to final output. For professional (and undistorted) work they are 100% necessary. VU meters respond to the average level of the program signal. They are a mechanical unit and, consequently, slow to react to peaks. The peaks come and go, and the VU meter does not give an accurate view of what is happening. Not realizing how high the peaks are can lead to overloading the mixer, crossover, power amplifiers and speakers. Of course, this means distortion. In the past, engineers have been aware of this problem and have designed "headroom" into most audio equipment. "Headroom" is the amount of safety margin above the "0" reading on the VU meter. A "live" sound mixer and a recording mixer should have a minimum of 20 dB between the "0" VU mark and overload, thus allowing all the peaks to be reproduced without distortion. (Note: 20 dB of headroom should be allowed throughout the entire sound system, including all electronic equipment, amplifiers and speakers. This means that the average operating level of the system is 20 dB below the maximum output of the system, therefore insuring low distortion.)

In recent years, peak indicators have been added to mixers and are extremely useful to the sound man. When a music peak hits the maximum output, a light flashes to warn the sound mixer. This is the best way to set mic-preamp gain and also an excellent method for checking the maximum output of a power amplifier. Peak indicators have an additional advantage in that they are all electronic and extremely rugged, unlike a VU meter.

(8) **How Many Outputs? What Kind?** As stated earlier, a mixer may be purchased with any number of outputs. The important thing is to determine what your needs are and what functions the mixer is to perform. Should the mixer be mono or stereo? Most systems today are mono. Stereo or quad systems have great difficulty getting the same sound mix to a large amount of people in a big room. Consequently, most systems are mono, but does that mean the mixer has only one output? Most big mixers have four outputs that are selectable at each mic-input channel. The sound engineer will probably subdivide all the vocals and instruments into four groups. For example: (a) output #1 is lead vocal, (b) output #2 is back-up vocals, (c) output #3 is lead instruments, (d) output #4 is the bass and drums. Of course, there are endless ways to subdivide, but the result is that the sound man can better control the mix.

What are the other standard outputs? If reverb is needed, then there should be a set of reverb-mix pots and a reverb output. If a stage-monitor mix is to be made at the board, then a set of monitor-mix pots and a monitor output is necessary. If recording is to be done during a "live" performance, it is necessary to have an output that is independent of the main output. A constant level is needed to go to the tape recorder, while the volume level to the power amplifiers on stage may need to be changed up and down throughout the performance. These



Allen and Heath Pop Mixer

BGW 500D Power Amplifier

two outputs need to be isolated so that the engineer running the tape recorder cannot make a mistake and cause problems in the main sound system.

Technically, the type of outputs should be low impedance, balanced, transformer-isolated. They should not be high impedance for the same reasons as listed previously. The outputs should be able to drive a 600-ohm load to a level of 20 to 24 dBm. The isolation transformer will help eliminate ground loops which cause hum and buzz (which makes the sound engineer look bad).

(9) **Overall Specifications? Other Indications of Quality?** All quality mixers should be low-distortion, low-noise, wide-frequency bandwidth and linear. Below is a list of minimum performance specifications:

(a) Distortion: less than 0.1% total harmonic distortion and less than 0.1% intermodulation distortion.

(b) Noise: buss levels at least 60 dB below "0" VU—equivalent input noise at least −120 dBm.

(c) Frequency Response: ±.5 dB from 100 Hz to 15 kHz and ±2 dB from 20 Hz to 20 kHz.

(d) Power Bandwidth: full output should be available at 20 kHz to insure good transient response.

A mixer should also be checked for quality construction. Heavy-duty metal construction with military-grade components will give a good indication that the mixer is well-built. An abundance of patch-points on the mic channels and output channels also illustrates that the design was probably carefully planned.

Power Amplifiers

Power amplifiers are much easier to choose because there are fewer variables to consider. The perfect power amplifier would not change or color the sound signal but would only amplify it. Electronically, the power amplifier needs to meet the following minimum specifications: distortion less than 0.1%, noise at least 80 dB below maximum output, frequency response from 20 Hz to 20 kHz ±.5 dB, and power bandwidth from 20 Hz to 20 kHz. If the amplifier meets or exceeds the above specifications, then it must be evaluated for quality construction and reliability. Once again, look at the grade of components and construction inside. High quality on the inside probably means that it is a good unit.

For a power amplifier to deliver high power to a load (300 watts and up), it must have a large power transformer and big heat sinks (except for several new configurations employing different design concepts that allow smaller and lighter component parts). Good components, big power transformers, and large heat sinks mean big money. A 300-watt-per-channel (into 4 ohms) amplifier will cost about $800, and a 150-watt-per-channel (into 4 ohms) amplifier will cost about $490. Beware of the salesman who tries to tell you that his amplifier will produce 600 watts and only costs $450. It may produce 600 watts at 50% distortion for five minutes before it blows up!

As for reliability, all amplifiers may blow up, but some blow up less than others. Do not buy an amplifier that is not output-protected. The amplifer should (a) be able to withstand a dead short across its output at full power, and (b) be capable of driving reactive loads without consequence.

After an amplifier is chosen, it must be operated correctly. If its minimum output load is 4 ohms, do not load it down beyond that point. It should also have proper ventilation for heat dissipation. If the amp is mounted in a rack with other amps and electronic gear, there will need to be adequate space between all the units, and a fan may be necessary to achieve ample flow of air. Many times, all works well until an outside job on a hot summer day is attempted. If the amps are in the sun they will probably overheat and "thermal off" in a hurry. This problem can be helped if all of the amplifiers are not loaded down with their maximum loads. Running with 8-ohm loads on the amps will almost always insure that they will not overheat.

Finally, choose amplifiers that meet the requirements. Do not put a 300-watt amplifer on a 30-watt tweeter. Even when the volume control is turned down, the amp is still capable of delivering full power if the wrong input signal is accidentally applied. An

Crown DC300A Power Amplifier

dbx 160 Compressor/Limiter

Urei LA-3A Audio Leveler (Limiter)

amplifer's power rating is given at a rated distortion value, and if the amplifier is pushed, it will put out much higher power. For example, an amp that produces 180 watts into 8 ohms at .01% distortion can produce about 360 watts when it is pushed to the maximum output that its power supply will deliver. At this point the distortion rises greatly, and the audible sound would be terrible. Still, if an accident occurs and places the wrong voltage on the amp's input, the result can be many destroyed speakers and drivers.

Electronic Crossovers

In Part I, many reasons were given for the use of an electronic crossover rather than a passive crossover. The electronic crossover should have the same high-quality construction and performance specification as the mixer and power amplifier. Its output impedance should be low enough to drive many amplifiers without difficulty.

Choose the crossover points to best divide the sound according to the frequency range of the speakers and horns that are going to be used. For a two-way system with a horn it will probably be between 500 Hz and 3500 Hz. For a three-way system using some type of a bass enclosure, a midrange speaker (probably 10″ or 12″) and a high-frequency horn, the crossover points should be approximately between 250 Hz to 500 Hz for the low and between 1200 Hz and 3,500 Hz for the high. To make the above into a four-way system using a super tweeter, an additional crossover point should be placed between 7,000 and 10,000 Hz. These crossover frequencies are standard, although the speaker components vary greatly from system to system. Beyond selecting the correct crossover points, the only other practical item to know is not to plug a low-frequency output into a high-frequency amplifier. Putting low frequencies into a horn driver will rupture the diaphragm in a very short time, and that costs money!

Compressors, Limiters and Noise Gates

Compressors, leveling amplifiers and limiters were once used only in recording and radio broadcast applications. Now, they are finding their way into P.A. systems.

These devices, whose above-mentioned names are often interchanged, allow automatic "gain-riding" of an audio signal. When used properly, these units minimize the amount of correction to volume levels that the sound operator must make to compensate for the dynamics found in music. They will greatly reduce "blasting" of the sound system by an exuberant scream from an excited musician, and make mixing of the program far easier.

In practice, the compressors or leveling amplifiers will give gentle, but firm, control of volume levels of signals connected into these units. They may be thought of as a very quick hand on the volume control; as the input signal increases in level, the electronically operated volume control will reduce its setting, helping to control dynamic levels. This is obviously useful on vocals, and optimally one compressor per vocalist should be used so that each one is individually controlled. Instrument signals can often benefit from compression, too.

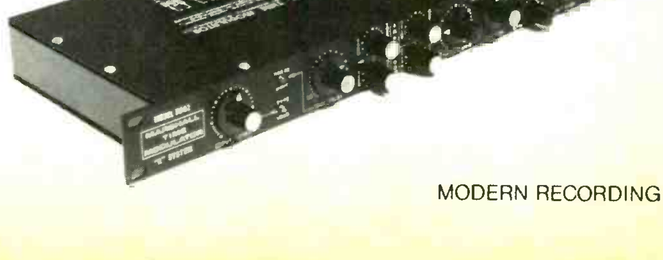
Limiters operate in a similar fashion, but for a different purpose. These units are for eliminating peaks in the program material that contribute little to the sound but can cause overdriving of the system amplifiers or loudspeakers.

Another dynamically controlled processor is the noise gate. This unit can be compared to the "squelch" circuit found on most CB radios. When no transmissions are being received, the squelch mutes the audio signal in the radio to eliminate the static normally present. When a transmission begins on the channel, the squelch turns the audio circuitry on again. The "squelch" control on the radio determines how strong a signal is necessary to trip the squelch.

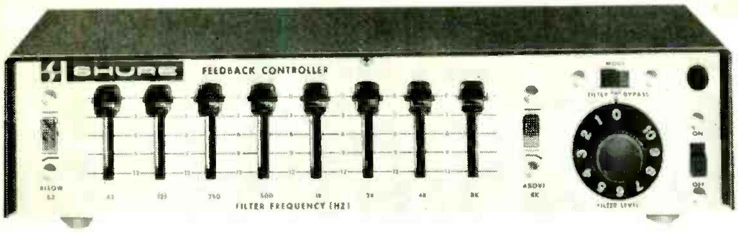
Professional audio noise gates are usually more sophisticated than those of a CB radio. Most have a threshold control (akin to the "squelch" control) and a release-time control that determines how long the unit takes to shut

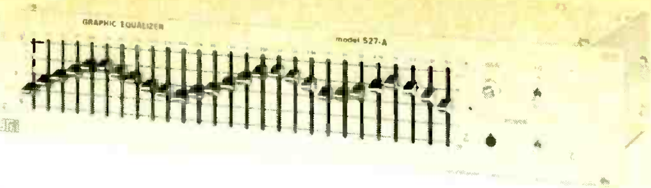
Orban/Parasound 621 Parametric Equalizer

Marshall Time Modulator 5002 "A" System



Shure M-610 Feedback Controller (Octave Equalizer)





Urei 527-A Graphic Equalizer

off the audio after the signal has dropped below the threshold established by the other control.

A noise gate eliminates the need to turn off a mic channel of a mixer when the primary signal being picked up by the microphone isn't present. This in turn eliminates peakage of other sounds received by the microphone during periods when the mic isn't being used. Consequently, a cleaner sound is usually produced by the sound system when sound leakage has been minimized with the noise gate.

One final comment: When checking for the feedback point of the sound system, be sure that the noise gates are keyed on, making all microphones "live." This will eliminate the embarrassment of a feedback howl when the noise gates are turned on by program material during the show.

Graphic Equalizers and Notch Filters

An equalizer is an accessory often used (and misused) in sound systems to correct for poor speakers or difficult acoustical conditions found in many rooms. The most common is the graphic equalizer which gives the user adjustment of individual frequency bands in the sound system. Boosting or attenuation of each band is allowed by five to 30 (or more) controls, each corresponding to a different portion of the audio spectrum.

While a graphic equalizer can improve the sound quality of a P.A. system and increase volume before feedback occurs, sophisticated test equipment is required to properly adjust the equalizer. Misadjustment can create more problems than are imaginable. If any boosts are applied to the system's frequency response, this requires that the amplifiers produce twice the power for every 3 dB of boost in that frequency band. This can put a strain on the horns and speakers, possibly causing failure of these devices under heavy-drive situations.

Another type of equalizer that has been gaining popularity is the tunable narrow-band notch filter. This equalizer is useful for "dipping" out feedback ring frequencies that plague all P.A. systems. In practice, this type of filter is adjusted as follows: the system's volume level is slowly increased until a feedback ring is heard. The frequency control on the filter is adjusted until it coincides with the feedback frequency. The filter circuitry will attenuate a narrow band of frequencies surrounding (and including) the selected frequency, reducing the offending ring. This procedure is repeated for other feedback nodes until all of the available notch sections in the filter have been utilized.

Digital Delays

Digital delay units (DDU) have been gradually decreasing in price, and consequently are finding wider applications throughout the audio industry. In addition to special-effects generation, these units can be used for correction of time-delay problems found in large sound systems. This becomes a problem in situations utilizing several arrays of speakers at a distance from the main stage sound system, such as is sometimes done to cover large audiences at outdoor music festivals.

This can be best understood by remembering that a sound wave takes time to move through the air since the speed of sound averages 1132 feet per second. Consequently, when multiple speaker stacks are used throughout the audience area, this time delay can cause disconcerting echoes as a listener hears the sound from a nearby loudspeaker and then from a more distant one. Since a DDU reproduces an audio signal after briefly storing it, a proper amount of delay can correct for the distance from the stage by delaying the audio signal to the remote speaker locations by an amount equal to the acoustical delay.

NEXT ISSUE: MR's *P.A. Primer will conclude with a discussion of speaker systems, mic technique, wiring, tips on power distribution and some true-life adventures experienced by all P.A. soundmen.*

Lexicon 102 Digital Delay System

Eventide 1745M Digital Delay Line

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CIRCLE 87 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

Me-Too-Ism in Audio

In the early days of the audio and recording industries there was a great deal of innovation and original research and development to be seen. In "live" recording work, there was no such thing as "standard" microphone placement. Each recording engineer considered himself something of an artist and mic placement was often the jealously guarded secret of each engineer. Many of us can remember the days when such individuality was actually overdone, leading to the need for dozens of playback equalization curves on home hi-fi preamplifiers before standard RIAA equalization was finally adopted world-wide.

The same situation held true when it came to design of audio hardware. Hi-fi equipment manufacturers encouraged innovation in circuitry as well as styling, and "next year's" prototypes were literally locked up in safes to prevent industrial spying. It was this attempt to be different from one's competitors that prompted the incredible growth of the industry during its earliest years and which resulted in a magnitude of improved equipment performance.

Now let's take a look at the present. More than a decade ago, Harman-Kardon developed the "blackout" tuning dial—that cosmetic treatment of the front face of a tuner or receiver which was promptly copied by just about every tuner and receiver manufacturer here and abroad. For a time, you had to read the small identifying logo on each panel to be able to tell the difference between one receiver and another, so closely were cosmetics and control layout followed. The sameness of style became so dull that finally consumers began to be bored. So a new format was developed, in which the dial area is now highly visible even with the power shut off, and numerals are now screened in dark colors against a light gold background. The moment this format appeared, other manufacturers followed suit almost as quickly as car makers added "opera windows" to their new automobile models.

Just a few years ago someone figured out how to stand a stereo cassette upright and the race for "front-loading" cassette decks was on. Does front-loading improve performance of the unit? Probably not. It does make for convenience in stacking components, but somehow advertising has lost sight of that simple advantage and front-loading capability has been made synonymous with high quality—even when it appears in the form of under-$200 cassette decks that offer mediocre measured performance and lack features of equivalently priced table-top units.

Nor has audio me-too-ism been confined to external appearance of products. No prominent manufacturer of audio amplifiers would dare offer a product today that does not feature direct coupling at the output, with no coupling capacitor required between output stages and speakers. The fact that safe use of these capacitorless output circuits has often meant the need for extensive and costly protection circuitry (to insure against DC appearing at the voice coils of direct- connected speakers if a power output transistor fails) is seldom, if ever, discussed in the literature—nor is the fact that direct coupling is not the *only* good way to get from the power output stages to the loudspeakers.

I am not suggesting here that manufacturers should refrain from using an improved circuit just because of the "NIH" factor (not invented here). Obviously, the use of MOS-FET's in FM tuner front ends has made possible the design of solid-state tuners which are more selective and less subject to cross-modulation and overload than earlier solid-state front-end circuits. Negative feedback tone controls *are* better than the old "losser" tone-control circuits, and are almost universally used in preamplifier designs today. The phase-lock-loop, as applied to stereo multiplex decoder circuits does provide better and more stable stereo separation than did earlier simple switching demodulator circuits (though not everyone's multiplex phase-lock-loop circuit is quite as good as everyone else's—catch-phrase advertising notwithstanding).

What I'm really suggesting is that the industry, at all levels, needs to devote part of its annual engineering budget to what is sometimes called "pure research"—research that is conducted without specific aims towards an end product: the kind of research that leads to major breakthroughs instead of annual product cosmetic changes.

I was delighted to learn that some companies have set aside monies and engineering time for just that kind of fundamental research. Recently, I was invited up to visit the facilities of AR in Norwood, Mass. Normally, I try to avoid such trips to plants since, in most cases, I find that the usual press release tells me all I want to know. But when I was told that this visit would *not* involve the presentation of yet another "new and revolutionary" speaker-system design, I accepted the invitation.

What I saw and heard represented the purest kind of basic research project it has been my pleasure to witness in some time. Under the guidance of AR's Bob Berkovitz, engineer Dave McIntosh had assembled what amounts to a computer system which divides up listening space into 16 spacial segments. In addition to the normal stereo speaker array, there were 16 additional speakers mounted at specific locations, each designed to handle the hall ambience or acoustic qualities produced by an appropriate area in a concert hall. The electronics of the system consisted of a highly sophisticated 16-channel digital time-delay system into which could be programmed the known reflective or reverberance characteristic of fictitious, theoretical or actual concert halls which had been accurately calculated.

Several kinds of recorded music (stereo) were auditioned and *real* as well as imagined concert hall acoustics were simulated by means of this incredible machine. In essence, the people at AR were examining what they consider to be the last significant difference between the "live" and reproduced listening experience—the psychoacoustic contribution of the concert hall itself. What to me is most significant about this work is the fact that neither Bob Berkovitz nor the management team at AR foresaw any immediate "product" evolving from this advanced work. Instead, they regard the work as a means of studying acoustical phenomena and feel that the system may be useful for acousticians who must design the concert halls of the future. By "plugging in" hall dimensions, absorptive coefficients of proposed wall surfaces and a myriad of other acoustical data, acoustical engineers may well be able to audition their proposed hall designs before they ever expend the money to build them physically. Conceivably, recording engineers could study the effects of artificially-added reverb in various software products they create, and compare such simple decay and reverb additions with simulations of true hall listening conditions.

Another non-consumer application suggests itself. Suppose a recording engineer wanted to achieve the acoustics of a "live" concert in a given recording. With the aid of AR's acoustic environment simulator and 16 small speakers, the typically confined recording studio room could be expanded through digital electronics to sound like the concert hall which the studio could not afford to rent for the recording session.

Many industry experts have been predicting for some time that digital electronics will someday take over in the world of audio, and AR's advanced work in acoustic environment simulation seems to be an important step in that direction—one that could not possibly have been done using analog techniques. If we, as an industry, are to continue to make technological strides, we had all better get away from the "me-too" philosophy and engage in some fundamental research that goes beyond new knob or front-panel designs.

MODERN RECORDING LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Akai GX-630D-SS Open-Reel Tape Recorder

General Description: The Akai GX-630D-SS is a four-channel and stereo open-reel tape recorder capable of handling up to 10½-inch diameter reels. A two-speed (7½ and 3¾ ips) model, it will record and play in four-channel (quadraphonic), two-channel stereo or mono; it also is equipped with the "quadra-sync" feature—similar to "simul-sync" on other models—that enables the recordist to lay down successive tracks in perfect synchronization for a variety of "overdub" applications. Input mixing of line and mic sources is possible during recording. "Fast-buttoning" also is possible with this machine—that is, the operator can go from one transport mode to another without first pressing the stop button; this feature also permits direct change to the recording mode for "punch-in" recording, or—as Akai terms it—"add-on" recording.

Also found on this deck is a pitch control designed to vary musical intervals by ±1 semi-tone (tape speed, ±5%), a feature that may be used in music instruction or for deliberately adjusting musical pitch when more than one instrumentalist is performing. The pitch control may be activated on both recording and playback.

The transport is powered by three motors: an AC servo motor for capstan drive, and two eddy-current motors for the reels. The head configuration is essentially the three-head line-up (erase, record, play) but there are actually two erase heads, one for the front channels and the other for the rear channels.

The deck may be installed horizontally or vertically; there are two sets of "feet" on the "bottom" and "back." The front panel is busy-looking but sensibly laid out. The reel "tables," of course, dominate the upper portion, and the head assembly is centered below them. The tape threads through a fairly sophisticated system of guides and swinging levers, common to this class of equipment. The tape index counter and reset-button are to the left of the head assembly, and the pitch control is just above the counter. Tape-speed selector, power off/on switch, and reel-size selector are just under the counter.

Transport controls are grouped to the right of head assembly, and include a pause control, plus the buttons for recording, rewind, stop, play and fast-forward.

Below the head assembly are four VU meters (average-reading) which operate on both recording and playback. Above each meter is an indicator lamp to show when that track is being recorded. To the left of the meter group is a tape-selector switch with positions for low-noise tape or "wide-range" tape which Akai identifies only as "tape of a grade higher than low-noise." The recommended standard tape for this machine, incidentally, is low-noise such as Akai LN-150-7 or Scotch 211. To the right of the meters is the tape/source monitor switch.

Below the meters are the four buttons for engaging the quadra-sync feature, and a fifth button for selecting either four-channel or two-channel mode (in the latter position, only the front channels are operative).

Ranged across the bottom of the panel are stereo headphone jacks (one for the front channels, one for the rear channels); dual-concentric knobs for mic and line input levels, separate on each channel; a dual-concentric output level adjustment (each portion in this case regulates both front and both rear channels); and the four microphone input jacks.

Line input and output jacks (eight in all, separate for each channel) are grouped on the rear panel (or the "bottom," should you care to install the deck horizontally). At the rear there also is a jack for using an optional remote-control (available as an accessory), the machine's fuse-holder and, of course, the power cord. The Akai GX-630D-SS is supplied in a wooden wrap-around and comes with two signal cables, a pair of 10½-inch reel adapters, and one empty 10½-inch reel.

Test Results: In general, MR found that the Akai GX-630D-SS met its published specifications, although there were a few areas where it fell slightly short. For instance, THD was measured as 0.7% against the claimed 0.5%; response at the slow speed made it out to 14 kHz (within $\pm$ 3 dB) as compared to the claimed 15 kHz. On the other hand, response at the 7½ ips speed was right on spec to 21 kHz; signal-to-noise came in "as claimed" and—very commendable—the deck had very ample signal headroom, up to 9 dB, for 3% THD.

General Info: Unit is 20.7 inches high, 17.3 inches wide, 9.4 inches deep. Weight is 45.5 pounds. Owner's instructions judged very good. Price is $995.

Individual Comment by N.E.: In the sometimes confusing galaxy of big open-reel tape recorders known loosely as "semi-pro" models, it is difficult to decide where to place this Akai. It has some of the advanced features many recordists want, such as quadra-sync, fast-buttoning, input mixing, "add-on" instant recording and, of course, the capacity for 10½-inch reels. On the other hand, its top speed is 7½ ips (not 15 ips); its monitor function covers all tracks rather than individual tracks; and it obviously lacks a sophisticated tape bias-and-equalization facility. In fact, the instructions are somewhat cryptic on this point: why couldn't Akai spell out what they mean by "tape of a grade higher than low-noise"—do they mean chrome, or ferrichrome or what?

Judging this recorder, however, by the perhaps less-demanding criteria of an advanced home-user tape enthusiast, one can regard it as a very good machine within its avowed design-and-feature framework, one that will make clean and faithful tapes of whatever you care to record on it, and which always has the option of both four-channel recording and playback, as well as the sync-ing of multiple tracks during recording. To go much beyond what the 630 does offer would, obviously, mean spending more (for a higher-end Akai or similar deck) than the "under-$1,000" price tag asked for this model.

Individual Comment by L.F.: Having had an opportunity to work with Akai's more expensive and earlier Model GX-400D-SS, I can tell you that "quadra-sync" (or "sel-synch," or any of the other trade names devised to describe synchronized multi-track recording capability in consumer-type open-reel tape decks) does not, in itself, make a "professional" tape deck. Certainly, the ability to synchronize newly recorded tracks with previously recorded ones is important for anyone wishing to do any serious recording beyond the transcription of already available program sources such as records or tapes, and the "quadra-sync" feature, as supplied on the GX-630D-SS is as versatile as on any other Akai or competitive machines.

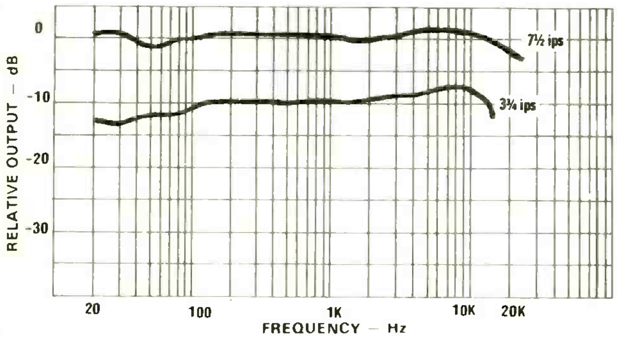
What bothers me most, however, is the transport arrangement. The absence of dual-capstan tape drive places this unit in the less-than-professional category. The wow-and-flutter readings which we measured are really "averaged" readings. But there were many moments during the readings when "spikes" of speed variation (due either to inconsistent tape tension, poor tape wind or any one of several other reasons) showed up clearly on the wow-and-flutter meter. Dual-capstan drive could have done much to smooth out these random speed variations.

The direct function-change controls (which enable you to switch from any transport mode to any other without having to go through "stop") are also nice to have, but the unduly long pause which occurs when using this feature make the operator wonder if anything is ever going to happen. Hand-rocking of the reels is not possible with this unit, which makes one dependent upon the "pause switch" for most editing functions—hardly an effective means of precision editing.

The pitch control is a useful feature. It permits a 5% speed variation during playback only, so that if you are using this machine in combination with another and wish to mix previously recorded program material (played on the GX-630D-SS) with material to be recorded onto the tracks of a second machine, you can adjust the pitch of the played-back material to match that of the instruments you are playing "live" in the subsequent dubbing or mix.

While we cannot account for the considerably better S/N readings that we obtained compared to published specifications (Akai claims only 54 dB whereas we measured over 60 dB), it is possible that Akai may not have been using a reference signal which produces 3% total harmonic distortion—common practice with professional machines. In any case, we were able to reach a +9 dB record level before the 3% THD point was reached and that is pretty good head room for any machine—home or professional.

In short, I feel that the Akai GX-630D-SS has excellent electronics and less-than-excellent mechanical and transport design. I doubt if it could stand up mechanically to the rigors of many hours of daily use, but if you want to get into multiple-track synchronized recording at home and can't afford more than the thousand dollars it takes to buy this model, it may be the way to "learn by doing" until you can afford something better—either from Akai or from some other manufacturer.



Akai GX-630D-SS: Record/play frequency response, using Maxell UD tape.

AKAI GX-630D-SS OPEN-REEL TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC (using Maxell UD tape, noted as suitable by mfr. for this deck)	LAB MEASUREMENT
Record/playback frequency response	
7½ ips	±3 dB, 25 Hz to 22.5 kHz
3¾ ips	±3 dB, 40 Hz to 14 kHz
Harmonic distortion	
+3 VU	0.7%
0 VU	0.7%
−3 VU	0.7%
Best S/N ratio	60.5 dB
Input sensitivity, mic	0.25 mV
line	65 mV
Output level, line	.78 V
headphone	33 mV/8 Ohms
Bias frequency	100 kHz
Speed accuracy	+0.5%
Wow & flutter (WRMS), 7½ ips	0.06%
3¾ ips	0.08%

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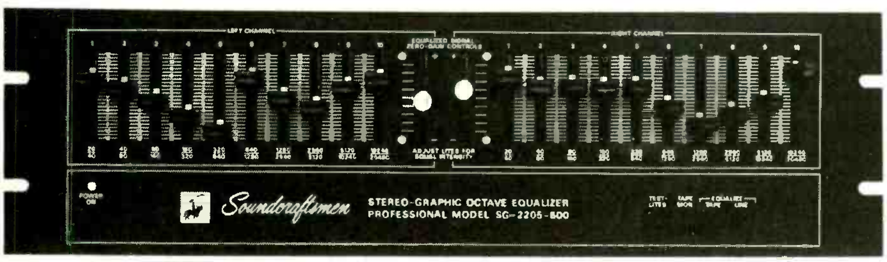


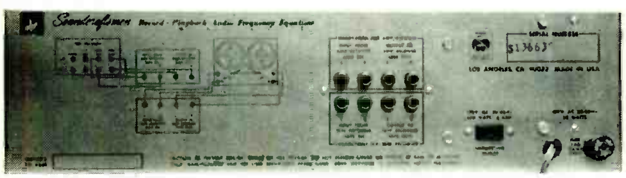
Soundcraftsmen SG-2205-600 Stereo-Graphic Octave Equalizer

General Description: The Soundcraftsmen SG-2205-600 is a two-channel (stereo) graphic octave-equalizer for adjusting tonal ranges in ten independent bands on each channel. Inasmuch as each set of ten controls operates independently of the other, the device may be used for a stereo system, or in applications involving two separate monophonic systems. The general functioning and format of the SG-2205 make it suitable for use in a wide assortment of sound systems, including commercial, studio and home setups. Its uses include "speaker/room equalizing" for playback systems and for P.A. systems, as well as "live"-performance reinforcement of desired frequency ranges, and deliberately altering the equalization or tonal characteristics when making tape recordings.

The front panel contains all operating controls and switches. For each channel there are ten sliders, calibrated from +12 to −12 (dB). The frequencies affected by each are labeled in actual octave steps starting with 20/40 Hz and running up to 10,240/20,480 Hz. Centered between the two main groups of sliders are four LED indicators and two additional sliders (two LED's and one slider relating to each channel). The LED's serve as balancing and "proper operation" indicators. The additional sliders serve to balance the overall signal after equalization so that its gain equals that of the incoming (unequalized) signal—as shown on the LED's. This is a useful and important feature inasmuch as the equalizer is not intended to serve as an amplifier.

Below the sliders on the left is the unit's power off/on button, while over at the right are four more buttons. One turns on the LED's; another is a tape-monitor option; the last two permit a choice of equalizing a tape while recording, or of equalizing the line inputs to the device (these two buttons cannot both be engaged at once).

At the rear are the input and output connections (phono or pin-type jacks). Provision is made for hook-up directly to and from a tape recorder, and to and from the tape-monitor and tape-out jacks on a standard receiver or amplifier. When the latter hook-up is used, the tape-monitor button on the front of the SG-2205 replaces the tape-monitor button on that receiver or amplifier. As an added convenience, the recommended hook-ups are clearly illustrated on the rear panel, and all connecting jacks are carefully labeled as to function. The rear panel also contains the unit's fuse-holder, power cord, and an unswitched AC outlet (rated for up to 600 watts, 5 amps) for powering other equipment.



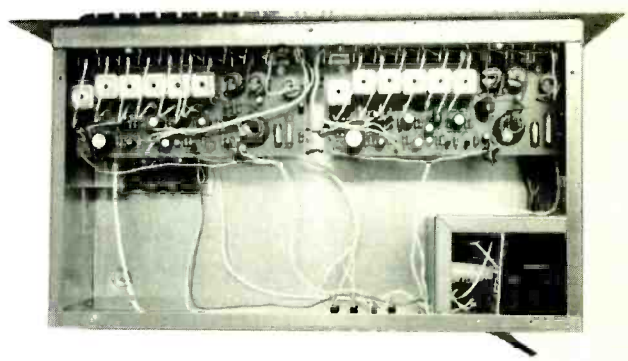
Soundcraftsmen SG-2205-600: Rear panel has hook-up diagram and extra jacks for reestablishing "tape-monitor" circuit that may have been used up on receiver or amplifier when installing the equalizer.

The device is supplied with a special disc recording that contains explanations, test procedures, and pink-noise signals for using the SG-2205 in room equalization. Three methods are given, in increasing order of complexity, from doing it by ear, through a more involved procedure in which speakers may be moved for a more accurate result, to a professional procedure using sound-pressure level measuring equipment. The same material is recorded on both sides of the record, which doubles its effective life as useful test material.

Also supplied with the device are "computone" charts for marking particular equalization settings that may be used in different applications requiring alternate control settings to those already established. By referring to the markings on the charts, the various equalization positions can be quickly repeated.

Test Results: On all counts, the Soundcraftsmen SG-2205 impressed MR as being a carefully thought-out and well-engineered product that not only does what it claims, but does so without demanding too much on the part of the user—including the non-technical owner. At the same time, it is a thoroughly professional-grade device that should also interest the more advanced or technically sophisticated user. Although not all the specs that came with our test sample were equaled in our tests, the differences were small indeed, generally on the order of a hundredth of a percent or so in terms of distortion, or a few Hz in terms of frequency response. More important, in MR's view, is the device's headroom which, even at 3 volts input (far more than is likely to be encountered in normal use), permits it to remain a "long way" from waveform clipping. Signal-to-noise ratio is excellent (actually we measured 93 dB against the 91 dB claimed), and response is likely as flat or uniform (within ± 0.25 dB) across the band from below 20 Hz to beyond 40 kHz as any other audio component in your system.

Internal construction, components layout, and wiring all are first-rate. The power supply for the device is located at a generous distance from the signal circuit boards and is very well shielded, which accounts for the excellent S/N measured. A check of the slider equalizer control effect verified the center frequencies as well as the boost and cut available on each slider. Additional tests confirmed the limitless myriad of response patterns available with the device. MR feels the unit to be reliable, useful, and of interest to the serious sound-enthusiast, as well as to the professional user. It may be rack-mounted, fitted into a cabinet cut-out (vertically or horizontally), or placed anywhere you choose "as is" or in an optional dress-up cabinet.



Soundcraftsmen SG-2205-600: Internal view discloses individual circuit boards (identical) for each channel. Filter circuits are LC type. Well-shielded power supply at lower left accounts for excellent S/N ratio measured.

General Info: Front panel is 19 by 5¼ inches. Chassis depth behind is 9½ inches; allow about an inch more for connectors at rear. Chassis height is 4⅝ inches. Weight is 28 lbs. Owner's instructions, very good. Supplied with test record and "computone" charts. Price is $399.50.

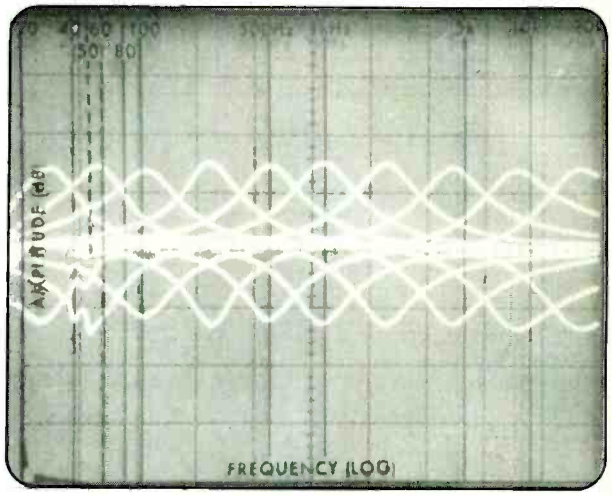
Individual Comment by L.F.: A graphic equalizer can turn out to be one of the most useful additions to a sound system, or it can prove to be a sonic disaster—it all depends on how it is used, and how experienced the owner is, or at least how willing the owner is to experiment with the device and follow the instructions supplied by its manufacturer. The people at Soundcraftsmen put it very well when they state: "As you use the equalizer and experiment with each octave control, the first impression may be how strange or 'bad' you can make things sound. . . ."

My own feeling in experimenting with the SG-2205-600 is that if you need the full range of any of the controls (either plus or minus 12 dB at the center frequency of the octave control in question) you had better start all over again. Either your listening room is beyond help or you own a piece of associated equipment that is degrading overall system response to such a degree that nothing is going to help. Used in moderation (a maximum correction of no more than 6 dB in either direction for any of the controls) a graphic equalizer (and particularly this one) can smooth out the response of your system to a degree that is just not possible with conventional tone controls.

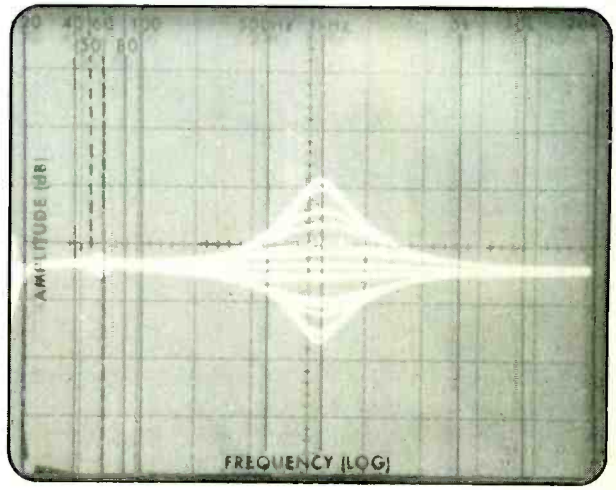
We did find it a bit difficult to pinpoint the precise mid-point of each octave slider control, and the incorporation of some sort of mechanical "click stop" detent at the center of the range of each control would have made it easier to locate the "flat" spot for those octave controls which never require any boost or cut. We were pleased to note that the controls of one channel are totally separate physically from those of the other, in recognition of the fact that often the two stereo channels require different equalization.

Individual Comment by N.E.: As listening tastes become more sophisticated and critical, and as tape-recording activities increase, along with other sound-application work such as public-address and sound-reinforcement, the rising interest in devices such as this becomes very understandable. The SG-2205 is designed to fill more than one bill and somehow manages, as a product, to suggest an appeal to the non-technical but sound-wise listener, as well as to the trained technician. I like the manner in which its use is explained (on the test-record), and the fact that you can get good results in about 20 minutes by using the simplest method. I also like the use of LED's as an easy way to establish "unity gain" through the device, regardless of how you have set the octave sliders. It also is easy to make an instant comparison between unequalized signals in, and equalized signals out.

A word on the specifications for this unit. The owner's manual that came with our test sample lists somewhat different figures from those appearing in a later bulletin. For instance, the early spec for THD (at 1 volt) was given as below 0.01% whereas the newer spec gives THD (at 1 volt) as 0.05%. Our test sample made it at 0.025% which is fairly in the middle of this range and good enough. The earlier spec for S/N was 91 dB; the newer spec is better at 96 dB. Again, our test sample showed 93 dB, also a nice in-between figure and certainly an excellent one for this characteristic.



Soundcraftsmen SG-2205-600: Available range of control (boost and cut) of each octave slider.



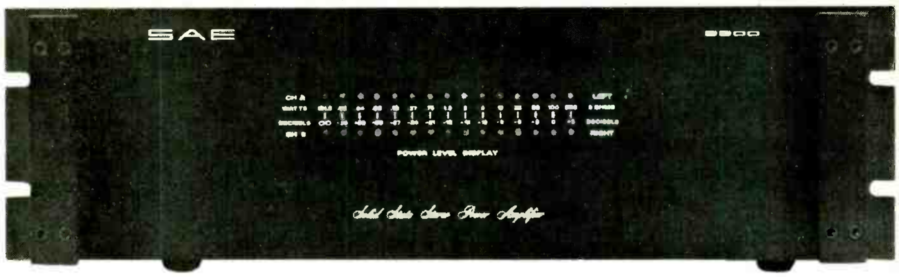
Soundcraftsmen SG-2205-600: Response setting obtained by varying the 640/1280 Hz octave control.

SOUNDCRAFTSMEN SG-2205-600 GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Harmonic distortion:	
1 volt in; 1 volt out	0.025%
3 volts in; 1 volt out	0.11%
3 volts in; 3 volts out	0.13%
1 volt in; 6 volts out (boost settings)	0.33%
IM distortion, 1-volt in; 1-volt out	0.14%
S/N ratio (below 1-volt output)	93 dB
Frequency response	±0.25 dB, 17 Hz to 48 kHz ±1 dB, 10 Hz to 54 kHz
Range of each octave control	±12 dB (see curves)

CIRCLE 11 ON READER SERVICE CARD

SAE 2200 Stereo Power Amplifier



General Description: The SAE 2200 is a two-channel basic or power amplifier designed for insertion in a sound system between a "high-level" source (such as a preamp) and loudspeakers. No controls are provided. The 2200 has no off/on power switch and its AC line-cord must be connected directly into an outlet capable of handling up to 500 watts, which is what the amplifier draws when driven to its rated power output (which is 100 watts per channel at 0.05% THD across the audio band from 20 Hz to 20 kHz, both channels driven into 4- or 8-ohm loads).

The front panel features a stereo (two-channel) LED display of power level for an 8-ohm load on each channel. There are 15 indicators per channel, calibrated in both watts (from "idle" up to 200), and in decibels from below -36 to +3. The front also is fitted with handles for insertion or removal in rack-mounts, although the unit also could be installed vertically, in a suitable cut-out with the extending edges of the front panel resting on a sturdy-enough supporting frame. In any case, some ventilation is recommended; the 2200 should not be installed with other heat-producing components in a completely enclosed space.

The rear contains a stereo pair of signal input jacks (phono or pin-type), a stereo pair of speaker output terminals (binding posts that also accept banana plugs), a five-amp fuse-holder, and the power cord.

SAE 2200: Section of rear panel has major performance specifications printed on it.

Examination of the interior indicates long-term reliability based on observations of the careful, high-quality manner in which the amplifier is constructed. The circuitry and components used bespeak "design conservatism"—such as the use of plug-in Mil-spec G-10 glass-epoxy circuit boards.

Test Results: In tests conducted by MR, the SAE 2200 met or exceeded its published specifications. Power into 8-ohm loads went up to 115 watts; into 4-ohm loads we measured 150 watts. Distortion consistently remained lower than claimed; ditto for hum and noise. In the very rigorous square-wave tests, the low-frequency response was virtually a replica of the input-test signal; the high-frequency response showed a little "rounding" of the leading edge but with no ringing which would indicate good transient performance.

MR deliberately pushed the amplifier into momentary overload and found that it clipped and recovered in exemplary manner. We also subjected the amplifier to "torture loads" (inductive, highly capacitive, and fully open circuit) and the results of these tests do indeed indicate that the unit is "unconditionally stable."

Readers may wonder why SAE chose to rate power output capability of 100 watts for both 8-ohm and 4-ohm loads, when it is widely known that in general an amp can deliver greater power into 4-ohm than it can into 8-ohm loads. The answer has to do with the FTC's pre-conditioning requirement which demands that an amplifier be able to run at one-third its rated power for one hour. For the SAE 2200, that means 33⅓ watts per channel or about 2 amperes of current delivered by each channel's output stages for an hour into an 8-ohm load.

Of course, the amplifier is capable of much greater power output at 4 ohms (we measured 150 watts even at 20 Hz), but the same 33⅓ watts of "preconditioning" power demanded by the FTC represents nearly 3 amperes of current into that lower impedance load. Had SAE chosen to rate the 4-ohm power output at a higher level, the preconditioning wattage would have risen proportionately, with a proportionate increase in current delivered by the output stages. The unit might not pass that test, even though that test is unrelated to actual listening conditions when a musical signal is amplified. We mention all this only to assure the prospective user who might hook up 4-ohm speakers that the SAE 2200 is indeed capable of delivering much more than its rated 100 watts per channel into that load impedance, FTC notwithstanding.

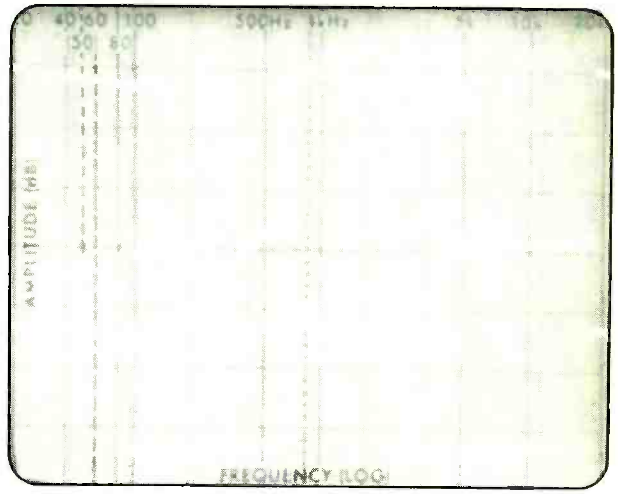
General Info: Front panel measures 19 inches by 5¼ inches. Chassis behind measures 17½ inches wide; 4⅝ inches high. Depth behind front panel is 8⅛ inches; for total depth, including front handles and rear connectors, allow 11 inches. Weight is 22 pounds. Owner's manual, good. Unit furnished in black metal case; optional cabinet is available. Price is $450.

Individual Comment by N.E.: There is something "Spartan" about the appearance of this amplifier. Tests in the lab and in the listening room do confirm that the SAE 2200 is among the very best as a performer, but there also is a sense of not making any concessions to the user as far as making things too easy or convenient. For instance, there is the lack of a power off/on switch that implies the 2200 must be connected to the "convenience" outlet on the back of a preamp, but how many such outlets around can handle the 500 or so watts of input power drawn by the 2200 when it is called on to deliver its top wattage output? The LED display is nicely worked out, and accurate, but you have to hunt in the instructions to find that with 4-ohm loads the power indications are doubled, and for 16-ohm loads they are divided by two. Why couldn't SAE supply this data in the form of a simple legend right on the front panel?

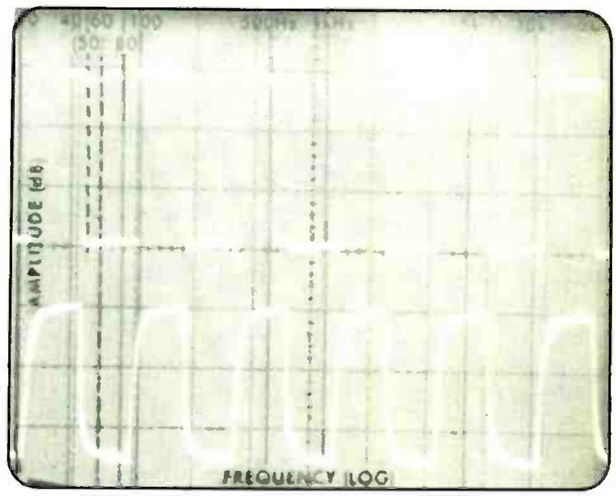
You could argue that the unit is designed primarily for professional users who don't need such frills. On the other hand, even the pro is entitled to a little "icing on the cake"—especially when the cake itself is as good as this one is. And, in any case, it would seem—on the basis of its high performance—that this amplifier will also appeal to many non-pro stereo fans, and these users need fairly explicit guides when using super, high-powered audio gear like this.

Individual Comment by L.F.: Whenever I am faced with the prospect of evaluating one of the new breed of high-powered basic amplifiers, I find it difficult to talk in terms of audible differences or sound qualities of one good basic amp compared with another. Granted that these differences exist, they are so much more subtle than the major differences one hears when comparing, say, two high-quality speaker systems, or two top-of-the-line phono pick-ups that I tend to read with cynicism some of the so-called subjective "reports" that appear in some of the so-called underground audio buff books. You will therefore not find me straining to come up with descriptive terms with which to describe the sound qualities of the SAE 2200. Suffice it to say that I was extremely pleased with what I heard.

On more tangible aspects, I like the array of LED indicators that grace the front panel of the 2200 (they are, in my view, preferable to meters for this application). The LED's do provide the meaningful peak-level indications, and we did find them to be accurate to within 1 dB all the way from 20 milliwatts to overload. But I also feel that there should be a power off/on switch on the front panel.



SAE 2200: Square-wave response with 20 Hz square-wave input. Upper trace is input signal.



SAE 2200: Square-wave response with 20 kHz square-wave input. Upper trace is input signal.

SAE 2200 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Continuous power output per channel, 20 Hz to 20 kHz	115 watts into 8-ohm loads 150 watts into 4-ohm loads
THD at rated output (100 watts), 1 kHz	0.016% into 8-ohms 0.025% into 4-ohms
Power bandwidth for rated THD (0.05%)	10 Hz to 42 kHz
Frequency response	±1 dB, 6 Hz to 58 kHz
Damping factor	>100
IM distortion at rated output (100 watts)	0.05%
Residual hum and noise below rated output	−105 dB
Square-wave transient response	see 'scope photos
Input sensitivity for rated output	1.5 volts for 8-ohm loads 1.0 volts for 4-ohm loads

CIRCLE 13 ON READER SERVICE CARD

Peavey 800 Stereo Mixer

General Description: The Peavey 800 is an eight-channel stereo mixer intended primarily for sound-reinforcement work, although its flexibility, relative portability, and inclusion of a stereo tape-deck output jack also makes it suitable for use in an "8-in/2-out" recording console. The unit's built-in Accutronics Hammond reverb delay line is a valuable addition, and the post- and pre-access to the reverb, effects, and monitor busses lend additional versatility, although in recording applications the "pre-" capability is likely to prove less vital than it would be for sound-reinforcement ("live" concert) monitoring work.

The main left- and right-channel output busses provide a choice of unbalanced or 600-ohm balanced outputs, so that these outputs can be connected to hi-fi amplifiers as well as to commercial balanced-input P.A. equipment. The rotary input attenuators have sufficient range of adjustment to permit virtually any type of microphone or line input to be connected, while full use of the sliders is maintained.

The model 800 includes low- and high-frequency equalizers. The former is rated for better than 15 dB boost or cut at 50 Hz; the latter, for 15 dB boost or cut at 5 kHz. A stereo-pan control also is provided to achieve the desired balance from each channel into the main output mixing busses.

The front panel is laid out logically and with a view toward facilitating operation for the busy soundman. The major portion is sloped upward, with a pair of VU meters at the upper right. The rear panel contains an ample assortment of in- and out-connectors for correctly and effectively integrating the model 800 into various possible systems using a wide variety of sound gear. The unit's main AC power switch has two "on" positions that effectively reverse the polarity of the line cord, if needed to minimize hum level and/or to ob-

Peavey 800: Rear panel.

tain the least noise when high-Z mics are used. The line cord itself is a heavy-duty type with a three-pin plug for use in three-hole (grounding) outlets. If used with a two-hole outlet, a three-to-two adapter must be added. For convenience when travelling with the 800, cable-wrapping brackets are provided at the rear.

Test Results: To test some key performance areas of the model 800, MR hooked it into a Teac A-6100 to use it as a recording mixer/console, and also made several instrument-test measurements. Frequency response was checked to exactly as claimed by the manufacturer (within ± 1 dB from 20 Hz to 20 kHz at a level of +4 dBm). Total harmonic distortion was lower than claimed, although channel crosstalk was shy of the specification by 5 dB (we measured −55 dB; Peavey states −50 dB). Other important characteristics such as equalization ranges and output levels all were "on spec."

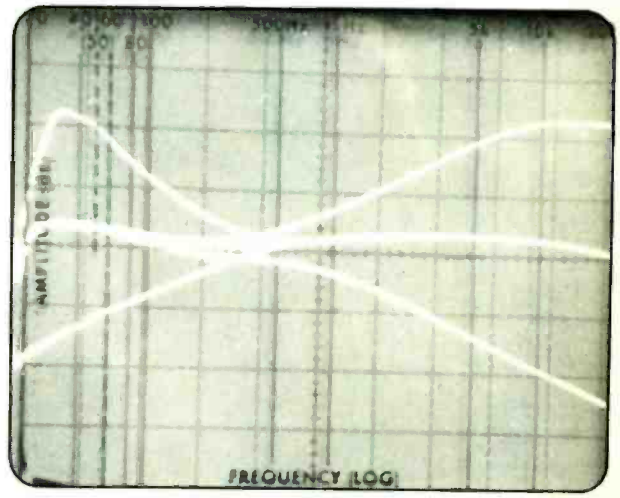
Individual input channel controls were found to be smooth-acting, and MR observed a high order of uniformity between the eight individual input modules.

All told, the Peavey 800 certainly is a recommendable product for its intended uses.

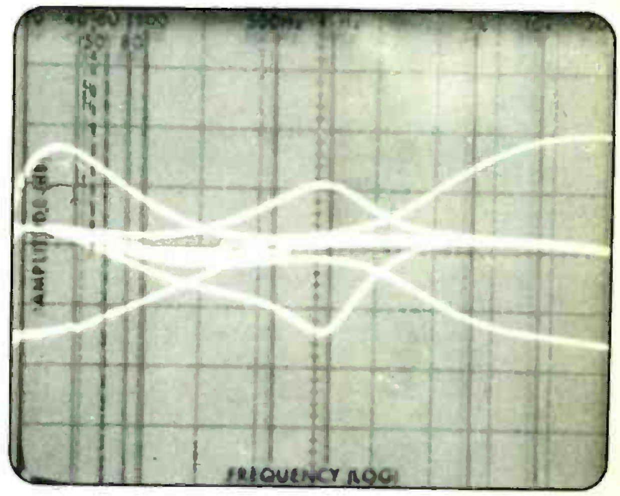
General Info: Peavey 800 is supplied in metal case with wooden sides. Overall dimensions are 28 inches wide by 23 inches deep by 10 inches high. Weight is 36 lbs. Owner's instructions are excellent. Price: $649.50.

Individual Comment by L.F.: Apparently, Peavey has thought of just about everything that could be included in an eight-channel mixer in this price range. There is one area, however, in which it seems to me that Peavey has provided almost too much flexibility—and that concerns the two-channel VU meters. These have front-panel adjustments that vary their sensitivity, presumably for the user who wants to adjust the meters so that their 0-dB reading agrees with the 0-dB readings of associated equipment (such as a tape deck, etc.). My question here is: What about the user who is dependent solely on the meters on the model 800? If he does not have test equipment whereby the mixer's own meters can be set up to read in standard VU or dBm, there is no way to properly calibrate these meters. A test-tone oscillator (delivering some standard reference level to the meter circuits), or at least a switch that lets you bypass calibration pots and automatically sets up standard VU readings would have been, in my opinion, a useful addition to what is otherwise a carefully planned mixer.

Individual Comment by N.E.: A particular interest during the use-tests of the model 800 hooked into a tape recorder was getting an idea of relative noise levels and action of the controls. We found that it is possible to set things up so that the residual noise level sometimes becomes audible—this comes about by pulling the input attenuator all the way down, thus requiring high settings of the slide controls, both individual and master). However, this may be mere nit-picking since if instructions are followed and the input attenuators are adjusted so that both input and output slide controls "hover" in their mid-position ranges, there will be no audible noise contributed by the Peavey 800 to tape recordings made through it.



Peavey 800: Low and high-frequency range for each input channel (each vertical division equals 10 dB).



Peavey 800: Low-, mid- and high-frequency range of master (output) section.

PEAVEY 800 MIXER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response (at 4 dBm)	±1 dB, 20 Hz to 20 kHz
Total harmonic distortion at −10 dBm	0.03%
Total harmonic distortion at 0 dBm	0.1%
Crosstalk (at 1 kHz)	−55 dB
Equalization, input channels	±15 dB at 50 Hz and at 5 kHz (see curves)
Equalization, output (master)	±15 dB at 50 Hz, at 800 Hz, and at 5 kHz (see curves)
Output level, balanced (600 Ohms)	3 V RMS (+12 dBm)
Output level, unbalanced (2K Ohms)	4 V RMS

CIRCLE 12 ON READER SERVICE CARD

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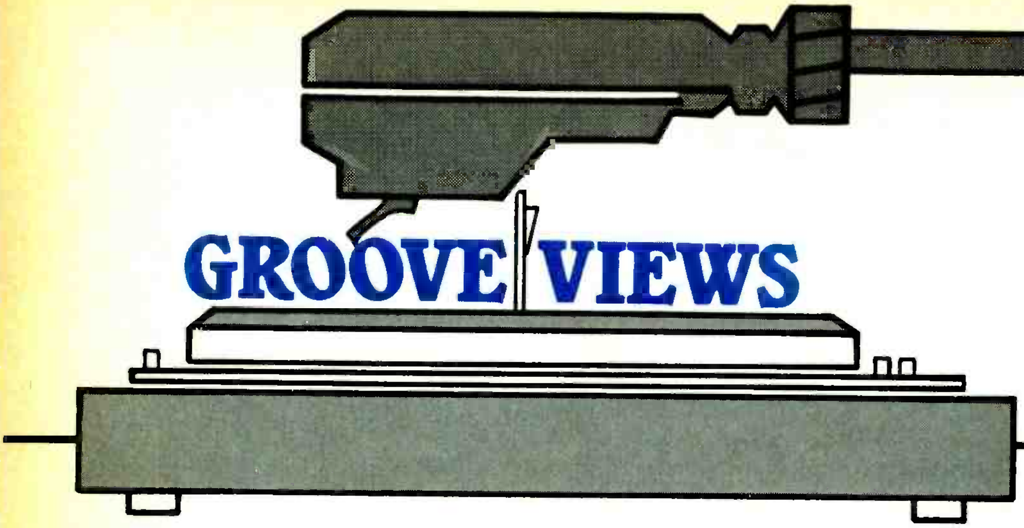

Reviewed by:
SEDGWICK CLARK
CHAS FARRELL-KIMBRELL
NAT HENTOFF
H.G. LA TORRE
IRA MAYER
STEVEN PEEPLES
GIL PODOLINSKY
PAULETTE WEISS

POPULAR

BE-BOP DELUXE: *Sunburst Finish.* [Bill Nelson, John Leckie, producers; John Leckie, engineer; recorded at E.M.I. Studios, London and Air London.] Harvest Records ST-11478.

Performance: **Consistently strong**
Recording: **Much improved**

Sunburst Finish is Be-Bop Deluxe's second American release and shows great progress from their *Futurama* album. For an introductory record to the U.S. (there is one lp prior to *Futurama*, released only in England), it demonstrated the promise of this latest adventure in English rock.

Sunburst Finish takes a definite step in fulfilling that promise. One marked improvement is the addition of Andrew Clark on keyboards, allowing Bill Nelson, the group's guitarist, vocalist and writer more emphasis in those areas and eliminating his elementary keyboard skills. Clark's presence successfully transforms the band from a high-power blues trio to one fine progressive group.

The production of *Sunburst* is also an improvement in many regards. The extensive use of echo and occasionally garbled vocals are eliminated. Further, the production is more in sync with the group's goals by having Nelson co-produce the album with engineer John Leckie. It is also worth mentioning that *Sunburst* was recorded at only two studios, whereas *Futurama* was born in three vastly different studios and settings by three different engineers and one non-affiliated producer. The result was that several good ideas were scuttled on *Futurama*. The emphasis on *Sunburst* is to showcase the talents of Be-Bop Deluxe equally and that decision will take them a long way. G.P.

JOE COCKER: *Stingray.* [Rob Fraboni, producer; Baker Bigsby, Neil Case and Rob Fraboni; recorded at Dynamic Sounds, Kingston, Jamaica; mixed at The Village Recorders, Los Angeles, Cal.] A & M SP-4574.

Performance: **Disappointing**
Recording: **Uninspired**

Joe Cocker has had to bear the cross of inconsistency for most of his career. True, this has been more apparent in his "live" performances than on vinyl, but inconsistency is nevertheless a difficult label to throw off. So, theoretically, what better way to assist Cocker than with some of the most consistently good session musicians in R & B and rock as a back-up group?

It had to have looked better than this on the drawing board.

The outcome of the master plan is that *no one* succeeds in standing out on the album. Now, this is not to say that Cocker drags down Mr. Tee, Dupree and friends. No, the problem is that the material is not challenging enough. Also, the arrangements don't allow any room to expand. The lack of dynamics makes it easy for each song to become indistinguishable from the others. If it weren't for changes in tempo, some parts would start to sound like a test tone. In short, the arrangements are predictable.

Richard Tee may have had trouble tailoring these compositions to a vocalist of Cocker's unpredictability—sometimes shouting, sometimes mumbling. To be fair, a vocalist of the caliber of Bobby "Blue" Bland would have been hard pressed to pull these selections up from the depths.

The more basic tracks—those with no, or at least limited, back-up vocalists (incredibly tiring on this album)—are the tracks that escape total anonymity: "You Came Along," "Worrier" and "The Jealous Kind." A vocalist with Joe Cocker's enunciation cannot survive a production plan that places his voice anywhere other than up-front in the mix.

It's not apparent why the album was recorded at Dynamic Sounds in Jamaica, unless simply to work with reggae-specialist Peter Tosh on "The Man in Me." Certainly the album lacks the sharply defined high and midrange tones or the earth-moving bottom and indefinable "feel" for which Dynamic has come to be known.

Despite it all, drummer Steve Gadd manages to come out clean. He is one of the few drummers who knows how to properly tune his drums to fit the environment. Therefore, his drums always sound (especially the snare) natural, rather than like large foam rubber pillows.

Ideally an album with such top-notch talent should allow room for that talent to show through both strong material and expanded settings. The ideal did not occur here, and although the outcome might have been acceptable from a different musical gathering, it becomes a hurried-sounding and disappointing work from Joe Cocker and friends. H.G.L.

JOE COCKER & FRIENDS: Back to the drawing board.

CHRIS HILLMAN: *Slippin' Away.* [Ron and Howard Albert, producers/engineers; recorded at Sound Labs and Cherokee Studios, Los Angeles, Cal.] Asylum 7E-1062.

Performance: **Consummate melancholy**
Recording: **Clean and alive**

Okay, so I wrote the story with the Alberts and will be the first to admit I think their sound is among the best in the country. So what? None of that really means a thing. What's important is that the Alberts and Chris Hillman have a track record longer than both of my arms put together. I've come to expect a great deal from them individually, and collectively I would expect even more. Those expectations have been exceeded after listening to the finished album. It's a consummate mix of earth and class.

And the studio-hopping had no adverse effect, something which I couldn't really determine until I heard the finished album. It's remarkably clean, very straightforward and vibrantly alive.

Most of the songs are apparently the results of his seemingly frequent mid-night connections with loneliness; they project a non-pathological melancholy. What's remarkable is that while the *songs* feel lonely, the arrangements and instrumental colorations exude a strong positive energy. Hillman the lonely eternal optimist.

My only complaint is that a couple of the tunes rhythmically approach reggae, which doesn't do anything for me at all. But that's a personal problem. If you like reggae, all the better for you. *Slippin' Away* is a consummate piece of work, but you don't have to take my word for it; once you put it on the turntable, it won't slip off without extra attention. S.P.

ANTONIO CARLOS JOBIM: *Urubu.* [Claus Ogerman, producer; Frank Laico, engineer; recorded at Columbia Studios, New York, N.Y.] Warner Bros. 2928.

Performance: **Disappointing**
Recording: **Superb**

Claus Ogerman has concocted for our delight the *piece de resistance* of recording art, a melding of the musician's creative talents with the sauce of the finest, most imaginative use of modern electronic reproduction possible. "Boto" is the glorious high point of Antonio Carlos Jobim's latest album, *Urubu.* Here, Ogerman, who also acts as orchestral arranger and conductor throughout, has masterfully blended accepted musical devices—swelling orchestral backgrounds, the gentle vocal duet of Jobim and Miucha, sparkling instrumental solos—with breathtaking creative touches. A tubby, echoing Brazilian string introduction captures the ear. The cry of the urubu, or turkey buzzard, imitated by a whistle with reverb, cuts into the music, sharp and startling. Claves "toc-toc" in sporadic, fascinating patterns. And the ending—an unexpected, electronically stretched piece of percussion after the music has faded—is as wonderful as it is astonishing. This complex, delicious cut must be heard and savored. "*Corrententeza* (the Stream)" is another morsel of happy melody enhanced by technical guidance.

Unfortunately, Brazilian composer-performer Jobim offers Ogerman less to work with in the remaining selections. The two other vocals on side one are dull and unmemorable. Particularly disappointing is side two, wholly devoted to four lush orchestral compositions with a rather saccharine similarity. No complaint about the electronics here, however. Recording engineer Frank Laico has created sound of exceptional clarity, with a true sense of aural realism. But the overriding genius is unquestionably that of Ogerman, whose presence is felt throughout. Even with inferior material, his production is masterful.

With a meaty piece of music, his brilliance is most impressive. P.W.

THE RAMONES: ***Ramones.*** [Craig Leon, producer; Rob Freeman, Don Hunerburg, engineers; recorded at Plaza Sound, New York, N.Y.] Sire SASD-7520.

Performance: **A Sow's Ear**
Recording: **A Silk Purse**

On initial hearing, the Ramones' debut album harkens back to the good old days when Blue Cheer would go into a studio and blow out a whole series of Marshalls while the knob-turners-in-residence, attempting to turn volume and feedback into primordial rock 'n' roll, would field complaints from the neighbors.

Going on the assumption that this music is basic in the extreme (I'm still listening for a second chord change) we find that those manning the boards have pulled off a minor miracle in turning potential slag into something highly listenable.

The non-stop wall of sound, which must be an ear-ringer "live," is intelligently mixed on all cuts, allowing for clean, undistorted guitar lines and the often-tricky setting of bass and drum levels. More than one heavy metal effort has been turned to sludge by producers who insist that "bottom" sound should stand on an equal basis with guitar and vocals. Happily, the Ramones' rhythm section is brought to the fore only for occasional fills and nothing self-indulgent.

A striking plus is the way vocals are set above the blast-furnace roar of the instruments. Joey Ramone's street-corner refrains are positioned in cleverly arranged relief that allows all elements to have a blow without stepping on each other's toes.

Based on this effort, I hereby award Messrs. Leon, Freeman and Hunerburg this year's "silk purse from a sow's ear" recording award. M.S.

TODD RUNDGREN: ***Faithful.*** [Todd Rundgren, producer/engineer; recorded at Secret Sound Studios, New York, N.Y., Bearsville Studios, Bearsville, N.Y., Ramport Studios, London, England, Agency Rec'g Studios, Cleveland, Ohio, Sound Masters (Div. of Nashville Sounds Inc.), Houston, Tex.,

The Thinking Man's Re-Issue Series: Blue Note/U.A.

By H. G. LaTorre

Whenever a record label begins its promotional campaign for a re-issue series I scurry for cover like a fox being chased by the hounds. Therefore, it took some clever baiting by stubborn associates to convince me to even poke my head out from hiding to look at the Blue Note re-issue series.

Surprise, surprise.

First, the performers are grade "A." From Lester Young to Chick Corea. Secondly, performances are unique, solid efforts ranging from "Bop" to "Modern." There are sessions by pianists Andrew Hill and Herbie Nichols—those by Nichols being some of the best music contained in the series. They shed light on some extremely appealing and musically complex compositions. The ideas and directions that Herbie Nichols developed in these tracks (consider the fact that all are under six minutes long) would still be strong and innovative. It's good to have these recordings back in the mainstream and available to everyone.

Other highlights are classic work by Sonny Rollins "live" and a most appealing T-Bone Walker blues album. The only questionable selection is the Paul Horn *In India*, showing Horn's and America's infatuation with Eastern Indian music during the Sixties, which seems to me a bit dated. Perhaps a re-issue of the album, in a different series would have been better.

Blue Note, of course, was a forerunner in the jazz recording movement and has one of the longest and respected histories in the field. Therefore, the amount of material is quite large, having been compiled since the 1930's.

Probably the most difficult task in the plan was identifying the artists and recording background. Few of the storage crates had been properly labeled (if at all), so production could begin only after thorough examination of each tape. Most of the tapes were very clean, considering storage procedures and age. However, the iron oxide had worn off on some, thereby creating "drop-out" problems where the tape had become transparent. Also, on many, the old splicing-tape adhesive had oozed out from pressure and made the tape stick together. This required reeling off the masters by hand because the machines would have snapped sections off. Then the old splicing tape was replaced with new.

Some of the original recordings had been done on ten-inch 78 rpm acetates which had to be cleaned before being transcribed onto a master tape. Filtering processes were employed only when absolutely necessary. Most of the originals, on the other hand, were recorded on mono and two-track machines. Believing that the performance is *the* thing, producers of the series Charlie Lourie, Michael Cascuna and Peter Welding, along with United Artists' studio manager Dino Lappis, opted to leave well enough alone and issued the releases in their original format. They also chose not to electronically re-channel the mono tracks. Many classic recordings have lost their singular intensity by being put into a much broader spatial environment than the original instrumentation could survive.

The album liner notes indicate that a number of the recordings were done in home or garage settings, which is true, but sophisticated equipment (for the period) was used and professionals were engineering. Many of the recordings were done by Mr. Rudy Van Gelder—the mysterious, low-profile genius—at his New Jersey studio. In the industry, to say that Van Gelder has done the tracks is to say that the job has been properly done.

Limiting the use of noise reduction did not affect the overall sound quality of any of the albums. The Lester Young tracks were originally an Aladdin Records release, so United Artists cannot be faulted for the increased noise level.

This is a re-issue series with creative foresight, taste and intelligence. Even included are instructive and informative liner notes. It all goes to show that a little care goes a long way.

A Fresh Legend and Other New Horizons

By Nat Hentoff

Endemic to the jazz microcosm are misty legends of musicians who have exiled themselves to those parts of the country where critics and A & R men seldom venture. At great removes in time, ecstatic word may come back to New York or Los Angeles from local musicians who have played with the legend; but words are not music. And so I am skeptical of these distant marvels.

Only most rarely is my skepticism demolished. An instant case is the first recording in ten years by Ira Sullivan, a onetime Chicago-based jammer with Charlie Parker and other undeniable legends (because they recorded a lot and the music is *there*). Sullivan, however, seldom recorded even when he played where the action was, and since he moved to Florida in 1962, it's almost all been hushed-word-of-mouth. But now, in *Ira Sullivan* (part of A & M's exemplary new Horizon jazz series), there is no question that the man is a post-graduate improviser with a freshness and wholeness of conception that makes me urge you to get this recording for the sheer and unusual pleasure of its authentic, unstrained originality.

In the set, Sullivan plays trumpet, alto, soprano, and flute. He is in thorough technical command of each; has a distinctive and not in the least distorted sound on each; and turns out to be, I now think, the most full-blooded flutist in jazz. On some tracks, he is given to overdubbing, a practice I usually consider a felony (or at best a misdemeanor), but Sullivan has the taste and musicianship to transmute this technological shell game into artistry. Sullivan also has an utterly secure sense of swinging time, but unlike some authoritative swingers, he is expansive rather than narrow in his mastery of jazz rhythms. That is, he can keep on surprising you without losing the pulse. So can his wholly unknown Florida associates, among them guitarist Joe Diorio, pianists Tony Castelano and Alex Darqui, and drummer Steve Bagby.

As is almost invariably the case in this Horizon jazz series which A & M has wisely commissioned, the effect of the engineering sounds to me as if the technician in charge were also somewhat of a musician. Not only is there clarity and spaciousness of sound, but also the manifest goal is to satisfy the musicians on the date rather than, let us say, a hi-fi salesman looking for a demo disc to use in the shop.

That quality of engineering was especially essential in the first studio recording by The Revolutionary Ensemble (*The People's Republic*). This astonishing trio—capable of a sustained, intense inventiveness that is almost on an overwhelming par with Cecil Taylor's—has been represented on record only by two "live" and hard-to-find sets. Recognizing the crucial difference between actually making a record and being taped "live," The Revolutionary Ensemble should acquire a good many new listeners. Leroy Jenkins, easily the preeminent violinist in jazz, doubles on viola and percussion; and his colleagues, also multiple-armed, are bassist Sirone and drummer Jerome Cooper. The trio primarily moves texturally, the rhythms exfoliating in fierce colors. But unlike many "avant-garde" jazz action painters, these three work within over-arching, explosively *organic* designs. That is, they are not jiving.

IRA SULLIVAN: *Ira Sullivan.* [John Snyder, producer; Mack Emmerman, engineer; recorded and mixed at Criteria Recording Studios, Miami, Fla.] Horizon SP-706.

REVOLUTIONARY ENSEMBLE: *The People's Republic.* [Ed Michel, producer; Baker Bigsby, engineer; recorded and mixed at Kendun Recorders, Burbank, Cal.] Horizon SP-708.

Streeterville, Chicago, Ill.] Bearsville BR 6963.

Performance: **Uneven**
Recording: **Technically strong**

The musicianship on this record is clear and solid, probably due to the group's reduced number of players. The line-up now consists of Todd, John Siegler, bass, John Wilcox, drums, and Roger Powell, assorted keyboards. Missing are keyboard men, Moogy Klingman and Ralph Schuckett. Also, it seems that Todd's "wizardry" on this album shines through more in the actual music than in the display of knowledge of electronics. Not that his engineering or production is found lacking—the engineering is in good taste rather than the barrage of synthesized madness prevalent in some of Todd's other endeavors of late (e.g., *Initiation*). Production remains on equal ground with the music.

The only saving grace on side one, which consists of remakes of other artists' material with no attempt at originality in arrangements, is an almost perfect rendering of "Good Vibrations" (Wilson, Love) which is so near the original version it's frightening. The others are pretty much a waste of time. Musically they are "faithful" to the originals (hence the title of the album?), but the vocals are very noticeably different. Therefore, they neither offer anything new nor serve as good reproductions. The worst mistake on this side is "If Six Was Nine" (Hendrix) which dies on the turntable for 4:55.

Todd obviously went to great lengths to get the music down so perfectly on these tracks, and even though the tracks are engineered brilliantly (keeping the players' individual style neatly out of the way) one must ask, "Why bother?" An artist with a writing ability of Todd's caliber needn't lean on rehashes of other people's material.

Side two improves 100%, sounding more like the Rundgren most of us like to remember. This side is a perfectly balanced set, starting with a good rock 'n' roll number, "Black & White," and ending with the cynical Mr. Rundgren at his cynical best with "Boogies (Hamburger Hell)." In between are tasty ballads in that unmistakeable Rundgren style.

The overall sound of the side is comparable to *Something/Anything* in performance, quality and material—

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reminiscent of Todd when he was "Runt." There's the slick production number "The Verb 'To Love' " and, in "Cliche," Rundgren playing the sensitive kid we haven't seen for awhile. The only weak spot on this side is "When I Pray," which sounds like a poor attempt at reggae. Even with this ear-sore, side two is a perfect Todd sampler for the uninitiated—produced, engineered and played with the quality we have come to expect from the Wizard.

Real Todd followers will find the album worthwhile for side two alone. However, others may feel a little cheated. If you replace "When I Pray" with "Good Vibrations," you end up with half an album at best. So, there it is, a weak side with one bright spot and a bright side with one weak spot. Let's trust that side one is a satisfied whim for Todd and that side two is the shape of things to come for us. C.F.-K.

TONY TRISCHKA: *Heartlands.* [Tony Trischka, producer; Bill Storm, Les Tyler, engineers; recorded at Pyramid Sound, Ithaca, N.Y.] Rounder 0062.

Performance: **Wonderfully crooked**
Recording: **A-1**

ANN MAYO MUIR, ED TRICKETT, GORDON BOK: *Turning Toward Morning.* [Sandy Paton, engineer; recorded at Folk-Legacy, Sharon, Conn.] Folk-Legacy FSI 56.

Performance: **Natural**
Recording: **Lovingly gentle**

The stories of these two albums are the stories of but two folk music-oriented labels that make use of local or home facilities in the production of first-class professional recordings. Indeed, because the runs on albums such as these are low when compared to the releases of the "major" labels, quality control is nearly absolute.

Rounder is a folk music collective based outside Boston, recording a variety of new artists as well as releasing re-issues of considerable historical interest. *Heartlands* is Trischka's second album for the company and represents as well as any the kind of freedom that these labels can offer an artist without forcing compromise. Trischka is a banjo player who, with the assistance of such noted session men as Kenny Kosek, Andy Statman and Roger Mason (once of David Bromberg's band) carries the basic bluegrass form ahead two hundred years. The sound is busy because everybody plays solos at the same time. Yet the clarity that one has come to expect from better known outfits is there and probably improved upon because the recordist is familiar with the style of playing and sympathetic to the artist's goals. (One wonders how many major-studio engineers are familiar with what they record before the musicians arrive at the studio.)

Folk-Legacy's perspective and output are a little more limited, but the integrity is there. Sandy and Caroline Paton remodeled and attached the barn alongside their house in Sharon, Connecticut and that large, natural wood room serves as their beautiful-sounding studio, Sandy at the controls of a four-track Tandberg either in the room itself or through a door at the dining room table.

The three performers heard on *Turning Toward the Morning* are traditionally-based musicians and singers with a particular love of the sea, singing as well of love and balladic heroes and playing two instrumentals. Again, the crispness of what was undoubtedly a "live" recording is omni-present and the loving involvement of all concerned can be heard throughout. Folk-Legacy, as with many folk music labels, also includes a booklet with lyrics and background information with each lp. I.M.

CLASSICAL

STRAVINSKY: *Concerto for Piano and Wind Instruments*; *"Ebony" Concerto*; *Symphonies of Wind Instruments*; *Octet for Wind Instruments.* Netherlands Wind Ensemble, Edo de Waart cond. [W. Holleg, producer; recorded in the Concertgebouw, Amsterdam, Holland.] Philips 6500.841.

Performance: **With it**
Recording: **Warm with good depth perspective**

STRAVINSKY: *Octet for Wind Instruments*; *Pastorale for Violin and Quartet of Wind Instruments*; *Ragtime for Eleven Instruments*; Septet; Concertino for Twelve Instruments. Boston Symphony Chamber Players [Thomas Mowrey, producer;

Hans-Peter Schweigmann, engineer; recorded at Symphony Hall, Boston, Mass.] DG 2530 551.

Performance: **Poker-faced**
Recording: **A bit flat**

These are both notable additions to the Stravinsky discography. It is heartening to see that new recordings are being made in repertoire which has been limited mostly to the composer's historic series on Columbia.

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The Philips disc seems the best to me. The *Concerto* is well played by Theo Bruins—a less percussive performance than some of a work which can be unduly severe. The Dutch are known for their love of American jazz and these players really make something out of the *"Ebony" Concerto*, unlike the composer's somewhat uptight recorded performance. The *Symphonies*, which the composer never recorded in stereo, is fine also. But in the case of the *Octet*, I miss the dry wit and pert accents of Stravinsky's own version. The Dutch and Boston players are certainly accomplished and more refined, but the composer gets more biting staccatos from his winds despite a rough moment or two.

The Boston disc is musicianly throughout, but I occasionally sense a lack of character, a rather literal response to Stravinsky's often outrageous musical humor. The composer's own *Ragtime* reading, for instance, seems about to trip over its own clownish feet, while the Boston version has every note in its over-careful place. Nevertheless, the record is well worth hearing—if only because it *is* better played.

Both discs are well-recorded, the Philips acoustic having more depth than the DG, where the players all appear to emanate from the same distance. Surfaces were excellent. S.C.

TCHAIKOVSKY: *Symphony No. 4.* New York Philharmonic, Leonard Bernstein cond. [John McClure, producer; Bud Graham, Ray Moore, John Johnson, engineers; recorded at Manhattan Penter, New York, N.Y.] Columbia XM 33886.

Performance: **Disappointing**
Recording: **Ill-defined**

What is one to say about such records as this? It had all the earmarks of a superb disc at the exciting concert performances which preceded the sessions—and then someone announced, "Take 1."

In general, the conception is adequate. The breadth of the first movement is impressive and the coda energetic, but it sounds like a run-through; it lacks the electricity of the

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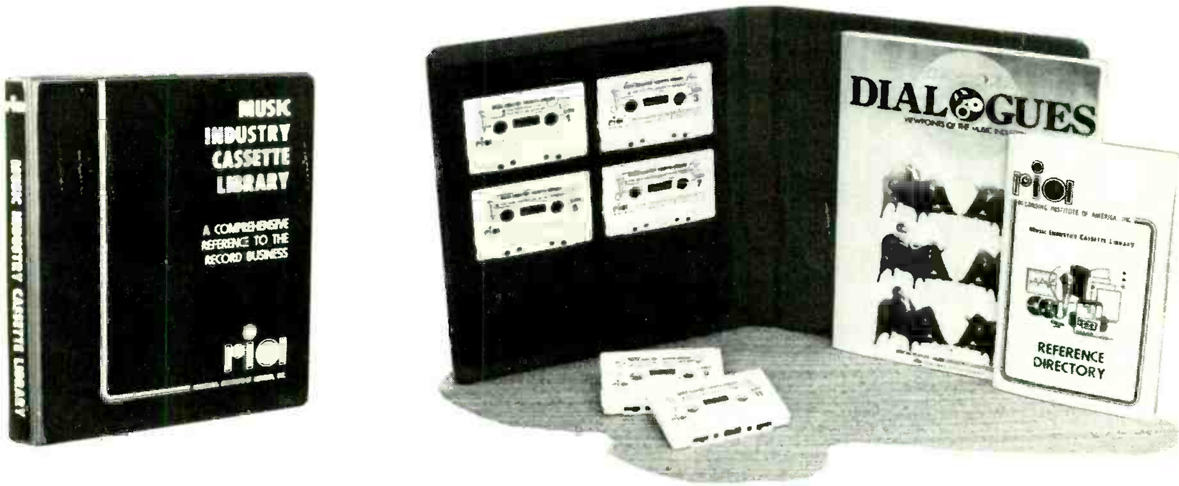

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"live" concert. Nowhere in the over-conducted second movement is Bernstein's careful balancing of wind passages—the major distinction to his performance—in evidence. The pizzicato strings in the third movement are appallingly fuzzy, especially the cellos and basses. The fourth movement goes well enough, even counting minor, unnecessary tempo adjustments and exaggerated pauses when the fate motive returns.

As for the sound, who would believe that the soft-focus murk on this disc could result from the same company, orchestra and recording location as Boulez's Stravinsky *Firebird* reviewed in the Apr/May issue? The major difference, of course, is the production.

Whereas Boulez's producer Andrew Kazdin and his engineers achieved rich, glowing textures, Bernstein's producer John McClure (with engineers Bud Graham and Ray Moore from the *Firebird* crew and John Johnson) has opted for a similarly plush, but diffuse, monochromatic sound. What seems to be conservative mic placement renders the strings almost totally without presence, and the backward balancing of the horns, ill-defined timpani banging away somewhere off in a corner, and general clouding of textures are quite different from what Bernstein achieved in the drier ambience of Avery Fisher Hall. Certainly the conductor should have heard at playback that his meticulous wind/string balances were not registering on tape. Producer McClure has written that Bernstein doesn't like to record at Manhattan Center. And, of course, the acoustics of these two locations are drastically different. But one should think that Boulez's production team has shown without doubt that superior results may be attained at this extremely resonant location.

A final sour note is the amount of extraneous studio noise from this most slovenly of major American orchestras. We are spared the loud cough in the first movement of Bernstein's 1958 recording with the Philharmonic of this symphony, but the third movement of the new recording contains an unbelievable amount of shuffling and banging of bows which all but drown out the woodwinds when they begin the first trio; later, a sharp blast of air (presumably, a wind player cleaning his instrument) is added to the proceedings. Does no one care? S.C.

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