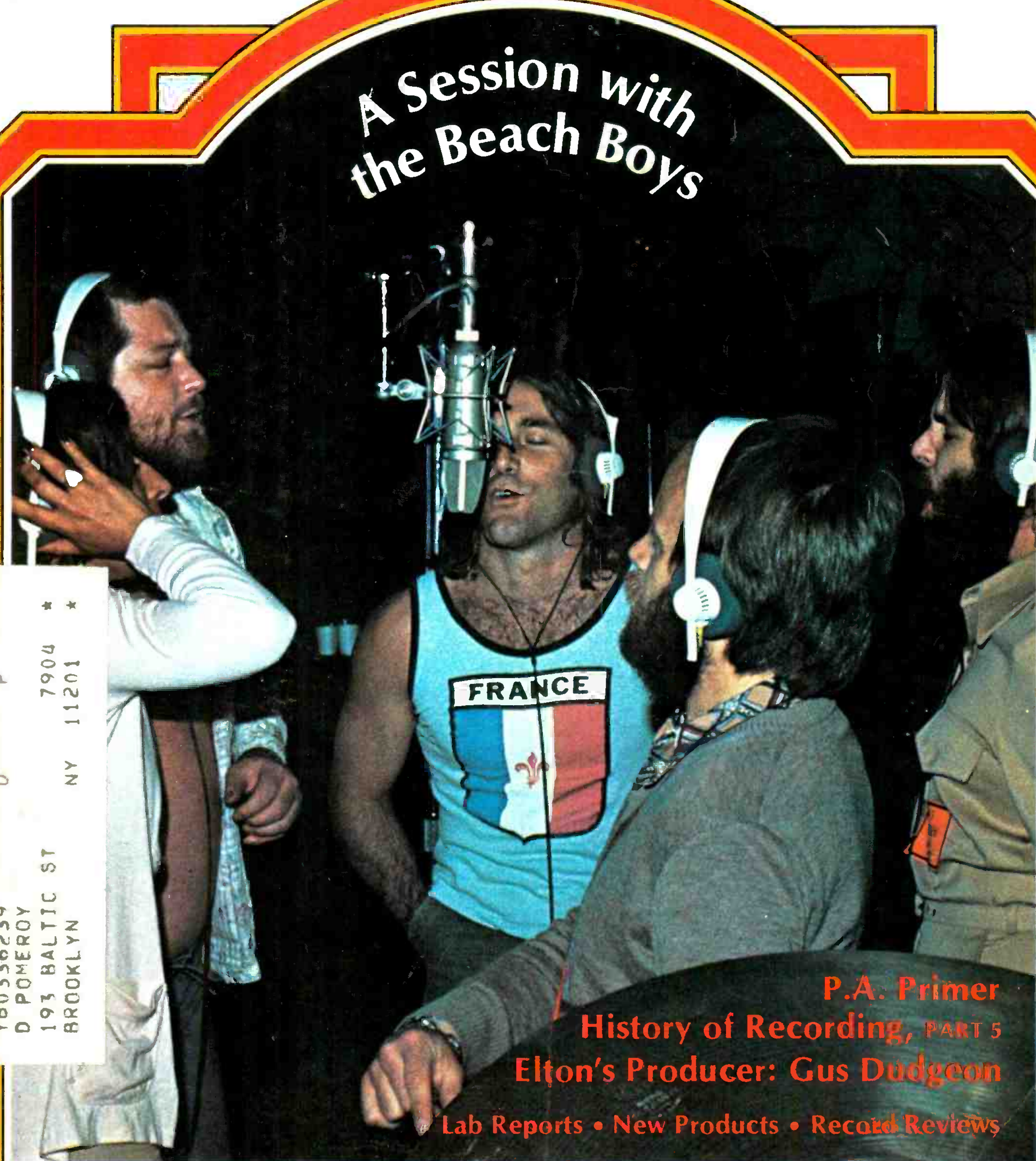
JUNE/JULY 1976

VOL. 1 NO. 5
$1.50

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY



A Session with the Beach Boys

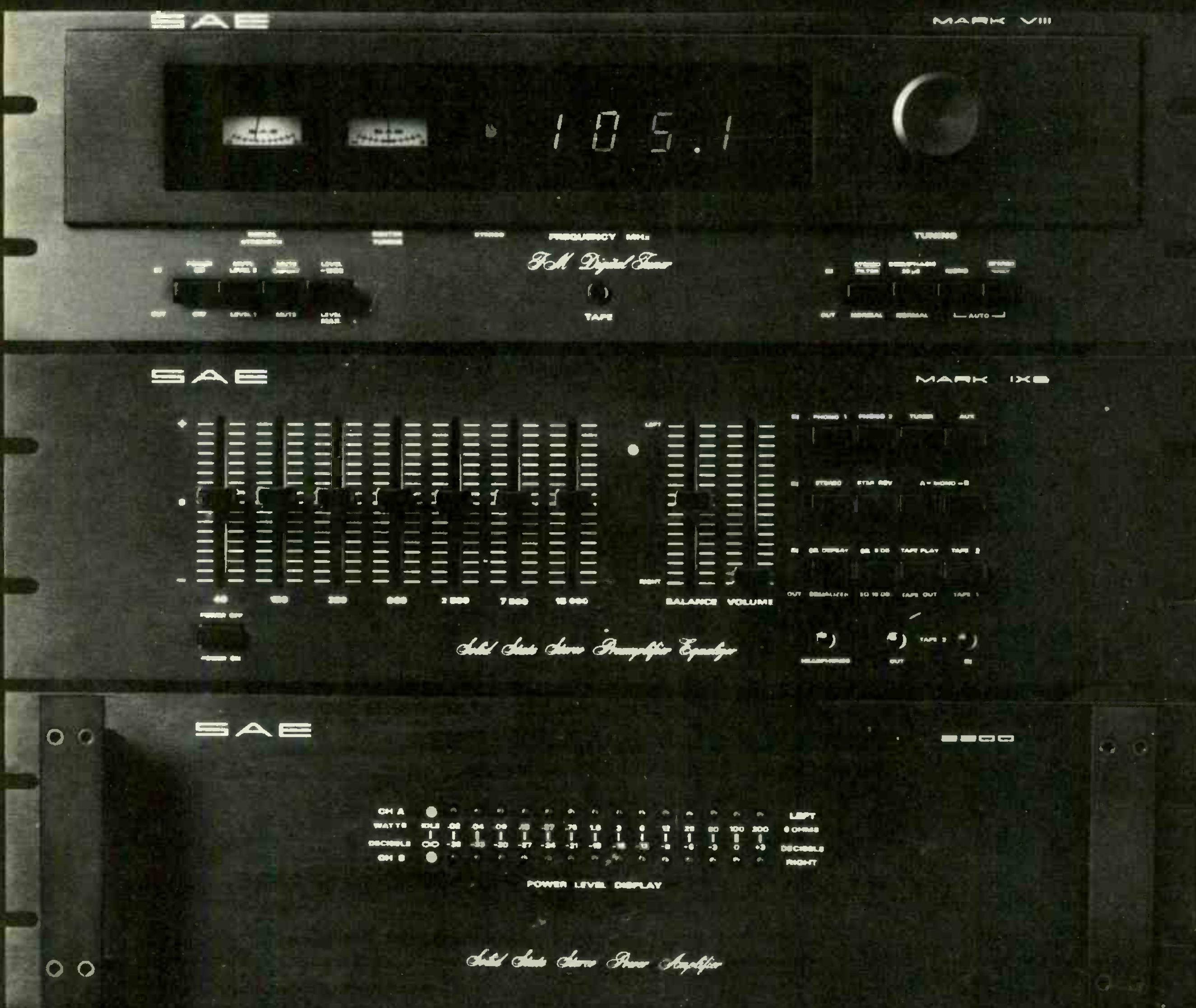
P.A. Primer

History of Recording, PART 5

Elton's Producer: Gus Dudgeon

Lab Reports • New Products • Record Reviews

In the Black



Suggest Retail of System shown-Mark VIII($650.00), Mark IXB($400.00), 2200($450.00)-Total System $1500.00

They say that you can't judge a book by its cover; that's true, but what a cover. What was, and is, the most beautiful look in the professional field is now the most daring look in general audio. BUT looks are not the whole story, in fact, not even the best part. Inside—that's where you find true SAE quality and performance. Here are just a few highlights of this SAE system:

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MR-6-76

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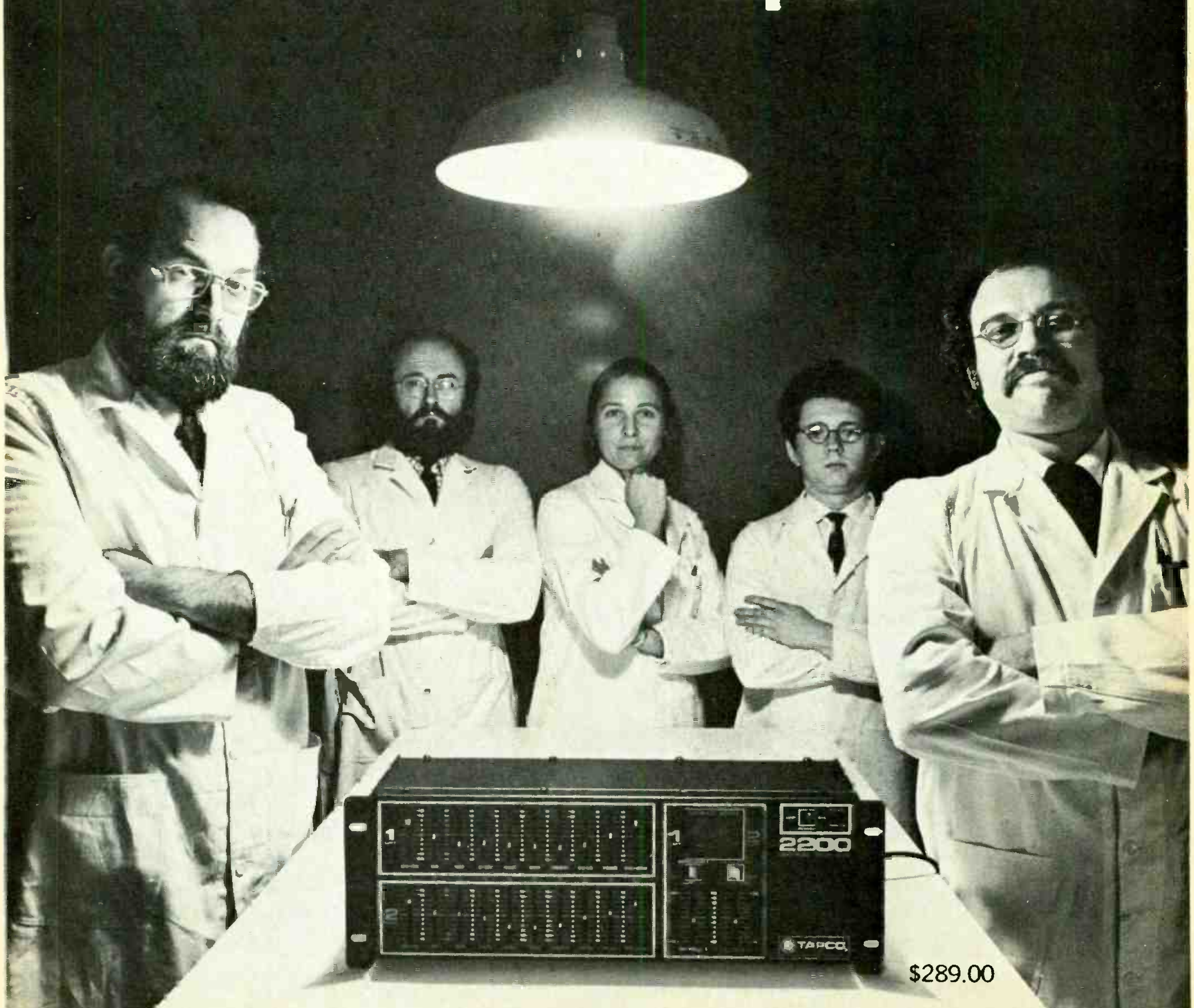

For more information write: Wayne Inouye, Tapco, 405 Howell Way, Edmonds, WA 98020 (206) 775-4411

CIRCLE 79 ON READER SERVICE CARD

JUNE/JULY 1976
VOL. 1 NO. 5

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

THE STAPLES

COMING NEXT ISSUE!
A "Live" Session with Fleetwood Mac
History of Recording, Part 6
Tanglewood's "Live" Broadcast Facilities
P.A. Primer, Part 2

Cover Photo by Ed Roach. Pictured (left to right): Marilyn Wilson, Brian Wison, Dennis Wilson, Carl Wilson, Al Jardine.

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The Editor's Mixdown

Regular readers of *MR* will notice immediately that this issue is considerably larger than our previous ones—due primarily to an increase in advertising, but also because of a broader range of editorial content. As our subscription and advertising bases increase, so too will our editorial portion expand and broaden. Letters to the Editor and Talkback questions are coming in with increasing rapidity. We are pleased to note, especially, that readers are beginning to offer answers from their own experience to Talkback questions. This is very important, for, if anything, we are particularly eager to establish *MR* as a means of *communication* between its many readers.

Our features this issue are worthy of note, I think. Robert Angus' popular History of Recording series is up to Part 5. This installment covers one of the most fascinating and dramatic events in the development of recording. Who's to say how long it would have taken American technicians to develop the tape recorder if Jack Mullin had taken the *other* road in the fork and returned to the U.S. after World War II without those famous German Magnetophons?

We also have an interview this issue with Gus Dudgeon, the acclaimed producer for Elton John and many other fine artists. Gus talked so much about his work with Elton that we felt the interview transcended our regular Profile format, which deals specifically with the personality being interviewed.

When we began planning the P.A. article, we decided that we'd better do it well or not at all. We commissioned two experts in the field who specialize in P.A. design—and from there the monster grew out of one issue's control. Hence, we have the principles and theory in this issue, and the practical application part will come next issue.

Our Session article this time is special, also. It was hard enough to find the Beach Boys in their studio, but this time we got 'em with their leader, Brian Wilson, in charge. Brian has only been intermittently active in the past few years, and his full-time return to the group adds a historical element beyond the routine session to this report.

Even our Lab Tests are exceptional this issue. Well, one of them, anyway, because *MR* is once again venturing into uncharted territory as we run a *comparative* test on two important new turntables—one a perfected model of the straight-arm tracking design and the other an epitome of automation. It's certainly among the most interesting (as well as the most exhaustive) lab tests yet to appear in *MR*. Let us know, readers, what you think about this two-in-one comparative review. Write us a letter, though—don't call by either by phone or CB. Soon after our third issue, we received an urgent call from a young man in Alabama who had just read our review of the McIntosh MA-6100 integrated amp. It seems he had been debating whether or not to buy the Mac unit when he saw our review. Since he had never seen a review of a Mac unit in any other magazine, he wanted Len Feldman to plan out all the rest of the components to match the amp. That is one service that we just can't provide!

S.C.

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We've taken the latest advances in electret technology one step further. By combining them with advanced acoustic technology to make professional condenser microphones more portable, more practical and less costly. A lot less.

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Letters to the Editor

More "How to" in MR?

Congratulations on the introduction of this new publication, *Modern Recording.* As an amateur recording enthusiast and audiophile, I find your magazine informative and very interesting reading. I enjoy your magazine from cover to cover. I like the fresh look at the role of recording in the making of modern music. I like your policy of equipment reviews, that of more subjective listening tests with less emphasis placed on ambiguous number specifications. Will you be including periodical construction features in your magazine, such as mixers, preamps, equalizers, etc.? Keep up the excellent work!!

—Delbert Udy
Calgary, Alta, Canada

"How to" features will of course play a major role in MR's editorial planning. The Dec/Jan 1976 issue contained an article entitled "How to Build Your Own Recording Studio for Under $500." This issue presents the first of two articles on the design and application of sound reinforcement systems. And we anticipate many more to come.

Wall of Sound Reverb

Your feature on the "Wall of Sound" (Feb/Mar 1976) took me "someplace I had never been before." Thank you and congratulations on producing the best mag. for all of us recording "nuts" out here.

—Charlie Musser
Manheim, Penn.

I hope you won't take too seriously Larry Levine's flat statement regarding successful artists sustaining their creative flow (Feb/Mar 1976).

I'm pretty sure with a little forethought Levine would not have made the gross generalization that "every creative artist, sooner or later, runs out of places to go." I tend to agree almost 100% with him but *only* as regards artists who let themselves be coerced by the all-too-common corporate "bottom-line" attitude to the point of changing for the sake of change (excuse my rambling).

Recognize please the very distinct difference between the relatively mature, creative being expressing himself because "the feelin's in the boy and it gots ta come out," regardless of criticism and the dude not unlike the junior executive whose every move is governed by the bottom line and his image in the eyes of his superiors.

This letter is written simply as an attempt to "unpaint" the bleak ending Levine has painted of the demise of "every creative artist." Lord knows a serious young artist needs no outside discouragement.

I in no way am attempting to demean Levine's own particular abilities but there is a world of contrast between merely "interpreting," and being a prime creative force such as Miles or Dali.

. . . really! Comparing the Beatles to Herb Alpert.

And another thing, you've (we've) got a good magazine. It fills a definite need.

Good luck and thanks for your time.

—Boot
Washington, Mi.

Recording Demystified

Thanks for the very rewarding article on "Recording Techniques: Then & Now" by Jim Furman [MR, Oct/Nov 1975]. It completely demystified the recording process for me.

Also, would you please review the Teac 3340S tape deck in Product Scene?

—Neil Hall
Pleasant Hill, Cal.

"The Affair" Lives

Advertising has just come to our attention quoting your January 1976 issue as follows:

"To quote from *Modern Recording* Jan. '76, 'The Oberheim is the only stereo synthesizer on the market, and Stevie is delighted with the added dimension. The instrument is polyphonic in that four sounds can be produced simultaneously. Most Arps and

The Vako Polyphonic Orchestron is the brainchild of Dave Vankoevering. Mr. Vankoevering pioneered the development of the market for solo-line synthesizers used in many contemporary groups. As a distributor of synthesizers, and later as marketing vice president for Moog Music Inc., he helped bring synthesizer technology to the performing musician.

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Moogs are able to produce only one sound at a time . . . With four more units on order, Stevie will soon have an eight-voice model.' "

"The Affair," product of CONCERT COMPANY (Model 8000—introduced in 1973), is a two-channel (stereo) synthesizer capable of real polyphonic performance in counter melody, including simultaneous production of as many notes as there are keyboard intervals available, achieving complex harmony structures.

Further, "The Affair" is multiphonic having the capability of simultaneously producing more than one voice-modulated wave shape, or mode as well.

Other poly-voiced products of CONCERT COMPANY go as high as 25 synthesized voices from which to select, with simultaneous production of tones at every interval.

We would appreciate your advising your readers of the correct facts, with a copy to us.

—D.S. McNally
General Manager
Concert Sales Company
Inglewood, Cal.

Experiences and Vice-versa

My cousin and I are budding musicians/composers who have realized that the only way to work out our compositions without spending hundreds of dollars in studio costs is to record demos ourselves, at home. We have no questions, but we figure there must be quite a few more people in the same situation who would profit from our experiences, as well as vice-versa. Anyone interested in trading solutions and questions can reach us at the following address:

R. & J. Manwiller
1422 Margaret Street
Laureldale, Pa. 19605

Our equipment consists of a Dokorder 8140 4-track, a Sony 630D 2-track, a Sony 330 2-track reel/cassette, a Peavey 800 8-to-2 mixing panel, and several cheap-to-moderately expensive AKG and Shure microphones.

—Richard Manwiller
Laureldale, Pa.

P.S. Does Michael "Tapes" Colchamira always sign his name twice?

That little mistake was a printer's stripping error. And that's why editors are more apoplectic than ordinary humankind. —Ed.

Readers Offer Own Talkback Solutions

In your Feb/Mar issue Talkback section, Eric Fussell asked a difficult question on recording with a Teac 3340S. I've owned a 3340S for over six months and I've found that if you patch directly from line out on track 2 to line in on track 3 before recording anything, you can record on tracks 1, 2 & 3 in sync. But be sure track 2 stays on source.

Glad I could help.

—Roy Peak
Aurora, Colo.

In Vol. 1 No. 3 [Feb/Mar 1976] there was a [talkback] from a person with a Teac 3340S who wanted to know how to make a tape he'd recorded on tracks 1 and 2 of his machine into a normal stereo tape. Although it is true that you can't transfer track 2 to track 3 in sync, this maneuver can be accomplished by first transferring track 1 to track 3 out of sync and then transferring track 2 to track 1 out of sync. This would keep Mr. Fussel from having to borrow a tape recorder.

—John Vandiver
Houston, Tex.

Stockhausen and Tangerine Dream

I'd be interested in seeing some articles on electronic music—tape technique, synthesis, etc.—and particularly something on the new studio Karlheinz Stockhausen is putting together, and generally what he is up to now. In the same vein, an article on the recording of Tangerine Dream's new album, *Ricochet*, might be instructional (they're not Stockhausen, but what they're doing is pretty interesting). The album was recorded "live," and I understand that a number of their recent concerts have been presented in cathedrals in France, Germany and Britain. It also seems that the performances were improvised, all of which would present some novel problems in recording.

Keep up the good work.

—Lin Naylor
Mount Joy, Pa.

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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.

Two leading hi-fi magazines working independently tested a wide variety of cassettes. In both tests, TDK SA clearly outperformed the other premium priced cassettes.

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CIRCLE 38 ON READER SERVICE CARD

TALK BACK

"Talkback" questions are answered by professional engineers, many of whose names you have probably seen listed on the credits of major pop albums. Their techniques are their own and might very well differ from another's. Thus, an answer in "Talkback" is certainly not necessarily the last word.

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

Monitoring a P.A.

Your policy of getting "talkback" answers from professionals in the field is certainly to be commended.

Here's a question: Many times I have heard it proposed that the level of an auditorium's sound system might be better monitored when a microphone in the listening area was used to feed an output indicator located at the amplifier controls.

When the controls are located backstage somewhere it does seem the idea would have merit—so the level wouldn't have to be set blindly (deafly?).

Has such a system been tried—and what are the pros and cons?

—Hugh Lineback
Siloam Springs, Ala.

Theoretically, that sounds like it would work. However, there are several problems with this approach.

The typical meter will show only average levels, whereas "live" program sources contain large peaks in level. Consequently, it is entirely possible for the sound system amplifiers to be driven into clipping distortion, while the meter indicates "all is well." Also, the meter will tell nothing about tonal balance or the program mix.

While it is possible to attach headphones to the monitoring circuitry, this still represents a very bad way to control a sound system. I cringe every time I see someone operating a sound system while listening with "cans"—I have yet to hear anything but garbage from a sound system controlled in this manner.

The only reliable method for operating a P.A. system is to do the mixing and monitoring in the room itself. Any other method has severe limitations.

If this is not possible, I suppose that the backstage metering set-up is better than nothing.

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

Noise Reduction and Pitch

I have a Teac 3340S deck which I use to make multi-track recordings of my original music. Six to 12 tracks are often required, which necessitates several mixdowns and add-to dubs. Obviously, noise becomes a problem. I've toyed with a Dolby unit I have, but find it too cumbersome for my use.

I've investigated both Burwen and dbx and find their operation to be less complex—Burwen having the least complicated approach of all. My question is this: I want to eliminate the noise (primarily hiss), but I don't want to have to lug one of these noise reduction units around with me for audition playbacks.

If I use either of these systems at all recommended stages of recording, including between the final playback mixdown to the recording stereo master, will that master retain the program integrity (with no noise or attenuation) when played back without being connected to the unit?

(2) I remember reading somewhere that a variable-frequency power supply could be constructed to act as a pitch control for a tape deck. Could you discuss how this might be constructed and any hazards that might be inherent in such a device?

—Jay Stewart
Cleveland, Ohio

(1) All noise reduction devices must be used during playback to reduce tape noise. Using a noise reduction device to encode the final mixdown on the stereo master recorder will significantly change the program's sound and be of no help in reducing noise if the complementary decoder is not used during playback. In all of the systems, it is the decoder not the encoder which reduces the noise.

An important function of the encoder used in compander-type systems is to counteract the effect the decoder would otherwise have on the program so that the program will emerge from the encode-decode cycle unchanged. Since the noise is not encoded, the decoder reduces it. Non-complementary noise reduction units such as Burwen's dynamic noise filter do not encode the program and are designed to filter out noise without too much effect on the program. They still must be used on playback, however.

Since you are using noise reduction at all recording, dubbing and mixdown stages, you have already eliminated the major sources of noise build-up. The solution to your problem is to use noise reduction on your stereo master, but decode that master on playback to make non-encoded dubs for auditions. The audition copies will have only one generation of non-reduced tape noise and should be quite acceptable.

(2) A variable frequency AC power supply can be used as a pitch control on any tape deck with a hysteresis-synchronous capstan motor. The operating speed of this type of motor is determined by the frequency of the AC driving it. A power supply of this

type is often called a VSO, for "variable speed oscillator," and is easily constructed by using a variable-frequency sine wave oscillator to drive a power amplifier such as a Bogen MO100, McIntosh MC275 or any other amplifier capable of a 115-volt audio output with about 35 watts of power.

Besides the obvious hazards of dealing with 115-volt power, care should be taken to connect only the capstan motor and not the tape deck electronics to the VSO, for the electronics may be adversely affected by variable-frequency power as well as any variations in VSO output voltage. Care should also be taken to prevent electrical connection between any part of the VSO chassis and the tape deck, console or other recording equipment, for this can cause ground loops and supersonic oscillations. The VSO has to be connected to the capstan motor and the AC power line, nothing else.

A voltmeter should be used to test for any voltage between the VSO chassis and the tape deck or console. If any exists, reverse the oscillator and/or power amp AC plugs in their wall sockets one at a time until this shock hazard is eliminated. Operating the capstan motor at too high or low a voltage or frequency can cause it to stall and overheat, reducing its life. The VSO output should be kept between 90 and 115 volts. Most motors will safely operate between 30 and 90 Hz, which produces a range of ½ to 1½ times normal speed.

—Robert E. Runstein
Author,
Modern Recording Techniques

Recording and Playback Levels

I am experiencing a situation in tape recording and I do not know if it is a result of my inability to use the equipment properly or if this is not possible to eliminate without more sophisticated equipment.

My components consist of Dual 1229, Pioneer 939 SX, Teac 3300's, JBL 100's.

The problem presents itself in my attempt to duplicate 78 r.p.m. records to tape.

I use Maxell UD 35-90 tape and Pickering's XV 15-400 cartridge with a mono stylus with the connections strapped out for this purpose.

When I first started recording in the stereo mode, the quality of tapes was very good. As I became more familiar

with recording technique and got into 78's, this is where I got lost.

My preliminary is to play a selection on phono and establish my peak VU to not generally exceed "0." Switching between Source and Monitor on the receiver, I bring the output to where I do not perceive a difference in the level of loudness in the speaker.

Up to this point, everything seems to be going well. Where I go into the record mode and all controls remaining in the initial settings, the VU meters show a good signal until I switch to tape monitor at which time they go to the total limit of VU.

My question is, does the output level have any effect on the tape as it is being recorded? Should the output be reduced to match the input signal?

I have tried recording both ways and I cannot mirror the sound quality of my phono onto the tape. In other words, my record sounds better than most of the tape. I am concerned over this because I don't know if I am over-recording and introducing the possibility of self-erasure, although I do not detect any dropouts on the tape.

Observing the meters bouncing like this startles me and I can record without monitoring, but then I don't know until playback whether or not I have laid down a good tape master.

Any information you may provide is most welcome.

—Dan Daniel
College Point, N.Y.

No, the output level pots will not affect what's being recorded on the tape, and so, in answer to the second part of your question, adjusting them will not improve your recordings in any way. However, if your machine electronics are properly calibrated, lowering the output to equal the input will accurately show you what level of signal is being recorded on the tape. In any case, do not rely on your ears alone to set your levels. That's what VU meters are for, and they are far more accurate.

In a properly aligned tape machine, playback levels are set by referencing to a standard level. A carefully recorded test tape is used to provide the standard. (If you want to check the calibration of your machine, the Teac YTT 1003 Test Tape for 7.5 ips would be adequate.) Once playback levels are set, they must not be altered, since now record levels will be set while recording and monitoring the tape playback on the VU meter (tape monitor). Once a good input level is established, the electronics are switched to monitor input (source monitor), in order to check the amount of gain going into the record amps. If properly calibrated, the machine's input and output should read the same (±2 dB is acceptable for most consumer machines.)

If you go this far, and you find that your machine is way out of alignment, I suggest that you either take it to a Teac service shop, or write for a Teac 3300S service manual and attempt to calibrate it yourself. I warn you, deal only with simple level adjustments; any further and you may be involved with bias, EQ, and other more complex calibrations—none of which should be touched without a thorough understanding of all principles involved.

Once you have straightened all that out, and you know the status of your machine, make a habit of monitoring tape playback at least part of the time while you are recording. (Switching between tape and source monitor will not affect what is on the tape.)

Whether you are recording 78's or regular lp's does not matter—a proper alignment will give you the optimum performance for your machine. If your problem still exists, check all components and lines feeding the input to your tape machine.

—Rob Freeman
Plaza Sound
New York, N.Y.

On-location Miking

The major problem that I have encountered with on-location recording is the method in which to mic the vocals. Until my mixer arrives I am only able to run four mics or lines (Teac A-7300-2T), and have had to resort to miking the P.A. columns. For now, and even after I have eight mics available, should I continue to mic the columns, in many cases ruining my separation and control, should I run direct lines from their mixer, or should I mic all vocals direct, adding problems to an on-axis signal and to space availability? Is it feasible to split-feed the line from the vocalist's mics (one for P.A. and one for my mixer)?

Also, could you briefly advise me as to mic placement and choice (dynamic or condenser) for the various instruments involved with "live" recording? Such as dynamic or condenser on vocals, drums, keyboards, etc.?

Please keep up the good work. Your magazine fills a large gap that has long been ignored—that between a consumer need and a studio's needs. Thank you.

—David B. Miller
Absolute Recording
Elkhart, Ind.

(1) If you are recording on-location projects as a business venture, I would suggest that you invest in enough equipment to cover your needs before your business disappears. Four microphone or line inputs will only give you the capability of capturing four sources properly. Unless you are recording a small act (two voices and two instruments, etc.), your best coverage will be obtained by continuing to mic the P.A. columns if you can live with the mix, equalization and quality presented by the system.

Since it is virtually impossible to mix for a "live" audience and a two-track tape machine with the same set of controls (unless the total system was designed with such capabilities), your idea to split-feed the microphone lines would provide the best possible solution. Your first consideration should be to determine your monetary limits and try to compromise between quality and ease of performing a particular function without going too far beyond your predetermined budget. There is a boundless supply of mixing consoles, recording machines and peripheral equipment available in a wide range of quality and price. Through wise selection of your equipment, you can do a professional job with a modest investment.

(2) The choice of microphones and their placement is a question that leads to other questions. The type of music and its content (referenced to levels and dynamic range) plays a greater role than one might expect. A preferred sound is often obtained by the engineer's personal choice of microphone and its placement. Let's not forget that the purpose of recording is for the storage of a performance as it is, so that it may be reproduced for later enjoyment. In "live" recording, you would probably get better results by using as many "direct boxes" as possible in order to get better separation.

These "direct boxes" can only be used on electric instruments and would save you money since they are cheaper than an average quality microphone. This would also eliminate a certain amount of noise due to the fact that no other sound would be able to enter these lines. As for vocals, my personal preference is to use condenser microphones because of their general ability to capture the higher frequencies in a clean manner. I would use a professional-quality dynamic microphone on the toms in the drum kit, with condensers on the cymbals. There are many high-quality microphones available that will do a good job. Nothing will replace the experience of having to work in different rooms to find the best microphone for a given situation. If you are looking for specifics, try to find an old AKG-D19 in good condition and mic the snare drum with it from the top.

—Richard H. Royall
MEGA Sound Studio
Bailey, N.C.

Bal and Unbal Line Wiring

Can you explain:
(a) What are 600-ohm Bal and unBal lines and how are they wired? Also, echo send and return lines.
(b) What type of wire is used for these lines?
(c) Is there any complete information concerning studio wiring guidance?

—Steve Mayland
Bend, Ore.

(a) We can break this question down into two parts: (1) What is a 600-ohm line? (2) What are balanced and unbalanced lines? 600 ohms is the resistance from the high to the low side, or ground. 600 ohms has emerged from electronic history as the semi-standard of most professional audio equipment, although others exist, such as 250 ohms for microphones and 50,000 to 100,000 ohms for input bridging.

A balanced line is a pair of wires with an identical signal traveling down each, hence balanced. Normally this pair would be shielded for audio use, so we call this a three-conductor cable. The advantage of a balanced line is the ability to reject clicks and external noise. The counterpart to the balanced line is the unbalanced or single-ended line. In this system the signal travels down just one (usually shielded) conductor. And this wire is referred to as a two-conductor cable.

(b) Most any good audio wire can be used for echo send and return, the only consideration being balanced or unbalanced circuitry. If the send or return is balanced, a three-conductor wire should be used, and two-conductor for unbalanced.

(c) One of the most complete books of information on studio wiring and all related audio information is *The Audio-Cyclopedia* by Howard Tremain, published by Howard W. Sams.

—Travis Turk
Freelance Engineer
Nashville, Tenn.

Tricky Teac 3340S Techniques

I own a Teac 3340S tape deck. In a booklet published by Teac called *Meet the Creator,* it states that seven individual tracks can be laid down *without going beyond second generation.* I assume the technique [involves] taking tracks one, two and three, and mixing them to track four. Then record new material on tracks one and three and mix them to track two, leaving tracks one and three open again for the remaining two parts, hence seven tracks.

But how are tracks one and three in sync with track four when they are being mixed down to track two? Or, what am I doing wrong? (I have Teac's mixdown panel AX20.)

—Bill Wojciechowski
Towaco, N.J.

You can actually record up to nine tracks in two generations on a four-track like the 3340S but it gets tricky.

Fill up channels 1, 2 and 3 in the normal manner. Then put channels 1, 2 and 3 in the tape position. Simul-Sync is in the normal position. The line outputs of your 3340S are feeding the inputs of an AX20. The AX20 switches for channels 1, 2 and 3 are in the left position. Use the output level controls on the 3340S to get the relative balance of the three tracks.

All three signals are coming out of the left output of the AX20 as a composite mono signal. This signal is fed into the line input #4 on the 3340S.

Meanwhile, you can add a fourth voice through the Mic input on channel 4. You can refer to Mic/line mixing in your owner's manual if you like, although it does not mention this application.

For this stage, you may need a friend to help you because it's rough to get a good mix and play at the same time. If you are having trouble, simply forget about the fourth voice and do a mix of tracks 1, 2 and 3 without adding a new "live" part. You may lose one part, but you may also save yourself a good amount of frustration.

Here comes the trick to make it work!

You are now going to record a new part on track 2. The trick is to have channels 1 and 3 in sync. You record track 5, listening only to channels 1 and 3. DO NOT MONITOR CHANNEL 4.

Record track 6, listening to channels 1 and 2 in sync. Still, do not monitor channel 4.

The next step is to put channels 2 and 3 in the tape position. The Simul-Sync switches are in the normal position. The line outputs of channels 2 and 3 are feeding your AX20. This time you connect your left line out of the AX20 to the line input of channel 1. You can add part 7 with Mic/line mixing.

Now you use channel 4 for sync monitoring!

Put channel 4 in the tape and Sync position. You record your composite mix of channels 2 and 3, plus your "live" part onto channel 1, while monitoring your earlier composite of parts 1, 2 and 3 on channel 4.

You can now record tracks 8 and 9 on channels 2 and 3. These are first generation and located on the inner tracks. This is good for vocal and other important parts.

Tracks 1-4 are on channel 4 and tracks 5-7 are on channel 1. These are only second generation. Track 8 is on channel 2, and track 9 on channel 3.

And the whole thing is in sync. It's a little tricky to do all this, but if you plan your parts carefully you can expand your creative horizons quite a ways.

—Theo Mayer III
Training Manager
Teac Corp.
Montebello, Cal.

Distancing

How does an engineer manage to make an instrument sound as though it is in the distance? I have heard the technique used mainly with reed instruments. Is echo/reverb used or are the instruments miked from a greater distance?

—Robert Chappell
Fort Wayne, Ind.

Yes, the use of echo/reverb, as well as miking the particular instrument from a somewhat greater distance will make an instrument sound as though it is in the distance. Also, by experimenting with the echo return delay, it will help in achieving the desired effect.

However, in miking the particular instrument from a greater distance, you should be especially careful not to obtain any unnecessary leakage.

—Stan Dacus, Chief Engineer
LeFevre Sound Corporation
Atlanta, Ga.

Slapped Back

What is slapback?

—Henry Rosen
Farmingdale, N.Y.

I think the "slapback technique" might have been what I was subjected to the last time a tape salesman informed me of a price increase.

The questioner is probably referring to a discrete echo effect derived from a time delay introduced into an audio path. It is often referred to as "slap echo." It occurs when a signal is delayed and then mixed with the undelayed signal so that two distinct sounds are heard. They may overlap, becoming confused, or they may be distinct punctuations of the "hello . . . hello" variety. This can only happen if the original signals are of short enough time duration, and spaced far enough apart.

The time delay can be obtained from a tape deck, digital delay unit, or by using the delays available from some of the new flanging devices. A tape recorder operating at 15 ips will generate delays of 6.67 milliseconds (thousandths of a second) for every $\frac{1}{10}$-inch separation between your record and playback head gaps. A couple of common studio machines have a gap spacing of about 2.5 inches. This yields a time delay of at least 166.2 ms, between $\frac{1}{10}$ and $\frac{2}{10}$ of a second. Such long delays are clearly audible and all too easily identifiable.

When the time delays get short enough, under 40 ms, then frequently the ear can't distinguish them as separate sounds and they begin to integrate. The effect then can simulate doubling the instrument (you may feel it sounds like two instruments playing together or that the instrument is being played in a different room). You hear a "slap echo" which is part of the acoustic presence of any "live" room. The separation of the two sounds is very much dependent on the duration of the original. A wood block can be broken into two wood blocks with time delays as short as 10 ms.

It can be quite interesting to double a snare drum back beat or to create new-sounding percussion instruments. Instrument doubling is not usually satisfactory because of the phase distortion caused by the delays. The synthesizer is an excellent instrument to utilize with slap echo in the creation of unusual sounds. It can be very impressive to add echo to the slap, only creating a large "ba-BOOOM" effect. Once you have delayed a signal and can mix it with the original you have opened a door which can be used to create many interesting sounds.

—Ed Rehm
The Ken Nordine Group
Chicago, Ill.

Decoding dbx

Does a dbx signal have to be decoded before mixing and equalization? Will I

need six channels of noise-reduction to mix four to two tracks? I'm trying to figure if the cost (about $1500?) of dbx would be sensible with my existing equipment (Sony 854-4 deck, Allen & Heath 'Mini' Mixer).

—Bill Mauchly
Ambler, Pa.

To ensure proper encoding and decoding of your dbx signal, all equalization and level mixing must be done after the signal has been decoded. Trying to alter an encoded dbx signal before decoding would result in the expansion stage not being able to decode (expand) the signal with the mirror image exactness needed to retain accuracy. This would require six channels of dbx for mixing four channels dbx to two channels dbx. The benefits of superior signal-to-noise ratio and dynamic range are well worth the cost, if maximum quality is your goal.

—Bill Mueller
Sheffield Recordings, Ltd.
Timonium, Md.

What Is "Disco" Sound?

I just received my first issue of *Modern Recording*—read it cover to cover, and *loved* it! It's interesting and superbly informative. I especially gained from your "talkback" section.

I have a question that I have been pondering for some time now, and I believe it will be of particular interest to a great many of your readers.

Since "disco" music is such an extremely lucrative and commercial field right now, I would like to know:

(1) What is the main difference in the "disco" sound from what we have been used to? Is the difference more in recording or mixing?

(2) Please show how to get the "disco" sound with special reference to mixing and panning the drums and bass.

—Jim Robson
Mesa, Ariz.

The differences between the disco and the standard rock sound can be as superficial as the same record with the words "DISCO VERS" on one side of the label, or as radical a difference as the entire concept—from the arrangement to the playing, recording, mixing and mastering techniques. But usually it is simply an unedited longer version of the commercial copy.

If that answer disappoints you, let's analyze it further. When studio-type people sit down between sessions and talk shop, the conversation usually begins, "Just what makes disco disco?" And each person in the room has a different opinion of what it is. Then, finally unable to verbalize it, someone sits down at the old trap set in the drum booth and starts banging on the sock with the (of late) familiar beat, accompanied by the appropriate rhythmic patterns on the snare and bass drum. And that, everyone agrees, is it . . . although no one has actually said exactly what *it* is.

In a conversation with a top producer of disco records who is currently on a roll, I posed that very question to him. There was total silence for a moment and then with a lot of stammering he had to admit that he didn't know the answer either; it was just something that evolved. Now here's a top disco producer who doesn't know and you're asking me!?

Apparently, disco isn't anything that different. But the noticeable things are a lush string section, simpler bass line, a very danceable beat—combined with the carry-over from bubble gum music of such sexual innuendoes as "You're just the right size." Exactly what makes each disco record will vary from artist to artist and from producer to producer.

So now we can assume that it is just ordinary music, and the technical end does not pose any special problems. If you process a disco record exactly as you would process a "standard" record, you will come out with a disco record. Remember that the bass line is a little simpler and the bass drum is very predominant to hold up the danceability; so you would want that out in front. If the bass drum isn't in the arrangement (where it should be) and you have to help it on the studio end, obviously you would make the low end as punchy as possible by emphasizing the upper lows for a more definitive and harder sound. Then, in addition, pull out the midrange to emphasize the lush strings and vocals.

So really what I'm saying is that the disco sound has to be in the arrangement and the recording will take care of itself (as with any recording)—or you have to make a patch job out of it (as with any recording).

In the mixdown—in any mixdown—panning the bass and bass drum to any place but dead center is lunacy (psychopathic in fact), but the rest of the drums are given the usual bigger-than-life spread across the stereo stage which produces the effect of a go-around with the drums . . . left tom on the left, right tom on the right, etc.

Often, the only place a disco recording differs from the "standard" is at the mastering stage (because the producer has no idea of how to produce a disco record and says to the mastering engineer, "Give it a disco sound"). This is not the place to fix things up for any type of recording, but more often than not that's where it happens. So you try to pull a hard, punchy sound out of the mud on the bottom and emphasize the midrange enough to almost shear your ears off. Perhaps a bit more limiting than the usual goes into it at this point also. But if the record were intended to be a disco from the beginning and is well arranged, you will have no problems.

—Dave Moyssiadis
Frankford/Wayne Recording Labs
Philadelphia, Pa.

THE POWER HOUSE

If you're looking for a rugged, dependable stereo power source for the studio or the concert stage, consider Acoustic's Model 400 Stereo Amplifier.

Designed for normal operation at 8, 4, or 2 ohms, this 19-inch rack-mountable industrial power amplifier features 2 totally independent channels rated at 375 watts RMS each. Because we've been building state-of-the-art professional sound reinforcement equipment for over a decade, we've got the design know-how to make the Model 400 virtually immune to the problems of overloading and overheating that plague most other units.

The Model 400 features L.E.D. overload indicators on the front panel to indicate "true clipping" in the amplifier and guard against distortion at the input. Plus the 400 never quits. It's fan-cooled and designed to operate under the most stressful conditions. If ever needed, a unique thermal switching circuit automatically steps down the power by 1/3 to prevent overheating, while leaving you with plenty of juice to keep on cooking. Like all Acoustic products, the Model 400 can be covered for life by our unique "Protection" Plan.

The Model 400 Stereo Amplifier . . . rugged enough to travel anywhere, and flexible enough to handle the requirements of the most sophisticated concert stage, industrial or recording studio systems. Now you know why we call it The Power House!

acoustic

Professional Sound Reinforcement.

For more information write to Dept. MR
Acoustic Control Corp., 7949 Woodley Ave., Van Nuys, CA 91406

CIRCLE 35 ON READER SERVICE CARD

MUSICAL NEWSICALS

NEWS . . . *Synapse* is a new monthly magazine ($10/year) devoted to electronic music. Vol. 1 No. 1 (Mar. 1976) contains an interview with Alan R. Pearlman, a feature on devices for controlling synthesizers, a calendar of electronic music performances throughout the country, an examination of graphic equalizers, and a new-products section. *Synapse* is published by Contempo Publishing (*Synapse*, P.O. Box 359, N. Hollywood, Cal. 91603).

KEYBOARDS . . . The Roland Corp., known for its electronic musical instruments, has just unveiled a new product called the "Piano Plus," which claims to be the first completely electronic piano with acoustic sound and feel. So proud is the company of its new offspring that 28 patent claims are being issued!

Some of the most important elements of the Piano Plus are: 27 different sound combinations may be obtained, including the sound of a grand piano, a light piano or a harpsichord (and various combinations of these primary tones). Tuning is unnecessary. Weight is only ⅓ that of a conventional piano. Headphones may be plugged in for private listening. The Piano Plus may be plugged into a large amp/speaker system for stage performances. The spinit design is both stylish and refined enough to allow it to fit into any home, however, and Roland promises that its "Revo Sound System . . . will fill your performance room with widespreading rich sounds." Price is $1575 for the 88-note model and $1395 for the 75-note model (Beckman Musical Instruments, Inc., 2925 S. Vail Ave., Los Angeles, Cal. 90040).

GUITARS . . . All Guild flat-tops, 12-string flat-tops and classic guitars are now available with factory-installed Barcus-Berry "Hot Dog" transducers. The pick-ups are placed inside the bridge with two pearl dots visible on either side of the bridge saddle. Output jack, end-pin and "Hot Dog" preamp box are provided (Guild Musical Instruments, 225 W. Grand St., Elizabeth, N.J. 07202).

The B.C. Rich "Seagull" solid-body electric ($995) was handcrafted by master guitar builder Bernard Roco. It features a one-piece neck which eliminates the heel, phase and splitter switches, and a six-position varitone circuit (L.D. Heater, Beaverton, Ore.).

The Artwood Artist solid-body electric guitar Model #2617 ($395) is now available from Ibanez. The ash body and maple neck are hand-shaped, and the rosewood fingerboard features pearl-abalone block-style inlays. The Artwood Artist has two humbucking pick-ups for increased frequency response and added punch and

presence. A three-way toggle switch controls the pick-ups along with individual tone and volume controls (Elger Co., Cornwell Heights, Pa.).

The Badass Bass Bridge ($40) is a massive-weight bridge designed to provide for substantial gains in attack and sustain. Especially recommended for Gibson and Fender basses, this bridge has 1¼-inch horizontal and ⅜-inch vertical travel and is made of high density zinc with machined brass inserts (Leo Quan Badass Musical Products, 2831 Seventh St., Room 21, Berkeley, Cal. 94710).

AMPS . . . The OMEC digital programmable amplifier ("about $1,700") enables the musician to preset four different combinations of effects. Treble,

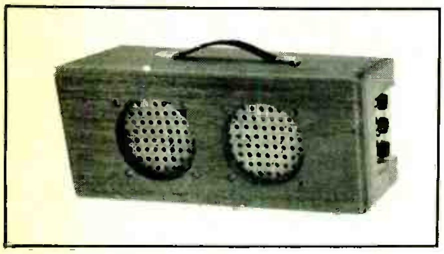
midrange, bass, reverb, sustain (compression-limiting), distortion and volume are independently programmed into each combination through the use of illuminated push-buttons and L.E.D. display. The desired combination can then be chosen and modified during performance with the accompanying foot-switch and mating cable. This compact (24″ x 30″ x 7″), lightweight (less than 30 lbs.) amplifier delivers 150 watts RMS and is compatible with any good musical instrument speaker system (Orange Musical and Electronic Corp., R. A. Neilson Co., 5001 Laurel Grove Ave., N. Hollywood, Cal. 91607).

The "Dwarf" ($124.95) and "Dwarf Mitchell" (in production) are mini-amps for guitar and bass, respectively. Housed in a hand-made ash cabinet, the "Dwarf" is a 12-foot tall, 7- watt amp, driving a 5¼-foot high performance speaker. The "Dwarf Mitchell" bass practice amp will be 18 inches tall and have 20 watts pushing an 8-inch bass speaker in a folded horn. Both units feature speaker plugs for private headsets or an external speaker (21st Century Products, 62B Hamilton Dr., Ignacio, Cal. 94947).

The Commander RG-120-210 ($479.50) produces 120 watts RMS at less than 1% THD. The twin 10-inch speakers are housed in a compact (27½" x 17" x 10") cabinet (Randall Instruments, Inc., Irvine, Cal.).

Lamb Labs' cable extension box ($600) has a single 200-foot cable connecting a multi-connector cable panel (15 female and five male XR connectors, with two in parallel) on stage to an equivalent panel back at the mixing console. This arrangement minimizes the problem of multiple cables from stage console (Lamb Laboratories, Inc., Syosset, N.Y.).

Electro-Harmonix has announced the Freedom Brothers Amp ($249.95). This small portable amp runs on a battery that recharges itself automatically when in use. It is rated at 15 watts RMS and has two heavy-duty

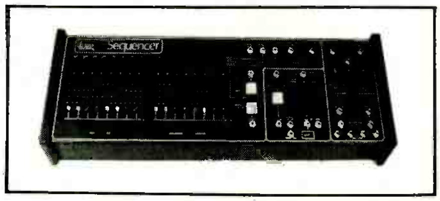


5½-inch speakers. Volume and tone controls are supplemented by a "bite" control, a 5,000 Hz boost which accentuates pick noise (Electro-Harmonix, New York, N.Y.).

MISC. INSTRUMENTS AND EQUIPMENT ... The QSC 1.0 portable six-channel mixer/preamp ($398) boasts bi-polar integrated circuits which enable it to handle anything from tape decks to heavy-duty amplifiers. This 6½-lb. unit has hi & lo impedance jacks, active gain control, five-band graphic EQ (18 dB range) available independently on each channel, reverb/effects send and return and two separate outputs with master volume controls (Quilter Sound Co., 1936 Placentia, Costa Mesa, Cal. 92627).

The Micromoog ($695) is a portable synthesizer (24" x 14" x 5½") featuring a zero inertia pitch-bending ribbon, sample & hold, separate contour generators for filter and amp, and reversible filter contours. The 20 lbs. Micromoog is a variable—as opposed to preset—instrument and is compatible with other synthesizers, instruments and accessories (Norlin Music, 7373 N. Cicero Ave., Lincolnwood, Ill. 60646).

ARP has designed a 16-note sequencer for "live" multiple-keyboard performance. Bass, percussion and other musical lines of up to 16 notes, or eight two-note chords can be programmed, in sequence or randomly,



freeing the musician's hands for other purposes. Dual-quantizer circuits permit fast, chromatically scaled tunings with variable accents, rhythm patterns and dynamic changes. The notes are tuned with linear sliders, and push-button switches allow the musician to skip, reset or start a sequence at any point The sequencer includes five gate outputs, a pulse width modulation control and pedal jacks for external control and interfacing with additional sequencers or synthesizers. The ARP sequencer should be available April 1, 1976 and will list at $795 (ARP Instruments, Newton, Mass.).

The Shure PE52 Vagabond noise-cancelling microphone ($105) is a close-up mic designed to shut out sound generated more than an inch away. This uni-directional dynamic microphone features built-in filters to minimize sibilance, breath noise and feedback. The unit comes wired for high impedance inputs and is accompanied by a 20-foot cable. (Shure Brothers Inc., 222 Hartrey Ave., Evanston, Ill. 60204).

The MXR ten-band graphic equalizer ($139.95) can be used as a sophisticated tone control for electric instruments or in P.A. systems to compensate for room acoustics or for feedback control. This AC-powered unit is capable of up to 12 dB boost or cut at each of ten frequency bands (31.2 Hz to 16 kHz by octaves). A stereo model is also available for the home or studio. (MXR Innovations, P.O. Box 722, Rochester, N.Y. 14603).

Kustom's new mixing consoles ($4,500 to $7,000) come with 12 to 24 inputs and 8 outputs, but the user can add as many modules ($220 each) as desired. The console has -0, -10 and -20 dB input pads, dual LED preamp overload indicators, echo send (pre/post), low-impedence inputs and five-band EQ (Kustom Electronics, Chanute, Kan.).

Leslie has two new stereo headphones. The W-2 ($14.95) is lightweight and audio-acoustically designed to eliminate outer ear variations in bass response. The W-4 ($44.95) offers three bass-response positions and a high fidelity driver that uses a thin mylar diaphragm vibrating like a piston to provide the sound (Electro Music, Pasadena, Cal.).

The WT10 Korg Tuner ($159.50) is a compact unit for audio and visual chromatic tuning of instruments over a three-octave range. The built-in microphone permits tuning of low and high volume instruments, while there is an input jack for silent tuning of electric instruments. The WT10 operates on four penlight batteries or AC current (Univox, Merson Musical Products, 75 Frost St., Westbury, N.Y. 11590).

ACCESSORIES ... The FRAP Wind Pick-up was designed for woodwinds, brass and drums. Like the FRAP Contact Pick-up for guitars and other acoustic instruments, there are no moving parts, so the pick-up is virtually insensitive to extraneous mechanical noise (key clicks, valve noise, breath noise, etc.). The kit consists of transducer, preamp and connecting cable. The transducer comes with a screw mounting for brass, most woodwinds and drums. For flutes, the cork-surrounded transducer is attached to a custom-fitted silver endplate. The FRAP Wind Pick-up comes in two models, determined by the preamp. The W-200 ($500), powered by four 9V batteries, is good for 100 hours, and the W-250 ($600), powered by 18 C-cell batteries, is good for 1,500 hours. Frequency response for both units is flat from 20-20,000 Hz (FRAP, Box 40097, San Francisco, Cal. 94140).

THE PRODUCT SCENE

By Norman Eisenberg

TDK DROPS CHROME; ADDS NEW TAPES

TDK has replaced its KR (chromium-dioxide) cassettes with a new tape called SA (for Super-Avilyn). SA cassettes are available in C-60 and C-90 sizes and are recommended for use with CrO_2 biasing. At the same time, the company has dropped its ED (normal bias) cassettes in favor of a new line known as Audua. According to TDK, several cassette deck manufacturers have begun to recommend the use of SA as an alternative to CrO_2 cassettes. (Advent, as you might expect, is not one of them.) The Audua tape, with and without special back treatment, also is available in open-reel format in lengths from 1,200 to 3,600 feet.

According to TDK, Audua tape offers improved high-frequency performance, lower noise levels, and greater reliability and durability than any previous tape. The coating is made of a "special" magnetic oxide powder using particles smaller than those found on other tapes. It is applied by a new process that forms a dense coating of uniform, tightly packed particles oriented along their axes.

CIRCLE 7 ON READER SERVICE CARD

SOUND GUARD SAVES RECORDS

The newest offering for cleaning records and keeping them clean is a product known as Sound Guard, introduced by the Ball Corporation. It is essentially a dry lubricant that comes in liquid spray form. According to Ball it contains "VacKote," a lubricant designed for moving parts on space craft, plus cleaning anti-static fluids. When sprayed on a disc surface and buffed, it leaves a coating said to be 5-millionths of an inch which, while it bonds to the record, does not bond to itself and so will not form a coating build-up that could interfere with frequency response.

Sound Guard has been lab-tested and is credited with preserving signal response over repeated playings of both stereo and discrete (CD-4) discs. Other findings include a reduction of the rate of harmonic distortion with repeated playings, and a smoothing effect on pulse noise and random spiking. The tests also indicate that because the use of Sound Guard makes anti-skating force compensation less critical, record wear on equipment not using anti-skating should be reduced.

Sound Guard comes in a kit that contains a two-ounce bottle with pump sprayer and a buffer. One kit is good for about twenty 12-inch microgroove discs and costs $4.95.

CIRCLE 11 ON READER SERVICE CARD

PHANTOM-POWERED MICROPHONES

A series of phantom-powered studio condenser microphones wth RF-biased transducer elements has been announced by Sennheiser. The mics are designed to operate with 48-volt phantom-powering systems where the positive supply voltage with reference to ground (pin 1) is supplied through the audio lines (pins 2 and 3) of the XLR connectors. Acoustically, the new mics are similar to previous Sennheiser mics in its MKH series. Three models are available now: the $529 MKH 416 with narrow super-cardioid pattern; the MK 406 with cardioid pattern and higher overload (up to 132 dB/SPL) and priced at $495; and the MKH 816 shotgun mic ($629). For installations not yet equipped with central phantom-powering, Sennheiser offers a separate dual power supply (model MZN, $176).

Other recent Sennheiser mics include electret modular condenser units built around a model K2U powering module that employs a 5.6-volt battery, said to last for about 600 hours. Shotgun and cardioid heads are available as well as alternate impedances and outputs, plus many accessories from windscreens to shockmounts. For more information, contact Mr. Kees Hofman at Sennheiser Electronics Corp., 10 West 37 St., New York, N.Y. 10018.

CIRCLE 16 ON READER SERVICE CARD

THE LINE F

Trine Corp. has announced a new stereo phase-flanger known as "The Pipe" (Model SF-3). Priced at $369, it offers variable resonance which is claimed to change tonal coloration to represent a pipe ½-inch in length to 32 feet in length, thereby simulating a tonal range from a flute to an organ. Its linear voltage-controlled delay range is said to be 1,000 to 1 which allows you "to create that accelerating swish through infinity." It also can add a natural (or unnatural) vibrato to voice or instrument. The device also provides control-voltage input and output, allowing up to 20 units to be ganged together or controlled from automated mix-down equipment. "The Pipe" has its own AC power supply. Other new items from Trine include a complementary multimode parametric equalizer; a noise-reduction system; and a sound-system measuring-package built around a pink-noise generator.

CIRCLE 9 ON READER SERVICE CARD

PEAVEY OFFERS EQUALIZER

The model EQ-10 is a new graphic equalizer from Peavey Electronics Corp., a Mississippi-based manufacturer. The device comes in a carrying case with handle and features ten-band equalization with nominal center-frequencies of 50 Hz, 100 Hz, 200 Hz, 320 Hz, 500 Hz, 800 Hz, 1.5 kHz, 3 kHz, 6 kHz and 12 kHz. Ten sliders handle the action and are calibrated from +12 dB to −12 dB in 2-dB increments. The front panel also has an input level knob plus switches for by-pass and for power. The EQ-10, with both balanced and unbalanced capability, retails for $200.

CIRCLE 15 ON READER SERVICE CARD

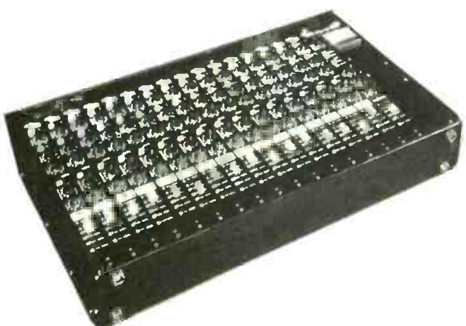
VERSATILE EQUALIZER/PREAMP

The PQ-3 by Furman Sound (of San Francisco) is a relatively low-cost parametric equalizer which includes a preamp. Self-powered, it may be used as a patchable outboard equalizer, as a musical instrument preamp, or as a general-purpose preamp to drive an external power amp in a reproducing or P.A. system. The parametric design enables the PQ-3 to tune to an exact frequency. The equalizer configuration covers three bands: bass (from 25 to 500 Hz), middles (from 150 to 2,500 Hz), and highs (from 600 Hz to 10 kHz). Three frequencies may be tuned and controlled simultaneously, and once a frequency is selected the exact amount of boost or cut may be adjusted.

Inputs and outputs are phone jacks and the unit may be installed in standard 19-inch rack mounts. Noise, with maximum gain and EQ out, is rated at -93 dBm; with EQ in and set flat, at -78 dBm. Distortion with EQ out is rated at 0.01%; with EQ in and set flat, 0.035%. The unit measures 19 inches wide, 8 inches deep, and 1¾ inches high. Advertised price is $250.

CIRCLE 20 ON READER SERVICE CARD

MODULAR MIXER



"Mom's Wholesome Audio" is the name given by Gately Electronics to its MicroMixer system which essentially is a modular set-up the buyer chooses to meet various needs in sound reinforcement and in recording work. Both input and output modules are selectable; price will vary, of course, with what is chosen. A descriptive brochure is available from the manufacturer, Gately Electronics, 57 W. Hillcrest Ave., Havertown, Penn. 19083.

CIRCLE 18 ON READER SERVICE CARD

SONAB BOWS IN WITH BROAD LINE

A name well-known in Europe and making a bid now for the U.S. audio market is Sonab. Owned wholly by the Swedish government, this manufacturer has set up sales reps across the country to supply dealers with its products.

Sonab offerings include speaker systems, headphones, receivers, turntables and a cassette recorder. In speakers there are several models using a patented "Ortho Acoustic" principle which is an omnidirectional concept involving bouncing sound off walls and ceilings. One of these, the OA-6, uses biamplification; price is $599. Top-of-the-line model is the OA-2212 which uses two woofers, two mid-range drivers and 12 tweeters; its price is $780. Other models range downward in price to $150.

The firm's entry into cassette recording is marked by the model C-500 with built-in Dolby, selector for standard and chrome tapes, and peak-value meters. Mic/line level inputs are concentrically arranged, and the recorder has a "center" input with its own level control in addition to normal left and right stereo inputs. Price is $399.

Sonab's final offering, at least up to now, is literary: the company has issued a 14-page booklet titled "Enjoying Music at Home" which may be obtained free at Sonab dealers or by writing to Sonab Electronics, 1185 Chess Drive, Foster City, Cal. 94404.

CIRCLE 13 ON READER SERVICE CARD

GERANTIUM PRODUCTS

Various speaker systems and a roster of electronic units are available from Gerantium Laboratories Associates, Inc., 45 York St., Brooklyn, N.Y. 11021. Speakers are grouped in the company's "Disco" line which now includes four models of which the top one is an amplified model (75-watt RMS amp driving a 15-inch woofer, an exponential midrange horn, and two ceramic element horn tweeters). Disco speakers are offered for high-level monitoring, discotheque, P.A., sound-reinforcement and similar uses.

Gerantium's electronic units include a signal processor (active equalizer); a preamp with special dubbing and patch jacks, plus three-band tone-equalizers and two sets of outputs for power amps; the model 3800 mixing console (facilities include two stereo phonographs, stereo playback and microphone, precue on inputs, etc.); a discomaster mixing consolette; a disco desk for installing turntables and related signal processing units; and the model 5880 mixing console that incorporates the functions of a phono preamp, mic preamp, line mixer, headphone amplifier and equalizer.

CIRCLE 19 ON READER SERVICE CARD

SPEAKING OF SPEAKERS . . .

There is, it seems, a fair amount of mythology dealing with speaker systems. Which of the following statements would you mark as true, which as false?

1. Strong directionality is desirable because it helps you localize specific sounds in a stereo program.
2. Big-size speaker cabinets are essential for big sound.
3. Several cheap speakers in one box can furnish sound as good as one or two expensive speakers.
4. A 100-watt speaker sounds louder than a 50-watt speaker.
5. Any speaker placed in a corner will produce more bass.

Here are my views on the above statements.

1. False. Strong directionality or "beaming" is actually a form of distortion that can fog the program and make it sound unclear. It also lends the sound a distinctly hard sheen when listening on axis; the sound off to a side is less prominent. Much better is some form of "controlled directionality" or "omnidirectionality."
2. True as far as it goes, but big sound can also be achieved with small speaker systems as long as they are fed enough high amplifier power (and assuming, of course, they are rugged enough to handle high power). Several small speakers, adequately powered, can blanket an area as effectively (maybe more so) as one or two big speakers.
3. Unless someone can come up with a miracle, chances are very good that an array of cheap speakers will emphasize their shortcomings over whatever benefit accrues from the lot. Also, do not be misled into thinking that the small drivers often used in some high-performing commercial systems are "cheap" versions you can pick up at a bargain counter in Radio Row.
4. Maybe. But if it does, it is not because of its 100-watt rating but rather because it is being fed with more amplifier power, or it is more efficient than the 50-watt speaker, or both. The "100-watt" rating refers not to the relative amount of sound a speaker puts out but only to its safe upper limit for power-handling ability.
5. True. The proximity of large surfaces (floor, walls, ceiling) will reinforce the bass output of any speaker.

Send in the "theories" you have been exposed to—whether or not you agree with them—and we'll try to discuss them in a forthcoming issue.

THE HISTORY OF RECORDING

By Robert Angus

PART 5: Mr. Mullin's Machine and Mr. Orr's Brown Paper Bag

For most Americans, the story of the tape recorder goes back only to World War II when two men brought it forcefully to the American consciousness by returning from Europe with a working recorder and with the means of making tape for it.

The two men were John T. Mullin and John Herbert Orr, both serving with the U.S. Signal Corps in occupied Germany. Their stories are typical of the drama which has surrounded the development of magnetic recording since its very beginning.

Just before the Allied invasion of Europe in 1944, Mullin was assigned to a Signal Corps laboratory somewhere on the southern coast of Britain. His assignment: to eliminate interference caused in a radio receiver the army was using. The unit had been developed in the United States, where it worked perfectly. But when it reached Britain, radar signals which didn't exist in the western hemisphere at the time began playing hob with reception. "It was not a difficult thing to solve, and besides I was intrigued with the job," Mullin recalled recently. "As a result, I used to stay up until two or three in the morning by myself in the lab. For companionship, I turned to the AM radio nearby and tried to find some entertainment. The BBC used to sign off at midnight, so I'd begin fishing around the dial until I found something else. It didn't take me long to discover that the best entertainment and music at that hour came from Germany."

Indeed, entertainment poured forth from German radio 'round the clock, interspersed with news and propaganda. "The intriguing thing to me," Mullin remembers, "is that the music was well played and obviously at times played by very large orchestras. That went on, hour after hour, all through the night. Well, of course, Hitler could have anything he wanted in those days. All he'd have to do was to say, 'You stay up and play tonight'—and they'd do it. But that didn't seem reasonable, even for a madman. The amazing thing about this music was that there wasn't any record scratch or breaks every few minutes to indicate that they were playing records. I was intrigued. I wanted to know how they did it."

Bing Crosby, in typical attire for show day in 1950, beside Mullin's two Ampex Model 200's.

Mullin was to find out the following summer when, as part of a Signal Corps team, he was assigned the job of following the retreating German army, picking up any electronic items of interest. "Another man and I had the assignment of checking out a tower on a mountain several miles north of Frankfurt. There wasn't much left when we got there, but there was a British officer on the same kind of assignment as ours. We walked around together, and somehow the subject of records and music and recording came up. He asked me if I had heard this machine down at Radio Frankfurt. I had heard tape recorders before—we'd been picking up abandoned battery portable jobs all across Europe behind the retreating Germans. So when he told me that this was a tape recorder, I thought that probably his ears weren't very good.

"When you come down the road from that mountain, you come to a main highway at the bottom. If you turn right, it takes you back to Paris. If you turn left, you're on your way to Bad Nauheim, which was where Radio Frankfurt was actually located. When we got to that junction, I turned left, following the guy's advice. That was probably the greatest decision of my life."

Radio Frankfurt at the time was being operated by the Armed Forces Radio Service. When Mullin and his companion pulled in late in the afternoon, they were greeted by the Signal Corps officers in charge. "They spoke in German to an assistant who clicked his heels and ran back into a room and came out with a roll of tape and put it on the machine. That's when I really flipped, because I had never heard anything like that. To my knowledge, there just hadn't been anything like that in recording before. You couldn't tell whether it was 'live' or playback, there was no background noise. I was thrilled. The man with me had a camera, and we went to work and photographed all the instruction manuals and all the schematics and took them back to Paris."

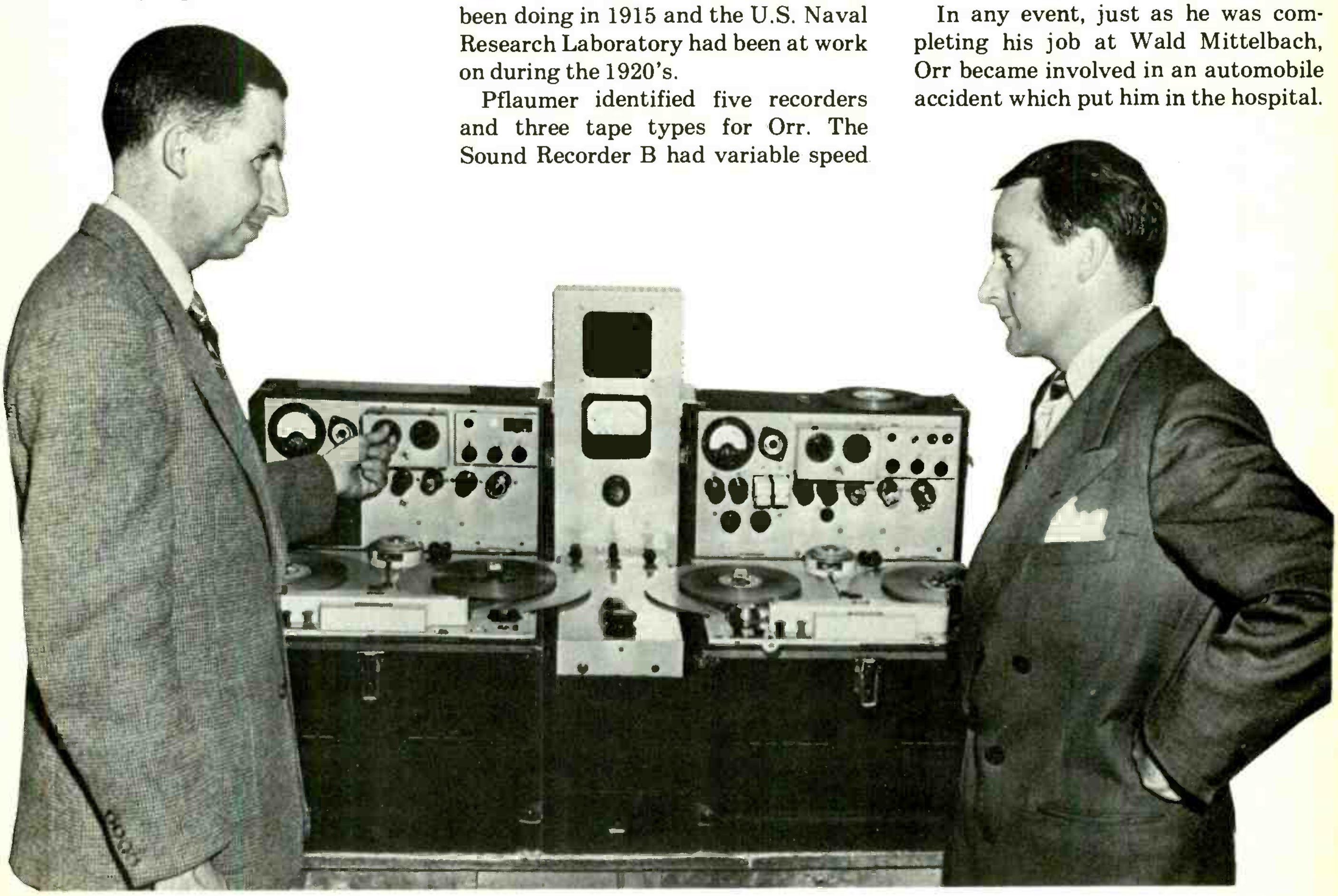
The Signal Corps wanted two of everything, so Mullin quickly rounded up two functioning Magnetophons, scrounged from Radio Frankfurt, Radio Munich and several other sources. Meanwhile, he also scrounged two units for his own use. These he disassembled carefully, crated, and mailed home to himself. "They had to be broken down into units small enough to fit in mail bags, or you couldn't send them," he recollects. "My biggest problem was getting the case into a mail bag, because the case had to enclose everything else. I made little wooden boxes for each of the motors and sent them separately." Altogether, there were 35 packages including tape, and fortunately all 35 arrived home safely.

In the months which followed, Mullin singlehandedly did what the president of Telefunken was unable to do—interest an American manufacturer in the Magnetophon. Telefunken's Dr. Heyne, trying to rebuild his war-torn company, took a Magnetophon to General Electric. That company called in a group of experts who took one look at it and told GE that the machine had no future, that it was impractical. Mullin was hampered because the only tape he had were the 50 reels he had shipped home to himself. At the time, nobody in the United States was making recording tape.

Unbeknownst to Mullin, another Signal Corpsman, Major John Herbert Orr, was doing something about that. Orr had been assigned the job of getting BASF's tape manufacturing plant at Wald Mittelbach back on its feet as quickly as possible. The plant had been established during the war in a rural area well away from Ludwigshafen, in order to avoid Allied bombing. Orr worked with Dr. Karl Pflaumer, a chemist in charge of the plant. Dr. Pflaumer told him that while the plant had kept on producing tape all during the war at a rate of some 3000 reels a month, the company really had no idea what the tape was being used for. AEG, he said, had been commissioned secretly to develop all sorts of specialized recording equipment, including broadcast playback sets, field dictation units and others. According to Pflaumer, the Germans visualized recordings transmitted by phone or radio at eight times recording speed to permit rapid, accurate communication—the same thing the shortwave station at Sayville, L.I. had been doing in 1915 and the U.S. Naval Research Laboratory had been at work on during the 1920's.

Pflaumer identified five recorders and three tape types for Orr. The Sound Recorder B had variable speed tape drive, and was produced for the Wehrmacht. At its slowest speed, it recorded for 70 minutes. The playback head rotated, permitting a recording to be made at one speed, then played back at a different speed without changing the pitch of the recorded signal. The fidelity was low. This was the machine Mullin kept finding all over western Europe. The Sound Recorder C was a portable knapsack model with spring motor drive for use by the Wehrmacht and radio correspondents. The Sound Recorder D was a combination recorder and transmitter for military and radio use. The RE-3, designed for the navy, was also used in broadcast work. Finally, there was the K-7, the machine Mullin found at Radio Frankfurt. Pflaumer said that there was an experimental AEG machine in addition which used a scanning-type head on an 8½-inch by 11-inch sheet of plastic impregnated with oxide. This, he said, could record up to eight hours at a time. No such machines were reported found. The three tapes were type C, dual layers of acetate; type L, a single layer of polyvinyl chloride; and LG, a dual-layered calender tape.

In any event, just as he was completing his job at Wald Mittelbach, Orr became involved in an automobile accident which put him in the hospital.



John T. Mullin (left) and Murdo McKenzie (right) flanking the two revolutionary German Magnetophons.

According to him, while there, he had an Irish nurse who helped him pass the time. In appreciation, he later named his company after her (it should be noted that Major Orr was an excellent storyteller, and at various times he has given other accounts of how he chose the name "Irish" for his tape).

Irish Paves the Way

When Orr finally was discharged from the hospital and reassigned home, Dr. Pflaumer gave him a going-away present—a brown paper bag. In it was what appeared to be iron rust. It proved to be the iron oxide formulation BASF had been using to make its tape. Orr returned to his home town of Opelika, Ala., itching to go into business for himself. Before the war, he had operated a small radio shop, and had a passionate interest in radio and electronics which had led to his Signal Corps commission. On his way home, Orr met Col. Richard Ranger, yet another Signal Corps officer, who said that he was thinking of copying one of those German machines and going into the tape recorder business.

When Orr got back to Opelika, he found an abandoned prisoner-of-war camp on the outskirts of town. He took a long-term lease, hired a research chemist, Herbert Hard, and moved in to make tape. There was no market in sight when they started—a fortunate fact, because it took Irish five years to lick the problem of putting oxide on a plastic backing so that it would stay. The process borrowed heavily from BASF's idea of calendering—using one layer of tape oxide to polish another—and Irish became the first to market a calendered recording tape.

While there's no gainsaying the importance of Mullin's and Orr's contributions to the development of tape recording, it would be an oversimplification to imply that nothing else had been happening. In fact, under the auspices of the U.S. Navy and others, there had been a fair amount of research under way since the early months of the war.

At about the same time, Armour Research Foundation in Chicago began manufacturing wire recorders for the army and navy. The recorders were the result of experimental work conducted by Marvin Camras, a recent graduate of Illinois Institute of Technology. It was in 1941, also, that Camras applied for a patent for a process of AC biasing of recordings, the technique used today in all professional and home recorders. In the months which followed, Webcor and General Electric began making wire recorders, using the Camras patents. In 1944, partly in response to a letter from Brush, Dr. Ralph Oace of Minnesota Mining & Manufacturing Co. began experimenting with recording oxides and processes for coating tape. By the end of the War, Armour had produced some 10,000 military wire recorders, and Brush had produced an additional 2,000.

With the coming of peace, it appeared that magnetic recording might yet be a reality for the amateur—but that he'd be using wire. Acting on this premise, Armour licensed Webster-Chicago, Sears-Roebuck, RCA and others to manufacture wire recorders for home use under its patents. In 1946, four young men from Armour Research decided to go into the wire recorder business for themselves. Their company, Magnecord, would produce high fidelity wire recorders for professional broadcast use. The first year, they sold ten Model SD-1 recorders, for $1500 each. The first was purchased by Glenn Martin Aircraft Co., which used it for telemetering rather than sound recording.

Although the wire recorder continued to sell well into 1948, its days were numbered. Wire simply didn't have the fidelity even of the phonograph record. It was very difficult to edit, and if the wire slipped off the spools, it became hopelessly tangled. Brands like Polyphonic, Pierce, Pentron, Brush, Silvertone, Dynaport, Crescent and WiRecorder were doomed to disappear almost as fast as they had appeared in the first place. By the end of 1948, they simply weren't selling. What was selling was tape recorders—from Ampex, Berlant, Crestwood, Brush Soundmirror, Ekotape and others.

A Momentous Event

But we're getting ahead of our story. It's May 16, 1946, and something momentous is about to happen. Bear in mind that the wire recorder boom is already on, with stores like Sears promoting them for home use. Remember that 3M has been working on producing magnetic tape for 2½ years, and Brush and Indiana Steel Company both are at work on similar projects. There's a meeting of the San Francisco chapter of the Institute of Radio Engineers scheduled for tonight at the studios of KFRC. The speaker: Jack Mullin. His subject: the two German tape recorders he's been using to do studio sound recording work for more than a year.

"The studio was really packed that

Ched Smiley (center) checking an early stereo master tape.

night," Mullin said recently. "There was a lot of interest in the subject. We prepared some tapes of orchestra, vocalists and pipe organ. I was still using the German tape I'd brought back, and the whole thing was very effective."

In the audience that night were Harold Lindsay and Myron Stolaroff, representing a small company called Ampex. After the demonstration, the two asked Mullin about the small German loudspeaker he had used as part of his demonstration. Ampex, they explained, had been making aircraft motors during the war, and the company was now looking for a peacetime product. The engineers who made up company management (there was a total of six people on the payroll at the time) leaned toward something in high fidelity, and the German speaker might be just the thing.

Lindsay reported to Ampex president Alexander Poniatoff, who quickly decided that it was the recorder, not the speaker, which was worth pursuing. But before committing himself, Poniatoff submitted the idea to a group of technical experts. "After careful analysis, listening to the machine, examining its construction," Poniatoff reported later, "they came to the unanimous conclusion that the product had no future whatsoever. Nobody would bother to thread tape through a very complicated system. Breakage of the tape would immediately discourage anybody from using it. A disc is so much simpler and better, they said." Nevertheless, Poniatoff had a gut feeling that tape had a tremendous future. So he disregarded the advice and started work on the first professional American-made tape recorder.

Mullin, however, had given a verbal commitment to Col. Richard Ranger, whom he'd met on his way home from Europe. Col. Ranger, based in New Jersey, had announced plans to build tape recorders also based on the Magnetophon, and had gone to Germany to get parts and to meet with officials of AEG. He was to write, "The center of Magnetophon production was the AEG in the part of Berlin which finally came under the French. I found that there were parts for 18 machines available which had not been assembled. The French agreed to let them be assembled, and the 18 were to be apportioned—six to the French, six to the British and six to the U.S. When I came back some weeks later, I found the first had gone to the French, the second to the British and the third was to go to the French. Well, we finally got that straightened out and five of ours did excellent service in our Army Broadcasting and the sixth one I brought back to Fort Monmouth along with some 20 cases of all kinds of technical equipment which would be of interest to the Signal Corps."

Meanwhile, word of the successful IRE meeting had found its way to Frank Healy of Bing Crosby Enterprises. Accordingly, Mullin was invited in June, 1947 to demonstrate his machine for Crosby and his technical director, Murdo McKenzie. "After the demonstration there was a lot of interest, but I didn't get the job. Instead, I was invited back in August to record the first show of the 1947-48 season, to see how it would go," Mullen recalls. "At the same time, they were recording the show on discs.

They also invited Col. Ranger to bring two of his machines out at the same time. He showed up with his first two. He had rushed them through to get them there in time for the demonstration. So he recorded on his, I recorded on the Magnetophons and the Crosby people recorded on discs. During the playback, there was a comparison, and the Crosby people made an immediate decision that they didn't want to hear any more of Ranger's. That was unfortunate because he hadn't had the time to get the bugs out of his."

Bing Crosby on Tape

At the time, the Crosby show was the only major network program which wasn't broadcast "live." Crosby had been off the air entirely in 1945, and had returned to work in 1946 only because the fledgling American Broadcasting Corporation had promised that he could record his show in advance. In 1946, that meant 16-inch transcriptions, revolving at 33⅓ rpm. Crosby, being a casual person, liked to do the show without worrying about timing or fluffs, then leave the job of trimming it down to size to somebody else later. That meant rerecording from one transcription to another, with a resulting loss in sound quality on each transfer. Since some transfers were made three or four times, the final version sometimes didn't sound so good. The sound in some cases was so poor that the sponsor blamed it for a fall-off in ratings. Crosby was warned that if he didn't do something about the poor quality of some of his shows, he'd have to go back to doing the program "live" or risk the loss of his sponsor.

"Well," Mullin continues, "the result of the demonstration was that the Crosby people wanted me to stay right there and go through an editing process, to make a broadcast out of it. I did, and they saw how easy it was with tape. The next thing I knew, I had a job recording the Bing Crosby show for the rest of the season."

Mullin was still using his original German recorders and 50 rolls of recording tape; it wasn't until April 1948 that Ampex delivered its first two recorders to Crosby, and Scotch tape didn't show up until Mullin was ready to record Program #27. "In the meantime, all I had was my original 50 rolls. I didn't dare throw anything away when I edited, because I didn't know where I could get any more. So after every show, I'd go through and take the tape apart, splicing all the little bits together so that I could use them over again."

Project engineer Harold Lindsay and the Ampex 200's which were closely modeled after Mullin's Magnetophons.

Crosby was so taken with the Magnetophons and later with the Ampexes that he became the West Coast distributor for Ampex professional products. He was instrumental in getting ABC to purchase the first 12 machines produced by Ampex.

"Once Ranger was out of the picture, I was free to accept Ampex's invitation to work with them," Mullin continues. "They borrowed my two machines, measured them, examined the parts and the construction, then produced the Model 200. It's not an exact copy, but it's very much like the German machines."

Magnetic Tape Experiments

At 3M, a management decision had been made the previous year to go ahead on the tape recording project, even though Mullin's machine had yet to make an appearance. The first problem was getting the oxide to stick to the tape backing. The company disposed of that one quickly, but decided that something could be done to improve the oxide coating which Brush was supplying for use with its machines.

Since machines were necessary to the sale of any magnetic tape, 3M began contacting firms which were working on machines and submitting samples of experimental tape to them. It was during this process that 3M found out what Mullin and Ampex were up to. The company had already decided that standardization of tape widths, speeds and reel sizes was essential if tape recording were ever to develop full scale. Since the Magnetophon's speed and tape size seemed satisfactory, 3M simply standardized around them. Unfortunately, the company's problems weren't over. It discovered that the red iron oxide that it had developed so laboriously was already covered by a patent held by Marvin Camras. 3M promptly sought an exclusive license, then tried to use it to discourage other manufacturers from entering the field.

Orr observed that his technology derived from German technology, for which patent rights were available to anybody through the U.S. Alien Property Office. But other early tape makers—Audio Devices and Reeves Soundcraft—found themselves in a round of legal battles to prove that they weren't infringing upon the Camras patent.

While Magnecord's decision to switch from wire recorders to tape in 1948 lacked the impact of the appearance of the Mullin machines, it sealed the doom of wire. John Boyers, one of the original partners, recalls how it came about: "Going into the tape recorder business on a professional basis was rather difficult since we had an investment in a wire recorder which promised to be a very good machine. But our New York representative convinced one of the other partners and me at two o'clock

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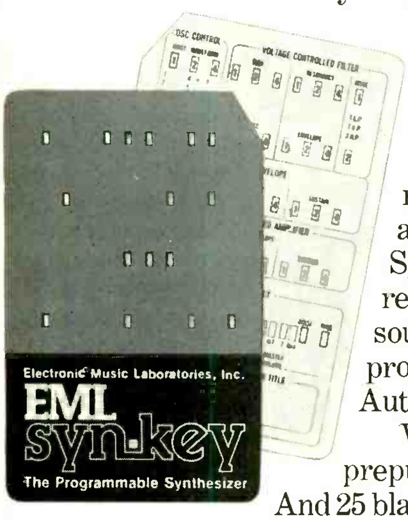

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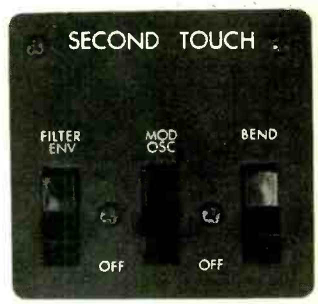

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one morning that we should make a tape recorder. By the time we got back to Chicago a day and a half later, we had outlined the genesis of the Magnecord PT-6. Two months later, we showed our first unit to the National Association of Broadcasters' convention."

Within weeks, no respectable broadcast or recording studio would be caught dead without either an Ampex 200 or Magnecord PT-6—or both—on the premises. The tape boom was on. As professional users switched to tape, dozens of home recorders began to appear—carrying names like Stenovox, Crestwood, Stancil-Hoffman and others.

The Dawn of Stereo

Then, one day in 1949, David Apps of General Motors Laboratories approached Magnecord with an unusual request. Could the company build a tape recorder which could record stereophonically? GM had been using a PT-6 to analyze automobile noise, but had found it unsatisfactory because of the limited perspective. Perhaps a stereo model might solve the problem. Magnecord responded with a PT-6 modified so that there were two three-head assemblies spaced $1\frac{5}{16}$ inches apart. The upper assembly recorded the left channel, the lower assembly the right. Having made one stereo machine, Magnecord made two more, for display at the 1949 Audio Fair in New York. The response was little short of fantastic. Immediately following the Fair, Magnecord built 12 machines to fill orders it had already received, then another 25 which were sold by the time they were assembled. After that the company sold hundreds.

Two early customers were Ched Smiley, of Livingston, N.J. and Emory Cook of Stamford, Conn. Both had experience as recording engineers, both were electronic experimenters and audio hobbyists. Cook used his to record trains, a thunderstorm, the sound of surf and eventually music in Boston, while Smiley took his to the May Festival in Florence, Italy in 1951 to record a symphony orchestra for commercial release on tape.

The musical applications of Magnecord's new machine were obvious to the company's founders, so in 1950, they hired a musical coordinator, a young recording engineer and audiophile named Bert Whyte. Whyte's first chore was to use the stereo machine to record as many different types of music as possible. As the Benny Goodman, Lionel Hampton, Woody Herman, Jimmy McPartland and other bands passed through Chicago, Whyte taped them all in stereo. Unfortunately, none of these tapes (many still in existence) has been released commercially, thanks in part to union problems.

It was Whyte, in fact, who had the chore of explaining stereo to James Caesar Petrillo, then president of the American Federation of Musicians. Since Petrillo had brought all commercial recording to a screeching halt in the United States only a few years before, he was a man feared by record companies and broadcast executives, and adored by his union's members. "I had been down to the University of Illinois at Urbana," Whyte said recently. "Leopold Stokowski was doing a concert there, and he had agreed to let me record it on an experimental basis. I came back to Chicago with the tape, and arranged for an appointment to play it for Mr. Petrillo. He sat there and listened, and as he listened I could see his face clouding up. When the tape ended, he scowled and said to me, 'Of course, you realize that we'll have to work out a new pay scale for stereophonic recordings. Since there are two channels on the tape, I'd consider that two recording sessions. Each man in the group would have to be paid twice.' "

Since stereo hadn't gotten to the point where anybody was considering recording it commercially, it was obvious to Whyte that any such suggestion would kill stereo before it ever got started. "Yes, Mr. Petrillo," he explained. "But don't you see? If stereo catches on, everything will have to be rerecorded since there are no stereo recordings now." As Petrillo contemplated what this would mean in work schedules for his musicians, the scowl slowly faded, to be replaced with a benign smile. "Yeah," he said, and the meeting was over.

AFM president James Petrillo, welcoming the advent of stereo with a benign smile.

(Left to right) Bert Whyte and Leopold Stokowski: A toast to the beneficence of AFM's Petrillo.

NEXT ISSUE: The Stereo Era

The end of the war between art and engineering.

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CIRCLE 5 ON READER SERVICE CARD

The control room is filling up. Running over background vocals, working out harmonies for "Don't Let the Sun," Dusty Springfield, Jessie Smith, Clydie King and Sherlie Mathews are watching Gus for cues. ☐ *Elton shows, meets everyone, then sits quietly on a stool in the back. Almost reserved.* ☐ *Bruce Johnson, Brian Wilson arrive. Cat Stevens and Danny from Three Dog Night fall in.* ☐ *An event.* ☐ *Elton has an idea. Everyone to the piano. "Billion-dollar background vocals," Gus chortles. "Imagine* that *liner note."* ☐ *The tape runs. It doesn't sound good. Too many voices. Too many stars. Two hours pass. Everyone in the control room is getting a bit nervous. Everyone except Gus. He's still smiling like the world's in his hands. The engineer's wondering what's going on and he keeps looking at Gus for signs. Gus hasn't said a word for an hour.* ☐ *Sometime after midnight he casually pushes the talkback. "I don't think this is going to work. Let's move on."* ☐ *On cue, as if from a movie director, everyone shifts gears. On to the next track. It might look chaotic but Gus is in control.*

Modern Recording: You seem incredibly relaxed in the studio. Are you? Is that real?

Gus Dudgeon: Oh, yeah! I often go to sleep. No—I'm always relaxed, always very relaxed in the studio, because there's nothing about a studio that phases me, unless the equipment goes wrong. Then, suddenly, I'm not very relaxed!

MR: Has it always been like that?

GD: Yeah, I think so, probably because I was an engineer first. I think once you've got to the point where the console can't do anything to fool you, once you know exactly what it does and what it can't do, it's like driving a car.

MR: Do you think you were a good engineer?

GD: Not really. I think what happened, really, was that I enjoyed it at the beginning. I thought it was fantastic. But I got bored with it very quickly because I realized it wasn't really what I wanted to do. Once I realized that it was a means to an end, I became a little more satisfied with it because I knew it was going to lead to something. But, of course, I became very frustrated. So, I started convincing people that maybe they should allow me to spend some of their money for them.

MR: And what was the first?

GD: Well, the very first thing I did was a "live" album with a guy called Zoot Money, who was around the time of Georgie Fame and Chris Farlow— sort of British R & B bag. And then I did the early Strawb's and it sort of built up from there. But the biggest shot in the arm I had was producing "Space Oddity" because that was the first time—you know, the David Bowie record—that somebody actually said, "Do exactly what you want. Just do it really, really well." And I thought, "Right, this is the one, this song is just so good."

So we sat down and we planned it very, very carefully, and when it came out, it actually was exactly what it was supposed to be. There are so few things in that record that even now I can fault, that I feel were mistakes. "Space Oddity," in fact, then led to Elton John coming to see me because he liked the record and was looking for a producer.

ROCKET MAN AT THE CONTROLS

AN INTERVIEW WITH ELTON JOHN'S PRODUCER GUS DUDGEON

BY RICHARD SANFORD ORSHOFF

"Space Oddity," in fact, then led to Elton John coming to see me because he liked the record and was looking for a producer.

MR: What had Elton been doing up until that point?

GD: Elton had made an album called *Empty Sky* which hadn't been totally successful. But he'd made some good records that I'd heard. And the important thing as far as I was concerned at the time was that, when he came in, I knew I liked the stuff of his I had already heard, and out of about twenty songs he played for me, I heard about twelve, as many as twelve, that just freaked me out. I just thought, "This guy—not only is he good, but he's got staying power." There was such a high percentage of his stuff that was really good. That was really what I'd been looking for, something that I could see going on for a period of time rather than just being one or two odd shots here and there.

MR: How did it come together from there?

GD: At the beginning—the very first album we planned for three weeks before—all of us, we sat down and planned it to the last, absolutely to the last detail. The whole album was done in a week.

MR: "All of us" meaning who?

GD: Well, that was Paul Buckmaster, myself, Steve Brown, Elton, and Bernie would drop by occasionally too. And we planned the whole album. After that, the second album, which was *Tumbleweed Connection*, that was done very much *a la carte.* We just went in with the guys from *Hookfoot* and laid about eight tracks down in about two days.

MR: Once you mentioned that you very seldom go into the studio with Elton having already written songs. You schedule an album without any songs to record.

GD: Oh, that's right. Let me think. He first started doing that when we went to the Chateau [a studio in France] and did *Honky Chateau.* And he wrote the songs at the Chateau. Ever since then, he's written them on location at the studio.

MR: How do you feel that affects studio work—or does it? Or does it make it more spontaneous?

GD: Well, I'll tell you what it does. It's one of those things that keeps you on your toes because when you start to work on the songs completely fresh—if you spend too much time thinking about something, you go stale on it, and very often the first thing you thought of was the best thing anyway. It's that feeling of, like, they're here to do this, and it's only just being created and it's all completely brand new and fresh and everybody's right on their toes. It creates a very good, high-pitched, tense situation. Making an album, then, you have no illusions about what you're going to do. I mean, I arrive in America, at a studio, with about as much idea of what kind of album it's going to be as I have about flying to the moon. I walk in the door and he says, "Here are the songs." Bang, bang, bang.

MR: They've been there a couple of days and he's been writing and playing?

GD: Yeah. And I just get an instant reaction. Yes. No. Yes. No. Whatever. Oh, that sounds—oh, when I pop my fingers—listen to that one again—and, maybe this—oh, that's a bit weird but I'll probably like it. And it all kind of comes bang, bang, bang, bang, bang, and then after a couple of days I'm completely into it. And everybody else is. And making an album is like one long, relaxing session, because you start off on the first day, on the first title, super-tense, thinking, "Jesus Christ, is it really going to be this good? Have we really got so many great songs?" And then, slowly as you go, as you work your way through the album, it becomes more and more relaxing, you slowly unwind, you get the final track out of the system and you know the album is done and all three tracks are done. The feeling is just amazing.

There doesn't seem to be any kind of real system to it. The only thing I do know that is consistent, is the fact that when you get there—even though some of the things on first hearing, you think, "What? He's gone mad!"—after a while it just seems to fall into place.

MR: The projects you're doing now—Howard Werth and Colin Blunstone—how do you find they differ from being in the studio with Elton?

GD: Oh well, every single artist is completely different. Apart from anything else, Elton is the most extraordinary artist that I've ever worked with as a producer or even as an engineer. And as an engineer, I worked with an incredible amount of different artists. You know, from John Mayall, Marianne Faithful, The Zombies, Small Faces, all kinds of people, Roy Orbison. Elton is just—I s'pose it sounds strange of me to say this, for me to compare artists that I work with, 'cause they might become offended— but, the fact is, in his own way, Elton is head and shoulders above them all, simply because he's just completely in charge of himself. He's—I could go on about him for hours—he's just got the right balance of humanity and ambition. And he just balances it out really nicely.

MR: In the case of these projects then, I would assume you find yourself participating more in production than in Elton's sessions?

GD: I find myself doing jobs, worrying about things with other people I work with—things that I just don't have to worry about with Elton. There are lots of things that Elton just takes care of. You don't think about it. You don't worry about it because you know he's got it well "sussed" out. To take a small example, when Elton is laying down his lead vocal, at the same time he's laying it down, he's already working out what his harmonies are going to be. And he comes in and says, "Okay, I want to try this harmony." And you go "fine." With most other people, they come in and you say, "Well, that's great. Now, let's try some harmonies." And that's the difference. So, in no way am I putting the other artists that I work with down, it's just that Elton deserves to be where he is, because he's that good. And other people maybe just need a little more help. And I think they'd be the first to admit it.

MR: On the other hand, you mentioned to me before that Elton doesn't work on the mixes of his records. How involved does he get in the production?

GD: That's very difficult to answer. He has absolutely nothing to say, usually. Let me say first of all, once he's left the studio he's done his piece. He's finished. That's it. He doesn't attend any mixes. He doesn't attend if we're going to use an orchestra. He doesn't even attend the routining, the decision of what the line-up should be. He gets a rough idea, 'cause I keep him informed.

MR: I assume this is by choice.

GD: Yes, by his choice. For instance, when we're doing the backing vocals, he never comes in on those sessions, for two reasons. Firstly, they can be very boring, especially for somebody who's just sitting there observing. I mean, a few hours of listening to someone doing an "ah-ing" can become a pain in the ass, unless you're physically doing it. And also, because being such a good vocalist himself, it's obviously frustrating for him to hear other people having a problem with something he wouldn't have any problem with. So therefore, he's just cool about it. And also, if they're in the studio and they see him standing there looking at them, and they know he could do what they're trying to do in one take, what they would, say, take four or five takes to do, they just get a bit hung up. So, he's very cool. He just goes, watches television or plays backgammon or pool or something, and we just get on with it, and he comes in and checks it out now and again for a couple of minutes, and then when it's all over, we play the whole thing back for him.

The mixes I've always done on my own. He often says, "This time, on this album, I'm going to come down and listen to some of the mixes," but he never actually shows up. If he does, all he says is, "How many have you done," and if I say, "Three," he says, "Can I hear them?" And I'll play them to him, and he says, "Great," and off he goes again.

MR: Is there any particular kind of sound you try to achieve as a producer?

GD: For me personally, the sort of sound I've always been after, right from when I started, has been the sort of sound we got on "Captain Fantastic." That, to me, was the ultimate. Of course, now I can already fault it, but at the time it was the nearest I've got to what I always wanted.

MR: Do you feel the equipment makes much difference? At least, did you in that case?

GD: Oh, I'm convinced. I mixed that all on MCI gear. You see, the thing is, I don't like Neve equipment at all. And a lot of people use it. And, for some reason, people are using it a lot in the States. I don't know, I think they're buying it because it's British, which is a pretty bloody stupid reason to buy anything.

The fact is, all the EQ is so sharp, if you start putting EQ on a 24-track tape with Neve, and then decide later you want to duck it out, there's no way—because it's too damn sharp, you can't get hold of it and pull it back into shape again.

And, what I've done at Caribou [a studio on a ranch in Colorado] which is Neve, is I record everything completely flat. I just tidy a few things up. Maybe I may use EQ on say a guitar or something—not on an acoustic, but I will on an electric, because an electric is not so difficult to change in a mix. But I never use it on things like bass or anything like that, because the EQ is just so sharp.

For me, the MCI is very, very musical, the EQ is extremely musical. And when you lift something, it just lifts nicely and it just sounds smooth and it doesn't sound EQ'ed. What I don't like about Neve is I can tell a Neve recording from 55 miles away. I s'pose I can tell an MCI from 55 miles away, but I happen to like that sound.

MR: I've seen some of your 24-track tapes with four and six tracks of drums. Elton's records consistently have a full and driving and yet, at the same time, musical drum sound. Are the two related?

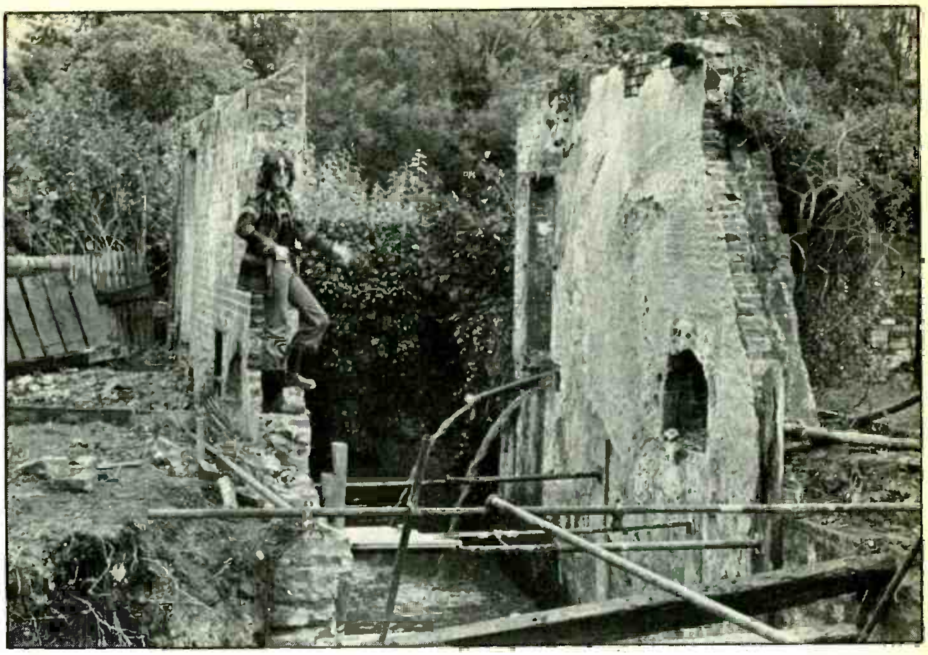
GD: Not really. This whole drum thing comes up all the time. It's one thing that everybody from taxi drivers to recording engineers comment on. And yet, it's something that I've never consciously—well, I s'pose a long time ago, I made a conscious attempt to do something with drums that nobody else had done. And that was, very simply, to record them properly—to record them the way they sound, not the way an equalizer can bend them. I really can't pin it down to any one stage. I'm sure I wasn't doing it when I was a recording engineer, because in those days it was four-track recording.

Anyway, I think it came about because I suddenly found that drummers were applying a certain kind of pressure on you which was very subtle. They were buying good kits—good kits that sounded really good "live"—and we were trying bass drum mics and that added something extra, and then we thought, "Well, Christ, if that bass drum mic has an effect, why don't we try it on tom toms." And then we started putting different mics on tom toms and fiddling about, and bit by bit, over a period of time while I was working at Trident, I found that I was getting somewhere and it was achieving something. Right from the beginning, as soon as I started producing, I always recorded the drums in stereo, which was something I knew that no one else was—well, maybe somebody else was doing it, but I didn't know of anybody else that was actually recording drums in stereo.

MR: That was eight-track in those days?

GD: Right. Eight-track at Trident. Just the fact that they were in stereo meant that you had to have more microphones. And then when you started employing more microphones you suddenly found you were hearing all kinds of overtones and things within the drums that you'd never even noticed before. And then along came Ringo Starr, who I think must be responsible more than any other drummer in the whole word, to revolutionize not only drumming as an art (rather than the techniques, because he's not a technical drummer, he's an artistic drummer), but he revolutionized it in terms of the sort of fills he played and his whole approach.

And it's just become a habit now. I automatically spend as much time as I need to on a drum kit. Sometimes it's only 15 minutes, and sometimes it's two days. It's been as long as two, three days to get the drum sound right. But the funny thing is, people will always talk about Nigel [Olssen's] drum sound because, I suppose, they probably know my work with him better than anybody else. And they're always saying, "How do you get that drum sound on Nigel?" And my answer, quite honestly, is, "I don't fucking know!" I s'pose if I got down to it and actually told them, they'd say, "Well, that's what I do!" Because I'm sure we all do the same. But, it seems to be the subtleties that make the difference. There seems to be something that I look for that maybe other people don't look for. I don't know.



Gus surveys an old watermill outside of London on the River Thames which he has recently purchased. His original plans to build a mixing room have expanded to include a complete 24-track quad studio, rehearsal studio and accomodations for guests and artists.

MR: There seems to be this consistent "shine" on Elton's tracks even from album to album, specifically drums and piano and vocal. This is really part of the same question. Is this something you work towards or is this something that has to do with the particular kind of echo you use, or what?

GD: Let me put it this way. If I was to go and produce the Beach Boys, we'd both get screwed up. Because the Beach Boys—I've talked to Carl Wilson about it—I would come in and I'd say, "How the hell do you do this? Why on earth do you—and, how do you get this?" And he's saying, "I don't know what you're talking about."

And he's saying to me, "Well, forget about—how do *you* get this piano sound?" And I'm going, "Well, I don't know." And he's saying, "Well, what kind of mics, where do you put the mics?" And I'm saying, "Well, I use this and I use that," and he's saying, "Well, that's what I use.

So, why is it different? I don't know. It seems to be that my ears, my taste, tell me that something is right or something is wrong. And I listen to somebody else's album and I think, "Well, that's probably the worst bass sound that I've ever heard in my life." But somebody else may listen to it and say, "Well, that's the greatest, that's just so great." A lot of people, for instance, really like the sound that Chris Squire gets with Yes. The bass sound. I think it's terrible. There's no way that I would ever want to get that sound. And if I tried to get it, I wouldn't know how to get it.

I think we did get it once for "Funeral for a Friend." There's just this very short section in "Funeral for a Friend" where Dee Murray specifically asked for that sound, and we did get it, but most of the work on getting the sound came from him, not from me, because I just looked at him and said, "Well, I don't know where to begin." It's the same with the Beach Boys. I'm sure if the Beach Boys tried to use my techniques, they wouldn't get it right. And if they did, they'd mess up what they do anyway. Because the way they do their thing is amazing. I just love the way the Beach Boys produce their records. But I feel that if I did it, I would just screw them up.

MR: There are some specific things that you do though. For example, you have described a rather involved enclosure system for miking piano. Others have tried that, but there is still something about Elton's piano sound that consistently rides above it.

GD: Well, that's a special one. I don't know what it is. All I can say is I think it's a combination of—number one, it's got to start out with the pianist and the way he plays. It could be something to do with the weight he plays at.

MR: Would you describe the set-up?

GD: I extend the piano. I think you would agree that the best way to get a piano sound is to get the mics as far away from the strings as possible. You get far too many overtones and far too much unnatural presence from a piano when the mic is three inches from the strings. After all, when you're playing the piano you haven't got your ear three inches from the strings. If you did you'd deafen yourself, and you'd hear all kinds of awful things going on. Ringings, overtones, harmonics, all kinds of gear you don't want to hear.

So, what I did, working on the principle that you don't want the mics further back, I built a box which is exactly the same size and shape as a piano, except that it's about four foot high and sits on top of the piano, so the piano has basically been heightened by about four foot. Now that means that, with the piano being about a foot inside, you've got about five or six feet that you can move up or down with the mics. The inside of this box is lined around the sides with some kind of dampening material. Usually we use fiberglas, which is the stuff you use in roof insulation—not that you have very much roof insulation in the States, because it's always warm enough. But in England we use it a lot.

The top of the piano, top of the lid as it were, on the inside, is left untreated, so it reflects a little bit. And then we insert the microphones in a hole which is cut in the "U," as you look at the piano on the right hand side, where you get the dip in the side of the piano. We cut a square there, which is big enough to allow the microphones to move completely up to the top of the box or right down to the strings. And, depending on the number, we move the mics around to get different effects.

MR: Do you generally use Neuman 87's?

GD: Generally, yes. Although, if I can find 67's, I prefer them, 'cause I think they're better.

MR: Less flat?

GD: Listen, I haven't the faintest idea what the difference is, all I know is, to my ears they just sound better. The same thing with the bloody drum sound.

MR: How many mics do you usually use on drums?

GD: Depends on the size of the kit. Nigel has used everything from enormous Slingerland kits down to quite small kits. I usually try if possible to mic everything individually, except I don't mic every cymbal individually but I mic, obviously, the snare, the bass drum and all the individual tom toms. Sometimes I mic above, sometimes below. But that, again, depends on the tuning of the drum and whether it has a bottom skin on or not.

MR: Are you still using kepex on drums?

GD: Very rarely. In fact, I've stopped using kepex almost entirely. I've also stopped using Dolbys as well.

MR: Really?

GD: I stopped using Dolbys with the *Solution* album. You should get a copy of it. It's a remarkable sound.

MR: Are you using any noise reduction?

GD: None at all. I don't like it. It changes the sound. And I don't care if Ray Dolby were to sit down and give me a 45-year lecture, it wouldn't make any difference. For a start, the various Dolby systems don't match. If you record through the old Dolbys they won't sound right through the new Dolbys, no matter what they say. My ears tell me that.

I have been recording all my 24-tracks and 16-tracks recently at 30 ips. And I shall continue to do so. I have been putting a slightly elevated level on the tape. I've been putting on 4 dB more than I used to. The *Solution* album was actually mixed through Dolbys. But from now on, I'm going to mix 30 as well. The only reason that I didn't mix 30 before was because there weren't any good cutting rooms in this country that had a 30 ips machine in their cutting room. But now, a cutting room that I use a lot and one particular engineer who cuts [all our master parts]—for the entire world because nobody else could match it. He now has a 30 ips machine, so I'm going to mix to 30 from now on.

MR: Getting the noise back doesn't bother you?

GD: All I can say is, if anybody can spot the noise to an irritating level, then I shall send them a personally autographed new mix of it.

MR: One last question: is there going to be a *real* "live" Elton album in the future?

GD: I don't know whether I'm supposed to tell you, but there will in fact be one. Yes.

MR: Has it already been planned, recorded?

GD: It's been recorded, because over the period of about two years I've been recording gigs quite frequently. The only reason I'm telling you that there will be one is because it's obvious, considering that the "B" side of "Lucy" was the "live" thing with Lennon. Obviously I didn't go along there and record one song. So anybody must figure out, "Well, there's a 'live' recording for a start." And it was common knowledge that the Royal Festival Hall gig was taped some time ago, the gig that Elton did for Princess Margaret.

MR: When will we be seeing that?

GD: That I don't know. It's in the can, but I haven't mixed it yet.

MR: All the recording has been done?

GD: I think we've got enough. Obviously, we've taken the same precautions that any "live" artist does when they do a "live" album, which is, you record a number of gigs which you think are going to be good and then you pick the best from that. I don't think people think it's done any other way.

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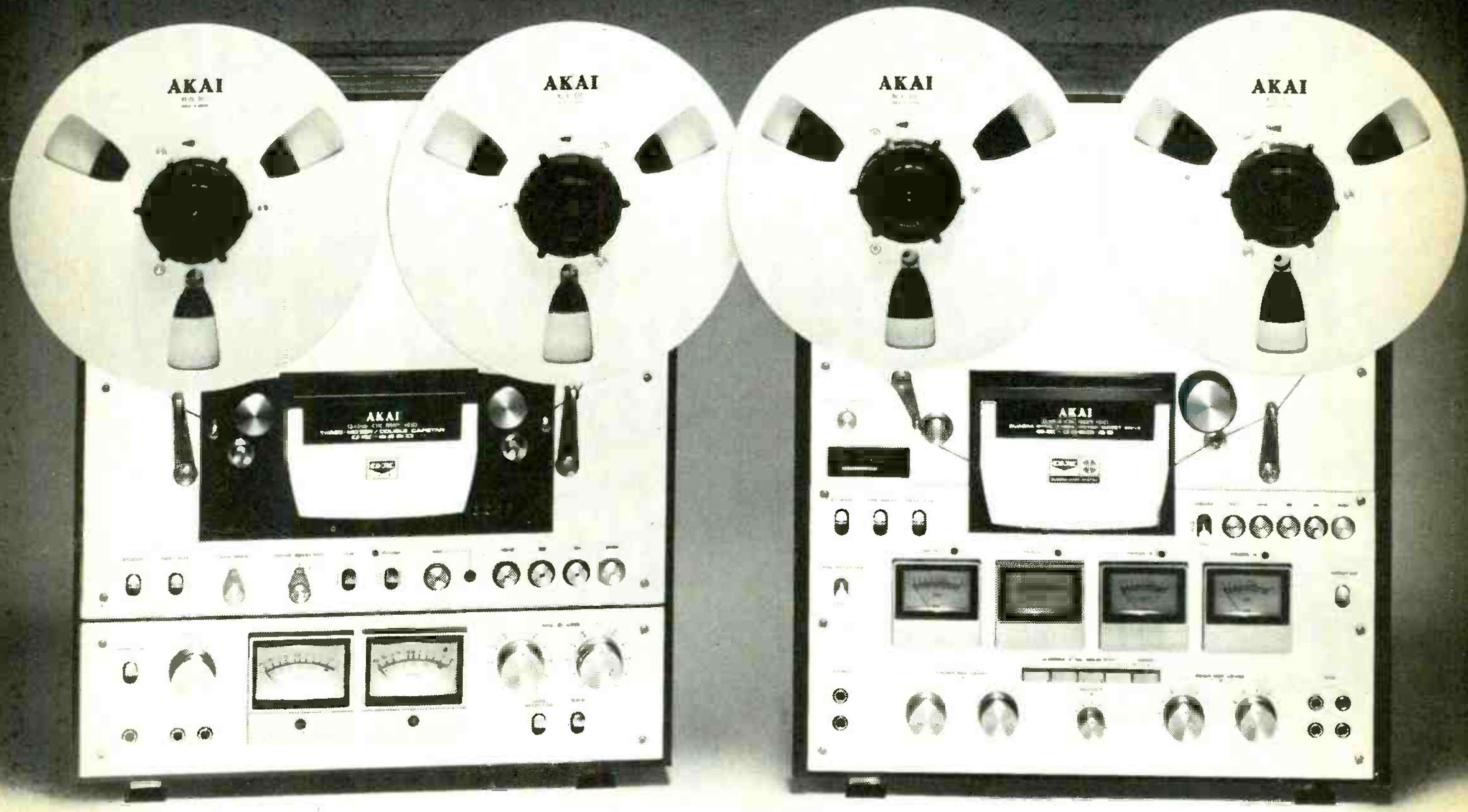

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CIRCLE 87 ON READER SERVICE CARD

MR: When will you be cutting the record? Can you tell me something about that? Like where and what?

GD: We should be going in the studio hopefully in the middle of March.

MR: Caribou?

GD: No. We were going to use Caribou, but we changed our minds, and decided to go probably to Canada. In fact, I'm coming over to the States in about two weeks time to have a look around some Canadian studios and pick one out. We had already decided after the last album, *Rock of the Westies*, that that would be the last time at Caribou, because—it's got nothing to do with the actual studio, even though I don't like working on a Neve desk; I've avoided the problems there by simply recording flat.

But the real reason we're moving is because we usually make a change about every two or three albums. It becomes too easy. If you keep going to the same place, you get to the point that you know it so well that you've only got to flick a certain switch and you know what's going to happen, it becomes a drag. It's much better to flick a switch and think, "What the fuck *is* going to happen?" It could be great or it could be quite boring. It's that thing that keeps you on your edge and keeps you thinking, "*Is* this going to work or isn't it?" As soon as we've started to feel very comfortable in a place and very kind of relaxed, we've moved. It keeps you on your toes. This also keeps it alive.

Gus (right), avoiding the flick of that certain switch.

FLASH! The day before this issue went to press there arrived on my desk a shipment of MCA records containing—lo and behold—a new Elton John album entitled Here and Now. *Side 1 is "live" at London's Royal Festival Hall and Side 2 is "live" at New York's Madison Square Garden.*—Ed.

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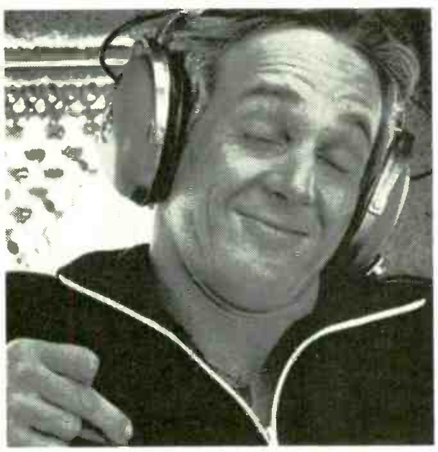



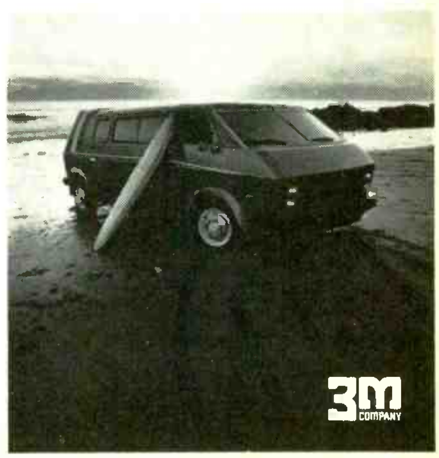

CIRCLE 86 ON READER SERVICE CARD

A SESSION WITH

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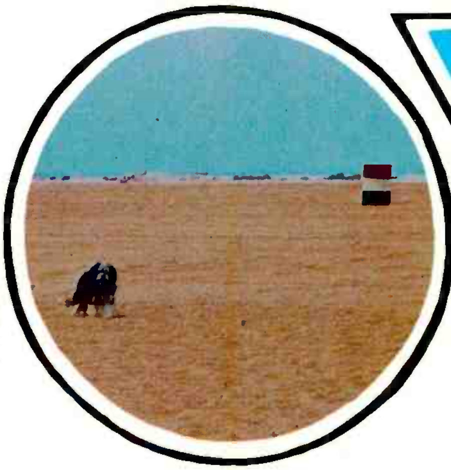


By Bob Weil

Photos by Ed Roach

The Beach Boys are among an elite group of musicians who enjoy the luxury of recording at their own fully equipped 24-track studio. Brother Studio is owned by Beach Boys Dennis and Carl Wilson. The facility, located, appropriately enough, in the sunny surf city of Santa Monica, Cal., gives the group the opportunity to create and record in a comfortable environment, free from the problems of scheduling, booking and, more importantly, paying for outside studio time—major considerations for most artists.

Over the past six months, the Beach Boys have rehearsed at Brother for their recent national tour; they have worked in the studio on individual projects ranging from soundtracks to producing other artists; they have worked intermittently in various combinations on their long-awaited album of original material; and, most recently, they have recorded a number of their favorite Oldies from the "Golden Age of Rock 'n' Roll." Some days's sessions have been cancelled simply because no one showed, and on other days the entire group has appeared at the studio totally unannounced, but ready to record. In a word, the situation there is "loose." The Beach Boys are Brother's *raison d'etre*, and the studio reflects the group's "go with the flow" philosophy.

The Beach Boys have been together since 1961 and the group still consists of the original five members: Brian, Dennis and Carl Wilson, their cousin Mike Love, and Al Jardine, a high-school friend. Today, all but Carl are in the studio working on the Phil Spector classic, "Chapel of Love." They have just completed the rhythm tracks and are gathered in the control room to listen back before putting on additional instruments and vocals. Dennis Wilson, Mike Love and Al Jardine seem pleased with the tracks so far, but they are even more enthused about the fact that Brian Wilson is really enjoying himself in the studio.

Throughout the Beach Boys's career, Brian Wilson has remained the group's creative leader. In the early '60's, he wrote the songs that captured the American adolescent dream world of sun, surf, cars and teenage romance. His high falsetto became the band's trademark, and after a few albums he had replaced his father, Murray Wilson, as the group's producer. By 1966, Brian had matured as a producer and the group had grown musically. The result was *Pet Sounds*, an album notable for its melodic an conceptual cohesiveness and for Brian's use of the studio as an active part of the creative process—something more than a means of preserving a "live" performance. In keeping with his reputation as an eccentric, Brian mixed the album in mono, although stereo was fully accepted at the time. And, in keeping with his professional reputation, *Pet Sounds* was, and is, a very successful album—from both popular and critical standpoints.

At this point, the group was recording all over Los Angeles—mainly at Gold Star, Columbia and United-Western—and, piece by piece, Brian put together *Good Vibrations*, a sea of overdubbed, heavily processed vocals, flowing in and out of numerous time changes. *Good Vibrations* is a landmark in that the song summed up the Beach Boys's celebration of the joyful life. It was the quintessence of Brian's richly layered production technique at the time, and it marked the beginning of his self-imposed withdrawal.

Brian stopped touring with the Beach Boys and (depending on your source) "stayed home and wrote a lot, and stepped back to get a new perspective on producing" (Brian); "took a well-deserved rest and stopped touring 'cause it's too fuckin' loud on stage" (Dennis); "was recuperating from a breakdown" (the press); or backed off from the pressures of a demanding record industry and of a worshipfully expectant public (pure conjecture).

Anyway, whatever he did, Brian gradually took a less active role in the group and Carl, Dennis, Mike and Al began doing more of the writing and producing. This more democratic arrangement resulted in some fine albums—most notably *Surf's Up* and *Holland*—but without Brian's leadership things slowed down. An indication of the group's admiration for (dependence on?) Brian comes from Dennis: "I look at Brian as *the* producer. Like Mike Love's master in transcendental meditation is Maharishi Mahesh Yogi—and he's a nice guy—my master is Brian Wilson in the studio. He blows my mind! I

love his music and emotionally go through changes when I hear it. I know it's weird talking about a brother like that, but he's a fucking genius."

It has been almost four years since the Beach Boys have recorded an album of original material. But, the Oldies album is different. Dennis explains: "We're recording it because *we* want to. It's a fun thing, which is what this group is all about. The album is more than just a warm-up for the original album, but it has a loose feel, for sure."

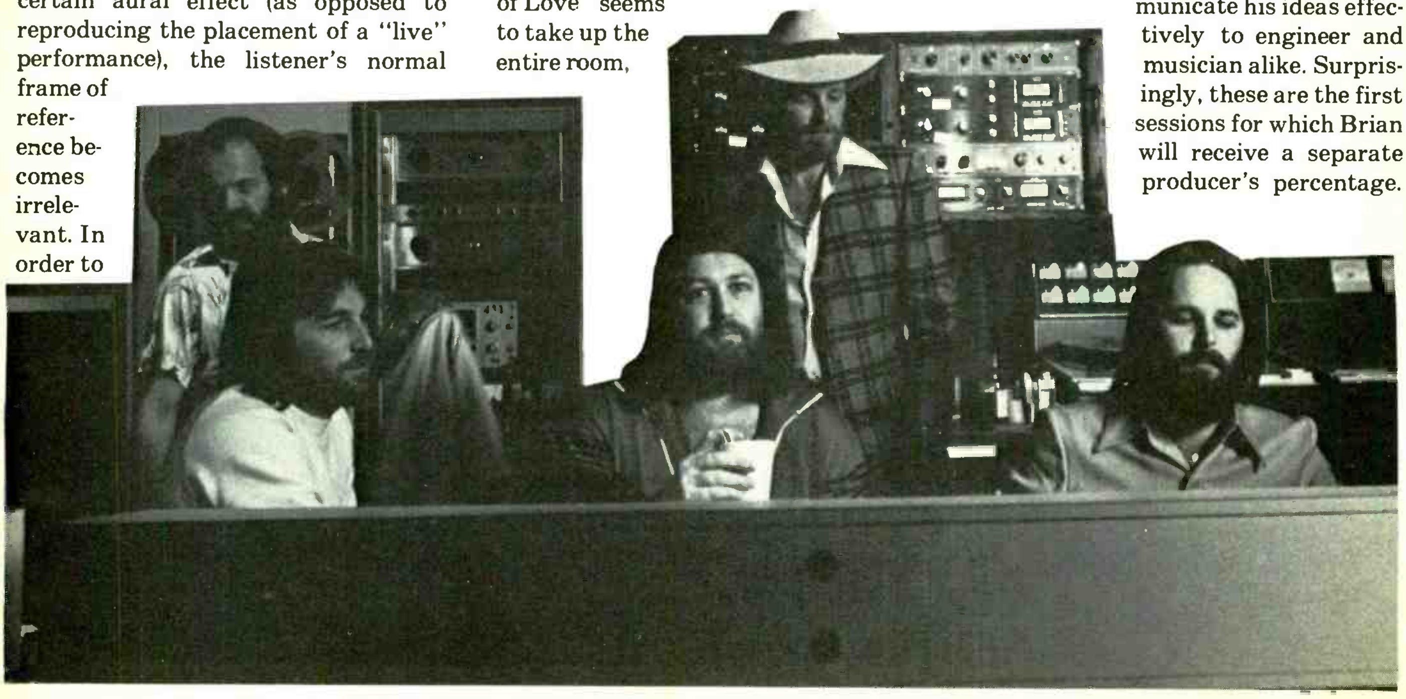
Brian obviously feels comfortable producing in Brother's easy atmosphere. Seated at the Clover Industries' board next to engineer Steve Moffitt, he has a casual confidence in his talent for choosing the right instrument or achieving the desired effect. As Brian positions and EQ's the tracks recorded so far (drums, percussion, autoharp, piano and guitar), "Chapel of Love" takes on a subtle but unmistakable Beach Boys sound, even before their distinctive vocals are added.

Steve Moffitt, who has been engineering for the group for about four years, explains, "The Wilsons all have this thing for grouping together instruments that are in the same register. This thickens the sound and produces some interesting phasing effects which we sometimes exaggerate with radical EQ. For example, we might boost 400 cycles and cut 500 cycles to get a lot of shift, which gives you that swimmy effect. When you look at most popular songs on an oscilloscope, you get a fairly linear pattern from bottom-left to top-right. That indicates everything's pretty much in phase. Most Beach Boys songs will produce a rounder, sunburst pattern which means there's a lot of shift. The Wilsons use that effect to get their particular sound."


Steven Moffitt

This unique approach to equalization and positioning results in a masking effect that tends to blend the instruments (and voices when they are added) into a homogeneous mixture. This is in direct opposition to most producers today who go for maximum definition on each track. And, since the Beach Boys place the tracks in the stereo spectrum in order to create a certain aural effect (as opposed to reproducing the placement of a "live" performance), the listener's normal frame of reference becomes irrelevant. In order to appreciate this effect, compare the swirling sound of most Beach Boys albums with the realistic sound of *Sunflower*, which was produced by the whole group in 1970. *Sunflower* was a major departure from the group's usual production (according to the liner notes) in that it was "recorded in true stereophonic sound; [it is] not 16 monophonic signals placed somewhere between right and left speakers, blended together with echo, but [it] capture[s] the ambience of the room and the sound in perspective as heard naturally by the ear." Conversely, the autoharp on the new album's "Chapel of Love" seems to take up the entire room, while the piano appears to be only a foot across, and there appear to be three drummers located throughout the studio.

Mike Love and Brian agree that "Chapel" has plenty of bottom end even without a bass track, but Brian decides bells should be added before moving on to the vocals. He tells percussionist Gene Estes to "just play all the tonics, except on the turnarounds," and it works on the first take. Brian Wilson's talent as a producer goes beyond an intuitive sense of what something will, or should, sound like. He has the rare ability to communicate his ideas effectively to engineer and musician alike. Surprisingly, these are the first sessions for which Brian will receive a separate producer's percentage.



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The instrumental tracks are complete and it is evident that keyboards have replaced the electric guitar as the instrumental cornerstone of the Beach Boys's music. Brian plays most of the keyboards on the album, although the other Wilsons sit in on some of the tracks. Various combinations of piano, Baldwin organ, MOOG and ARP String Ensemble synthesizer give the cuts a rich sound. Brian calls the synthesizer the "greatest addition to the Beach-Boys sound to come along," and he actually prefers the String Ensemble's unnaturally full sound to an real string section for most songs.

This emphasis on keyboards is a far cry from the early Beach Boys hits which almost single-handedly popularized the Fender Stratocaster. That self-contained electric-combo sound was a radical departure at the time from the orchestral pop/rock sound of the early '60's, and some people think that it helped pave the way for the Beatles and Progressive Rock.

This is not to say that the guitar has been phased out of the Beach Boys' music entirely. Carl has become an excellent guitarist, and, in spite of a bad back, he has been coming in regularly to lay down guitar tracks. In a couple of cases he is literally "laying down" the tracks, since the only position he can play in without considerable pain is flat on his back on the studio floor. Other guitarists featured on the album include studio guitarist Ben Benay, and Ed Carter and Billy Hinsche who are members of the Beach Boys backup band on the road.

Meanwhile, back at the "Chapel", Steve Moffitt sets up the Neuman U-67's for the vocals, and Earl Mankey, Brother's other staff engineer, explains, "The U-67's tube sound is usually perfect for the group's vocal blend, but we will occasionally go to a Sony C-12 when we want a solo vocal to stand out."

Meanwhile, the group is in the midst of deciding who will sing lead. Most artists have a fairly specific plan for vocals before the rhythm tracks are finished but, according to Dennis, "We've all been through the mill together, and we know what to expect. Strengths, weaknesses .... So whatever happens, whoever takes the lead will be the Beach Boy who is right for it at the time."

Brian asks Mike Love if he wants to sing lead on "Chapel," but Mike says he can really hear Brian singing it. Brian, who has not sung a lead in nobody-knows-how-many years, casually agrees and leaves the control room for the studio. Once in front of the microphone, he appears to have some reservations and asks that the control room be cleared of everybody but Beach Boys and engineers.

Through the (almost) soundproof studio door, we can barely hear Brian's still familiar falsetto running through the song. By the third time through he is singing pretty confidently, they have a take, and all of us non-essential personnel have trickled back into the control room.

Listening back, Brian's voice sounds a little rusty in a couple of spots, but basically the track feels good. After some discussion of possible back-

ground vocal lines and of who is going to take the high part, Al and Mike move to the studio. As it turns out, Mike (who usually sings the lower leads) takes the high harmony, but the basic quality of the Beach Boys's harmony comes through.

Moffitt, who also mixes for the group on the road, says, "The Beach Boys have naturally unique vocals without even touching them. Their blend is usually accented further by having two or three of them sing backgrounds around the same mic, and by doubling the voices—sometimes with other voices, sometimes with instruments. EQ and echo are added, and the vocal tracks are usually limited a little, then mixed up front." There is nothing really extraordinary in the way the Beach Boys's vocals are recorded any more than there is a trick to recording Eric Clapton's guitar or Miles Davis' trumpet. More often than not, good production is more a matter of highlighting something that is already special than of taking something ordinary and trying to make it special.

When Mike, Al and Brian are satisfied with the back-up vocals, Steve makes some cassette copies of the rough mix for the band to listen to outside the studio. It has been a productive and eventful day and everyone's spirits are high.

As he gets ready to leave, Brian is raving about Phil Spector: "The man is just a hero. He gave rock just what it needed at the time and obviously influenced us a lot. We're doing two more Spector songs on the album: 'Just Once in My Life' which he cowrote with Carole King and Jerry Goffin, and 'He's So Fine' [written by Margo, Margo, Medress and Siegel]. I guess I chose most of the ballads on the album. Carl and Dennis are more into rock 'n' roll and picked 'Rock 'n' Roll Music,' 'Mony Mony' and 'Palisades Park.' Al and Mike are the R & B fans and they suggested 'Blueberry Hill,' 'A Casual Look' and 'On Broadway.' "

It is easy to see that Brian feels good about today's session and the Oldies album, in general. The group is scheduled to have some publicity shots taken the following day and won't be coming in until late afternoon. This gives a perfect opportunity to sit down with Steve and Earl to get the engineer's viewpoint.

Steve has been with the Beach Boys since Carl and Dennis called him in late 1971 to work on *So Tough*—a Beach-Boys album recorded under the pseudonym of "Carl and the Passions."

"I didn't know it then, but my freelancing days were over. First I spent a week figuring out Brian's home studio, then we did the album. Then we packed up the whole studio and flew to Holland where we set it back up and recorded the album *Holland.* When we got back, the equipment went into storage for about a year before Carl and Dennis asked me to help them put Brother together. Since then it's been a 20-hour-a-day job for me."

Moffitt and Gordon Rudd (of Clover Industries) designed and built the

Brother console, then Steve went to work on the monitors with a specific goal in mind: "Most speakers' tone coloration changes drastically as you change volume. This is mainly due to midrange design—say 200 to 5,000 cycles. So, I looked for a midrange speaker that had flat frequency response with no cabinet affecting it. When I found one, I experimented with various enclosures and materials until I was satisfied. The crossover network is pretty unconventional in that it isn't perfect theoretically, but it works with the system. And I think we ended up with monitors that come close to reproducing the signal that goes in, instead of sounding like that signal coming out of a box, which is what most monitors sound like to me."

The Beach Boys have come to respect Steve as an engineer and they do not hesitate to ask his opinion of the music. Brian calls Steve an "efficient engineer who has a stabilizing effect on the group." He adds, "We haven't worked with Earl as much yet, but more and more, recently. I really respect the fact that he doesn't hesitate to tell me exactly how he feels about the music he's recording for us—whether he likes it or feels it could be better."

Like many engineers, Earl Mankey is a musician who tired of life on the road. A former member of the group Sparks, he came to Brother mainly because of his fascination with Brian's production. "The thing about Brian is that he is concerned with the song as a whole, not just with being able to hear a 'hot guitar lick' here or a 'meaningful lyric' there." Earl obviously looks at engineering as a step toward producing, and he can often be found in the studio, after hours, doing numerous overdubs on his own material.

Earl and Steve have very different ideas about drum-miking. Steve likes to close-mic the drums in order to minimize leakage and maximize his control over the balance and tone of the set. Earl leans toward distant mikings, especially if the set's natural sound is right for the song.

Several days earlier, Dennis (who plays most of the drums on the album) was out with a sprained wrist, so Brian called in premier studio-drummer Hal Blaine. Moffitt says that he was very surprised that he could not get a full drum sound on Blaine's drums. "They had a sort of this cardboard sound that I couldn't get rid of." On the other hand (or microphone), Earl liked Blaine's drums and on "Blueberry Hill" he miked them with a pair of Neuman KM84's overhead, a Shure SM56 on the snare and a Neuman U-67 as the more distant room mic.

"A lot of times when you get into close miking you have to spend a lot of time taping or padding the drums to get rid of the ring. Then you've got to EQ the hell out of them to restore some of the sound you've taken away with the tape or pads. If the drums sound good anyway, then I'd rather leave them alone—mic them from a distance, and live with the leakage."

The group is due back from the photo session and the studio personnel are just starting to get a little impatient when Tricia Roach, Brother Studio's Girl Friday, announces that "The Boys just called and they won't be in because the photo session was sort of a hassle, but they'll be in the usual time tomorrow."

But they cancel again the following day, and the next day only Dennis comes in for a short session. Over the course of the next two weeks, the Oldies sessions continue to lose momentum, and the atmosphere at Brother reflects the change. Steve, Earl, Tricia and the rest of the studio personnel are used to these fluctuations in the group's recording habits, but there is still a letdown after the high energy of the previous sessions.

The album is almost finished, however, with most of the tunes ready to be mixed. Although the sessions have definitely slowed down, most days at least one of the Beach Boys comes in to do some mixing, with Carl taking up a lot of the slack. In the past, Steve has worked more extensively with Carl than with the other members, so it goes fairly smoothly. They work well together, with Carl doing most of the EQ, positioning and level changes and Steve handling the outboard equipment. Brian goes for the final sound on each track during recording, so very few effects are added while mixing down, other than limiting the whole mix a little "so that the radio stations don't play with it too much."

Steve says many of the Beach Boys's albums were mixed on this kind of individual basis, and that Carl's and Brian's production techniques are close enough that Brian will probably not have to remix any of the songs. So, it looks like the Oldies album will be wrapped up soon after all—although without the exuberance of the earlier sessions.

It is easy for an outside observer to blame the lull on a loss of interest in the relatively unchallenging Oldies, or on impatience to get into the original material, or on a rebellion against pressure from Warner Brothers to put out an album. But that is all guesswork. The Beach Boys just are not into putting a lot of energy into recording at the moment, and there is nothing that says they have to. They all seem (at least) comfortable financially, so there is no urgency to finish an album on that level. They own the studio, so there is no deadline or heavy expense when they cancel. The Oldies album may be released as initially planned—simultaneously with the presently unfinished record of original Beach Boy compositions—or the two albums may come out in a double package, which was also discussed. Warners might rush the Oldies album out on its own, or perhaps the tapes will sit in the proverbial "can" forever.

The Beach Boys may be rolling again tomorrow, or it may never happen again. They are in the enviable position of waiting until they are ready to go back into the studio. The rest of us will simply have to wait and see whether this kind of artistic freedom makes it too easy for the group to lay back permanently and rest on their laurels—or if it gives them an opportunity to truly explore their potential, free from the usual outside pressures.

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A GUIDE TO SOUND REINFORCEMENT

PART I

Introduction

Most everyone becomes involved with the sound business because of some special love for music or the music industry. The majority of people doing sound-system work are in the business almost by accident and have very little training. In many cases, sound-system techniques have evolved from years of trading ideas from company to company. Many of the ideas have been based on rumors, superstitions, advertising or some "soundman" for a band that passed through town. Consequently, most band P.A. systems or small sound systems are constructed and operated incorrectly. The goal of this article is to divide this large problem into small problems and give some general "rule of thumb" procedures and examples to eliminate these errors.

The article will be divided into two parts. Part One will set down some basic rules and explanations about different pieces of sound equipment. In the next issue, Part Two will cover practical application of all the equipment, and will include the following:

(a) Selection of amplifiers, crossovers, speakers, boxes, horns and drivers.

(b) What to look for in mixers, limiters, equalizers, noise gates, digital delay lines, etc.

(c) Stage monitoring systems.

(d) Hardware, cables, snakes and connectors.

(e) Mics, direct boxes and technique.

In order to be able to make good choices out of Part Two the reader must understand the fundamentals of Part One.

The Size of the Job?

Before a sound system is built, the type and size of sound reinforcement jobs that will be encountered must be considered. The answers to these questions will describe the type and quantity of the hardware, and dollars needed to construct the system.

The following questions must be considered:

(1) Is the job (or system) indoors or outdoors?

(2) What size is the room, auditorium or arena?

(3) How reverberant is the room?

(4) How many people?

(5) How loud must the system be?

(6) What kind of music?

If the job is outdoors, the system will have to be approximately four to eight times larger to provide the same sound pressure level to the same number of people as if the job were indoors. This is due to the fact that in an

By Jim Ford and Brian A. Roth

Drawings by Steffon A. Kachocki

enclosed room the sound will reflect off the walls, ceiling and floor causing reverberation, and this room reverb adds to the volume of the sound system. Obviously, there are very few reflective surfaces outdoors, and, consequently, there is practically no reverb to help the sound system. Although it takes a larger system outdoors, the sound quality is more pleasing because the fewer reflections cause less deterioration of the sound and/or less feedback.

Indoors you will find everything from huge concrete "barns" (arenas) to small, low-ceiling, heavily-carpeted clubs. A "live" room with all hard surfaces (concrete, tile, wood, etc.) will have a high reverberation level (long reverb time) and demonstrate the following characteristics: (a) a very boomy sound, (b) poor clarity and definition of the sound, (c) speaking or singing will be difficult to understand, (d) the sound system will feedback easily, and (e) the sound level throughout the room will be nearly equal. A "dead" room with soft surfaces (carpet, padded seats, curtains, etc.) will have a low reverb level (short reverb time) in comparison. The sound in this room will not be as loud or as evenly and equally dispersed, but it will allow for better clarity and definition from the sound system. This room will

The number of people and type of music should additionally help to determine how loud the system must be. Don't take a sound system built to play for 200 country and western fans in a small club and try to provide sound outdoors for 20,000 rock 'n' roll fans. An average sound system should be able to produce a sound pressure level throughout the room of about 110 dB ("A" scale) for rock 'n' roll and about 100 dB for country and western. Although there are systems that will produce higher sound levels, it is not recommended due to the damage and loss of hearing that will result; nor is there much evidence that the audience actually enjoys the performance more because it is louder (with the possible exception of a few masochists).

Facts About Sound and Loudspeakers

Now that the size and type of job has been defined, the type and quantity of components need to be selected. As many facts as possible about sound and its transmission should be examined in order to build the exact system to accomplish the job. Here are some simple explanations of terms, along with rules that are frequently used in the sound industry.

(1) The decay of the volume level of sound as it moves away from the sound source is described by the "inverse square law." This means that the sound level decreases as the distance is increased from the sound system. The exact law is that for every doubling of the distance from the sound source the sound pressure level is reduced by 6 dB. Example 1: If a sound system produced a level of 100 dB at 20 feet, it would follow that the level would be 94 dB at 40 feet, 88 dB at 80 feet, 82 dB at 160 feet, etc.

(2) Each time the power input to a speaker is doubled the resulting sound pressure level is increased by 3 dB, or each time the power to a speaker is halved the SPL is decreased by 3 dB. Example 2: If a speaker with a 100-watt-maximum power rating were being powered by a 100-watt amplifier, and the speaker were producing 90 dB at 10 feet, then it would take two identical speakers being powered by 200 watts to produce 93 dB at 10 feet. It becomes obvious that gaining a 3 dB increase in sound pressure level can become a very expensive problem when it requires doubling the entire sound system each time.

Example of 1 and 2: If a speaker produces 100-dB SPL at 10 feet with 10 watts of power, how high a level would be measured at 40 feet with 20 watts of power applied? First, since the power was doubled the SPL would increase 3 dB at 10 feet to a total of 103 dB. Secondly, since the distance was doubled, twice the SPL would be reduced by 12 dB to a final level of 91 dB. Now, the above examples did not include the effect of the reverberation in the room, but the general concepts are a good starting point for calculating how many speakers and how much power is required to produce a certain sound level at a certain distance.

(3) Efficiency is defined as the ratio of the power output of a unit, divided by the power put into the unit, multiplied by 100. In simple terms, efficiency is a measure of how well a machine works. If 10 electrical watts of power were applied to a speaker and 5 acoustical watts were given out by the speaker, then the speaker would be 50% efficient. Most speakers and horns are 20% efficient or less. Example 4: 100 watts are applied to speaker X, and it produces 100 dB at 10 feet. 100 watts are applied to speaker Y and it produces 94 dB at 10 feet. It is seen that speaker X is 6 dB more efficient (louder) with the same amount of power applied. Now, if speaker Y is to produce the same loudness, the power input of Y would be (using example 2) double the power applied twice. 200 watts of power would produce 97 dB and 400 watts of power would produce 100 dB. If speaker Y were capable of handling 100 watts then it would take four speakers to match the power. In short, it would take four times the power amplifiers and four times the speakers to get the same SPL as speaker X. That also means four times the money would be spent, and four times as much equipment would have to be carried to each job. Now, it does not take long to see that it is advantageous to use the most efficient speaker system that is available. Another important conclusion that can be drawn from this is that it is not how much power a speaker will take, but how efficient it is. There are many 200- and 300-watt high-power speakers on the market that will not produce as much SPL as some speakers with 50- and 100-watt capacity.

(4) Distortion is the addition to the original signal of extraneous signals that were not in the original. If a 1000 Hz tone were applied to a power amplifier, and at the output of the power amplifier the 1000 Hz signal plus other tones were measured, then the amplifier would be adding distortion. Of course, this is not desirable; equipment should exhibit the lowest possible distortion. Distortion is generally expressed as a percentage of the output signal. If the distortion products are related to the input signal harmonically, then the distortion is called "harmonic." If the distortion products are generated by the interaction of different frequencies, it is called intermodulation (IM) distortion. With today's technology, all amplifiers and mixers should add less than .1% distortion to the signal. Many good amplifiers available today produce .05% distortion or less.

As for speakers, most manufacturers will not even give distortion specifications, due to the fact that typical figures are 5 and 10% or higher. However, there are several speaker manufacturers who are dedicated to the lowest distortion specifications possible, and these are the transducers that must be used if the sound system is going to sound

In the development of our $4000 state-of-the-art Servo Statik 1A we came across new principles which we determined to incorporate into less expensive speakers—principles concerning crossovers, and particularly the phase relationships of component drivers in a system. By properly balancing the phase leads and lags of each driver, and by scrupulous design of the drivers themselves, we are able to reproduce recorded temporal information absolutely accurately in a speaker of modest size and price.

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realistic and not cause listening fatigue to members of the audience.

Another item to look for when purchasing speakers: How linear is the frequency response? For example, two different speaker manufacturers may be selling a speaker that is advertised as having a frequency response of 40 to 16,000 Hz, and yet manufacturer A's speaker costs $100 more than manufacturer B's. The difference may lie in the fact that the frequency response of speaker A will deviate from any peak to any dip by only 4 dB maximum, whereas the frequency response of speaker B may deviate as much as 10 to 15 dB. Not only will the more linear speaker sound better, but it will not be as prone to feedback.

(5) All speakers and horns exhibit patterns of coverage peculiar to their individual design. (See figure 1.) Many times these coverage patterns are useful, and a working knowledge of how each speaker and horn operates is necessary for good results. All speakers or horns will be more efficient when positioned in a certain direction. If the unit is pointed correctly, the sound can be aimed in a desired direction to optimize the coverage.

This means that the sound system can concentrate the majority of its power toward the audience and not onto the walls and ceiling. The less sound reflecting from the walls and ceiling, the smaller the reverberation level in the room—which will let the audience hear more direct sound from the sound system. The result of this is a much improved ratio of direct-to-reflected sound, which will give better clarity, definition and intelligibility. Now it should be obvious that correct selection of coverage patterns and good aiming of the horns and speakers can greatly improve the end result.

(6) Generally, all cone speakers will have a dispersion pattern that is conical, and will be from about 90 degrees to 120 degrees wide. As the frequency of the sound goes up, this pattern usually narrows, which shows that speakers are more directional at high frequencies. This is called "beaming," and the result is that as the listener moves off the center axis of the speaker the sound appears to have more bass and less treble.

(7) Horns come in all shapes, sizes and lengths, and, consequently, a great variety of coverage patterns are available. Average coverage patterns used are (a) 40 degrees vertical by 90 degrees horizontal (see figure 2), (b) 40

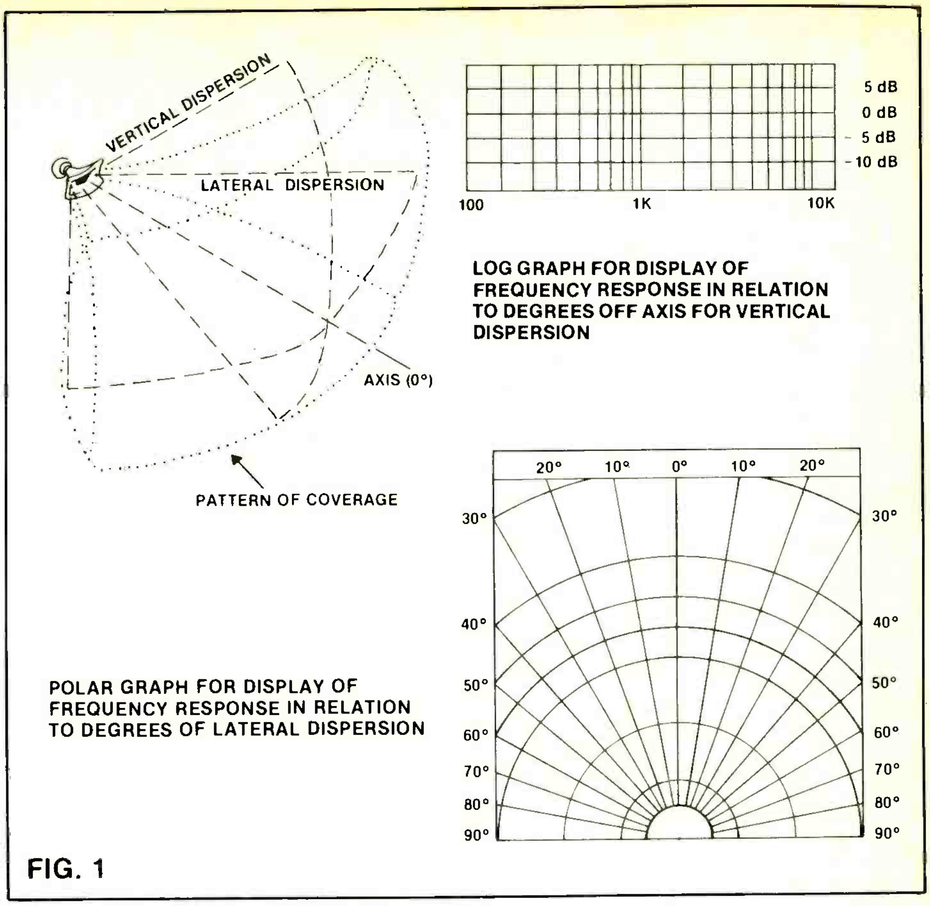


FIG. 1

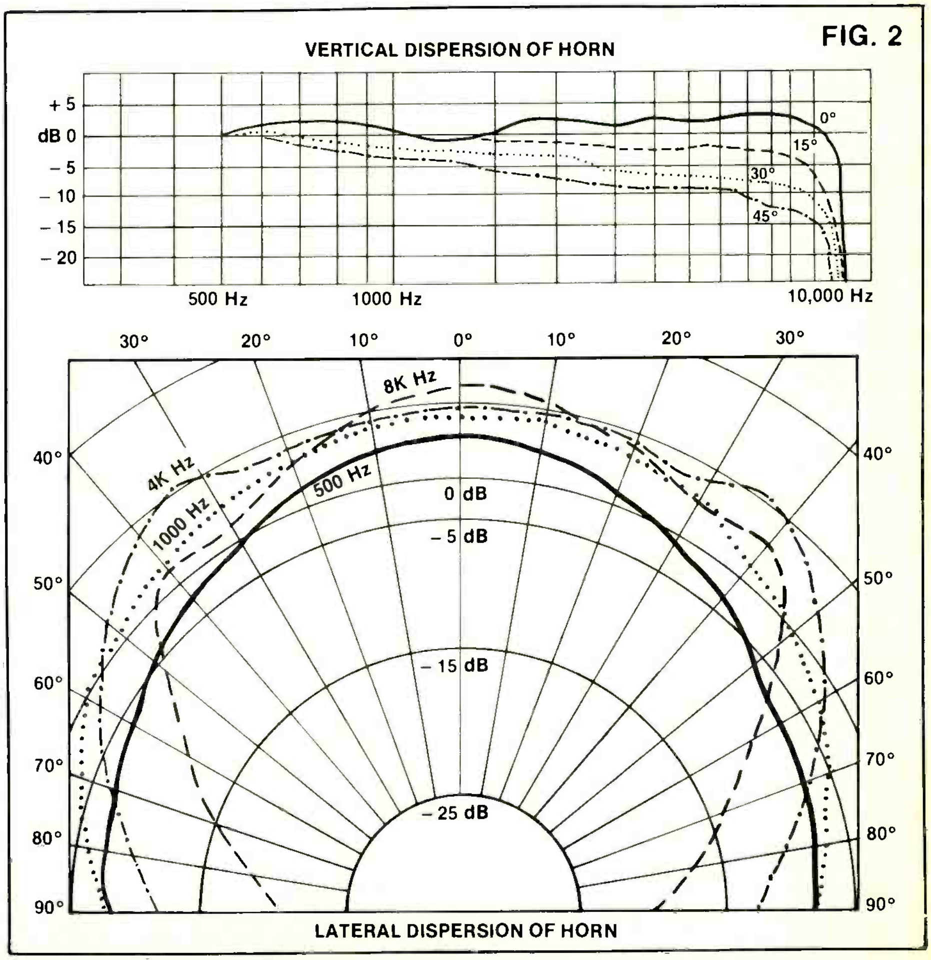
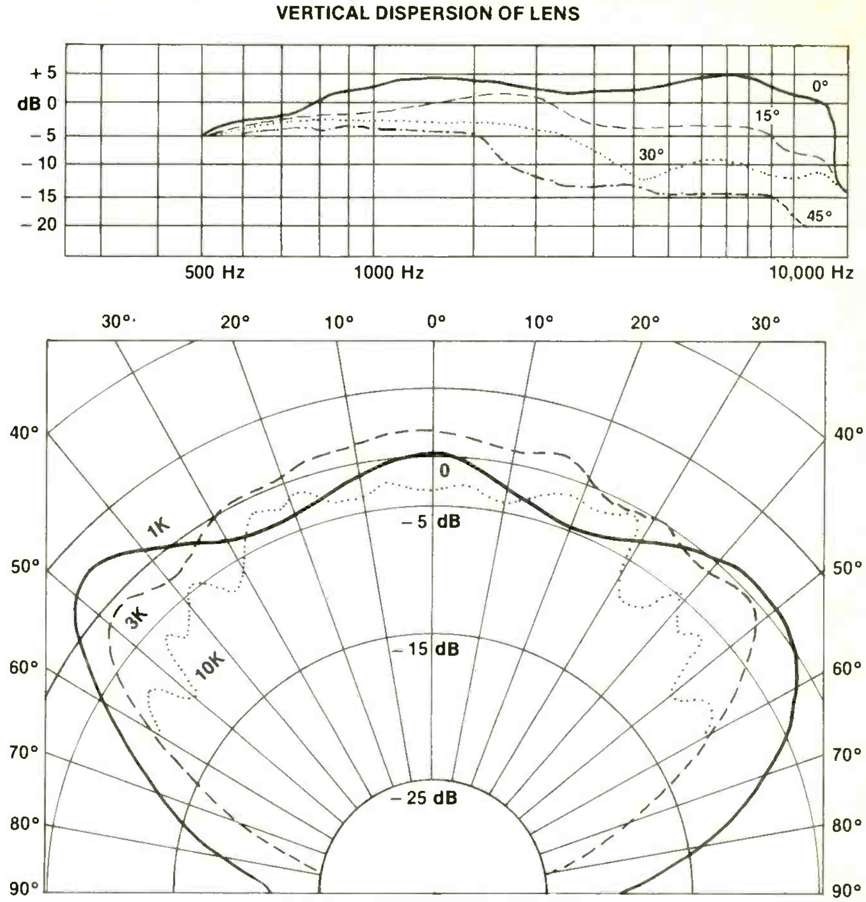


FIG. 2

FIG. 3



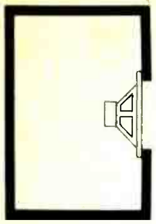
degrees vertical by 60 degrees horizontal, and (c) 20 degrees vertical by 40 degrees horizontal. Multi-cell horns are made up of 20-degree-by-20-degree cells that are put together to get the desired pattern. The above coverage patterns vary considerably with frequency, but are still useful in system design. Depending on (a) the length, (b) the size of the mouth, (c) the size of the throat, and (d) the rate of flare, the operating parameters of a horn are determined. Usually the longer and larger the horn is, the lower the frequency it will reproduce. This lower limit of operation is called the low- frequency cut-off, and below this frequency, the horn loses control of the sound. All of the coverage patterns are for frequencies above the cut-off point; the specification sheets show that horns also tend to be beamy at high frequencies. Use of the horn below the cut-off frequency is likely to lead to the destruction of the diaphragm in the horn driver due to acoustic unloading.

A horn is primarily an acoustic transformer, which helps couple the movement of the driver diaphragm to the air. This improved coupling makes the horn/driver combination very efficient. An average horn is 10 dB more efficient than a speaker, and that is a big difference. As the coverage pattern of a horn narrows, the efficiency goes up because the same amount of power from the driver is being concentrated into a smaller area. A 20-degree-by-40-degree radial horn can be as much as 20 dB more efficient than a cone speaker. However, there is a basic problem found in most horns that is the result of the high pressures developed in the horn throat. The high pressures cause the walls of the horn to flex and vibrate which adds distortion to the original signal. This is usually called "throat distortion." Many people prefer not to use horns because the throat distortion "colors" the sound, and it sounds "horny."

To summarize, horns are used because (a) they are very efficient and require little power, and (b) they have controlled coverage patterns.

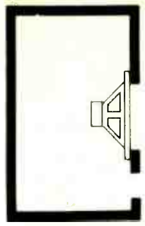
(8) Acoustical lenses (see figure 3) have appeared on the scene lately, although they have been around for more than 30 years. Basically, the lens is an attachment on the end of a horn.

FIG. 4



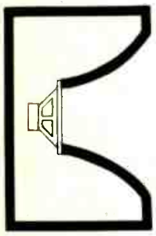
INFINITE BAFFLE OR SEALED BOX

FIG. 5



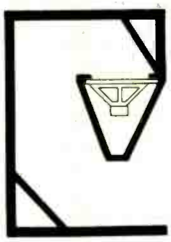
BASS-REFLEX BAFFLE

FIG. 6



BASS HORN

FIG. 7



FOLDED HORN

The horns are special in that they form the sound wave-front correctly in order to enter the lens. The problem with high frequencies being very directional is partially solved by the use of a lens. It operates almost like an optical lens that disperses a light beam into a wide pattern. An acoustical lens disperses high frequencies into a wide pattern and generally eliminates beaming.

(9) Speaker enclosures also come in many different sizes and shapes. There are three basic types: (a) infinite or sealed baffle (figure 4), (b) bass reflex (figure 5), and (c) horn (figure 6).

All cone speakers need to be enclosed so that the sound wave off the rear of the cone will not interfere with the sound wave from the front. This can be accomplished by putting the speaker in a sealed box. However, it was discovered that if a hole (port) were placed in the front baffle board near the speaker, the bass output of the unit was increased. By either changing the size of the hole, the volume of the enclosure, or the resonant frequency of the speaker, the overall bass response could be increased and smoothed out. At high frequencies, the port has a high impedance and does not work; therefore, this bass-reflex approach is only effective from about 250 Hz down. If a speaker is being used for a midrange unit, the volume of the box can be small, and it should be sealed. Note that at lower frequencies, however, a properly designed bass reflex box is preferred for P.A. use due to its higher efficiency.

Bass horns are designed on the same principles as high-frequency horns, and exhibit the same advantages. Unfortunately, as the usable frequency is lowered, the size of the horn must be increased, and this becomes an important factor. Bass horns that are usable to 100 Hz are about 5 feet high, 3 feet wide and 3 feet deep. If they are constructed of wood, they become extremely heavy.

A good bass horn can increase the efficiency of a speaker by about 6 dB, which means one bass horn can equal four other speakers and four times the power. As with high-frequency horns, there is still the problem of throat distortion, with the offsetting advantage of better directional control. So, it can be seen that bass horns have good and bad points.

Folded bass horns (figure 7) have also been in use for many years. The advantage is that a big horn can be built into a smaller package. Unfortunately, high frequencies do not turn corners very well, and, therefore, folded horns should not be used above 300 Hz. For use in a two-way system up to 800 Hz, a straight horn is much better.

Facts About the Crossover

The most simple amplifier/speaker system will use a full range speaker(s) connected directly to the power amplifier (see figure 8). While inexpensive in concept, this type of system places great demands on the loudspeaker in that it must reproduce the entire frequency spectrum faithfully, and thus turns out to be a very difficult design problem.

Consequently, the majority of full-range speaker systems are composed of two (or more) different transducers, each optimized to cover only a portion of the audio spectrum. Usually when these speakers are operated outside of their design frequency range, their acoustical output drops. In the case of high-frequency transducers (tweeters), the transducer will be damaged or destroyed if it is operated at a frequency below its range.

To divide the music signal into its proper transducer, a frequency-sensitive network called a crossover is placed between the amplifier and the speakers (see figure 9). For a two-way speaker system, the crossover divides the sound into bass and treble (low frequencies and high frequencies). The bass is sent to the bass speaker, and the treble is sent to the horn or tweeter. The dividing point between the bass and the treble is called the "crossover frequency." The crossover frequency for most two-way systems is between 500 and 3,500 Hz. At the crossover frequency, the bass section starts to attenuate the high frequencies, and the treble section starts to attenuate the low frequencies (figure 10). The rate of their attenuation or "roll-off" is called the slope of the crossover and is usually 12 or 18 dB per octave. (A one-octave change is defined as doubling the frequency.)

The crossover as described above is called a "passive crossover," and although it offers a great improvement over a single speaker system, there is a better method.

A passive crossover has the following deficiencies:

(a) It is built out of capacitors, inductors and resistors. Because the crossover is between the amplifier and the speaker, it must be able to handle high power levels. Consequently, the

FIG. 8

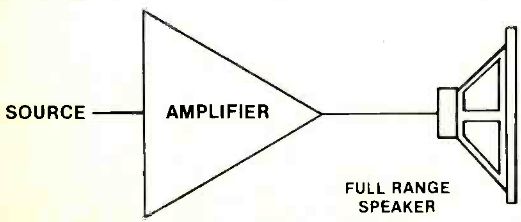
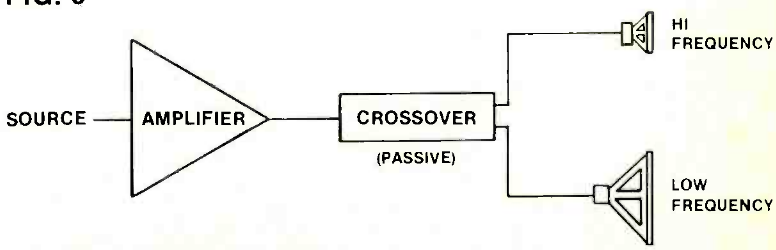


FIG. 9



capacitors, inductors and resistors must be very large, which means the crossover will be large, heavy and expensive.

(b) The inductors in the low-frequency section reduce the damping factor of the amplifier. (Damping factor is a measure of how well the amplifier exerts control over the speaker cone movement.)

(c) In most two-way systems a horn and a speaker are used, and the horn is usually 6 to 10 dB more efficient than the speaker. A resistor is used in the crossover to cut the power to the horn so that its sound level will match the speaker's sound level. This is a great waste of amplifier power. (Sound power going up in heat!)

(d) Passive crossovers are designed to work into a certain load (or impedance) which is usually 8 or 16 ohms. If the right combination of speakers is not attached, the crossover will not function properly. For large systems using multiple speakers and horns, many crossovers must be used and matched correctly.

Now that a passive crossover has been thoroughly discredited (except as a method of improvement over a full-range single-speaker system), here is a better solution.

Referring to figure 11, the frequency-dividing network has been moved ahead of two amplifiers. This gives direct coupling of the amplifiers to the speakers, eliminating the damping-factor problems. The high-frequency power output from the amplifier is not wasted in a resistor. Additionally, by dealing with the frequency at a lower voltage level, electronic techniques can be used in the filter to eliminate the inductors, which will increase performance and lower distortion. One electronic crossover can drive a number of amplifiers without strain, thus eliminating the impedance-matching problem and the multiple passive units needed for larger systems. Now for the clincher: Use of an electronic crossover and multiple amplifiers is a much more efficient method. In a bi-amplified system, the increase in power theoretically can be twice as much as that of an identical system powered by one amplifier. This can be explained by examining figure 12. Two different frequencies at equal amplitudes are linearly mixed. This is similar to what occurs when high and low frequencies in music are mixed together. The resultant signal has twice the voltage. Now, it would take one amplifier at 40 volts peak-to-peak to equal two amplifiers each at 20 volts peak-to-peak. Using the formula for power—power is equal to the voltage squared, divided by the impedance—the result is that two 50-watt amplifiers are equal to one 200-watt amplifier. Once again it is obvious that this can make a big difference in the amount of dollars spent on power amplifiers.

If a system is tri-amplified, the power advantage becomes 3 to 1, and it becomes 4 to 1 for a four-way system. For example, four 100-watt amplifiers will produce the same power as one 1,600-watt amplifier! That is why 99% of all professional systems use an electronic crossover and multiple amplifiers.

FIG. 11

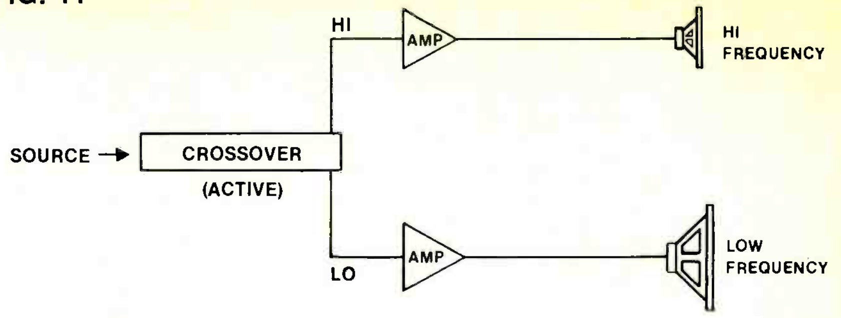
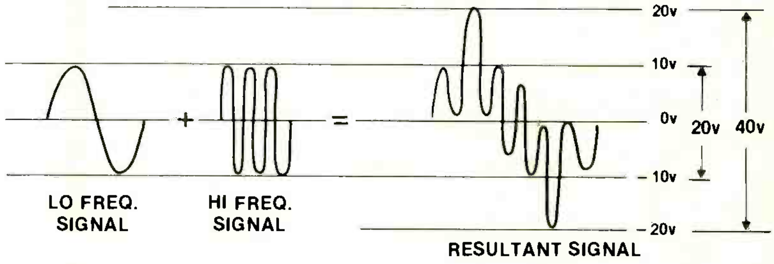


FIG. 12



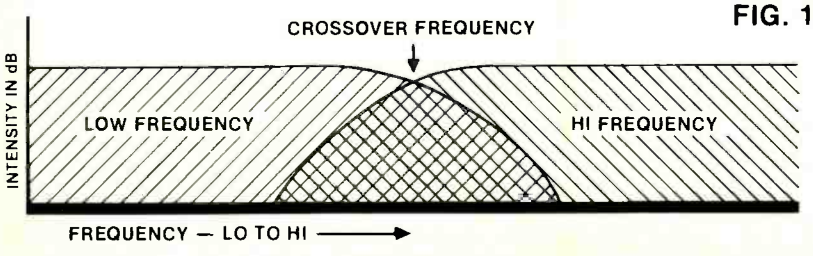
Now, in real life, the above ratios are not always realized. Due to different speakers and horn efficiencies, commercially available power amplifiers, and the varying sources of program material (music), it is difficult to pick a crossover frequency that will always equally divide the highs and lows.

The crossover frequency will greatly affect the degree of success in achieving the desired increase of power. This can be proven by varying the two voltage values discussed before, so that they are not equal. The greater the difference, the lesser the amount of effective power gain. There is considerable variation in the energy distribution from one type of audio source to another, and, in most music, the majority of the needed power is at lower (under 1,000 Hz) frequencies. This dictates a low crossover frequency for equal division and optimum effective power increase.

In the two-way system comprising only a woofer and a tweeter, this poses quite a challenge since the typical high-frequency unit in a P.A. system is a horn with compression driver. Usually these units must be crossed over at 800 Hz or higher in systems when they are operated anywhere near the rated power of the driver. This requires more "soup" for the bottom end since it is dealing with the majority of the acoustical energy.

From an energy-distribution standpoint and keeping available speaker

FIG. 10



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hardware in mind, it appears that a three-way split would be in order with a lower range (approximately 250 Hz and below), midrange using 10- or 12-inch diameter core transducers (typically 250 Hz to 1500 Hz), and a high-range horn (1500 Hz and above). From tests, this appears to be a much better distribution than an 800-Hz split for the woofer and midrange and an 8000-Hz split for the mid- and high-end often found in use. The main advantage of the 800/8000-Hz division is that typical horn/driver combinations that perform well at lower frequencies exhibit high-frequency-response roll-off. Consequently, the 8000 Hz high-end crossover point allows a specialized driver to be used to extend the top-end response to beyond the range of hearing. Although in the preferred method the high-frequency unit is required to cover one or two more octaves than the other units, the top two octaves (5K-10 kHz, 10K-20 kHz) do not contain nearly as much energy as the lower regions. If flat response to 20 kHz and beyond is necessary, a four-way system is required.

As discussed earlier, at frequencies near the low-frequency cut-off point of a horn/driver combination, distortion increases rapidly as the horn loses control of the diaphragm in the driver. By operating the transducer at 1500 Hz and higher, this problem is reduced without requiring a physically large horn (which would have a lower cut-off point). Also, the diaphragm in the driver is less likely to "blow" because of the better acoustic damping given by the horn at higher frequencies.

One of the original advantages of so-called "bi-amplification"—utilizing a crossover before the power amplifiers—was that much less intermodulation (IM) distortion would occur in the amplifiers. Since IM distortion involves interaction of two frequencies, separating the frequencies into two separate amplifiers would result in a drastic decrease in IM. However, this argument is not quite as important as it was 15 years ago when the original proposals for this type of system were made. Current state-of-the-art amplifiers exhibit remarkably low IM distortion (.1% or less) at high power output levels compared to the average 2-10% levels of yesteryear. Nevertheless, the elimination of intermodulation distortion between the two (or more) bands cannot be overlooked. It is yet another advantage for active crossovers ahead of the power amplifiers.

Concluding, here is a tabulation of advantages for "active" crossover usage:

(1) Elimination of the passive crossover and its problems.

(2) Direct coupling of amplifier to loudspeaker.

(3) Full high-frequency power delivered without the "watt-eating" attenuator resistor.

(4) Low-level (active) crossover filters which are generally lower in distortion, more accurate, smaller and lighter.

(5) Only one crossover needed for a system of practically any size.

(6) Decreased power-amplifier requirements since "bi-amplification" gives approximately twice the effective power of full-range amplification, and "tri-amplification" can deliver up to three times the effective power of full-range amplification.

(7) Lower intermodulation (IM) distortion due to the power amplifiers.

Well, that is the end of Part One. In the next issue, the above information will be used to design a real-life sound system—and that should be interesting!

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Ambient Sound

BY LEN FELDMAN

Dynamic Range—The Last Audio Frontier

If one traces the important progressive steps in the history of high fidelity reproduced sound, the sequence of events consists of a quest for flat frequency response (achieved some time in the 1950's), followed by attempts to reduce *measured* distortion (achieved at least in the electronic components of a sound system sometime during the mid- or late-1960's). The quest for sonically accurate transducers seems to go on forever and, despite so-called "breakthroughs" (which occur at the rate of about one a month), will probably continue for at least the rest of our lives.

There is one element involved in the reproduction of sound, however, that seems to be the new concern of engineers, technicians and lay audio fans alike in recent years. That is the field of dynamic range expansion. Today, we are offered all sorts of add-on gadgets designed to "restore that missing dynamic range." We have the dbx system, in all its various formats. We have Dolby "A" and Dolby "B" which are used primarily as noise reduction systems but which qualify in this discussion because dynamic expansion need not be confined only to "upward expansion." If we can make the soft sounds softer (or prevent them from being masked in residual noise levels), then we have expanded the dynamic range just as surely as if we had made the loud sounds louder. There are one-sided or single-ended expansion systems, such as the new Pioneer RG-1 Dynamic Signal Processor tested by Norman Eisenberg and myself for this issue, and we have the sophisticated Phase Linear Auto-Correlator with its "downward expander" ancillary circuitry.

It stands to reason that if these various forms of dynamic expansion have a place in the audio world, then somewhere along the line someone or something must have done one hell of a lot of *dynamic compression*—else why would we all be searching for better and better means of dynamic expansion?

Musical Dynamic Range in Real Life

Indeed, most of our presently available recorded or transmitted musical program sources are severely restricted in dynamic range compared with music heard in a concert hall. In a properly designed concert hall, during a symphony concert, we can be treated to dynamic ranges of greater than 80 dB from softest soft to loudest loud. In absolute terms of sound pressure level (SPL), we are talking about soft sounds which are barely 20 dB above the threshold of human hearing and loud sounds that may be well over 100 dB SPL. It is this extremely wide dynamic range which, many people feel, is the main distinguishing factor between "live" and "canned" sound in this era of ultra-low distortion and super-flat frequency response.

Reproduced Dynamic Range

Now, consider the capabilities of all the popular forms of reproduced music. Some FM tuners and receivers boast a signal-to-noise capability of 70 dB or more, under ideal strong signal conditions. Unfortunately, few FM broadcast stations are capable of transmitting signals with that much S/N, and even if they could, what would they use as source material? Disc recordings, at best, offer approximately 60 dB of signal-to-noise ratio. More often than not they are several generations removed from the tape masters which might have had signal-to-noise ratios which were slightly in excess of 60 dB before being remixed and edited for final use in cutting the master disc.

So there we are, with more than 80 dB of dynamic range wanting to be captured or preserved, and the media available for its preservation is missing at least 20 dB (and possibly a good deal more) if it is to do the job without sacrificing dynamic range. Obviously, something had to give, and that "give" takes the form of manual or automatic compression—at radio stations and in recording studios. The trouble is that no two compression methods are alike. Some engineers may elect to raise the level of the quiet passages and lower the passages of the loudest musical passages. Others may treat only the loud passages while letting the soft passages fall where they may. Still others may "limit" rather than compress. That is, they may let everything fall where it may until a certain level is reached, beyond which any further increase in input level is prevented from creating any increase in recorded (or broadcast) level.

The possibilities are endless. And because of that, I contend that unless some standards are arrived at in the near future, no single system of dynamic expansion will ever truly restore precise dynamic balance as it existed in the original musical performance. How can any single expansion device hope to cope with the situation—especially since data is hardly ever available regarding just what forms of compression were used to begin with and at what threshold levels? In terms of standards for expansion we are as much in the dark ages as we were in terms of record equalization twenty years ago, before the industry finally agreed to use the now-standard RIAA equalization curves for recording and playback of phonograph discs.

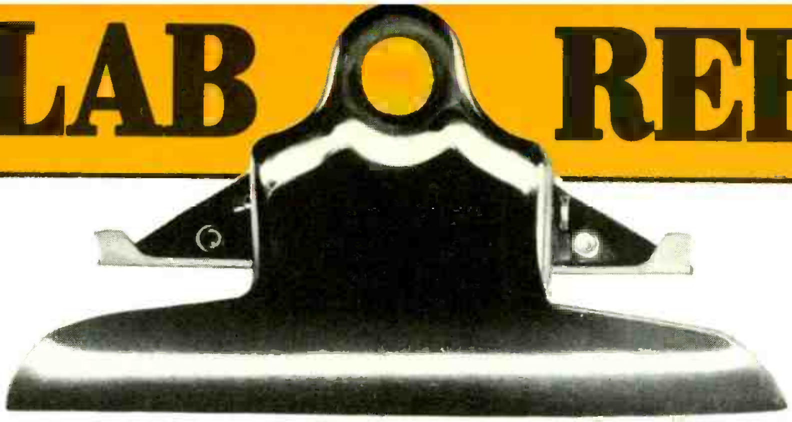
Other Expansion Problems

Even if all segments of the audio industry were to adopt a uniform set of standards for compression of stored or broadcast music, we would still have problems in attempting to reverse the process with some uniform expansion system. That is because any type of expander (be it a straight 2:1 system such as dbx, or some form of predetermined threshold system, such as that used in the Pioneer RG-1 or in JVC's newly announced Super-ANRS noise reduction system) must have some means of sensing what it is that it has to do. So long as that sensing involves analysis of the audio composite signal itself, there are going to be time constants involved. And time constants mean we are going to hear the expansion taking place, however minimal the "pumping" or "breathing" may be. And it means that some musical waveforms, complex as they are, may well "trick" the system so that its effectiveness is reduced. This is even more true if we include synthesized or electronic music (and how can we exclude it these days?) which is totally outside the sphere of statistical analysis that can be used to analyze "natural" musical waveforms.

What I am suggesting, therefore, is that no system of compression and expansion, using analog signals, will ever be capable of "perfect" reconstruction of the full dynamic range of music without any side effects. That's not to imply that some of the efforts being put forth in this area today should be abandoned. On the contrary, I never fail to be impressed with what dbx or Dolby or ANRS or any of the other systems have managed to accomplish in terms of satisfying noise reduction and very listenable improvement in dynamic range. It seems to me, however, that the proliferation of devices of this sort may lead to the same sort of consumer and industry confusion that was in great part responsible for the sudden stunting in the acceptance of four-channel sound in this country and abroad.

If an orderly attempt to standardize compression/expansion methods were made right now—perhaps even using a combination of digital (for the sensing or instruction signal) and analog technology—the result might well be new breakthroughs in this exciting domain, bringing us an important step closer to the avowed goal of total realism in reproduced sound.

MODERN RECORDING LAB REPORT



NORMAN EISENBERG AND LEN FELDMAN

Two New Radically Different Turntables

ADC Accutrac 4000

Harman-Kardon Rabco ST-7

Special Introduction: The turntable, of course, is a stable item of every sound studio as well as being used conspicuously by professionals and amateurs alike. Even though most recordings today are made on tape, the major dissemination of music is on vinyl.

Both the ADC and the Rabco are state-of-the-art products of recent and innovative design that merit serious attention. The ADC comes with a pick-up, but when you add the cost of a suitably high-quality pick-up to the Rabco, both tables fall roughly in the same price bracket. Each model offers something unique .... That is to say, while each is different from the other, both are different from all other turntables. Finally, and this may be the most intriguing aspect of all, their very differences (which superficially seem to belie a valid comparison) form a basis for comparison—in terms of what kind of buyer might prefer one or the other.

The twin reports here follow the usual MR format with these changes: first, we present the write-up on each, separately. Then, in the "individual comments," we discuss both units. Finally, in the "vital statistics" chart, we present the lab test data on each side by side.

General Description, ADC Accutrac 4000: ADC's first venture into turntables (the company has been long known for its phono pick-ups and speaker systems) is a two-speed (33 and 45 rpm) direct-drive model with integral arm and base for playing a single record with a degree of automation wholly unprecedented in turntable history. Thanks to a built-in computerized memory bank, the Accutrac can be programmed to play specific cuts on a disc in any sequence chosen, including repeats of the same cut up to 24 times. Once the order of selection is "punched in," the unit will perform as directed and shut itself off at the end. During the sequence, the operator can override the program, change it or cancel it.

The controls for operating the unit are ranged across the front top surface of the chassis. But in addition, there are duplicate buttons on a remote-control selector which may be used at some distance from the turntable. This hand-held device sends an ultrasonic signal to a sensing receiver which, plugged into the turntable, will activate its automatic functions. The r/c attachment (sensor and control box) is supplied as part of the complete package, which also includes an ADC phono pick-up, specially modified to contain a tiny electro-optical sensing system which responds to blank or smooth areas of a cut disc thereby enabling the automatic selection, set-down and lift-off to work. When the sensor in the pick-up "sees" a modulated groove it "tells" the arm to keep moving. For very heavily modulated grooves which could present a relatively smooth surface and possibly "fool" the sensor, there is a sensor adjustment that compensates, thus enabling the scanning system to function as intended.

The ADC Accutrac 4000 may be used automatically via the controls on the chassis or by the remote-control attachment. It also may be used manually, with cueing accomplished either via the cueing control or by regular set-down by hand.

The platter itself is a 12-inch casting weighing a bit over 3 lbs. Made for ADC by the Technics division of Panasonic, it is powered by direct-drive from an electronically controlled DC motor. Speed change is accomplished electronically and the two speeds have their own fine-speed adjustments that may be made with the aid of a built-in strobe light that works in conjunction with raised markings on the outer rim of the platter. The tone-arm, also from Technics, is a low-mass metal tubular type with non-removable head and decoupled counterweight. Vertical tracking force is set by rotating a section of the counterweight, which has a direct readout scale for VTF engraved on it. Anti-skating force is set by a separate adjustment to the right. Tracking force range is 0 to 4 grams. With the ADC LMA-1 cartridge supplied, the recommended range is ¾ to 1½ grams. Cable capacitance is less than 20 pF per channel, making the unit compatible with CD-4 playback. The cartridge itself is a modified ADC, fitted with a nude diamond elliptical stylus.

The full complement of controls includes four knobs for power off/on, scanning sensor, 33 rpm pitch adjust, and 45 rpm pitch adjust; five buttons for clear, play, reject, cue and repeat; and 14 more buttons for selecting recorded tracks, with the last button marked "all" for simply playing the entire side in the order it was cut. Table and arm come on a wooden base that "floats" inside the wooden surround. The hinged cover (which may be removed if desired) may be set down to cover the table and arm during use. The control panel is outside the cover and so the arm-to-record relationship may be changed without lifting the cover.

Test Results, ADC Accutrac 4000: Both the novelty and the myriad of automatic functions of this turntable intrigued MR's testers, and two samples were subjected to considerable use. They were put through their paces time and again, and several efforts were made to "fool" the system. The verdict is that with the unit adjusted correctly, and used as instructed, it works exactly as claimed. You can "punch up" the control panel, activate the unit, and let it run. The arm will scan the record and settle down on the cut selected, play it, then lift up and scan the record again to find the next selected band, and play that, and so. This same automation can be had using the r/c option, which we were achieving from distances up to 30 feet between the transmitter and the receptor.

In terms of normal turntable performance criteria, the ADC Accutrac measured up very nicely. Rumble (using the DIN "B" standard) was well down at better than −70 dB; wow and flutter were insignificant at less than 0.03% (weighted). Speed could be adjusted to pinpoint accuracy with the built-in strobe adjustment, or varied deliberately by ±4.5%. The platter took a bit less than 2 seconds to come up to full speed.

Tone-arm friction was very low, within the 5 to 7 grams specified; VTF and anti-skating adjustments were accurate. The cartridge itself produced a response that was measured to be within ±1.5 dB from 20 Hz to 20 kHz. Channel separation (at 1 kHz) was 29 dB. Output signal measured 3.6 millivolts re: 5.5 cm/sec velocity.

The entire unit gave the impression of quality and durability, combined with handsome styling and ease of operation. While the automation here is undoubtedly the Accutrac's strongest attraction, it is evident that ADC also has paid attention to the normal desiderata of high-quality turntable/arm performance.

General Info, ADC Accutrac 4000: Supplied as complete record playing system (turntable, arm, pick-up, base, dust cover) plus remote-control sensor and transmitter, 45-rpm adapter, and necessary cables. Overall dimensions: 18½ inches wide; 17⅜ inches front-to-back; height is 6 inches with cover down (cover raises to maximum height of 16¾ inches). Net weight is 20 lbs., 4 oz. R/C transmitter is 6 by 3 by 1¼ inches; R/C receiver is 4¾ inches high, 4½ inches in diameter. Price, complete, $499.95.

General Description, Harman-Kardon Rabco ST-7: This unit is a greatly improved and refined version of the straight-line tracking arm/turntable assembly brought out a few years ago by Harman-Kardon. It is a two-speed (33 and 45 rpm) manual, integrated with a radial-tracking arm that permits a phono cartridge to track a record groove with literally zero lateral tracking-angle error, with no need for anti-skating compensation, with no pivotal friction, and at lower-than-usual vertical tracking forces for a given pick-up. Thus, a possible source of playback distortion, and another of record and/or stylus wear, is greatly reduced. At the same time, since there is no pivoting arm, there is no pivotal friction and the need to compensate for skating does not exist since there is no skating effect (skating exists only as a function of a pivoted tone-arm).

The ST-7 must be recognized as considerably more than an "update" of a previous product. From the motor driving the platter to the manner in which the pick-up is installed in the arm, it is evident that the whole system has been rethought.

The arm body itself is a metal tube of low mass. Balance, for whatever pick-up is used, is first accomplished by adjusting a rear counterweight; VTF is set by a sliding weight along the arm body, which has line-markings for 0 to 3 grams. A control arm extends from the carriage assembly and is fitted with a "restrictor" piece just before the pick-up head. The control arm is used for positioning the tone-arm to cue a disc; the restrictor piece aids in aligning the pick-up and in maintaining a suitable height during lift-up. Adjustments are provided for varying the arm length (to suit different pick-up dimensions) and for fine-adjusting the lateral tracking. The carriage assembly, in addition to the visible upper guide rod, rides on a lower bar that rotates when the motor is turned on, being belt-driven from the platter. The arm begins to travel across only when the pick-up stylus is engaged by the record groove. A built-in cueing lever may be used to lower or raise the arm; for manually moving the arm to a given part of a record the cueing lever must be placed in the "stylus up" position. At the end of play, the arm lifts up against the restrictor piece.

The platter used in the ST-7 is an 11⅞-inch aluminum casting, weighing 2.4 pounds. It is belt-driven from a brushless DC (Hall effect) motor. Speed change is done electronically, and each speed has its own fine-speed adjustment that may be made with the aid of a built-in illuminated strobe system, visible through a small window. The turntable has a main AC power off/on switch under the base, near the front. In addition, it has a touch-to-operate 45-rpm/stop/33 rpm control bar, with separately illuminated indicators for each function. Total cable capacitance is below the 100-pF level and so is suitable for CD-4 playback.

The hinged cover may be raised to any convenient height or it may be removed entirely. The unit will operate with the cover in place.

Test Results, H-K Rabco ST-7: Setting up the Rabco ST-7, especially assembling the arm and correctly fitting one's pick-up, takes a little more than usual time and care but the instructions and gauge supplied do guide you correctly through this chore. Once done, it all works smoothly and exactly as claimed in the literature. The pick-up does move across the record in a true radius, and perfect groove-tracking is realized without the need for anti-skating adjustment, and with a VTF that may be ¼ to ½ gram less than used before for the same pick-up.

As for the standard turntable test measurements, the ST-7 is an obviously improved performer vis-a-vis earlier units, and indeed can stand up to any top-of-the-line manual turntable. Rumble, by the DIN "B" standard, approached the −70 dB mark; wow and flutter were insignificant at 0.035% (weighted). Speed is adjustable to on-the-nose accuracy via the strobe adjustment, and it can be varied deliberately over a ±5% range. Starting torque is especially good, with the platter coming up to full speed in only 0.6 seconds.

Arm friction is nonexistent in the usual sense since there is no pivot; the possible friction along the carriage guides is effectively cancelled vertically by a special bearing with counter-rotating bands, laterally by the absence of relative rotation between the stylus and the far (driven) end of the arm assembly.

The entire unit is handsomely styled and sturdily built. The owner must install it perfectly level, and occasionally clean the tracking shaft and a few other exposed parts, as instructed in the owner's manual.

General Info, H-K Rabco ST-7: Turntable and arm supplied with integral base and hinged dust-cover, plus 45-rpm adapter and cables. Overall dimensions: 16½ inches wide; 16¼ inches front-to-back; 6¾ inches high with cover down (cover raises to maximum height of 18¾ inches). Weight is 22.2 lbs. Price, $400.

Individual Comment by L.F. (both units): I showed the Accutrac turntable system to about a dozen visitors during the two-week period I had the unit for testing. Reactions were of two sorts. Some exclaimed that the Accutrac's computer-programming capability and its wireless remote-control option were exactly what they had been waiting for.

On the other hand, others felt that absolutely no benefit was to be derived from the band-programming and other automated functions performed (flawlessly) by the Accutrac.

I suspect that classical music lovers will see no reason to spend extra money on a unit of this type, while those who play specific bands of pop or rock discs will see the desirability of the automatic programming at once.

Rather than attempt to take sides in this argument, I prefer to confine my comments to the machine itself. To begin with, it performs exactly as claimed, both in regard to its programming features and to its published specs. However, I wonder why ADC did not select Technics' best direct-drive motor and turntable, if the Model 4000 represents the "top-of-the-line" Accutrac entry. We have been advised that lower-priced Accutracs (with all of the same programming capability and remote control) are on the way. If that is so, and if the Accutrac idea is to appeal to the audiophile who demands the very best in "pure" turntable performance (as well as to less precision-minded listeners), ADC might well have chosen better turntable and arm construction (and a more powerful drive motor) for this top entry. The tone-arm and turntable/drive system simply do not match the sophistication of the rest of the mechanism, in my view.

I also wish that cycling were a bit faster. It seems to take an inordinately long time to move from one band to another (bear in mind that the tone-arm must return to its rest position each time a new band is to be played, unless the "all" button has been programmed). The somewhat confusing nomenclature of the programming buttons (one must press "CUE" to lift the arm and press it again to resume play, and one must press "PLAY" before any action following band programming begins) will probably be corrected before mass production of the Accutrac begins.

As for the H-K Rabco Straight-Line Tracking System, Model ST-7, the numeral at the end of the model number suggests that there have been other ST's before. The problem with earlier Rabco models is that while in theory they always seemed right, in actual production they ran into quite a bit of trouble. Indeed, even the present ST-7 (which, as far as I could detect, has licked all those earlier mechanical problems) impresses me as a fairly delicate device—one I wouldn't dare handle in too rough a manner. Following the well-written assembly instructions, however, I encountered no difficulty in setting up the machine (which comes with turntable in place and requires installation of cartridge and tone-arm assembly) and got the unit to work first time around. I found it a bit difficult to correctly position the slide bar over the lead-in grooves of a record since there is some lack of precise verticality in the descending action of the arm on our sample, but after a few tries I learned to compensate for this error. The cueing action is not nearly as precise as that of the Accutrac. That is, the descending stylus does not return exactly to the groove at which playing had been interrupted when cueing lift-up is employed. Still, not having to fuss with anti-skate adjustments is indeed a pleasure. The tracking system is a marvel of simplicity and does not require any servo-correction systems to keep that arm moving along regardless of groove pitch.

I noted that a cartridge which normally had to track at downward forces of 1.25 to 1.5 grams was now able to track at just under 1 gram when properly set up in the tone-arm of the ST-7. That is a more significant improvement than might first seem true by comparing these sets of tracking numbers.

Even though the Accutrac uses less than the most expensive direct-drive components, it edges out the ST-7 in rumble by a couple of dB, but when you are talking about rumble figures of 68 and 70 dB (DIN "B") the differences are insignificant as far as audibility is concerned. If hand-cueing and quick attainment of correct rotational speed is important to you, the ST-7 wins hands down. If you are more casual about your phono system requirements, you won't mind the several seconds it takes before sound begins using the Accutrac.

Honestly, I can't tell you whether I like one more than the other anymore than I can tell you whether or not pears are better than apples. Each of these devices is sure to have a dedicated following among the audio cognoscenti. It really boils down to a decision as to whether you want *all* your turntable dollars invested in the turntable and tone-arm performance or whether you care about convenience features enough to apportion a fairly substantial percentage of those dollars into the convenience-aspects of record playing.

Individual Comment by N.E. (both units): My visitors also expressed amazement commingled with varied comments about the Accutrac. We even had some fun aiming the r/c transmitter away from its receptor to determine bounce-angles for the signal to activate the unit.

But once past the initial "shock" of the functioning of the Accutrac, questions did arise regarding its ultimate usefulness. I don't think it can be divided simply between classical and pop/rock buffs. There are many classical records that have several bands—either medleys of short selections, or something like the Elgar *"Enigma" Variations* which (on Philips 6500.481) has the main theme plus 13 variations on one side. In general, however, I would have to agree that most classical music listeners will want to play a record straight through most of the time, which would of course make the automatic band-selection feature of the Accutrac less-than-important to them. For other listeners, the pre-programming of a disc side could be a welcome convenience—playing various cuts in any sequence chosen, omitting some, repeating others, etc. without constant attendance on the turntable.

L.F. raises an interesting question regarding the choice of turntable and arm used in this system. The rumble and other measurements indicate that it is one of the quietest turntables around, and it seems to me that the motor chosen is matched to the chores the unit performs. Cycling is, however, on the slow side; but apparently the time taken is what the "brain" employed needs to get the message and translate it into action. As for the naming of the buttons, I suspect ADC designers had a time deciding what to call the "Clear" and "Reject" buttons since under certain conditions both will do the same thing.

The owner of the Accutrac, it should be pointed out, is limited to the use of the particular ADC pick-up supplied with it, since this pick-up had to be specially modified to accomodate the sensor beam device incorporated in it. This being the case, it perhaps is incumbent on ADC to substantiate that the arm supplied is indeed optimally mated to this pick-up—something which other companies offering a one pick-up/arm combination seem to have been doing.

How many of these units will sell is anyone's guess. My first thought, after the possible rock/pop market, would be the Niemann-Marcus shopper, or the audio buff who loves to practice one-upmanship. The future of this product, including its anticipated lower-priced versions, will be interesting to watch.

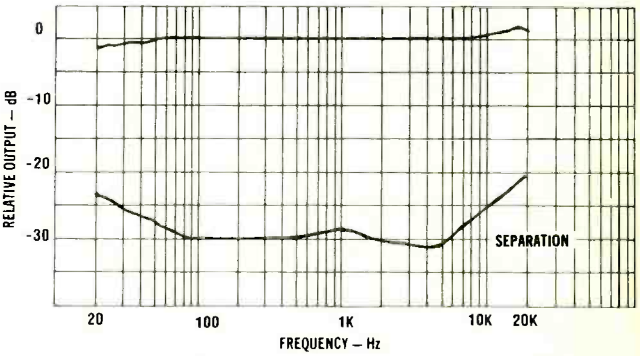
The Rabco ST-7 represents a new level of refinement for a product concept that dates back to 1968 and had, as its first commercial offering, a separate arm—the SL-8 which sold for $150. The arm evolved through various spin-offs, was combined with a table and came out as the ST-4 in late 1970. Comments then about the product ran something like: "Love the arm, can't stand the platter." In my view, the present version—the ST-7—is bound to engender the "love verdict" for both elements. It is on all counts a precision-made, quality product, and weeks of heavy use find it in as healthy shape as when I first took it out of the carton and set it up—which, by the way, takes some fussing to get the proper seating of the cartridge. Perhaps overshadowing this chore is the fact that the arm will accept any make of cartridge and will permit tracking at lower stylus forces than otherwise required, and without the need for anti-skating.

Listening reactions can be tricky, but methinks I do detect, using the same cartridge previously installed in a pivoted arm, a better definition of very low frequencies and a somewhat firmer stereo image.

On other characteristics, the ST-7's cueing action could be more precise; it tends to return the stylus to a few bars away from the point in the music at which you lifted it. As for positioning the arm to hit the lead-in groove, it is no different actually from any arm using a cueing lever (the most precise cueing-in is still by hand, albeit a steady hand with a light touch!); the difference here is that you tend to look at the pick-up head-on instead of from a side as in conventional arms. With a little practice, this minor problem can be overcome.

It is interesting to note that both ADC and H-K specify rumble by the DIN "B" standard which happens to be the most inflated rumble rating around, and one that gives seemingly better figures than the more rigorous (and, in my view, more accurate) CBS-ARLL rating. There is no direct translation from DIN "B" to ARLL but at 320 Hz, the DIN "B" number will be 6 dB "better" and at 50 Hz, about 5 dB "better." In any event, our own lab measurements show that even by the ARLL standard, both of these turntables would do better than −60 dB for rumble, which is an excellent mark for any turntable.

Either of these units would do a stereo system proud. The Accutrac boasts an unprecedented automation (for a single record side); the Rabco ST-7 boasts true radial tracking for any pick-up of your choice. The Accutrac can be remotely controlled; the ST-7 offers faster start-up time and lower VTF. For my taste, and speaking very personally, if I had to make a choice between the two I would pick the ST-7 by a narrow margin because it permits improved pick-up performance. I still must marvel, however, at the Accutrac "brain child." It seems that the "personalities" of these two products will appeal to different human personalities who doubtless will support these two new "revolutions" in turntable design.



ADC Accutrac 4000: Frequency response and separation, ADC LMA-1 cartridge.

VITAL STATISTICS

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT	
	ACCUTRAC 4000	RABCO ST-7
Turntable rumble (DIN "B")	−71 dB	−68 dB
Wow and flutter (WRMS)	0.027%	0.035%
Speeds available	33; 45	33; 45
Speed adjustment range	±4.5%	±5.0%
Speed build-up time	1.8 seconds	0.6 second
Platter weight	3 lbs., 2 oz.	2 lbs., 6.4 oz.
Motor/drive system	DC brushless/direct	DC brushless (Hall effect)/belt
Tone-arm type	Moderate "S" curve; effective length, $9\frac{5}{16}$ in.	Straight tube; straight-line (radial) tracking; length adjustable for pick-up.
Arm bearing friction (pivotal)	Lateral, c. 6 grams Vertical, c. 6 grams	Not applicable (see text)
Tracking force range	0 to 4 grams	0 to 3 grams
Stylus overhang	$\frac{19}{32}$ inch	Not applicable (see text)
Cartridge response	±1.5 dB, 20 Hz to 20 kHz	
Cartridge output	3.6 mV/5.5 cm/sec.	Not applicable; depends on cartridge installed.
Channel separation	29 dB at 1 kHz	

CIRCLE 8 ON READER SERVICE CARD

CIRCLE 12 ON READER SERVICE CARD

Tandberg 10XD Open-Reel Tape Recorder

General Description: The model 10XD is a recent offering in open-reel tape recorders from the well-known Norwegian firm of Tandberg. A three-speed model (15, 7½ and 3¾ ips), it is—the manufacturer points out—the world's first (and so far only) open-reel recorder that accepts 10½-inch reels and has a Dolby circuit that operates at all three speeds (in other recorders that have both 15 ips and Dolby, the Dolby does not operate at the 15 ips speed).

The Tandberg 10XD also employs Cross-Field biasing which is applied during recording via a special head. The tape actually passes between this head and the recording head. There is a separate erase head to the left and a separate play head to the right of the record/bias heads, making a total of four heads. The tape path from supply to take-up reels follows a carefully planned system of swinging arms, guides and capstans. Spring-loaded locks hold tape reels in place, and adapters (supplied) may be fitted over the spindles to accept the large-size (NAB) reels. The transport and electronics come in a handsome rosewood case that may be installed vertically, horizontally or at any intermediate angle.

In addition to mono and stereo recording and playback, the Tandberg has provisions for sound-on-sound recordings, echo recordings and input mixing (two stereo programs from different sources, or up to four mono sources). The unit's built-in preamp may be used as a microphone amp or as a mixer (mic plus other source) for distribution to loads other than the recorder itself. In fact, the 10XD can be used to play a mono tape and serve as a mic-amp at the same time.

The Dolby system included in the Tandberg will encode in recording and decode on playback. It also has provisions for copying FM programs (and other tapes) which themselves may be Dolbyized or not.

The transport action features fast-buttoning from any mode to any other (except from wind or rewind to record). Thanks to an automatic motion sensor, the transfers from one mode to another are quite rapid. During normal play mode, you can go directly into record at the touch of a button for the "flying start" kind of operation that permits "punch-in" recording or "electronic editing."

The Tandberg uses peak-reading VU meters that permit recording at relatively high signal levels without running into overload. They show signal levels after equalization. When used with Tandberg's HL tape or equivalent, zero dB on the meters is the guide-mark; when recording on other types of low-noise tape, the operator is advised to avoid needle deflections beyond the -2.5 dB mark on the VU meters. The meters, which remain in the record circuit as long as the deck is recording, will show input levels while monitoring the tape, regardless of whether the monitor switch is in source or tape position. Only when the deck is changed to playback mode will the meters read playback level. Thus, "visual control" of the input signal is always available when recording.

Layout of the deck is orderly and distinctly "contemporary" in visual appeal. Controls include the main power off/on switch; tape-speed selector; 10½-inch reel button; edit/cue button (which may be used in fast wind modes as well as when moving the reels by hand); the usual transport selector buttons; a pair of sliders for separate mic input level on each channel; a similar pair for line inputs; a third pair for output levels; record-select buttons for each channel; source/tape buttons for each channel; the sound-on-sound control; the Dolby system control. The deck has a four-digit tape counter with reset button. During use, a soft green pilot lamp indicates power on; the meters light up, and a small indicator glows over the mode key pressed. The sound-on-sound feature has its own pilot lamp. Mic inputs (phone jacks, balanced inputs) are on the front panel; the headphone output jack also is here. In and out line connectors (pin jacks) are at the rear where there are also connectors for a DIN cable and an optional remote-control accessory.

The Tandberg 10XD is supplied with an empty 10½-inch reel; two NAB hub adapters; signal cables. The owner's manual is outstanding for its thoroughness, clarity and two-color illustrations. It is printed on quality stock and in addition to containing carefully written instructions for using the deck in all its modes, it contains valuable hints on microphone use and placement, tape copying, language learning, mixing, editing and splicing, simple maintenance, and a very good rundown on the Dolby system.

Test Results: Using low-noise tape (Maxell UD, which is one of the tapes Tandberg states is suitable for use on the 10XD), MR's testers put the deck through its paces and exceeded Tandberg's performance claims in every instance. Mechanically and electrically, the deck ran beautifully and easily justified Tandberg's claim as being the best tape recorder they have yet produced.

Without doubt, in the view of MR's testers, the use of the Cross-Field biasing system can be credited with improved response, and especially so at the lowest speed of 3¾ ips which, on this deck, is comparable to 7½ ips on many other models, particularly when used with the unit's built-in Dolby system.

Indeed, response at all speeds—as well as the deck's low distortion, superb signal-to-noise ratio, and overall mechanical excellence—rivals or exceeds many "pro" decks. Wow and flutter, for instance, were among the lowest MR has yet measured; response, among the most linear and extended; and so on. The metering system here —once the user gets accustomed to it— actually enables taping at higher input levels without the danger of tape overload, and the self- adjusting mic inputs could be of help to the "live" recordist. The preamp circuitry is especially quiet, and adds no hiss even when the input pots are run wide open.

MR notes too that the Tandberg 10XD is "spec'd" to within ± 2 dB which is more rigorous than the ± 3dB used for many other models and which, when checked out in MR's lab, did indeed conform to the test findings.

General Info: Deck is supplied in rosewood wraparound. Overall dimensions are 17 by 17⅝ by 7⅛ inches. Weight is 36 lbs. Owner's manual is excellent. Price: Quarter-track version (unit tested), $1,399; half-track version, $1,499.

Individual Comment by L.F.: When checking out a recorder costing over $1,000 the question arises: is this a professional deck with home-use features, or is it a home machine with professional features? The requirements of the studio engineer are not, of course, always the same as those of the serious home recordist. For instance, a pro wants to be able to "punch in" the recording function instantly, even when the transport is running in the playback mode. This facility—supplied in a home machine—could produce accidental erasure of recorded tapes if the user is not aware of it and prepared to use it with care. In this instance, Tandberg has chosen to go the professional route with the model 10XD.

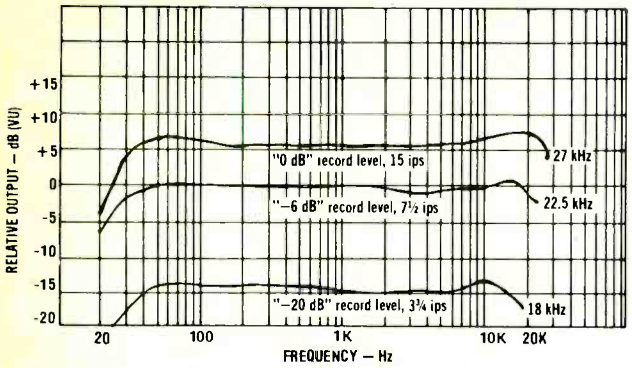
On the other hand, a truly professional tape machine normally has XLR three-terminal balanced line and mic inputs and outputs, whereas the 10XD has unbalanced two-terminal line inputs and outputs (using standard home-type pin jacks), and balanced mic inputs which are self-adjusting depending on mic impedance but which do use three-terminal phone jacks for access to the mic preamp circuitry.

While the 10XD certainly performs like the best pro machine I have ever encountered, its "orientation" as a product is obviously to the home recordist. The true pro machine would have adjustments for bias and equalization and for Dolby calibration; the 10XD has none of these. On the other hand, this simplified approach which still provides uncompromised audio and mechanical performance (providing the proper tape is used) may well endear the 10XD to a great many tape recording enthusiasts who, while lacking the test instruments to make studio-precision adjustments, still want (and can get with this unit) *ne plus ultra* specifications and reliability. Our test results indicate this is a tape recorder of superior capabilities, and—as Tandberg's best to date—the manufacturer's enthusiasm and references to "professionalism" in the literature are understandable.

Individual Comment by N.E.: I agree that there are certain features normally found on all-out studio tape machines that are not obvious on the Tandberg 10XD, but this is sort of like saying that Model Z is a beautifully performing car with both sport and passenger attributes except that it won't compete too well at the Indianapolis races. Tandberg needs no apologies, and I do not need to assume the role of their apologist, but the company his its own ideas of tape recorder design. The kind of recorder they offer with the model 10XD seems to me to represent a nice balance of features and options that should appeal to a significant market which, for want of a better term, might be called the "semi-pro" or "advanced consumer" tape buff. To me, the type of connector is less important than the quality of the signal passing through that connector and on this count, not to mention mechanical smoothness and reliability, the 10XD can stand up to anything I know of in its price class, or even at higher cost.

On specific points, the Tandberg's metering system is more carefully worked out than in most home-type machines, and it helps in recording tapes with optimal S/N characteristics. The mic inputs permit running cables over 1,000 feet in length, and the outputs (not the DIN socket, of course) have enough dB margin to permit cables about as long. The Dolby system has inherently excellent "tracking" and, since the level controls come after the Dolby circuitry, there really is no need to recalibrate. There is, in any event, an internal Dolby adjustment for a qualified technician (should the need arise). Bias is not "user-adjustable" although the Cross-Field head under the head-cover can be adjusted, again by a qualified tech using proper instruments. Equalization is very precise here and "follows" bias closely, obviating the need for a separate EQ adjustment.

In short, the obvious "studio" features are, in this unit, deliberately "hidden" since Tandberg feels they could be loused up by inexpert hands. "As is," the 10XD is one beautiful machine, deliberately designed to Tandberg's tape-recorder "philosophy" and intended to appeal to a broad range of users that straddles some "pro" and some non-pro areas. Call it what you want, the Tandberg 10XD is one hell of a tape recorder.

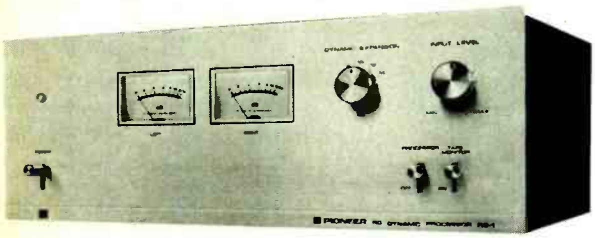


Tandberg 10XD: Record/playback response, at three available speeds, using Maxell-UD tape.

TANDBERG 10XD TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC (using Maxell UD tape, noted as suitable by mfr. for this deck)	LAB MEASUREMENT
Record/playback frequency response	
15 ips, 0 dB	±2 dB, 29 Hz to 27 kHz
7½ ips, –6 dB	±2 dB, 32 Hz to 22.5 kHz
3¾ ips, –20 dB	±2 dB, 37 Hz to 18 kHz
Harmonic distortion	
+3 VU	0.7%
0 VU	0.5%
Best S/N ratio	61 dB (unwtd; less Dolby)
Input sensitivity, mic	1 to 130 mW (self-adjusting to impedance)
line	180 mV (0.1 to 5.0 V, automatically adjustable)
Output level, line	1.8 V (at 150-ohm impedance)
headphone	5 mW (at 8 ohms)
Bias frequency	c. 125 kHz
Speed accuracy	±0.2%
Wow & flutter (WRMS), 15 ips	0.018%
7½ ips	0.030%
3¾ ips	0.075%

Pioneer RG-1 Dynamic Range Expander



General Description: The Pioneer RG-1 is a self-powered stereo (two-channel) dynamic range expander for use in a playback system. It works on ordinary (unprocessed) signals from tape, disc or broadcasts to help enhance loud passages while leaving softer passages unaffected. The audible effect is intended to enhance the impact of playback by effectively restoring a measure of the dynamic range that may have been compromised or limited during the original recording or broadcast (or both, in some instances). A side benefit of this process is the relative reduction of background noise (hiss, surface noise, etc.) by the enhanced upper signal level.

A special "sensor" circuit is designed to prevent the expansion action from being stimulated by noise elements in the signal, and the amount of expansion itself is selectable by the operator in 2 dB increments from 6 dB to 14 dB via a front panel control. The degree of expansion is displayed on left and right channel front-panel meters. Another control knob selects the input level signal to match the system level without the Processor in the circuit. A switch that puts the unit in and out of the system is available for instant comparisons (by ear).

The RG-1 is designed to be patched into the tape record/monitor jacks of a normal stereo amplifier or receiver; a duplicate set of jacks to restore this facility to the stereo system is provided at the rear of the unit.

Test Results: One of the most gratifying things about the RG-1 in use is its freedom from audible "pumping" or "breathing." MR could not measure the attack and decay time of this expander, but apparently its designers have chosen the unit's operating parameters very cannily. To sense and control the desired variations in signal gain, the range of frequencies that is used to develop the controlling signals lies primarily above the fundamentals of most musical instruments. When played loudly, says Pioneer, these instruments tend to produce more harmonics than when played at lower volume. The key, then, to the RG-1's action is the presence or absence of frequencies in the region of 2.5 kHz rather than overall sound without regard to frequency. In MR's tests, the effect of the RG-1 in the system did seem to bear out this theory by enhancing the dynamic range of much program material played through it.

From the standpoint of undesirable contributions to the signal (noise and distortion in the device itself), the RG-1 presents no problem. At an output of 1 volt, the unit produced a mere 0.05% THD (half of the 1% allowed by the manufacturer), and its signal-to-noise ratio was measured as 98 dB, close enough to the 100 dB claimed by Pioneer. Its maximum output is 6.6 volts, which tops by .1 volt the level claimed by the manufacturer.

General Info: The RG-1 dimensions are 13¾ inches wide; 12⅝ inches deep; 5⅜ inches high. Weight is 11 lbs., 10 oz. Unit is supplied in metal case with brushed aluminum front panel. Price is $175.

Individual Comment by L.F.: Two things about the RG-1 make it different from other single-ended expanders I have encountered. And there's one thing that makes it a bit trickier to use than some others.

First and most important, the expansion degree is variable and selectable by the user. I approve, since no audio enthusiast likely to use such a device is naive enough to suppose that every recording engineer employs the same degree of compression for every recording. Or that even the same recording engineer will compress the same for all of his own recordings. Of the five expansion options in the RG-1, I generally chose the 8-dB or 10-dB positions for classical listening, while for pop and rock material I preferred the higher settings of 12 dB and sometimes 14 dB.

Another thing I liked in this unit is its ability to reduce noise by as much as 6 dB as a side benefit of expanded dynamic range. But to enjoy this feature, the unit must be correctly set, which leads me to the one flaw I feel the RG-1 has.

And that concerns the setting of the front-panel input level control. From reading the operating instructions, you might get the idea that setting this control is not at all critical, and that it may be done once, and casually at that. Nothing could be farther from the truth. In my opinion, that control should be adjusted each time you choose to move the expansion control from one of its settings to another, and the adjustment takes careful doing. It was only after considerable experimentation that I realized that simply reading the meters is not enough to set up the RG-1 for optimum signal processing. I finally hit on a method that involves both careful listening and study of the meters. The idea is to keep adjusting the input level control until *average* loudness levels remain essentially unchanged as you switch the RG-1 in and out (using the Processor/Off switch). It may take some time to establish just when average music levels are being experienced but with patience it can be done. It was only after going through this procedure to set correct input levels that I was able to realize full benefit from the RG-1. That benefit, I would emphasize, is great: crescendos come through with full impact, while softer passages remain unchanged and seem audibly freer of noise than before.

Individual Comment by N.E.: If one is using program material that already is "noise-processed" and plays it through a reciprocal decoding circuit (such as with the Dolby system), then a device such as the RG-1 becomes a redundancy. However, all that we are likely to listen to has not been so processed and it is for that large amount of source material that the RG-1 is intended, as a kind of "universal" expander to restore the dynamic range that has been compressed in the recording and/or broadcast studio.

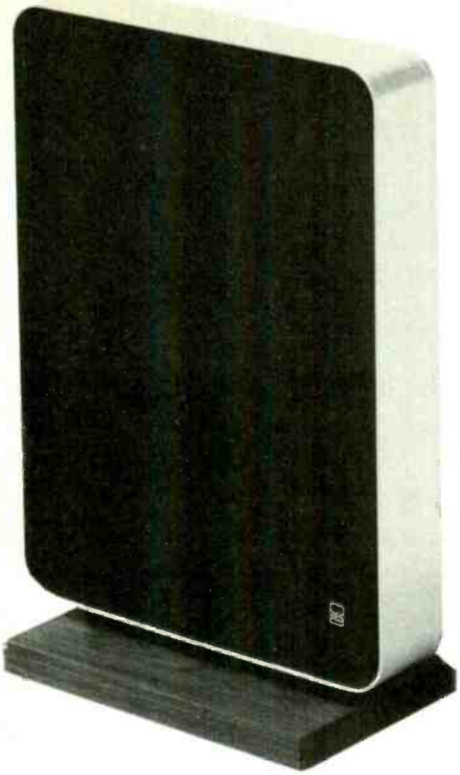
In my view, this is theoretically desirable, although I do not regard it as the most important or even the most readily attainable way to improve the realism of a playback system. I would say a more basic step would be to improve the dynamic range capabilities of loudspeakers. In a playback system of given performance ability, with all other things staying equal, a dramatic and audible improvement can be realized by changing to speakers with greater dynamic range. This also suggests a negative corollary, or at least raises a nice question: how can a signal source whose own dynamic range has been expanded be reproduced successfully by a loudspeaker whose own dynamic range remains—because of its inherent design—limited vis-a-vis the signal it is getting?

However this question be answered, I feel there are others regarding the RG-1. On the matter of setting the adjustment—the question raised by L.F. in his comments—I also note that the signal meters do not register when the Processor switch is moved to its "off" position. Thus, the basic matching of signal levels—essential to the proper working of the RG-1—must be done by ear. The meters actually are only monitoring what the RG-1 is doing but since this is "after the fact" and cannot be related to another meter reading for comparison, the whole process is highly subjective, and you get a feeling that the meters are suddenly less-than-very-important here. L.F. rightly criticizes the owner's manual instructions on this point and I would underscore that comment, adding too that there are other areas of the manual that could stand some clarification—and this applies both to text and illustrations.

In general, my overall reactions were not as enthusiastic as L.F.'s. In my listening tests, the RG-1 proved somewhat effective on well-recorded rock and pop material, less so on classical stuff and least of all on small ensembles such as chamber music groups. When noise content has appreciable midrange components, such as FM hiss on weak stations, the RG-1 brings up everything including the hiss, and the net effect is rather like having someone automatically raise the volume control and then lower it. It would seem that the effectiveness of the RG-1, in addition to the critical setting up indicated by L.F., depends too in some measure on how good and clean the original source material is to begin with.

PIONEER RG-1 SIGNAL PROCESSOR: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Maximum output for 1.0% THD, at 1 kHz, 14 dB expansion	6.5 V
THD at 1 V out, 1 kHz, 14 dB expansion	0.05%
S/N ratio, 1 kHz, 14 dB expansion	−98 dB
Dynamic expansion (nominally 6, 8, 10, 12, 14 dB)	see comments in text
Attack time (nominally 0.5 msec)	see comments in text
Release time (nominally 80.0 msec)	see comments in text
Power consumption	12 watts



B.E.S. "Geostatic" Speaker Model U-60

General Description: The U-60 is one of a series of speaker systems (the letters B.E.S. stand for Bertagni Electroacoustic Systems, Inc., the manufacturer) known by the generic title of "geostatic" and which utilize as the diaphragm (sound-producing element) a relatively large section of expanded polymer material held within a surrounding aluminum frame. The frame (which theoretically could be of any material) serves only to support the polymer. There is no need, in this design, for an enclosure in the usual sense. Driven by a ceramic magnet, the entire polymer section loads directly to the room from both sides and thereby radiates sound from its nominal "front" and "rear."

The design makes for an extremely thin speaker that is light enough in weight to be wall-mounted or suspended from the ceiling, although the particular unit sent for testing came with a small pedestal base to which it may be attached and placed on the floor.

The unit is quite attractive and presents a neat contemporary-style look. Size and weight do not preclude experimenting for best placement in a given room. The U-60 is recommended for use with a minimum amplifier power of 10 watts (RMS per channel); maximum recommended RMS amplifier power per channel is 100 watts, with a continuous RMS power rating of 40 watts. Input impedance is 8 ohms, and connections are made to a pair of screw terminals near the bottom of one side. No controls are provided.

Test Results: The manufacturer makes several performance claims for this speaker which MR could only partly confirm. To begin with, the impedance measured was indeed 8 ohms or higher across much of the audio band, dipping slightly to about 6 ohms from 2 kHz to 10 kHz and then coming closer to 8 ohms again in the 20 kHz region.

To measure response, special techniques had to be used because of the unconventional nature of this speaker. Wide-band pink noise was used as a program source. A sweeping narrow-band filter in the spectrum analyzer was connected to a calibrated microphone positioned more than 1 meter away from the front of the U-60 to produce the spectrum curve shown. The results are somewhat meaningful only from around 50 Hz to 15,000 Hz. (The published specs claim response within ± 3 dB from 38 Hz to 18,000 Hz; MR could not achieve this with the test samples it had).

Sensitivity claimed for the U-60 is 91 dB/SPL with 1 watt input at 1 meter. MR's tests measured 87 dB at the very best—and we had to integrate several readings to obtain that result. This figure, admittedly, might be in error, but when the U-60 was compared with another speaker system having a known sensitivity of 92 dB (on various source material including white noise, pink noise and music), the U-60 simply sounded far less efficient regardless of its position in the room. The speaker is as low-efficiency a reproducer as its testers have ever encountered.

Of course, low efficiency as such is not necessarily a demerit in a speaker, provided it can handle enough amplifier power to produce adequate sound levels. In MR's view, it would take pretty close to the maximum 100 watts for the U-60 to reach such levels in most rooms. Naturally, the smaller the room the less power demanded for comparable levels.

The most impressive aspect of the U-60's performance seemed to MR to be its excellent dispersion. One can literally walk around the speaker without experiencing the slightest variation in high-frequency energy content while the system is playing. When listening to a pair reproducing stereo, the stereo "imaging" is as good as anything heard from conventional speakers and perhaps a bit better. Like all omnidirectional systems, of course, placement to achieve this effect can be critical—in MR's tests, best results were obtained in a 14-foot by 20-foot room with the pair of speakers positioned along the *long* wall, each a little more than 1 foot from the wall, and spaced about 10 feet apart.

General Info: The U-60 is 23½ inches high, 17⅝ inches wide, and 3¾ inches deep. The wooden base to which it may be fastened is 14½ inches wide, 9½ inches deep, and it raises the total height by 2 inches. Price is $189.

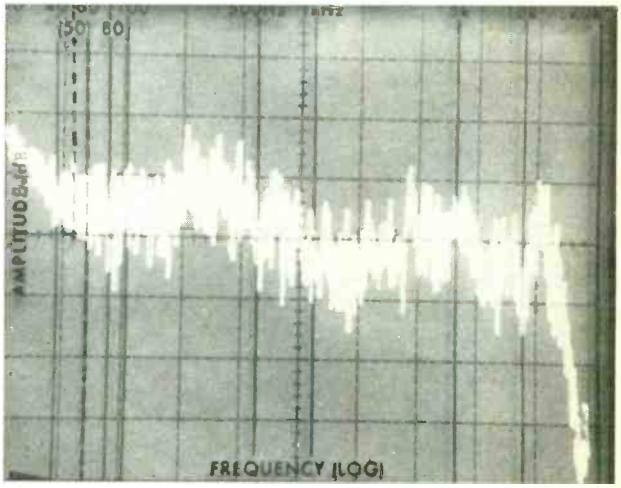
Individual Comment by L.F.: The dream of a "flat speaker" that does not require a box-type enclosure is almost as old as high fidelity itself, and closely parallels that of the flat-screen home TV set. I must admit that I, too, was taken by this dream several years ago and anxiously auditioned a flat-speaker product once offered by Fisher Radio, known as the Sound Panel. That speaker was developed by an Argentinian professor of electronics and audio engineering by the name of José Bertagni. It was very flat indeed, and could be hung on a wall, much like a painting (in fact, Fisher Radio offered grille variations ranging from realistic to abstract art, as I recall). There was only one thing wrong with those sound panels. They had little bass response and weren't terribly efficient. The highs were not particularly outstanding, either.

Well, the years have passed, and Professor Bertagni has been hard at work improving his patented concept of the enclosureless speaker. Recently, he established a company in the U.S. called Bertagni Electroacoustic Systems (B.E.S.) which currently produces four models of "flat" speaker systems. To my taste, the new Geostatic units (as they are called) are aesthetically much more pleasing than the old sound panels.

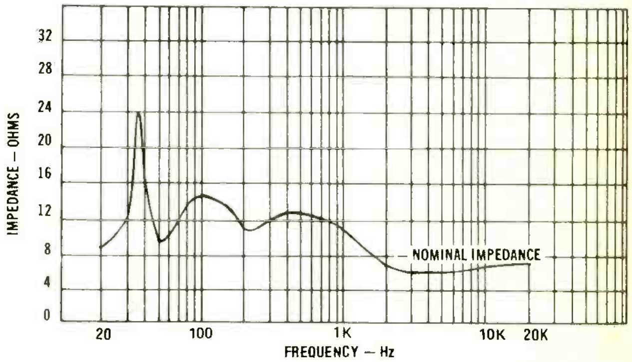
The sound is generally better too, but the U-60 simply does not go down as low in frequency as I would like it to. I seemed to be missing at least a full octave of lower bass compared with almost any other reasonably priced system I listened to in making comparative A/B tests against the U-60. In all fairness, I must state that clarity of sound and absence of distortion are a plus for this system, but it just doesn't have the response I would expect from a speaker selling for more than $150.

I will concede that the present version of the B.E.S. Geostatic Speaker System runs rings around that earlier version made for Fisher Radio—but even excellence is relative, and I believe that the day of the definitive "flat" speaker is still to come. It may well come from Professor Bertagni at that, if past improvements are any indication of what's ahead.

Individual Comment by N.E.: The pair of U-60 speakers I auditioned were so different in tonal output that I must conclude one of them had to be defective. It simply did not make it with any appreciable energy output beyond about the 2,000-Hz frequency mark. Reproducing music it sounded muffled and "gutless." I must attribute this to a serious defect, since the other unit sounded much better, with a relatively prominent midrange and high end out to at least 15,000 Hz, and with superior dispersion in the upper frequency region. With either unit, efficiency is quite low and in a relatively large room one wants to crank up the amplifier gain to get decent sound levels, except that if you try for too much output there is a tendency to sonic "breakup" on complex orchestral passages. In a smaller room, where less volume is needed (or can be tolerated) and you can sort of "snuggle up" to the speaker, it makes a more acceptable kind of sound. In the bass region I found that doubling became evident at about 50 Hz but that the speaker would respond somehow to tones down to 30 Hz. Interestingly enough, with the unit that had a defective midrange and high end, even the bass on music sounded thin, demonstrating that the overtones contributed by upper frequency response are germane to a full perception of bass. My present feeling about this speaker is that it is an interesting "work in progress" that has considerable promise as a possible future marvel, but in its present form, at least on the basis of the units tested, it is not my audio cup of tea—although I am willing to admit it may be someone else's.



B.E.S. U-60: Wide-band pink-noise response.



B.E.S. U-60: Impedance vs. frequency.

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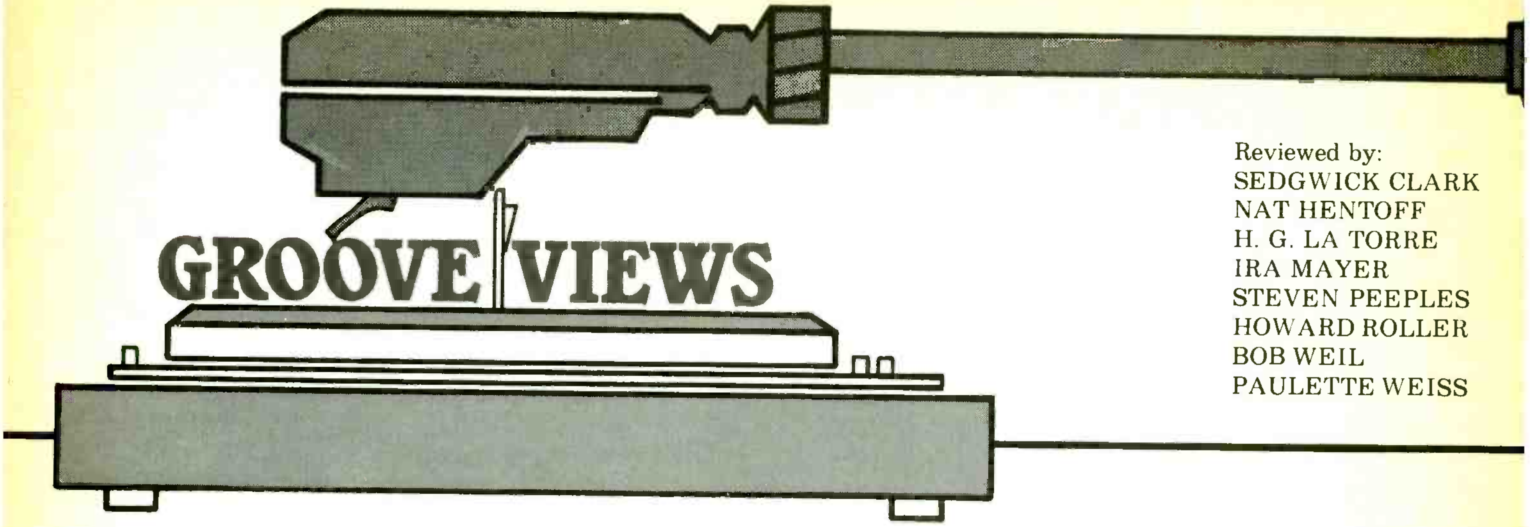


GROOVE VIEWS

Reviewed by:
SEDGWICK CLARK
NAT HENTOFF
H. G. LA TORRE
IRA MAYER
STEVEN PEEPLES
HOWARD ROLLER
BOB WEIL
PAULETTE WEISS

POPULAR

ALLMAN BROTHERS BAND: *Win, Lose or Draw.* [Johnny Sandlin and the Allman Brothers Band, producers; Sam Whiteside and Carolyn Harriss, engineers; recorded at Capricorn Sound Studios and the Record Plant, Los Angeles, Cal.] Capricorn Records CP0156.

Performance: **Excellent**
Recording: **Their best studio sound to date**

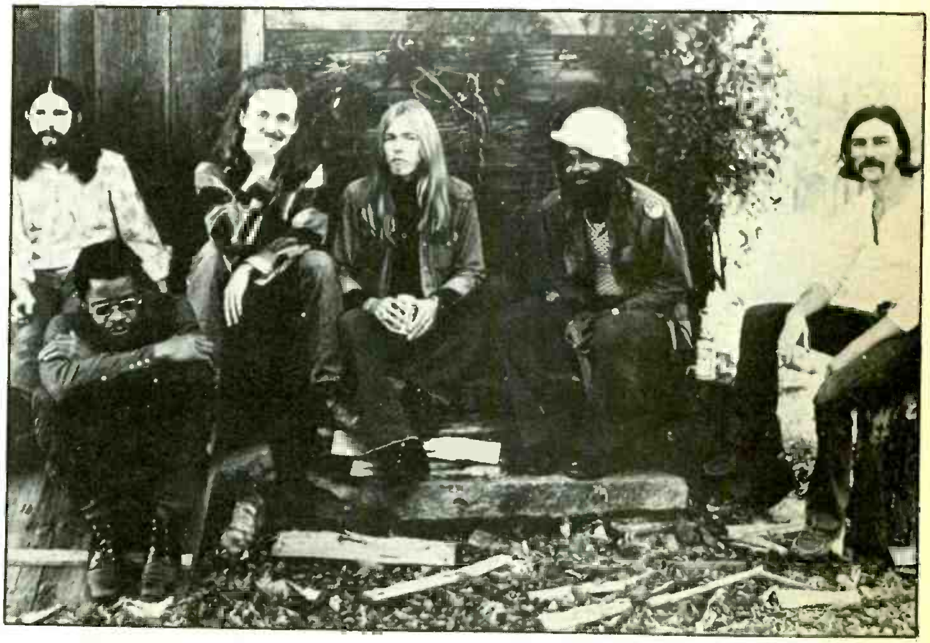
Who would have thought that the Allman Brothers Band would fall victim to Einstein's Theory of Relativity? Well, it's true, and like old Al said: "It all depends on your frame of reference." If one listens to *Win, Lose or Draw* as an isolated entity, it comes off as a good, solid album that successfully blends blues, country and light jazz with polished rock 'n' roll. The problem lies in listening to this album in the context of the Allman Brothers' previous accomplishments.

There is nothing on *W, L or D* that comes close to Allman classics like "Whippin' Post," "You Don't Love Me" or "Blue Sky." The blues cuts (contributed by Muddy Waters, Billy Joe Shaver and Greg Allman) seem impersonal without Duane Allman's incomparable slide guitar. While Richard (formerly Dickie) Betts is a technically excellent guitarist, his clear, rich melodic lines are less suited to the blues than to his own country-influenced compositions. Of these, "Just Another Love Song" could be retitled "Son of Ramblin' Man," and "Louisiana Lou and Three-Card Monty John" is good basic country boogie. The extended instrumental "High Falls" has its moments, but lacks the energy and cohesiveness of "Jessica" (its forerunner). The title cut is a Greg Allman composition that might be the high point of the album if it didn't bring his "Jessica" to mind.

The Allman's musicianship and Sandlin's production are flawless throughout *W, L or D,* and even when things get busy, the instruments never get in each other's way. Bassist Lamar Williams and pianist Chuck Leavell have added their own styles to the group's sound, and the New Allman Brothers Band really uses the studio (vs. going in and recording a basically "live" gig as they used to).

It's time, however, that this New Allman Brothers Band explored some uncharted territory instead of sticking to the same old trails. Perhaps then this talented group will move back into the winner's circle, and not have to settle for another draw. B.W.

ERIC CARMEN: *Eric Carmen.* [Jimmy Ienner, producer; Jack Sherdel, engineer; recorded at O.D.O. Sound Studios, New York, N.Y.] Arista AL4057.



ALLMAN BROTHERS: Flawless musicianship and production

Performance: **McCartneyesque**
Recording: **A bit carried away**

As lead singer, composer and rhythm guitarist for Raspberries, Eric Carmen was instrumental in putting together some of the best light rock since *Rubber Soul*. Unfortunately, Raspberries (along with Bread and Badfinger) was dismissed by many "rock aficionados" as lightweight, presumably due to its concentration on melodies and harmonies as opposed to heavy lyrics and instrumental breaks. So, the Raspberries never got the chance to ripen, and Carmen hopped off the proverbial bush and went solo.

Carmen has teamed up with the veteran producer/engineer team of Ienner-Sherdel (Three Dog Night, Grand Funk) to put together a highly polished work that features some real gems, which are occasionally buried in a beautiful but distracting setting. "Sunrise," "Last Night," "My Girl" and "No Hard Feelings" (which chronicles the Raspberries' career) achieve a happy balance between pop and rock that brings back the early Beatles, Beachboys, Turtles and Hollies. But when Carmen moves over to keyboards on "Everything," "Never Gonna Fall in Love Again" and the overplayed "All By Myself," he and Ienner follow that "Long and Winding Road" off "The Bridge Over Troubled Waters" into the sea of overproduction.

Then again, most former group members get a bit carried away their first time out on their own, and the production here is so clean that it never feels cluttered—just a little over-dramatic. So, it's up to Carmen and Ienner to show us next time around whether *Eric Carmen* is a step towards some really memorable music or just another pretty lp. B.W.

KATE & ANNA McGARRIGLE: *Kate & Anna McGarrigle.* [Joe Boyd and Greg Prestopino, producers; John Wood, engineer; recorded at A & R Studios, New York, and Sunwest, Hollywood, Cal.] Warner Bros. BS 2862.

Performance: **Highly emotional**
Recording: **Sparkling**

French-Canadian by birth, the McGarrigles combine that unique hybrid background with American folk, rock, religious and Cajun influences, developing a writing style that is at once individually ethnic and universal. Though each writes separately, the sentiments are at one. And the blending of the two high voices is quite unusual for contemporary popdom.

Kate and Anna have seen their original compositions "Heart Like a Wheel," "Cool River" and "Work Song" recorded by Linda Ronstadt and Maria Muldaur. For their own first recording, however, they do not rely on the familiar. "Heart" is included, but in an arrangement dissimilar to Ms. Ronstadt's. Yet, deeply touching as "Heart" is, there are still other songs of equally somber character that are even more disturbing in their impact.

"Go Leave," for example, stands out for the simplicity of the song itself, as well as for the delivery—just vocal and guitar. "My Town" and "Mendocino" are just as compelling, but the scope is much grander, the arrangements much fuller. "Foolish You" and "Swimming Song" are included to keep the sisters from getting downright morbid, and the album ends with "Travelling on for Jesus," very much a song of hope.

Boyd and Prestopino have avoided overburdening the McGarrigles and the result is a crystal-clear recording that highlights every nuance. Few debuts are as masterfully executed.

I.M.

KISS: ***Destroyer.*** [Bob Ezrin, producer; Jay Messina & Corky Stasiak, engineers; recorded at Record Plant, New York, N.Y.] Casablanca NBLP 7025.

Performance: **Frothy**
Recording: **Studio-brilliant!**

For those interested in the function of the producer on rock 'n' roll albums, few groups offer as vivid an illustration as Kiss. Compare the current lp to the band's previous efforts and it is obvious that more than the expected "maturing" has taken place. Listen to what happened to Alice Cooper between "Pretties for You" and "Love It to Death." The phenomenal changes in both cases can be traced directly to Bob Ezrin.

Regardless of what you think of hard-edged rock or of the outrageousness of Kiss or Alice Cooper's stage performances, there *is* something very musical going on. And if rock has become theater, Ezrin is probably the single ablest person making the transition from stage to vinyl workable.

On the musical side, he teaches bands how to play their instruments and use their voices for record. (He once told a reporter that in the early days of Alice Cooper he found himself putting a group member's fingers on the proper frets of his guitar.) And he frequently uses studio musicians to obtain the sound he wants. Too, he polishes melodies so that each song, while sometimes deceptively cacophonic (you can't listen to this music at low volume), has its own natural flow.

Technically, Ezrin has mastered the studio. It is not just that there are multiple tracks available to him, but there are layers upon layers of sound on *Destroyer* that subtly reveal themselves as you become more familiar with the record—or as you listen in different ways. The opening car sequence, for example, if heard through headphones, puts you behind the wheel. And at various points there are voices or solos that are virtually undetectable until you think you've heard everything that's there.

None of this is to understate Kiss's own contribution. The material they've come up with is exceptionally strong and there's a ballad, "Beth," that's surprisingly gentle and appealing. But for anyone seriously interested in the state-of-the-art of rock-cum-technology, knowledge of what a man like Ezrin can do in a studio is absolutely essential. I.M.

NEW RIDERS OF THE PURPLE SAGE: *Oh, What a Mighty Time.* [Bob Johnston, producer; Ben Tallent, engineer; recorded at the Record Plant, Sausalito, Cal.] Columbia PC-33688.

Performance: **Poor**
Recording: **Decent, considering**

THE FLYING BURRITO BROTHERS: *Flying Again.* [Norbert Putnam and Glen Spreen, producers; Glen Kolotkin, engineer; recorded at CBS Studios, San Francisco, Cal.] Columbia PC-33817.

Performance: **Rather inspired**
Recording: **Complimentary and uncluttered**

It's nice to have a lot of vocal assistance on an album, but it's pointless when they clutter the tracks, as is the case with the last Columbia effort by the New Riders. Additionally, the lack of original material does little to help. Covers of tunes like "Take a Letter, Maria" and "La Bamba" should have been left in the can.

The Riders started out as an extension of patriarch Jerry Garcia's pre-Dead folk/country/bluegrass predelictions, and therein lies their credibility. When they try, on this album at least, to rock 'n' roll, they lose credibility. This is not to say that they shouldn't experiment, but on this album the rock 'n' roll ventures don't consummate themselves well at all. "Strangers on a Train" is a prime example. But even more country-oriented material by Rusty Wier and Ray Wylie Hubbard is lackluster.

As a whole, the album is unnecessarily busy, due probably to the large number of guest voices, and lacks coherence.

On the other side of the fence is "Flying Again" by the Burritos, newly refried (with two original members). The album's backgrounds are much less cluttered than the Riders' album, and the Burritos have a greater sense of artistic direction and care. Putnam's clean production allows the Burritos' five members to sound like five musicians doing well, while the Rider's album gives the impression of 20 musicians doing creative injustice to themselves.

There are several standout tracks on the Burritos' effort that show they continue, as individual artists and as a group, to evidence a firm grasp of the progressive area between hard-core country and hard-core rock 'n' roll first advanced by original Burrito co-founders Gram Parsons and Chris Hillman six years ago. The Burritos of today sound somewhat different, but nonetheless high and lonesome. S.P.

HARRY NILSSON: *Sandman.* [Harry Nilsson, producer; Richie Schmitt, associate producer and recording engineer; recorded at RCA studios.] RCA APL 1-1031.

Performance: **Good**
Recording: **Good to muddy**

Let's not pussyfoot around here—Harry Nilsson is the greatest thing since Contadina stuffed those eight great tomatoes in that itty-bitty can. A uniquely creative force in popular music he is most closely associated with the rock mainstream, yet is responsible for the straight pop classic "Without Her" and the delightful *A Little Touch of Schmilsson in the Night* album, a collection of standards sung in a sweet old-fashioned tenor.

Perhaps his most endearing quality is a loony, devil-may-care attitude toward most things, including himself and his music. This gets a bit out of hand on *Sandman*, however. His parodies of diverse musical styles are fun, many of them witty on several levels, but none are as musically satisfying as the originals. And that's the key: who is supposed to be satisfied by all this? I found the disc generally enjoyable, but I suspect that Harry & Co. had a ball—and I further suspect this has been the trend of the last three albums. Not an unfounded suspicion, this, for Nilsson, playing two drunks on a rap piece called "The Flying Saucer Song" has himself say to himself, "It's your record . . . carry on, do whatever you want."

The recorded sound is a bit muddier than usual here. Nilsson has always kept a strong hand in, whether producing his material or not, and the results have sparkled. Perhaps this lack of clarity is due to the numerous assisting rock stars on *Sandman* (Ringo Starr, Joe Cocker and Leon Russell amongst them) who may have stuck a finger or two in themselves. Perhaps it's a matter of trying to cram too many tomatoes into the can. But at least there's not a real lemon in the bunch. P.W.

PURE PRAIRIE LEAGUE: *If the Shoe Fits.* [John Boylan, producer; Paul Grupp, engineer; recorded at the Record Plants, Sausalito and Los Angeles, and Capitol Studios, Hollywood, Cal.] RCA APL1-1247.

Performance: **Tasteful country rock**
Recording: **Unobtrusive**

At some point, one has to draw a line of demarcation between slick, lifeless production/performance and smoothly laid-back production/performance. PPL's latest, their fourth, is not really as raw as their first two, but shows a continuing vocal and instrumental maturation that was hinted at in their last album, "Two Lane Highway."

With a slightly more serious approach to song-writing, and a 60/40 ratio of vocal mix over instrumental mix, this album has a coherence that is fresh, to say the least. Boylan's relatively low- key production allows the vocal harmonizations to fully develop and at no point interferes with extraneous backgrounds.

PPL, as a group, evidences a greater understanding and control of their diverse influences— country, rock, blues, and a hint of swing jazz here and there—resulting in their most tasteful album yet. A solid effort. Laid-back, yes; lifeless, certainly not. S.P.

JAZZ

JOHN LEE & GERRY BROWN: *Mango Sunrise.* [Skip Drinkwater, producer; Emiel Elsen, Mike Bobak, Ralph Moss, engineers; recorded at Dureco Sound Studio, Weesp, Holland, Morgan Sound Studio, Brussels, Belgium, and Electric Lady Studios, New York, N.Y.] Blue Note BN-LA 541-G.

Performance: **Admittedly good**
Recording: **A producer's dream**

This album won't grow on you. You'll either love it from the start or save it for your next Sunday outing as a frisbee.

Bassist John Lee and Drummer Gerry Brown are presently members of Larry Coryell's Eleventh House. Along with producer Drinkwater, they have released an album of contrasts. The production verges on the lavish ("Mango Sunrise," "Keep It Real"), includes Arp string arrangements, and also features very raw, rhythmic tracks and solos. Amazing for an album recorded in only two months.

Much credit to all the engineers and studios for keeping things so clean and balanced. But many of the pieces are repetitious, making this one of the most mechanical-sounding albums created in a great while. Yet, it does contain some excellent guitar work—most of it in a somewhat frenetic, intense, free-form setting—specifically by Eff Alberts.

If you like synthesizers galore and frantic funk, then pick this one up. If not, buy a frisbee—it's cheaper.

H.G.L.

DAVID SANCIOUS: *Transformation (The Speed of Love).* [Bruce Botnick and David Sancious, producers; Bruce Botnick, engineer; recorded at Caribou Ranch, Nederland, Colo.] Epic PE 33939.

Performance: **Invigorating**
Recording: **Commendable**

Since leaving Bruce Springsteen's side, David Sancious has in a few short months issued an outstanding premier album, played album dates with (among others) Stanley Clarke and Billy Cobham and now recorded a second effort.

The growth of Sancious as a group leader, musician and composer is remarkable. He has managed to take jazz, rock and blues and fuse the different musical idioms in an intelligent manner, avoiding the pitfall of placing speed (dexterity) and technique above style and composition. Here, there is none of the ear-splitting, frenetic results others have reached when dealing in this form of music.

For this release, Sancious has gone to exotic Caribou Ranch in Colorado and teamed up with the well-known engineer/producer Bruce Botnick. The move has produced a crisp, sharply defined sound on both guitar and drums, while the keyboard instruments (synthesizers included) elicit a clear-cut individuality. Often, the over-dubbing of synthesizer tracks onto existing piano and organ tracks results in a carnival calliope effect.

The band, which has been named "Tone," includes Gerald Carboy, bass, and Ernest "Boom" Carter, drums. It has matured with Sancious and they perform more tightly together on the new selections. Especially impressive is the track "Sky Church Hymn #9," dedicated "To James Marshall Hendrix & 'Daddy E' (The Spirit of the Blues)." Engineer Botnick is to be especially commended for the unlayered, natural guitar sound captured on this track.

Transformation is a very impressive release.

H.G.L.

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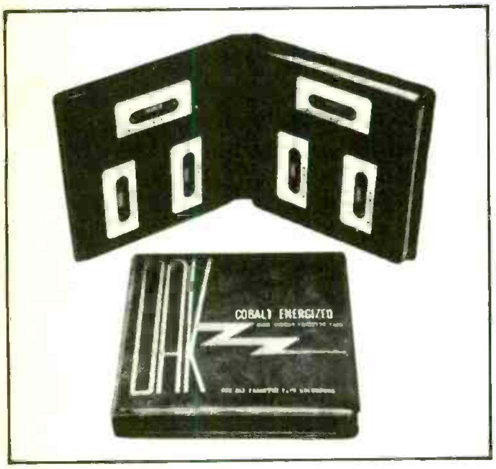

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The Duke's Realm and Evans's Lyrical Sculpting

By Nat Hentoff

Duke Ellington did indeed have his own realm. The title was more than an honorific one. No orchestra in all of jazz history was so continually and completely distinctive as his, and this was true in just about every respect. Duke, for instance, invented no radically new chordal language; but his way of shaping and blending harmonies resulted in a much richer, more evocative, and more *mysterious* sound-ambience than that of any other jazz band. And so it was with the rhythms of Ellingtonia. Other orchestras swung hard or crisply or (as in the case of the Basie band of the '30's and early '40's) actually floated. But Duke's time was different from all the rest—more subtle, more insinuating, more protean in that the collective pulsation he devised kept changing according to the need of the piece. Just plain swinging all night long would have been much too boring for Duke. And then, of course, there was Ellington's melodic conception—as myriad as his experiences and fantasies. No jazz composer has so far come anywhere close to the scope of Ellington's melodic imagination.

All these qualities are evident in the newly released transformations of travel reminiscences, *The Afro- Eurasian Eclipse*, recorded in 1971 when Ellington still had a substantial nucleus of key players, from Cootie Williams to Harry Carney. The recording is well-balanced, though not as vibrant as I would have liked. However, it is such a delight to come to a new Ellington find that I shan't cavil. There are few experiences in jazz as overwhelming as the majestic surge of an Ellington orchestra, with the composer leading the charge from the piano. Duke's was music of a separate jazz microcosm, a terrain of utterly singular colors and contours, a land now forever of the past.

On a much smaller scale than Duke, Bill Evans has also created and sustained a singular jazz space, particularly since he has devoted his years to leading his own trios rather than working intermittently as a prestigious sideman (as he did earlier in his career). Evans is quintessentially a lyrical improviser, but his is a rigorous lyricism based on unusually thoughtful harmonic designs and the kind of melodic sculpting which, while certainly spontaneous, is so ineluctably logical that, in retrospect, his best work sounds as if it had been carefully scored. In sum, Evans is a good deal more of a perfectionist than many jazz pianists, and that accounts for some of the intensely searching quality which characterizes his playing. That is, his work is a nearly equal fusion of both visceral and intellectual drives and so it simultaneously rewards both such needs of the listener.

Since We Met, a superior illumination of the jazz territory Evans has claimed, was recorded at New York's Village Vanguard in 1974. I happened to have been there on one of the recording nights and was grumbling that drummer Marty Morell's heaviness of attack was bludgeoning the balance between him, Evans, and bassist Eddie Gomez. In the recording van outside the club, I ventured that opinion and was told not to worry. And sure enough, through some rather ingenious mixing, Morell sounds here as he should have that night. Is the recording, therefore, faithful to the "live" experience? An irrelevant question, Pierre Boulez would say, for a recording is *not* a "live" experience even when it is *of* a "live" experience. And that's true.

DUKE ELLINGTON: *The Afro-Eurasian Eclipse.* [Duke Ellington, producer; Roger Rhodes, engineer; recorded at National Recording Studio, New York, N.Y.] Fantasy F-9498.

THE BILL EVANS TRIO: *Since We Met.* [Helen Keane and Orrin Keepnews, producers; Michael De Lugg, engineer; recorded in performance at the Village Vanguard, New York, N.Y.] Fantasy F-9501.

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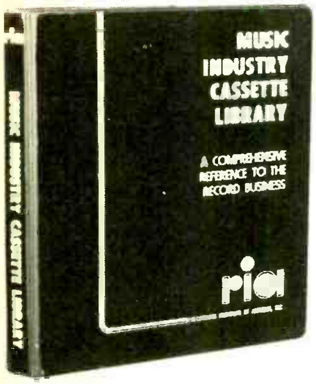

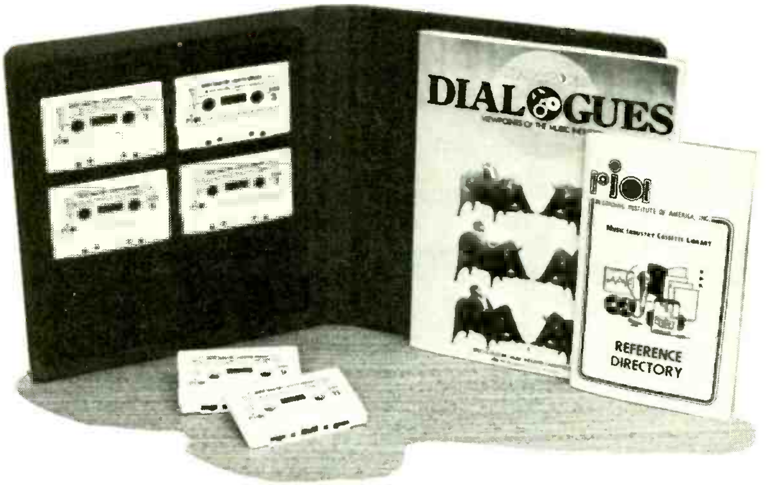

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CLASSICAL

DEBUSSY: Piano Music. Aldo Ciccolini, pianist. [Recorded in France.] Seraphim S-60253.

Performance: **Fresh and alive**
Recording: **Good**

Bravo for Aldo Ciccolini! He has recently recorded a disc of Debussy piano music that should send everyone scurrying to buy it. Who is Aldo Ciccolini you say? No, it's not a pseudonym for Chico Marx. This talented pianist's most notable achievement on records to date has been six volumes of Satie piano music. Except for recent releases of Rossini and Granados piano works, his output is almost exclusively French music.

All his previous recordings show skill and sensitivity, but now with Debussy he has really come into his own. Of all French composers, Debussy is best known and perhaps deservedly so. In recording Satie or Rossini there is little competition, but Walter Gieseking set the standard for Debussy nearly forty years ago. Most subsequent versions of this music pale in comparison. Now Ciccolini emerges as a worthy successor to Gieseking. Even such standards as *Clair de Lune* and *Reverie* are fresh and alive here. My personal favorite is *Danse (Tarantelle styrienne),* a piece largely neglected in its piano version.

At the budget price of $3.98 (most stores sell it for less) this disc is irresistible. One can only hope this recording will sell so that we may get more Debussy from Mr. Ciccolini. H.R.

RAVEL: *Gaspard de la Nuit*; *Sonatine*; *Valses nobles et sentimentales.* Martha Argerich, piano. [Rainer Brock, producer; Heinz Wildhagen, engineer.] DG 2530 540.

Performance: **Breathtaking!**
Recording: **Vivid**

This extraordinary disc contains the finest playing of Ravel piano music I have ever heard. Argerich's control of rhythm, dynamics and expression is simply breathtaking! One looks at the score in astonishment as she flawlessly executes hairpin dynamics that other pianists dare not attempt for fear of rupturing the crucial ongoing flow of Ravel's devilishly difficult writing.

Argerich tackles *Gaspard* as only the greatest virtuosos could. Her technique is impeccable, as is necessary, but she also invests the work with a depth and expression I hear in no other recording. *Le gibet*, the macabre second movement which

MARTHA ARGERICH
Ravel interpreter extraordinaire

portrays a gallows corpse swaying quietly in the wind as a bell tolls monotonously in the distance, is taken at an achingly slow but beautifully sustained tempo.

The overside *Sonatine* and *Valses nobles et sentimentales* share many of the same attributes as *Gaspard*, although some of the *Valses* are a bit fussy rhythmically. From the evidence here, Martha Argerich should be recording a complete Ravel cycle. Let us hope that DG agrees. S.C.

ROSSINI: Overtures. Academy of St. Martin-in-the-Fields, Neville Marriner cond. [Vittorio Negri, producer; recorded at Wembley, London.] Philips 6500.878.

Performance: **Delightful**
Recording: **Clean and realistic**

This is a delightful record. Go out and buy it at once! The joy of hearing eight Rossini overtures played by this small orchestra with such zip and elan, and recorded by the Philips engineers with such realistic presence and depth, should not be missed.

Of conductors using full symphonic forces, Toscanini, Szell and, in some instances, Bernstein, should not be en-

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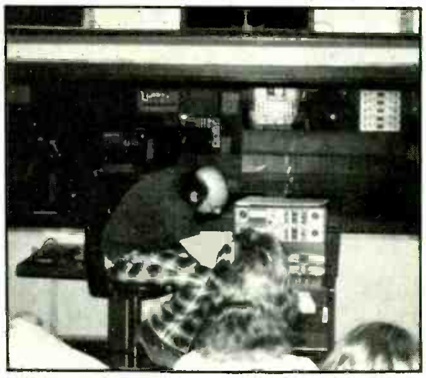

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tirely disregarded, but Marriner's reduced orchestra makes much more sense, is more authentic (if one cares about these things) and allows the winds their proper balance. Marriner obliges with interpretations which yield nothing in point, clarity and wit to his more experienced colleagues.

The overtures on this disc are to *Il barbiere di Siviglia, L'Italiana in Algeri, La cambiale di matrimonio, La scala di seta, Tancredi, Il Signor Bruschino, Il Turco in Italia* and *L'inganno felice.* There are rumors that another disc is on its way. The sooner, the better! S.C.

SHOWS and SOUNDTRACKS

MAURICE JARRE: *The Man Who Would Be King.* National Philharmonic Orchestra, Maurice Jarre cond. [Carl Prager, producer; Eric Tomlinson, engineer.] Capitol SW-11474.

Performance: **Good for background**
Recording: **Appropriate**

One of the problems with soundtrack recordings is that even though you saw the movie, you're really buying a pig in a poke. Most film music is written after the picture is finished or at least in "rough cut" stages. Hence, the music is conceived primarily to support the footage it accompanies. Any intrinsic value outside of the film's context is pure gravy.

So what usually happens is this: we view a film; two or three themes in the score affect us in connection with the film; we decide we like the music. Then we obtain the recording and what do we find? The two or three themes we like are repeated four or five times each on the record with slight variation in arrangement, tonality or rhythm. Then there is still half a disc of music we've never even noticed before. Why? Well, . . . some of it is extra; maybe a minute and a half of drum beats or bugle blats were needed for the film itself, but the composer generously donates five minutes of it for the recording. The rest is music that was very quietly underscoring the movie—heard full blast on the disc.

The recent soundtrack from *The Man Who Would Be King* illustrates this perfectly. The main theme (the standard British barracks song "Minstrel Boy") is repeated at least four times on the album. There are one or two marching themes of moderate interest and half a disc of pseudo- Indian music which is only good for background. Is this a bad movie score? No. For the film it is both functional and effective. Is this something anyone should buy? Equally, no. The film is very good, but without it the music is virtually devoid of interest. Save your money. H.R.

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