AUDIO

DECEMBER/1964
60¢

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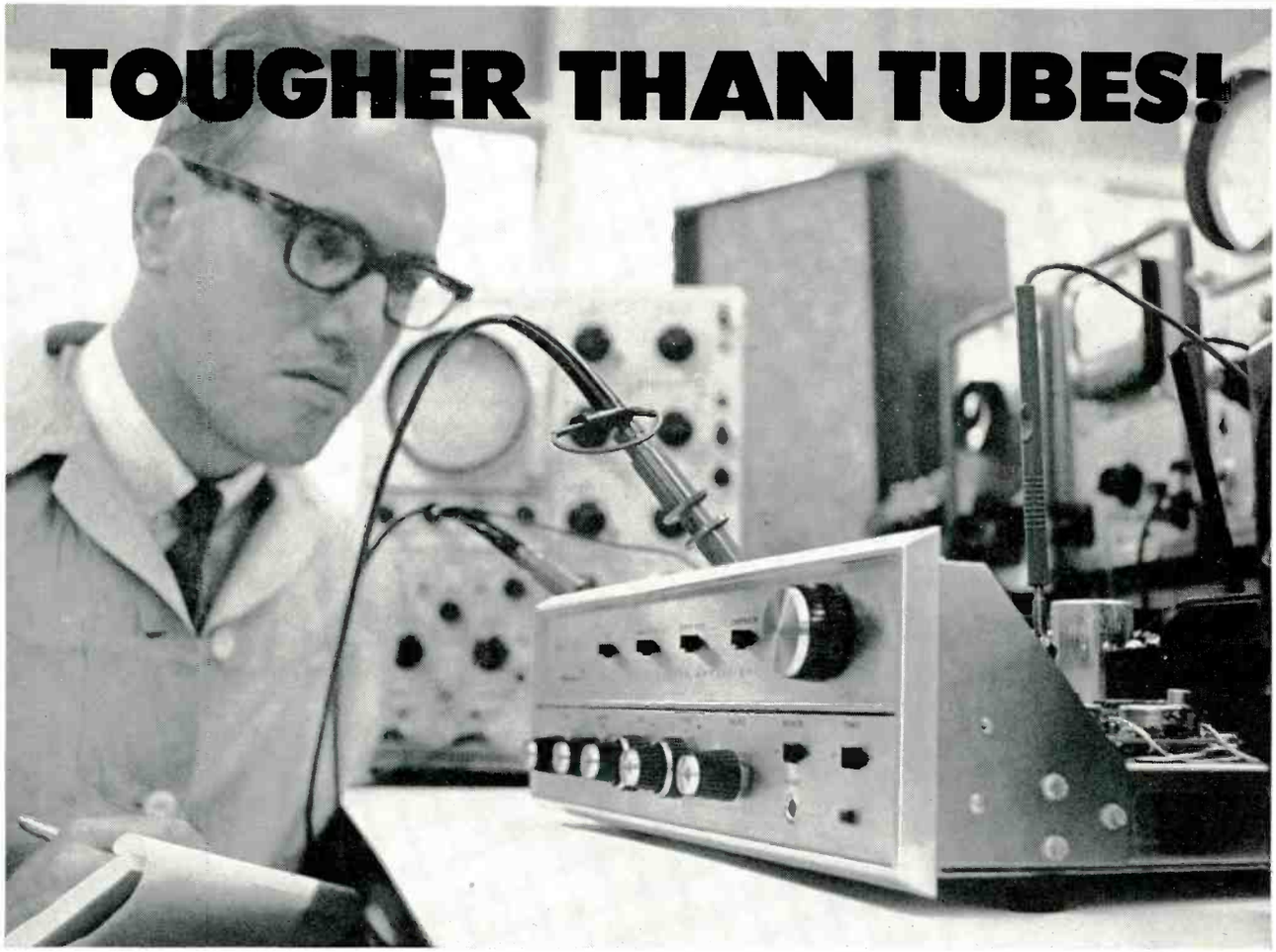

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Now you can own a powerful 80 watt solid state amplifier constructed to standards unique in the high fidelity industry. The new Scott 260 uses rugged pre-tested heavy-duty components, including massive heat sinks, heavy printed circuit boards and new silicon output transistors. Critical electrolytics are hand selected and have operating capabilities far exceeding circuit requirements.

To insure the thoroughness of its quality control procedures, H. H. Scott called in transistor specialists with many years experience in the design of critical military components. Rugged tests were devised to subject the amplifier to conditions far more severe than encountered in normal use. These "torture tests" include: Applying a "step-stress-test" to a selected sample of all components used, simulating hundreds of hours of normal operating life and showing up any components that might fail; applying a unique "surge and cycle" test, normally performed only on rugged military equipment, to simulate stresses the amplifier may be subjected to under the most severe home conditions; elaborate pre-test and checkout of all components, including transistors, to insure that components will not fail in service.

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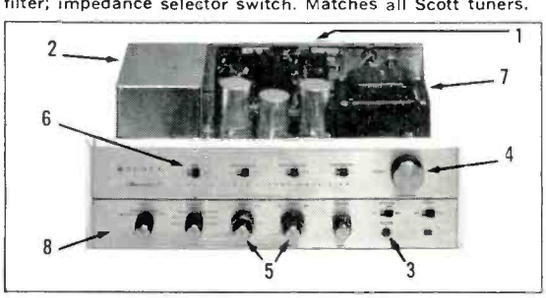

(1) Rugged silicon direct-coupled transistor output stage (2) Preamps on separate modular-type printed circuit boards (3) Stereo headset output (4) Master volume control (5) Separate bass and treble controls (6) Complete tape recorder input and output facilities (7) Massive power supply provides high power surges when music demands it (8) Handsome styling matches Scott Transistor Tuner model 312

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AUDIO

DEC., 1964 VOL. 48, No. 12

Successor to RADIO, Est. 1917

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Coming

Construction

• Matrixing Amplifier for Two-Channel Stereo Signals. Wayne B. Denny. A matrixing amplifier for altering the "stereo spread" of two-track tapes recorded at a performance of an amateur group.

Sound Reinforcement

• A Basic Course in Commercial Sound. Norman Crowhurst. Chapter 10. Final chapter of this series, tying up all loose ends and reviewing the previous chapters.

• Two Engineered Temporary Sound Systems. David S. Klepper. The Billy Graham Greater Boston Crusade required temporary systems, both indoors and out. Here's how it was done.

Profiles

• JFD FM-Stereo Antenna, Model LPL-10
• JBL Stereo Control Center, Model SG520
• Heathkit Solid-State AM-FM-Stereo Tuner, Model AJ43

In the January Issue

On the newsstands, at your favorite audio dealer's, or in your own mailbox.

AUDIO CLINIC

Joseph Giovanelli

Send questions to:
Joseph Giovanelli
2819 Newkirk Ave.
Brooklyn 26, N. Y.
Include stamped, self-addressed envelope.

Overloaded FM Tuner

Q. I have a monophonic tuner of fairly recent origin and a monophonic amplifier. My antenna consists of two folded dipoles at right angles to each other, connected in parallel and located in the attic of my one-story house.

Within a radius of five miles are the transmitters of at least two TV stations and three or four other AM and FM stations. I have no interest in hearing most of them.

My problem is that on several points on the dial I receive two stations simultaneously, and at least two points I receive three, usually with considerable distortion and odd noises. WASH-FM is the chief offender. It is located less than a mile away from me. It happens that one of the stations I would like to receive is WBAL-FM Baltimore, about 40 miles away, but at that location on the dial all I can get is WASH, with an admixture of some other station I cannot identify.

I might mention that a line to Baltimore on the map would pass approximately through the WASH transmitter tower.

In the past year, the tuner has been aligned three times, the last time by the manufacturer who replaced a tube or two and installed some new front-end stuff to bring up the tuner's performance to the level of their current models. Their treatment has improved matters—I used to get WRFL, Winchester, Va. instead of interstation noise over the band from 100 mc down to 88 mc, WGMS used to interfere at more places than it does now, and I used to get three stations at a time at more places than I do now, but whereas I used to get WITH-FM Baltimore when the tuner was new, now I can't get it at all, and whereas before I was able to get WBAL faintly and with distortion, along with a lot of WASH, I cannot find it at all any more.

Until 6 months or so ago, I accepted the nonselectivity and failure to reject these local signals as unavoidable. (I remember one of the claimed advantages of FM over AM as it was trumpeted some years ago was that FM had made it impossible for the FM tuner to pick up more than one station at a time. That's beside the point.) Aware as I was of the proximity of powerful transmitters which make hash of AM reception here, I more or less lived with the situation. Then I talked with a neighbor, telling him of my troubles. He was much surprised, and told me that his old-model FM tuner at his home a few blocks from the WASH transmitter, had no trouble (with a pair of rabbit ears) in picking up WBAL.

What should I do? Stanley Metalits, Kensington, Maryland.

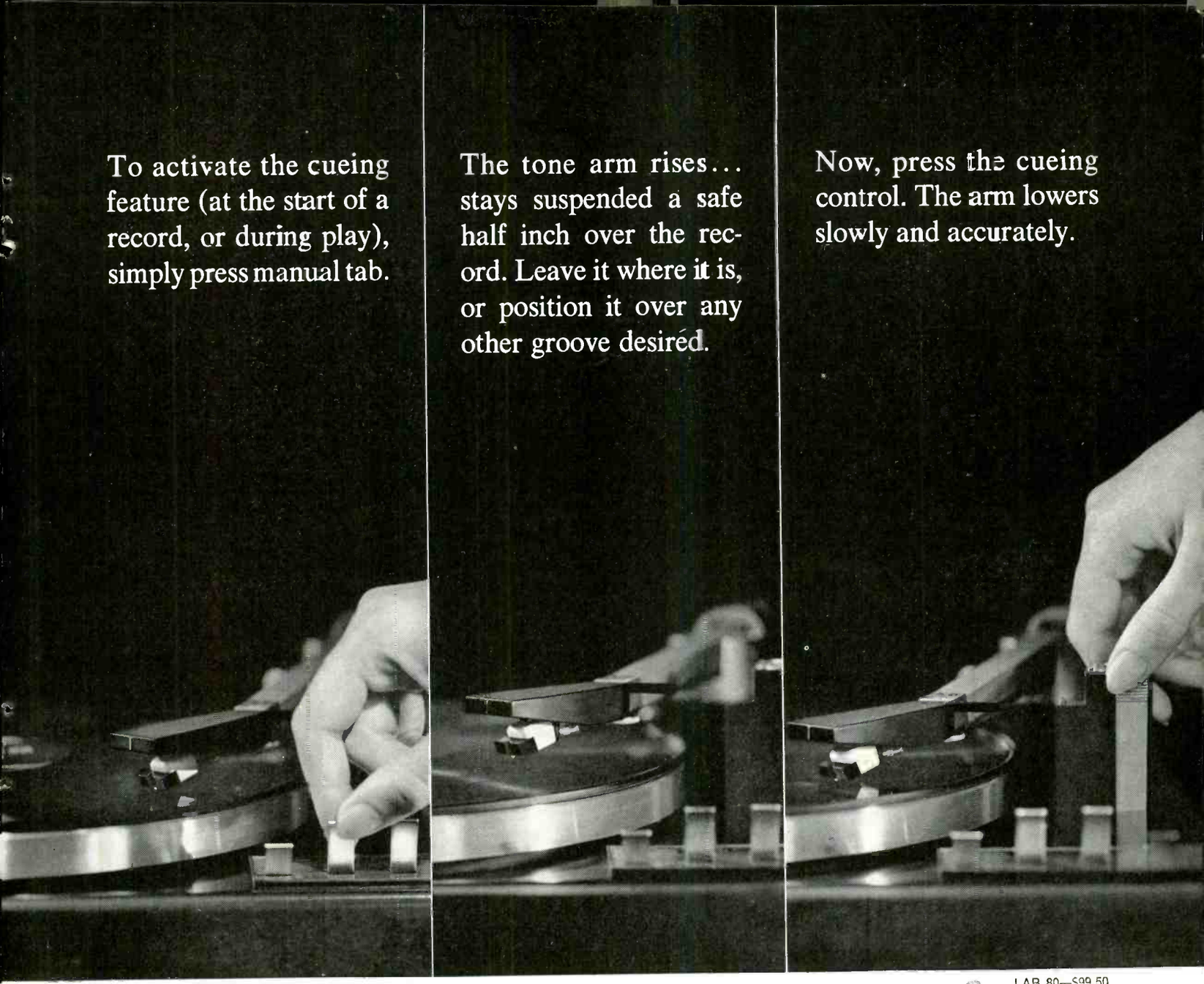
A. As time goes on, tuners have improved remarkably in sensitivity to weak signals. This is not without price however. The more sensitive a front end becomes, the more susceptible it becomes to overloading in the vicinity of strong signals. Once the front end is overloaded, all the rules about capture ratio go out the window because the signal feeding the detector is a composite.

To explain this better, assume that a weak signal is passing into the front end of your tuner. It will get into the i. f.'s in the customary manner but because the front end is saturated with the strong signals around it, there is a mixing process taking place in the front end which results in the output being made up of not one signal but a combination. The signal passing into the i.f. amplifier is made up of the weak signal, influenced in character by a signal possibly half the dial scale away. The i.f. amplifier and detector circuit do not know this; they are fooled into thinking that there is just one signal.

This is not all that takes place when two signals on the same or adjacent channels are passed into the detector, assuming no front-end overload. To make this simple, two signals still have separate identities, and the stronger one will be separated out by the detector and presented to the audio stage free from interference. The full story of capture ratio is, unfortunately, too detailed to go into in this context.

Probably the reason your tuner works

(Continued on page 66)



LAB 80—$99.50

Integral tone arm cueing is featured in Garrard's new Lab 80, the first Automatic Transcription Turntable!

Here's how it works:

To play a single record: Press the manual tab. This starts the motor and activates the cueing feature. Then, position the tone arm above the first (or any) groove. The arm is suspended a safe half inch above the record. Now, press the cueing control and the stylus lowers slowly into the groove.

To cue a record during manual or automatic play: Press the manual tab. The arm rises and suspends a half inch over the record. Move the arm to the band or groove desired, and press the cueing control. The stylus lowers gently into the groove.

To pause during manual or automatic play: Press the manual tab. The arm rises directly over the record groove. The turntable continues to revolve. When you are ready, press the cueing control. The stylus descends safely. The music continues from where you left off.

As tracking forces have become lighter, and stylus assemblies more delicate, so has the danger of damage from manual handling increased. Now, in the Lab 80, Garrard has incorporated the convenience and safety of this ingenious cueing control . . . built into an automatic unit for the first time. This, and other advanced features of the Lab 80 are explained in Garrard's new 32 page Comparator Guide covering the entire line. For a complimentary copy, write Dept. GX-14 Port Washington, N.Y.

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LETTERS

Stereo Enhancement—Channel A

SIR:

I was surprised to read Mr. Canby's article in the October issue in which he rather heartily endorses that strange process called "stereo enhancement." This process purports to revitalize a mono recording so that it offers "improved" sound when played on stereo systems.

The author makes several statements that would not seem to bear up under close scrutiny. While it is true that identical signals in each speaker may seem to emanate from a point midway between them, this is true only when one is listening from a point equidistant from both speakers. In his article, Mr. Canby implies that all listening is done from this mid-point. Actually, this is rarely the case. Most people are closer to one speaker than the other in many situations. Thus, that "squashed-up sound" he refers to is not always present.

He also says that the layout of the classical orchestra can almost be heard on the enhanced recording (violins on the left, the bass to the right). But what about the poor brass and woodwinds that are often seated in the center? What about the unearthly sound of the human voice whose high tones are heard in one speaker while the fundamentals are coming from the other, 8-ft. away?

Finally, the question arises as to what is more natural, the spread of sound, or its balance and unity upon reaching the ear? Often, the enhanced recordings make the speakers sound out of phase. And show me the mature listener who would prefer the disembodied sound of out-of-phase music merely to achieve some sort of unnatural spread! Perhaps in our preoccupation with this newest advance in audio (and I refer to real stereo, not "enhancement") we fail to hear the music through the "spread."

ANASTASIO A. ROSSI
Coordinator of Music
Mount Pleasant Central School
District #1
Thornwood, New York

Stereo Enhancement—Channel B

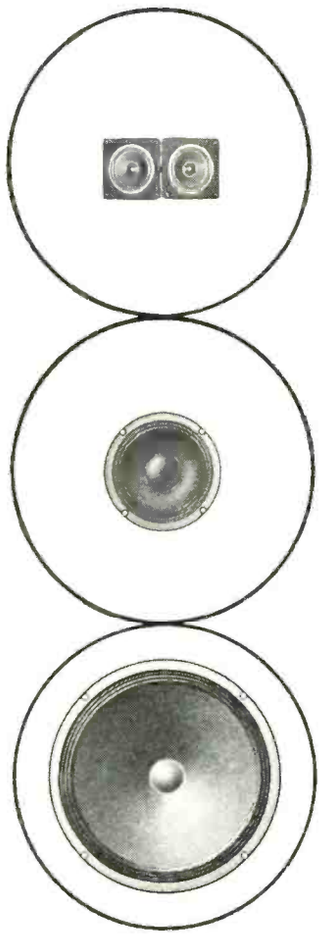
SIR:

In answer to Mr. Rossi, I would say first that the "bunched up" sound is not to be got rid of so easily—it follows you wherever you may sit. Makes no difference at all; the bunching merely moves from side to side as you do (or as you turn your balance control). For my ear it is unpleasant in any position it may seem to take.

Further, I think he is making an old semantic mistake, the all-or-nothing. His criticisms of "enhanced" sound are real; there are genuine dangers. But like so many artistic matters, this one involves a balance between desirables and undesirables and, more important, it invites ingenuity of a constructive, inventive sort. The question isn't "Is 'enhancement' (all) good or (all) bad" but "what can we learn to do with this potentially useful new technique?"

Finally, Mr. Rossi in his last paragraph almost makes the mistake I avoided, the comparison between real stereo and "enhancement." It isn't a substitute for stereo, this process, but a substitute for mono. What is important, I think, is to see what *is being done* by those who are experimenting with it. Not what ought or ought not to be done.

E. T. CANBY



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LIGHT LISTENING

Chester Santon

Disc and tape reviewers are not immune to impressions of recorded sound when they visit a major annual High Fidelity Show. The 1964 New York event left me with a few thoughts pertinent enough to be included in this month's column. As the man who sets up my copy in type will recall, I was somewhat taken with the cleancut job done on records by transistor amplifiers and preamplifiers at the 1963 Hi Fi Show. Possibly because of exposure to equipment then limited pretty much to the top price range, I came away from the '63 event with a very favorable impression of solid-state gear. After listening to records and tapes at the '64 show on transitsor equipment offered in a wider range of prices, I was struck by a certain sameness in the sound. So much of the recorded material had a similarity in the basic nature of its sound, I'm beginning to wonder if transistor gear subtracts some of the characteristic, color-producing quirks of musical instruments in attaining its undeniably clean sound. The question, for all its subjective nature, would be easier to answer if the recording industry could provide us with stereo discs whose average in audio quality is more in keeping with the specs already attained by solid-state equipment. As matters stand now, I spend part of my listening time still wondering why some records, even on major labels, make my top-notch tube equipment sound only a trifle better in frequency range than my neighbor's five year old console.

Alec Guinness in Dylan (Original Broadway Cast)

Columbia Stereo Tape DOQ 666

Columbia Records has joined the other companies in offering the spoken word on 4-track stereo tape at 3¾ IPS. The slower speed is no longer a novelty on commercial releases. London Records established 3¾ as a perfectly legitimate medium for dramatic presentations on its lengthy series of Shakespeare plays recorded by the Marlowe Dramatic Society. As my listening time has permitted—a good part of an evening can be spent with one of these reels—I've been sampling some of the Shakespeare tapes at 3¾, finding them more than suitable for true reproduction of the human voice in a wide range of stage emotions. Since no stretch of the imagination can quite put a group of Shakespeare plays in the light listening category, I haven't reviewed any drama on 3¾ until this appearance of Columbia's recording of "Dylan," starring the great Alec Guiness in the role of the roistering Welsh poet. It's heartening to discover that more than one label can get very good results on speech in commercial releases at 3¾. It hasn't been too difficult to demonstrate acceptable frequency response at tape speeds of 3¾ and less at press showings of systems carried over from the lab on a soft pillow. Making 3¾ work under random sampling of an assembly line's output has been something else again.

Heard with only one interruption in the tape version, the twenty scenes of the Sidney Michaels' play take on their natural momentum as Guinness creates a Dylan Thomas a bit more convincing to the ear than his stage makeup was to the eye. Stereo effects throughout the play are just as realistic as any I've heard on 7½ tapes. Columbia's listing of the reel's playing time is somewhat confusing. The box claims three and one half hours playing time for this new series of plays recorded at 3¾ IPS. This is true enough in the case of a companion tape release of Richard Burton's "Hamlet." Once the "Dylan" tape box is opened, the folder inside lists the reel at the more accurate figure of two and a half hours. With a price tag no higher than that of the LP album counterparts, Columbia's tape release of "Dylan," "Hamlet" and "Who's Afraid of Virginia Woolf?" represents a darn good buy for Broadway-minded tape fans.

Let's Ring Bells Around the Christmas Tree

RCA Victor LSP 2914

RCA has again called upon carillon virtuoso, John Klein, in planning an unusual Yuletide release. Here he performs on location at the console of what is claimed to be the world's largest modern carillon installed at the Coca-Cola pavilion at the New York World's Fair. The instrument is an electro-mechanical device built by Schulmerich, Inc., of Sellersville, Penn. This latest creation, similar to carillons built by the firm for the 1958 World's Fair in Brussels and the 1962 Seattle World's Fair has a total of 610 bells. The bronze bells, tiny in size when related to those used in conventional carillons, produce tones barely audible to the human ear. Modern high fidelity equipment amplifies the sound and, in the case of the Coca-Cola installation, feeds it to an array of fifty-seven directional loudspeakers mounted on the 120-foot tower. Approximately 3,000 watts of audio power are brought into play when the system is operated in the mode that spreads sounds to the entire backgrounds. In producing this album, RCA Victor recorded the Henri Rene orchestra and chorus at Webster Hall in Manhattan before taking the tape to Flushing Meadow where Klein added the sound of bells to a collection of traditional carols and Christmas songs. The blend, handled with great fitness by engineer Ed Begley, makes the carillon a natural member of a rich-sounding cast.

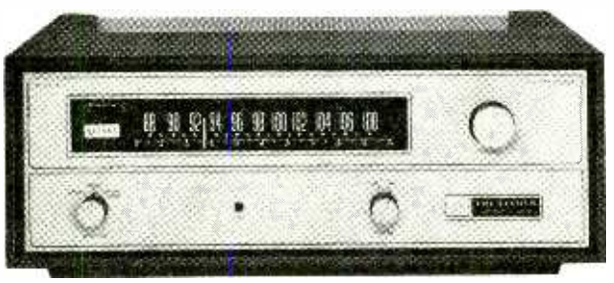
Garbo

M-G-M E 4201 P

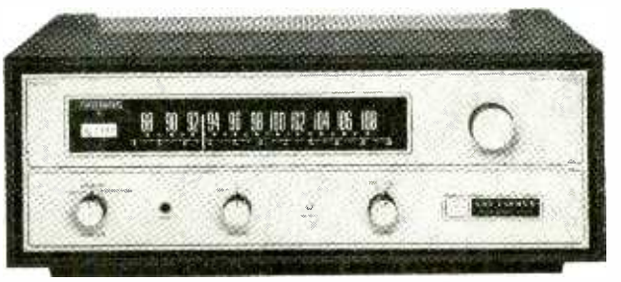
Every now and then a recording comes along to remind us how much a voice can convey within a limited frequency range. Here is a succession of scenes from famous MGM motion pictures starring the immortal Greta Garbo. During the decade when these films were made—1930 to 1939—sound recording was still a bit of a novelty at Hollywood studios. For that matter, a few men now prominent in the high fidelity industry were probably still attending high school. When Garbo made her first appearance in a talking picture (Anna Christie in 1930) MGM's sound technicians had to content themselves with a frequency response that sounds pretty primitive by today's LP standards. Yet the essence of Garbo as most of us remember her comes through in the nine different sound tracks selected for this recording. If you haven't seen any of her films in their recent revivals, listening to this record can be an uncanny experience. Far more than any photograph, the sound of Garbo pronouncing a given word in her unique way can throw one right back into the Thirties. A heartening by-product of this collection of scenes is the realization that we no longer have to endure the fancy scenarios and stilted leading men that once were commonplace in the movies. Æ

Before multiplex came along, you could get by with an average tuner.

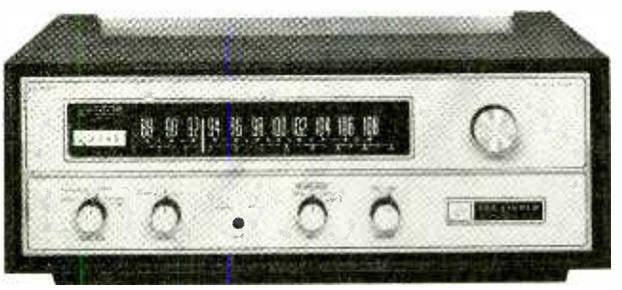
Now you need a Fisher.



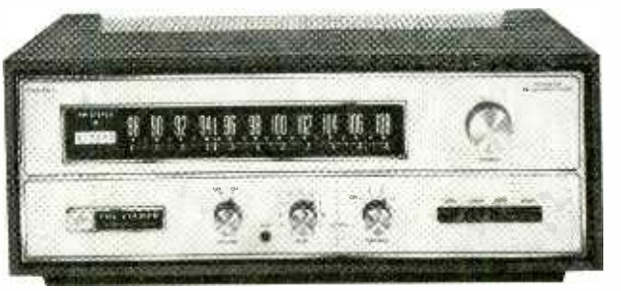
The Fisher FM-90-B
FM-Multiplex Stereo Tuner
with STEREOSCAN†, GOLDEN SYNCHRODE† front end, 3 IF stages, 2 limiters, 2 μv sensitivity; $179.50*



The Fisher FM-100-C
FM-Multiplex Stereo Tuner
with STEREO BEACON†, *Nuvistor*-GOLDEN SYNCHRODE† front end, 4 IF stages, 3 limiters, solid-state multiplex, 1.8 μv sensitivity; $249.50*



The Fisher FM-200-C
FM-Multiplex Stereo Tuner
with STEREO BEACON†, *Nuvistor*-GOLDEN SYNCHRODE† front end, 5 IF stages, 4 limiters, 1.8 μv sensitivity, solid-state multiplex, AUTOSCAN† automatic stereo scanner, MICROTUNE® tuning; $299.50*



The Fisher TFM-300
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In the good old days of strictly monophonic FM, any reasonably well-engineered tuner brought in at least your local stations with listenable fidelity. Of course, a Fisher tuner still made quite a difference in sound quality; but it was a difference in degree, not in kind. Multiplex has changed all that.

Even though FM-stereo has potentially much greater sonic realism than FM-mono, a multiplex broadcast can actually sound badly distorted unless received through an absolutely first-rate tuner. And to hear a stereo program *exactly* as it was monitored in the FM station's control room requires the sensitivity, wide-band design, low distortion and unequaled channel separation of a Fisher multiplex tuner. Nothing less will do; the medium itself has become more demanding.

The seven stereo tuners currently made by Fisher feature ultrasensitive front ends, three to six wide-band IF stages, two to five wide-band limiters, wide-band ratio detectors, multiplex circuitry of the superior time-division type, and — in six models—automatic mono-stereo switching via the Fisher STEREO BEACON†. The IHF sensitivity of all models is in the range from 2 to 1.5 microvolts; FM-stereo separation at 1 kc ranges from 35 to 40-plus db. Prices start at $179.50, less cabinet.

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JAZZ and all that

Bertram Stanleigh

In March, 1940 Woody Guthrie made a series of more than 30 discs for the Library of Congress. The recordings, made at the Department of the Interior recording studio in Washington on an old Presto disc recorder, consist of an interview between Woody and Alan Lomax, interspersed with a substantial number of folk songs. It was originally thought that these discs would be the basis for a radio program, but the plan was never realized, and the discs remained in the Library's archives for nearly a quarter of a century. Now, thanks to Elektra Records, these recordings have become available to the public on three long-playing discs. The process from the original recordings to the finished LP's was a long one involving months of painstaking effort.

After surmounting a number of legal obstacles, work began in Washington, where Robert B. Carneal, chief engineer at the Library of Congress recording laboratory transferred the original disc recordings to 15 ips tape. Recorded at 33⅓ rpm on 12-in. acetates at 110 lines per inch, they had about eight minutes on each side and had been cut up to a 3-in. center diameter. The original lacquers had been stored in old metal containers, and they were scratched, dusty and had been played a number of times with the kind of playback equipment that was generally available in 1940. Several different types of blank discs had been used in cutting the original records, and consequently there was a difference in thickness that resulted in varying groove depth and width. Before transcribing each disc, it was carefully washed, and Carneal checked the groove dimensions with a binocular microscope. From his collection of more than a hundred different playback styli, he was able to select the one that matched each groove contour most closely, and the sound was rebalanced in transfer, utilizing a Klein & Hummel EQ 1000 equalizer.

Fig. 2. Woody Guthrie.

In New York, a 7½ ips copy was made of the tape transfer, and work began on putting three days of recording into shape as a three-hour commercial recording. Many of the songs had been re-

(Continued on page 74)

Fig. 1. Jac Holzman (left) and Sid Feldmen cutting the LP masters for the Woody Guthrie/Library of Congress Recordings.

Can you recognize these three great stars?

The first two are obvious to movie goers: Rock Hudson and Doris Day starring in Universal's new comedy, "Send Me No Flowers." The third is ours and it's on the table. It's the Incomparable Concertone 800. This unique stereo tape recorder plays a vital role in this hilarious new movie. When the script called for a tape recorder, Universal Pictures wanted the finest, most modern machine available. That's why they selected the Concertone 800 to appear. It's as modern and sophisticated as the live stars of the movie. □ The Concertone 800 is the only stereo tape recorder with double Reverse-o-matic® and six heads—allowing you to record and play continuously without reel turnover. The 800 (portable or tape deck) is priced realistically and provides an exclusive combination of features. It starts as low as $379.95. Send for a free Concertone 800 brochure and the name of your nearest dealer. Write to Concertone, Box 3162, South El Monte, Calif.

CONCERTONE

ROCK HUDSON and DORIS DAY star in "SEND ME NO FLOWERS"

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AUDIO ETC.

Edward Tatnall Canby

THE BIG SHOW

1. Velvety, Soothing, Relaxing

Maybe it'll seem late and after-the-fact to comment at this stage upon the autumn hi fi shows when winter is already hard upon us. But deadlines being facts of life, it must be (it always has been) now or never. The fall merchandise, after all, was labeled "1965" and there are still a few weeks to go until that halcyon time arrives. So herewith my late '64 comments.

My show, of course, was the New York International High Fidelity Music Show, if I remember the fancy title correctly. Others weren't very different, I expect. As for this New York show for 1964—er, 1965—I really enjoyed every moment of it. Was I surprised.

Much as we love our hi fi and the very idea of componentry and home musicry (to invent a ghastly term), many of us reporters do privately groan each year in anticipation of the big annual noise makers. Yet we're always there, you may be sure, because we can't afford not to. The shows are too important for us. The public may learn a great deal—we learn much more. We see Everybody who is Anybody in the flesh, of course, which is worthwhile. And we see every product that Anybody of Any Importance is making or about to launch, which is even more important. If not in the flesh, then in metal, glass and plastic. And last but NOT least, we hear. We hear just about everything, or the output of everything—the throughput, if you wish—in terms of of loudspeaker and earphone sound.

We hear with unusual efficency, too; for as everybody knows, there are seldom less than three audible signals available to the ears at any geographic point in a hi fi show and sometimes five or six. To adjust relative volume, one need merely walk a few feet one way or the other. Easy.

Yes, even so, I *did* enjoy this year's show. Perhaps it was the hoped-for result of my personal audio drought, my "sabbatical" of the previous months; it was a kind of coming-back party for me. But I think there was more to it than that. What struck me—and I'm not yet going deaf—was that the big show wasn't noisy.

Definitely, the noise level was not only more circumspect in the large average this year; it was better controlled, better channelled into usefulness, with better "separation" than I ever remember from previous year's. So it was in New York, anyhow.

Am I dreaming? I don't think so! Very seldom, this year, did I hear that familiar horrid, hollow, bumping and thumping that betokened a mammoth loudspeaker on the loose, through the wall in the next booth. Last year, many a good exhibit was ruined for me, as for others, by that sort of dog-eat-dog competition. Maybe the knife twisted both ways last year—anyhow, it was good to feel this year that mutual respect had somehow managed to rise a bit and/or big noise no longer seemed to be the selling point it once was. More power to you, gents—I mean *less* power. . .

The result of this unaccustomed kindness toward us dumb animals was, as I sensed it, that a great many more listening rooms presented a *good* case for their enclosed products. A large number of exhibitors were able to run remarkably quiet demonstrations without hopeless interference, their volume levels down to a pleasurable conversational range. (Perhaps this was significant—in ye olde hi fi days constructive conversations in most show rooms were impossible. No sale—couldn't hear yourself think.) People came in and made relatively unhurried evaluations of what they saw and heard; they talked, to each other and to the people in attendance. They could even sit down—another good sign. A lot of the booths featured comfy chairs and they were always full.

The still-lively interest in earphone listening lent a novel touch to all this relative quiet. (Relative, mind you. I wouldn't have called it exactly the silence of the tomb.) Imagine walking into a display to find dozens of people, earphones on heads, lost in silence, living in a private shared world of their own: simultaneous smiles, frowns, beat-tapping—and the newly entered person (me, for instance) could scarcely wait out of curiosity to hear what they were all smiling, or frowning about.

That was one area where signal interference was absolutely at zero. Collectively, the intra-earphone territory at the show must have added up to quite a sizeable plot, maybe a quarter acre.

My favorite memory, though, was the Most Significant Booth of All, occupied by that excellent maker of high quality throughput equipment (amplifiers, preamps, et al)—McIntosh. On the day I visited McIntosh, at least, there was no reproduced sound at all in his exhibit. Not a trace. Was I impressed, especially since the place was absolutely jammed with people. Must prove something or other. Was it the all-visual appeal of the lovely McIntosh instruments? Or that velvety, soothing, relaxing SILENCE?

2. The Sound-Alikes

What was Most Significant at the fall hi fi show—other than the improved signal-to-noise factor?

Well, I'll skip transistors because everybody knows *they're* significant and getting better and better, etc etc. The thing that struck me above all else (even above the astonishing number of home tape recorders in evidence) was simply the sound of a thousand loudspeakers. For this year the sound was not only less loud but also significantly DIFFERENT. It was the year of the sound-alikes.

In fact, I've just invented a new club, the Sound-Alike Club, for speaker manufacturers, and I've appointed a whole slew of charter members, whose names are strictly restricted to my own private thoughts. A lot of the sound-alike speakers were brand new; but they joined a growing coterie of loudspeakers already familiar to us, to make an increasingly large and unmistakeably important element in the listening spectrum—at the hi fi show and in a million dealers' show rooms too.

An old thesis of mine, as you'll remember, is the platitude that if all loudspeakers were 100 per cent hi fi they would all sound exactly alike. For we do have an ideal, after all, however unattainable in practice; and that ideal allows for no tolerance whatsoever in the faithful reproduction of hopefully idealistic sound.

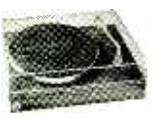
In practice, we live with a great deal of tolerance, complicated by factors of taste and preference which are important to all of us. Nevertheless, we all continue to huff and puff, year by year, in our gradual approach to that ever-shining ideal of absolute fi, in our speakers as in other elements of the whole hi fi kit and caboodle.

And so the new and larger group of sound-alike speakers, of every shape and size, in every price range imaginable, tended to bear out my thoughts of last year that there was a strong chance speaker sound would be getting better pretty soon. It is getting better, in all areas, top to bottom. That is, it's getting more uniform.

If you think you can't afford the best high fidelity components available, bar none, check these three independent magazine surveys.*

They agree on their choices of the *best* turntable and the *best* loudspeakers – moderately priced AR's.

	TURNTABLE	LOUDSPEAKERS
Popular Science (Sept. 1963)	AR two-speed	AR-3's
Bravo! (Fall 1963)	AR two-speed	AR-3's
hi-fi/tape systems (1964)	AR two-speed	AR-3's



The AR turntable—less than ½ the cost of other arm-turntable systems over which it was chosen.

AR-3 loudspeakers—less than ⅓ the cost of other speakers over which they were chosen.

*The Bravo survey introduced its selection of top components with: "If music is so deeply your passion that it makes you intolerant of all compromise...you may enter that rarified area of audio where nothing matters but the dedicated pursuit of perfection."

The **Popular Science** panel tried to eliminate frills, and limited its choice to compact speakers for reasons of practicality in the home, but stated: "Where there was a more expensive component that produced a detectable improvement in sound, it was chosen."

The **Hi-Fi/Tape Systems** survey referred to its choices as "the least expensive way to obtain state-of-the-art performance."

The **Popular Science** survey also recommended Roy Allison's **High Fidelity Systems – A User's Guide** (AR Library Vol. 1, $1). This book may be purchased at many AR dealers', or you may order it directly with the coupon below.

ACOUSTIC RESEARCH, INC., *24 Thorndike Street, Cambridge, Massachusetts 02141*

☐ Please send me Allison's **High Fidelity Systems—A User's Guide.** I enclose $1 in cash or check only, and /or

☐ Please send me free literature on AR products, plus the complete lists of components chosen by each magazine.

*Name*_____ *Address*_____

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Gray Flannel Sound

What I heard as a fact at the show was what I had anticipated in theory in my earlier article. This year there is clearly a sharp difference between the two major groups of loudspeakers, and it is an ear-difference. One group is the sound-alikes, a growing number of speakers that tend in major respects to sound more or less alike in over-all sound quality, without sharply noticeable differences from one to the next. They are neutral, as related to each other. The bigger, fancier ones have splendid specs in detail—wide spectrum, ultra-clear high highs, smooth middles, crisp lows; the cheap items may lack a lot in the top and boast very little low bass at all, but the over-all principle holds: in spite of these differences, these speakers *do sound alike,* and strikingly alike, for the instant ear. This is a value judgment that is too quick for fancy analysis; it hits you automatically, without effort on your part; it is good for any ear and especially the ears of the inquiring neophyte, the awaited customer.

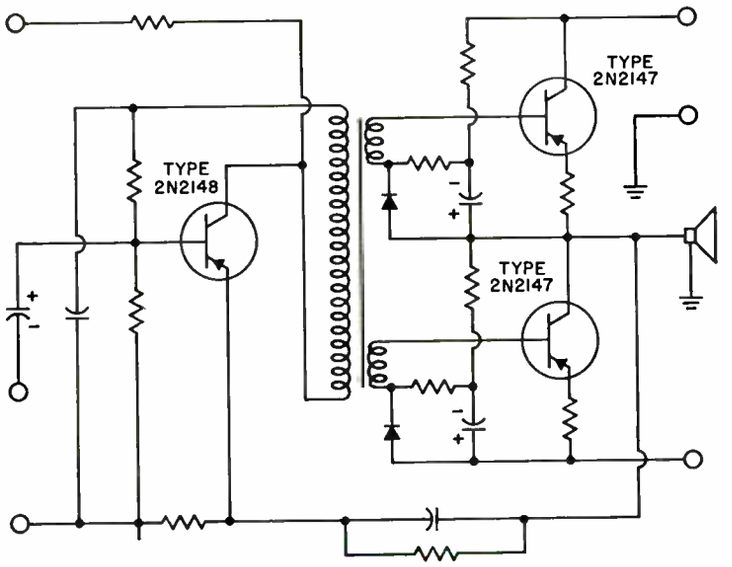
In contrast, the other group of speakers are non-conformist. Unlike the colorless, conformist, neutral-minded sound-alikes, as much resembling each other as so many gray flannel suits on Madison Avenue, these strong individualist speakers are daily becoming more noticeably "different", more consciously special for the average ear. The sound-alikes blend into a colorless nonentity wherever you hear them. You forget the speakers and listen only to the program material emanating from them, which they pass slavishly on to you, without comment, unchanged.

The individualist speakers are rugged individuals of the old school; they tend *not* to sound alike. And that, admittedly, has long been their strongest virtue. Each has its own distinguishing character, its own sound color, its own type of presence, its special enchanced bass, lively treble or golden mid-range. Strong-minded speakers! From one maker to the next, often from model to model within a given line, these rugged speakers show differences that in the continual AB comparisons at a hi fi show are quite startling, from one booth to the next.

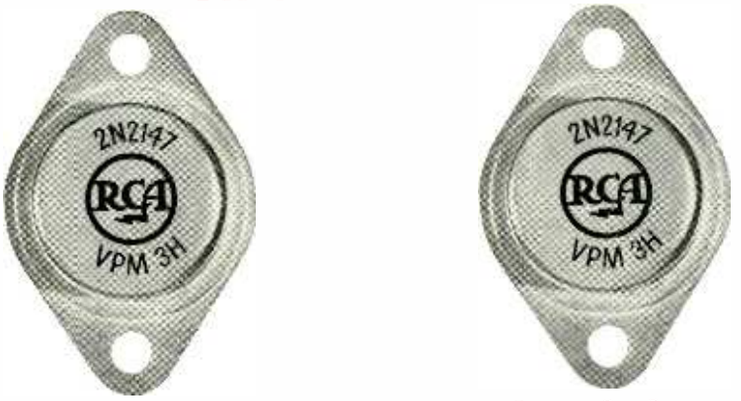
I noticed them, you may be sure. So did a lot of other show visitors, you may also be sure. Simply because they sound so *unlike* the sound-alikes with their conformist neutrality, utterly colorless, servile, undramatic.

You see, we are in a period of change, here as in the great areas of our national life. It is no longer an age of rugged individualism. This is the new age of conformity. I sometimes feel like shedding a tear for the distinguished old-line speakers of yore, and their heirs of today, because after all, I'm a non-con-

(Continued on page 57)

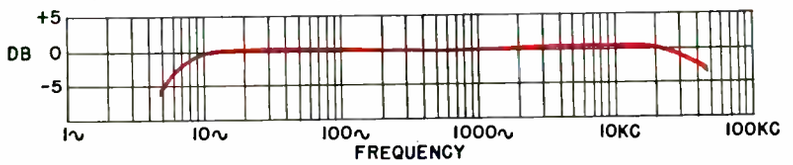


This circuit...



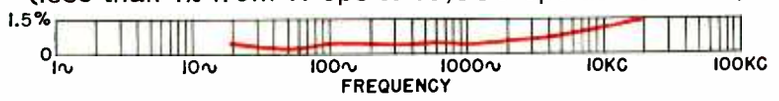
with these two low-cost output transistors gives you audio power output like this

(25 watts—virtually flat from 8 to 30,000 cps!)



with total harmonic distortion like this

(less than 1% from 11 cps to 15,000 cps at 25 watts)



...plus the magic of "Transistor Sound"

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Low cost is only part of the story.

Additional benefits—as in all RCA germanium audio transistors—include exceptional uniformity, high linearity, excellent stability, broad frequency response and high beta.

Positive proof: most of the originators of quality solid-state hi-fi units use RCA audio transistors. And the "new generation" of RCA audio transistors now available deliver still better performance at lower cost.

RCA Top Value transistors and silicon rectifiers include all the solid-state devices you need to design virtually anything from a 1.0-watt-per-channel commercial phonograph to a 35-watt-per-channel professional stereo amplifier. For information on specific types or the complete line, see your RCA Field Representative. Or write: Commercial Engineering, Dept. EZC-12 RCA Electronic Components and Devices, Harrison, N. J.

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ABOUT MUSIC

Harold Lawrence

How To Sit Four Feet From A Symphony Orchestra — And Like It.

THE GENERAL ASSEMBLY HALL of the United Nations is not designed for music. Its absorptive surfaces soak up sound like a giant acoustical blotter, to make speech intelligible over the public address system and through the headphones for simultaneous translation located near each seat. At symphonic programs, some listeners are reminded of the dry acoustics of N.B.C.'s Studio 8-H. An orchestral player summed up his reaction to the General Assembly Hall: "It's like having six blankets thrown over my instrument." Nevertheless a concert at the U.N. is usually a gala affair, not to be dampened by the hall's acoustical shortcomings.

Such an event took place on October 24th when the London Symphony Orchestra capped a brilliantly successful week of appearances in New York with a concert in celebration of United Nations Day. By the time we arrived at the Hall only a few seats in the first row were vacant and we found ourselves on the right, looking down the f-hole of the third cello. The player greeted me and shook his head in mock solicitation: "Too bad about your seat. . . . Would you like me to play on one hair of my bow to cut down on the buzzing?" I turned down his offer; it wouldn't have helped anyway. For the rest of the afternoon I was to hear *all* the cellos with startling, if unwanted, clarity.

George Solti now strode into the hall, hopped onto the podium, bowed smartly, and turned to the players, who covered every plank of the stage. The position of my seat allowed me to observe Solti as he faced the musicians. With a conductor whose translation of musical rhythm into corporeal rhythm is always vivid and dynamic, the concert promised to be a good one, visually if not acoustically. And the visual element was enhanced by the fact that the lights remained at full strength due to the presence of television cameras.

There is a great deal to be said for this kind of ringside listening, both for the layman and the musician. True, the balance is impossible: instruments are either too close or too distant, and there is never any sense of a homogeneous ensemble at work. But the effect of such close proximity can transform the spectator into a participant.

Fig. 1. George Solti conducting.

From my seat a few inches from the cello section, nearly every movement of the conductor could be seen. During the Bruch Concerto, violinist Isaac Stern had reached a climactic phrase which was to be accompanied by pizzicato cellos. The attack of the first pizzicato notes was not precise enough for Solti. Instantly he turned to the cellos and, with raised eyebrows, beat time as if each beat were the snap of a whip; the ensemble straightened out at once. It was as if we in the audience had been flashed the signal.

Under these circumstances it was easy to identify with the cello section. But the sense of involvement extended to the soloist as well. Solti had just launched into a vigorous *tutti* when Stern, his back to the audience, playfully beat time aloud for the conductor: "One . . . two . . . three . . . four." Solti endured it for a few bars, then broke into a wide grin at the kibitzing. At another point, Stern missed a high note in his cadenza; the two men exchanged a quick glance, and Stern winked at Solti.

Elsewhere, soloist and conductor seemed to forget themselves in the score; and when Solti was not blocking Stern from view, their profiles were juxtaposed in ways that a cameraman couldn't have planned better—Stern's robust face, slightly reddening in the heat of a key phrase; and Solti's agile features and hunched athlete's shoulders darting back and forth. Unfortunately, none of the many cameras focused on the performers from the booths above the stage were in line for such a shot.

For the rest of the program, Solti and the orchestra held the spotlight in Bartók's Concerto for Orchestra, and we were given the opportunity to study this conductor's podium technique at close hand. "The gesture," wrote the late conductor, Nicolai Malko, "is the fundamental element of the conductor's technique just as touch is to the pianist." In the world of conducting, podium choreography has ranged from the style of Fritz Reiner, who cultivated a dry vocabulary of tight, precise movements, to the celebrated exuberance of Leonard Bernstein. Solti belongs to the expressive school, far from the vest-pocket approach or the one-armed manner advocated by Richard Strauss, who asserted that the left hand should be left in the pants pocket. The Hungarian-born conductor uses not only both arms, but his entire body in directing the orchestra. His directions range from the tiniest movements to those of a calisthenics instructor.

Solti seems capable of producing the slightest dynamic gradation with almost imperceptible gestures. In the first movement, the entrance of the three trumpets in bar 39 was too soft. Solti

(*Continued on page 70*)

"...THE FINEST LOW-POWERED AMPLIFIER ON THE MARKET..."*

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$99.95 kit
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Now there's a Dynakit for everyone. Long acknowledged as the quality leader of the high fidelity industry, with performance directly comparable to the most extravagant designs, Dynakit now introduces high quality performance concepts at a remarkably low price. ■ Take the time to listen to this latest Dynakit on your favorite speakers. Compare it to amplifiers at least twice its cost. Even with low efficiency speakers, the SCA-35 sounds like a big, powerful, expensive amplifier. ■ Why does the SCA-35 sound so much better than higher priced designs? As detailed and as fine as its specifications are, they cannot adequately define absolute performance. Dyna's painstaking engineering and consummate concern with quality have evolved unique circuitry which fully utilizes the superior performance characteristics of patented Dynaco output transformers—transformers frequently used in far more expensive equipment than our own. Years of refinement have produced a new design with the stability, low noise, low distortion, full power bandwidth, and excellent overload characteristics usually reserved for much more costly equipment. ■ One look inside the SCA-35 will convince you that this is the easiest of all such amplifiers to build. A clean, uncluttered layout and three factory assembled etched circuit boards speed assembly and assure consistent performance from unit to unit, even in the hands of novices. Detailed pictorial diagrams and step-by-step instructions leave nothing to chance.

***Hi Fi Tape Systems Annual,** in their Editor's Choice of Hi Fi Systems, selected the SCA-35 and the FM-3 Dynatuner as offering the "Most Fi per Dollar" (after choosing other Dynakits unanimously for higher priced categories) with the following comments: "The SCA-35 is the finest low powered amplifier on the market, delivers 16 watts (on each channel) from 20 to 20,000 cycles with less than 1% distortion, and below 3 or 4 watts the distortion is unmeasurable."

High Fidelity Magazine (May 1964) reported: "A kit-built version of the SCA-35 proved to be an outstanding performer among low power amplifiers. (It) offers performance that belies its cost, meets or exceeds its specifications, and is in general an excellent high fidelity component."

Audio Magazine (March 1964) concludes: "The SCA-35 . . . is perfect for a small installation where excellent quality, simplicity of construction and operation, and attractive appearance are requisites."

FM-3
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The FM-3 Stereomatic tuner is the ideal companion to the SCA-35 for flawless mono and stereo FM automatically!

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Should Sherwood's new solid-state amplifier be rated at 150 watts?...300 watts?...or 100 watts?

Audio power *should* be one of your major criteria of amplifier performance. The important thing is to use the same yardstick of comparison.

Among responsible component manufacturers, the commonly-accepted expression of audio power today is "MUSIC POWER"—the amplifier's output capability across the full spectrum of orchestral sound.

If you simply like to play with bigger numbers, multiply MUSIC POWER by two (the way some manufacturers do) and you get "PEAK POWER". It's exactly the same rating but it *looks* twice as powerful.

But the really important measurement is "CONTINUOUS SINE-WAVE POWER" with both channels operating simultaneously. This is the *meaningful* measurement, used in laboratory work. It separates the wheat from the chaff.

Sherwood's new S-9000 delivers 150 watts of MUSIC POWER . . . 300 watts of PEAK POWER . . . and 100 watts of CONTINUOUS SINE-WAVE POWER at less than ½% harmonic distortion. (At normal levels, distortion never exceeds 0.15%).

Unequalled power—by *any* standard—is just one of the important engineering advances built into the new Sherwood solid-state amplifiers. Here are some more:

Military-type Silicon Transistors. Used exclusively throughout Sherwood circuitry. Twice the heat-reliability of ordinary germanium transistors. Safe for even the most confined custom installations.

Exclusive transistor short-circuit protection. (Pat. Pend.) New system virtually eliminates transistor failure or fuse replacement due to shorted speaker terminals or other improper operation.

Additional features: Phono input noise less than −65db., with no microphonics or hum / Professional Baxandall tone controls / Tape monitoring and tape-head playback facilities / Stereo headphone jack with speaker disabling switch / Glass epoxy circuit boards / Compact size—14" x 4" x 12½" deep.

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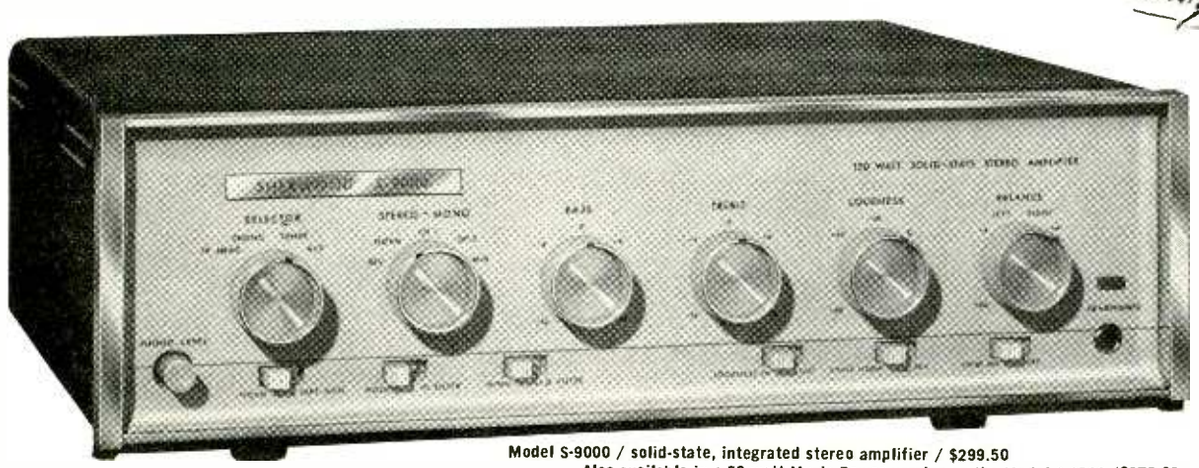

Model S-9000 / solid-state, integrated stereo amplifier / $299.50
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Low-Cost High-Performance Volume Compressor

S. S. ANDREWS

More than 60 db of compression at less than $15, with fast attack, slow decay, and low distortion. The heart of this scheme is a photoconductor.

VOLUME COMPRESSION is one of those areas with which most audio fans are familiar and seldom do anything about. The advantages are rarely realized, and if they are, most feel that it must be expensive to be good. In the past this was true. In order to have good dynamic range, fast attack, slow decay, and little distortion a compressor had to be expensive. Now new devices have made it possible to have a low cost compressor with all of the above excellent characteristics.

The Need for Compression

Why compress? The amplitude range of live musical presentations is often greater than many recorders will handle efficiently. Therefore, continual "gain riding" must be done. However most people cannot react in 10 milliseconds and the end result is overload resulting in distortion.

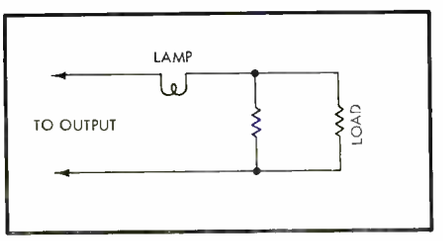

Fig. 1. Series lamp compressor scheme.

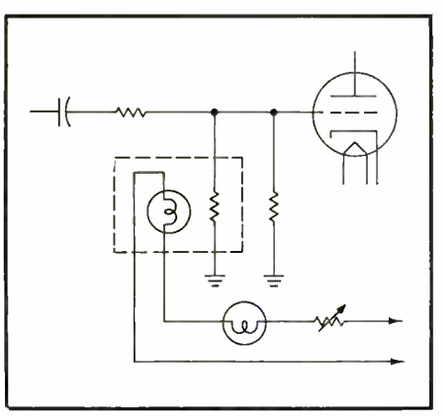

Fig. 2. Raysistor compressor scheme.

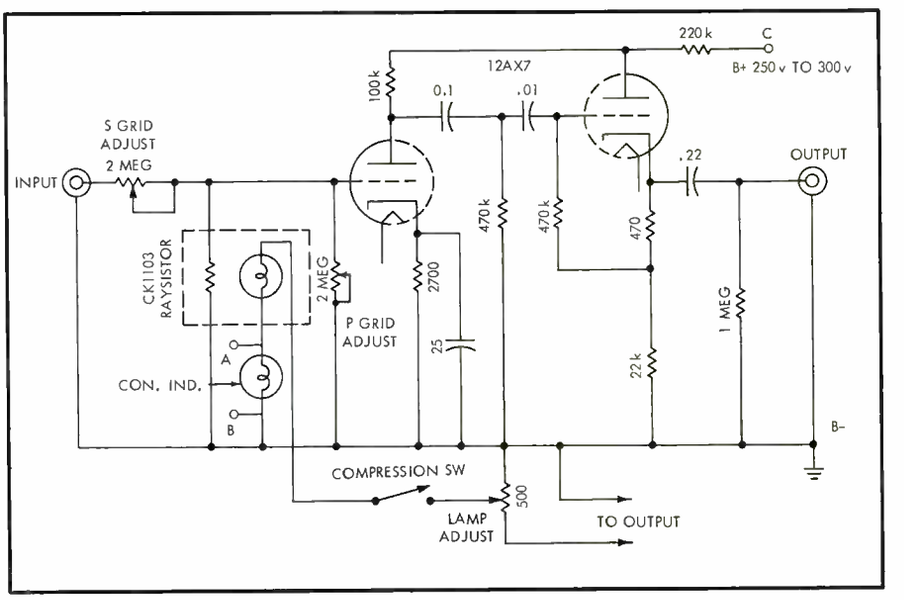

Fig. 3. Schematic of compressor-amplifier.

It is obvious that a need for automatic compression is very desirable. Most commercial recording studios and broadcasting stations use some type of compressing device. This narrows the range of amplitude variations to controlled limits. Also, in background music applications such as for restaurants, hotels, department stores, and so on, wide amplitude variations are not desired.

Various designs have evolved over a period of years for compressing and/or limiting the audio signal for recording. The simplest of these methods, no doubt, was using a pilot lamp in series with the output circuit (*Fig.* 1). The characteristic of pilot lamps is to increase in resistance as they become brighter (more signal), and decrease in resistance as they dim. Therefore, high amplitude levels will cause the resistance of the lamp to increase and less signal will arrive at the output than was fed to the lamp. Conversely, at low levels, the lamp will have less resistance and proportionately more signal will reach the output. The advantage of this system is obvious; the cost is extremely low. The disadvantages, however, are numerous: First, it uses a great deal of power; second, impedance matching is difficult; third, it is most effective when the lamp is glowing rather brightly; fourth, it has an extremely small dynamic range.

Another commonly used method is to rectify the audio signal and use this potential to bias a pentode electron tube and feed the signal through the same electron tube. As the audio signal changes amplitude, the rectified biasing potential changes and therefore controls the output of this tube. This method, of course, has a great deal of merit over the lamp circuit. It does not consume power from the output circuitry. The amount of compression can be controlled rather accurately and, by means of an RC network, the attack time may be varied over a large range. It also has disadvantages. First, the circuit is rather complex and involves numerous components, including a power supply, which makes the cost rather considerable. Second, the audio levels presented

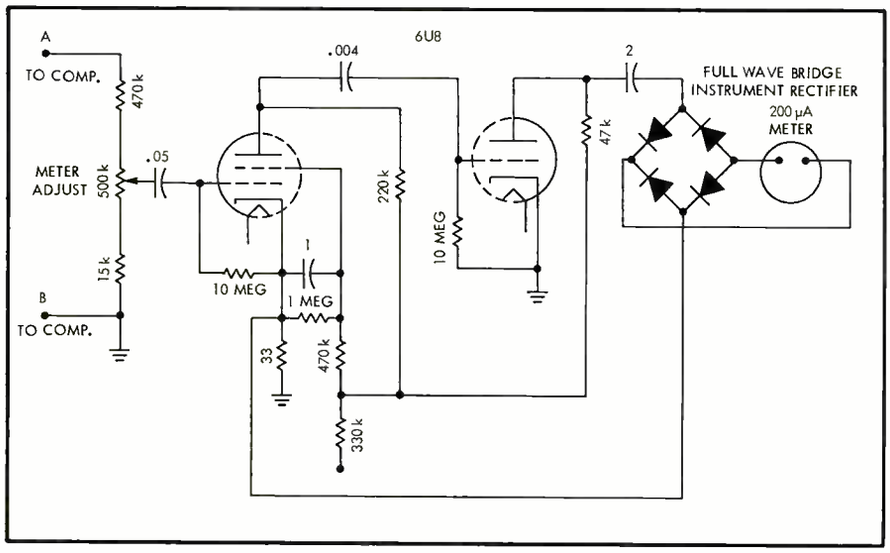

Fig. 4. A. C. VTVM schematic.

to the pentode should operate over a small dynamic range so as not to overload the tube. When this is done, the problem of hum becomes extremely troublesome. Special techniques must be employed with regard to power supply and layout. Third, if higher amplitude audio levels are used at a higher level point in the circuit, then the tube is not operating over a linear portion of its curve and distortion may become excessive. There are many more methods more or less sophisticated than the above mentioned. However, these examples are two of the more commonly found.

A New Device

A new approach is to use a Raysistor. The Raysistor is a Raytheon trade name for an opto-electronic component designed for applications in a variety of control functions, providing comparatively noise free control of a.c. or d.c. signals over a wide dynamic range, without transients or contact chatter and with a high insulation and electrical isolation between the signal and control circuit. It consists of a light source and a photo-resistive element assembled in a light tight case.

The characteristics of the Raysistor which made it very appealing for use in compression are as follows: 1. Fast "ON" action (10 ms); 2. Slow "OFF" action (800 ms); 3. Lamp filament slow response provides an averaging effect; 4. Noise free control; 5. Automatic control; 6. Wide dynamic range.

Placing the photocell portion (a variable resistance element) in the grid circuit of an audio amplifier (*Fig.* 2), makes it possible to easily regulate the signal to the grid circuit since the lamp, which is across the output terminals of the amplifier, glows brighter during loud passages and dimmer or off during soft passages. The photocell has a high resistance when the lamp is off (10^7 ohms) and a low resistance when the lamp is on (50 ohms).

Design Considerations

There are certain design considerations which must be taken in account. I have chosen the CK1103 which has a control lamp with a 5 volt filament drawing 175 ma. Most amplifiers are capable of producing voltages which will far exceed this although it is true that as the amplifier increases in output the input is decreasing by the action of the photocell. However, when low-efficiency speaker systems are utilized, if the bias of the photocell is set to a point where the output would be an acceptable level, then the voltage will exceed lamp specifications. So it is apparent that some means has to be provided to decrease the voltage to the lamp.

A reasonable solution is to place a variable resistor plus a lamp of approximately the same voltage and current characteristics as the one in the Raysistor in series with the Raysistor. The series lamp would allow the voltage level to be determined and also could be used to give indication of the amount of compression taking place at any instant. The series potentiometer could be varied to give the predetermined level.

The circuit can be made as simple or as complicated as desired. In my testing procedure I wanted to have greater control so that I might study some of the effects of the compressor. I decided to place an a.c. voltmeter across the lamp circuit so as to have minimum effect on circuit operation. I used a VTVM circuit. This circuit, when incorporated in the unit, will give a continuous indication of the voltage across the lamp and give a continuous observation during operation. The VTVM circuit is by no means necessary.

The Circuit

About the circuit: The first half of the triode in the amplifier section is a standard voltage amplifier to compensate for circuit losses. The output of the next stage is less than unity. The next stage is a conventional cathode follower which is used for impedance matching purposes and to keep the impedance low so that the high-frequency cutoff is beyond the audible range if cable lengths are fairly long.

The output of the amplifier is fed to the control portion of the Raysistor so that as the output becomes greater the resistance becomes less, thus shunting the grid and reducing the signal and the output. It takes only 10 ms for this to

(*Continued on page 71*)

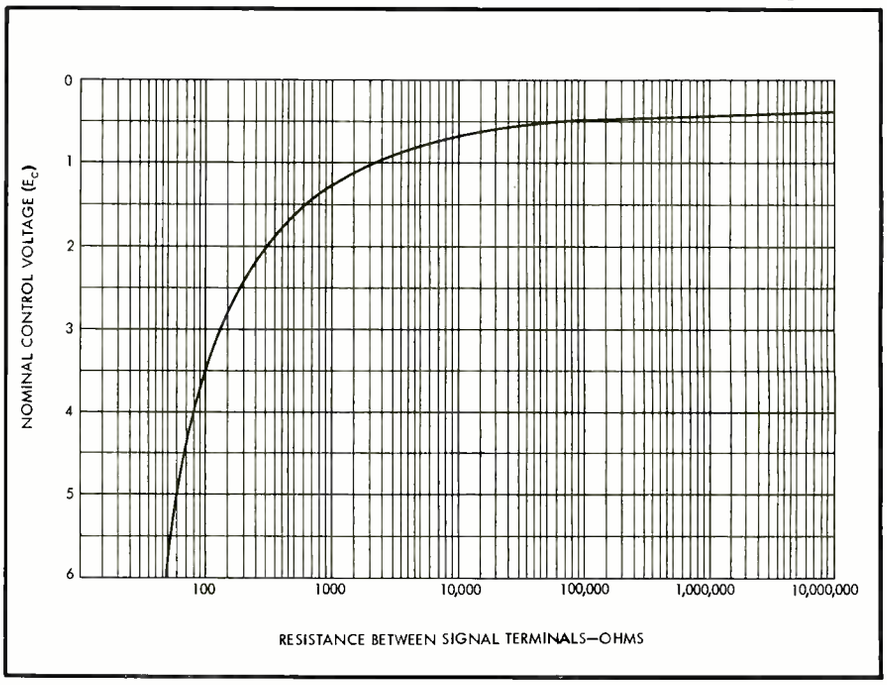

Fig. 5. Typical curve of resistance versus control voltage.

Recording the Muse on Tape

ARTHUR C. MATTHEWS*

Techniques for recording dramatic productions of an amateur group

"WE'RE DOING a play down at the school, could you record it for us?"

"I'll admit that the convention hall is a poor place to do drama, but we've just got to find a way. How about recording it?"

"Well, I wrote this play, see, and I'd sorta like to hear what it sounds like. Could you get a group together and tape it in stereo?"

If you haven't been approached yet, you may be, then what? If you know something about recording music or musicals, the equipment may be the same, but the techniques definitely are not.

As a result of six years of recording drama in stereo, I'd like to give you a few hints that I've found valuable. Remember that the primary purpose of recording drama is that it be heard. If it cannot be heard, and heard clearly, without strain in an ordinary living room, much of its effect will be lost.

Acoustic Environment

Acoustic environment is the feeling of the room which a good recording in stereo conveys. The wall of the listening room disappears and the listener is transported into a concert hall, in the case of music, or into another room, an outdoor scene, or whatever acting area is required in the case of a play.

A room has a characteristic sound. If you've ever slept in a strange room overnight, you know the sensation of waking up in a different acoustic environment. Another indication of room differences is demonstrated if you are brought into a room which is dark or if you are blindfolded. In most cases you can tell the difference between a small room and a large one.

When staging drama for stereo recording you ought to decide what kind of acoustic environment is necessary for the play you are producing.

An outstanding example of annoying acoustic environment is the disc that Basil Rathbone made of Poe stories. Mr. Rathbone sounds as if he were recording in an enlarged rain barrel. The extra reverberation might be suitable for the catacomb scene in "A Cask of Amontillado," but it hardly suits the other stories.

Imagine Mr. Rathbone standing on the edge of an empty loft, on a slightly raised platform, book before him, microphone on the floor about ten feet away. He gazes into a murky Poe vastness, empty except for a recording engineer off in a corner somewhere. Too bad nobody came to absorb some of the sound. Such a vast hall is annoying if there is no artistic reason for that much space.

Acoustic environment ought to consider "living-room-ability." How will it sound to a few people in a living room somewhere? Music can get away with an "enlarged" space because we know that an orchestra requires a large performance area and space for the sound to develop.

Live Recording

Not much can be done about acoustic environment if you make a live recording, but there are some advantages. The hams (actors to you) can go on about their business in front of an audience, and usually give a better performance.

The biggest difficulty is miking. A minor problem is audience noise. Should it be minimal or part of the recording?

The answer to the noise question is psychological. If you are just making a "memories" recording for the cast and director, they simply want to hear how they sounded. Audience noise isn't a problem as long as it doesn't interfere with intelligibility.

If you're going commercial either for broadcast or disc, then the psychology of the listening situation is important.

Comedy *seems* to require laughter. Many TV programs add "canned" laughter to their comic gems, assuming that laughter will encourage the audience at home to laugh too. There is a squared correlation in opening the can: $I = L^2$, where I equals the inanity of the material, and L equals the quantity of the laughter added. In your recording you'll be stuck with the laughter, or lack of it, that a real audience supplies. You can always doctor the recording.

Still, my own experience leads me to believe that laughter, canned or live, does not add anything to the home listener's enjoyment of a play. While a matter of taste, you will decide which solution you prefer, for the answer dictates the mike setup.

There are several mike patterns: 1. Omnidirectional mikes pick up well from all directions; 2. Bidirectional mikes pick up from the front and back, in a figure 8 pattern; 3. Cardioid mikes pick up in a "heart shaped" pattern from the front; 4. Super-directional mikes pick up in a very narrow beam.

For most rooms, bidirectional and cardioid microphones give best results. If the room is small, an omnidirectional mike picks up the reflections off the walls. If you have an ideal recording studio, the omnidirectional pattern makes an excellent recording. If the studio is less than ideal, the other two types are better choices. Cardioid and bidirectional mikes are best if you want to keep audience participation to a minimum.

At Luther North where I recorded a number of plays, I also installed a sound re-enforcement system. The auditorium is one of those multi-purpose rooms which are better named no purpose. A good theatre usually needs an acoustically live ceiling. A cafeteria-study hall-playpen requires a dead ceiling. Guess who wins?

The low ceiling (13 ft. 6 in.) combined with the 3-ft. stage height put the actor within 4 ft. of the acoustic tile ceiling. Unless an actor has iron lungs, nobody can hear him 20 ft. away. If he can be heard, he'll probably end up in an iron lung after one performance. Sound re-enforcement is necessary, no matter how much the director may object.

The re-enforcement at Luther is stereo. Four Electro-Voice Sound Spot (644) mikes are spaced equally across the 30-ft. stage opening. The four mikes are jacked into the inputs of a Crown 700 series recorder. The outputs of the recorder feed a Dynakit preamp-amp combination. The amplifiers supply a series of speakers above the proscenium arch. A slight blend of channel A and B is used.

Fortunately the mikes hang relatively low. If, however, the stage you're going to work on has a higher opening, try to come as close to the actors as possible.

This may mean dropping the mikes down into the stage setting. Of course the director may complain that you are ruining his decor. Altec lipstick mikes are one solution. The other is to place the

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mikes in the footlight trough (if there is one, they are fast disappearing) or on small floor stands across the front of stage. Be sure the front curtain doesn't take the mikes with it when it opens or closes. For each 10 ft. of width beyond 30 feet, add one mike for each channel (rule of thumb).

If the acting area is more than 15 ft. deep, you *may* have to add a line of microphones for the back of the stage. Try to keep the mike cables away from the lighting cables, you may induce some hum.

If you use sound re-enforcement which has a means of separating the line after preamplification and before the final amplifier stage, I suggest some form of compression.

A compressor boosts the apparent loudness of low volume material and cuts the apparent loudness of high level material. A compressor tends to make both loud and soft passages move toward a middle ground. In recording music, large volume variations tend to produce exciting music. In recording plays, such large volume variations make some parts inaudible and others too audible.

Compression will not only help the re-enforcement, but will permit you to ride gain a little higher and pay more attention to the play and less to meter needles or flashing eyes. The flashing eyes of the leading lady are much more interesting.

A simple way to compress without ruining the budget is to insert a Fairchild Compander, a Knight KN 777 expander-compressor, or build your own (see Andrews article in this issue), between the preamp and the amp. (See *Fig.* 1.) A "Y" connector at the output of the compressor feeds both the re-enforcement amplifier and the tape recorder. Compression can be added to the playback, but compression before the recording helps make a better signal-to-noise ratio.

WARNING: a compressor by its nature increases background noise level during quiet passages.

If you've decided that laughter is needed to complete a comic illusion, bidirectional microphones are necessary. Omni-directional mikes are possible but require a good acoustic environment.

I recorded a version of "The Doctor in Spite of Himself" (Molière) with the North Park College Players using two bidirectional ribbon mikes. We had a separate re-enforcement system. The mikes were placed on stands directly in front of the stage and feed into a Concertone 505 recorder.

Two problems: 1. Two mikes weren't enough, especially since there was a great deal of action upstage; 2. The front of the stage is too far front for bidirectional mikes (they tend to pick up the audience better than the play).

It seems that Jon and Lars (this is a

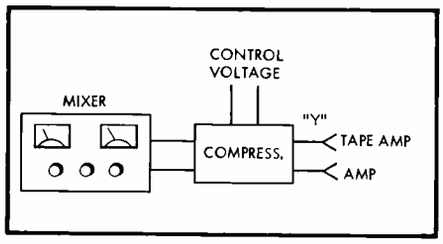


Fig. 1. A compressor helps sound reinforcement and taping.

Swedish school or they might have been Pat and Mike) were sitting in the front row. Kids have few inhibitions so they laughed and laughed and talked about the play. As a director I was happy. As a recording engineer-listner, I was not. Two enthusiastic audience members giggling in the front row are annoying. A better perspective on the laughter was possible by hanging the mikes from the top of the proscenium arch pointing so that the laughter is picked up as a bounce from the ceiling.

I have never been able to find a satisfactory compromise between live and studio recording. If you record live you have noise and level problems as well as an artificial sound. If you record in a studio the actors lose part of the edge to their performance.

Studio Recording

A recording in a good acoustic environment seems preferable to one with the actors projecting for a large audience (to listeners in a small room). Perhaps a special recording session with an audience can be arranged. Shoot all people with smoker's hack, hay fever, and common colds at the door!

Deciding that studio recording is the lesser of two evils brings you smack against three major problems: 1. Studio-playback area; 2. mike type and placement; 3. stereo movement, how much and how.

The studio need not be a professional recording studio complete with floor suspension, but it should be located where the noise level is low. Probably at least a block in all directions from main traffic arteries. The further away from the mikes the actors work, the higher you have to raise the level of the mikes. Noise in the background comes way up too. If you have a noisy background, work close to the mikes.

In recording "My Own Song" I was faced with a shift in flight pattern. O'Hare International Airport sends up jets like clouds of mosquitoes. The mosquitoes do not ordinarily bite, but when the wind shifts, every jet in the U. S. makes a pass over the "studio." We consulted the augurers about the future and moved the session to a more propitious time.

The size and shape of the room are important too. You might think offhand that a recording on the stage is best. But this isn't so in many cases.

The recent recordings of "Strange Interlude" (Eugene O'Neil) and "Who's Afraid of Virginia Wolf" (Edward Albee) are good examples. The O'Neil shifts acoustic environment for indoor and outdoor effects. Virginia's tinkling ice cubes are sometimes irritating but effective. The recording of "Waiting for Godot" (Becket) while not in stereo, has an excellent perspective (distance of actors from the mike).

Since most plays take place in a room, the studio should probably sound like a room, not Howe's Cave or The Chicago Stadium where Al Melgard, the organist, plays for ten minutes and then rests ten to let the sound catch up. Take the mikes to several locations. Remember, if the sound is too dead the voice will not seem to be in a room and the stereo effect is lost.

Reverberation can be added artificially, but I have yet to hear a satisfactory reverberation system for voice. I may have heard one so clever that I didn't notice, but most are abominable.

The room should sound "big enough" to be different from the listener's normal environment, but not cavernous. Symphony Hall in Boston is ideal for the Berlioz *Requium* but a little large for a play.

It is impossible to be too exact about the kind of room that will sound best. A room smaller than 10×15 ft. with an 8-ft. ceiling is probably too small. Rooms with acoustic tile ceilings are probably out in most cases.

The ceiling of a room has a great deal to do with how sound develops. A room with plaster walls, wooden floor, without furniture or carpets will probably be too live. Brick and cement block interiors, because of the roughness of their surfaces, tend to be interesting acoustically.

There is no really ideal room. As director you'll have to decide what kind of sound you're looking for and then listen for a room. But since your ears aren't at all like a microphone, you'll have to perform some tests on the room.

To test a room, place your mikes across the short side of the room at one end. Make a "take" (recording) of some material from the play; move the mikes closer to the far wall, retake. Do this several times. Then make tests across the wide side of the room. Try shades up and down. Drapes opened and closed.

DO NOT listen to the playback in the same room. The listening room should be an "average" living room (whatever that may be).

By all means try several rooms. And don't be confused by looks . . . sound is what counts. Bruno Walter recorded his California dates in an absolutely hideous looking room; the sound was excellent.

When you are satisfied with the hall,

be prepared for changes as the humidity and temperature change.

You might start your tests on stage by closing the front drape. The stage may sound too big . . . add some more drapes to the sides. Try recording with the actors playing with their backs to the rear wall, talking towards the front drape. The bounce off the back wall might add just enough extra liveness. Experimentation is the only way to be sure. There are too many variables.

In considering the studio, consider where you want to put the recording and playback equipment. There are two theories here. Some directors like to be in the same room with the actors and wear earphones. The director must hear how the recording sounds, not how the actors sound in the actual room.

Earphones are dangerous unless there is a device like the Bauer circuit incorporated. Without some mixing of the two channels you will hear "binaural" (two ear) recording through the headphones. The separation between channels is made greater than the listener at home will hear. The Bauer circuit mixes L and R and puts you back into a room. At best, earphones are dangerous.

The room in which the director listens should be as much like a normal living room as possible. The speakers should not emphasize a particular frequency range. The director and engineer may have to be located some distance from the recording area. An intercom may be necessary.

Preliminary and final testing of the recording should be done by someone who is not familiar with what you did or are doing. He (or she) should be someone you believe to have taste, someone perhaps familiar with drama, but not with the play. Your judge should be a "new ear" which will not be hearing things *not* on the tape.

I often despair as a director because I'm so close to the show that I can't hear it any more. Just recently my Christian Theatre Company performed in the Valparaiso University Gymnasium before an audience of 600 people. I'd heard the play so often that I simply could not tell whether the sound re-enforcement system helped at all. Fortunately it did, but I wouldn't have known. I'd directed the play with two groups and had written it. You can't find a more prejudiced critic.

A quick review then: A studio with right liveness; a playback room with right deadness; intercom. Fine you say, now how do I set up the mikes (Always questions). How you set them up will depend (as you've guessed) on the kind of effect you are looking for.

Microphone Placement

There are two basic techniques I use . . . curtain and three area. In the curtain approach, the microphones are set

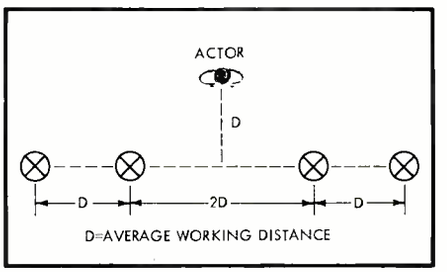


Fig. 2. Spacing microphones based on average working distance.

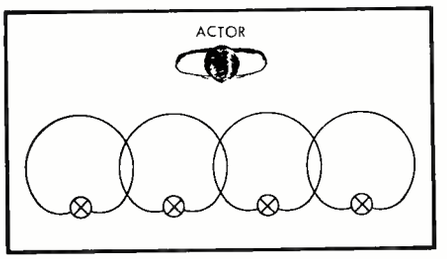


Fig. 3. Using cardioids for the "curtain" approach.

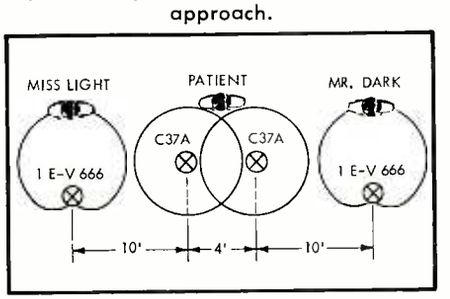


Fig. 4. Using cardioids and omni's for "three-area" recording.

up so that an *even* pickup of sound seems to spread across the listening room. This might be accomplished with four (or three) omnidirectional microphones. I have used Sony C37A's with some success.

The exact spacing will depend on the size of the room and the kind of acoustic environment you want.

The distance between the mikes (*D*) should be the same as the *average working distance of the actors from the mike.* This average (middle) distance is the one at which much of the acting is done. A shouty play will probably have a greater "*D*" than a quiet play. Start by placing the center mikes 2*D* from each other, and the left and right mike 1*D* on either side of the two center mikes (*Fig.* 2). The 2*D* figure is not a sure bet, but it *is* a place to start. You may have to move the mikes closer together, approaching 1*D* or less.

If you use three mikes, the single center mike will have to have two jacks on the far end of the cable. One jack will plug into each channel. The three mikes will be separated by 1*D* each. NOTE: The distance factor will change if the actors are closer than two feet to the mikes, since the square law does not apply in this case.

The curtain of sound can be created (*Fig.* 3) using cardioid and bidirectional patterns. Experimentation with spacing is necessary. This "*D*" should not probably be over four feet for most types of mikes. Remember, the reason you went to studio recording was so that the actors wouldn't have to project and seem forced in their speaking.

In recording a live performance of "Christmas 1960" with a speech chorus of 100, intermixed with various solo speakers and small groups, I was able to use spacing greater than 4 ft. Two mikes were placed about 6 ft. in front of the chorus, up to full height on the stand. The mikes were bidirectional ribbon mikes (B&O 50's). The mikes were separated about ten feet from each other.

Since the recording was done in a gymnasium, the reflection off the opposite wall and ceiling was picked up by the back of the mike. For the solo voices we used two E-V 664 cardioid mikes separated about 30 ft for a real L and R isolation. The left and right mikes also picked up the ends of the chorus, although they were 12 ft from the front of the group.

An ideal solution, which I haven't tried but am looking forward to is two or four AKG C-12 microphones. This type of microphone has a control for complete pattern variability from cardioid to omnidirectional. Test recordings in various pattern positions can be made and the best one chosen.

Another experiment involves four bidirectional mikes mounted in an unusual way. Each channel consists of a pair mounted one above the other (colinear) with patterns at 90 deg. to each other. The pair jacks into two separate inputs of the same recorder channel. B&O-Dynaco makes a colinear unit (model 200) which might be ideal. Consider the "front" of the microphone the pattern which the actor faces. It is operated at normal level. The "side" of the microphone (picking up reflected sound) is varied in volume until the kind of acoustic environment needed is achieved. The experiment ends up with a sort of variable omnidirectional mike. The "side" mike can be rotated until the angle between it and the "front" mike is 0-deg. All sorts of possibilities!

Finding the correct "*D*" is critical. If you place the mikes too far apart, the actors will drop in and out as they move from L to R. This simply sounds like the microphones are too far apart.

If the mikes are too close, the L and R channels will blend and the sense of movement will be reduced. Don't make the sound curtain too wide from L to R. A wide curtain could expand the listening room to disturbing proportions.

Three Area

The three-area approach uses three or four cardioid mikes or two bidirectional mikes with a spacer. In recording *The Rick Mark Show,* about a teenage

hero, the separation problem was solved by the nature of the play. The stage represented the dressing room of the star of our show and the dressing room next to it, where Carlotta Parks and Arthur Hopkins waited to go on Rick's show.

For this recording I used four E-V 644 Sound Spots arranged so that the actors played close to the right mike when in Mark's room, and close left mike when in the other room. The center mikes picked up a little of the talk in both cases to give the in-a-room effect. This is really a two-area recording.

"Welcome Angels" presented a different problem. Two Angels, Miss Light and Mr. Dark, have come for the Patient, who is dying in a chair center. When he dies at last, his final words are so equivocal (neither hot nor cold) that Dark and Light can't decide who gets him.

Because I wanted to contrast good, evil, and the poor confused patient in the middle I used the following setup: L mike E-V 666 cardioid, two center mikes (10 ft. from L mike, 10 ft from R mike, separated by 4 ft) Sony C37A's in omindirectional pattern, R mike EV 666 cardioid (*Fig.* 4).

The three-microphone three-area technique uses three identical mikes, separated 2*D* or more. The center mike terminates in two jacks, one plugged into L and one into R channel. There is some danger in this three-mike technique that the center channel will have a different frequency response because of the "bridging" device.

Two Dynaco-B&O mikes with an acoustic spacer are used in the three-area technique. The mikes are not set up as they would be for ordinary stereo. The mikes are turned so that at the front they face the centerline of the spacer, giving a close center channel. The backs of the mikes are turned away from the spacer which acts as an acoustic barrier between L and R channels. Thus there is a dead spot as an actor moves from left-center-right. I put adhesive or masking tape on the floor to indicate to the actors areas to keep out of. The actors moved between areas during pauses. They worked from 2½ to 4 ft from the mikes depending on the scene and the power of their voices.

One point I learned during this production, that the floor should be marked with radial lines around the mikes so that the actors can find a position and keep it. Exact positioning helps in editing.

A variation on this setup uses the two bidirectional mikes and an omnidirectional suspended over the basic mike setup (7½ to 8 ft. off the floor) with a double jack on the end feeding a sort of center room air channel. The experiment was not too satisfactory. The ceiling of the room was too low and the room too small. A little experimentation in a larger room might prove out the technique.

Amateur actors do not have the ability to control their voice volume in the way a professional actor or announcer can. I know it took me a full summer at WFIU in Bloomington to get the technique. After the play was edited, I found the volume variation from scene-to-scene and voice-to-voice too much and compressed about 20 db in playback.

There are then several mike setups for each of the two basic techniques. Curtain of sound is the most difficult because it requires an eveness which takes experiment to achieve. The three area, L-C-R, has something to recommend it, especially with amateur recordists without all kinds of time and equipment.

Movement in three-area is constricted, but it is thereby easier for the audience to visualize exactly what is going on. In a theatre the audience sees the actors moving . . . at home the listener has no way to tag who is where.

Plays with large numbers of characters and constant violent action are probably better left out of your recording career until you can handle the medium.

Problems of movement and problems of voice color are questions of taste, but some microphone problems are questions of mechanics. Some condenser mikes seem to favor "S" sounds. With ribbon mikes plosive sounds (p, b, t, d) and fricatives (f, v) are likely to drive the ribbon out of its normal vibration circuit and set up low-frequency pops or thuds. The closer the actor is to the mike, the worse the problem. Back the actors off, or teach them to control their enthusiasm on p, b, t, d, f and v. Dynamic microphones are not as sensitive to fricative problems, but may be popped by plosives under some circumstances. Fire all poppy, hissy, or ffffffy actors.

Staging for Recording

Now some aesthetic problems. Too much movement can be annoying. The listener in his home without a book cannot be following hopping, skipping, and jumping actors. As a director, sit down with the book and decide who should be where, when. If the scene is intimate, play it with the actors close to mike and close together. Shouting scenes (for the sake of audience, actors, and engineers) are best played further away than "*D*".

Most plays break up into what directors call "French Scenes." A French Scene begins or ends every time an actor enters the set or leaves the set. In some cases, a different scene begins if there is a major change in idea. Breaking the play into French Scenes can be helpful in deciding movement.

The director must clearly establish entrances and exits. For example, the kitchen is middle, right; the door to outside is center, far distance; the bedroom is near left. Unless you want a peculiar effect, the actors should not turn their backs as they exit. Move out backing up.

In the three-area approach, divide the play into French Scenes and decide which of the three major areas you intend to use for each scene. Restrict movement in most cases to movement to or away from the mike. Once the actors know where to stand, *make them stay there.* Movements of heads or feet can cause an audible shift, especially with the two bidirectional microphones and spacer technique. The center area in this setup is particularly sensitive.

Forget any experience in directing for the stage. Remember what you learned from radio, adapt it. As in radio, keep the actors close to the mike, especially in a poor acoustic environment. Avoid the stage director's temptation to move actors back and forth. The actors can fade into or out of a scene, but do not play important scenes far from the mike "*D*" or less!

Alternate close, medium, and far distance scenes. The very close-to-the-mike method can be used for 18th century plays where asides abound and in Shakespeare for soliloquies. Again experiment will tell.

Another directing problem: Should the recording be made from beginning to end without changes of scene order? If the actors have rehearsed for some time together, yes. Even though they do not have an audience, some sense of performance is possible.

But in recording with a drama class, for example, usually it is not possible to schedule four-hour sessions for everyone at once. Break the play into French Scenes (it's surprising how often only two or three characters are necesary at one time); match the actor's time against the scenes, and set up the schedule.

Under these circumstances, it's probably best to record the middle scenes of the play first. Film producers often use middle, beginning, end as their filming order. At first the actors aren't quite in character. If you record first things first, their characters are not set sufficiently to project the personality. If you record the last scenes, they haven't built the emotional intensity for the scene. The middle scenes, then, seem to be safest.

The pieces and bits recording method should be scheduled: several complete read-throughs of the play, marking scripts with ideas of character, emotional climax, important lines (point lines), speed, and mood. Help the actors get an idea of the character they are to play.

In working with individual scenes, my approach is to assume that actors are intelligent human beings who respond to someone who helps them release and

(*Continued on page 62*)

Electronic Organ Tone Coloring

D. WOLKOV

Electronic organs achieve the required tone coloring by a variety of processes. Here is a description of these techniques.

A FASCINATING, AND STILL DEFINITIVE, research effort on musical scaling, harmonic overtones and beats is described by Helmholtz in his "On the Sensations of Tone as a Psychological Basis for the Theory of Music."[1] Particularly, those readers who desire a deeper understanding of what electronic engineers need to know about music so that oscillators and filters can be converted into a musical instrument should refer to Part III, pages 234–371 of this book.

Although his book was written about 100 years ago, the theories and explanations have withstood the tests of inspection and challenge. Without describing how these well understood (but not defined here) laws pertain to the physiology and psychology of music, we will direct our interest in this article to: 1. Harmonic analysis. 2. Harmonic synthesis. 3. Passive electrical filters as analogs to mechanical formants. We will also discuss how these techniques are used in modern electronic organs.

Later we will see that the basic constituent of sound is the sine wave. To understand sound, we must first understand *simple harmonic motion*. The mathematics and basic concepts can be found in any text on mechanics.

An elastic body undergoing vibration, be it a violin string, or a flute, vibrates in a complex mode, but it can be shown that the complex mode can be translated into a system of simple harmonic motions by harmonic analysis.

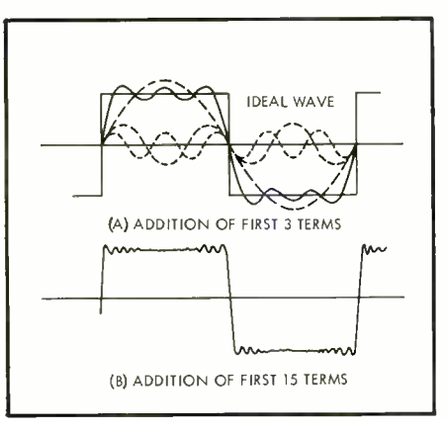


Fig. 1. Example of harmonic synthesis.

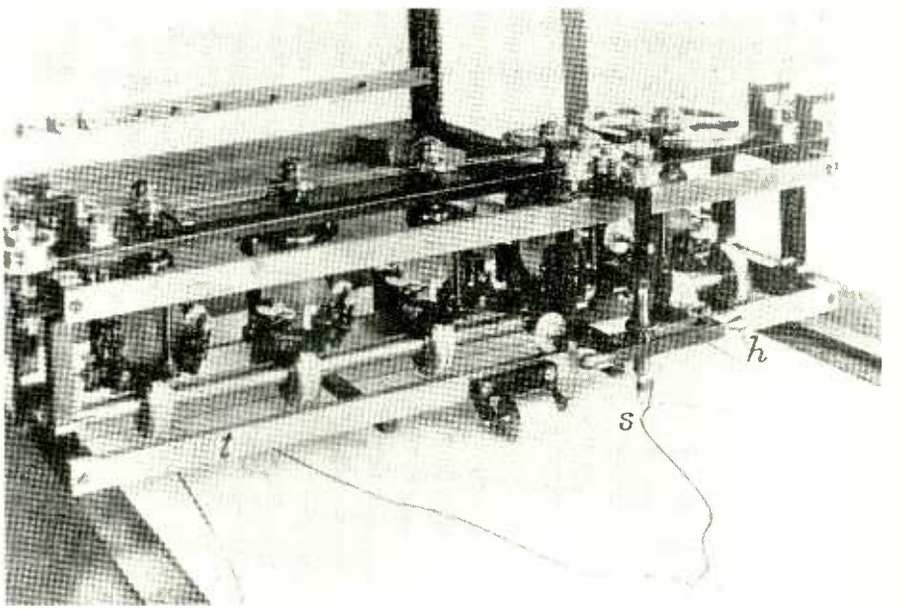


Fig. 2. Mechanical harmonic analyzer. **(Courtesy Case Inst. of Technology.)**

Dr. D. C. Miller in his book, "Science of Musical Sounds,"[2] provides a fascinating picture of techniques of harmonic analysis before high-quality microphones, amplifiers, and tape recorders were known.

One of the more interesting devices described is Koenig's apparatus (1862) in which a burning gas jet was modulated by sound. The combination of a microphone by Bell and an oscillograph by Blandell (1893) is also described by Miller. He concludes "that mechanical and electromagnetic factors produce appreciable alterations of the wave forms" (low fidelity?). Also of interest is the Scripture machine which made optical traces from phonograph records—one turn of the 78-rpm record in five hours and an additional optical magnification of 300. The first real technical breakthrough was Miller's phonodiek, built in the early 1900's, an optical-mechanical microphone, which was flat from 0 to 10 kc.

Fourier's Theorem, published in Paris in 1822 by Baron J.B.J. Fourier, states that a curve, having a wavelength l, can always be reproduced in one particular way by compounding simple harmonic curves of suitable amplitude and phase having the same axis and having the wavelength l (fundamental) $l/2$ (first harmonic) $l/3$, $l/4$ . . . l/n (n−1 harmonic). There are other restrictions on the theory for which the reader is referred to Wood,[3] Pender/McIllwain,[4] and Woods.[5]

It should be noted that Fourier's theorem assumes two conditions regarding the form of the complex vibration: 1. The displacement must be single-valued and continuous. (It is obvious that this condition is fulfilled in all cases of mechanical vibrations since a

[1] Helmholtz, "On the Sensations of Tone as a psychological basis for the Theory of Music," Dover Publications, N. Y., 1954.

[2] D. C. Miller, The Science of Musical Sounds, Macmillan, N. Y., 1916.

[3] A. B. Wood, A Textbook of Sound, The Macmillan Co., 1937.

[4] Pender/McIlwain, Electrical Engineer's Handbook, John Wiley Sons, N. Y., 1950.

[5] F. S. Woods, Advanced Calculus, Ginn & Co., New York, 1934.

particle cannot actually have two different displacements at the same instant of time); 2. The displacement must always have a finite value (this condition also is clearly fulfilled in the case of sound).

Taking the square wave as an example of the application of Fourier's theories, *Fig.* 1 shows the graphical addition (harmonic synthesis) for 3 and 15 terms. It should be noted that at 15 terms the square wave is fairly well approximated. A rigorous mathematical proof and further demonstrating examples are contained in Rainville's text.[6]

In a similar manner, we can analytically find the harmonics of any periodic wave if we can describe its function. If we cannot then we must resort to a mechanical analysis (which is a numerical approximation method using Fourier Analysis) or a point-by-point integration using a digital computer.

Back in the early 1900's, digital computers were an unforeseen art. This did not stop the researchers in the harmonic analysis of sound. Instead they built analog computers. Many accurate machines of this type and of great accuracy were built. *Figure* 2 shows a five-coefficient harmonic analyzer and *Fig.* 3 shows the details of a rolling-sphere integrator. *Figure* 4(A) shows an oscillogram of an organ pipe sound and *Fig.* 4(B) shows its harmonic content when analyzed by a mechanical harmonic analyzer. *Figure* 5 shows the recheck when the function is recomputed from the coefficients, frequencies, and phase shifts given by the analyzer.

When an oscillograph of the sound from a bell is inspected it turns out to be a non-periodic curve, thus, Fourier techniques are not applicable. This is evidenced in the practical art: Schulmerich supplies mechanical chimes and Artisan supplies bells and other percussion instruments as adjuncts to electronic organs. We conclude that periodic sounds can be synthesized by harmonic oscillators and aperiodic sounds cannot.

Further investigation reveals that it is important to plot the magnitudes of the amplitudes against the frequency of the harmonics. The relative amplitudes of harmonics for the violin, french horn, and flute were determined by Dr. Miller using mechanical recording and mechanical integration.

By 1929, the Bell Telephone Laboratories were able to determine amplitudes and frequencies by electronic means.[7] They used the apparatus shown in a block diagram in *Fig.* 6. The fourteen bandpass filters were designed so that they presented a uniform impedance of

[6] Rainville, Elementary Differential Equations, Macmillan, N. Y., 1958.

[7] Sivian, Dunn & White, Absolute Amplitudes and Spectra of Certain Musical Instruments and Orchestras, *J. Acous. Soc. Am.*, January, 1931.

PERFECTIONISM IS NOT INEXPENSIVE, EVER

Over two years ago, word leaked out (as it will) that Shure, one of the world's most respected manufacturers of high fidelity components, had embarked on an epochal project: the creation of a perfectionist's compact stereo system. A high fidelity system that conjoined optimum sound and minimal size.

A formidable and dedicated group of development and design engineers was given carte blanche.

Independent high fidelity authorities were flown in for repeated consultations. Their recommendations and comments provided a demanding framework for Shure engineers.

Prototypes were modified and improved . . . and improved again . . . and again.

The monies and skills appropriated to the project were sizable — by ANY standards. Perfectionism is not inexpensive, ever.

THE M100 SYSTEM

The result of this highly disciplined creativity is Shure's M100 Maximum Performance component high fidelity system. It is not just good, it is great.

It will re-create your favorite records with a sound quality that is remarkable in its naturalness and exciting in its impact. It is unlike anything previously available in compact systems—save perhaps a highly-inspired custom component rig costing two (or more) times as much.

Because of the singular standards for sound and for size that were set for the M100, few components were judged to be satisfactory for inclusion in the unit.

Consequently, the Shure design staff (that developed the unique Dynetic cartridge which made true high fidelity stereo a practical reality) designed many of their own components. In addition, quality is controlled by the famous Shure Master Quality Control Program.

The M100 embodies significant and unique engineering considerations, such as a solid-state pre-amplifier/amplifier developed and produced by Shure with more than ample power to drive its two total-range, ultra-compact multi speaker systems, also designed by Shure. And, it features the renowned Shure Bi-Radial Elliptical Stylus in the famed V-15 Stereo Dynetic 15° tracking cartridge. Naturally, it cannot scratch records. Dual's finest precision Model 1009 automatic turntable is standard.

The M100 can be used with AM, FM, Multiplex tuners, or be used as a public address system. Ideally suited for every home-listening requirement, the M100 is also recommended for use in schools, hospitals, and other institutions. It is Underwriters' Laboratories, Inc. listed.

Space is not sufficient here to list all the M100's pertinent specifications. They are for a system of this size, singular. We will be happy to send complete specifications to technically oriented music lovers.

LIMITED QUANTITIES

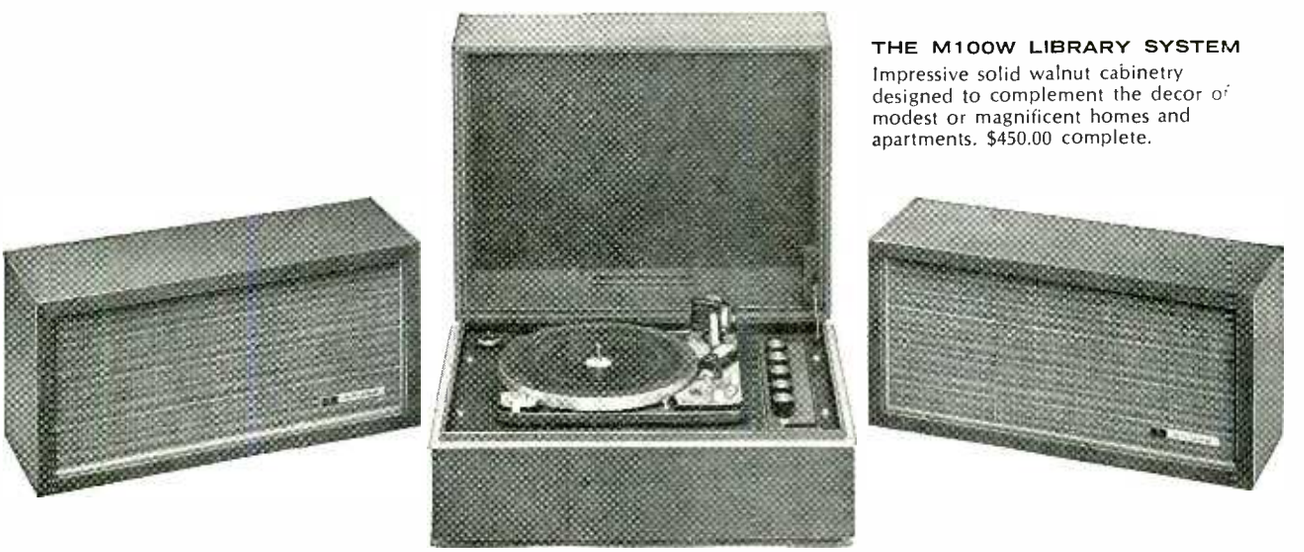
Because of the detailed craftsmanship, large amount of hand-labor, and rigid quality control and inspection techniques involved, the number of M100 systems manufactured will be few. They will be available only through a select group of the most experienced high fidelity consultants and dealers. We cannot promise immediate delivery. For perfectionism cannot be mass produced.

THE M100L PORTATIVE SYSTEM

In two instrument cases. Recommended for serious listening among music lovers on the go: armed forces members, boat owners, college students, school music directors, or anybody given to travel. Or summer cottage owners, of course. $389.00.

THE M100W LIBRARY SYSTEM

Impressive solid walnut cabinetry designed to complement the decor of modest or magnificent homes and apartments. $450.00 complete.



SHURE M100

THE SHURE ENGINEERED SYSTEM OF LABORATORY MATCHED HIGH FIDELITY COMPONENTS

Shure Brothers, Inc., 222 Hartrey Ave., Evanston, Illinois

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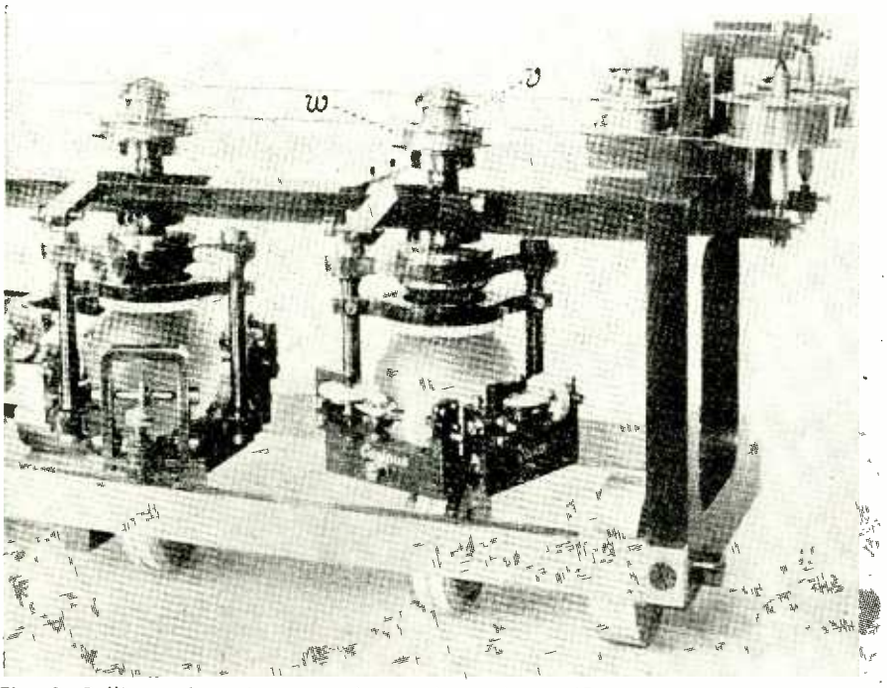


Fig. 3. Rolling sphere integrator-harmonic analyzer. **(Courtesy Case Inst. of Technology.)**

600 ohms at all frequencies. Only one filter at a time was used, this one being connected to the 600-ohm measuring circuit while all the others were terminated by 600-ohm resistances.

Today, harmonic analysis is a continuing engineering effort not necessarily for music but for the resolution of flutter problems in missile and high-performance aircraft.

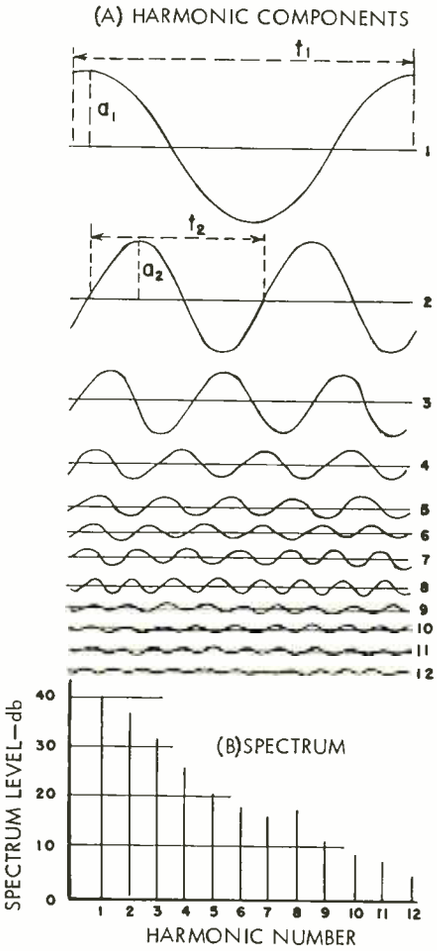


Fig. 4. Harmonic analysis. **(Baldwin Piano Co.)**

I do not know whether harmonic analysis of musical sounds is being done anywhere today as a research project, but if it were, how simply it could be done using high-fidelity techniques. The sound from the source would impinge on a high-quality recording-studio microphone, be amplified in conventional manner, and recorded on tape, for immediate or delayed playback onto an oscillograph. The harmonic coefficients and phase shifts could be determined easily by computor techniques.

Intellectually, we conclude that the theorem of Fourier has shown it to be mathematically possible to consider a musical tone as a sum of simple tones. We do need to enquire as to whether these partial constituents of a musical tone can be reconstituted into a complex wave and will ear hear the original sound. Helmholtz said:[1]

"That this is indeed the case, that this analysis has a meaning in nature independently of theory, is rendered probable by the fact that the ear really effects the same analysis, and also by the circumstance already named, that this kind of analysis has been found so much more advantageous in mathematical investigations than any other."

Formants

As has been mentioned earlier, the harmonic content of a tone is dependent upon the fundamental frequency. For some instruments such as the oboe, bassoon, French horn, and trumpet, the dependence is pronounced. To the extent that this dependence follows a definite pattern the tone color is said to possess a "formant" characteristic.

The formant may be simply defined as a frequency range in which the harmonic components of a complex wave are prominent relative to harmonics at neighboring frequencies. A given tone color may possess more than one formant. *Figure* 7 is an illustrative example of the effect of a formant characteristic upon the spectrum of a tone of variable fundamental frequency.

Although the tone spectrum of an organ stop or of a single instrument will vary considerably with fundamental frequency, there are large differences between the spectra of various stops or instruments for the same fundamental frequency. For tones in the octave range above middle C, for example, the spectra shown in *Fig.* 8 for flutes, reeds and strings are fairly typical.

Each successive harmonic of a flute tone wave will be ten db or more lower in level than the preceding harmonic. For higher pitches the higher harmonics are even less apparent. Reed instruments

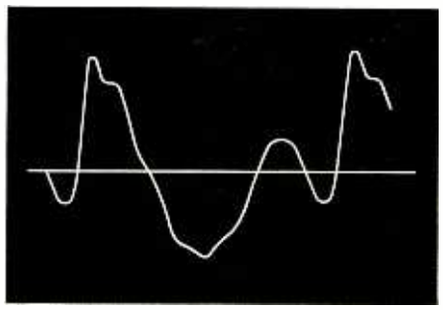


Fig. 5. Harmonic synthesis—proof of harmonic analysis.

and the reed class of organ stops generally have well-defined formant characteristics while strings have a large number of harmonics of comparable intensity. The string spectrum shown is for a tone on a violin E string.

A fine summary of the mathematics of music appears in an article by Sir James Jeans.[8]

Let us return to the amplitudes of harmonics. If we had a bank of sine-wave oscillators available to us, we could key in appropriate frequencies and attenuate them to the harmonic pattern of various instruments. The effect would be frequency addition with each proper harmonic having its proper amplitude. Our electrical addition would recall the technique shown in *Fig.* 1 when we attempted to duplicate a square wave.

In theory, and in practice, the sound that one would hear from a loudspeaker connected to these attenuated and added frequencies would effectively duplicate the original sound.

[8] The World of Mathematics, Volume 4, New York, 1956.

[9] Baron Rayleigh, The Theory of Sound, 2 volumes, Dover Publications, New York, 1945.

ALTEC ALL-TRANSISTOR PLAYBACK *Three Sixty* STEREO AMPLIFIER

The Only All-Transistor Stereo Amplifier Backed by 6 Years of Leadership in Solid-State Circuitry!

"Professionals in the recording and broadcasting industries know that Altec has been making solid-state amplifiers for some time now—for professional use only. The Three Sixty reflects the experience Altec has achieved in making these amplifiers." AUDIO *Magazine, April, 1964*

At a time when most amplifiers were of the vacuum-tube type, we marketed our first all-transistor amplifier. Then, five years later, we presented the *Three Sixty*. In the past year, it has been proved again and again by satisfied users. Because you asked for it, we have improved the styling of the instrument, making it more modern, good-looking, more convenient to operate.

CHECK THESE REASONS. YOU'LL SEE WHY THE ALTEC *Three Sixty* IS SO SUPERIOR.

✓ Altec's wide experience in designing solid-state circuitry for audio frequencies has given us a lead over other companies. This experience made the *Three Sixty* possible. Over five years ago, we designed the first all-transistor amplifier (the 351A) for high quality applications. We also developed the first successful all-transistor repeater amplifiers for use by telephone companies. We also were the first to combine transistors with vacuum tubes in the famous 708A "Astro".

✓ All-transistor circuitry of the *Three Sixty* offers greatest possible durability because, unlike heat-generating vacuum tube amplifiers, it always runs cool. Hence, there's no deterioration of quality caused by heating and cooling of vital circuit components.

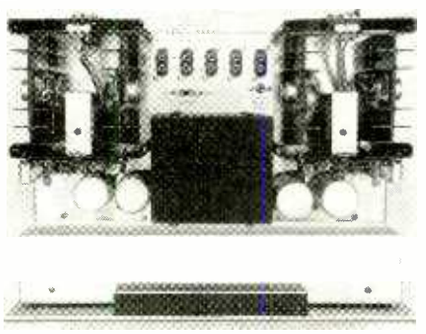
✓ The *Three Sixty* is a genuine Altec PLAYBACK Component. It is part of a line that has won acceptance by leading recording and broadcast studios. There's a world of difference between equipment designed for and used by professionals, and ordinary hi fi components made strictly for home use.

✓ You'll like the looks as well as the sound of the *Three Sixty*. And, its size is smallest of any integrated stereo amp/preamp on the market today. The *Three Sixty* is priced at $389.00; matching cabinet $11.00 extra.

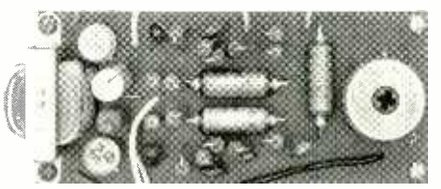
In addition to those shown below, other studio-users of Altec PLAYBACK Equipment include: ABC, Universal Recordings, Columbia, Sam Goldwyn, Glen Glenn, United Walton, and just about every major concert hall, auditorium and theatre in the nation, including all Cinerama Theatres. At the New York World's Fair, some of the exhibitors who selected Altec equipment include I.B.M., Du Pont, Chrysler, Ford, General Electric, Heinz, Cinerama, Billy Graham, Texas Pavilion, Johnson's Wax, and many more.

Visit your nearest Altec Distributor (Yellow Pages) and hear the finest equipment in the world of sound: Altec PLAYBACK Equipment. Be sure to ask for your courtesy copy of Altec PLAYBACK and Speech Input Equipment for Recording and Broadcast Studios. Although prepared specifically for the recording/broadcasting industry, the conclusions to be drawn about your own home music center will be obvious. Or, for free copy, write DEPT. A12B.

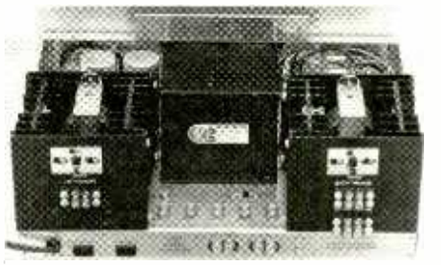
Four extra-heavy heat sinks in Altec *Three Sixty* make possible continuous operation with virtually no rise in temperature. Internal-external heat is kept well below industry standards to guarantee *lifetime* trouble-free, service-free operation. Two power output transistors (four per channel) on each heat sink.



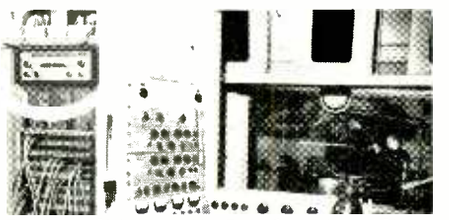
Top view of *Three Sixty*. All-transistor circuitry eliminates hum and hiss common with vacuum tube amplifiers. Features 3 automatic resetting circuit breakers—one in main power circuit, one in each speaker output circuit.



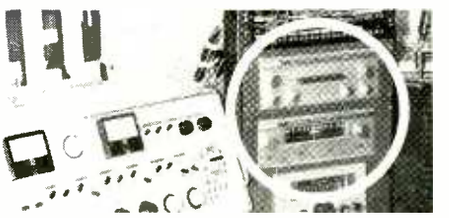
Modular preamplifiers are completely shielded from output and power circuits to assure long life, minimum maintenance. Another Altec exclusive!



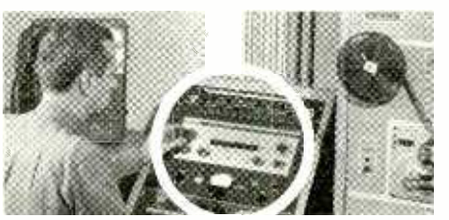
Plug-in facilities provide independent voltage output to drive separate remote power or booster amplifiers (Altec all-transistor 351B) for patio, poolside, recreation room, other remote areas. Impedance selector switch eliminates confusion about multiple speaker terminals.



The *Three Sixty* is used by Dave Sarser's and Skitch Henderson's famous "Studio 3" in New York.



Perfect partners! Rack-mounted Altec *Three Sixty* works with Altec 314A FM Multiplex Tuner at Sim-O-Rama Recording Studio, N.J.



The film production service at Virginia State Department of Education relies on *Three Sixty* for power in conjunction with other Altec PLAYBACK amplifiers and controls.

ALTEC® LANSING CORPORATION

ALTEC LANSING CORPORATION

LTV *A Subsidiary of Ling-Temco-Vought, Inc.*

ANAHEIM, CALIFORNIA

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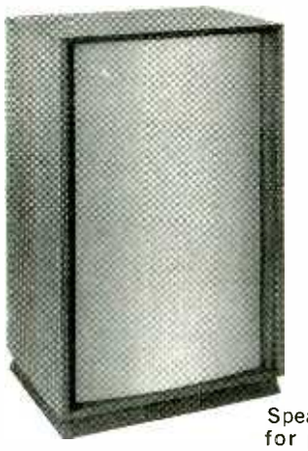
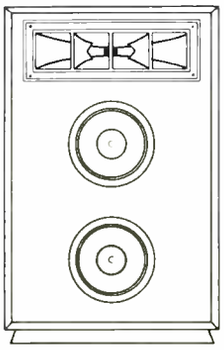
THIS MUCH FLOOR SPACE

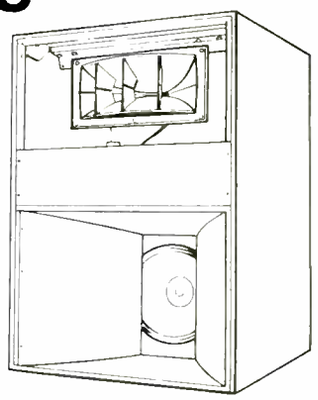
FOR THIS MUCH MUSIC

. . . with Altec Full-Size PLAYBACK Speaker Systems





ALTEC 843A "MALIBU" contains the same speaker components as the new 844A Monitor & PLAYBACK Speaker System (shown at far right) for professional recording and broadcast studios: a pair of low resonance Altec 414A bass speakers, a cast aluminum sectoral horn powered by an Altec 804A high frequency driver, and a two-section dividing network. Dimensions: 40" H, 25" W, 18" D. Price: $356.00 in Walnut. Low-boy model—the 838A "Carmel"—is also available.



"VOICE OF THE THEATRE"® Speaker Systems now come fully clothed, ready for your home or high quality applications in public places where both styling and excellent sound are the goal. Available as the A7W or A7-500W models, these are the identical PLAYBACK speakers used by leading recording studios. Dimensions: 46" H, 30" W, 24" D. Price: A7W Speaker System, Walnut Finish—$384.00; A7-500W Speaker System, Walnut Finish—$411.00.

NOTE for do-it-yourself decorators and recording engineers: The A7 and A7-500 are available as usual in their economical utility cabinets at $288.00 and $315.00 respectively.

ENJOY SOUND WITHOUT COMPROMISE WITH THESE NEW FULL-SIZE PLAYBACK SPEAKER SYSTEMS FROM ALTEC:

These new Altec PLAYBACK speaker systems contain all of the elements that are essential to give you *no-compromise big sound*. Each is large enough to hold a low-cutoff sectoral horn which permits the simplicity of a two-way system with a single crossover. Use of a 90° horn provides perfectly controlled, wide angle dispersion of both the *mid and high frequencies* to achieve *big sound*. This subject of "big sound" is fully covered by both proponents in THE GREAT DEBATE, mentioned elsewhere in this advertisement.

Both the 843A "Malibu" and the "Voice of the Theatre" Systems are full-size, floor-standing PLAYBACK units with impressive cabinets in walnut. They are styled to do credit as an impressive furniture piece in any living room. In fact, these are loudspeakers that you can display proudly... and listen to by the hour.

Circle 120 on Reader Service Card

NEW FULL-SIZE PLAYBACK® SPEAKERS FROM ALTEC NEED ABOUT 3 SQ. FT. OF FLOOR SPACE TO GIVE YOU NO-DISTORTION MID-RANGE WITH LOWS & HIGHS TO MATCH

THE ALL-IMPORTANT MID-RANGE

Almost any good speaker has good lows and highs because so much attention has been given to these extremes of the frequency spectrum in recent designs. *But very few speakers have really good mid-frequencies.* Yet, it is the mid-range that holds the primary attention of the recording engineer because *this region embraces 90% of all musical material.* Most fundamentals and all of the rich lower harmonics are in this critical range. It is the meaty part of music and is essential for life-like reproduction.

When you judge one of the new Altec PLAYBACK speaker systems through A-B comparison listening tests, we urge you to especially notice their clean, no-distortion mid-range. Their smooth, no-distortion reproduction in this region makes a subtle, though readily discernible, difference – a difference that explains why so many major recording studios depend on Altec PLAYBACK speakers for monitoring and playback in a continual comparison of the live rendition to the freshly recorded version.

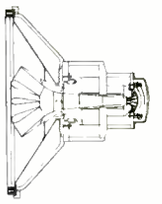
While listening, ask to hear a full orchestration of many pieces performing through a wide dynamic range. *This is the acid test for good mid-range.* It will quickly expose any existence of "mid-range muddiness"—a distortion which has crept into many speakers of recent design due to the attention concentrated on highs and lows, with little or no regard for the mid-range.

THE GREAT DEBATE ABOUT BIG VS. LITTLE SPEAKERS

As was inevitable, the controversy about big vs. little speakers had to be settled sooner or later. Now, the tiresome argument is over, with expert proponents stating the case for each side. We're of course referring to "THE GREAT DEBATE" which appeared in the August issue of *HiFi/Stereo Review*, titled "IS A GOOD BIG SPEAKER BETTER THAN A GOOD LITTLE SPEAKER?". If you haven't yet read it, just let us know and we'll gladly send you this reprint giving both sides.

Not surprisingly, we were asked to speak up for the affirmative – that a good big speaker is indeed *much* better than the best little speaker. We are certain that if you want the best there is in musical reproduction you will give up some floor space for our good full-size speaker systems. Write Dept. A12.

ALTEC LANSING CORPORATION
A Subsidiary of Ling-Temco-Vought, Inc.
ANAHEIM, CALIFORNIA



THE 604 "DUPLEX"® IS BACK! The most famous single speaker in history of high fidelity is back, packed with all the new engineering knowledge that has been acquired since its original design two decades ago. The new SUPER "Duplex" 604E is an updated version of the original and famed 604A, B, C, and D Models (you'll find more of these speakers still in use in quality recording and broadcast PLAYBACK and monitoring than any other speaker ever made).

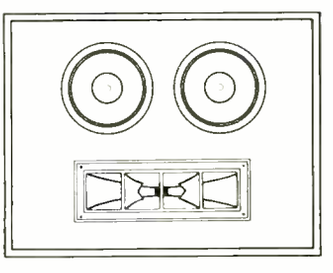
The SUPER "Duplex" offers highest efficiency like all Altec speaker systems with full capability of reproducing the entire dynamic range of music with today's medium-power transistor amplifiers. Also check the 604E for purity of mid-range, exceptional attack time, and no-distortion 20-22,000 cycle frequency range. With a dual magnetic structure that *weighs 26 pounds, 13 ounces,* the SUPER "Duplex" 604E is the most efficient speaker offered to the home music market. Price: $199.00 including two-section dividing network.

For optimum performance, we recommend the "Malibu" furniture-styled enclosure for the SUPER "Duplex". It is available as the 855A Cabinet and comes with pre-cut baffle for easy installation. The 855A is priced at $126.00 and is also recommended for use with any other 15" Altec speaker.

TAKE A CUE FROM THE RECORDING & BROADCAST STUDIOS: SELECT A NO-COMPROMISE SPEAKER SYSTEM

Professionals in sound – people whose careers as performers, directors, and recording engineers depend on the quality of their equipment – have for years relied on Altec PLAYBACK equipment in their studios. In fact, in the days before the term "hi fi" was ever coined, Altec was already producing studio-quality PLAYBACK components. And, as another fact, high fidelity as we know it today was born right in those same recording, broadcast, and motion picture studios.

You can bet your bottom dollar that the studio professional not only expects, but knows where to get sound quality that approaches the "live"... and no compromises tolerated. Perhaps that's why so much of our income comes from the professional and commercial sound industries. Here's an example of our latest design for the professional market:



NEW! SPECIFICALLY FOR RECORDING & BROADCAST USE: STUDIO VERSION OF THE "MALIBU" & "CARMEL" • Designed especially for recording and broadcast studios, the 844A Monitor & PLAYBACK Speaker System contains the same speaker components as the 843A "Malibu" and 838A "Carmel". Comes in studio grey cabinet with sectoral horn mounted below the low frequency speakers so that the unit may be mounted above the observation window in studio control rooms. Dimensions: 24" H, 31" W, 16" D. Price: $327.00.

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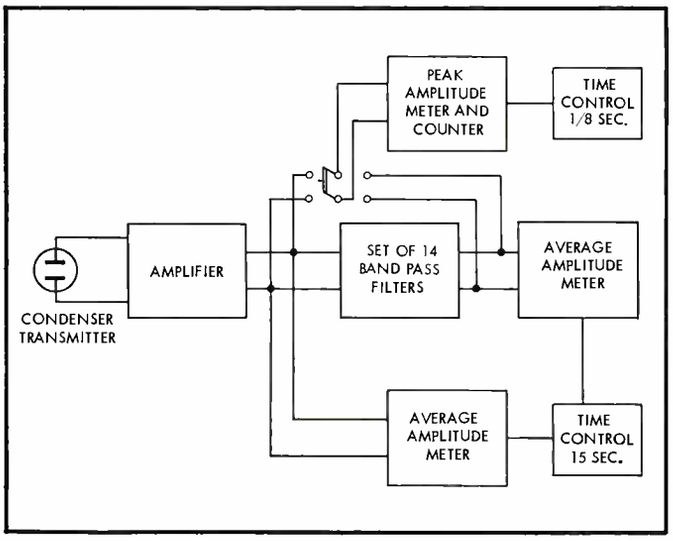

Fig. 6. Circuit for measuring peak and average power.

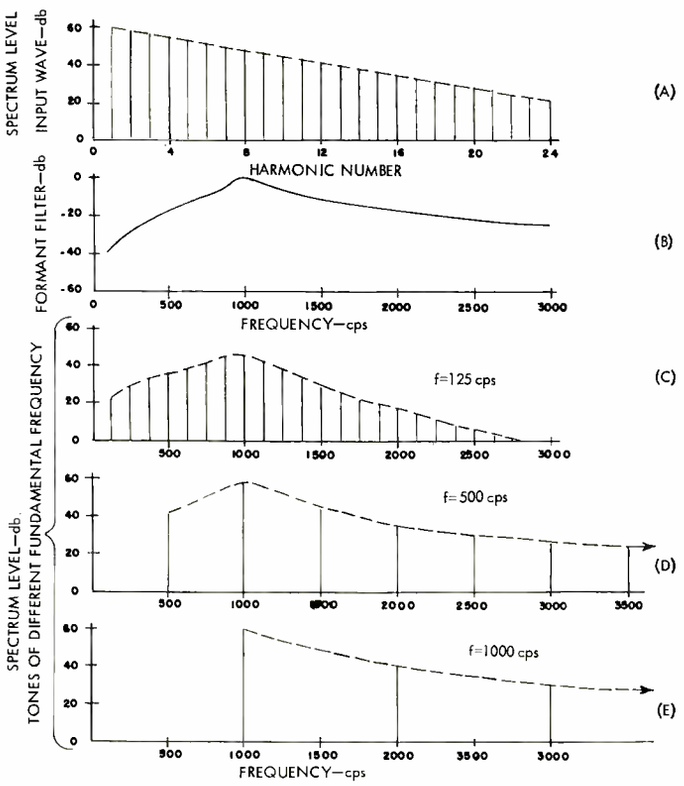

Fig. 7. Formant effects on tone spectrum. **(Baldwin Piano Co.)**

If we were bold experimenters, we might try changing the relative worth of the harmonics and we might even add the 7th of the fundamental or the 5th of the 3rd harmonic. The Hammond organ is built on this principle. The drawbars are potentiometers which control the attentuation of the appropriate harmonics and partials. *Figure* 9 shows graphically the frequency addition that takes place in the Hammond circuit.

The Hammond scheme provides the musician with great flexibility. However, one might like to have the formants or harmonic patterns preset by internal wiring so that engaging a key would pick up the entire harmonic pattern. These we would call stops.

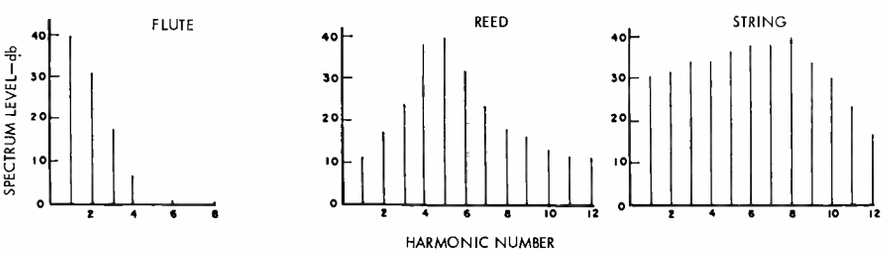

Fig. 8. Spectra of various instruments. **(Baldwin Piano Co.)**

The Conn and Allen organs also depend to an extent on additive techniques. The electrical addition at frequencies is accomplished by well designed mechanical switches, which are quite similar to telephone crossbars. *Figure* 10 shows a Conn switch.

Such techniques are extremely useful in generating flute sounds whose waveforms simulate sine waves. The clarinet and stopped organ pipe sounds which sound hollow and woody contain a minimum of even harmonics. When these stops are required, the switching mechanisms do not pick up the even harmonics.

Filters

The passage of an alternating electric current through a long line or cable is

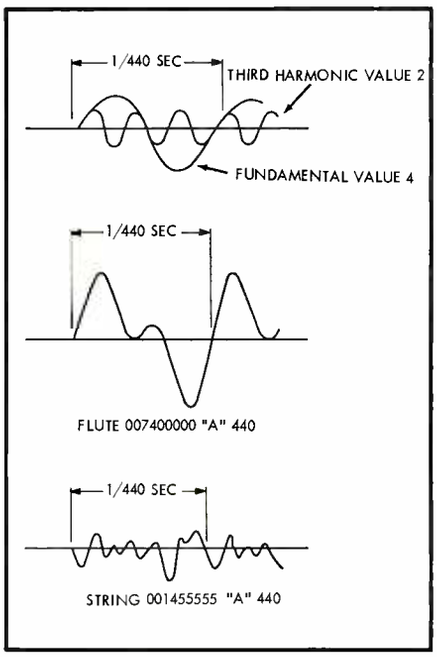

Fig. 9. Oscillograms demonstrating Hammond tone addition.

usually accompanied by waveform distortion. This effect is particularly noticeable in the transmission of complex waveforms. Heaviside showed that this distortion was due to the distributed electrical capacitance of the line filtering out certain components of frequency from the wave during transmission.

In 1924, G. A. Campbell at Bell Laboratories extended Heaviside's theory and showed how it was possible in practice to distribute inductance and capacitance in electrical transmission lines so that certain selected ranges of frequency are transmitted with negligible attenuation, while other frequencies are suppressed almost entirely.

In the simple form of the theory, the transmission line is regarded as a repeated network or chain of impedances arranged in series and in shunt. From this we derive the low-pass, high-pass, and band-pass filters needed in elec-

Meet the new Royal Grenadier . . . . . world's most perfect speaker system. Pretty soon every stereo system 'round will be featuring this revolutionary divergent lens speaker system. The first loudspeaker ever designed and engineered for stereophonic reproduction. Lets you sit anywhere—hear everything.

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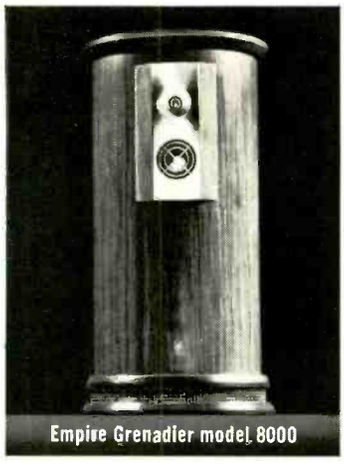
Years ahead in design and engineering the Grenadier projects a majestic sound unlike any you've heard before. Its cylindrical shape creates a system relatively free from room standing waves and approaches acoustically flat frequency response. Sound level and tone remain constant virtually anywhere in the room. Its three divergent acoustic lenses achieve unparalleled stereo separation. With the Empire Grenadier . . . speaker placement becomes non-critical.

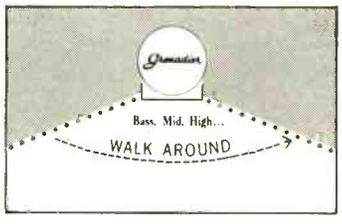
Model 9000M outstanding features:

1. 15" mass loaded woofer with floating suspension and 4" voice coil.
2. Sound absorbent rear loading.
3. Die-cast mid frequency-high frequency full dispersion acoustic lens.
4. Hand rubbed satin walnut finish.
5. Imported Italian Perlata marble.
6. Ultra-sonic domed tweeter.
7. Full presence mid range direct radiator.
8. Exclusive non-resonant rigidized heptagonal sonic column.
9. World's largest (18 lbs.) speaker ceramic magnet structure.
10. Front loaded Horn–360° aperture throat.
11. Complete symmetry of design with terminals concealed underneath.
12. Dimensions: height 29" — diameter 22".



Empire Grenadier model 8000

Started a new era in speaker systems. Measures 29" high with a 15¼" diameter. Its features are virtually the same as the 9000 plus the exclusive Empire Dynamic Bass Reflex . . . high Q reflex tuned columns for in-phase low frequency reinforcement. The scientifically accurate gradients and vented ports provide unbelievably enriched base response.



Try this simple test.

You will notice no change in sound level of bass, mid range, and highs. Full frequency and separation is assured by Empire's exclusive divergent acoustic lens system.

Try this same test with any other brand of speaker. Some speakers will only have a narrow angle of high frequency sound propagation. Some may have 2 or even 3 bands of high frequency sound. With these or other speakers, slight shifts of position, turning one's head, or even leaning to one side may cause sharp changes in the listening tone and level. Not so with the Empire Grenadier.

Acoustically engineered to let you sit anywhere — hear everything. The Empire Grenadier is decorator-designed to fit any decor . . . from warm elegance to stark modern . . . fit in corners or against walls.

Its satin walnut finish is designed to blend with all furnishings. An imported Italian Perlata marble top is optional for added elegance on the model 9000. The Empire Grenadier is a truly beautiful and functional achievement in sight and sound.

For a sound demonstration of the Empire family of "most perfect" products, go 'round to your dealer or write for complete literature.

EMPIRE

"World's Most Perfect High Fidelity Components"

The incomparable Troubadors —

The model 498 — tailor-made for console or equipment cabinets . . . the famous Empire 398 — outstanding — too handsomely finished to hide behind cabinet doors. High Fidelity reports on the Troubador: ". . . precision engineered product of the highest quality . . . one of the finest, handsomest record players available."

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Audio Magazine stated "...truly excellent...the finest cartridge tested." Frequency response 8 to 30,000 cps. Compliance 20 X 10—6cm/dyne. Empire 880pe comes with a biradial elliptical hand polished .2 X.9 mil diamond.

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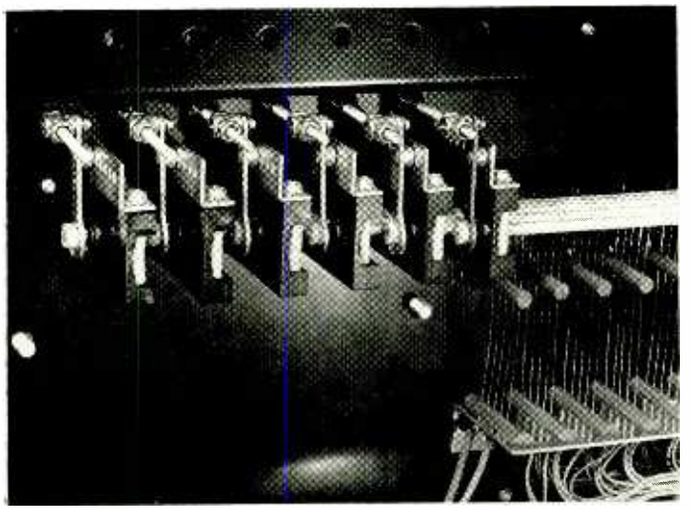


Fig. 10. Keyboard assembly. Left end of Great Manual.

tronic organs. A rigorous proof for filter theory and m-derived filters is given in "Communication Engineering."[10]

We have been discussing two seemingly unrelated subjects, the first was harmonic analysis and the second filter design. Whether the energy distribution is determined from the Fourier coefficients (either numerical or mechanical integration) or by direct measurement it is possible to draw the "frequency output curve." Let us design a filter whose frequency response curve is identical to this frequency output curve (the formant).

Since the filter is made up of resistors, capacitors and inductors, we should be able to change these at will. We can do this analytically or we can actually twiddle dials or solder different components into this circuit.

Now we will impress the output of a sawtooth oscillator upon this filter. As we turn dials, our variable filter may convert the harmonic-rich sawtooth to unpleasant sounds. Some of our experiments may generate sounds more pleasant than the original.

So, while the first approach to electronic organ sounding may be duplicate air-organ sounds, the consequence of our research has been to free us to generate air-organ-like sounds which are beyond the bounds of air-organ design.

There are other classes of organ tone color which require special filtering since they depend on harmonic suppression. They are the diapason family, and "stopped" tone colors. Both of these classes have alternately prominent harmonics rather than groups of strong adjacent harmonics. The even harmonics, especially the second, are more prominent in diapason tone. These harmonics are omitted in additive-design organs. In filter type organs, the removal of alternate harmonics requires ingenuity.

In electronic organs such as the Baldwin and Schober, production of these tone color classes is accomplished by addition and subtraction of the proper amount of complex wave of twice fundamental frequency. *Figure* 11 shows the principle of subtraction (adding out of phase) which is used for stopped tone colors. The spectrum of waveform A containing a complete series of harmonics is compared to that of a waveform A + B where B has twice the frequency of A, half the amplitude and opposite phase. Spectrum A + B has the even harmonics missing.

Electronic organs using fixed-frequency tone generators are divided into two classes: formant, and tone synthesizing. We can now see that there are merits to both designs. Some organ manufacturers utilize both within one instrument; the Allen Sheraton Model, the Schober Concert Model. The organ using sine wave additive technique must also include circuits for differentiating and shaping the sine wave in order to get rich harmonics which are too numerous to obtain by addition. Organs depending on sawtooth waveforms as the basic wave shape must have additional circuits for obtaining a sine wave from a sawtooth in order to generate flute tones. This is done through integrating circuits.

It is a matter of personal taste when comparing the synthesizing techniques of the Hammond or some Allen instruments to filtering techniques of a Baldwin or Conn organ. The filter organ provides an array of filters which can be shunt simply and directly to the inputs of the preamplifiers.

In tone synthesis, complex switching arrangements are required to pick up the appropriate harmonics coming from many tone generators. Frequently, the harmonics must be appropriately attenuated in order that the proper tone color can be effected.

We see now that some instruments such as Hammond do not try to "imitate" air-organ sounds; the second group such as Baldwin, Conn, Allen, Wurlitzer, Gulbronsen and others provide tones which are analogs to air-organ sound. In addition, these organs have sounds which their designers consider to be improve-

(Continued on page 68)

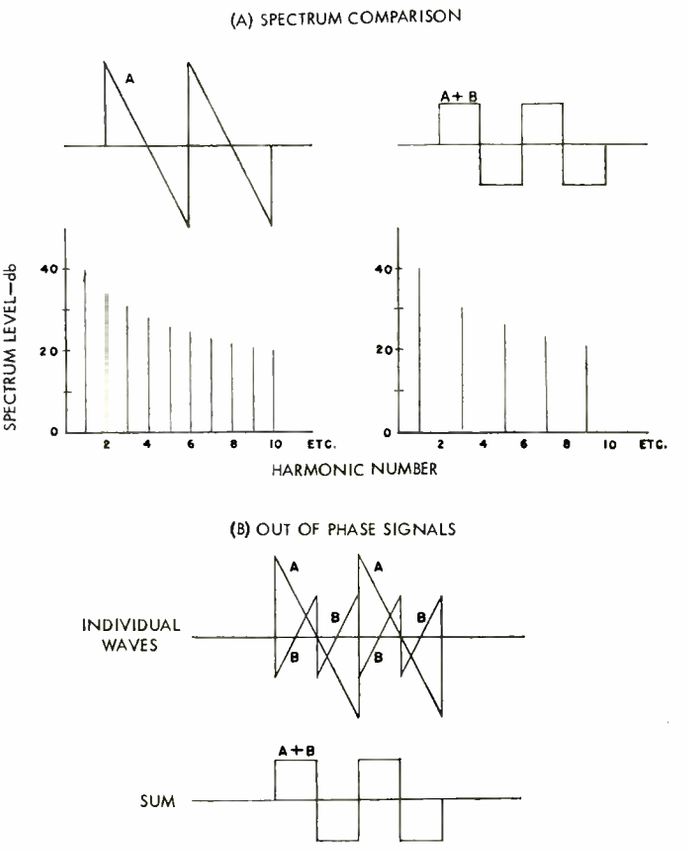
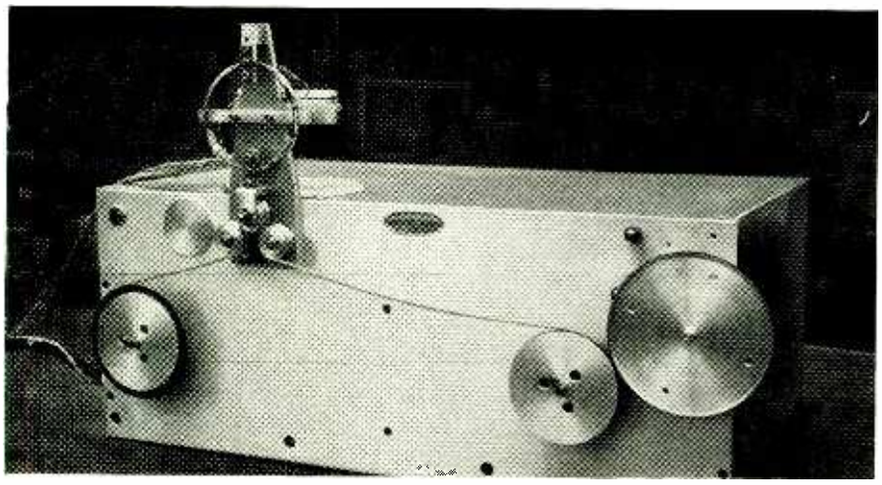


Fig. 11. Use of out-of-phase signals to synthesize desired wave form. **(Baldwin Piano Co.)**

[10] W. L. Everitt, Communication Engineering, McGraw-Hill, New York, 1937.

Some plain talk from Kodak about tape:

physical testing and tape performance



The High Speed Tensile Tester is designed to break tape under load . . . and gather a lot of useful data besides.

Magnetic tape is subject—day-in, day-out—to a wide variety of stresses and strains. That's why we are more than casually interested in its tensile properties. Tape is much like a rubber band. Put under tension, it will stretch. When the tension is released, it will snap back to its original shape. It will, that is, unless you've stretched it beyond its yield point. For if over a certain amount of longitudinal stress has been placed on a tape, the tape will lose its ability to recover and will, in fact, remain permanently elongated. Stretch it even further and, naturally, the tape will break.

Deformed tape will not reproduce sound faithfully. And tape that breaks too easily is just a plain nuisance. So we set our sights high and developed a special triacetate—called Durol base—that's exceptionally tough, yet breaks clean without "necking down." In order to prove its worth *and keep it that way*, we developed a tight set of specs for our quality-control boys—specs which were a direct outgrowth of the conditions under which a tape is to be used. For example, the shock of going from fast wind to fast rewind. Or the shock generated on a running recorder when the supply spool jams.

We think that tape should be able to take this sort of punishment routinely. So, of course, we double, triple, even quadruple the requirements! And just to make sure that the tape performs we build torture tests that would have delighted Attila the Hun.

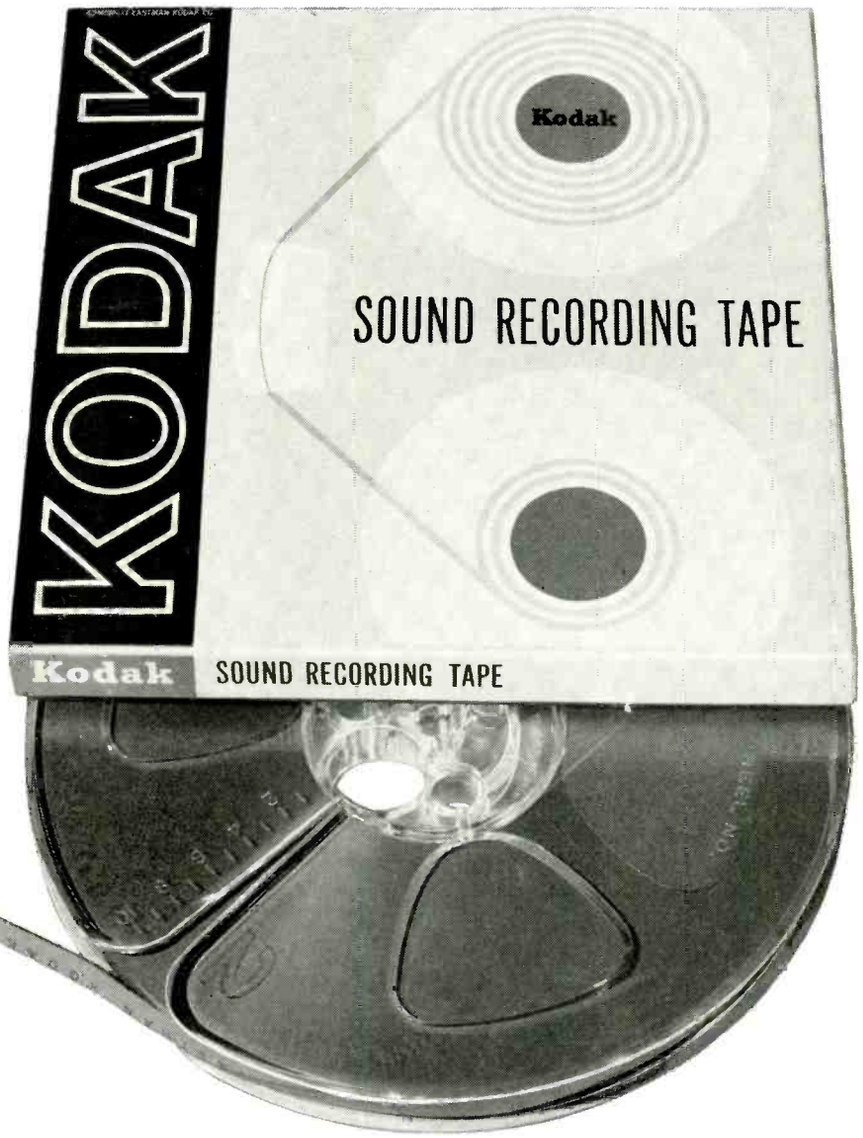
Here is one tester that is outstanding in its fiendishness. It's called the High Speed Tensile Tester and is designed to break tape under load.

But like any good one-man band, it does a lot more than just one job. It not only breaks tape but gathers scads of very useful data as well—data which completely describes a tape's tensile properties. Here's how it works. It's built like a tape deck with the tape attached to one half of a split-ring electrical strain gauge. We run the deck and then jam on the brakes on the supply reel but keep the take-up reel going. The strain gauge takes the full load and the split ring spreads and deforms. This deformation causes the gauge to change resistance and causes the DC voltage on it to pulse. We monitor the pulse on a scope and measure the duration. This gives us a figure of merit in terms of tensile strength.

Just how good is Durol base? Well, consider this data. Yield strength for Durol base is 47% greater than regular triacetate and 70% greater than diacetate (the two most common plastic support materials). Break strength is 43% greater than triacetate and 80% greater than diacetate. And this is the kind of test that almost duplicates actual use conditions on your tape equipment.

But any torture test one engineer can devise, another engineer can improve upon. Take the Toughness Tester, for example. This is an instrument designed to determine a tape's strength (toughness) by measuring the force required to break a sample. A measured length of tape is held securely between two clamps. Then it is struck and broken by a falling pendulum. Because it has been raised to a fixed height, the

pendulum always delivers a precise and repeatable amount of impact. The energy absorbed by the tape at impact controls the height of the pendulum's backswing. Thus, a measurement of backswing height is a direct measurement of toughness. The strain rate that this device imposes is on the order of magnitude of 200,000% per minute—enough to break any acetate-based tape. How does Durol base compare to conventional acetates? Well, it comes through this test, too, like an Olympic star. In test after test, Durol base proves to be about 40% stronger. This toughness test also provides a valuable measure of permanent elongation. Durol base's unique "shear-pin" action lets it break clean with minimum elongation (less than 1% compared to 10% for other acetates). These are only two of the more interesting physical tests routinely performed on random samples of Kodak tape. There are dozens more, of course. And we haven't even gotten into electronic testing yet. But we'll save those for another day.



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A Basic Course in Commercial Sound

NORMAN H. CROWHURST

Chapter IX

Output Matching.

A few minor points need clearing up that were crowded out of the last installment. The broad principle of output matching was covered in Chapter 5. A feature that can be helpful is the provision of multiple output taps on the amplifier. Smaller amplifiers, up to say 25 or 50 watts, will provide the same kind of output taps usually found on a high fidelity amplifier, 4, 8 and 16 ohms. This facilitates connecting various numbers of 16-ohm speakers directly.

Larger amplifiers are designed for constant-voltage line operation. Theoretically, they only need one output pair, to provide the needed 70 volts nominal, at an impedance to correspond with the power rating. But it is an asset, especially in portable systems, to have extra tappings for other voltages, particularly lower ones, so the amplifier can have optimum power output where the total rating of connected speakers and that of the amplifier do not happen to agree as conveniently as may be expected.

Provision of multiple taps also facilitates varying the power delivered to individual speakers in different locations. However, this increases the demand on wiring (*Fig.* 9–1). A more direct way to make such adjustment possible is by means of multiple tapped-line transformers. These are available from several manufacturers nowadays. However, changing the amplifier tap is a convenient way of varying the relative power sent to whole lines of speakers.

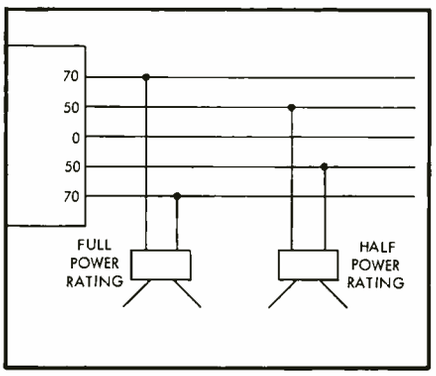

Fig. 9–1. Using different amplifier taps on constant-voltage line type output as bus connectors for varying feed to different speakers, or groups of speakers.

Stereo Installations.

Sometimes the customer will question whether stereo is feasible, because he fancies the notion. At other times, the provision of stereo may be advisable, to achieve something that would be more difficult monophonically. The questions to be answered in deciding on a method of achieving the desired end in either case are similar. To be effective, stereo must achieve its objective, of providing separate "sources" for different parts of the composite sound presented, otherwise it may only add to confusion bad enough already, in mono!

How is the stereo effect achieved? By intensity differences between channels (or between separate sound sources) or by time differences between them? Or is it by a combination of both intensity and time differences? If so, which is more important in the evolving of a satisfactory system?

The fact is that both contribute, at different component frequencies, and also (which is not always realised) according to environmental circumstances. In a small room, with little reverberation, intensity differences can never be very great from point to point, so human hearing becomes acutely sensitive to minute time differences in this environment, particularly at lower frequencies (below about 1000 cps). At higher frequencies (above about 2000 cps) the obstacle effect of the human head creates an intensity difference, to which the hearing faculty pays more attention, because time is indeterminate when such small waves are involved.

In larger rooms, reverberation becomes more dominant, whether we want it to or not. We may be using stereo, in effect, to reduce the apparent effect of reverberation, but still its presence will change the sensitive factors on which hearing relies. To ignore reverberation in interpreting sound heard, the hearing faculty now pays more attention to the leading edges of sound wave patterns, and much less to the sustained "follow

through'' tones. But variation of distance, due to different audience locations relative to speakers for the separate channels, can interfere badly with intended timing effects. So the important thing will usually be to control relative loudness from the respective channels, so that the desired channel is only heard from the desired side.

In a long hall, the method is simple: feed left channel down the left side and right channel down the right side (*Fig.* 9–2). Whether stereo presentation is better than delay treatment of monophonic program for this kind of situation depends on both the type of hall (its reverberation characteristics) and on the type of program. Stereo can seldom do very much for presentation of a single speech. For discussions it is much more helpful. It helps the audience identify who is speaking and, if more than one person speaks at a time, it helps the audience resolve the confusion of sound this tends to create.

Stereo may also be desired for the presentation of musical entertainment, either from records or from live performance. Again, the left-right method would be good for the long hall. But what about rooms that either spread out the wrong way, or are of more complicated shape? Then it requires careful thought about what will be the best way to achieve the objective.

In a large, spread-out area, such as a restaurant, especially if this has a fairly low ceiling, the best way may be to install speakers as uniformly as possible throughout the ceiling surface. With mono, this creates no particular problem of connection: they all radiate at low level and the audience is often unaware of any sense of ''source''; sound just fills the room pleasantly. But to get stereo, so all the audience can appreciate it, requires that alternate speakers be connected to

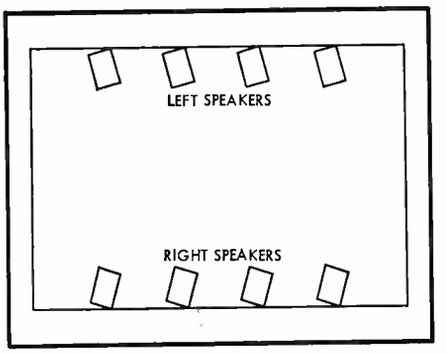

Fig. 9–2. A long narrow hall is simple for stereo installation.

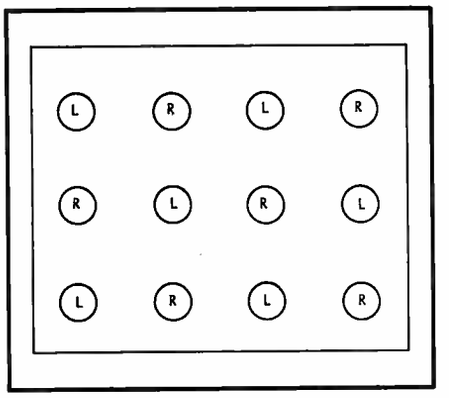

Fig. 9–3. Ceiling plan for connections to get a form of stereo in a large, spread-out floor arrangement with low ceiling, such as a restaurant.

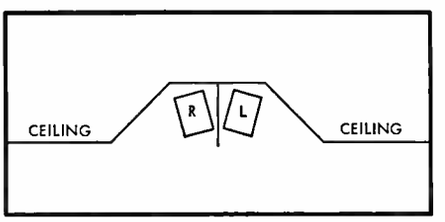

Fig. 9–4. Modification to the ceiling, where proper sound location is needed in a large-floor, low-ceiling installation.

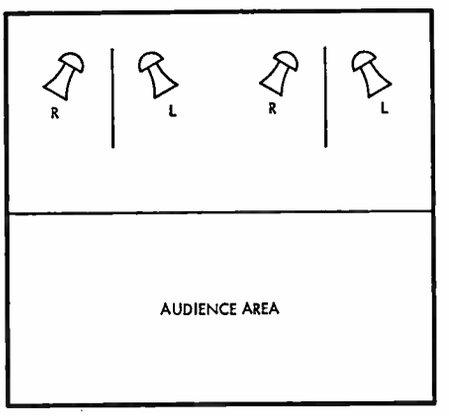

Fig. 9–5. Using pairs of projector horns in an arena installation to achieve stereo illusion.

alternate channels, so that, whatever pair of speakers an individual listener may happen to be between, one is ''left'' and the other is ''right'' (*Fig.* 9–3).

Such an arrangement may provide good separation, but it is no good if specific identification of source direction is required. For music, this means one cannot present the type of demonstration which says, ''the violins are now playing from the right,'' because they may appear to be to the right for some listeners and to the left for others. Also it cannot be used for live relay, because the natural association, based on positions visually observed, will be violated by the sound at some listening locations.

One method of overcoming this is to direct sound by some means, so each speaker can only be heard from the side for which it is intended (*Fig.* 9–4). In a large arena, the same kind of treatment will usually be desirable (*Fig.* 9–5). Note that baffles or barriers in this usage need not act as observers. The system will be more efficient if they work as reflectors.

A particular kind of installation where the term stereo is often abused occurs with juke boxes. Most of the juke box chains now make available machines to play the new stereo discs. The average installer tends to put the speakers at opposite ends of the establishment, which may well be so far apart that few listeners stand any chance of hearing both speakers at once (*Fig.* 9–6). A listener nearer either speaker will only hear that speaker, while a listener nearer the middle will hear more of local conversation and gossip than of the juke box stereo!

Restaurants are not the easiest places to locate speakers in, at the best of times. But to get true stereo they are most often next to impossible. However, a proprietor likes to know he's getting what he's paid for. If the juke box concessionaire has sold him a stereo system, he's disconcerted to have a local stereo enthusiast come in and tell him

*names on request

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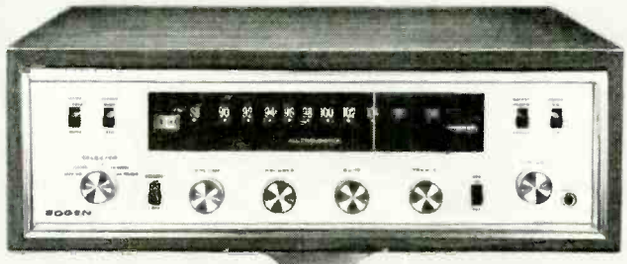

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Whether compared to vacuum tube or solid state receivers, the FM-Stereo 60-watt RT6000, is the "dream receiver" come to life. Solid state power for precisely defined, truly transparent sound; unlimited control versatility; response above and beyond audibility; extreme FM sensitivity; reliability—on every standard, a consummate *professional*. And only Bogen—with its ten-year head-start in solid state technology—could have produced it at such moderate cost.

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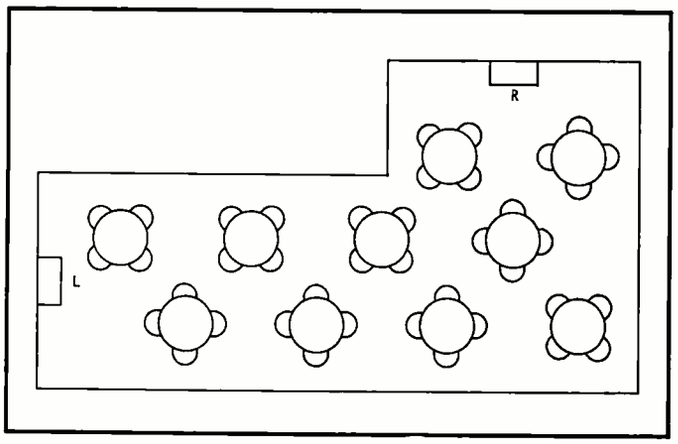


Fig. 9–6. A poor installation sometimes encountered in diners and restaurants that are supposed to have "stereo"!

it doesn't sound in the least like stereo! So what can he do, he asks you?

On this it is difficult to formulate any fixed rules or suggestions. Every installation will pose different arrangements of the same or similar problems. The only thing to do is to walk around the area to be served and try to judge how much of it can reasonably be served with pairs of speakers so that a satisfactory stereo illusion is obtained. Probably a useful device here, if it can be used, is the planar type speaker, in which the diaphragm is flat and larger than the usual cone speaker, and the speaker can be placed edge-on, out in the audience area.

With this arrangement, each speaker creates a sound field around it, whose intensity does not appear to originate from the unit itself, except maybe at a distance. Uniform low-level distribution is possible, without sound becoming inaudible at greater distances, and the combined fields work together to give good stereo illusion almost throughout the room. Suitable choice of complementary positions for the stereo units can result in a very satisfactory system. A good way to provide locations is the construction or suitable placement of decorative dividers of the open type.

Church Choirs.

Here is another opportunity, that many clergymen will appreciate. Nothing is more conducive to the building up of larger congregations than impressive religious music. Small choirs in large churches always seem somewhat inadequate. A public address system, with provision for phono or tape, can enable the humblest church to present the best selections of religious music in stereo, which can be a wonderful experience, if the system handles it well.

Here again, the planar speaker is a useful adjunct, if the acoustics permit. In larger churches, column type units may be necessary to avoid feedback when the system is used with mikes, either for the sermon or for a live choir. For a live choir, directional mikes are a must, because long range pickup with a bad feedback problem means omnidirectional types are virtually useless.

A good comprehensive tone control is also essential, because of the highly colored acoustic properties inherent in most churches. Working a sound system close to feedback point, or even quite a little below it, will exaggerate these acoustic deficiencies, so that what previously appeared to be characteristically the tone of a church ("cathedral quality"), becomes something almost unbearably grotesque. Unfortunately, you cannot 'blame' this on the acoustics, because the layman is not equipped to understand your quite correct explanation. All he knows is that he liked the sound without your system, except that sometimes he couldn't understand what he heard, while with your system it sounds horrible!

The tone control will help minimize the unwanted exaggerations, so the acoustics still seem about what they should be, while the audience can now hear better. This takes skillful and painstaking setting up, but once the controls have been set to suit the acoustics of the particular church, they will not need much changing.

If you have much of this class of work, it may be worth while to set up a fairly comprehensive and elaborate tone control system, which can be used to optimize response and then, from it, design a less elaborate compensation circuit that will achieve the same result, with provision for a little variation, using a much simpler control circuit.

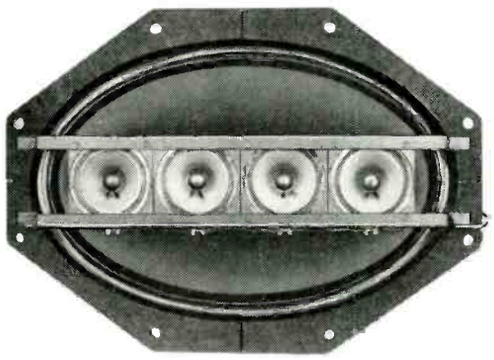
Installation Headaches.

We've covered most of the things that can cause headaches in installations: wrong impedance or level, poor matching, and so on. Sometimes a bad connection can cause difficulties that take time to trace. But most of those things are relatively obvious. The one headache that stands out as a real nuisance is when unexpected hum appears. We'll assume you've checked each item of equipment and found it performs without undue him. Then when you've got the system all connected up, there it is: a snorting hum, fit to kill! The only thing that seems to stop it is switching off the system—admittedly not a practical remedy!

Assuming it is not an ordinary input hum, due to unshielded leads, or mike leads taken over a bad route—which is easy to verify by turning the gain down to zero, which will cut out picked up hum at the inputs—your trouble most likely arises in the system's ground connections. It's either "not

(*Continued on page 56*)

An even more "dangerous" loudspeaker ...with the world's largest elliptical woofer!



THE EMI 901 CONSOLE

This new EMI console Model 901 loudspeaker system is designed with one purpose in mind . . . to provide professional studio sound in a cabinet suitable for the home. The 19" x 14" elliptical loudspeaker has a low, low bass and a particularly smooth mid-frequency response so essential to any good high fidelity system.

The four high-frequency units mounted within the center of the elliptical woofer are designed to give a comparable high quality finish to the bass and midrange sections. The high frequency units at either end have a reduced sensitivity to those in the center . . . to improve the polar horizontal response and limit the polar vertical response. Cast alloy chassis have been used to give a resonance-free response.

All this is required of professional studio sound and this is precisely what you get in the new EMI 901. The hand-rubbed, oiled walnut cabinet is well-constructed and mounted on concealed casters for easy room placement. It measures 34" high, 28" wide and 19 3/4" deep and features EMI's exclusive woven metal grille for superior dispersion of sound. Listen to the new EMI 901 console at your dealer . . . and get professional studio sound for just $395.00* or $285.00* without the cabinet, for custom installation.

*All prices slightly higher in South, West and Canada.

EMI

(Makers of the Dangerous Loudspeakers)

SCOPE ELECTRONICS CORPORATION

235 East 42nd Street, New York, N. Y. 10017

Distributed in Canada by: Hartone Electronics Industries, Ltd., 298 Bridgeland Avenue, Toronto 19, Ontario.

Circle 124 on Reader Service Card

The "Dynagroove" System

HARRY F. OLSON*

In Two Parts—Part Two

An arrangement similar to *Fig.* 5 was used to carry out the project outlined above. The small room was similar in all respects to the typical living room in the home. As a result of the subjective experiments, a Dynamic Spectrum Equalizer was developed, which, in the broadest terms, translates the sonic equalities of the original performance into stimuli which will project that performance into the perception of the home listener with the greatest possible proficiency. A block diagram of the Dynamic Spectrum Equalizer is shown in *Fig.* 13. The system operates in a continuous manner to change the response frequency characteristic as a function of the amplitude. Typical response frequency characteristics for various levels are shown in *Fig.* 14. There is a continuous variation in response from one level to another level. The response frequency characteristics are varied for different types of musical selections. In effect, when the levels are low, the low-frequency components are accentuated. For medium levels there are slight accentuations in the low-frequency region and the presence region of 2000 to 6000 cps and a reduction in response in the region from 400 to 1000 cps. For high sound levels there is accentuation in response in the presence region and a reduction in the frequency range below 1000 cps. When the sound level of the program is low the objective is to raise the sound level of the appropriate frequency regions so that the music can be appreciated under the ambient noise and surround conditions of the average residence. When the sound level of the program is high, the level of the presence region is raised and the level of the low-frequency range is lowered. The procedure does not upset the dynamic balance but rather enhances this aspect of sound reproduction in a small room.

The Dynamic Spectrum Equalizer provides a dynamic alteration of the projection qualities of sound so that under conditions of playback, which differ from those in which the music was performed, the best perception of the qualities of the original performance is obtained. The soft passages of the music

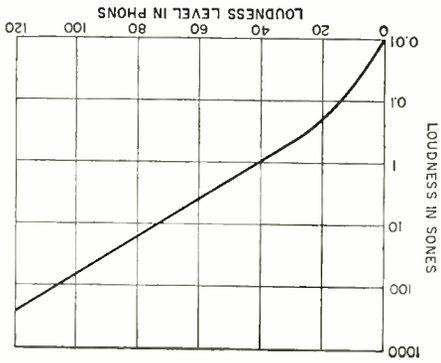


Fig. 12. The relationship between loudness in sones and the loudness in phons.

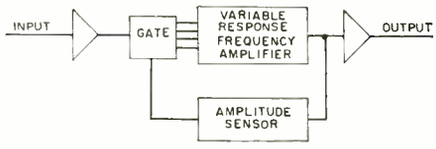


Fig. 13. Schematic block diagram of the Dynamic Spectrum Equalizer.

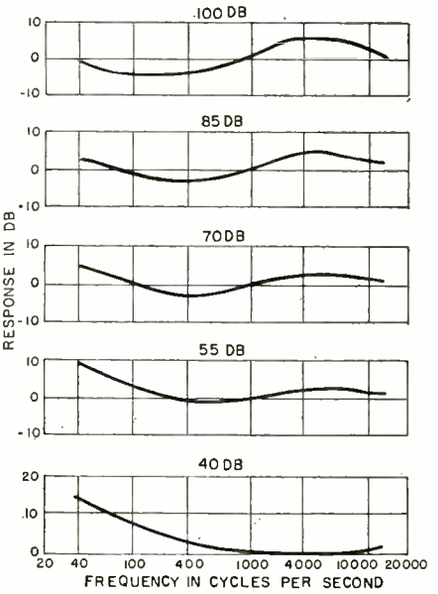


Fig. 14. Response frequency characteristics of the Dynamic Spectrum Equalizer for various sound level of the program.

are given full body and minute detail plus a breadth and expansion which offers the listener a complete appreciation of softness. Throughout the full dynamic range of the performance the music grows in structure and sonority and the change in harmonic content which is so characteristic of every instrument with change in volume is immediately apparent. The loudest sections of the music are projected with extreme intensity. The full realization of this dynamic range, which gives music its dramatic impulse and its sonic structure, is perceived by the listener in his home surroundings with unique impact and intense realism.

Recording Overload Indicator[21]

The Recording Overload Indicator was designed to provide indications of the maximum allowable signal which can be applied in the cutting of the master stereophonic disc record. A schematic block diagram of the recording overload indicator for one channel is shown in *Fig.* 15. The recording overload indicator is provided with two separate indicating meters in each of the two stereophonic channels. One meter is calibrated to show the occurrence of program peaks which will cause curvature overloading[22] which occurs in the high-frequency range. This is accomplished by the use of a differentiator and meter which indicates curvature overload. The other meter is calibrated to show displacement overloading which occurs in the low-frequency range. This is accomplished by the use of an integrator which indicates amplitude overload.

A recording console has been designed for recording the submaster magnetic tape from the master magnetic tape. The submaster recording console contains the Dynamic Spectrum Equalizer and employs the Recording Overload Indicator as well as the auxiliary control equipment for producing the submaster magnetic tape.

The audio monitoring of the recording of the submaster magnetic tape is carried out in a room with dimensions and acoustics similar to those of a typical living room. A sound level meter is used to check the level of the reproduced sound.

* *RCA Laboratories, Princeton, New Jersey.*

[21] The Recording Overload Indicator was developed by R. W. George, J. G. Woodward and E. C. Fox.

[22] L. W. Septmeyer, "A Curvature Meter for Use in Disk Recording," Journal of the Acoustical Society of America, Vol. 19, No. 1, p. 161, 1947.

Mr. Saul Marantz discusses his revolutionary new model 10-B FM Stereo Tuner

Q. Mr. Marantz, your new 10-B tuner is quite revolutionary. Do you feel it will obsolete all other tuners?

Mr. Marantz: In one sense, yes. The performance of this tuner is so dramatically superior to conventional tuners that anyone who wants or needs perfect FM reception today has no choice but to use the model 10-B. Its superiority, however, does not necessarily *obsolete* conventional tuners. Rolls Royce, of course, makes superior cars, but they haven't obsoleted Chevrolets.

Q. Is this superior performance discernible to the average listener?

Mr. Marantz: Very much so. The difference is quite dramatic. As you know, conventional tuners have never been able to pick up and reproduce broadcasts which could match the quality of a fine disc or tape playback system. This has often been blamed on *broadcasting* quality. But the new 10-B disproves this theory. It reproduces the *broadcast* of a disc or a tape with the same clarity and separation as if played through a playback system — proving that broadcast quality is generally excellent.

Q. Is this true with weak broadcast signals also?

Mr. Marantz: Yes. In fact the model 10-B will reach 55 db quieting at only 3 microvolts! This is better than most conventional tuners will reach at 1000 microvolts. With a 25 microvolts station the Model 10-B reaches a phenomenal 70 db quieting which is about 20 db better than most conventional tuners can achieve at *any* signal strength. This means that with the Model 10-B there will be excellent reception even in fringe areas, particularly so because of the tuner's high sensitivity, its extremely sharp selectivity and reduced susceptibility to multipath effects, which on other tuners cause distortion.

Q. How are such improvements accomplished?

Mr. Marantz: The answer to that question is very complex, because the 10-B is far more than an improved tuning system; it is a completely new *design concept* with *many* technical innovations developed by Marantz engineers.

Q. Can you give us some examples?

Mr. Marantz: Yes. The RF section, for example, contains a balanced-bridge diode mixer — a technique used in modern sensitive radar designs to eliminate a major source of noise, harmonic distortion and other spurious interference. The whole RF circuit is balanced-tuned, using a precision tuning capacitor with four double sections, for further reduction of spurious images.

For the critical IF strip, we've developed the first commercial application of the "Butterworth," or phase-linear filter. This new concept provides a number of distinct characteristics essential for good results. The passband, for example, is phase-linear for extremely low distortion — especially at high frequencies — and it remains essentially phase-linear at all signal levels.

Cutoff slopes beyond the passband are extremely steep, allowing unprecedented selectivity; it is much less subject to the effects of multipath, and it doesn't require realignment with tube changes or aging. The old standby coupled IF circuits currently in use do not have any of these characteristics.

Q. Are there any innovations designed specifically for multiplex?

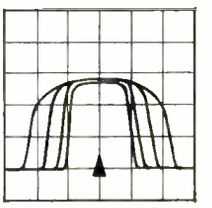
Mr. Marantz: Yes. For multiplex reception we've developed our own unique variation of stereo demodulator, which permits phase correction to maintain a very advanced order of stereo separation throughout the whole audio band.

Q. What is the purpose of the tuning and multipath indicator?

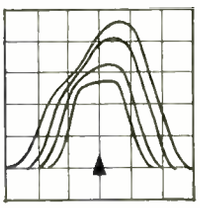
Mr. Marantz: This oscilloscope device is so versatile its single trace tells many easily understood stories. It shows when a station is tuned exactly to the center of the passband. The height of the pattern shows the signal strength. The indicator shows how much multipath is present, making it easy to adjust the antenna for best reception. It shows if the station is creating distortion by overmodulating. Also, technically informed users can check stereo separation of transmissions, discs and other sources.

Q. And how soon will the model 10-B be available in quantities?

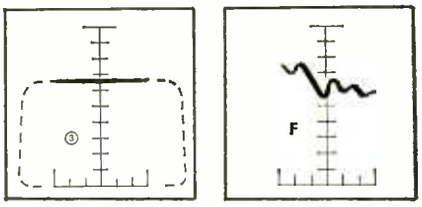
Mr. Marantz: The Model 10-B is a laboratory instrument of extremely high quality which will never be *mass* produced in the usual sense. However, production has been stepped up fourfold and all back-orders are now being filled by Marantz franchised dealers.



IF Passband retains phase linearity and sharp slopes at any signal strength for low distortion, sharp selectivity.



Conventional mutually-coupled IF circuits change characteristics drastically depending on signal strength.



MARANTZ MULTIPATH/TUNING INDICATOR

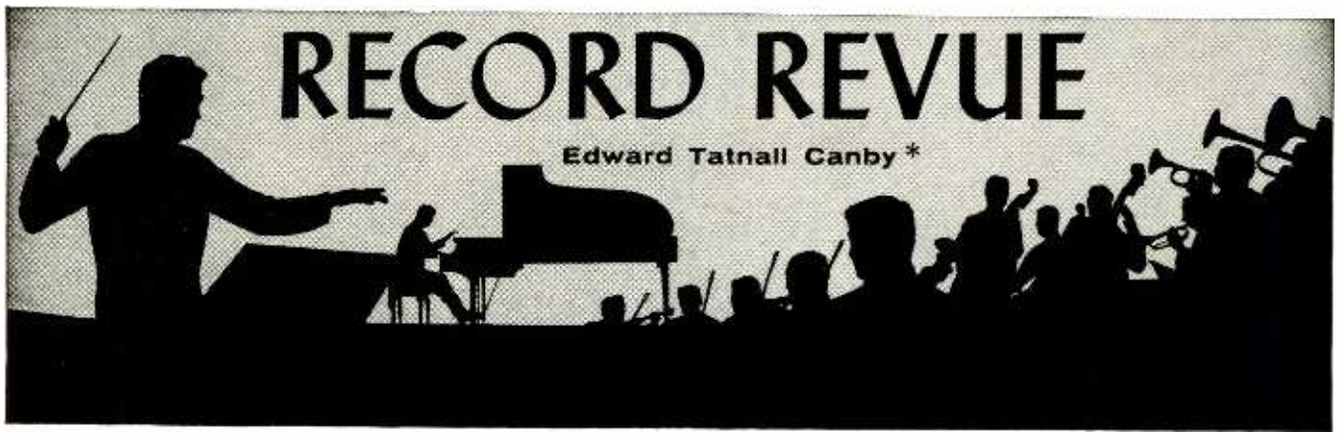
Station tuning is simply and accurately adjusted by centering the trace.

Multipath (Ghosts) shows up as 'wiggles' on the tuning trace. Antenna is simply rotated until trace is smooth.

marantz

MARANTZ, INC., SUBSIDIARY OF SUPERSCOPE, INC., SUN VALLEY, CALIF.

Circle 125 on Reader Service Card



Chopin Waltzes. Artur Rubinstein
RCA Victor LSC 2726 stereo

Old age is so rarely a time of supreme fruition—except, it seems, among musicians! For many long years, Artur Rubinstein was RCA's most brilliant piano wheelhorse (and his concert agent's, too), turning out millions of minutes of steely, self-assured, powerhouse pianism. Frankly, I used to dislike him rather vehemently.

No longer. Where once—if I can believe my memory—Rubinstein's Chopin was the epitome of hard, brilliant whanging and banging, now in this new album it is absolutely lovely. Self-assured of course, and utterly confident no matter what the technical difficulty. But musically effortless too, flowing like oil, sunny, relaxed, alive, fluent. Never has this kind of Chopin sounded to better effect.

For an artist like Rubinstein, it seems to take the final coming-to-terms-with-life of approaching old age to bring effort and sincerity into focus with sheer technical know-how. In the last few years, Rubinstein's recordings have shown such a wise, human mellowness—minus bangs and whangs—as only a man in his gifted seventies could produce.

By golly, you can even use this one as background music. Takes to it like a fish in blue water and Chopin's dignity never suffers. (Remember—his was the original "salon music".) That's saying a lot, both for Rubinstein and for RCA's mellow piano recording.

XMAS-CELLANY

Beethoven: The Four Overtures to Fidelio. Philharmonia Orch., Klemperer.
Angel 36209 stereo

Beethoven wrote only one opera—but managed to turn out four different overtures for it, in a humped curve; first an enormous piece, ten times too big for *any* opera; then a concentration of that, shorter but even more overpowering. Common sense, at last, led to a mild third version, mostly but not all new—it was never used. And finally, for an extensively recast revival years later, he composed the perfect overture, musically entirely unrelated to the first three and, at last, on a small enough scale to fit into its place at the beginning of an evening's operatic fare.

Thanks to mixed-up publishing dates, these are, in order, Leonore Overtures Nos. 2, 3 and 1, and the Fidelio Overture. (The opera's name had shifted from Leonore to Fidelio.) For many a year I've been hoping to find a really first rate dramatic performance of all of these in a single recording. Here it is.

Not many conductors are left who can give the music the Germanic sweep and clarity of Klemperer's performances. Almost nobody (almost—to be safe) can reach the intensity of these playings without hammering, thrashing, roughing-up the musical edges. A maximum of power with a minimum of harshness and sloppiness. An unerring sense of style, precisely suited to Beethoven in 1965, combining modern terseness with a classic take-your-time Romanticism . . . . . Enough generalities! Go and listen for yourself.

Liszt: Piano Concerto No. 1 ("Triangle"). Saint-Saëns: Piano Concerto No. 2, Op. 22. Ruth Slenczyska; (a) Orch. of Vienna, Melles; (b) Symphony of the Air, Swoboda.
Decca DL 710084 stereo

This lady with the consonentally Polish name was a round, chunky little-girl prodigy back before the War, right out of somewhere in the good old U.S.A., if I remember rightly. She retired into limbo, grew up and in due time, not so far back, reappeared for a grown-up try. She seems to have made it OK. She's getting around now.

In these two show-piece war-horses she shows herself oddly still a solid, chunky, determined pianist, not much of a poetess in the drawn-out lyric parts but a powerhouse when the music at last turns rapid. The high points in both of these concerti, decidedly, are the scherzo-like movements, in the middle of the unbroken Liszt, the second movement of the Saint-Saëns, where she lets out like a chubby race horse and flies over the keys with terrific enthusiasm!

I don't think much of either accompaniment. The anonymous Viennese orchestra ought to know the Liszt better than it does; the Symphony of the Air sounds as though maybe it were almost-sight-reading the Saint-Saëns. Both are OK, but on the routine side. Ruth, with her 1920's boyish bob, carries the music for them.

Albéniz: Suite Española; Pavaña-Capricho Cantos de España. Alicia de Larrocha, piano.
Columbia MS 6603 stereo

This is an interesting import by Columbia, recording by Hispavox in Spain itself. The all-Albéniz program, played by a leading lady pianist of that country, has an unusually authentic Spanish flavor, easily evident on the hearing.

It's not always a total pleasure, I'd say. This lady plays with immense fluency and fine musicianship but her style (for our ears, at least) is somewhat old fashioned and tends to be a bit over-dramatic. A great deal of fussy, if fluent rubato (slowings-down, unevenness in tempo, for emotional effect). Quite a drama of big-bang endings—one can almost see the lady throwing her hands up in the air in triumph at the end of each item! Very stagey, definitely. But what brothers a casual listener most, I think, is a certain blowing-up of music that is intrinsically not too profound into something too big, too nationalistic for its own content.

The Suite Española, mostly lyric and fairly quiet, gets nothing worse than a poetic, old-fashioned Romantic treatment. Pleasing, decidedly. But the other works, on the second side, are of slightly sterner stuff and here the lady plays almost furiously. Rather hard on the stylus.

I wouldn't suggest this as a disc for tasteful background listening (which is where most Albeniz finds itself these days) and that, I think, is my best compliment to Alicia de Larrocha. She is no mere parlor pianist.

O Great Mystery. Unaccompanied Choral Music of the 16th and 17th Centuries. The Canby Singers, Edward Tatnall Canby.
Nonesuch H-71026 stereo

As my *alter ego* the record reviewer I give brief mention to this item which, in my capacities as conductor, recording director and tape editor took me a total of four years' on-and-off work and most of last summer to complete. Very good for a reviewer's too-callous soul, to see how hard it is to *make* records.

The music is all unaccompanied, featuring on Side 1 three settings of the Christmas text O Magnum Mysterium (O great mystery, that the animals should see the birth of Christ. . . .) by Victoria, Morales and Byrd, plus works by Lassus and another Spaniard, Guerrero. Side 2 has a variety of music, more contrasted, including a couple of slightly zany love songs—Waelrant, Schütz, Handl, Melchior Franck, Schein and Monteverdi are the composers.

What *I* like about the music is the perfect tuning and clarity of the harmonies, the lack of vibrato in the voices (I picked 'em that way) and the consequently pleasurable ease in the listening, abetted by the fine acoustics of the chapel of the General Seminary in New York. As the ads say, you'll enjoy. . . . etc etc.

Maurindo Almeida—Guitar Music from the Romantic Era.
Capitol Duophonic DP 8601

The "Duophonic" series is made from mono originals treated for synthetic two-channel recording for playing on stereo machines. (See "Audio, E.T.C.," October 1964.) The solo guitar definitely benefits, on my equipment at least, from the more immediate sense of space and presence added here by the synthetic "stereo." For a solo instrument—a "point-source"—this enhancement might logically seem unnecessary. But it does help, not in spreading the instrument out but in setting it acoustically within a more immediately sensed space.

My only technical objection here applies both to mono and "stereo" versions; the original recording seems to have an unconscionable amount of bass for a mere guitar and (perhaps as a consequence) a rather dull sounding treble. Could be miking, or even the frequency response of the original recording equipment; it also might be no more than faulty equalization, trusting to formula rather than the evidence of the ear. Who knows?

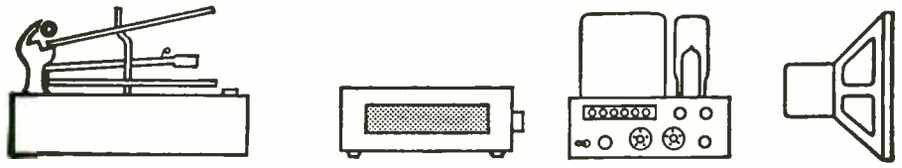
Nice playing of a batch of classical transcriptions of Romantic-perod tid-bits by Beethoven, Schumann, Chopin, Masseet, Grieg, Debussy, and one Mozart theme put to guitar variations by the ever-present Fernando Sor.

Piano "Pops" Promenade. Leonid Hambro, Jascha Zayde, duo-pianists.
Command CC 11023 SD stereo

Amazing how musical talent and energy, these days, can erupt in unlikely ways such as this, a battery of two-piano "arrangements" that come close to being re-compositions for two pianos. One, indeed, is just that, a set of Variations on a brace of earlier

(Continued on page 61)

EQUIPMENT PROFILE



BOGEN SOLID-STATE FM-STEREO RECEIVER, MODEL RT-6000

The name Bogen has graced many fine high fidelity instruments over the years, extending back to the origins of high fidelity. Throughout this period they have been known as a maker of well-designed and well-engineered products. When Bogen literature specified performance characteristics, you could be sure that the product you bought would perform that well, if not better.

The RT-6000 is no exception; it performs exceedingly well, as claimed.

The RT-6000 is an FM-stereo tuner a stereo control center, and a 50-watt rms stereo amplifier, all on one 16″ × 14″ chassis which is 5½″ high.

We should restate that slightly: The RT-6000 is a *completely solid-state* FM-stereo tuner, control center, . . .

Only a few months ago there was nary a completely solid-state receiver to be found. Without going into a lengthy discussion of the subject we must point out that it is much more difficult to make a completely solid-state tuner perform at compenent high fidelity standards than a tube unit. The Bogen RT-6000 is evidence that the problem can be solved.

Also, at the amplifier end of the instrument, the RT-6000 demonstrates a refinement of performance as compared to previous solid-state amplifiers with a similar output configuration. This we would expect. After all circuits of this kind have been available for several years now and so have the devices.

Control Center

The RT-6000 incorporates inputs and controls for accepting signals from a stereo tape head, a stereo magnetic cartridge, and a stereo high-level source, in addition to the built-in tuner. Outputs are provided for stereo recording and monitoring the recording. A headphone jack is provided on the front panel and slide switches are provided to select speaker or phone output as well as the monitor mode.

In addition to the usual tuning, volume balance, bass, and treble controls slide switches are provided for a.f.c. defeat, loudness defeat, stereo or mono mode selector, and stereo reverse. The source selector also permits selection of FM Mono or FM Auto.

In the FM Auto position of the selector, the set is automatically switched between the mono and stereo modes of FM reception depending on the availability of an FM-stereo source. Switching is completely silent and undetectable audibly.

A meter is provided to indicate FM signal strength.

Before leaving the front panel, it should be noted that the styling is quite handsome. The entire front panel is brushed gold and the knobs are also gold-colored. Its a pity the slide switches aren't gold too, instead of the usual dark brown.

The slide-rule tuning dial is of the new "expanded" persuasion, measuring almost 7-in. in length. In addition, a logging scale is provided just below the megacycle scale. Very convenient for explaining the location of stations to the non-technical user. Certainly 4 is easier to use than 96.3 mc.

Fig. 1. Bogen RT-6000 solid-state FM-Stereo Receiver.

FM Circuit

Viewing this solid-state FM circuit and comparing it with tube FM circuits, one is struck by the basic similarity. Except for some circuit constants related to tubes or transistors as devices, one could superimpose tubes in place of transistors throughout. If one takes off the bottom plate and examines the underchassis, it is very difficult to tell whether there are tubes or transistors on the top side except for the absence of tube sockets. The same rats nest of point-to-point wiring exists as has existed for many years.

Thus we note a common base (grounded grid in tubes) r.f. stage, straightforward mixer and oscillator Following are four i.f. stages with a tap at the last stage to drive the meter circuit. A.f.c. voltage is taken from the output of the ratio detector.

The multiplex circuit is also straightforward and reflects the latest thinking. A circuit is included which decides when there is sufficient 19 kc pilot signal present to indicate a stereo program (some 19 kc is almost always present as harmonics of the audio signal or from other sources, but much lower in level than the pilot signal). If there is not enough pilot signal, the circuit "switches" the output to mono. The switching is electronic, which accounts for its speed and quietness.

Amplifier Circuit

The amplifier circuit is also quite usual, employing feedback networks to achieve tape and phono equalization as well as bass and treble compensation. The phono input is loaded with 47k which is quite usual. The aux input is divided down to make it less sensitive rather than have it enter the circuit past the preamplifier stages. Thus if a sensitive uncompensated input is required, a microphone input for example, all one need do is remove some resistors.

The first two preamplifier stages are a 2N2613 and a 2N591 in that order. An output is then taken for the tape output which has another stage of amplification, a 2N408. The ganged volume control (all basic controls such as volume, bass, and treble are ganged) and the loudness compensation circuit precede the next amplifier stage, a 2N408. The following tone driver stage is also a 2N408.

The power amplifier section follows, with the signal going through a 2N2614

and a 2N408 to reach the driver transistor, a 2N2148. This is a rather husky driver section and is undoubtedly an important reason for the good performance of this circuit. The driver transistor feeds a transformer which acts as a driver to the push-pull output stage. consisting of a pair of 2N2147 transistors. A positive and negative supply of 22 volts is provided the output stage by a bridge rectifier in the power supply. Both supplies are fused as are the speaker output leads. There is no coupling capacitor to the speakers, thus eliminating a possible trouble spot.

The power supply for the remainder of the circuit is provided by a rectifier consisting of two diodes and then through a 40050 transistor in emitter follower configuration which supplies –38 volts, and then through two 2N408 transistors, also emitter followers, which supply –25 volts.

The amplifier circuit is designed to provide optimum power at 4 ohms, as we will note in the performance characteristics.

Performance

Starting with the tuner section we found sensitivity to be 2.7 μv (IHF), crossmodulation index 65 db, selectivity (alternate channel) 37 db, capture ratio 4 db, AM rejection 55 db, and excellent pulse noise rejection. The RT-6000 pulled in 34 stations loud and clear on our standard antenna.

The amplifier section exhibited a very smooth response, being only 2 db down at 15 cps, rising to 0-db at 1000 cps and continuing smoothly to a level of 1 db at 10,000 cps, returning gradually to the the 0 level at 20,000 cps, and then going down 1 db at 50,000 cps. This response was obtained at 20 watts rms at an impedance of 5 ohms. Power output at 5 ohms was 24 watts rms at a distortion of 0.7 per cent. At 8 ohms the output was less by about 20 per cent. Power bandwidth was 20 to 20,000 cps. IM distortion was just under 1 per cent with a 60 and 7000 cps signal mixed 4-to-1 at an equivalent output of 20 watts.

Altogether, the Bogen RT-6000 is a fine performer at a surprisingly low price. As far as we know its the first solid-state component-quality receiver at less than $400. **Circle 211**

ACOUSTIC RESEARCH AR-4 SPEAKER SYSTEM

Acoustic Research has been the leader in the fight for acceptance of compact speaker systems on their own merits without regard to price. While there are still holdouts, most responsible music listeners acknowledge the AR-3, Acoustic Research's senior system, as a major contribution to the art of reproduction. Certainly, at least, the $14 \times 25 \times 11\frac{3}{8}$-in., $225, AR-3 is established as a well

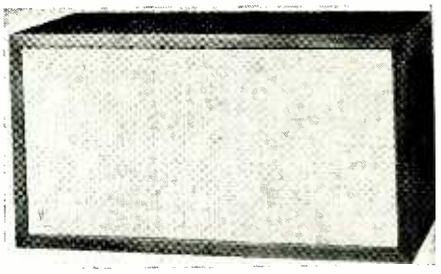


Fig. 2. AR-4 Speaker System.

known speaker of (incidentally) modest dimensions.

Now comes the AR-4, their latest entry. This system is $19 \times 10 \times 9$-in. and costs a mere $57 in oiled walnut and $51 in unfinished pine. And, as incredible as it may seem, these two systems are sonically related.

The AR-4 is a two-way system employing an 8-in. woofer with what is commonly called acoustic suspension. This is crossed over to a 3.5-in. wide-dispersion tweeter. This tweeter is controlled by a rear panel pot that permits tailoring of the system to room acoustics. Efficiency of the system is moderately low. AR recommends, and we would agree with them, that a minimum of 15 watts rms should be used to drive the speaker. Connection is made to the 8-ohm tap of an amplifier.

How it Sounds

For any speaker system in this price class it would be an easy task to essay what is *wrong* with it, from an absolute sound point of view. The AR-4 is no exception in this respect. Yet, it might also be said that it would be a most difficult task to find anything *right* with so low-priced a product. This is *not* the case here. Let it be stated immediately that the AR-4 is a most listenable speaker musically.

Subjective listening tests reveal that it simply does not have deep bass. And there seems to be a rounding off of the extreme high end as well. But these effects are not obvious as those two sentences would lead you to believe. The musical balance of top end to bass is so well controlled that there is the *effect* of deeper bass than actually exists. The top end too, is similarly affected.

The AR-4 sounds exactly the way you would expect a scaled down AR-3 to sound. The family resemblance is unmistakable. The over-all sound is subdued and a bit pinched, yet not in any way difficult to listen to. There is no raggedness or roughness in the AR-4. Male voices lacked fullest roundness and the female voice was dulled when the AR-4 was compared with several systems costing four to five times its asking price.

Transient response also suffered when compared, but only slightly. Yet it is amazing how well this speaker did hold up against the high-priced stuff.

As with the senior AR systems, this one sounds better the louder you play it. And, it will not break up under any signal likely to be presented to it under home conditions.

The AR-4 was frequency swept in an effort to correlate what was heard against an established standard. Subjective evaluation of the sweep showed that the AR-4 dropped off sharply below 50 cps. If lower frequencies were forced into the speaker, there was slight doubling, but for all practical listening purposes there was no response below 50 cps. Above that frequency, however, response was smooth, without obvious peaks or valleys, to well above audibility. Our system was adjusted, via the rear tweeter control, to suit our preferences. At this setting we found that highs were all there, but that there was a slight rise in response from 2000–6000 cps. This, and not any actual rolloff, we suspect, was responsible for the musical effect of slightly diminished high-end response. On an over-all basis, we judged this speaker to have an outstanding frequency response and sweep characteristic. And without regard to its price level.

In conclusion, then, the AR-4 is *not* one of the great speakers. But we doubt that this is what was intended by Acoustic Research, if, in fact, a *great* speaker can be made for so low a price. But the AR-4 is a speaker designed for a budget that, at the same time, wants unusually high musical quality. The AR-4 can be listened to for hours on end, without fatigue. **Circle 212**

WEATHERS "TOWNSEND" STEREO TURNTABLE

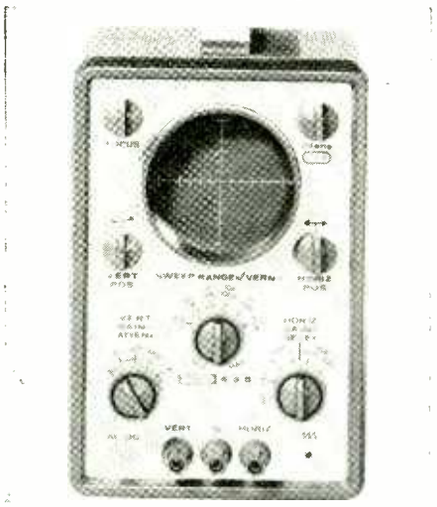
To this new single-speed turntable one could readily apply the old expression "Handsome is as handsome does," for it is certainly most attractive in appearance and its performance should enhance many a stereo system. With its over-all height of just a bit over five inches and base dimensions of $17\frac{3}{8}$ by $14\frac{1}{4}$ inches, it is first of all compact, yet capable of high-quality performance. With its satin-finished aluminum chassis, walnut base, and walnut arm with aluminum fittings, it is modern and functional in appearance.

The concept of a constant-speed synchronous motor of small dimensions driving directly a light-weight platter has been delineated before in these pages, and is accepted as valid. In the "Townsend" a 12-pole synchronous motor of clock proportions is fitted with a 11/16" dia. rubber drive puck on its shaft which transmits its motion directly to the inner rim of the aluminum turntable and the spindle rides in a Delain bearing. The

(Continued on page 62)

NEW PRODUCTS

• **Oscilloscope Kit.** Eico's new 3-in. a.c.-d.c. scope kit features distortionless vertical and horizontal trace expansion to several times screen diameter and drift-free positioning allows examination of waveforms with as much detail as with 5" scopes. Vertical response is flat +1 −3db from d.c. to 4.5 mc. Sensitivity is 18 mv per centimeter rms and 50 mv/c peak-to-peak. Input impedance is one megohm shunted by 35 pf. Horizontal response is +1, −3 db 1 cps to 500 kc with a sensitivity of 0.7 volts per centimeter rms. Input impedance is 4 megohms shunted by 40 pf. Intensity modulation input sensitivity is 3 volts rms for blanking and input impedance is 2.2 megohms. A 10

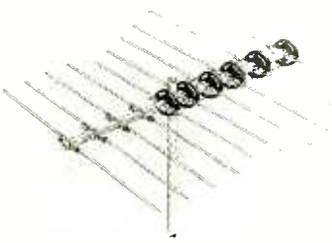


volt peak-to-peak sawtooth, with an output impedance of 300 ohms is available from 10 to 100 kc. The cathode ray accelerating potential is 1500 volts. The same gun as in a 5-in. tube is used which results in greater sharpness and brightness. Extremely fast retrace permits viewing a complete single cycle at the highest sweep frequencies. Also provided is a 200 mv peak-to-peak calibrating voltage zener regulated to ±5%. Weight is 15 lbs. It is 8½-in. high, 5¾-in. wide, and 12⅝-in. deep. The Eico 435 oscilloscope is priced at $99.95 in kit form and $149.95 factory wired.
Circle 200

• **Improved Stereo Receiver.** Sherwood's new S-7700III 80 watt music power FM stereo tuner-amplifier is the first such product to incorporate circuits and built-in test points for connection of an oscilloscope for direct visual display of incoming signal characteristics. A high fidelity installer can thus identify the presence, degree and direction of multipath interference. Armed with this information, he can

orient the antenna and take other necessary steps to eliminate this serious stereo reception problem. This all-inclusive component provides AM, FM and FM multiplex broadcast reception, as well as inputs for record players and tape decks. Two 40-watt music-power channels are provided with outputs for 4-, 8-, and 16-ohm speakers. A separate powered center channel speaker output is also provided. The front panel contains a low-impedance earphone jack. Important specifications include: 36 watts per channel rms output; IM distortion is claimed at a maximum of 1.5 percent; at full rated power output response is 20 to 20,000 cps ± 0.5 db. Dimensions of the component measure 16¼ × 4 × 14-in. D. Cost is $374.50. Also available as optional accessories are a leatherette-on-metal case for $9.50 and a hand-rubbed walnut case for $29.50.
Circle 201

• **Periodic Principle FM Antennas.** Based on a principle of antenna design already successfully used in a line of TV antennas, Jerrold Electronics has announced three new antennas specifically cut for the full FM band. Known as FM Paralogs, these are highly directional arrays. A high degree of rejection of unwanted signals is claimed for this line. As a result, a significant reduction of the problems of



multipath FM-stereo interference will be realized. Mechanically, these antennas feature Cycolac insulating mounts. This is a very tough plastic also used for golf club heads. A wedge-snap lock provides permanent joint connections which tighten with wind vibration. Made by Jerrold's TACO subsidiary, they are priced from $29.95 to $59.95.
Circle 202

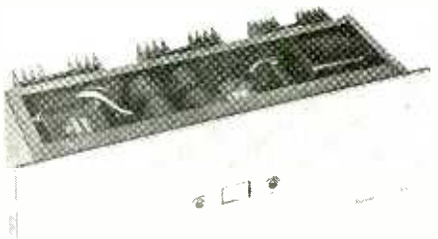
• **Pushbutton Stereo Recorder.** Recently announced by Lafayette Radio is a new four-track stereo recorder, the model RK-675. Selling for $179.95, the recorder plays and records four-track stereo and monophonic tapes at both 7.5 and 3.75 ips. Two level meters, a digital tape counter, and a pause control are incorporated for maximum recording convenience. Frequency response at the faster speed is 40–18,000 cps. At the slow speed top-end response is reduced to 12,000 cps with no change in bass response. Built-in amplifiers give an

audio output of 3 watts per channel. Stereo separation of 45 db is claimed at the amplifier output. This is maintained by built-in speaker flaps that disperse the sound from the two 6 × 4-in. speakers on either side of the recorder. Wow and flutter is stated as 0.2% at 7.5 ips and 0.25% at 3.75 ips. The RK-675 comes complete with two dynamic microphones, cables and take-up reel, all contained within a simulated leather case measuring 16⅛ W × 7 H × 12⅛ D.
Circle 203

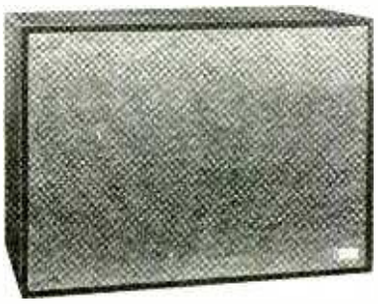
• **Amplifier Kit.** H. H. Scott has just announced the LK-72B 80-watt stereo amplifier kit. This is an integrated kit with facilities for phono and tape decks, separate channel tone controls, a derived center channel speaker output and Scott's subsonic filter circuit to prevent waste of useful power. In keeping with the latest styling from this company, the front escutcheon has been redesigned as an ex-

truded aluminum plate. A full-size and full color instruction manual, with matching Parts-Charts make for fool-proof kit assembly. All mechanical parts, terminal boards, tube sockets and the like have been pre-riveted at the factory. All wires are pre-cut to length and pre-stripped. Important specifications include an output power rating of 40 watts per channel (IHF), and a power bandwidth of 20–20,000 cps ± 1 db. Frequency response is also stated as ±1db from 20–20,000 cps with harmonic distortion of 0.8%.
Circle 204

• **All Transistor 100-watt Amplifier.** Rauland-Borg Corporation has made available the Model TA100 all-transistorized 100-watt amplifier designed for continuous-duty service in commercial and industrial application. Companion Model TA50, a 50-watt solid-state amplifier, is also available. The Model TA100 features: 100-watts rms



(200-watts peak) output; thermostatically controlled protective relay; back-up fast-acting overload protective relay; instant operation—no warm-up required; low power consumption; approximately 75 per cent less heat dissipation than in tube amplifiers; frequency response, plus or minus 1½ db, 50–15,000 cps; distortion less than 5 per cent at rated output (less than 3 per cent at 80 watts); noise level, 80 db below rated output.
Circle 205

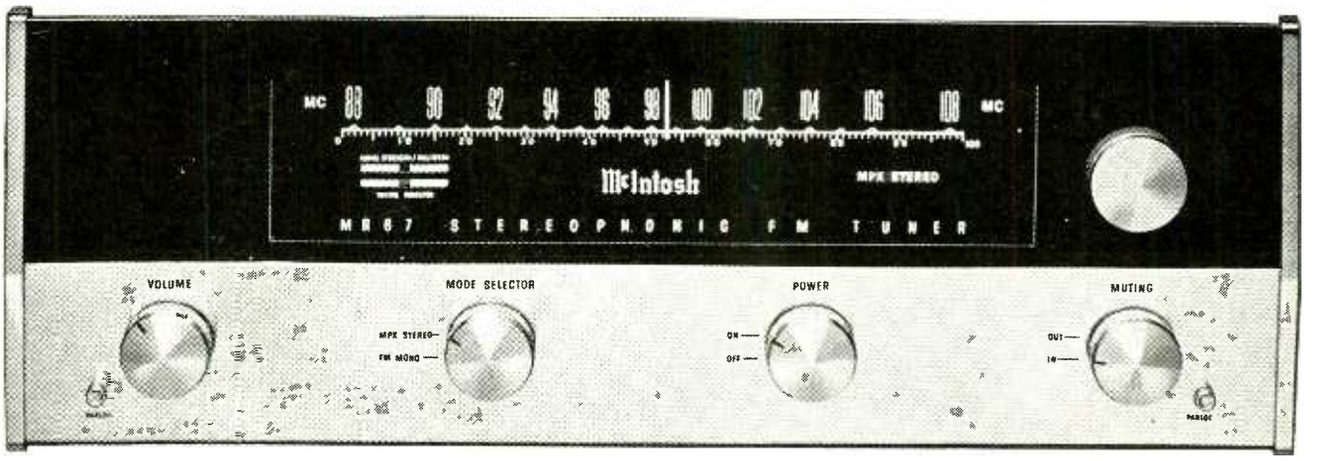


• **New Medallion Speaker System.** In a move to supply a speaker system with the sound of its Medallion and with its changeable grilles, University recently introduced the Medallion Monitor. The Monitor is a fixed grille bookshelf version of the Medallion XII with improved versions of the three speakers presently being used. It consists of a 12 in. woofer, an 8 in. mid-range and the Sphericon Super Tweeter. Response of the unit is claimed to be virtually undistorted from 20 to 40,000 cps. Finished in oiled walnut, the Medallion Monitor is priced at $129.00 net. LTV University.
Circle 206

"UNEXCELLED

by any other Tuner!"

Audio, February, 1964



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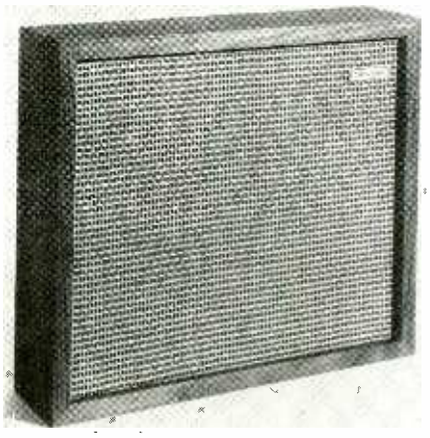
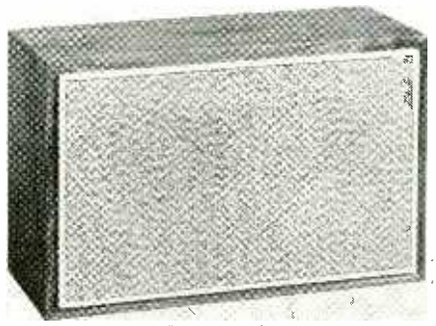
Circle 128 on Reader Service Card

● **New Trusonic Speaker.** Trusonic has introduced "Lyra" as the first of its new line of packaged speakers. The new unit measures 14" × 16" × 4". This size was planned to make Lyra fit into any space demanding a compact speaker. Within each cabinet are two miniature speakers. Each speaker,



a new type named the CR-50, will handle over 25 watts of power with smooth sound from 25 to 20,000 cps. The cabinet is made of select walnut. A black hairline is set into the sides to enhance the Italian finish. Also featured is Trusonic's lifetime warranty. **Circle 207**

● **Scott Speaker System.** H. H. Scott has announced the introduction of its S-5 speaker system. The S-5 was especially designed for use in the Belgian Village at the New York World's Fair, and is a two-way loudspeaker system employing a specially-designed low-resonance woofer and



a high-frequency tweeter. Specifications: 8-ohm impedance; frequency response ± 5 db from 60 to 15,000 cps. Dimensions: 10" × 16" × 6¾". Weight: 12 lbs. Price: less than $60. **Circle 208**

NEW LITERATURE

● **Magnetic Tapes Bulletin.** Electromagnetic and physical properties of magnetic tapes are described in a new six-page brochure now available from Reeves Soundcraft. The color catalog provides both properties and complete specifications for Reeves Mylar and cellulose acetate base tapes for sound recording. The bulletin is listed as RS-64-18. **Circle 209**

● **Sonotone Product Catalog.** Sonotone's Electronic Applications Division is distributing its new audio product catalog, covering the firm's OEM, distributor and consumer products in the hi-fi and electronic field. The 16-page catalog illustrates, in detail, Sonotone's complete line of ceramic and crystal cartridges, replacement needles, tonearms, ceramic microphones (including low-impedance types) and learning lab headset/microphone units. It also shows the Sonotone speakers and new speaker enclosure systems as well as the company's rechargeable flashlight battery cartridges **Circle 210**

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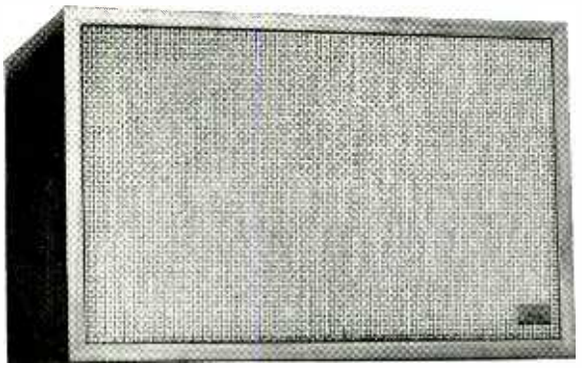
3" mid-range,
3½" tweeter.
24x13½x11½"
D. $79.50



UNIVERSITY SENIOR II
Ultra-Linear 12" woofer, 3½" mid-range, Sphericon
Super-Tweeter; 25x15⅝x12⅛" D. $99.50

UNIVERSITY MINI-FLEX
6½" woofer, 3" mid-range
3½" tweeter. 15x9⅞
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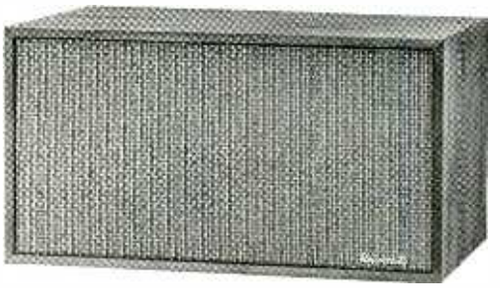

UNIVERSITY COMPANIONETTE
Ultra-Linear 8" woofer,
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21¾x11½x8⅝" D. $69.95

...and listen and listen. New? Revolutionary? Yes—when you consider how many people buy speakers based on the recommendation of others. Sound involves **subjective** criteria. The sound that pleases a friend, (a hi-fi editor or salesman, for that matter) will not necessarily please you. Therefore . . . hear and compare many systems. For the largest selection, **start with University.** Choose the superb University model that best meets your requirements, then compare it to all other brands of its type. For example—if it's a full-size bookshelf you want, ask your dealer to demonstrate the Senior II vs. the AR, KLH, and other bookshelf systems of similar size. You'll hear the difference. Especially in the mid-range. Especially in the Senior's complete absence of restraint, that tell-tale drawback of so many other bookshelf systems. Unlike other systems, **the** sound of the Senior, the Companion, or of **every** University system, large or small—is **free** and **open.** The bass is cleanly defined; the mid-range punches through for greater presence; the highs literally have wings. Want proof? (Of course you do) Visit your dealer . . . and **listen.** University sounds better. Free 1964 Guide to Component Stereo! Write: Dept. R-12.

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Circle 115 on Reader Service Card

COMMERCIAL SOUND

(from page 42)

enough'' ground, or ''too much!''

The system should have just one straightforward ground connection, from input to output, and with no extra connections bypassing the ''official'' one. If there is a break in the intended connection, or one piece of equipment fails to get its ground at all, or even if the official ground takes an unduly lengthy route (longer than the signal path, for example) this can cause hum. Check your ground connection with a continuity tester. You'll find a break or omission that way.

Then if there is already a complete ground connection and another is inadvertently connected (or even deliberately, with the idea of providing ''good measure'') you may even get worse hum than that due to a partially missing ground. This you can check by breaking the intended ground path and using the continuity tester to see whether it really does break, or whether there's another connection you don't know about, or forgot about.

Suppose, for example, your system includes tape recording facilities. For complete grounding according to rule, when recording the tape recorder's ground should be picked up at its input, from the ground point on the system feeding it (the tape output on the amplifier); on the other hand, when the tape

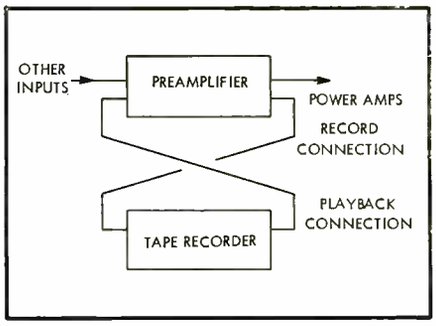


Fig. 9–7. Connection of recorder to system discussed with reference to double-grounding hum problem.

is being used on playback, the ground should come from the tape recorder's output to the amplifier input to which it connects. Switching is usually incorporated to separate these functions, so that both connections can be made permanently, as far as the relevant connectors are concerned. Unless the switching breaks the ground, as well as the signal circuits, there may be hum because of the ground loop so created.

If it is not practical to have the switching break ground connections, you will have to find a way of connecting so there is only one ground connection. Usually the safest thing is to break the ground connection associated with the transfer that occurs at the higher level, and let the lower level ground carry for both. The high-level ground tie operating with the low-level transfer is more likely to cause hum than vice versa. (*Fig.* 9–7).

So, if the output from the tape recorder is at lower level on playback than the signal fed to it on record, use the playback ground connection and break the record connection. On the other hand, if the input to the recorder for record is at lower level than its playback output, use the record mode ground, and break the playback connection. Where the difference in level is not great, or both are equal, you may find that you have to either change connections every time, or devise switching that does.

AUDIO ETC

(*from page 12*)

formist myself in this audio world of ours and I treasure almost any old individuality that manages to survive in our dun-colored time.

The individualist speakers have a distinguished tradition behind them. Many an old line's reputation has been built upon the maker's very special sound, and for many long years people have liked these specialties. (Just as they once liked the Chevy radiator pattern, or the Ford, or the Packard or Auburn, so clearly different and characteristic from each manufacturer.) People have bought thousands of dollars' worth of speakery for their home componentry (and its classical musicry), all on the basis of these familiar and well-established styles of speaker sound. That was, and is, their privilege.

It is also the manufacturer's privilege to cater to these solidly differing tastes, to build and to plug his own type of speaker sound and continue to sell it to every willing customer who finds it impressive, as plenty still do. That's the way the system has worked, and still does.

It still does, but alas, individuality is weakening, I'm afraid. The new factor on the scene today, the audible scene, is the gray flannel suit in speaker guise. And it is spreading fast. The end result is going to be inevitable.

The more of these conformist, soundalike, speakers there are, the more will the neutral, conformist sound become *normal.* And the funnier will the rugged-individualist speakers sound. Too bad, in a way. It is the end of an era.

You see, in earlier years we have had a number of neutralist, gray-flannel speakers around—for a long time, in fact—but the average ear couldn't really tell them apart from the others. With so few of them, relatively, their particular color didn't seem special one way or another; they were simply one more type of sound, among many to be heard. But that was before the Sound-Alike Club, my private club, came into existence.

Now, you see, the big change is upon us. That's what I heard at the autumn hi fi show. Everywhere, in room after room, more of those gray-flannel sound-alike speakers. Until as I moved about I became attuned to this new sound, so neutral, so uninteresting and unobtrusive, I clean forgot all about it through sheer repetition and my interest shifted to the many other fascinating aspects of the hi fi scene. (Same with other people. Couldn't keep their minds on these colorless speakers.)

That is—*until* I entered a room where one or two of the non-conformist speakers were still going full blast, with their special sound—and what a shock! It hit you in the face, after the gray flannel effect, and it hit you unpleasantly. Sometimes, looking at the distinguished names, I was incredulous; I could scarcely believe my tired ears. That's what happens after too much continuous exposure to gray flannel sound. Color—any color—sounds awful.

Don't think the public isn't going through this very same experience. It may take awhile longer to sink in. But there'll be no avoiding the trend. Gray flannel sound is IN. It is rapidly be-

coming NORMAL sound. The colorful speaker of old, with all its assorted enhancements, mellownesses, brilliances, presences, is on the way out via the public's own ear, and that is that.

So, Mr. Maker, if your speakers are noted for their distinguished individuality, you'd better join my gray flannel club quick, the Sound-Alike Club. It's getting less exclusive every day.

3. Novelties

I am certainly not going to throw myself to the wolves at this point by listing all the members of my Sound-Alike Club, nor even its Associate Members, those who almost made the grade. (*You* go listen, *you* decide for yourself with your own excellent ears.) But, having for once been systematic via penciled notes on the hi fi show exhibits that caught my fancy, I can't resist passing on a few items to you as novelties. Strictly personal highlights—no room for *all* my penciled notes, either.

(First note: the way to cover a hi fi show is to cross the fingers of your right hand and point to the nearest door on your right—right side of the corridor. Enter—*then get back out of the same door,* or you're lost. Continue with fingers crossed until you complete a circuit—then reverse engines and go the other direction, on the other side of the corridor. Never uncross those fingers and *never shake hands.* If you do, you're a goner. Worked fine for me—I visited every room in the New York show, on four floors. But, alas, my fingers are still crossed; they got stuck that way after a couple of hours.)

At least two tape recorder exhibitors did a simple show-trick this year, which I recommend as both dramatic and informative. They set up a battery of identical recorders each playing at a different speed, and so marked in large numbers. A switch allowed ABCD comparison of the sound. Crown's exhibit was the trickiest, for they had big machines running at 7½, 3¾, 1⅞. 15/16 *and* 5/16! All music, too. The 5/16 speed sounded like a good telephone broadcast. The music scarcely wavered. Amazing.

Marantz's sensational true-tracking phono arm, the SLT-12, was the object of my fascinated inspection. Being me, I touched it and threw the thing off its trolley for fair. (It was tracking a violently eccentric single groove on a shiny acetate disc.) This isn't the first arm that manages to remain tangent to the grooves throughout its play, but it surely is the most ingenious mechanically. Put your fingers together as in prayer, keeping them straight. Bring the joined tips towards you, still touching. That's the principle (i.e., the pickup travels in a straight line across the record, like your fingertips). It works, too, and soundly, for this mechanical arrangement allows for great "lever-

age" against the bearings (at your wrists) and thus minimizes the drag against the groove. But there are no less than twelve bearings involved—I counted them—and that's a lot. Nevertheless, the Marantz arm is one of those honest jobs that anybody who has a mechanical sense is bound to admire. More power (and less friction) to Marantz.

The smallest speaker in the show was a whiz, the tiny "Maximus I" from Goodmans. It's slightly bigger than a box of shredded wheat standing on end, and a pair of them sound like something you won't believe. Hope to try some later on. Smallish speakers are making a big spurt this year and, natch, virtually all of them are flaunting some variety of "acoustic suspension." That term now seems to mean any speaker enclosure that doesn't have a hole in it, not counting the speaker hole. We used to call them infinite baffles. Some of the new ones do, indeed, take advantage of the soft-cone, big-air-spring system and the results are understandably good, here and there, if not always.

The new smaller "bookshelf" size listings are much enriched this year, with significant items from many makers. I enjoyed, very briefly, the KSC models, the new KLH Model Seventeen and (how we do run to initials!) the new AR-4; also the Sonotone Sonomaster RM-1 (not to be confused with E-V's quite different Sono*caster*, an outdoor waterproof carryable job).

All of these new smaller models are of a size that really promote convenience and a much-needed flexibility in stereo listening, for they invite being moved around—and we *should* move our speakers around to suit different sorts of recorded material. The sound, too, while naturally compromised from absolute perfection, is something we would not have believed ten years ago. It can be done.

(If you still want your 30 cycles clean and your 50,000 too at 100 watts, stick to the monsters. They're still around, if a bit shrunken in cubic footage.)

I was intrigued by Leak's "perfect piston" sandwich, a speaker cone of featherweight foam about a quarter-inch thick, with two hard-faced silvered surfaces. He passed out pie-shaped piece of the stuff, one of which I have before me. Nice. Fisher had an aluminum cone, (partially) but I only saw it, didn't hear it.

Then there was the inevitable upside down record player—this time it was Dual. I'll have something to say later about the tendency to jar which is inevitable in these new feather-touch systems, even when upside down. Loose floorboards don't like them. And then there was the backwards tape recorder, taking us straight back to the beginning of home tape. Remember the ancient Twin-Trax? It started the "half track"

system, playing frontwards, then backwards to the beginning. This year it's Ampex and Viking who are doing it. But the fatal flaw in the older reversible systems, the moveable head, is now thankfully absent. The heads are fastened down to stay. You just have more of them. Concertone's Reverse-o-matic started the new trend awhile back; now that Ampex has joined up, we'll probably be seeing a lot of mirror stuff in the tape department. No more reel changing for "Side 2" and a good idea if you don't mind the added machinery and the electronics. (If you do, you can still buy the old turn-it-over kind.)

Trickiest speaker at the show was that huge lampshade affair, big modern-style vase lamp with the sound coming direct from the lamp shade itself—an electrostatic speaker. (Bass from the lamp base.) Surprisingly, those lamps really sounded pretty good. I'd enter them in my Sound-Alike Club even on brief audition, over people's craning necks. The Acoustica Omnisonic 360—better write that down 'cause you'll never remember it. (Panasonic? Trusonic? Duosonic? Supersonic? Subsonic??)

Of course almost any virtuous electrostatic today sounds pretty good at least in the upper end. Definitely within sound-alike tolerances, by their very nature JansZen's electrostatics were still around and still sounding good; he had a neat reflector to spread the sound of a single electrostatic tweeter "cell." Modest price on it. Pickering, alias Stanton, was using one of the old Pickering electrostatics, the unit mounted overhead on stilts with a woofer box below; it sounded wonderful and so did a new Stanton model in another room, not yet in production. Acoustech has the newest Janszen-designed monster, a pair of door-sized full-range electrostatics (like the KLH Nines) which will take 900 watts of audio out of built-in amplifiers. All you need is 1700 smackers.

How I do go on and on . . . room for one more minor novelty; Sonotone's newest ceramic stereo hi fi cartridge,which has two points, flipover-style, on a single stylus and is provided with neatly miniaturized equalizers so that the thing plugs right into any old magnetic input for instant play. Ingenious and probably sounds pretty good, too. Have to skip all those marvelous 15-degree magnetics; but you'll hear plenty about them elsewhere.

My bet for the zaniest name at the show—and no reflection on the product itself—is the Cipher. That means zero, or nothing. Now where do you suppose they got *that* idea? It's a line of tape recorders, and a lot more than nothing. With that cipherous thought, I bid you fair-well until next year's big show, for 1966. Æ

RECORDS

(from page 48)

Variations on a familiar theme, the well-known Paganini Caprice.

Jascha Zayde is the half of this team who lavishes his musical skill on these elaborate re-compositions. His style is marked on every one of them, superimposed on the original composer's ideas—Johann Strauss and waltz music from "Die Fledermaus," the familiar "Invitation to the Dance" of Weber, well-worn music by Rachmaninoff and Dvorak, every piece scintillating in its two-piano form, smooth as silk and glassy as glass, sometimes pretty hard and brittle, more often sparkling and light, and sometimes downright oily in sentiment.

It's as close as you can come to cafe candlelight music for two pianos; but the classics survive nicely. And Mr. Zayde's own variations on the Paganini theme, out-Brahmsing Brahms, out-Rachmaninoffing R., are exciting to follow in their sheer exuberance of technique even though the slightly cream-puff Zayde harmonies may give you musical indigestion. **AE**

AUDIO & HI FI IN WEST GERMANY

Emendations and Additions

As might be anticipated, we made a few minor errors in our article of this name in the November issue—none of them earth-shaking, but all correctable, as follows:

The DUAL people advise us that their top lines are made in St. Georgen and *Messkirch*, and the record players at a *third* factory in Dunningen, and that their total daily output of changers and record players is *2600* instead of the 1600 we credited them with.

We are sorry to learn that Mr. Oskar Steidinger, who was Managing Director at the time of our visit, passed away during the first week of October.

Baron Hornstein of UHER tells us that the company was founded by Count Toerring and his *son*, and not a younger brother, and he feels that apartments in Germany should more likely run from *$40* to *$60* instead of the $20 to $40 we reported (maybe he's accustomed to living in better apartments than we are) and that *$20* would be a more realistic figure for Vienna (though Fritz Sippl of AKG OK'd our $2 figure.) The Baron also counts *35* "roofs" at the Hanover Messe, whereas we counted from the "map" of the grounds only the 25 we reported—if it makes much difference.

Mr. Paul Metz of *METZ* tells that now a *second* FM program can be heard all over Germany. ELAC now has an output of *750* record players and changers per day, an increase of 50 over the 700 we said—an indication of some growth in just the past few months—and that Dr. *Rudolph's* (note the corrected spelling) son is general sales manager.

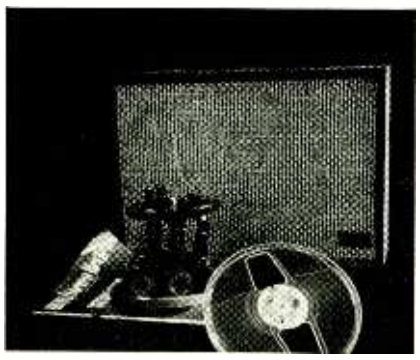
PERPETUUM-EBNER says we short-changed St. Georgen's population—*12,000* instead of 11,000. *CGMcP*

Not too long ago, I spent several days wandering the halls of the New York Trade Show Building, where four floors were taken up by an annual rite called the New York High Fidelity Show. It formerly was called the NYHF *Music* Show. Apparently the title was much too cumbersome, so they decided to drop the *expendable* word.

Anyway, some 60-plus components manufacturers occupied rooms spread over four floors of the building. And spread is the correct word. I have never

But for the manufacturer to get his money's worth, he must reach more than *some* who are actively in the market for merchandise. Compounding these economics is the fact that attendance has been shrinking over the past few years. This balanced against efforts to expand the market for components, raises, of itself, grave doubts as to the efficacy of presently constituted audio shows.

Previously, up to a few years ago, the shows served the purpose of stimula-



The Tape Guide

tion for the future buyer. These shows used to crawl with the *hobbyists,* the *kids,* the customers for the *next* five years. But they seemed to be absent this year. Is it partly because the compenents industry has outgrown its hobby customers? Let us hope not. There is a large, and as yet, untapped market among *music* enthusiasts *per se* (after all, what is all this equipment for, if not the reproduction of music) and they are the bulk of the future market, but the hobbyist remains the backbone. Break him, and the skeleton will collapse.

But to put meat on the bones, the music listener market must be the target of attack. At this stage, I doubt that anyone really denies the sonic superiority of components over the bulk of the packaged goods, but what is being done to get this message across? Is the large, expensive audio show (designed primarily for insiders) the answer?

The inescapable conclusion that must come from observations of this scene is manifestly clear. The great, centralized, and high-priced product exposition has grown obsolete in terms of the type of customer it attracts and, for that matter, fails to attract. But product expositions are, of themselves, *not* wrong. What is needed, and desperately so, are smaller, more regional, shows. It must be made possible for more people to get to see the components and talk to experts who are capable of conversation with lay, semi-knowledgeable and hobbyist groups. Selling the *concept* of componentry should be the aim, not selling the sound, at best a confused jumble at these shows. I dare say that few components have been sold, solely on the basis of how they sounded at a show.

It is time, I believe, for organizations such as the Institute of High Fidelity, that NAM of componentry, to sit down and indulge in some self-examination on this whole question. I for one, will nostalgically miss the big shows, but for the health of the industry, I believe they must go. **Æ**

AUDIO CLINIC

(*from page 2*)

poorly and your neighbor's tuner works well is that he is using a poor antenna and a tuner with a less critical front end. If you were to take his tuner and your tuner and place them side by side in a fringe area, your tuner would out-perform his. It might well be that you would be better off obtaining an old tuner with adequate sensitivity to bring WBAL and little more. Almost any tuner should be able to pull in signals 40 miles away. This might sound like taking a step backward, but it really is not when you consider the other factors in your particular case.

The only other thing I can suggest is that you disconnect your present antenna system from the tuner and connect a short length of wire to the rear terminals. All you need use is one terminal. Perhaps this arrangement will make it possible for you to pick up WBAL reasonably well. The antenna, being very poor in performance, will attenuate the strength of all signals, and might thereby permit your front end to operate normally. WBAL may provide sufficient strength to be heard with this antenna. It's certainly worth trying before discarding your tuner.

Transient Clicks

Q. For a couple of months now, while listening to a program on disc or tape, there occasionally comes from the speakers a very sudden and sharp single crack like that of a rifle shot. Perhaps it can be more appropriately described here as a kind of static or capacitive spark discharge somewhere, amplified and fed into the speakers. Also, the sound is not necessarily of the same intensity each time, and it seems to be quite independent of the volume control setting. It has even

been known to occur when nothing at all was being played, such as when a record is being changed.

This effect, coincidentally or otherwise was first seen to occur at the onset of winter. I was thus led to believe that air dryness resulting from indoor heating created static somewhere, somehow. I grounded all components to earth by way of a heavy lead to a waterpipe or heating radiator. There was no apparent effect, or lessening of the fault.

In consideration of the fact that the source of this "bug" might well be the a.c. power line itself, may I mention that I live in an apartment, one of eight in a single unit. The cause could be the starting up of a refrigerator or other appliance or perhaps even an intermittent shorting or defect in the main wiring somewhere, resulting in abrupt line surges. I state this because I have, at times, noted a sudden dimming of the room lights, the instant that the offending "crack" sounded from the speakers, a dimming such as is common when an appliance is switched on, although, of course, I have not been able to determine whether one phenomenon causes the other or is the result of it. John P. Wright, Montreal, Canada.

A. I definitely rule out static as the cause of your difficulty. I have known of instances where static electricity builds up on moving tapes, discs, and turntables. However, you have indicated that the "crack" can be heard when nothing is playing. I have not known static to cause interference to audio systems from any other cause.

I suspect that the trouble is caused by the transient created along the power line when the furnace ignites. If it is not the furnace, it is probably a refrigerator. The fact that the lights dim down at the time the transient is heard indicates a substantial drain along the power line.

It is probably the transient nature of the voltage demand which causes the noise in your system, rather than the drop in voltage. The problem then resolves itself into how to eliminate the transient condition from your equipment. There are on the market some LC filters which are designed to be placed between your amplifier and the line. Possibly such filters can eliminate your trouble. I am not altogether confident that their use will help.

It is far better to get right to the source of the trouble and eliminate the problem there. A capacitor-resistor combination placed across the contacts of the thermostat or probably better yet, across the relay contacts which actuate the burner motor might be successful. An effective filter can be made using a 0.02 μf capacitor placed in series with 200 ohms. The capacitor should be rated at 1 kv and the resistor should be rated at two watts. This combination is then placed across the switch contacts of the offending appliance. The addition of this combination will not impair the performance of the appliance.

A refrigerator is harder to correct because of the inaccessibility of the thermostat and the relay (if the refrigerator happens to use a relay—some do).

I do believe that the furnace is most likely the culprit because the onset of the problem concides with the onset of winter.

I have seen some cures effected by reorienting leads in the preamplifier so as to keep them away from a.c. power lines. Sometimes bypassing each side of the line to the chassis helps. 0.02 μf is probably a good choice for such a purpose, rated at 600 v.d.c.

In my previous apartment I used to have this trouble and I just had to live with it. The furnace was responsible first, but was added to by some old refrigerators whose thermostats needed to be replaced. In my case, however, grounding of the music system components did help somewhat, but not enough for my liking.

Has any reader completely solved a similar problem? Æ

ELECTRONIC ORGAN TONE COLORING

(*from page 35*)

ments over air-organ sounds.

These approaches give the organ purchaser a tremendous freedom of choice of organ sound when selecting an electronic organ.

Before closing, we might add a word about devices that have been designed to hasten the mechanics of picking up stops during rapidly played passages. The Crescendo pedal will be discussed in detail in a later article. Here we will only say that it is a mechanism for picking up stops in a programmed manner. A foot operated cam engages stops sequentially as they are required. In pistons, groups of stops are operated as one, so that effectively the stops are ganged.

A relatively simple device is used by Kinsman for this purpose—a punched card programs the stops when the card is inserted. (See *Fig.* 12).

Fig. 12. Kinsman stop-programmer.

This completes our discussion on tone generators and tone-shaping networks. In the next article, we will discuss techniques and devices for modifying the sounds we have discussed and means for generating special sounds.

TO BE CONTINUED

RCA VICTOR "DYNAGROOVE" SYSTEM

(*from page 46*)

geometrical configuration of the cutter of *Fig.* 18. A major portion of this difference in angles may be attributed to bending of the recording stylus resulting from the drag force produced by the record material being removed from the groove as shown in *Fig.* 20. A minor portion of the difference in tracking angle disparity may be attributed to longitudinal lacquer springback. The two effects introduce essentially the same type of tracking angle deviation.

The tentative standard angle for stereophonic disc records is 15 degrees. Accordingly, a suitable nominal geometrical tilt angle was introduced in the Wextrex Recorder and other modifications of the cutter were incorporated so that an effective angle of 15 deg. was cut in the master record. The "Dynagroove" records are cut under this standard condition.

Lacquer Original Recording Console

A recording console has been designed for recording the lacquer original from the submaster tape. The submaster tape is recorded in a manner designed for transfer to the original lacquer record without any appreciable modifications except for that introduced by the Dynamic Styli Correlator. Therefore, the manual controls are comparatively simple. The lacquer original recording console includes the Dynamic Styli Correlator and Recording Overload Indicator. The Recording Overload Indicator is included to check on the signal applied to the lacquer record.

Disc Record

The minimum number of, but nevertheless significant, steps between the original lacquer and the final disc record are depicted in *Fig.* 1. Studies of the plastic for the production of the disc record has been carried out in a joint research program with the manufacturer of the plastic. One significant development has been the new revolutionary electrical conducting plastic[31] which helps to keep the record dust free and thereby reduces surface noise. Everyone is familiar with the attraction and accumulation of dust on conventional records due to the tremendous electrostatic charges and resultant voltages developed on the surface of the records. As a result of the research and development in the production of records, the signal-to-noise ratio of the surface of the commercial "Dynagroove" records is now more than 65 db.

Reproduction of Sound from "Dynagroove" Records in the Home

In the Radio Corporation of America there are many listening rooms which simulate the acoustic of living, recreation and music rooms in homes. These are located in the RCA Laboratories, RCA Victor Record Division, the RCA Victor Home Instruments Division, the Broadcast and Communications Division and the National Broadcasting Company. All of the listening rooms are equipped with disc record reproducing equipment of high quality termed "reference sound reproducing systems."

The turntable used is the RCA BQ-2C. The wow and flutter are less than 0.25 per cent peak-to-peak.

The pickup is the RCA MI 11866-7. The response frequency characteristic is uniform to within ±1 decibel from 30 to 15,000 cps. The crosstalk separation between channels is more than adequate to insure stereophonic sound reproduction with good auditory perspective.

The amplifier used in each channel of the system exhibits less than 0.1 per cent nonlinear distortion up to 20-watts output. The response is uniform to within a fraction of a decibel over the frequency range of 30 to 15,000 cps.

The loudspeaker mechanism used is the RCA LC1A mounted in the RCA LS11A cabinet. The design of the cabinet has been carried out with the objective of reducing the deleterious effects of difraction. The response is very uniform from 30 to 15,000 cps. Another important loudspeaker characteristic in stereophonic sound reproduction is the directivity pattern, which should be independent of the frequency over the listening area in order to provide faithful audi-

[31] G. P. Humfeld, "Control of Static Electricity on a Phonograph Record," *Journal* of the Audio Engineering Society, Vol. 10, No. 4, p. 290, 1962.

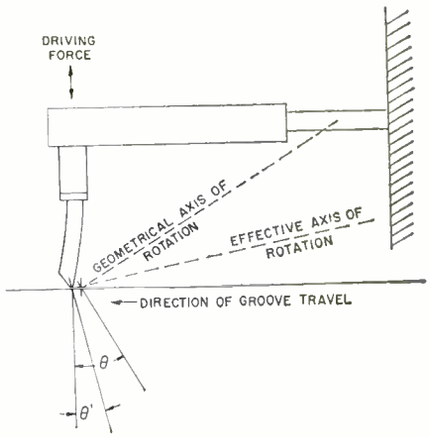


Fig. 20. Diagram showing the geometrical axis and effective axis of rotation of a stereophonic disc recorder.

tory perspective. Over a total angle of 90 deg. which covers more than the listening area there is no significant frequency discrimination due to the directivity characteristic. The nonlinear distortion is another important loudspeaker characteristic. As mentioned in the section on the Dynamic Spectrum Equalizer the peak sound level of reproduction in the home is 80 decibels. The RCA LC1A loudspeaker will deliver sound levels of 80, 90 and 100 decibels in a typical living room for electrical inputs of 0.05, 0.5 and 5 watts respectively. For all of the inputs the nonlinear distortion above 150 cps is a fraction of a per cent. If the distribution of the components in music with frequency are considered the resultant distortions will be imperceptible even for a level of 100 decibels.

In addition to the subjective tests on the reference equipment both objective and subjective measurements were also carried out on all manner of commercial phonographs again with the main objective in mind.

The general conclusion was that the original objective has been achieved, that is to provide the listener in his home environment with sound reproduction which exhibits the highest order of artistic and subjective resemblance to that of the live rendition. Æ

Acknowledgments

The "Dynagroove" concept was initiated, developed, directed and implemented by George Marek, Vice President, RCA Record Division. In a project of this magnitude it is impossible to list all who contributed to make it a success. Those intimately concerned with the artistic elements and tasks of the "Dynagroove" project include: John Pfeiffer and J. A. Somer of the RCA Record Division. Those intimately concerned with the scientific elements and tasks of the "Dynagroove" project include: R. L. McClay, H. E. Roys and D. L. Richter of the RCA Record Division and J. G. Woodward, John Volkmann, E. C. Fox and R. W. George of the RCA Laboratories.

ABOUT MUSIC

(*from page 14*)

crooked his left index finger, beckoning for more, and the trumpets immediately obliged with a shade more level on the second phrase. The clarinet melody beginning in bar 401 was a decibel or so too loud, and phrased with not quite enough pliancy; Solti tilted his chin in the direction of the clarinet, made a curvy movement with his left hand, and brought about a reduction in level and a smoother legato. It should be pointed out, of course, that most of Solti's indications merely remind the players of instructions given during rehearsals.

At the other end of the dynamic spectrum, Solti aimed his arm at the brass in a rapier thrust, and the brass attacks hit the ear with thrilling force. And to produce a major crescendo, Solti would lower both arms to his waist, then raise them gradually, as if lifting a massive stone. One could hear the players translate Solti's pantomime into aural terms.

Always in motion, Solti seemed to think of as many different movements as the nuances he evoked from his complex instrument. He was a tennis player negotiating a deft forward stroke, a juggler balancing a vase in the palm of his hand, a violinist with his finger quivering in vibrato, a fencing master lunging toward his opponent, a forester swinging a huge ax, a cat clawing at a sofa, or a frugue dancer at a discothèque, jerking his head and elbows.

To the layman, it might have seemed as if Solti performed his podium dance exclusively for the audience; none of the players appeared to pay much attention to him. Orchestral musicians, however, develop amazing peripheral vision, which enables them to keep both the score and the baton in sight. "It's as if we had another pair of eyes in our scalps," a bassoonist put it. What about the dance itself, the layman might ask; is the conductor merely acting out the musical score for his own pleasure? According to the English conductor-author, Frederick Goldbeck, "dancing gesture stands paramount as a means to make the players dance with the maestro —that is, express the music as he wants it to be expressed."

For well-balanced sound, avoid the front rows at an orchestral concert, by all means. But if you're forced to sit up close, you might have a good time, provided the conductor's mobile and you're not sitting directly behind him. Æ

COMPRESSOR

(*from page 20*)

happen. Naturally since the output has been reduced, the control light becomes dimmer and allows the signal to be greater. However, the light has a slow recovery time, 800 ms. Therefore, a general tapering off occurs. A switch can be placed in the lamp circuit to disable the lamp when it is desired to have the circuit function normally without compression.

The a.c. VTVM circuit is mainly conventional, however no provision for calibration is incorporated and a control is placed in the input circuit so accuracy is minimal. The VTVM is not used for accurate measurements but rather as an indicating device so that relative voltages in the lamp circuit may be observed. The VTVM is used in a manner similar to a VU meter.

Installation of the Compressor

If the system in which the compressor is to be incorporated has a program amplifier and a preamp, the ideal method of connecting would be to place the "signal" portion (photocell) between the preamp and the program amplifier and connecting the "control" portion (lamp) to the output.

This circuit could also be placed at low level inputs. However, care should be taken with regard to shielding so that hum and noise will not be at an objectionable level.

Before making connections to the amplifier, the following should be observed: First the amplifier should have no signal present; second connect the input section first; third the control potentiometer should be adjusted to short circuit the lamp; then a signal should be introduced into the amplifier to establish the proper operating level; then increase the signal slightly and adjust the control potentiometer until the signal is restored to the proper level by action of the compressor. The grid control potentiometer and the lamp control potentiometer interact so repeated adjustments will be necessary to establish the desired operating parameters. It is most important that the indicating series light be observed carefully during all adjustments so that the filament specifications will not be exceeded.

Using a Raysistor, or equivalent device, for the heart of the system the compressor can be made as elaborate or simple as the constructor desires. The Raysistor alone connected to the output and input circuit gives you 60 db of compression for less than fifteen dollars.

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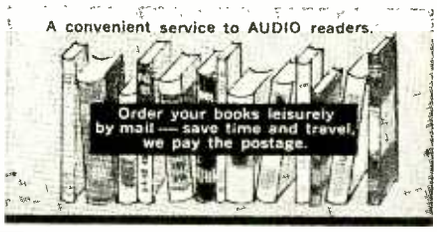

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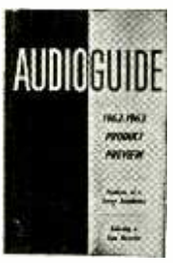

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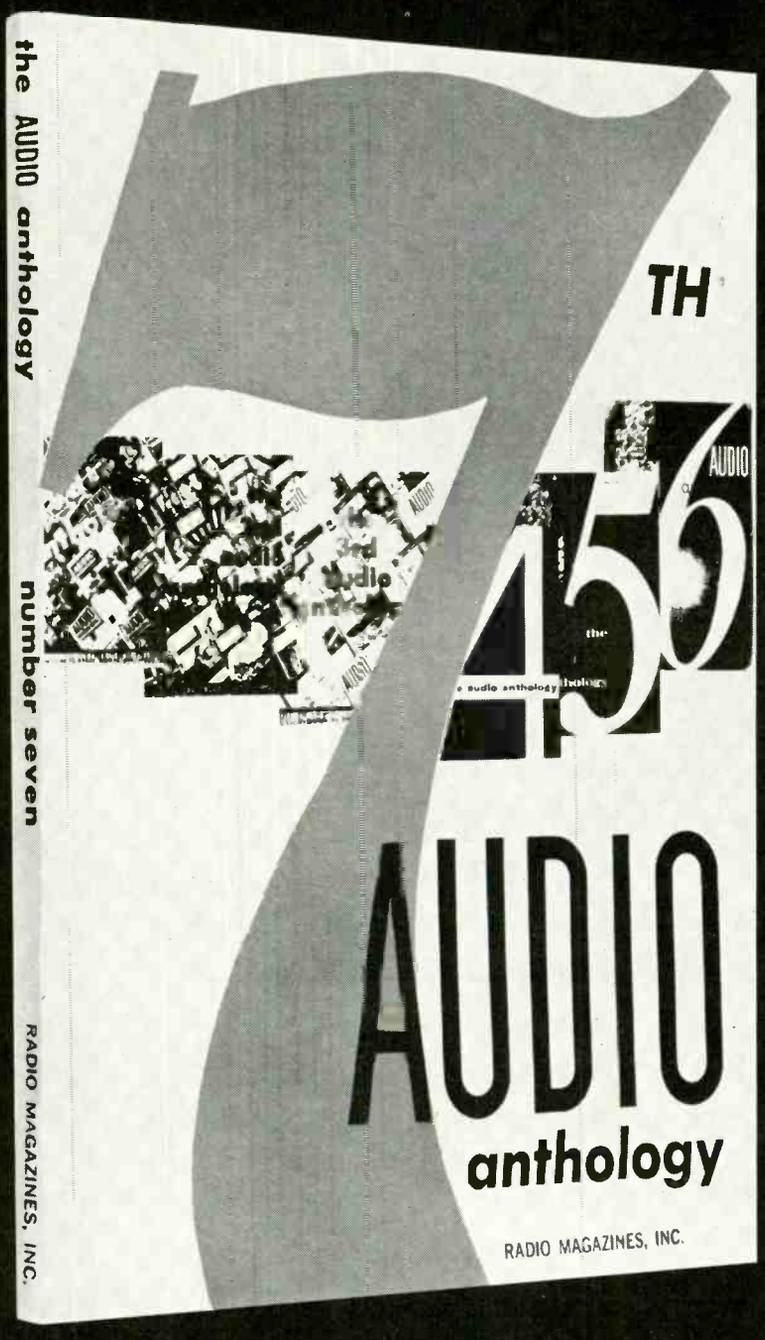

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JAZZ

(*from page 8*)

corded several times, and it was desirable to use the best portions of each take to avoid as much of the inner groove distortion and the various noises that were on the originals. First a court stenographer was called in to make a complete transcript of all of the words in the songs. This transcript ran to about 150 pages. Next, Jac Holzman, head of Elektra, carefully marked the transcribed text, using a special code he has devised, to note each inflection used by Woody. And from this marked text, an editing diagram was prepared, indicating where each cut would be made. An editor spent five days making a rough cut from this diagram.

In three days of recording, it was to be expected that much of the same material was covered several times over. Finally, the splices were all fine cut, and pauses were eliminated. Removing clicks and pops was the next operation; over 1,000 such sounds were cut out of the master tape.

Before the master was ready for cutting; however, it was necessary to prepare a cutting cue sheet so that portions of songs taken from different discs could be balanced and filtered to smooth out differences between sections with swishes, hisses, rumble and fry and those sections that were clear. Armed with his cue sheet, Holzman went to Mastertone Studios, where he and engineer Sid Feldman spent an entire day cutting the set. Holzman propped his 2-in. thick cutting script on a music stand and cued Feldman on levels while making balance adjustments at a bank of Pultec equalizers. For about ten hours, two pairs of hands turned knobs and flipped switches with split second precision.

The next step was to get a set of the discs to Guthrie, who has been hospital-

ized for a long time. He had been told of the plan to release these recordings, and he had insisted that he be the first to hear the final pressings. According to Lou Gordon, Harold Leventhal and Pete Seeger, who are the trustees of the Guthrie Children's Trust Fund, to which all royalties from this recording will be paid, Woody was deeply moved at hearing them.

What emerges from these three hours of long playing interview and song is more than a portrait of a fine folk singer, song writer and itinerant social critic. This recording is a powerful documentation of the depression years in the Oklahoma and Texas Dust Bowl. Through these conversations and songs, a slice of American history that had become a pale memory to most of us emerges with fresh immediacy. The southwestern United States in the thirties is graphically delineated; such matters as poverty, bank foreclosures, the Colored "situation," migration to California, chain gangs and hoboing are described in anecdote and song.

Some other recordings may offer a more polished representation of Woody as a performer, but never before has it been possible to see so clearly and closely the depth, charm and open honesty of this man who Alan Lomax describes in his introduction to the set's attractive booklet as "the best ballad-maker to come down the American 'pike . . . in many ways . . . the most truthful and most talented man of his generation." (Elektra Records EKL-271/272)

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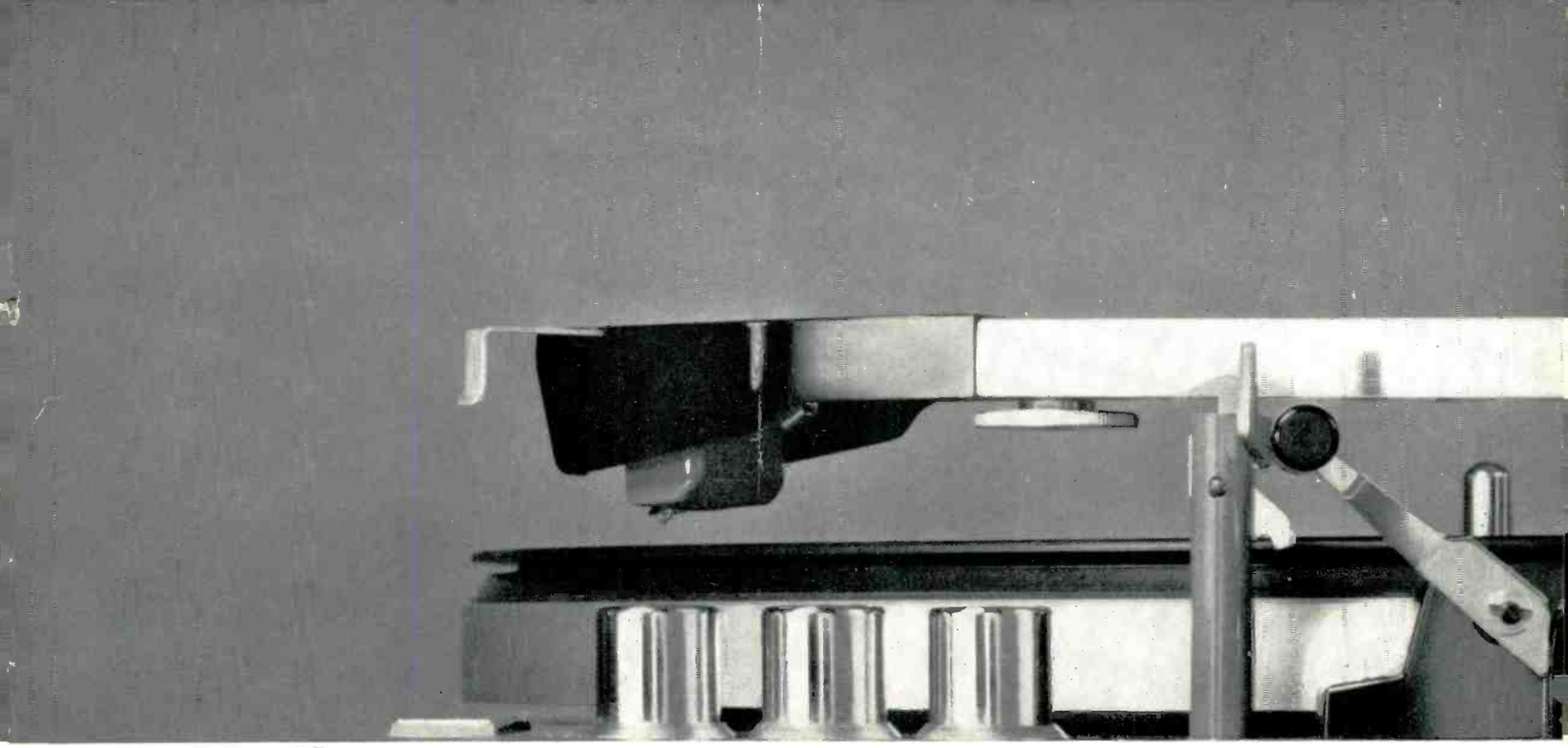

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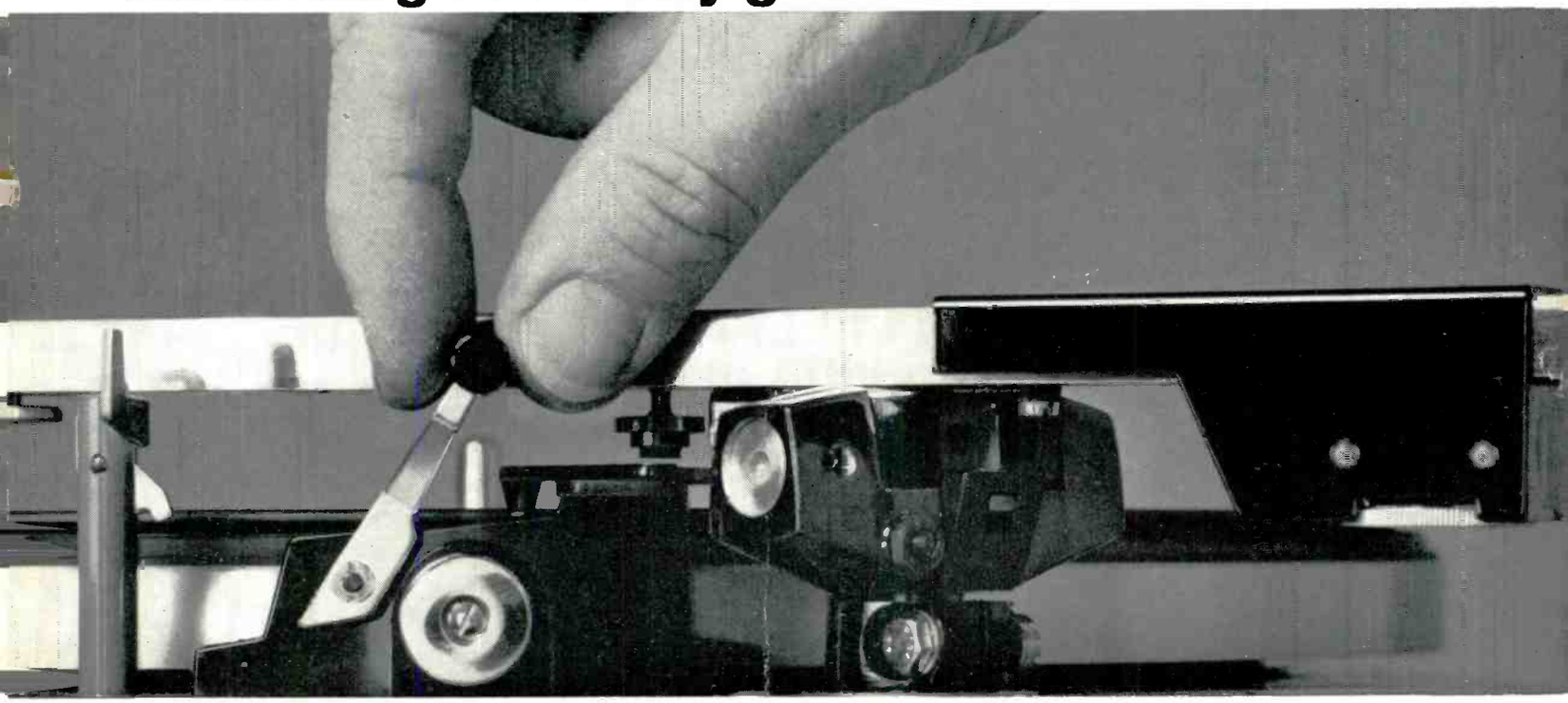

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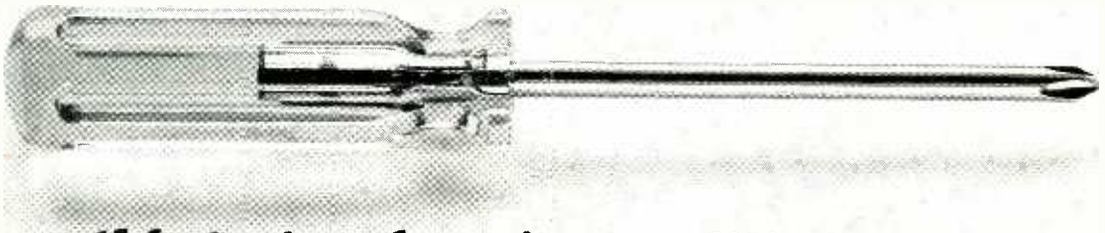
(A)  (B)

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