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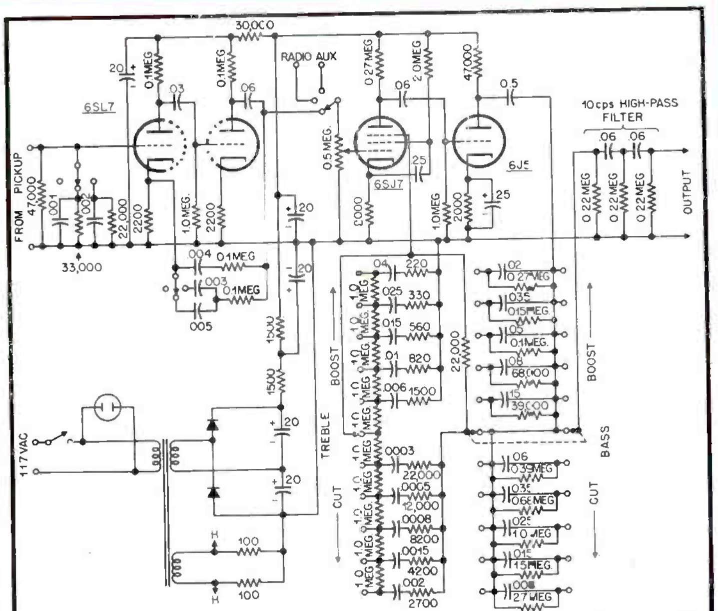
AUDIO

ENGINEERING MUSIC SOUND REPRODUCTION



Tailoring a "front end" to your own specifications can be fairly simple if you take the trouble to study types of equalizers and tone controls as one author did before building the one he liked best. See page 17.

Thorough study of the effect of stereophonic sound on listeners is important before developing the service commercially. See page 56.

LOUDSPEAKERS and ENCLOSURES -- A 31-page section

SIMPLE HIGH-QUALITY PHONO AMPLIFIER

STEREOPHONIC LISTENER TESTS

ACCURATE DESIGN OF PHONO EQUALIZERS

audiotape

TRADE MARK

on Mylar* polyester film offers you these important new advantages

*Du Pont Trade Mark

1. many times stronger
2. withstands extreme temperatures
3. impervious to moisture
4. maximum storage life
5. most permanent magnetic recording medium ever developed

Audiotape on "Mylar" polyester film provides a degree of permanence and durability unattainable with any other base material.

Its exceptional mechanical strength makes it practically unbreakable in normal use. Polyester remains stable over a temperature range from 58° below zero to 302° Fahrenheit. It is virtually immune to humidity or moisture in any concentration – can be stored for long periods of time without embrittling of the base material.

The new polyester Audiotape has exactly the same magnetic characteristics as the standard plastic-base Audiotape – assures the same BALANCED PERFORMANCE and faithful reproduction that have made it first choice with so many professional recordists all over the world.

If you have been troubled with tape breakage, high humidity or dryness, Audiotape on "Mylar" will prove well worth the somewhat higher price. In standard thickness (1½ mil), for example, the cost is only 50% more than regular plastic base tape.

Ask your dealer for our new folder describing Audiotape on "Mylar". Or write to Audio Devices, Inc.

PHYSICAL PROPERTIES

"Mylar" polyester film compared to ordinary plastic base material (cellulose acetate)

PROPERTY	1 Mil "MYLAR"	1.5 Mil "MYLAR"	2 Mil "MYLAR"	1.5 Mil Acetate
Tensile Strength, psi	25,000	25,000	25,000	11,000
Impact Strength, kg-cm	90	170	200	10
Tear Strength, grams	22	35	75	5
Break Elongation, %	80	95	105	20
Softening Point, °F	464-473	464-473	464-473	149-230
Moisture Absorption, % (at 100% RH)	0.3	0.3	0.3	9.0
Bending Modulus, psi	500,000	500,000	500,000	350,000
Flex Life, cycles at 0° F	20,000	–	–	500

AUDIO DEVICES, Inc.

444 Madison Ave., New York, 22, N. Y.

Export Dept., 13 East 40th St., New York 16, N. Y., Cables "ARLAB"

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AUGUST, 1954 VOL. 38, No. 8
Successor to RADIO, Est. 1917.

AUDIO

ENGINEERING MUSIC SOUND REPRODUCTION

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CONTENTS

AUDIO (title registered U. S. Pat. Off.) is published monthly by Radio Magazines, Inc., Henry A. Schober, President; C. G. McProud, Secretary. Executive and Editorial Offices, 204 Front St., Mineola, N. Y. Subscription rates—United States, U. S. Possessions and Canada, $3.00 for one year, $5.00 for 2 years; elsewhere, $4.00 per year. Single copies 50c. Printed in U. S. A. at Business Press, Inc., 10 McGovern Ave., Lancaster, Pa. Entered as Second Class Matter February 9, 1950 at the Post Office, Lancaster, Pa. under the Act of March 3, 1879.

RADIO MAGAZINES, INC., P. O. Box 629, MINEOLA, N. Y.

AUDIO PATENTS

RICHARD H. DORF*

A NEW VOLUME-LIMITER CIRCUIT invented by Edwin C. Miller of Havre, Mont., is of interest among this month's crop of patents because it is simple, yet highly effective. Volume limiters are used in broadcasting and recording to prevent overmodulation or to keep average modulation fairly constant. This one is particularly useful for additional purposes because of its very close control of output over a large input range. The circuit will maintain audio output constant within 1 db over a 20-db range of input voltages. This makes it useful, for example, in test generators with inherently nonlinear amplitude-frequency curves and, in one of my own special applications, in an electronic musical instrument in which pitch is varied by substituting capacitors across a tuning coil. At each frequency the Q of the coil is different and in most oscillators this causes a variation in level. The Miller circuit would keep level constant from tone to tone.

The patent number is 2,679,626, and it is assigned to Northwest Radio Consultant Service, a partnership in which Mr. Miller probably participates.

Figure 1 shows the single-tube circuit of the limiter; the two triodes are in the envelope of a 6SL7-GT. There are two paths for the signal from the input. One path goes through C_2, C_1, R_7, and C_4 directly to the output terminals. This is the main signal path, the capacitors and resistor being included for d.c. and a.c. isolation, respectively, from other elements of the circuit.

V_2 is placed directly across the output and its plate resistance constitutes the shunt leg of a voltage divider of which R_7 is the series leg. Knowing that, it is easy to see that the output amplitude can be varied by varying the plate resistance of V_2, easily done by varying the d.c. on its grid.

Input signal also passes through R_1 to the grid of V_1. Since the V_1 plate is connected directly to B-plus it looks at first glance like a cathode follower, but the cathode resistor R_5 is bypassed by C_6. The high-value cathode resistor causes a high bias which places the point of operation of V_1 close to cutoff and well below the linear portion of the tube

* *Audio Consultant, 255 West 84th Street, New York 24, N. Y.*

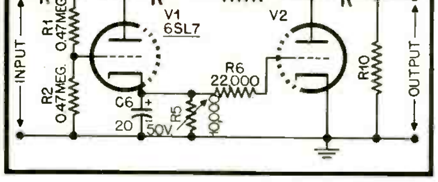


Fig. 1

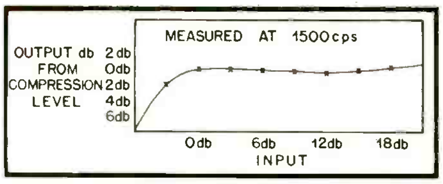


Fig. 2

signal on the V_1 grid is to produce a positive d.c. cathode voltage which is proportional to the audio input amplitude.

The d.c. V_1 cathode voltage is applied to the grid-cathode circuit of V_2, causing grid-current flow. As the input audio amplitude rises the effective grid cathode resistance of V_2 is reduced approximately logarithmically. This lowers its plate resistance; since this plate resistance is the shunt leg of the signal output voltage divider, the audio output is likewise reduced.

This is an entirely practical circuit which anyone can build without difficulty (technically speaking—the legal difficulties are up to you). Adjustment is very simple. Apply a 1,000-cps audio voltage to the input terminals. Set R_5 (which you might make a 15,000-ohm potentiometer, as the value given in the diagram is the approximate setting value) at maximum resistance and increase the audio input to the point where compression begins—where output no longer rises along with input. Measure the input voltage; then increase the input voltage to five times the measured value and adjust R_5 to provide an output voltage equal to one fifth the new input voltage. *Figure* 2 shows roughly the variation in output for given input variations.

The patent specification shows a second, somewhat more complex circuit designed along the same lines. It may be more suitable for program applications in that it produces a fairly fast compression on peaks and a slow recovery. As usual, 25 cents sent to the Commissioner of Patents, Washington 25, D. C., will get you a copy.

The Official Gazette

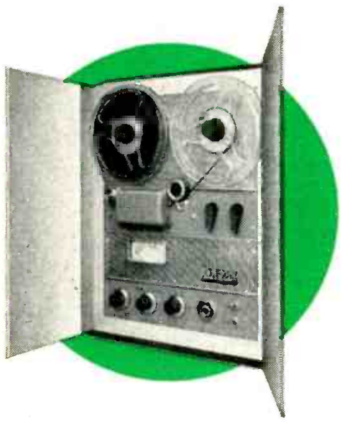
There may be those among habitual readers of this department who wonder how the writer comes across all the suitable patents to describe. The source is the Official Gazette of the U. S. Patent Office, issued weekly and on hand at all major libraries (also available by subscription).

The Patent Office Gazette lists all patents granted for the week, showing the major claim and the main drawing for each. Divided, as it is now, into General and Mechanical, Chemical, and Electrical sections, it is a prime source of information and inspiration for workers in every field, and is heartily recommended to the attention of possessors of inquisitive and original minds.

an investment in performance and pride

the AMPEX 600

The Ampex 600 is a design triumph that is sure to please adventurers in recorded sound everywhere. The performance that has made Ampex the supreme name in professional recording has been engineered into a machine of truly portable size at a reasonable price. You will love your new "600"; your friends will admire it; it will give you faithful service for many years to come.



From its portable case, you can easily transfer the Ampex 600 to a permanent installation in your home music system.

CONCERT HALL QUALITY . . . *living room comfort*

The Ampex 600 captures the full glories of the human voice or the sweeping majesty of symphonic orchestrations. It records and reproduces with such clarity and flawless accuracy that highly perceptive ears seldom can distinguish the tape recordings from the live performances. This superlative quality results from a broad frequency response, wide dynamic range, imperceptible distortion, uniform tape motion, and freedom from amplifier hum and noise.

- *Frequency response 40 to 15,000 cps at a tape speed of 7½ in/sec.*
- *Signal-to-noise ratio over 55 db.*
- *Flutter and wow under 0.25% rms.*
- *Timing accuracy ± 3.6 sec. in a full half hour recording.*

Never before could so much be recorded so well on a machine so small.

HOBBIES IN SOUND for *pleasure and pride*

Weighing less than 28 pounds, the Ampex 600 can be used anywhere, by anyone: no professional recording engineer is needed. Use it at home to record the musician in your family, the choir at your church, "wedding bells" and parties, or the performances of your talented friends. Use it "outside" to record bird calls, waterfront and railroad whistles, foreign dialects, folk songs, and other rare voices, music and sounds.

- *Separate microphone and line input recording channels.*
- *Quantitative VU meter for accurate program levels.*
- *Phones jack for monitoring during recording and "mixing."*
- *Record safety button to prevent unintentional erasure.*

Your Ampex 600 is engineered for years of true high-fidelity performance at a truly reasonable price, $545.

For the Finest in Recorded Sound, see your local Ampex Distributor today

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DISTRIBUTORS IN PRINCIPAL CITIES

(Listed in the Telephone Directory under "Recording Equipment")

NEW LITERATURE

• **Shure Brothers, Inc.,** 225 W. Huron St., Chicago 10, Ill., has greatly eased the problem of determining microphone sensitivity with a new Microphone Sensitivity Conversion Chart, which permits interchange of values in the three most commonly used systems. The chart is recommended to those engaged in buying, selling, or using microphones. Relative ratings can be obtained in a few seconds. The chart will be mailed on request.

• **G & H Wood Products Company,** 75 N. 11th St. Brooklyn 11, N. Y., announces the publication of a new, colorful catalog of the Cabinart line of high-fidelity equipment cabinets, speaker enclosures, and unfinished kits. This booklet is exceptionally well-produced, including excellent illustrations, dimensional information, finish specifications, and prices. Copy may be obtained from local dealers in high-fidelity equipment, or by writing direct to the company.

• **Frank L. Capps & Co., Inc.,** 20 Addison Place, Valley Stream, N. Y., will mail on request literature describing a new series of Capps condenser microphones. Data and characteristics are given on three new models, as well as description of associated power supplies and cables.

• **International Rectifier Corporation,** 1521 E. Grand Ave., El Segundo, Calif., lists ratings and specifications on the company's germanium diodes in Bulletin GD-1A. Included in the bulletin is a complete replacement guide of International germanium diodes for replacing various RETMA types. Available on request.

• **Yale Industries Corp.,** 82-09 251st St., Bellerose 26, N. Y., announces the release of a catalog sheet which fully describes the construction and operation of its Model TS-4 cutter-splicer for magnetic recording tape. The TS-4 cuts rounded indentations in the tape edges, giving the tape a "Gibson Girl" shape, leaving the edges free of adhesive. Copy will be mailed on request.

• **Photocircuits Corporation,** Glen Cove, N. Y., has prepared one of the most valuable publications of its kind in a new brochure which describes printed circuits, their functions, fabrication, and applications. The two-color eight-page booklet includes comprehensive information on printed-circuit materials, electrical characteristics and components. Assembly with dip soldering and plated-through holes is also described. Design improvements and lower production costs for many types of equipment are amply suggested in this excellent booklet which will be mailed on request, free of charge.

• **G. H. Leland, Inc.,** Dayton 2, Ohio, is distributing on a monthly basis a series of engineering sketch sheets with the purpose of acquainting design engineers with various applications for the company's Ledex rotary solenoids and circuit selectors. These reports should be of considerable value to engineers who face problems involving snap-action power or circuit selection. To get on the mailing list engineers should write the address shown above.

• **Telex Inc.,** Telex Park, St. Paul, Minn., describes the new Telex dynamic and magnetic pillow speakers which the company has recently introduced, in a two-color catalog sheet which will be mailed on request. Specifications and advantages of both units are listed along with suggested uses in hospitals, homes, hotels, and commercial transportation vehicles.

• **Rhein Sound Systems, Inc.,** 2 Coburn Ave., Orlando, Fla., lists the company's complete line of 27 amplifiers and associated equipment in a new catalog folder which will be mailed on request. Featured in the literature is a well-planned amplifier selection chart which will greatly assist both distributors and users in choosing the proper amplifier for a given application.

• **Insulated Circuits, Inc.,** 115 Roosevelt Ave., Belleville, N. J., announces the availability of Bulletin No. 106 on Carry-Through Printed Circuits, a four-page leaflet which describes many facets of printed circuit production. Included are detailed photographs and drawings of printed circuit patterns. Copy will be mailed on request.

Only the new MIRACORD XA-100

has the "MAGIC WAND" plus every other feature you have ever wanted in an automatic record changer

The "MAGIC WAND"

In the MIRACORD XA-100 you find a three-speed automatic changer where every component is held to the closest tolerances of high precision—where wow and rumble are eliminated. The results achieved provide everything the most exacting individual could wish for.

The MIRACORD XA-100 comes complete with the "Magic Wand" and single play spindles. An automatic record spindle for 45 rpm is available as an accessory. Every unit is shipped completely assembled with leads and plugs attached ready for operation. The MIRACORD XA-100 is superbly finished in rich burgundy with surf-white trim.

The "MAGIC WAND" spindle has been developed after many years of research, and its operation is remarkable. It eliminates stabilizing plates and pusher arms. The life of your records is infinitely prolonged!

THESE ARE THE FEATURES YOU'VE ALWAYS WANTED!

- Simple control-at-a-glance through four push-buttons: Start, Pause, Filter and Repeat.
- Completely automatic operation at 33⅓, 45 and 78 rpm; and ten and twelve-inch records can be intermixed.
- Built in filter to suppress surface noise of old records.
- Continuous record repeat if desired.
- Adjustable mechanism to pre-set pause between records.
- Simple thumb screw adjustment compensates for cartridge weight (No tools necessary).
- Specially engineered four pole motor insures rumble-free operation.
- White rubber matted turntable.
- Silent automatic shutoff.
- Constant speed guaranteed whether one or ten records are on the turntable—insures correct pitch—eliminates wow.
- Record capacity: 8 12" records or 10 10" records (or a corresponding mixture of both) or 10 7" records.
- Ball bearing suspension of turntable.
- The plug-in head has been specially designed to accommodate your choice of cartridge.
- By inverting the single play spindle the same record will repeat continuously.
- The tone arm is constructed of a specially developed noise-reducing plastic and also has ball bearing suspension which eliminates any displacement of the stylus and guarantees equal pressure of the stylus.
- Replacement parts and accessories are available for immediate delivery.
- The MIRACORD XA-100 and MIRAPHON XM-110 (see below) are guaranteed for six months against defective materials and workmanship.

AUTOMATIC CONTROL!

START: This button starts the mechanism. It also enables you at any time to change to the next record.

FILTER: Push this button to eliminate surface noise caused by old records.

REPEAT: Push this button to repeat the record being played.

PAUSE: Now you can automatically select a "pause" time between records of from five seconds to five minutes! The selection setting appears at a window directly above the push button. "Pausamatic" is a unique MIRACORD XA-100 feature. Here is your choice of pause lengths (in seconds) between records:

SETTINGS

RPM	0	1	2	3	4
78	5 sec	35 sec	70 sec	105 sec	140 sec
45	9 sec	60 sec	120 sec	180 sec	240 sec
33⅓	12 sec	82 sec	164 sec	246 sec	328 sec

The manually operated
MIRAPHON XM 110
incorporates all the MIRACORD's high quality sound reproduction characteristics in a single play unit.

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Londoner's Letter

RICHARD ARBIB*

Mr. Arbib, who has contributed so many London Letters *to our recent issues of* AUDIO, *has just completed a six-weeks' trip in the U.S.A. Before leaving for England he sent us a dictated tape which gives a Londoner's impression of American audio.*

EVEN A BRIEF EXAMINATION of American audio makes it clear that you have advantages and disadvantages compared with us in England. You have, of course, much larger population and a more wealthy middle class. A visit to some of your radio shops, audio show rooms, and record stores was a revelation to an Englishman. Visits to England of Sam Poncher (Newark Electric Company, Chicago) and Dave Ormont (Hudson Radio, New York) during the past year provided an excellent excuse for me to see them on their home ground. I do not believe that in Europe we have any radio store as large as Newark Electric Company in Chicago, and I believe that that is not necessarily the largest in the whole of America. The method of using telephone relays at the Hudson showrooms to enable instant comaprisons to be made between twenty-five tuners, twenty-five amplifiers, and as many loudspeakers and record players was indeed very ingenious and far more elaborate and efficacious than anything I have seen elsewhere.

I found it equally interesting to see how one and a half million records could be sold per year in not a very large store. I refer, of course, to Sam Goody's in New York, which was quite unique—as far as I was concerned—in that records were being sold without being heard but with the proviso that the customer could, if he or she desired, bring the record back after one playing and exchange it for another. Visits to stores of this kind cannot be other than invigorating to an Englishman, for our market is so much smaller that no English shop can operate on such a large scale as your American ones. On the other hand, conditions in England do not make it essential for people to have to expend so much on records and hi-fi equipment.

I thought that the average quality heard from an American television set was really appalling. With the exception of the Hoffman television receiver, which I heard in Los Angeles, most of your American sets appear to use speakers of only five or six inches in diameter. As your sound is broadcast on FM, this really seems to be a terrible waste of facilities which are available for good quality. On the other hand, to an Englishman, so many of the programs are mediocre that I can well imagine that many Americans turn to records for good quality.

Although all hi-fi enthusiasts are not necessarily lovers of classical music, I imagine that most of them are so. Consequently, your lack of live broadcasts of symphony orchestras on radio or television places your hi-fi enthusiasts at a disadvantage compared with us in England. We can always rely on at least one symphony concert being broadcast each day live from one of the three BBC programs. Consequently, the urge for us to buy records is to a great extent not as imperative as it is in America.

On the other hand, there are quite a number of commercially produced radio-gramophones in England—what you call combinations in America—which can really be classed in the hi-fi market. Practically every British radio and television manufacturer markets at least one of such instruments, whereas it seems that many of your radio and television manufacturers did not include any combinations in their line of models. Advertised claims by some American manufacturers that their products were "high fidelity" seemed in some instances to be an exaggeration, and I imagine that you must be careful that disrepute is not brought upon high fidelity by exaggerated claims. Some of the statements in regard to frequency range of tape recorders at slow speeds, to me seemed to be very remarkable. You either have engineers who are able to produce quite different results from tape than any other technicians, or else it is just the result of American advertising technique.

I found that so-called music was used as a background much more in America than in England. I thought that often the best advantage was not made of the facilities offered. For example, on the Super Chief train there are loudspeakers, quite well baffled, in practically all the rooms on the train. It must have been an expensive matter to complete this installation, including intricate switching in each room so that the occupant can avail himself of radio programs or classical or light music programs. Yet the quality of the classical and light music programs was appallingly poor. An investigation of the method by which the programs were provided showed that it was taken off wire recordings, probably designed four or five years ago. Surely, this should be replaced by modern tape machines. I was privileged to visit the local headquarters of the Muzak organization in Los Angeles and was more than interested to learn that the new tape equipment will have a frequency range extending to 15,000 cps. Yet in many of the places in which the Muzak service is used, I feel certain advantage is not taken of the high-quality reproduction which should be available. The future for hi-fi in America cannot be other than bright, and I was interested to be able to buy one of the first tape recordings issued by RCA. Obviously, there is going to be one problem with these tape recordings and that is—the matter of length. I was disappointed to find that a reel of the Tchaikovsky *Fifth* was only occupied by music on one and two-thirds sides. The purchaser of a gramophone record can see quite clearly if only part of one side is filled up with music, whereas the buyer of a tape is unable to discover this fact without first hearing it. I was also very surprised to see that the reel was not labeled in any way. Surely, reels of recorded tape should look different from unrecorded reels. They should be of a different color and the name of the piece and the composer should be stenciled or permanently printed thereon.

It is obvious that in a country where there are two hundred different record manufacturers and so many producers of tuners and amplifiers and loudspeakers that the outlook for hi-fi cannot be other than bright. It is certainly most invigorating for an Englishman to be privileged to examine, unfortunately only too briefly, the audio market in America.

* *Multicore Solders, Ltd., Hemel Hempstead, Herts., England.*

We'll sell many customers
through its rich sound...
the "artistic" performance
of this entirely new recording tape!
Just listen and compare to appreciate
its freedom from distortion and noise...
its balanced high and low frequency response...
its unvarying output from reel to reel...

Many others will buy it
for its scientific engineering
and material quality...literally every
inch precision tested and laboratory bonded
for uniformity, optimum sensitivity,
and adherence of oxide coating...
your continuous satisfaction guaranteed
through maximum tensile strength,
shelf-life and splice-ability.

Still others will insist upon it
for convenience and value built-in
above and beyond its superior
audio performance. Every reel comes
to you in a resealable polyphane
plastic bag, with spliced color-coded
leader for extra cue time,
and 5 full inches of Tuck splicing tape...
all these extra features at no extra cost!

...and this
extra performance
and extra value
will make new

Encore

the household word in recording tapes

(Encore means more in recorded results time after time...Just try it!)

Developed and manufactured by the Magnetic Products Division TECHNICAL TAPE CORPORATION Morris Heights 53, N. Y.

LETTERS

Ears, Speakers, and So On

SIR:

I would like to thank Professor Joseph J. Antonitis for his valuable information on the construction of the human ear (LETTERS, April, 1954) in which he mentions that on the surface of the basilar memberane are complex formations of cells, and the most important of which is the Organ of Corti which includes the hair cells. These are thought to generate impulses in the auditory nerve when set in motion.

I wonder if Professor Antonitis ever stops to ponder on the fact that the cells, *thought* to generate impulses in the auditory nerve when set in motion, *might* be numerous small independent resonators and transducers that are *capable of converting mechanical energy to electrical energy.* (I strongly believe that the nerve system is, in fact, an electrical system in a human body.)

As I said before in my article, "A Perfect Loudspeaker," what we aim to do is to create a completely new electro-mechanical transducer, (or driver, as Prof. Antonitis puts it), which, by *imitating the characteristics* of the human ear system, can reproduce sound of various frequencies, in the form of true replica. I am at a loss to understand what induced Prof. A to think that I did suggest to build a replica of the ear as my dream loudspeaker. His suggestion that a *conventional* electro-mechanical driver should be used to build a completely new system of electro-mechanical transducer would need a sounder theory to make it even feasible.

Scientifically speaking, I should think that we ought to be able to find our way in this world of sciences, with the blessing of inspiration of course, to look for something more than meets the eye.

LEUNG, CHO YUK
52, North Side,
Clapham Common,
London, S.W. 4,
England

The Inscrutable East

SIR:

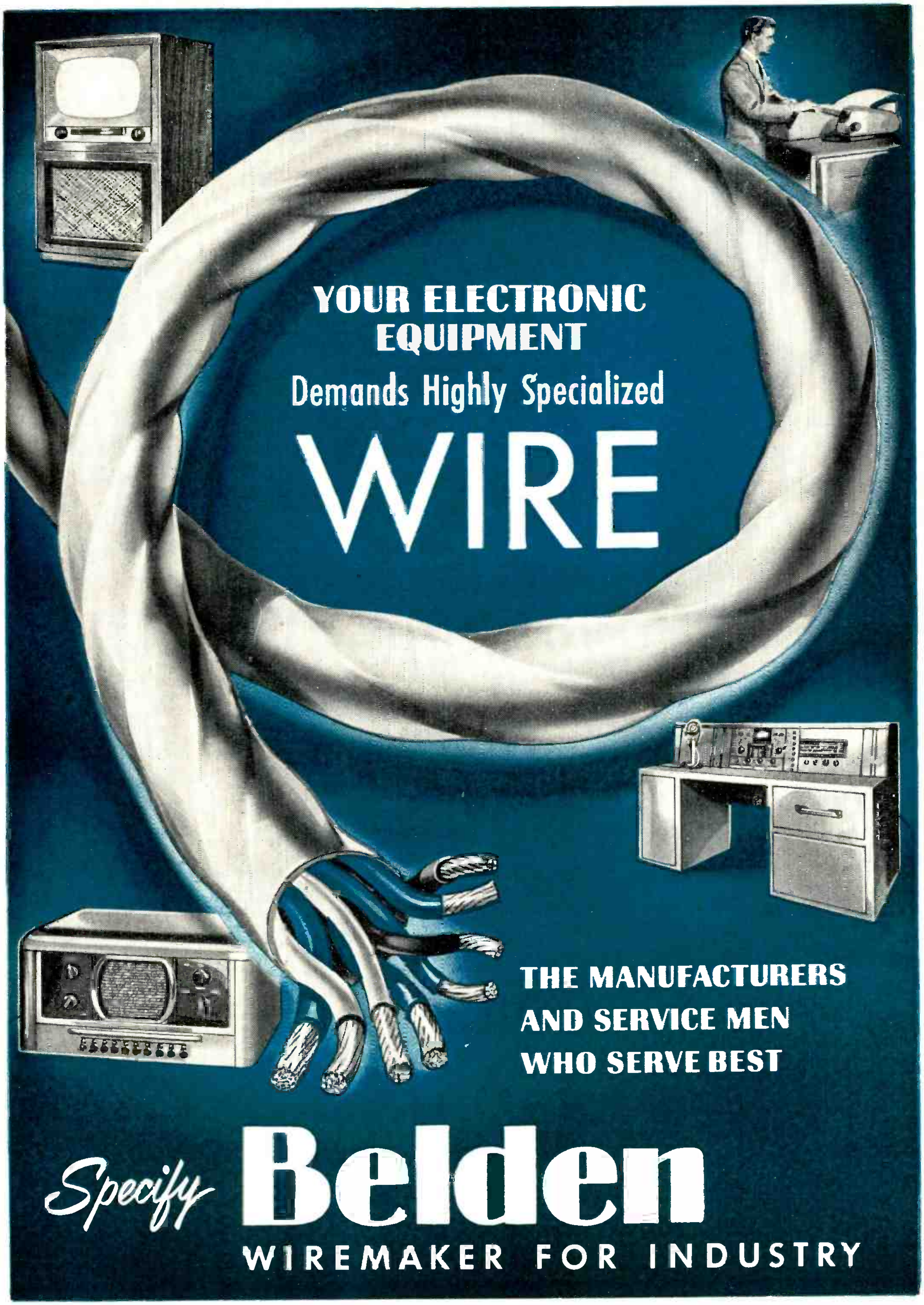
Having been globe-trotting these many moons, I have been away from my dear old AUDIO and have just gotten through six monthly issues. I thought "The Great Loudspeaker Mystery" would put the cat among the pigeons, and enjoyed Mr. Leung Cho Yuk's esoteric contribution on the subject. Since Professor Antonitis so competently disposed of the physiological aspect of the matter, may I content myself with disposing of *his* whole thesis in a few words?

To design a speaker on the basis of the design of the human ear is what a metaphysician would call a pseudo-science because it is a science based on a fallacy. The ear is a device whereby sound waves are caused to actuate the nervous system and send signals to the brain. Nobody knows what those signals are; we know only that the effect on the brain is such that mentally we develop the concept that what has actuated the ear is something we call "sound." A speaker, on the other hand, causes the air to vibrate when electrical currents are applied to it, and the best we can venture to say is that if the ear's impulses through the nerves are electrical currents, then a speaker is the reverse of the ear, and should therefore be designed like a mouth, while a microphone should be designed like the ear. (*May we interject the mention of the reciprocity theorum?* ED.) Mr. Leung seems to suspect this in the earlier part of his article, but then turns a somersault, like Karl Marx turning Hegel on his head.

Oddly enough, my notion of the flexible diaphragm seems to fill the bill all ways, for it could on the one hand be considered as an essembly of Mr. Leung's 24,000 discs, while on the other it somewhat resembles Professor Antonitis' sheet of fairly soft gelatine. As any scientific theory can be considered a valid working hypothesis until an exception has been found, I humbly persevere with my curious notions until someone trips me up.

H. A. HARTLEY
62, Latymer Court,
Hammersmith,
London, W. 6, England

(It is to be hoped that since these two correspondents are, if not countrymen, at least in the same city, they might possibly get together and hammer something out that the Smiths and Joneses can understand. To carry this a bit along, let's further hope that neither gets so vehement in his theories that the local constabulary might clap 'im in some common gaol for bodily assault. And may we mouth an "'ere's 'ow" to cheer them on. ED.)


YOUR ELECTRONIC EQUIPMENT
Demands Highly Specialized
WIRE
THE MANUFACTURERS
AND SERVICE MEN
WHO SERVE BEST
Specify Belden
WIREMAKER FOR INDUSTRY

Phase Inverter

SIR:

This is in defence of the cross-coupled input and phase inverter condemned by Mr. George Varkonyi in the May issue of AUDIO. In my experience there is none finer. Among the "self-balancing" inverters, the cross-coupler is the only one that really does it, low frequencies or high. With special H.F. tubes, matched as to their capacitances, it reaches into the sonar region where the split-load inverter cannot follow because "the total shunt capacitances across the two channels necessarily differ."

May I suggest that Mr. Varkonyi's unsatisfactory experiences are due to the associated circuits rather than to the cross-coupled inverter? Indeed, Mr. Marshall's "Golden-Ear" amplifier, quite admirable in conception, shows all the earmarks of blind experimentation without any 'rithmetic or design behind it. One instance of this is the curious inconsistency in specifying ± 1% components and then wantonly upsetting the balance thus achieved by a unilateral feedback loop which shunts the input cathode to earth and thus produces an ohmic unbalance of 3000 to 2680, or 11%. Even so the results might not have been so disastrous as observed by Mr. Varkonyi but for the direct coupling to the drivers which extends the initial static unbalance to three stages and even increases it.

Unfortunately, this is not the only source of distortion. Far more serious is the slapdash treatment of the whole feedback economy which actually creates conditions inimical to clean reproduction on all but the very lowest levels. At a superficial estimate, the total amount of feedback, not counting that of the cathode followers, is 35 db. But about two-thirds of this constistute uncontrolled current feedback in the unbypassed cathodes: 6.2 db in the voltage amplifiers, 16.2 db in the drivers. Owing to the reduced gain, the inner loop carries, not the 16 db envisaged by Mr. Marshall, but merely 0.8 db, i.e. a futile amount not worth the price of those resistors and capacitors. The external loop, similarly weakened, delivers the moderate amount of 12 db. Nor is there any possibility of testing the amplifier with 40 db of over-all feedback as Mr. Marshall claims to have done: after all those other feedbacks, there simply is not that much gain.

This unusual design of the feedback system places both amplifying stages at a grave disadvantage especially as Mr. Marshall favors output tubes with a 60-plus volts input. At the reduced supply voltage available, a 6SN7, or a 12BH7, or a 5687 tube in the driver stage could possibly do the job with a 6 or 8 per cent distortion (which would be reduced to one-seventh of this amount by the 16.2 db plus 0.8 db feedback). But the 9002 tube chosen by Mr. Marshall is not cut out for such strenuous work: it will distort considerably more. However, worse is still to come. In view of the reduced gain in the driver stage, the phase inverter (or its voltage amplifier) is called upon to deliver 25 volts per plate. But this it was never meant to do, in spite of Mr. Marshall's breezy assurances that the inverter would deliver 50 volts "distortion-free." If we consult the Sylvania Technical Manual, we find that, with a supply of 100 volts, the 12AX7 is capable of delivering 4.6 volts with a 4.1 per cent distortion, and this at the grid current point beyond which angels fear to tread. With the 150 volts provided in the "Golden-Ear," we may expect 9–10 volts, which is a far cry from the 25 volts required by the design. Nor may we expect help from the feedback (6.2 db plus 12 db at this point). The latter is calculated to decrease gain and distortion eight times as well as to make the grids accept an eightfold burden. But as soon as we press the tube beyond the output indicated, distortion increases by leaps and bounds. So we have no hope of limiting the distortion of this stage to 0.5 per cent (i.e. 4.1 per cent divided by 8). However, with a higher supply voltage, the tube might have a chance: at 250 volts, it could deliver 24.5 v. with a 0.6 per cent distortion (i.e. 4.95 per cent divided by 8). And with a supply of 300–400 volts the results would be even better. But Mr. Marshall has thrown away invaluable volts in order to operate his gimmick of a VR tube as a "stabiliser" in a circuit which is perfectly stable without it. So, if Mr. Varkonyi has noted as much as 18 per cent of intermodulation distortion and, in square wave tests, found "almost unrecognisable" waveshapes, the explanation is not far to seek. But it is equally clear that the cross-coupled inverter has nothing to do with it.

In fact, Mr. Varkonyi himself supplies some evidence of the inverter's inherent excellence. With the unbalanced feedback loop connected to the off grid of the input stage (and, presumably, with a more reasonable circuitry in the driver stage), he reports 0.2 per cent of intermodulation distortion with 18 db of feedback at 10 watts. Such a result is very HiFi, in fact, better than Leak's "Point One" (which only claims 0.1 per cent of harmonic distortion). Similar evidence was recently offered by Mr. Charles P. Boegli, of the Cincinnati Research Company, whose tests, published in a contemporary, show that the cross-coupled inverter (6SL7) delivers 15 volts with 0.8 per cent of intermodulation without any external feedback. With an 18 db loop, this would result in 0.1 per cent.

But the most conclusive evidence is offered by Mr. Ulric J. Childs' amplifier from which both White's "Powertron" and Marshall's "Golden-Ear" derive their inspiration. Featured three years ago in one of your contemporaries, it was carefully tested and re-examined by an independent investigator last year. The square wave patterns published in this connection have very straight sides and no rounding of corners, with little noticeable overshooting even at higher frequencies. Mr. Varkonyi may be interested to learn that, even at the phenominally high square-wave frequency of 40,000 cps at 30 watts, the square-wave shape is clearly recognisable. Of the cross-coupled inverter, which is largely responsible for these results, the author says: "It leaves little to be desired. . . . The intermodulation distortion was so low that it was impossible to obtain a reading on the analyser."

Of course, the phase inverter alone can work no magic unless the associated circuits are worthy of it. A cursory examination of Mr. Childs' brain child reveals that it was designed as carefully as the famous Williamson. To begin with, it does not sacrifice efficiency on the altar of an extra direct coupling whose benefits would be illusory. In spite of two balanced double loops, the inverter operates painlessly at 300 volts, and the drivers eschew undue distortion as they are cursed neither with 6AR6's nor with 16 db of wild current feedback. It is apparently dangerous to tamper with such an integrated design, as Mr. Williamson has pointed out elsewhere. But there is no reason why Mr. Marshall's (or Mr. White's) idea, even with that extra direct coupling, could not be made to work equally well, provided equal pains were taken to define and to integrate the numerous variables.

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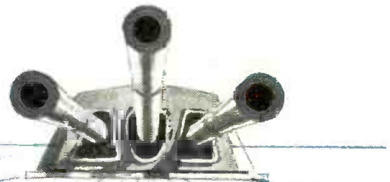

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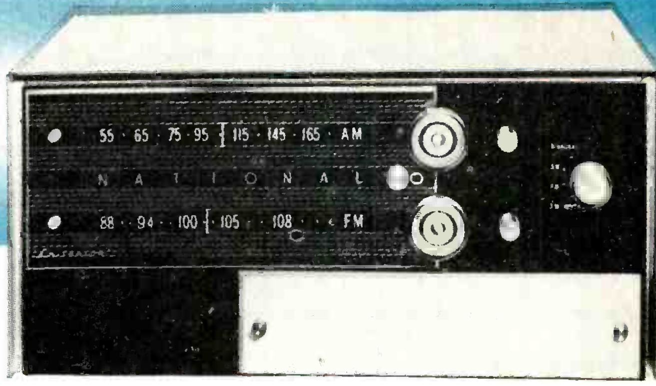

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Incorporating the revolutionary new unity-coupled circuit in a 10-watt amplifier design, the HORIZON 10 offers performance never before achieved at such a moderate price!

The built-in preamp-control unit offers a choice of 3 inputs, 3 record equalization curves, a loudness control and separate bass and treble controls.

Harmonic distortion is less than .5%; intermodulation distortion, less than 2% at rated output. Frequency response is ±1 db, 20 cps to 20 kcs.; power response, ±2 db, 20 cps to 20 kcs. Hum and noise are better than 70 db below rated output on high-level input, better than 50 db on low level input.

HORIZON *5*

PREAMPLIFIER-CONTROL UNIT $49.95

The HORIZON 5 achieves a new high in frequency response (±1 db, 20 cps to 100 kcs) and voltage output (up to 10 volts) — a new low in distortion (less than .2% harmonic, .3% intermodulation)!

Four inputs, 7 record equalization curves, a loudness-volume control and bass and treble controls are provided.

Entire unit slips quickly, easily into either the tuner or 20-watt amplifier.

NATIONAL CO., INC., 61 SHERMAN ST., MALDEN 48, MASS.

EDITOR'S REPORT

FM and GOOD MUSIC STATIONS

We are always gratified when we receive a response to remarks in these columns, and it appears that the July page awoke some interesting reactions. The moan by many broadcasters about "Good Music" not paying did bring a definite reaction from a few of the more successful FM stations throughout the country. Our own preference for good music, reasonably uninterrupted by inane commercials—not that all commercials are inane, but no one will deny that many are—quite often constrains us to beg for more music, more FM stations, and better quality of transmission. But against claims that stations cannot live on good music alone, we have no immediate argument, for relatively few business men cherish the idea of operating a radio station entirely on a philanthropic basis.

However, C. M. Edmonds of KCMS-FM in Manitou Springs, Colorado ends his letter with "Does 'Good Music' Pay? You double-darn betcha." KCMS-FM is located in a community of 30,000 radio families; there are two AM stations, two post-freeze TV stations, and the one FM station which is independent. Mr. Edmonds' programming is 60 per cent classical, 5 per cent jazz, and the remainder light concert. He says the competition gives the listeners hill billy and pop vocal and spot announcements by the hundreds while he gives them Beethoven's Fifth for dinner. Thirty per cent of the families that have FM receivers have made their wishes and likes felt and as a result KCMS-FM is in the black.

To all this we say "Hurrah." Perhaps it is even possible in a city of seven million to entertain enough listeners with nothing but the more esoteric of modern composers and still keep alive—although one station in New York tried it and failed—but when a station can survive because of the listening likes of enough listeners *who have FM* in a community of 30,000 radio families, it must prove that there is a market for a quality product, even in the entertainment field. We believe more FM stations should follow Mr. Edmonds' example. Incidentally, he reports that KOA (Denver) recently got 22,000 replies from 46 states about their programs from 11:00 p.m. until 5:30 a.m. (M.S.T.)—over 90 per cent were for serious music.

FM stations might take a leaf from Muzak's book—this organization programs for all its licensees, each one playing the same selection at the same *local* time. A similar programming service could well serve hundreds of radio stations at a fraction of the cost of individual programing, and would probably do a better job of it. We still await the announcement of such a service.

Another project we have a longing for is the establishment of a chair in audio engineering at some university—we believe the industry would gain in the long run by such a step.

FEDERAL ENGINEERING JOBS

It is regrettable that it is unlawful for the Federal Government to purchase advertising space or to solicit advertising where the applicant for the job offered is expected to pay a fee for referral. It would seem that advertising media might possibly receive postal credits or an offset against income tax for the use of space for such advertising. It is not the case, however, so notices of job availability must be carried by magazines and newspapers as a service to their readers.

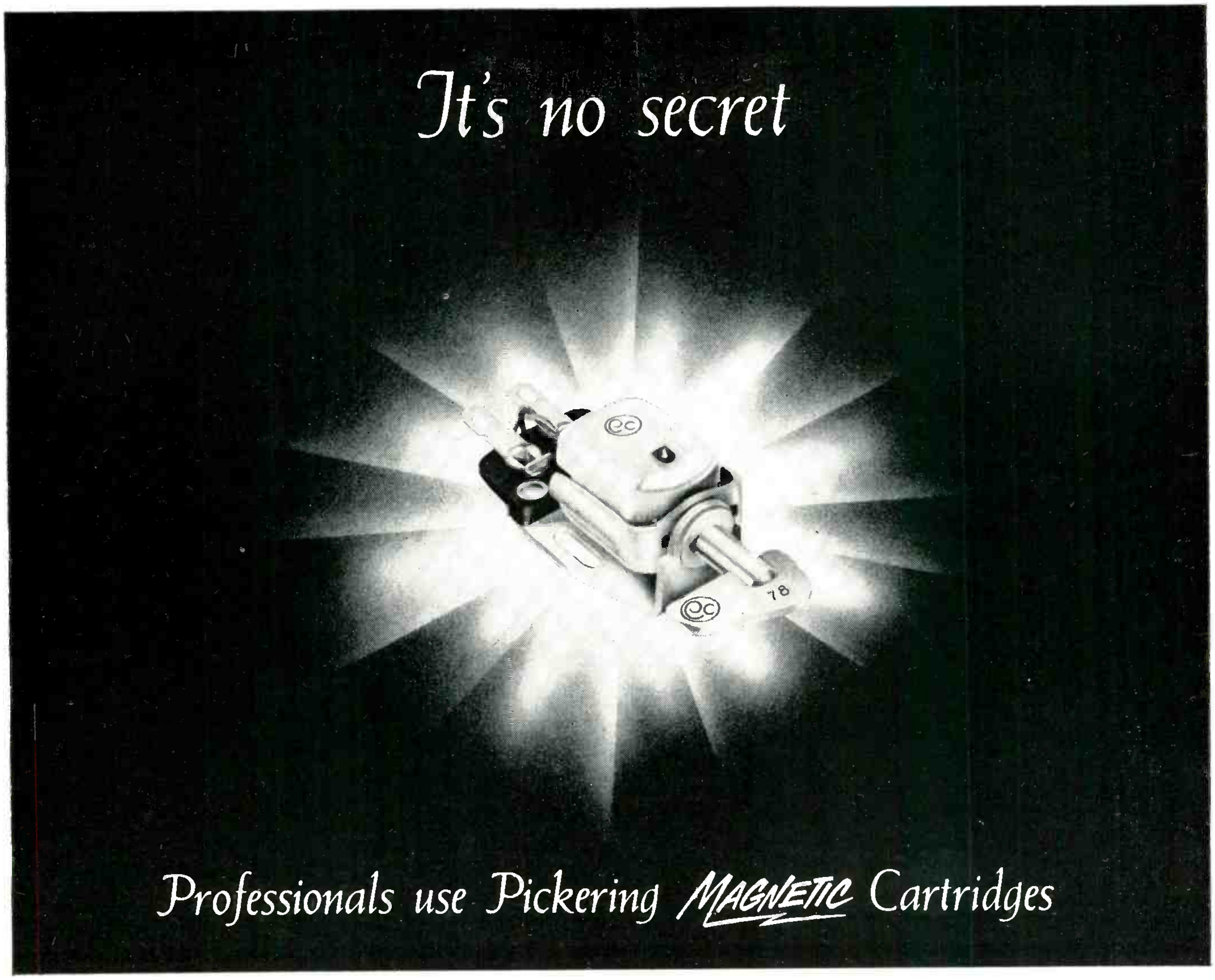
The U. S. Naval Air Station at Point Mugu, Port Hueneme, California, offers a long list of positions for electronic engineers, electronic scientists, physicists, and general engineers at salaries ranging from $8360 down to $3410 per year. A list of the available openings may be obtained by writing Employment Superintendent at the above address.

The Air Force urgently needs Civilian Electronic Engineers and Physicists. If you have experience or training in electronic research and development or in the installation and maintenance of fixed plant facilities, you may qualify for one of these jobs at salaries from $9600 down to $3410. Free on-the-job graduate degree training is offered by Syracuse University College of Engineering. For further information, write Professional and Scientific Recruiter, Placement and Employee Relations Branch, Rome Air Force Depot, Rome, New York.

HOWARD C. ANTHONY

The entire electronics industry was saddened to learn of the death of Howard C. Anthony of Heath Company, Benton Harbor, Michigan, in an airplane crash near Dayton, Tennessee, on July 23. Mr. Anthony and a party of friends, which included Gordon Paxson and the Company pilot Lawrence Durham, were making a demonstration flight in a new 8-place private plane when some undetermined trouble occasioned the crash during a thunderstorm.

Heath Company, of which Mr. Anthony was president, is known principally for its line of test equipment kits, which has expanded in recent years to include audio amplifiers and communication gear. Mr. Anthony, 42, leaves a widow, Helen C. Anthony, who was also his partner in the business. They had no children.



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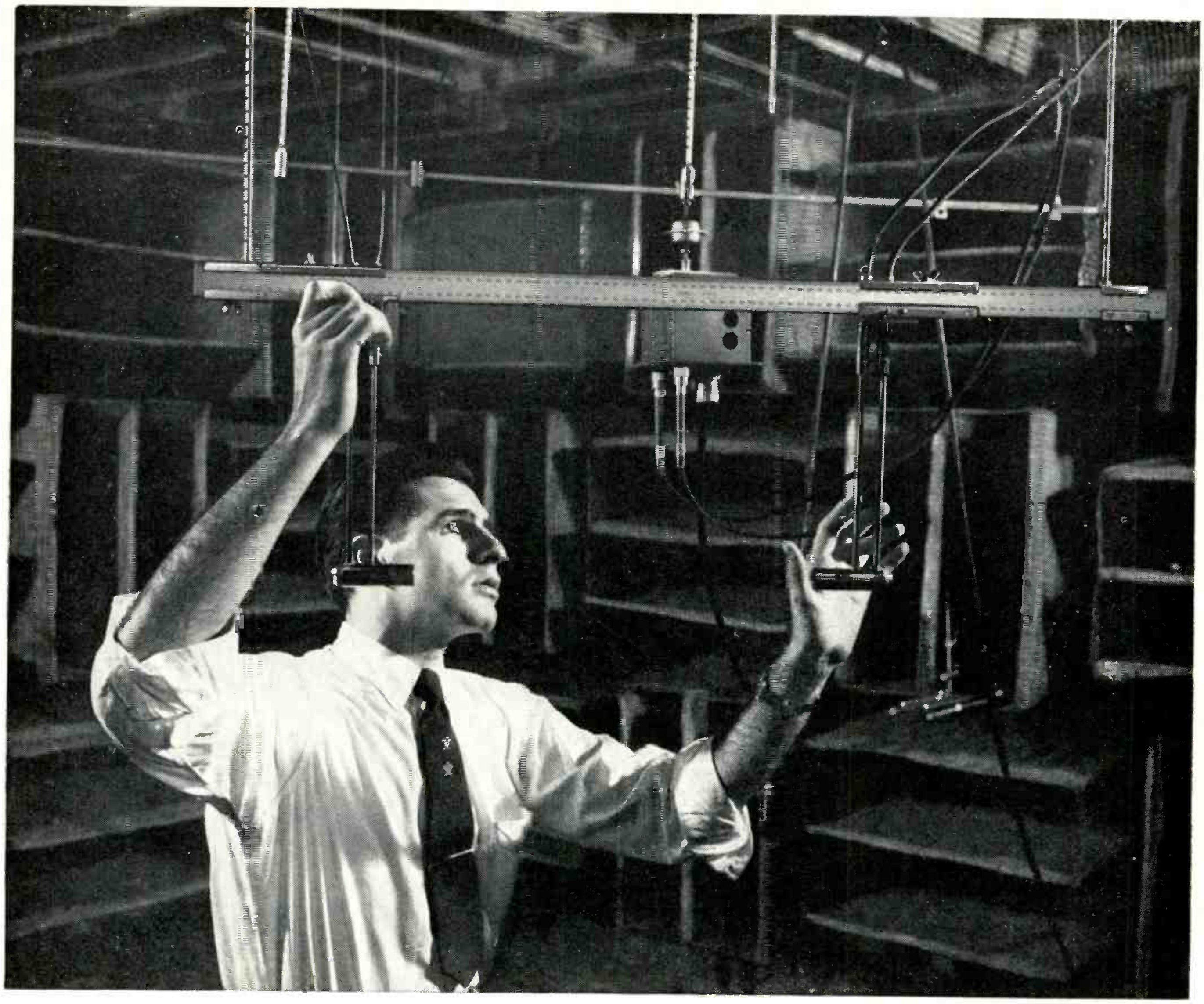

In a quiet room at Bell Laboratories an engineer scales off the distance between two condenser microphones during a calibrating test. Able to measure air pressure variations of a few billionths of an atmosphere, such microphones play a crucial role in the scientific study of telephone instruments.

SOUND STEPS ON THE SCALES

Those small cylinders facing each other are condenser microphones—measuring tools that play a vital part in making your telephone easier to hear and talk through.

They are being calibrated by an engineer at Bell Telephone Laboratories to give extremely accurate information on the kind of sound your telephone company handles. Armed with these vital fundamental data on what sound *is*, Bell Laboratories scientists devise the instruments and equipment that transmit it best.

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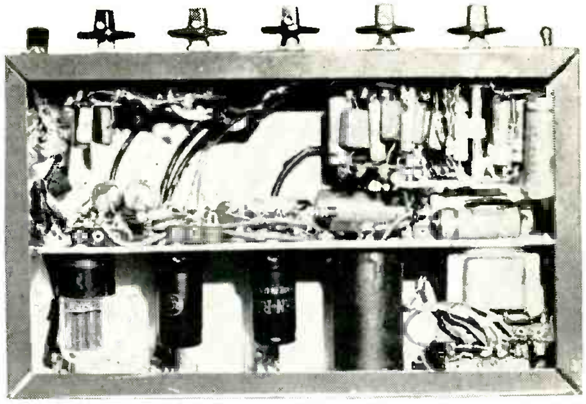

External and internal views of the author's self-powered control unit.

A Low-Distortion Tone-Control Amplifier

W. B. BERNARD, Cdr., USN*

A well-designed "front end" which should provide sufficient flexibility for anyone, since the curves can be tailored to fit by judicious selection of capacitors and resistors.

Over the past few years the author has been working to develop an input amplifier and tone control circuit with fewer than the average number of faults. This article is a story of the evolution of such a circuit and *Fig.* 1 is the family tree of the circuit.

The circuit shown at (A) in *Fig.* 1 is a widely used circuit which provides for boost and cut for both bass and treble. There is no interaction between the two tone controls but the circuit requires that three grids be operated at low signal levels with the attendant danger of hum and at high levels the three triodes without feedback are likely to produce more distortion than the output amplifier that follows them.

The circuit of (B) is a simplification using one high-mu triode. Its only advantage is simplification since the distortion produced was quite high as shown in the author's article on distortion in voltage amplifiers[1] and the low- and high-frequency controls suffered from interaction.

To alleviate these difficulties the circuit of (C) in *Fig.* 1 was developed. The inverse feedback reduced the distortion to a reasonable level and the resistor between the sliders of the bass and treble controls minimized the interaction of these controls.

In general the circuit of (C) was a great improvement over anything previously tried but it was considered desirable to separate completely the bass and treble controls so the treble control was moved to the feedback network and the amplifier of (D) resulted. As the slider on the treble control is moved to the left the amount of high-frequency feedback is reduced and therefore the gain of the amplifier is increased at these frequencies. If the slider is moved to the right the amount of high frequency feedback is increased thus reducing the treble gain of the amplifier.

All the foregoing circuits suffer from one fault. At the midfrequencies there are one or more points in the circuit where a tube is required to operate at 20 db above the desired output level in order to allow for a 20 db boost at high or low frequencies. This excess level adds considerable unnecessary distortion so the next step was to eliminate the necessity for operation at such a high level. The circuit shown at (E) in *Fig.* 1 was designed to produce all compensation but bass cut by variation of the feedback circuit. Because of the practical difficulties of obtaining bass cut by variation of

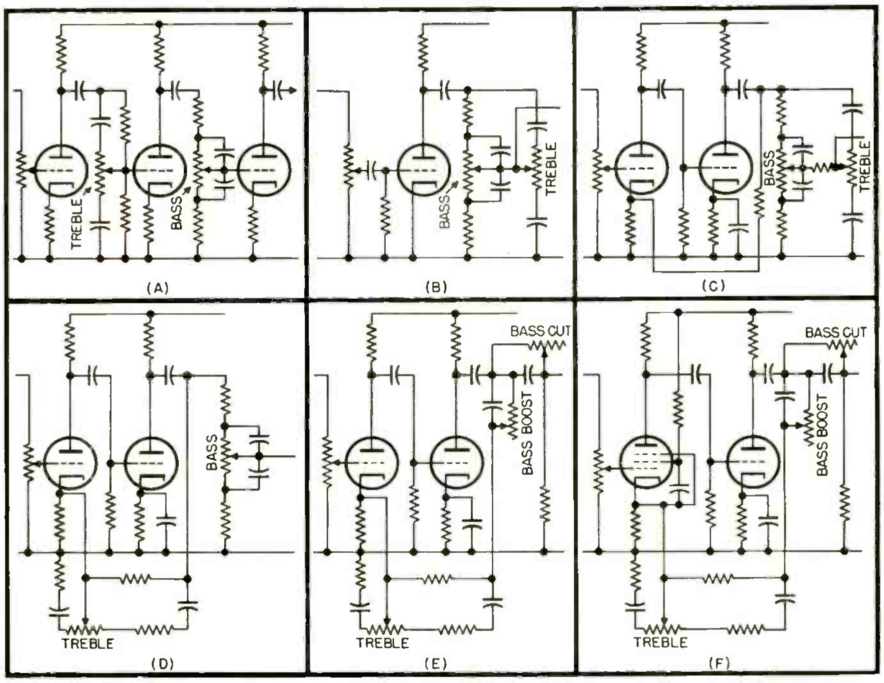

Fig. 1. Six steps in the development of the control unit described by the author. Each of the circuits from (A) to (E) had some faults; all are presumed to be eliminated in (F).

* *4420 Narragansett Ave., San Diego 7, Calif.*

[1] Audio Engineering, Feb., 1953.

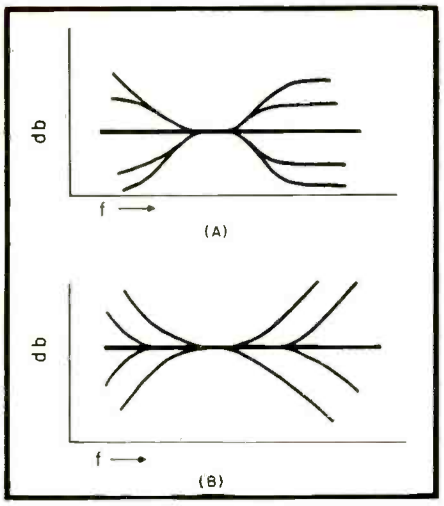

Fig. 2. Using a variable resistor and fixed capacitors in a tone control gives curves like (A). By changing capacitors, leaving the resistor fixed, variable-turnover curves may be achieved, as in (B).

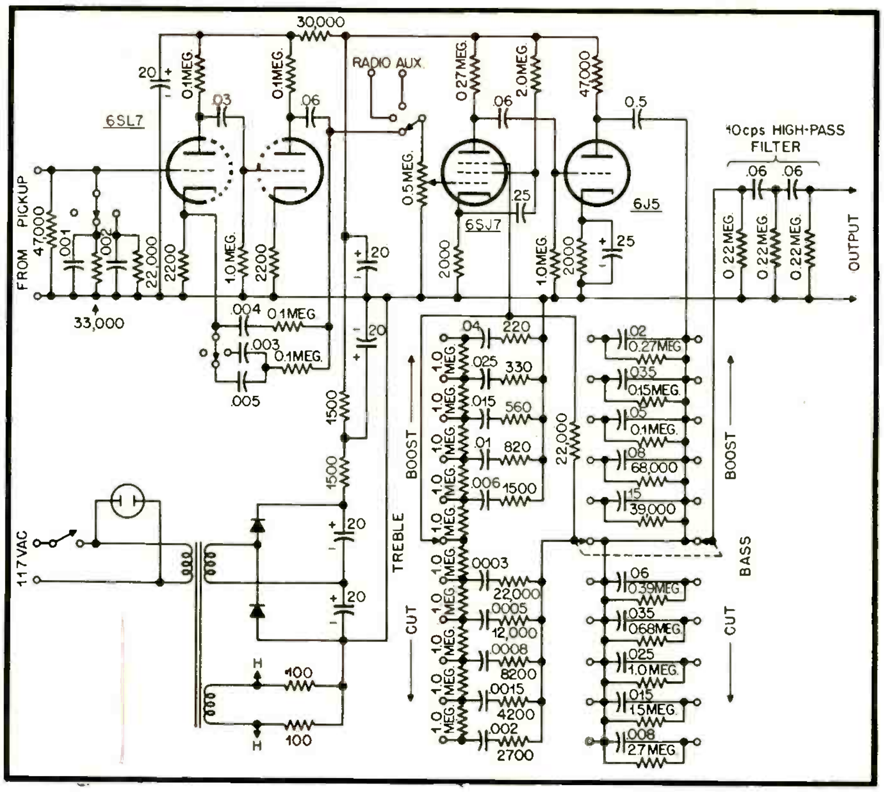

Fig. 4. Over-all schematic of the tone-control amplifier, combined with ordinary preamplifier.

feedback it is accomplished by an RC circuit connected to the output of the amplifier. For the sake of illustration separate bass boost and cut controls are shown but it is of course desirable to combine these on one shaft.

The circuit of (E) was a step forward but it still did not seem to be the optimum design. The requirements for a voltage gain of 10, tone compensation of 20 db, and about 20 db of negative feedback, were not compatible with the operation of the high-mu triodes with reasonable loads. These difficulties were overcome in the circuit of (F). Here a pentode voltage amplifier is used for the first tube and a low-mu triode is used for the second tube.

This circuit fulfills the requirements listed in the paragraph above and has additional advantages. The pentode grid which is fed from the volume control has a very low input capacitance when compared to a triode where the Miller effect has to be considered and the application of the inverse feedback further reduces this input capacitance. As a result, the major portion of the capacitance load on the volume control comes from the wiring and consequently there is no problem of loss of high-frequency response at mid-settings of the volume control. The output impedance of the low-mu triode is quite low and it is further reduced by the inverse feedback so any length of output cable within reason will have no appreciable effect on the high-frequency response. A calculation shows that under the worst condition (maximum treble boost of 20 db) the output impedance of the amplifier is about 600 ohms. This goes down to less than 100 ohms if flat response or treble cut is used. These impedances are as good as or better than would be realized by a cathode follower using the same tube.

The fact that a single potentiometer can not be used for both bass boost and cut may be considered to be a disadvantage but the use of a step-type control also has some advantages in that each step can be designed to give any desired curve up to the maximum of 6 db per octave correction. A variable resistor with one fixed capacitor gives curves which all have the same turnover frequency and initial slope as shown at (A) in *Fig. 2*. By changing the capacitor and leaving the resistor fixed we get curves which have the same slope but different turnover frequencies as shown at (B). Only by varying both *R* and *C* can we obtain the curves with different slopes which is the result usually desired. This makes it desirable to use a step-type control for treble boost and cut although in this case a single potentiometer would serve. By a special arrangement of switch wafers the resulting assemblies are quite compact. The maximum compensation and one intermediate curve are shown in *Fig. 3*.

Figure 4 shows the diagram of the complete unit which was constructed. The phonograph preamplifier section is conventional and will not be described in detail. In addition to the three choices of low-frequency turnover compensation, a sharp-cutoff filter as described by Pickering[2] is incorporated at the input of the preamplifier. The values shown are for a GE cartridge. A 10-cps high-pass filter is incorporated at the output of the amplifier to reduce turntable rumble and to prevent low-frequency transients from unduly disturbing the power amplifier or causing speaker cone excursions of undesirable amplitude. Such transients are prone to occur if a radio tuner which is operated into the system is tuned from station to station with the volume control turned up to a reasonable level. It may be stated as a general rule that there is no reason to feed into the main amplifier any

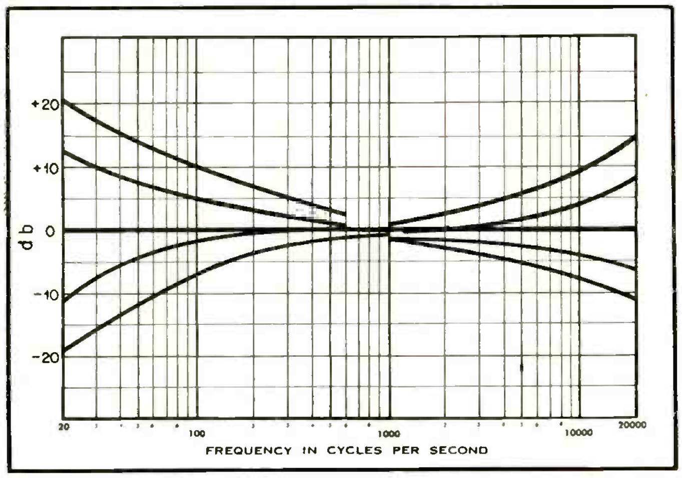

Fig. 3. Limit curves from the author's unit, with one intermediate curve shown to indicate trend of control operation.

[2] N. C. Pickering, "Effect of load impedance on magnetic pickup response," AUDIO ENGINEERING, March, 1953.

(*Continued on page 77*)

Sound Diffusion In Rooms

M. RETTINGER*

A review of the methods used to describe the acoustic quality of a room or auditorium by a simple "figure of merit," together with a discussion of the effect of acoustic treatment on the various methods.

THE EXPERIMENTAL EVALUATION of room acoustics has for a long time rested on the determination of the reverberation characteristic of the enclosure. Such obvious acoustic defects as echoes and sound concentrations were easily detected by the simple expedient of clapping one's hands at some distance from the suspected surfaces and listening for any echo effects. Noise level tests, too, could readily be made with any one of a number of sound level meters available on the market.

But quietude, absence of echoes, and an acceptable reverberation characteristic, it was felt, were not the sole criteria for good room acoustics. A condition known as "diffusion" also constituted an important acoustic factor, because it tended to provide more nearly the same acoustic conditions everywhere in an enclosure.

Since the end of the war, much work has been done, mostly in Europe, to arrive at some criterion for this condition of sound homogeneity in a room which would recognize the variations in sound-pressure peaks and valleys that prevail throughout a bounded space when a loudspeaker is energized by a steady sinusoidal signal. Such peaks and dips can for a single frequency be plotted in the form of a topographic map showing, usually in a plane, the distribution of the sound pressure. Similarly, when the signal frequency is varied slowly and continuously, and the microphone position remains fixed, the sound pressure variations appear in the form of an apparently haphazard series of peaks and dips. *Figure* 1 is an example. These, it was felt, had some bearing on the hearing conditions of a room because apparently similar rooms, alike in reverberation time and volume, showed sometimes totally unlike steady-state transmission characteristics, and were also known sometimes to have unlike hearing conditions. This phenomenon was first discussed by E. C. Wente.[1] While he drew no particularly pertinent conclusions from his investigations, short of showing that these variations became less with increasing absorption in the room, he led to the establishment of several techniques for the experimental evaluation of rooms in this manner.

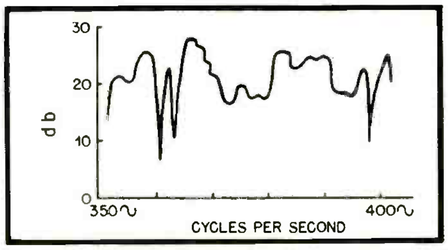

Fig. 1. Typical curve showing sound pressure variations over the audio spectrum between 350 and 400 cps.

A room which has a relatively great steady-state transmission irregularity (defined below) is said to exhibit a "roughness" in sound reproduction, at least at prolonged passages of speech and music where steady-state conditions tend to prevail. The "roughness" is sometimes dependent on the point of observation in the room. These irregularities are dependent not only on the amount of sound absorption in the enclosure—decreasing with increasing absorption—but also on the geometry of the room, decreasing, at least above a certain frequency and up to a certain point, with a more irregular wall contour. Because the amount of absorption in a room of conventionally good design (no parallel reflective surfaces, no concave walls, etc) is fixed by the optimal reverberation period which has been well established for rooms of different volume and fixed purpose (sound-recording, lecturing, etc.) it remained to consider the transmission irregularity in terms of effective room geometry.

Bolt and Roop[2] investigated this problem partly theoretically and partly experimentally, and introduced the term "frequency irregularity" (defined below). Before discussing the results of these investigations, it may be desirable to consider the causes of these fluctuations in sound pressure as a function of position and frequency. In the classic theory of sound, which availed itself of sound "rays" in the investigation of room acoustics—in other words, in terms of geometrical acoustics—such effects are spoken of as interference effects. One sound ray interferes with another, depending on the amplitude and phase relationship of the two. The more modern theory of room acoustics speaks of the sound pressure at a point in the enclosure as the vector sum of the pressures associated with each of the normal modes. This concept becomes of particular importance only when it is desired to evaluate the frequency irregularity theoretically, in terms of statistics, because it lends itself more readily to mathematical computation. In this discussion, however, only experimental investigations will be considered

Evaluation Factors

We are now prepared to consider four factors employed in the evaluation of steady-state sound pressure distribution measurements

1. *Transmission irregularity.* In an arbitrary frequency band, the transmission irregularity is the sum of all the "crest" values, in decibels, minus the sum of all the "valley" values, in decibels. Mathematically it can be stated as

$$T.I. = \Sigma P - \Sigma p$$

when P = pressure maximum in db and p = pressure minimum in db.

This is a convenient experimental quantity, though a wholly arbitrary one.

E. C. Wente, who first investigated and defined transmission irregularity, learned that the variations decreased with increasing absorption in a room. He also showed that they depended on the geometry of the room, at least insofar as the distribution of the absorbent material in the room was concerned. He drew attention to the requirement of a high recorder speed and of varying the signal frequency very slowly (by means of an auxiliary incremental capacitor in the usual beat-frequency oscillator used for the tests).

2. *Frequency irregularity.* This term, introduced by Bolt and Roop, refers to the transmission irregularity per one-cycle band width, and employs the unit db per cycle. Mathematically, the term may be written

$$F.I. = \frac{\Sigma P - \Sigma p}{\Delta f}$$

where P = pressure maximum in db, p = pressure minimum in db, and f = frequency interval in cycles per second.

The rate of frequency sweeping selected for the tests was 1.3 cps and the total irregularity was measured in successive 25-cps bands.

(*Continued on page 61*)

* *RCA Victor Division, 1560 N. Vine Street, Hollywood, California*

[1] E. C. Wente, "The Characteristics of Sound Transmission in Rooms," *J. Acous. Soc. Am.*, V. 7, No. 2, Pt. 1, Oct. 1935, p. 123.

[2] R. H. Bolt and R. W. Roop, "Frequency Response Fluctuations in Rooms," *J. Acous. Soc. Am.*, V. 22, No. 2, March 1950, p. 280.

A Simple High-Quality Phono Amplifier

R. D. MIDDLEBROOK*

An inexpensive audio amplifier with sufficient gain to drive a loudspeaker from a magnetic pickup and which uses an unusually large amount of negative feedback over the output tube and transformer.

THE BEST AUDIO AMPLIFIER is wasted if the associated equipment can not take advantage of it. Many high-quality amplifiers are available commercially, all of them using push-pull output stages. Their capabilities are impressive—and so are their prices. Of the thousands of non-technical people who have bought these amplifiers, how many have spent a *proportionate* amount of money on their loudspeaker systems?

There is room for a cheaper and simpler audio amplifier which can still be considered high-quality. It is not generally believed that a single-ended output stage is capable of producing results in keeping with present-day standards; yet it can be done. This article will describe how excellent listening quality can be obtained from a circuit which, at first glance, doesn't look as though it would sound any better than a small table radio.

The secret of this transformation is negative feedback. It is the writer's belief that most of the unpleasantness of the reproduction from an ordinary single tetrode or pentode output is due to the low speaker damping, and not to the harmonic distortion. Without entering into the fierce discussion on the merits or otherwise of negative output impedance of amplifiers, it can be stated with some confidence that an output resistance somewhat less than the speaker resistance is desirable in order to obtain satisfactory damping. By using enough negative feedback this can be achieved with a tetrode output tube, with all the attendant advantages of reduced distortion and extended frequency response. The amplifier to be described uses a single 6V6 beam tetrode tube, with an ordinary universal output transformer. The performance figures show that the results obtained are definitely in the "high-quality" class:

Maximum output:	3 watts
Total distortion:	1 per cent at 3 watts (1000 cps)
Frequency range:	3 db down at 20 cps and 25,000 cps
Output resistance:	0.1 ohm

The maximum power output of 3 watts is obtained at middle frequencies; the maximum power obtainable at extreme low and high frequencies depends on the quality of the output transformer. No amount of negative feedback can effect any improvement in this respect. Using the Thordarson Type T22S58 universal output transformer, the maximum available power output falls to 3 db below 3 watts at 80 cps and at 8000 cps.

Since most present-day amplifiers profesing to be "high fidelity" are capable of at least 10 watts output, a word in defense of 3 watts may not be out of place. 10 or 20 watts certainly give a considerable margin of safety, so that even the peaks of the signal give rise to distortion less than the figures quoted for maximum output. What the loudspeaker does to the transient is, of course, another story. In this amplifier, owing to the large amount of negative feedback, the signal contains very small percentages of distortion right up to the maximum output of 3 watts, and if the signal is increased further, the peaks are clipped. This effect is clearly visible on an oscilloscope, not only for a sine wave but also for the complex waves of music or speech. As mentioned previously, the guiding principle in the design of the amplifier is economy in construction while maintaining adequate performance for ordinary home listening: the acid test for success is therefore a listening test. If an oscilloscope is connected across the loudspeaker, while reproducing heavily-recorded music, it will be seen that the gain can be increased until the volume is sufficient to drown all conversation before the peaks start to clip. In other words, all activity must cease while the music is at full volume—a condition which must surely warm the heart of any hi-fi enthusiast. Need more be said about the adequacy of 3 watts?

The complete amplifier has sufficient gain to produce full output from a magnetic pickup, such as the General Electric variable reluctance type. *Figure* 1 is a block diagram of the various stages, and it is seen that the circuit can be divided into two distinct parts, each embodying heavy overall negative feedback.

* 1928 Cooley Ave., East Palo Alto, Calif.

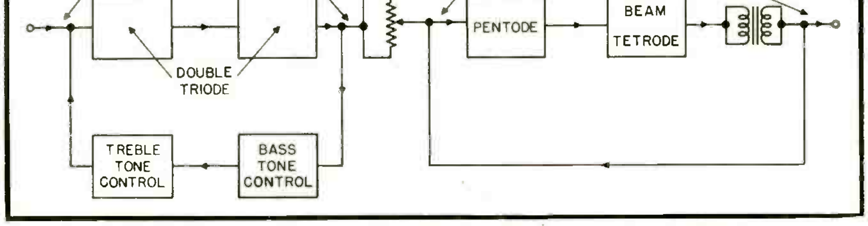

Fig. 1. Block diagram of the amplifier, showing the two negative feedback loops.

Circuit Description

Figure 2 shows the complete circuit in detail. The grid resistor R_1 is chosen to be a suitable load for the pickup. The first two triode sections form a straight cascaded amplifier with a gain of 1400 times, which is reduced to an over-all gain of 100 times by the negative feedback from the plate of V_2 to the cathode of V_1. This link contains the tone control circuits. Various tone control arrangements are possible, and can be chosen to suit the individual: however, two networks which have been found adequate are described here.

Figure 3 is the bass control, which gives a maximum of 12 db bass boost, the magnitude of the rise being controlled by the potentiometer. The point at which the rise begins is selected by the switch. These turnover frequencies are 80, 140, 200, 300, 400, and 500 cps. The bass boost is sufficient to compensate for the recording characteristic of LP records, and in practise it will be found that usually the maximum amount of boost is unnecessary.

Figure 4 shows the treble control, which has one position giving 3 db rise, one flat position, and four positions giving treble cut. The potentiometer determines the degree of cut, and the switch selects the turnover frequency. These turnover frequencies are 1500, 2000, 4000, and 7000 cps. The 0.1 megohm resistor is merely to prevent a thump at the speaker when switching from the flat to the lift position.

Raised in level by 40 db by V_1 and V_2, the signal is fed to the volume control and then to the grid of V_3. This tube has the unusually high plate load, R_{10}, of 2.2 megohms, and is direct-coupled to the grid of V_4. Using a 2.2 megohm load enables a gain of 340 times to be realized; if a.c. coupling were used, the grid resistor of V_4 would have to be at least 10 megohms to avoid shunting the load of V_3, and this exceeds the tube manufacturer's maximum permissible value of 1.0 megohm. Using direct-coupling also has the advantage that no low-frequency phase shift is introduced.

For an 8-ohm loudspeaker load, an output transformer turns ratio of 32 to 1 should be used. The voltage "gain" from the grid of V_4 to the transformer secondary is approximately 0.76, thereby giving a gain from the grid of V_3 to the transformer secondary of $340 \times 0.76 = 260$ times. Since the feedback voltage returned to the cathode of V_3 is one fourth of the output voltage, by virtue of the 470- and 1500-ohm resistors forming a voltage divider, the over-all gain is reduced from 260 times to 4 times: that is we have $20\log(260/4)$, or 36 db of negative feedback.

V_3 obtains its bias voltage from the very small grid current flowing through the 4.7 megohm grid resistor; this is commonly described as "contact potential." Bias amounts to about minus 0.75 volt. The voltage developed across the 470-ohm cathode resistor is only some 0.07 volt, and is therefore negligible. The reason for using this method of biasing is that it avoids the low-frequency phase shift which would be introduced by the conventional cathode resistor and bypass capacitor.

One of the troubles inherent in direct-coupled stages is the drift of operating point. This is avoided here by deriving the screen voltage of V_3 from the cathode of V_4, through the 0.47-meg resistor. The operation is as follows: suppose the plate current of V_3 decreases, then its plate voltage increases, carrying the grid of V_4 with it. The plate current of V_4 then rises, and its cathode voltage increases. This also raises the screen voltage of V_3, tending to increase the plate current of V_3 and restore the operating point. Thus we have in effect an internal negative feedback loop which stabilizes the working points of V_3 and V_4. This feedback loop is made inoperative for signal voltages by decoupling the screen of V_3.

The resistor R_{12} and capacitor C_7 between the plate of V_4 and the cathode of V_3 are to suppress high-frequency oscillations. The capacitor should be a high-quality mica type rated at 1000 volts. If different taps are used on the output transformer, or if a different transformer is used, the values of these two components may need to be adjusted. The series grid resistor of V_3 and the screen resistor of V_4 are included to guard against parasitic oscillations.

The total current drain of the complete amplifier is only about 40 milliamperes, and hence the power pack requirements are modest. Using a 350-0-350 volt high voltage winding on the power transformer, sufficient voltage is available to dispense with a choke in favor of the much less expensive resistor for smoothing purposes. The less efficient smoothing thus obtained is nevertheless quite adequate, because the large amount of negative feedback reduces the 120-cycle hum from the B+ line to complete inaudibility. It is desirable to use one or more of the standard methods for reducing 60-cycle hum from the heater of V_1, such as tapping a ground connection across the heater winding, (as shown) or raising the heater winding to some positive potential with respect to ground.

Only one other point need be mentioned in connection with practical construction. The cathode resistor, both grid resistors, and grid capacitor of V_3 should be mounted as close as possible to the tube socket. This is because the input impedance to V_3 is extremely high and the grid leads are very sensitive to hum pickup.

All figures quoted above are measured values, not calculated. Aural results are excellent: the cleanness of the bass response is particularly pleasing. The entire amplifier can be built on a chassis 8 by 6 inches, and the quality of its reproduction bears comparison with that of amplifiers many times more expensive.

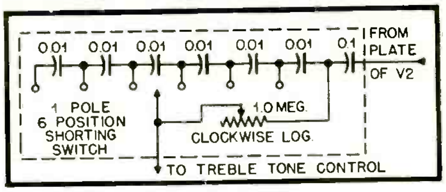

Fig. 3. A suggested bass-boost circuit. This section corresponds to box "A" in Fig. 2.

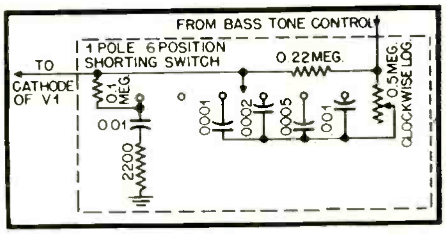

Fig. 4. A suggested treble tone-control circuit. This corresponds to box "B" in Fig. 2.

PARTS LIST—AMPLIFIER

C_1	0.1 µf, 600 v, paper
C_2	50 µf, 6 v, electrolytic
C_3	.01 µf, 600 v, paper
C_4	.001 µf, 500 v, mica
C_5	0.5 µf, 200 v, paper
C_6	500 µf, 50 v, electrolytic
C_7	100 µµf, 1000 v, mica
$C_{8a, b}$	10-10 µf, 450 v, electrolytic
C_9, C_{10}	20 µf, 500 v, electrolytic
R_1	pickup load resistor (see text)
R_2, R_5	0.12 meg, 1 watt
R_3, R_6	2200 ohms, ½ watt
R_4, R_{14}	0.47 meg, ½ watt
R_7	0.5 meg audio taper potentiometer
R_8	4.7 meg, ½ watt
R_9, R_{21}	27,000 ohms, ½ watt
R_{10}	2.2 meg, 1 watt
R_{11}	470 ohms, ½ watt
R_{12}	56,000 ohms, ½ watt
R_{13}	330 ohms, ½ watt
R_{15}	1250 ohms, 5 watts
R_{16}	1500 ohms, 1 watt
R_{17}	47,000 ohms, 1 watt
R_{18}	10,000 ohms, 1 watt
R_{19}	1000 ohms, 5 watts
R_{20}	0.27 megs, ½ watt
T_1	Output transformer, "Universal" type, 32:1 ratio for 8-ohm loudspeaker
T_2	Power transformer. 350-0-350 v at 50 ma; 6.3 v at 2.0 amps

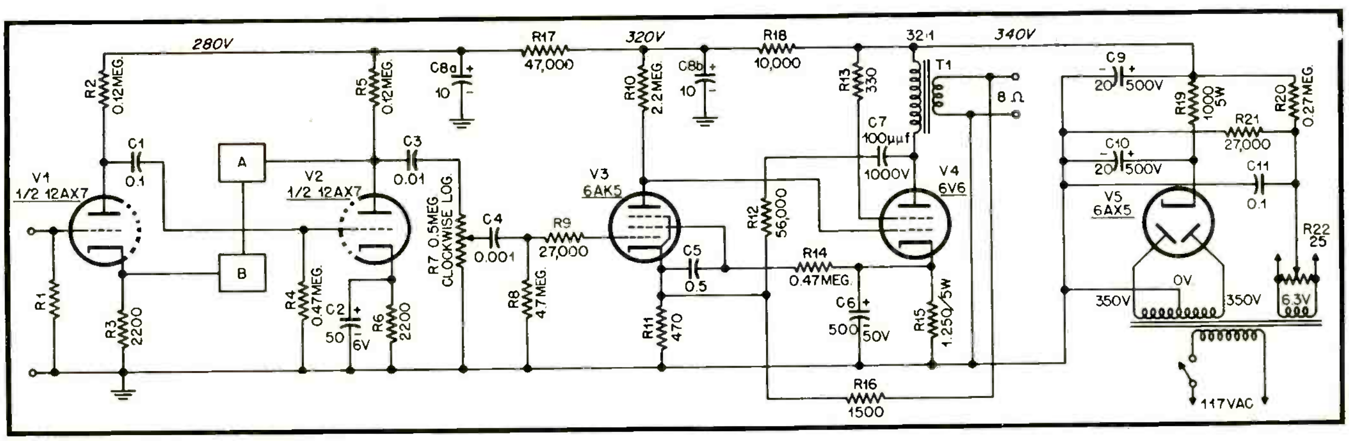

Fig. 2. The complete schematic of the amplifier.

Accurate Design of Phono Equalizers

CHARLES F. HEMPSTEAD* AND HAMILTON BARHYDT*

In Two Parts—Part 1

Feedback equalizers can be designed as easily as forward-path types by anyone not afraid to work with a few simple numbers.

During the past few years most of the articles which have appeared in this and similar magazines describing the design of phonograph preamplifier equalizers have merely given results in the form of a final circuit with a rough statement of its performance. The more detailed treatments have involved only the passive-attenuator-type equalizer. For the feedback-type preamplifier equalizer, which the authors prefer for bass boost because of its greater freedom from distortion, only approximate equations and rules of thumb have been given, with the implication that an accurate treatment is too complicated to be used in practice. However, we have found that the characteristics of this type of equalizer can be expressed by simple equations. The graphical presentation of the results simplifies the design procedure to the point where there is no reason not to carry out the design accurately. These rigorous expressions should also help to settle the controversies which have arisen over the amount of amplifier gain needed to obtain proper bass equalization down to a given frequency.

Equalization of recording characteristics requires an amplifier having a gain-frequency curve as indicated in *Fig.* 1, where the variation in gain is often described by "bass boost" or "treble roll-off," depending on the region of the spectrum involved. This separation into bass and treble regions is made necessary by the fact that recording engineers use two or more "turnover frequencies" about which they vary the recorded sound amplitude in a specified manner[1]. It is the function of the equalizer to perform the inverse of this process and make the electrical output as nearly an exact replica of the original sound input as possible. Fortunately, the separation of the turnover frequencies is great enough that we can treat the bass and treble regions independently and thus simplify the design problem. Between these two regions the curves are smooth and fit the desired characteristics very closely.

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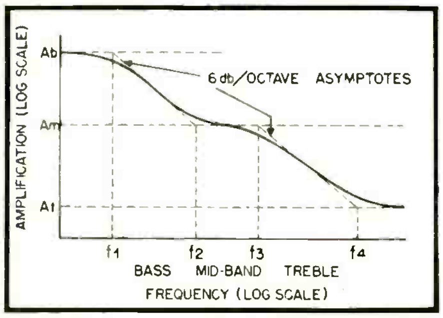

Fig. 1. Typical generalized equalization curve used in record reproduction.

Bass Boost

We shall first treat the case of bass boost in some detail as an example of the design of an equalizer. A typical feedback circuit for obtaining bass boost is shown in *Fig.* 2. The basic amplifier has a voltage amplification without feedback of A_o, considered to be independent of frequency. The over-all amplification with feedback at frequency f is

$$A(f) = \frac{A_o}{1 + A_o\beta(f)}$$

where $\beta(f)$ is the feedback facor. At sufficiently high frequencies $\beta(f)$ becomes

$$\beta_m = \frac{R_1}{R_1 + R_2}$$

and $A(f)$ becomes the midband gain

$$A_m = \frac{A_o}{1 + A_o\beta_m}.$$

At very low frequencies $\beta(f)$ becomes

$$\beta_b = \frac{R_1}{R_1 + R_2 + R_3}$$

and $A(f)$ takes on its maximum value of

$$A_b = \frac{A_o}{1 + A_o\beta_b}.$$

The ratio A_b/A_m is the maximum amount of bass boost available. The amount of bass boost at any frequency can be expressed by the following equation

$$\left|\frac{A(f)}{A_m}\right| = \frac{A_b}{A_m}\sqrt{\frac{1 + (f/f_2)^2}{1 + (A_b/A_m \; f/f_2)^2}}$$

where the turnover frequency at which bass boost starts is

$$f_2 = \frac{R_1 + R_2 + R_3}{2\pi C R_3 (R_1 + R_2)}.$$

The left portion of the curve in *Fig.* 1 shows the general form of response. The lower turnover frequency f_1 at which the bass boost stops can be determined from the equation

$$f_1 = \frac{A_m}{A_b} f_2.$$

The value of f_1 depends on both the resistance R_3 shunting the feedback capacitor C and the limiting effect of the maximum amplification A_o that is available from the amplifier. It is interesting to notice that the shape of the curve depends only on the A_b/A_m; thus the effect of R_3 and A_o on the shape of the curve is the same.

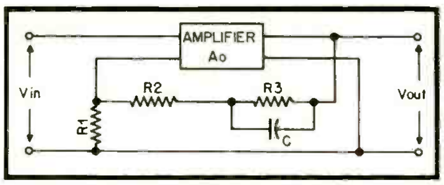

Fig. 2. The equalizer discussed in this article handles the bass boost and has this configuration.

[1] R. C. Moyer, "Evolution of a recording curve," Audio Engineering, July 1953, page 19.

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Graphical Design

The authors prefer to carry out the design of an equalizer graphically since it is much easier to see the sizes of any errors involved and the frequencies at which they occur. For this reason the bass boost curve has been plotted for various values of A_b/A_m in *Fig.* 3. To find the best value of A_b/A_m for a specific case the desired equalization curve is plotted and compared with this family of curves. This can be done conveniently by replotting *Fig.* 3 on a larger piece of paper and plotting the desired equalization curve to the same scale on a piece of tracing paper. The curve on the tracing paper can be directly compared with those replotted from *Fig.* 3 by placing the tracing paper on top. Once one of the response curves has been selected by this method, we can see exactly the places where the response differs from the desired equalization curve and the amount of the difference. Also, the turnover frequency f_2 can be found by locating the frequency on the tracing-paper curve which lies over f_2.

For an illustrative example let us deal with a case involving the maximum amount of bass boost. This amounts to a 6 db per octave boost starting at a 600-cps turnover frequency. In order to minimize the necessary gain and obtain the maximum stability, the smallest value of A_b/A_m which will match the desired equalization curve down to the lowest frequency to be reproduced is chosen. Few people have equipment which will give good reproduction below 30 cps, so we shall choose this as our lower limit. We can fit the desired equalization curve with an error of less than 1 db at 30 cps by choosing $A_b/A_m = 40$.

For a given value of A_o the maximum midband gain A_m is obtained when A_b is made equal to A_o. This occurs when R_3 is made infinite; i. e., R_3 is left out. However there is danger of very-low-frequency instability if there is not a d.c. feedback path such as that provided by R_3 and it is desirable to obtain the benefits of feedback even at the lowest frequencies. Thus we should like to have at least 6 db of feedback in the circuit as a minimum, making the ratio A_o/A_b at least as large as 2. Individual preference on this point may vary, and the relative values of gain can be altered to suit these preferences. Choosing $A_o/A_b = 2$ and $A_b/A_m = 40$ gives $A_o/A_m = 80$. If we have an amplifier with a gain A_o of 2000, which can be obtained with a cascaded 12AX7 driving a cathode follower, then a midband gain A_m of 25 is obtained, giving an output level of 0.25 volt if a GE cartridge is used. This is not enough signal voltage to drive most power amplifiers to full output without additional amplification. If contrary to the authors' preferences A_b is made equal to A_o by making R_3 very large or omitting it altogether, then for the same A_o of 2000 one can obtain an A_m of 50.

After A_o, A_m, and A_b have been chosen, the values of the components in the feedback loop are calculated from the following equations, which can be derived from those given in an earlier paragraph. R_1 is also the bias resistor of the first amplifier stage and is determined by the design of the amplifier.

$$R_2 = \left(\frac{A_o A_m}{A_o - A_m}\right) R_1 - R_1$$

$$R_3 = \left(\frac{A_o A_b}{A_o - A_b}\right) R_1 - (R_1 + R_2)$$

and

$$C = \frac{R_1 + R_2 + R_3}{2\pi f_2 R_3 (R_1 + R_2)}.$$

So far we have neglected the loading effect of the feedback loop on the gain A_o of the amplifier without feedback. If the amplifier has a cathode-follower output stage which has a very low output impedance, then the loading effect will usually be negligible. Moreover the low output impedance and high input impedance of the cathode follower simplify the combination of the preamplifier with other circuits. If it is desired, however, to omit the cathode follower, resulting in an amplifier with a high output impedance, the loading of the feedback loop will reduce the gain without feedback. The effect is difficult to calculate exactly, since the loading depends on frequency. Fortunately, the over-all gain with feedback at midband frequencies depends only slightly on the gain without feedback, since at these frequencies the feedback factor is relatively large. At low frequencies the feedback factor becomes small and A_o does have an important effect on the over-all gain with feedback, but the loading effect is reduced at these frequencies because of the increased reactance of the capacitor in the network. Hence, for practical purposes A_o can be calculated by considering the amplifier as working into a load of

$$R_4 = \frac{(R_g)(R_f)}{(R_g + R_f)}$$

where R_g is the input impedance of the following stage and R_f is the impedance of the feedback loop at some low frequency, say f_1, where $R_f = R_1 + R_2 + 0.7\,R_3$. The authors would like to point out that this is the only crude approximation used in this article and that it can be avoided by the use of a cathode follower.

At this point we should like to make a comparison between the passive R-C attenuator-type equalizer, a type preferred by many designers, and the feedback type of equalizer for obtaining bass boost. Since the general equations for the two types are exactly the same, the amount of over-all gain needed to obtain a given bass boost and midband gain is the same for either type. In one case the extra gain is lost in the R-C attenuator, while in the other case it is profitably used for inverse feedback, reducing the distortion generated in the preamplifier. The results of intermodulation distortion measurements on voltage amplifiers[2] indicate that a preamplifier incorporating an amplifier without feedback and an R-C attenuator equalizer would have about 1 per cent or more intermodulation distortion at 1 volt output. The application of negative feedback reduces the distortion by a factor of $1/1 + A_o\beta$, which is about .01 for a typical preamplifier. The preamplifier without feedback has the advantage that the cathode bias resistor in the first stage, whcih usually has to be left unbypassed in the feedback type in order to apply inverse feedback conveniently, can be bypassed, reducing hum and increasing gain. However, by careful construction techniques the hum can be reduced to a very low value even with a.c. on the heaters[3] and can be reduced even more if desired by using d.c. on the heaters, not too large a price to pay for the reduction of distortion afforded by inverse feedback. Or, if extra gain is needed, a low-noise amplifier can be used ahead of the equalizer amplifier.

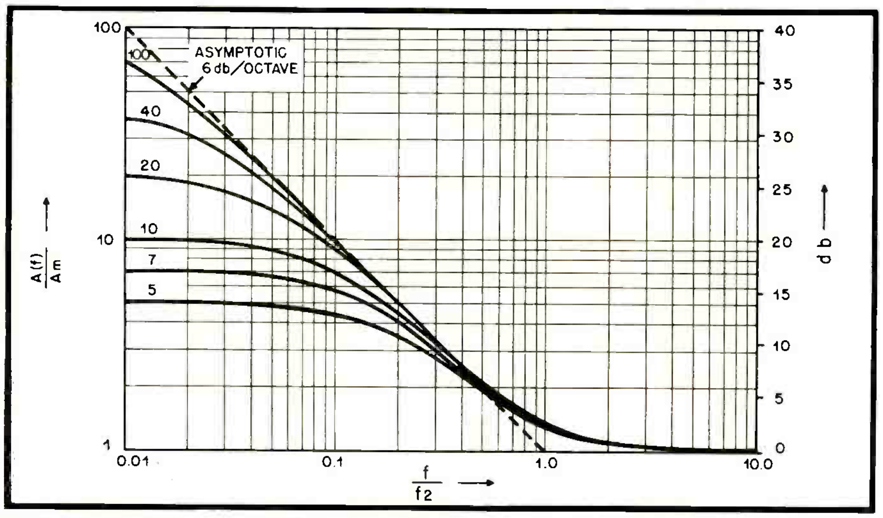

Fig. 3. Set of curves constructed by the authors can be used for designing the bass equalizer graphically.

[2] W. B. Bernard, "Distortion in voltage amplifiers," Audio Engineering, February 1953, page 28.

[3] F. W. Smith, "Heater supplies for amplifier hum reduction," Audio Engineering, August 1948, page 26.

LOUDSPEAKERS
and
ENCLOSURES

A SECTION OF

AUDIO

AUGUST, 1954

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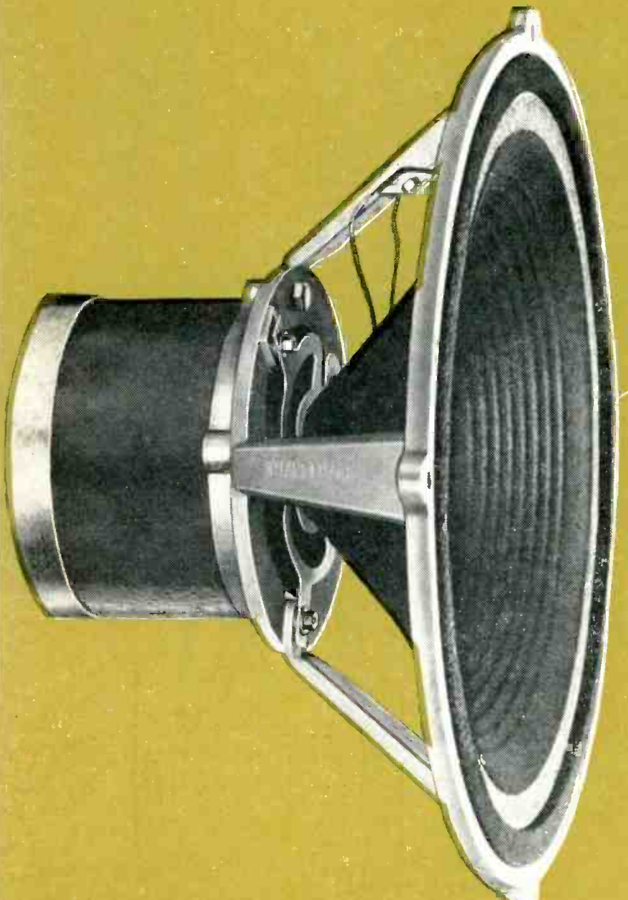

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THERE ARE FEW if any components of home music systems which have such individuality or are chosen so much on the basis of an almost emotional preference as the loudspeaker system. That is hardly surprising. Loudspeakers, practically alone, are still not "perfected" to the point where good is good and bad is bad, and the good is characterless. Amplifiers today can easily be built with inaudible distortion and practically any power output one desires. Pickups have wide frequency and dynamic ranges and the best of them do not distort beyond the point of negligibility. Many tuners are available which, as far as the sound goes, are identical in their flawless performance. But nobody can say anything like the same about loudspeakers.

No; speakers, speaker systems, and speaker enclosures still give the largest rein to the enthusiast who wants to express a preference or a dislike or sit and compare with quizzical expression reflecting critical ears. Speaker systems have individuality—one sounds different from the next even though by whatever standards we can evolve in an effort to be scientific both are equally good. And, too, because speakers are not purely electronic in nature—and enclosures are entirely nonelectronic—even many engineers trained in electronics tend to place them in a category smacking just a little of the Black Arts—somewhere between the good, clean electronic circuit in which a schematic diagram tells all and the mechanical assembly in which lever A moves gear B, which causes steam cock C to open and inflate balloon D—all of which can be seen with the naked eye. As a result they usually avoid the subject entirely and proceed by guess—not a very rigorous technique.

And when it comes finally to enclosures—ah, what cults grow up around these mediums' boxes with their ports and vents sized just so and their internal labyrinths of mysteriously shaped pieces of wood glued and screwed in crazy-quilt patterns, each with its unfathomable purpose and its exotic title!

In this section of AUDIO this month we intend to try removing some of the mystery from loudspeaker systems. We are aware that in this emotion-charged subdivision of the audio field we will be praised and damned for opinions we do not express, largely by the fringe of very special enthusiasts who listen only to drums and triangles—never music—to test speaker systems, and who never will find a system they can like for more than a week at a time. But we feel strongly that the solid substance of audio people—those who like music, want it to sound as good as possible, and are willing to learn a bit about how to do it—might like to absorb whatever correct and objective information we can furnish on a subject not too often exposed to public view.

To that end, AUDIO has created this special section. We will begin with the beginning—what a speaker is and how it works. From there we will go into enclosures, telling the factual story of why each type is used and how it functions. And to finish off, we will engage in a review of a great many of the speakers, systems, and components now on the market, telling you to the best of our ability in text English (rather than Advertising English) what each one of these representative selections from manufacturers' lines is all about.

It is our aim to make this section the Compleat Loudspeaker Guide for you, the aforesaid solid audio people, for which purpose we shall engage in no Engineerese; this section is not for engineers (who are normally among our favorite people), but for audio enthusiasts who want some easily absorbed background knowledge to help them in their quest for fine sound and the most enjoyable listening.

We have been helped a great deal by the speaker and enclosure manufacturers who have submitted data to us. Some of these companies show their wares in advertisements on these pages; many of the ads are interesting and helpful and we invite your attention to them. But please don't expect to find out what our own preference is—if indeed we have one. The facts (so far as they can be determined) are what we're after.

And so concludes the Prologue. Let the play begin!

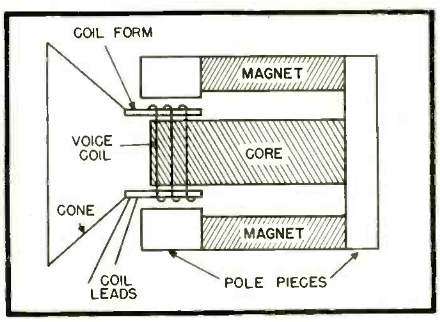

Fig. 1. Basic loudspeaker structure. The voice coil is wound on a form which is attached to the voice coil apex and slipped over the core of the field magnet.

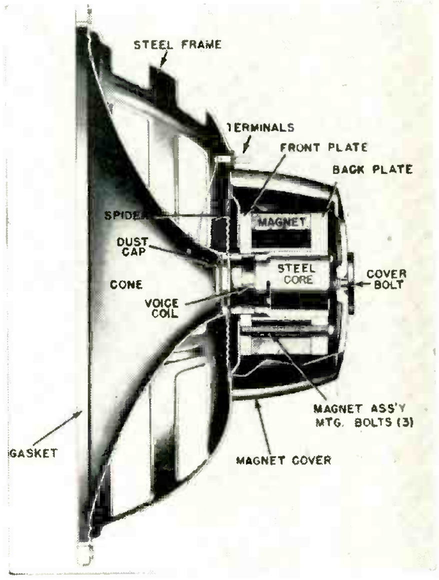

Fig. 2. Cross-section of GE S1201D speaker shows actual makeup of a typical speaker assembly. Other speakers vary in details but all function similarly.

Loudspeaker Construction

A loudspeaker is a d.c. motor with a field and an armature. The motion of the armature, however, is reciprocating—back and forth—rather than rotational. The armature is called a voice coil, and the field in all modern high-quality speakers is furnished by a permanent magnet. *Figure* 1 illustrates the basic principle of operation. The field magnet is cylindrical in shape and a cylindrical coil form moves back and forth within it. The voice coil is wound toward one end of the coil form and is connected to the output of the amplifier. To the left end of the coil form is connected a cone of paper or some material with a paper or paper-like base.

The voice-coil form fits loosely over the steel core, which is one pole piece of the magnet. The other pole piece concentrates the magnetic field through the voice coil. The a.c. (audio) passed through the coil causes magnetic lines of force of changing direction every half cycle. These lines of force interact with the field due to the magnet, as in an electric motor, and push the coil alternately up and down over the core. Each time the coil moves it pushes or pulls the cone, which in turn compresses or rarifies the air in front of the cone. When the compression and rarification takes place at an audio rate—faster than perhaps 20 cps—the result is a sound wave in air—and that's how the loudspeaker is born.

How these parts look in an actual speaker is illustrated by the cross-section view of *Fig. 2* and the exploded view of *Fig.* 3 of the General Electric S1201D, a low-priced speaker of excellent quality intended for home music systems. Other speakers differ in the details but this speaker is a good illustration of construction in general.

The frame is made of steel and is the foundation of the speaker. The front plate (note that it is disc-shaped, with a hole in the center) is welded to the rear of the frame. The magnet butts against it, and the back plate, screwed to the front plate with long screws, holds the magnet in place.

A steel core is held in place through the center of the magnet and this core continues the magnetic circuit, creating an intense magnetic field in the annular opening between the front pole piece and the core. In this annular opening is slipped a coil form on which the voice coil is wound. Permanently attached to the voice coil form is the spider, a pleated circle of flexible material, the outer edges of which are cemented (or clamped, in some speakers) to the frame. The small end of the cone in turn is cemented to the outer edge of the voice coil form. When the coil moves, it tends to push and pull the cone, but it is held in place axially by the spider, which is flexible so that the coil can move back and forth in the annular opening or slot without interference. The outer (large) edge of the cone is cemented to the frame, and there is a pleat in it so that it, too, remains centered axially but is fairly free to move back and forth.

The small dust cap simply covers the outer end of the voice coil form so that dust cannot enter and prevent the form from moving smoothly in the slot. The magnet cover is there for the obvious purpose of protecting the magnet assembly and improving appearance. The gasket is a circle of cardboard or fiber which is somewhat compressible so that when the front of the frame is screwed to a baffle the joint will be air tight.

Practically all speakers are made in about this way, including the cheap kind used where fidelity does not matter. What, then, are some of the quality factors that distinguish a high-quality speaker from a poor one?

One is the field magnet. Alnico V, an aluminum-nickel-cobalt-iron alloy, is almost universally used today in speakers. But, up to a point, the larger the magnet the better the speaker, all other things being equal. As in other parts of the audio system, output must be proportional to input to avoid distortion. In the case of the speaker this means that the motion of the voice coil must be proportional to the applied audio power over the entire span of the motion. Since motion depends on the interaction of the permanent-magnet flux and the electromagnetic voice-coil flux, certainly the permanent-magnet flux must remain constant over whatever distance the voice coil is allowed to move. A small magnet does not create enough flux to do this; a large magnet creates uniform flux density over a larger area, so that more power can be applied and the voice coil can move farther before distortion.

Another quality factor is the voice coil. The impedance of almost all voice coils is relatively low—perhaps 3 to 16 ohms—because to obtain a higher impedance more turns would be necessary. This would make the coil heavier and the additional weight would add inertia restricting the ability of the assembly to respond exactly to the applied audio.

Impedance of a coil is made up of at least two factors: d.c. resistance (what you would measure with an ohmeter) and inductive reactance. Every coil has both, including a voice coil. The resistive part of the voice coil impedance, made up simply of the resistance of the length of wire used, does nothing useful; it simply causes heat when current is passed through the coil. The higher the resistance, for a given current, the more of the applied power is radiated in useless heat, for it is only the inductive part of the impedance that is useful. Inductive reactance itself consumes no power, so the larger that part of the impedance which is purely inductive reactance, the more of the applied power is translated into mechanical motion.

The worst possible practice would be to make the voice coil out of resistance wire. Even using ordinary copper wire, however, there is a resistance of anything from 1 to 12 ohms, as you know if you have ever applied an ohmmeter to a voice coil. For an impedance of 16 ohms, for example, a 12-ohm resistance is a good part of the total and it is easy to see why the average speaker is rather inefficient. Some of the highest-quality speakers, therefore, employ aluminum ribbon, wound on edge, for the voice coil. Because the aluminum has a large area its resistance is low; and because it is light, it does not add undesirable inertia.

Loudspeaker damping is more important even than efficiency, and damping also requires low d.c. resistance. Let us see how damping works. Referring again to *Fig.* 1, suppose a sudden sharp drumbeat comes along and causes the voice coil (and cone) to move outward. We might assume that when the drumbeat is finished the voice coil would keep right on coasting and slowly come to a stop, there being little to keep it from doing so. But let us suppose that the impedance of the amplifier and transformer feeding the voice coil is very low, so that at all times the voice coil is practically short-circuited. Now if the drumbeat is over and the coil continues moving, the voice coil—a conductor—is moving through a magnetic field. As we all learned in high school, when a conductor cuts magnetic lines of force a voltage is induced in the conductor. If the coil is almost short circuited, the current circulating through the coil as a result of this induced voltage is very high. And high current passing through a coil sets up a magnetic field about the coil. The polarity of the field set up is such as to interact with the original field of the magnet and prevent further movement of the coil! All this happens when the drumbeat is over—or, indeed, even if the signal is turned off and you make the cone move by hand. In other words, interaction of coil and magnet tend to cause resistance to any coil movement not caused by audio power input to the leads.

But the amount of this opposition to free movement depends on how much current can flow through the coil when voltage is induced in it—and that, in turn, depends on how much electrical resistance exists in the coil winding and in the audio source attached to the leads. Amplifiers today have low effective output impedances, obtained mainly by high negative feedback. But if the coil itself has large resistance things are still bad. So a low-resistance voice coil allows cleaner sound, simply because it allows maximum damping effect: the voice coil movement is almost nil except in response to the input signals which are supposed to move it. In a word, the system cannot *boingggggg!*

We have mentioned that a large magnet is necessary to allow large movement of the voice coil. Let us see why large movement is necessary and when. When we apply d.c. to the voice coil, the only limitations on the distance of travel are the mechanical stops that exist

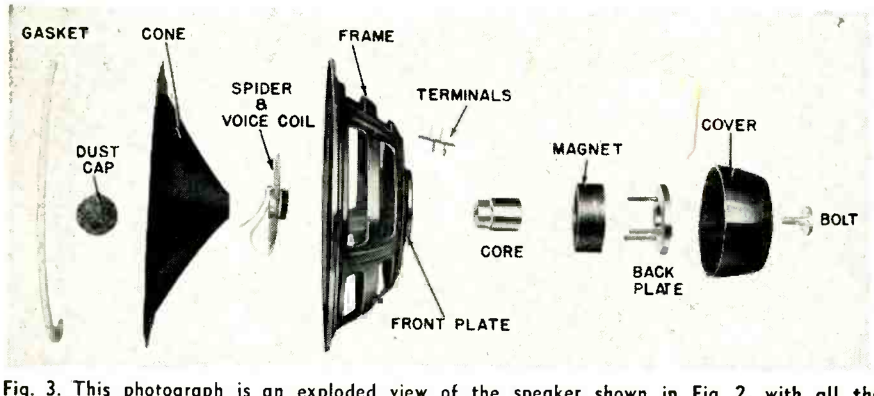

Fig. 3. This photograph is an exploded view of the speaker shown in Fig. 2, with all the parts shown in position to fit together when a giant hand (equipped with the proper tools) accordions them together. This figure makes clear the exact shapes of some parts—the field magnet, for example, which is obviously cylindrical. The spider-voice coil assembly picture is especially useful, since it shows how the coil form is supported fairly rigidly, yet can move axially.

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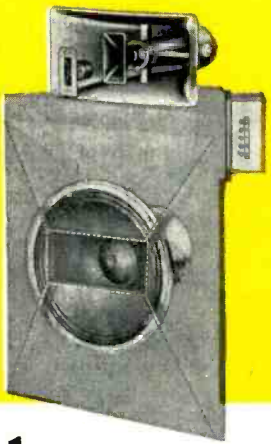

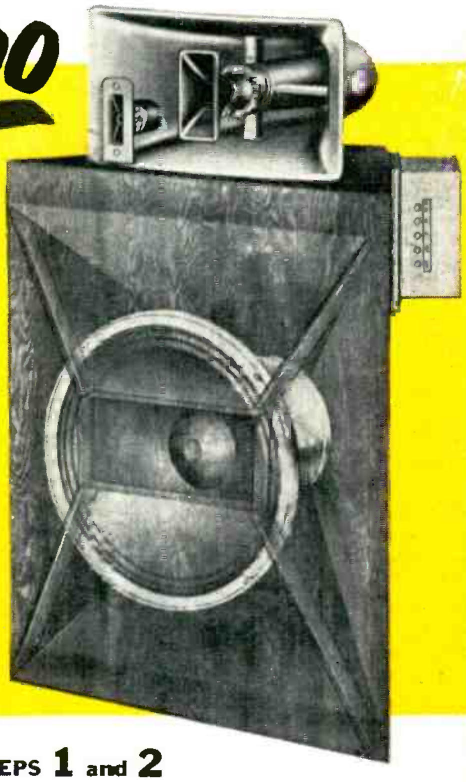

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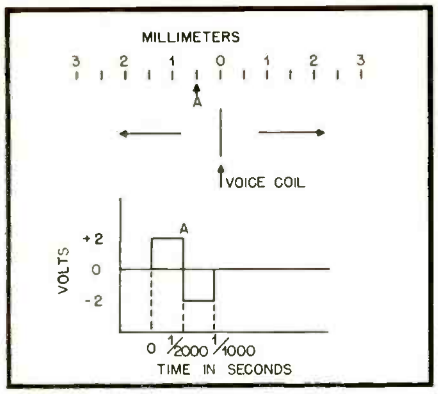


Fig. 4. A constant a.c. voltage applied to the voice coil means constant velocity of movement and direction of movement depends on the instantaneous polarity or phase of the applied a.c.

in the structure. However, there is one more way of limiting coil travel—by applying the d.c. for only the length of time required for the coil to travel the desired distance. This is exactly what happens when we apply a.c. (audio) to the coil.

Begin with the axiom that the speed or velocity of a motor like this is exactly proportional to the applied voltage (as long as the magnetic flux of the field remains constant). Suppose, for example, that for 2 volts applied to the coil, the coil moves at the rate of 1 meter per second. (These are not actual figures, merely illustrations with round numbers.) If a 2-volt square-wave signal at 1,000 cps is being applied the coil can move in one direction only during one half of each cycle, which takes 1/2000 second, allowing a total movement of only 0.5 mm. to A, as in *Fig.* 4. At the end of that time the phase of the voltage changes so that it must begin moving in the other direction, again for a time of 1/2000 second and a distance of 0.5 mm. If we applied 4 volts, the total movement would be 1 mm since the coil would move faster in the same length of time.

But suppose we hold the voltage at 2, so that speed remains at 1 meter per second, and we reduce the frequency to 500 cps. Now a half-cycle takes twice as

Fig. 7. These small cone speakers made by Bozak are used as high-frequency units. Their cones are quite stiff and small so that resonant frequency is high and moving mass is very small. Several may be used in a single speaker system.

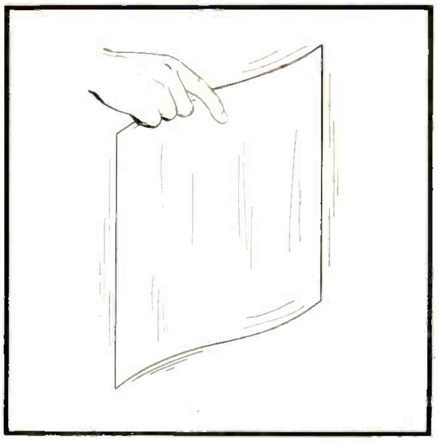


Fig. 5. Holding a piece of paper by one corner like this and moving it shows what happens to a limp (high-compliance) speaker cone. It moves well when the motions are long and slow (low frequencies) but tends to stand in one place and just ripple when the hand oscillates fast.

long—1/1000 second—so the coil has time to move twice as far in each direction—1 mm. And if we reduce frequency to 100 cps, it can move 10 times as far in the time of 1/200 second which is allowed, or 5 mm.

What happens, then, is simply that the distance of voice coil (and cone) travel is inversely proportional to frequency. This is logical if we expect the same power output at all frequencies. Power output does work, moving air. At high frequencies air is not pushed very far, but it is pushed often; at low frequencies it is pushed far but not very often. The work in both cases is the same, but the bass does pose the extra requirement of relatively large coil and cone movement and a large magnet for a big, uniform flux.

Perhaps the greatest difference between speakers of comparable over-all quality is in the cone—its material, shape, and mounting. To reproduce all frequencies a cone should be both stiff and flexible; it should be mounted rigidly and loosely. Obviously these requirements conflict, and the best compromise has been the subject of much search and many schools of thought.

The differing requirements of bass and treble begin with the cone mounting. If the cone is to travel far, as it must at low frequencies, it should be somewhat limp and very flexibly attached to the frame of the speaker. If it were rigid and rigidly attached it simply couldn't move very far. At the high frequencies, however, it should be stiff so that it can take the rapid reversals of direction without breaking up into little unrelated waves of paper movement.

You can demonstrate this with a piece of ordinary stationery. Take the paper by one corner as in *Fig.* 5 and wave it through the air in only one direction. The paper will follow if the movement is large. Now move the paper just as fast but in very short strokes in both directions. Because your hold on the paper is very flexible, everything but the corner you hold in your hand will tend to stay in one place—it is simply too limp to follow such rapid motions

Fig. 6. A piece of cardboard held in both hands simulates a stiff cone. It remains fairly rigid when the hands move fast, acting as a piston to displace air. While good for high frequencies, its low compliance as a cone raises resonance.

against the resistance of the air. Actually, the surface of the paper will break up into a series of smaller wave motions. The sum of all these is very small and very little air is actually moved—and this is exactly what happens in a speaker with a limp cone at high frequencies; it is known as cone break-up.

You can, however, move air successfully at high frequencies if you will substitute for the paper a piece of fairly stiff cardboard, and especially if you will hold it by two opposite edges as in *Fig.* 6. Then the cardboard resembles a stiff piston. It moves as a body and pushes air whenever it is moved, even when it is moved very fast. High-frequency cone speakers have stiff cones, and some of the best high-frequency "tweeters" are those with metallic or stiff plastic diaphragms.

Figure 7 shows a pair of cone-type tweeters made by Bozak. The cones are small and stiff, which is ideal for reproduction of treble. A diaphragm-type tweeter made by Stephens appears in *Fig.* 8. The driver (field and voice coil and mechanical assembly) is in the cylindrical housing. The voice coil drives a fairly flat diaphragm rather than a flaring cone and the multicellular sound

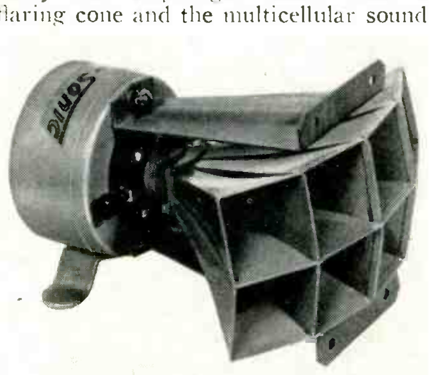


Fig. 8. Appearance of a typical horn-type tweeter made by Stephens. The cylindrical driver contains voice coil, magnet, and diaphragm. The multicellular horn disperses the sound.

Fig. 9. One method of attacking the dual problem of high- and low-frequency reproduction with a single speaker is illustrated by this Permoflux speaker. The cone is relatively rigid for good treble but slitted at the frame for bass compliance.

dispersing assembly attached to the front of the driver couples the diaphragm movement to the surrounding air by using eight horns (of which more later), all pointed in somewhat different directions for maximum dispersal.

Figure 9 shows a Permoflux cone speaker which is designed to reproduce the entire band. Here one form of the compromise in cone design is shown by the slits along the outer edge of the cone where it is joined to the frame. The slits make for a rather flexible joint so that the cone is fairly free to move at low frequencies, but the cone itself can be fairly stiff so as to reproduce the highs without breaking up. The designs used in cones for this purpose are too numerous to mention—combinations of cone material stiffnesses, flare shapes, mountings, and sometimes metallic diaphragms at the apexes of the cones.

One more factor in the performance of speaker moving assemblies should be understood before we go further—resonance and its effects. Most people know that every speaker has a resonant frequency—usually somewhere between 30 and 250 cps for fair-sized cone speakers—and that sound output drops off sharply below the resonant frequency. In an article like this we would be short-changing you if we let matters stand at that, especially since it is really so easy to explain why all that is true.

Analysis of Speaker Operation

The moving assembly of a speaker—cone and voice coil—is a rather simple mechanical system. It has three dynamic quantities—those that affect its performance while moving. These are (1) the stiffness of the assembly (the cone itself and its mounting to the frame and voice coil), (2) the mass of the assembly (the effective weight of the moving parts when moving), and (3) the resistance to movement caused by the air it pushes and any friction or other difficulty caused by mounting methods.

We can break down the speaker as in *Fig.* 10. The voice coil is the primary moving force. The link is the mechanical connection between the voice coil and various parts of the cone. The link naturally includes parts of the cone itself. The air is the load on the system which causes resistance, and the cone and link and voice coil combined have weight and therefore are the mass of the system.

The modern science of electronics is very useful in analyzing what happens in a mechanical system like this, because ordinary electrical quantities can be made to analogize—substitute for—the mechanical quantities involved. For example, we know from high-school physics that capacitance is an electrical quantity which opposes a change in voltage or *force*. Now, when the voice coil of *Fig.* 10 is moving back and forth, continuous changes in force are transmitted by the link to the cone. Suppose the link is very stiff—visualize a rod of steel. Then changes in force are very easily and precisely transmitted along it. It has, in other words, very little opposition to changes in force, and we can therefore liken it to a very small capacitor.

If, on the other hand, it is very limp—suppose it is a spring—then it will not be very good at transmitting changes of force to the load (cone and air). It has, we can say, a high opposition to force changes, and is the equivalent of a large capacitor.

In *Fig.* 11 we have drawn the electronic analogy of our mechanical "circuit" of *Fig.* 10. The voice coil, which generates mechanical energy in *Fig.* 10, is represented as a generator of a.c. in *Fig* 11. The compliance (opposite of stiffness) of the link is represented by a capacitor.

Inductance in electricity is a quantity which opposes a change in electromagnetic energy or in current. In the speaker system energy is required to make the system move. If it is heavy (large mass) it will, of course, be harder to move or once in motion it will be harder to stop. It has, in a word, a high inertia. Since inductance opposes a change of energy, we can represent the mass-caused inertia of the mechanical system of *Fig.* 10 by the inductance of *Fig.* 11, which will be larger as the speaker mass increases and smaller if the speaker mass is small.

When air is moved by the mechanical system, energy is expended in mechani-

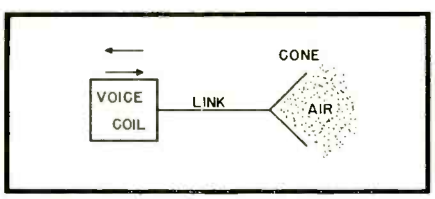


Fig. 10. This diagram shows the quantities translatable to electrical symbols. The moving voice coil (mechanical generator) moves the compliant link (capacitance) to drive the cone mass (inductance) which does power-absorbing work to drive air.

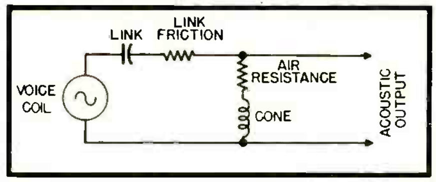


Fig. 11. These are the electrical symbols which analogize the mechanical system shown in Fig. 10. The series capacitance represents the compliance of the link and the series resistance represents power absorbed by incidental frictions. The inductor is the mass of the cone and assembly. The shunt resistor is the power-absorbing air load.

cal work. To expend energy in the electrical circuit of *Fig.* 11, we must have the current passing through a resistance. The larger the mechanical work done, the larger the equivalent resistance, which we appropriately place in series with the inductor representing the moving mass. An additional (very small) resistance is in series with the capacitor representing the link compliance to show that there is probably some friction preventing the link from moving perfectly freely. It is just as true in a mechanical as in an electrical system that there is never a capacitor with unity power factor or an inductor with zero resistance.

The circuit of *Fig.* 11 should look familiar. It is a high-pass filter of the resonant type, the resonant circuit having rather low efficiency because of the large resistance in series with the inductor representing the work done in pushing air. At resonance, the output will be at maximum because of resonant rise of voltage. Above resonance plenty of output exists because inductive reactance gets higher with frequency. But below resonance, output drops off sharply because inductive reactance becomes lower rapidly (tending to short the output) and the series capacitive reactance gets steadily larger.

The resistance representing work done in moving air is not constant over the frequency range, nor is it the same when the speaker is placed in a baffle or enclosure. A speaker by itself is not coupled to the air very efficiently at low frequencies, as we shall explain later. The resistance is therefore lower at the bass. When it is coupled better, with use of an enclosure, the resistance rises because more work is being done. This lowers resonant frequency.

Lest an engineer or two has been sneaking a look at this article, let us state for the record that this explanation of resonance is not rigid. It is simply the easy way to understand it, and it is close enough to show the following points:

(1) Why output drops off below resonance;

(2) That resonant frequency gets higher as the system is stiffer;

(3) That resonant frequency becomes lower as the mass increases;

(4) That resonant frequency gets lower when coupling to the air is improved at low frequencies.

Two- and Three-Way Systems

The paradoxes we spoke of earlier

which require different mechanical structures for optimum reproduction of high and low frequencies are not so depressing as they sound. When you can't lick 'em, join 'em. And that is exactly what sound enthusiasts do when they accumulate sufficient funds in the kitty to come right out with the decision that a single speaker is not enough. And it is quite true that, while a number of excellent single-unit speakers exist today, it is almost impossible to get the utmost in clean, wide-range, distortion-free reproduction in a single speaker. So, having looked that fact in the face, we simply walk around the problem by using separate speakers for different parts of the range, employing each one in the range where it is most satisfactory.

The low-frequency speaker or "woofer" is always a cone speaker. As you can imagine from what we have said earlier, it has a rather flexible cone and mountings and its mass may be relatively high, so that it has a low resonant frequency for extended bass response. It has as large a cone as the buyer can afford because the larger the cone the more air it will contact and move. Keep in mind that to obtain equivalent power output at bass where air is not moved very often, it must either be moved faster and farther (bass boost in the amplifier) or more of it must be moved (bigger cone). The Electro-Voice type 15WK shown in *Fig.* 12 is a typical high-quality woofer, with its 15-inch diameter, 5¼-lb magnet, and 32-cps resonance.

The high-frequency speaker ("tweeter") in a high-quality system is usually a combination of metallic-diaphragm driver and multicellular-horn dispersion unit such as the Stephens in *Fig.* 8. *Figure* 13 shows the University 4408, which is a smaller and less expensive tweeter of the same type, often purchased later and added to existing cone-speaker installations. Lately, high-frequency cone speakers like those in *Fig.* 4 are being used. Combinations like this of two speakers in a single cabinet are called two-way systems. Three-way systems, with a woofer

Fig. 13. This is the University type 4408 treble speaker (tweeter), a moderately priced unit. It is constructed on the same principle as that of the Stephens in Fig. 8 but has a smaller driver and a horn with only two sections.

for the bass, a tweeter for the treble, and a mid-range speaker, are also in use, but only when the ultimate in quality is demanded.

With a two- or three-way system it is not enough merely to connect the speakers in parallel or series and let it go at that. You remember the patient who asked the doctor how, when he swallowed the headache pill, the pill would know enough to go to his head and not to his stomach. In our case, the treble does *not* know enough to go only to the tweeter and the bass only to the woofer—we must use a *crossover* network to channel the right ranges to the right speakers.

Figure 14 shows the simplest possible arrangement for this purpose. The woofer is placed directly across the amplifier output, receiving all frequencies. A capacitor is placed in series with one lead to the tweeter; it is made small enough to prevent low frequencies from reaching the tweeter. The only purpose of doing this is that low-frequencies would damage the tweeter, which is not made to deal with them. A variable resistor may also be placed in series with the tweeter so that balance between bass and treble can be adjusted.

This is far from an ideal arrangement. One of the reasons for using a high-frequency speaker is to relieve the woofer from having to deal with the treble which causes cone break-up and distortion. The

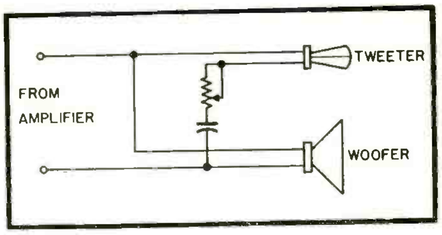


Fig. 14. The simplest way to connect woofer and tweeter is to connect the woofer directly to the amplifier output and place a capacitor (and level control) in series with the tweeter.

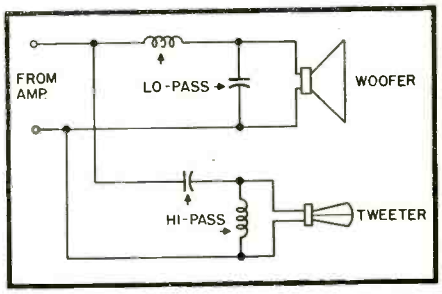


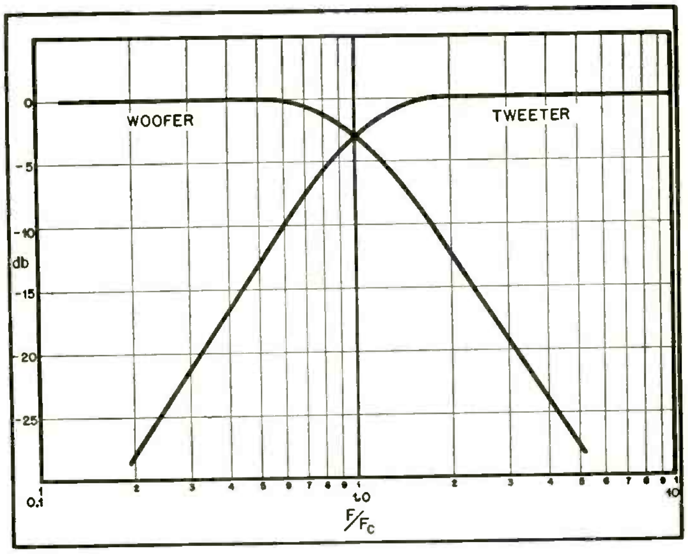
Fig. 15. A full-fledged crossover network is the way to take full advantage of a two-way system. The filters are present to see that only a single band of frequencies is channeled to each speaker so neither does the work of the other.

simple capacitor network does nothing about this. In addition, we now have two speakers sending out treble. Because of the short wavelengths involved, the two speakers may be far enough apart so that there may be acoustic phase interference —the crest of a high-frequency wave may reach your ears from the woofer at the same time as a trough of the same wave may reach you from the tweeter. This would, of course, cause cancellation and you would hear nothing. In practice, of course, phase would rarely be 180 deg. out, but even small differences would destroy uniformity of reproduction.

The ideal result is to have each speaker reproduce only its own range, with the acoustic output of the two equal. This can be done with a network like that of *Fig.* 15. The woofer is fed through a resonant-type low-pass filter, response of which drops off sharply (about 12 db per octave) above the cutoff frequency. The tweeter is fed through a similar high-

Fig. 12. This photograph shows a typical high-quality woofer—low-frequency speaker. This one is the Electro-Voice 15WK, a 15-inch cone unit with resonance at low 32 cps and a 5¼-lb. field magnet to give uniform flux over a large area.

Fig. 16 These curves show how the audio band is divided between the two speakers in a system with a crossover network like that of Fig. 15.



pass filter, whose response drops off sharply below its cutoff. The results are shown by *Fig.* 16 for a system with a 1,000-cps crossover frequency. Sometimes a potentiometer is connected across the output of the high-frequency filter of *Fig.* 15 so that treble balance can be adjusted, but this is neither necessary nor beneficial when the complete system comes from one manufacturer since efficiencies are usually balanced in the manufacture.

The choice of crossover frequency is not haphazard. Ideally, it would be a low one. perhaps 600 cps or so, so that the cone woofer would not have to handle anything that would interfere with the possibility of designing its cone for the single purpose of reproducing the lowest frequencies For home music systems this is not usually possible. The tweeter would have to be rather large so that its own resonant frequency could be below that 600-cps point, a problem largely of the diaphragm, the nature of which gives it a high-frequency resonance. Also the horns for dispersing the sound would have to be large, since horn size is inversely related to frequency. *Figure* 17 shows a Stephens high-frequency horn designed for a 400-cps crossover; it is 31 inches deep, 37 inches wide, and 16¼ inches high, and its shipping weight is 160 lbs. This is all without the driver, which is a cylindrical unit weighing another 22 lbs. and adding 4⅝ inches in depth. Such a tweeter is more suitable for theatres than for the home.

Crossover frequencies from about 800 to 4,000 cps are common for home systems. The upper figure is really somewhat too high for best results, but excellent results can be had with frequencies somewhat higher than 800 cps. In general, this is a quality and space factor—you get what you want to pay for and what the lady of the house will let you keep in the living room.

Coaxial Speakers

There is a third type of loudspeaker which is a two-way system in itself—the coaxial speaker. One type of coaxial speaker, manufactured just after the war by Altec, bears a good measure of responsibility for the "high fidelity" boom, along with the GE variable-reluctance pickup. Both these units were the first of their type to offer relatively high performance at consumer prices.

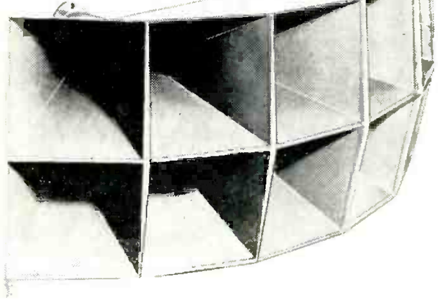


Fig. 17. A low crossover frequency is desirable since it channels less of the high frequencies that cause cone breakup to the woofer. A frequency of 400 cps, however, requires a mammoth tweeter assembly like this Stephens, for example, over a yard wide and almost as deep.

Fig. 18. The coaxial speaker is a two-way system in minimum space and often at minimum expense. This Altec 604C is the latest version of one of the first and best coaxials. It requires a crossover network just like separate dual systems.

The latest model of the original Altec coaxial speaker appears in *Fig.* 18, the 604C. This is a standard 15-inch cone speaker used as the woofer, with a tweeter and multicellular horn mounted coaxially. You will remember from our earlier description of a basic speaker that there is an area within the center of the voice coil which is used only for a core. Within this area—inside the woofer voice coil, which is made larger than usual for the purpose, there is space for the mouth of the high-frequency horn, and the voice-coil and magnet assembly of the high-frequency speaker is located behind that of the low-frequency unit.

Another type of coaxial speaker consists of a small high-frequency cone speaker mounted in the center of the low-frequency cone—sometimes flush with the front of the frame, and sometimes as a continuation of the conical section of the low-frequency cone.

Figure 19 is a cutaway photo of the Jensen G-610 triaxial speaker, which is a combination of three speakers in a single frame—a complete three-way system. To see how this is done refer to *Fig* 20, a cross-section drawing of the speaker of *Fig.* 19. The low-frequency speaker consists of a large voice coil and a cone. The mid-frequency speaker consists of a voice-coil-and-field assembly behind the big one and a short horn extending through the big voice coil. This short separate horn ends in a diffuser which is attached to a point near the apex of the low-frequency cone. The sound output from this mid-frequency diffuser goes out through an area bounded by the big cone, which acts as a flared horn to improve air coupling. The high-frequency unit is entirely separate, with a diaphragm, driver, and housing supported at the mouth of the big cone by a crossbar attached to the main frame.

Two- and three-way coaxial speakers are the equivalent of multi-unit systems employing completely separate units. They require crossover networks in the same way and for the same reasons. In most designs, there is probably some loss in quality over completely separate systems simply because some compromises are necessary to get all that material in a small space. The principle advantage is simply one of smaller required mounting space, and much simpler mounting in the enclosure. Another claimed advantage is that both bass and treble emanate from identical points so that the weird effect that comes about when the two are separated appreciably is avoided. However, the only real requirement here

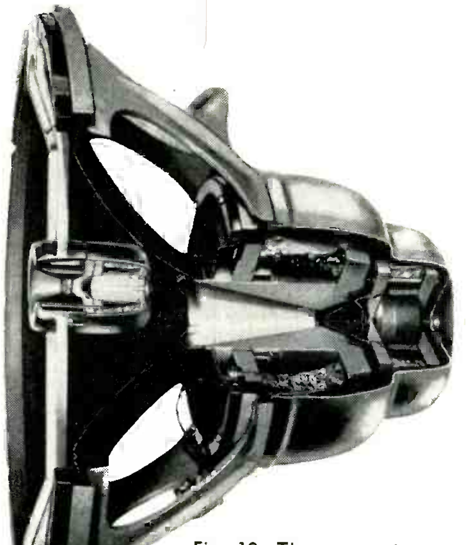


Fig. 19. The approach to construction of coaxial speakers is illustrated by this cutaway view of the Jensen G-610 "triaxial" which includes the same types of units as a coaxial plus a third mounted on supports out near the frame front.

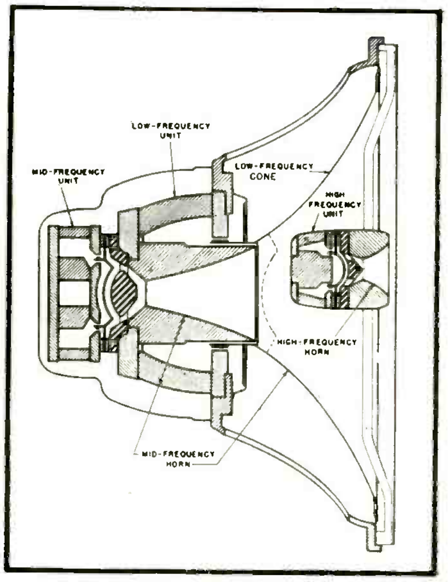


Fig. 20. The cutaway drawing of the speaker of Fig. 19 makes the construction clearer. The midfrequency horn goes right through the center of the woofer voice-coil and magnet assembly and is driven by an entirely separate coil-magnet assembly mounted behind the large one.

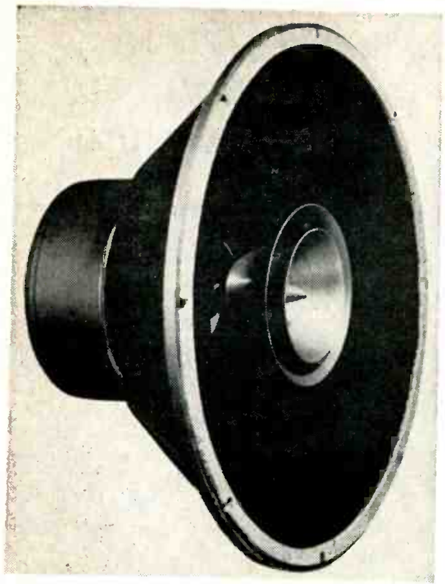


Fig. 21. One of the prime problems with high frequencies is not only reproducing them in the first place but destroying their tendency to be very directional. This Stephens "cospiral" is an attempt to disperse highs without employing a separate tweeter mechanism. It is not a "coaxial."

is that the distance between either speaker and the listener be several times that between speakers, and just about any two-unit system can easily be mounted in this way. In most cases, coaxial units are somewhat cheaper than separate units, which is, of course, some advantage.

One other type of speaker has only one voice-coil assembly but includes some special provision for the high frequencies. Such a one is the Stephens 101FR illustrated in *Fig.* 21. Known as a "cospiral" speaker, it has a differential diffuser composed of two concentric horns designed to diffuse the high frequencies, which in the normal single-cone speaker are very directional and can be heard best when the listener is on a line directly on the axis of the speaker. While speakers of this type have some advantages, they are very definitely *not* coaxial or two-way systems in any sense and do not have the characteristics or advantages of two-way systems.

The Need for Baffling

The best woofer ever made will sound like a dime-store squawkbox if you use it without a baffle or enclosure. With a baffle or enclosure it may reproduce the lowest bass notes on that new organ record you bought yesterday. Obviously, any component which makes that kind of a difference is not just a piece of furniture; it is an essential and critical part of the system just as much as and in the same sense as the amplifier or pickup. It is not very difficult to understand why this is so and how that vast improvement is made by a variety of different enclosures types. And once you have that information you can make your choice more intelligently.

Sound itself is, as you undoubtedly know, waves of compression and rarefaction of air. As it arrives at your ear the result is variations on pressure of the air against your eardrums, so that the eardrums vibrate like diaphragms and through some process as mysterious to us as it is to you (unless you are a physician) the brain perceives sound.

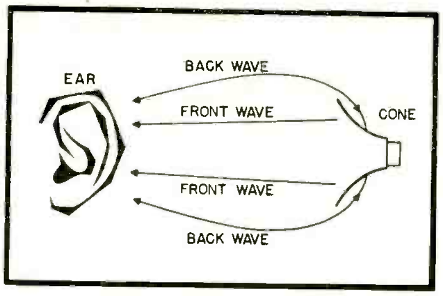


Fig. 22. This illustration shows the basic reason for a baffle or cabinet of some sort. The backwave, originating rear of the speaker cone, is 180 degrees out of phase with the front wave. If both reach the ear at about the same time, the net sound pressure at the ear will be zero.

It is the business of a speaker cone to compress and rarefy the air which is touching the cone. Each time this happens a wavetrain is started, which travels away from the speaker. The wavetrain begins losing strength—the compressions and rarefactions are no longer so intense—as it travels away from the speaker, and after some distance may not be perceptible at all, depending on the sensitivity of the instrument used to detect them—your ears.

Figure 22 is a rough sketch of a speaker cone and an ear. The sketch shows that when the cone vibrates there are two paths through which sound reaches the ear. If the cone moves toward the ear it compresses the air in front of it, and that starts a direct wave train traveling toward the ear. But at the same instant the rear of the speaker cone is *rarefying the air behind it*—and this starts a second wave train which travels around the back of the speaker toward your ear. Except at high frequencies, sound travels in pretty well every direction from its source, so that we can assume both the front and back waves are traveling in all directions. But we are interested only in what goes to the ear, so we have shown sound paths from the speaker only to the ear.

Now, if both front and back waves are generated at the same time, as must be true, and if, as you can see from *Fig. 22*, the back wave reaches the ear over almost as short a path as the front wave, the fact that one begins with a compression and the other with a rarefaction means that they are 180 degrees out of phase and will almost cancel; the ear will hear very little.

If, on the other hand, there is a substantial difference between the times when the waves reach the ear, cancellation will not necessarily occur. In fact, if the time difference is equal to one half-cycle, both will arrive at the ear in phase and will reinforce. The actual difference in arrival time depends, of course, on what extra distance the backwave must travel in terms of the wavelength of the particular frequency in air.

This is illustrated in *Fig.* 23. We have taken the two paths and straightened them out so that we just show the extra distance to be traveled by the back wave. The front wave starts with a maximum compression and travels to the ear. We want the backwave to reinforce it, so the backwave must be in phase with the front wave, and we have drawn it that way.

We know that if the front wave starts with a maximum compression, the backwave starts with a maximum rarefaction. Obviously it does that at a distance one half-wavelength farther back than the front wave, (point A) and, of course, one and a half wavelengths farther back (point C). We can then draw the conclusion that, for this frequency, if we want reinforcement of the two waves instead of partial or complete cancellation, the extra distance to be traveled by the backwave must be at least one half-wavelength. As a matter of hard fact, we know that the extra distance any wave will actually travel is simply around the edge of the speaker cone—and that is assuming that we are just talking about

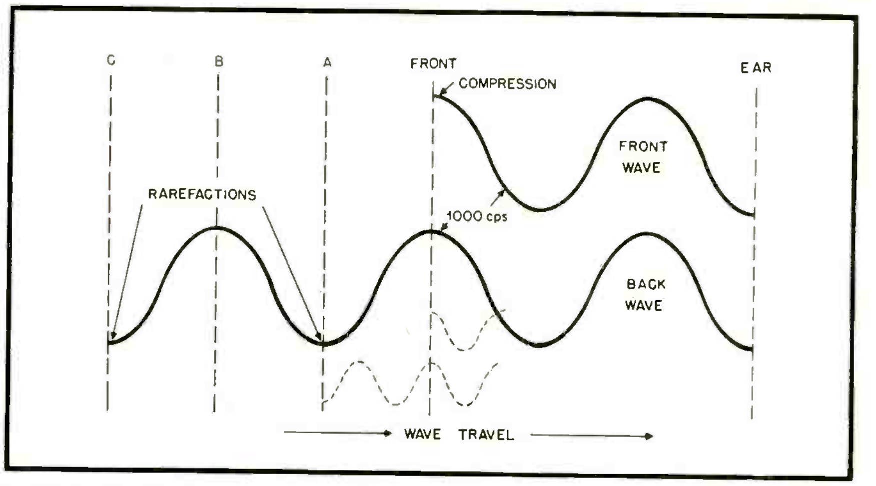


Fig. 23. In this drawing we have straightened out the path of the backwave to make phase relations clearer. If the back and front waves are to reinforce rather than cancel at the ear, the backwave must travel an extra distance equal to a half wavelength at the frequency of the tone. This means that either the speaker cone must be big enough to afford this extra distance or a baffle board must be used to increase the distance artificially. A baffle is not the ultimate answer.

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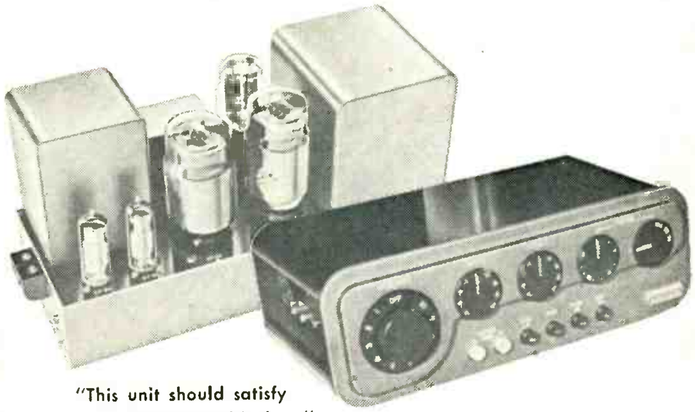

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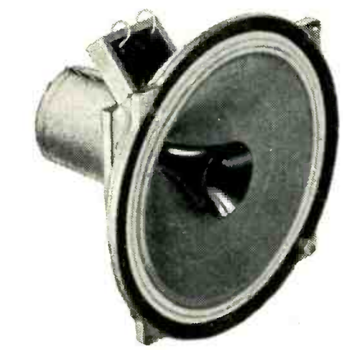

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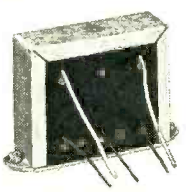

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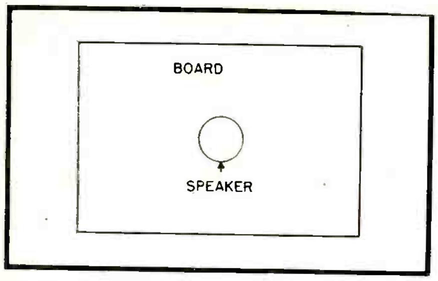

Fig. 24. Mounting the speaker on a flat baffle increases the distance the backwave must travel and makes it possible for front and back waves to arrive at the ear in phase at lower frequencies. Response is still spotty, however.

waves set in motion by a point very near the apex of the rear of the cone.

Let us assume a speaker with a 12-inch cone. The extra distance traveled by the backwave is a maximum of about 4 inches. Figuring 1100 feet per second for the speed of sound waves in air, we find that 4 inches is a halfwave at about 730 cps. If all our figures are correct, this means that 730 cps is the lowest frequency at which we could expect maximum sound output. Of course, the figures and assumptions are not correct, for many other factors enter, but in general the idea is quite correct and is the reason why no bass can be heard fom unbaffled speakers.

The obvious first step in curing this situation is to extend artificially the extra distance which the backwave must travel by mounting the speaker on a baffle board as in *Fig.* 24. Now, by making the baffle as large as possible, we can lower the lowest frequency at which reinforcement will take place because the backwave now has to travel from the back of the speaker over the back of the baffle to the edge before it can join the front wave. The flat baffle is often used in paging systems and the like where quality does not count for too much, but it is still not the ultimate answer.

To see why, refer again to *Fig.* 23. Suppose the lowest frequency at which reinforcement of the waves takes place is 1,000 cps. which is the frequency at which the length of a halfwave is equal to the distance between point A (the back of the speaker) and the front. The next frequency at which there is complete reinforcement is 3,000 cps, (dotted lines), at which the A-to-front distance is 1½ wavelengths. At every higher frequency at which that distance is equal to an odd number of half-wavelengths there will be complete reinforcement, and at frequencies between, there will be everything ranging from complete cancellation to almost complete reinforcement. Obviously the effective frequency response of the unbaffled speaker is very, very spotty.

The same is true of the baffled speaker. At every frequency at which the distance from the back of the speaker, across the board, and around the edge to the front wave is an odd number of half-wavelengths there is reinforcement, and at frequencies in between, everything else happens. The only improvement is that the lowest frequency of reinforcement is very much lowered so that it may perhaps be anything from 30 to 200 cps, depending on the size of the board. Actually, of course, the lowest useful frequency is lower than that at which reinforcement takes place because the ear can stand some cancellation before it yells for more bass. A common practice is to figure the distance from speaker to the nearest edge of the baffle at one-quarter wavelength at the lowest desired frequency.

There is only one way in which a baffle can give the results desired—complete elimination of any backwave interference at all. That is by making it infinitely large, so that the backwave never does reach the front. You may say that a baffle—or anything else for that matter—of infinite size is impossible. But you overlook the engineering profession which somehow seems to manage so many impossible things, like making internal-combustion engines work and arranging for big, bulky, obviously earthbound structures to take off like a bird and fly.

In this case if we can't beat 'em, we surround 'em. We take a box and seal it up tight except for one hole, against the inside of which we mount the speaker. Now, obviously, the backwave has no way to get out and interfere with the front wave, so the box is obviously an infinite baffle! Output is just as uniform as the electrical and mechanical structure of the speaker allow, as long as three conditions are fulfilled. First, the box must have sufficient space inside so that movements of the rear of the speaker cone do not tend to be impeded by air it compresses. If the box is very small, it will not contain much air. Any movement of the cone will appreciably compress that small volume of air and the air will then resist cone movement. To prevent this from happening, the inside volume should be somewhere between 6½ and 15 cubic feet. The volume cannot be too great but it can be too small, so infinite baffles, as these boxes are called, should be as big as possible.

Secondly, the box must be very sturdy and solid, so that vibration of the speaker cone is not transmitted to and by the box walls. If the walls vibrate, they simply become extensions of the cone and our troubles begin all over again.

Third, at least one of each two parallel walls of the inside of the box should be lined with some very absorbent material such as Ozite (the stuff you put under rugs), heavy felt, or thick Celotex. Preferably all walls should be lined. If they are not, sound waves from the rear of the speaker cone will strike the hard inside walls of the cabinet and bounce back against the cone, creating interferences of a random or unpredictable nature.

But if all these conditions are adhered to—and they are really very simple ones—the infinite baffle is one of the most satisfactory and effective speaker enclosures known. It does not radiate bass quite as well as some of the more complex enclosures such as horns, because there is still no provision for matching the acoustic impedance of the cone to that of the air, but the infinite baffle has an honored place in some of the very best home music systems in the country. It has the added advantage of being almost the only type of enclosure which will bring out the best in any speaker for which the cabinet is big enough, so that it is the safest buy when speaker and enclosure are bought separately. This is not true of many other enclosures; even folded horns work best with one or more particular speakers with which they were designed.

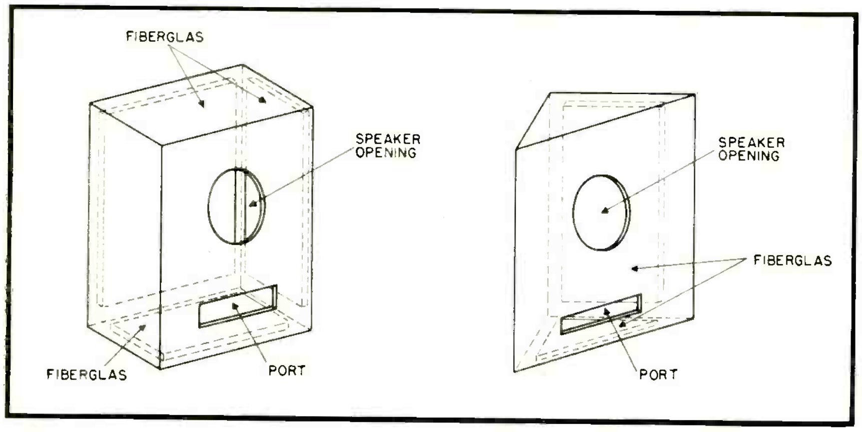

Fig. 25. The bass-reflex cabinet, originally made by Jensen before the war, calls for the box size and the port to form a Helmholtz resonator at a frequency equal to or slightly below the resonance of the speaker itself. At this resonant box frequency the backwave emerges from the port out of phase with the front wave and tends to cancel the resonant peak of the speaker, flattening out the response and tending to extend it somewhat lower in frequency. Bass-reflex enclosures may be made for corner placement, an ideal location for any speaker.

Bass-Reflex Enclosures

The output-vs-frequency characteristic of all loudspeakers (and the impedance-vs-frequency characteristic, which is easier to measure) shows a more or less large peak at the resonant frequency. If this peak is not reduced by the enclosure, a sort of one-note-bass effect is heard. The infinite baffle does tend to depress this peak to some extent, but the bass-reflex type of enclosure does it to a much greater degree *if it is correctly designed.*

The bass-reflex cabinet is a box like the infinite baffle, but it has a port cut into the front wall. This is illustrated by the two drawings of *Fig.* 25, showing bass-reflex cabinet shapes for wall and for corner placement. *Figure* 26 shows

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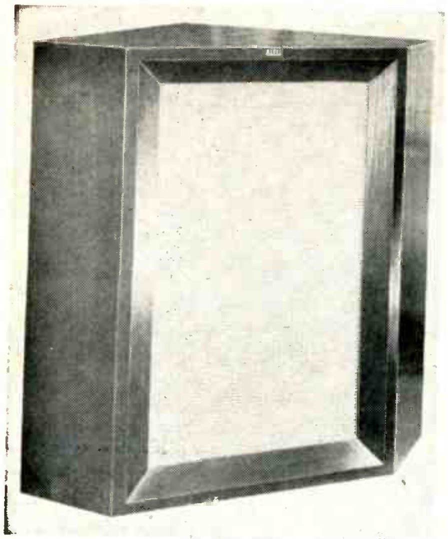


Fig. 26. This is a finished bass-reflex enclosure which is found in many living rooms. Like most enclosures its front appearance gives little clue to what its innards look like, but it is simply a box with a speaker hole and a port correctly proportioned for particular speakers.

a finished bass-reflex corner cabinet, the Altec 606, built according to the drawing in *Fig. 25*.

The port makes use of the back wave generated by the rear of the speaker cone to oppose the front wave at the frequency of speaker resonance. The result is a substantial dip in the resonant peak. This is illustrated by the two curves of speaker impedance against frequency shown in *Fig. 27*. These were taken on an Altec 602A speaker, with a resonance of 45 cps. Curve A was taken with the speaker unmounted and clearly shows the resonance. Curve B was taken in a correctly designed bass-reflex cabinet and the former point of resonance now has a decided dip, so that the over-all result is considerably flatter and smoother than before.

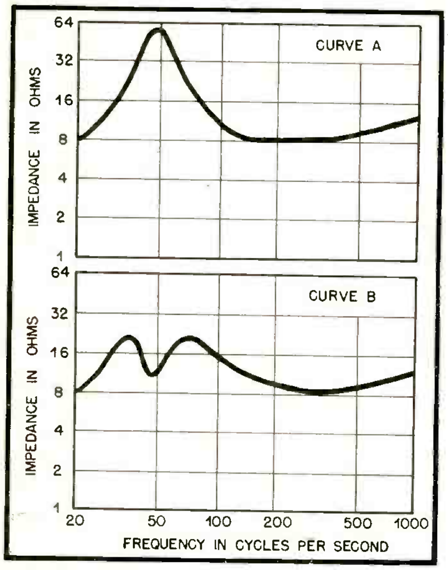


Fig. 27. These curves show in a very forceful way how a correctly designed bass-reflex enclosure functions. Curve A shows the very prominent response peak at speaker resonance and Curve B shows how that peak is suppressed and the entire characteristic smoothed out.

This is done by designing the shape and size of the box together with the shape and size of the port so that the resonance of box and port takes place at the same frequency as resonance of the speaker. At that frequency, then, there will be considerable radiation of sound from the rear of the speaker through the port. Since the rear of the speaker cone is now doing work at the resonant frequency, it is said to be back-loaded and the additional resistance caused by its having to move air reduces its ability to *boinggg* at resonance! It can also be looked at another way—that the inverse-phase radiation from the port at resonance tends to cancel the normal high power output from the cone front at that frequency.

Bass-reflex enclosures are sometimes used in another way, in situations where an infinite baffle of correct size would be too large for comfort or where speaker resonance is at too high a frequency to give the desired bass. Here the enclosure is made to resonate at a frequency somewhat *below* speaker resonance. The enclosure resonance then damps out part of the speaker resonance and also creates an extended bass response by causing resonant peaks of its own to boost the falling speaker curve. An example of this is *Fig. 28*, where a speaker with a 58-cps resonance is used in a bass-reflex cabinet with a resonance at about 52 cps. Note the additional small peak between 30 and 40 cps which extends the bass response.

When a bass-reflex enclosure is carefully designed by very competent people who take sufficient pains and when it is designed with and for one particular speaker or line of speakers, it does the job it sets out to do. Ported enclosures are, however, extremely critical and simply cannot give best results when used with just any speaker. It is important to note that the bass-reflex is a resonant enclosure and if the resonance is not damped by placing soft material against the inside walls of the cabinet there will very definitely be a boomy, "boxy" sound. Milton S. Snitzer's article, "Adventures with a bass-reflex," on page 26 of the January 1954 issue of AUDIO is a very good source of information on just how critical a bass-reflex enclosure is.

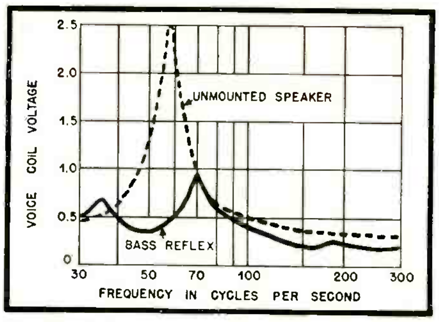


Fig. 28. Taken from the Snitzer article referred to in the text, this curve shows how a speaker with a 58-cps resonance was smoothed out by a well designed bass-reflex enclosure. The article shows in detail how much care and experimentation were necessary to obtain this final result and it is recommended reading on the subject.

Horns

Theoretically the most satisfactory way of radiating sound from any loudspeaker is to place it at the apex of an approximately exponential horn. This is a horn whose cross-sectional area is multiplied by some constant number for each unit of axial length from the apex. For instance if the number is 2, the area is multiplied by 2 for each unit of length. If the chosen unit is 5 inches, then area doubles for each 5 inches of length: If the area is 1 square inch at the apex, it is 2 square inches at 5-inch length; 4 square inches at 10-inch length, 8 square inches at 15-inch length, 16 square inches at 20-inch length, 32 square inches at 25-inch length, and so on. A horn of this type is shown in cross-section in *Fig. 29*.

We know from electricity that the most efficient transfer of power from a source to a load takes place when the impedances of the two are equal. The same is true in a mechanical or acoustic system, for these, too, have impedances.

It would be unnecessarily complex to go through the rigid explanation which would show why impedance of the speaker cone and the air in contact with it do not match. But we can explain it

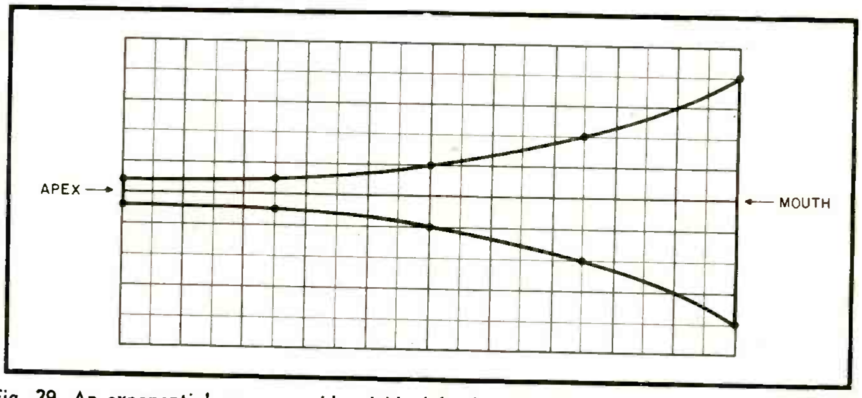


Fig. 29. An exponential curve, considered ideal for horns is shown in outline here. In primitive form—that of a straight-sided cone—the horn has been used for hundreds of years to improve the efficiency of sound transmission. Prime example is the megaphone, which made possible communication between ships at sea before any kind of electronic sound reinforcement was invented. An even more primitive example is simply cupping of hands around the mouth. When correctly shaped, a horn is the most efficient method of coupling point-source sound to the surrounding air.

briefly in such a way as to leave a clear picture in your mind.

We would like to transfer all the power of the speaker cone movement into acoustical movement of the air. The cone is, compared to the air, relatively stiff, and when a force is applied to a small part of it, the apex, the whole cone moves. Air, on the other hand, is a highly compressible medium. When we apply force to a small part of the air in a room—that portion of the air touching the speaker cone—the compliance of the air tends to dissipate that force without making much of the surrounding air move. It is as if you tried to make the entire body of a foam rubber cushion move by poking it with your finger. The point where you push just compresses without imparting much energy to the rubber around it.

To cause much of the cushion to move, you might set the big end of a solid cone against the cushion and apply your finger poke to the apex of the cone; in other words, you would be transforming your point-source energy to a larger area of cushion to compensate for its resiliency—and this is what a horn does for a speaker.

A small speaker diaphragm is applied to the apex of the horn of *Fig.* 29. It moves only the small amount of air present at the apex, but it moves that air rather efficiently because it is confined by the small area inside the horn walls and cannot easily compress as it could if not confined.

The small volume of air moved at the apex in turn moves a volume of the air a bit further toward the mouth. This second volume is somewhat larger but still confined—and it is being moved by the first air volume, which is practically its own size. As we move further up the horn, there is a very gradual and constant *transition* from the small, highly resistant air mass at the apex to the large, very resilient air mass at the horn's mouth.

Look at it another way. In electricity Ohm's law says that $R = E/I$, E being (electromotive) force and I being (electrical) current. At the horn's apex, E, force per unit volume is very high because all the force of the speaker diaphragm is being expended against a very small column of air; I, the amount of air being moved is very small, because there just isn't much air in such a small space. Therefore, with E high and I low, R, the acoustical resistance at the apex, must be high.

As we get toward the mouth of the horn E is much smaller because the same initial force is moving a much larger volume of air and the force expended against any individual segment of that volume is small. But I is higher because there is more air to move and therefore more moves. Thus R, the air resistance, is much lower. At the horn's mouth it is the lowest of any place in the horn and comes nearest to matching the impedance of the free air in the room. The horn has thus acted in a way as an impedance transformer matching the low-impedance room air to the high-impedance speaker diaphragm just as a voice coil is matched by the output transformer to the plates of the power tubes. The difference is that the transformer impedance change is abrupt; it just happens to be possible to wind coils to do the job. The acoustic impedance change must be made by gradual transition—and incidentally, exactly the same technique is used for impedance changing of microwave electricity—waveguide transitions whose cross-sectional area gradually changes.

When we have achieved an impedance match in this way, the entire system can be very efficient, as much as 50 per cent of the electrical power fed to the voice coil resulting in useful acoustic output, compared to just a few per cent with simple cones and the resulting impedance mismatch. The improvement which results is principally in the low frequencies, because the same power is expended in driving a small amount of air fast as in driving a large amount slowly. The speaker couples enough power in the fast high-frequency vibrations to the small volume of air near the cone, and the cone itself has a hornlike flare. But note that most tweeters use horns to couple the rather small diaphragms to a larger volume of air.

Every exponential horn has a low-frequency cutoff, below which it will not operate satisfactorily. This is determined by the flare rate (the distance in which cross-section area doubles) and the mouth diameter. A practical horn for use down to around 50 cps might have a length of 16 feet or more and a mouth diameter of perhaps 14 feet. Such a monster would hardly fit into the average living room, so straight horns are to be found only in the possession of those whom we will for the sake of politeness call very special people.

It is possible, however, to fold the horn so that it occupies much less space, and then to use the walls of a corner of the room as an extension of it. This is a considerable compromise, for the flare is no longer gradual and the bends introduce side effects. However, it is practical and there are some very fine folded-horn enclosures

One of the earliest to become popular and still among the best is that invented by Paul W. Klipsch, sold by the inventor's own firm as the Klipschorn and by other firms under license by other names. The Klipschorn is shown in *Fig.* 30. The top section holds a tweeter, while the lower section encloses the horn.

Unfortunately, the inside of a folded-horn enclosure is too complex to be shown clearly other than by a rather complete set of drawings—and even then it takes a mechanical engineer to get a mental picture of what it is. The cutaway drawing of *Fig.* 31 is of an early Klipsch folded horn and simply gives some idea of the complexity of the inside. The general idea is to enclose the speaker in a fairly small space inside the cabinet and then lead the sound out through a sort of winding tunnel or labyrinth whose cross-section area grows larger as gradually as possible. In the Klipsch design the sound emerges at the sides of the enclosure and the walls of the room are used as a horn extension. The enclosure is placed in a corner for this purpose.

The photograph of *Fig.* 32 shows the LEE catenoid folded horn, an enclosure of the horn type using a rate of flare computed according to the so-called catenoid curve, which we will not bother to explain mathematically. A vertical

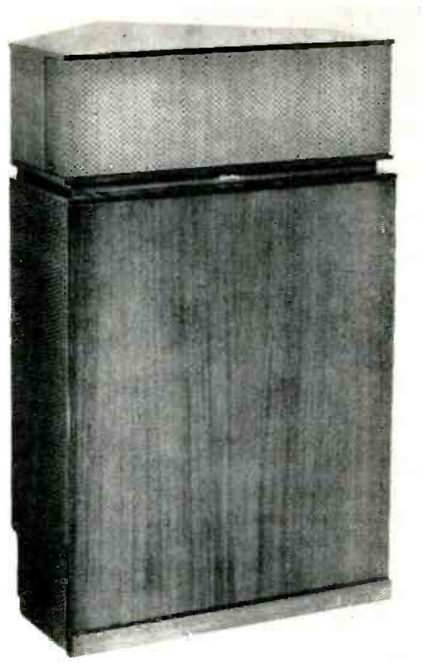


Fig. 30. The Klipschorn is one of the most popular and best sounding embodiments of the folded-horn idea. Designed for room-corner placement, it leads sound from the speaker cone front through a path of gradually increasing cross-section, then lets it emerge from the enclosure sides so that the room walls act as extensions of the horn.

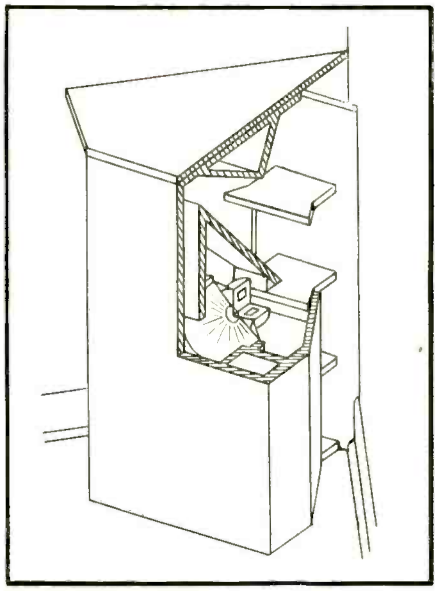


Fig. 31. This partially cutaway view of an early Klipsch horn gives some idea of the complexity of the inside panels, through like most drawings of folded horns fails to illustrate clearly the exact path which the sound follows.

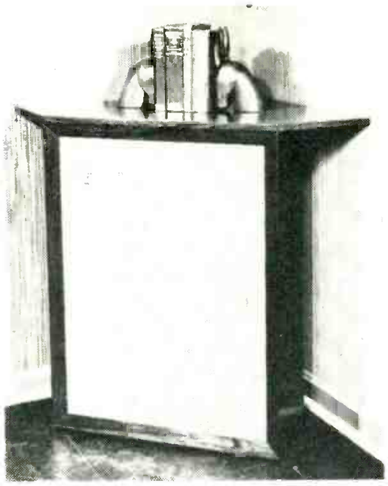


Fig. 32. This is a folded-horn enclosure made by Laboratory of Electronic Engineering, Inc. Its rate of cross-section area increase is not exponential but is based on the catenoid curve, claimed to have certain advantages.

cross-section of this horn is shown in *Fig.* 33 giving some idea of how the internal horn is constructed. *Figure* 34 shows a transparent model of the enclosure with arrows giving an idea of the path taken by the sound. As photographed the apex of the triangular enclosure which fits into the room corner is facing the reader. Many horn-type enclosures use the horn only for bass reinforcement. These have the speaker attached over a hole in the front panel so that there is direct radiation, then lead the back wave from the rear of the speaker through a folded horn and into the wall corner.

Choosing and Installing Speaker Systems

While the information we have given up to this point is a help in that the purchaser of almost any article does better if he knows how the gadget he is buying works, selection of anyone's ideal speaker system is still a matter of choice and compromise from many angles.

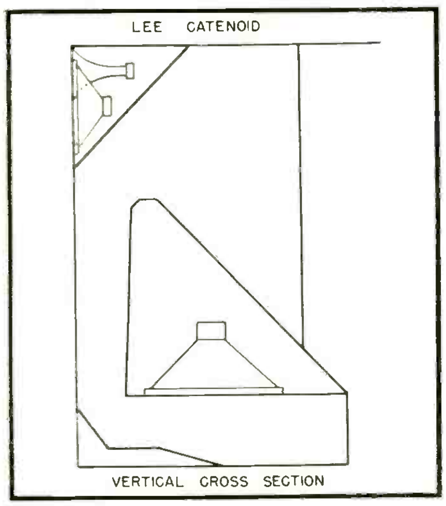


Fig. 33. A horizontal cross-section of the LEE catenoid-curve horn gives some idea of the configuration inside the cabinet to give gradual horn effect. The horn mouth does not expand relatively as rapidly as this drawing indicates; some components cannot be shown in this drawing

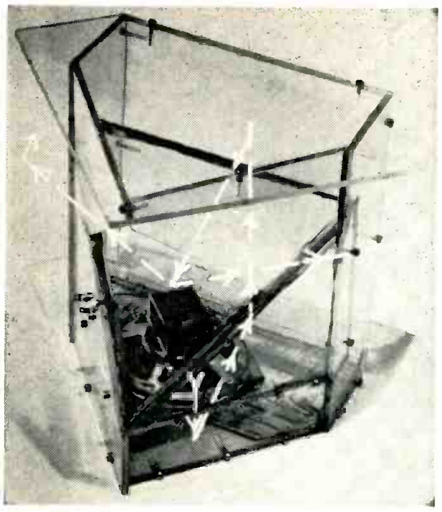


Fig. 34. LEE has made up a transparent model of the catenoid enclosure which in this photo has its rear (the part going into the corner) facing the reader. The arrows attempt to show the path taken by the sound. The woofer is clearly shown at the bottom of the assembly.

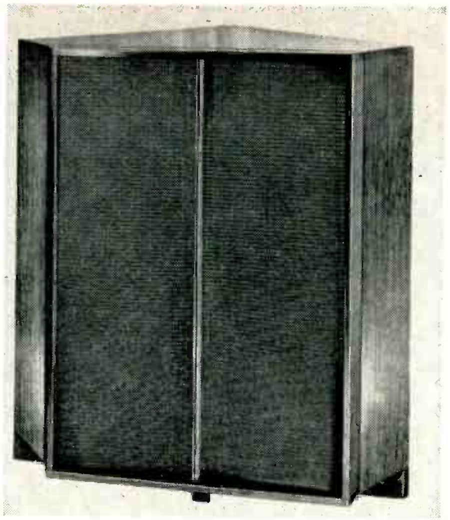


Fig. 35. The front or living-room view of the Altec 820A speaker system. This enclosure, like many of the best, is intended for corner use, an excellent idea for any cabinet.

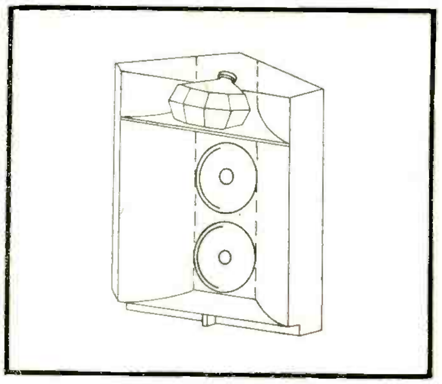


Fig. 36. Altec's description of the 820A enclosure as a "direct-radiator horn" is made clear by this outline drawing. The bass speakers are mounted at the apex of a simple horn formed of two curved side panels, and the corner placement lets the room walls act as a continuation of the horn. This type of horn is rudimentary and not particularly greatly effective as such, though it helps. There is space enough behind the speakers for infinite-baffle effect.

The first question is, of course, what the budget amounts to. In the speaker field as elsewhere you get pretty much the quality you pay for. When building an enclosure you will get pretty much the quality your industry and skill allow. For instance, it is easy to build a good infinite baffle; but unless you are pretty skilled with tools you will have a hard time duplicating a folded horn. Bass-reflex cabinets should not be attempted unless you are willing to go through a cut-and-try procedure after you are finished.

When the budget is low, a single speaker or a coaxial will probably be the necessary choice, together with an infinite baffle (hard to find on the market, surprisingly) or bass-reflex. Spend the most you can afford for the speaker and if you want a bass-reflex enclosure try to find one designed for your speaker.

When the budget is higher, a two- or three-way system is in order. The trick then is to go to an audio showroom and spend a good deal of time just listening, preferably with an amplifier and pickup similar to what your system has or will have. Speaker systems have individuality in the sound and you will like some better than others. But don't decide after less than about 30 minutes of listening. It should be much more but at that rate none of the stores would have time to do any business.

Folded horns are fairly common for the quality trade. Many are made, and others are made with variations and combinations of principles. Again, sound counts. In any case some kind of corner enclosure is usually best if you have a corner to put it in, even if it isn't a horn. The walls of the corner tend to act a little like a horn and improve almost anything; and a corner is the ideal location for putting you on or near the beam of the speaker almost no matter where you sit. If your tweeter has a 90-degree or more dispersion angle and it's in a corner, it will cover every part of the room. If it's on a wall, you can't sit next to it and hear the highs.

It's a small point but be sure the speaker system you buy has an impedance your amplifier output can furnish. Most do, but make sure. Don't put any level controls between amplifier and speaker (L and T pads are sold for this purpose) because they will destroy the damping of your speaker system and muddy the sound. If you have a bass-reflex or infinite-baffle enclosure (or one of the others, particularly the miniatures) don't be afraid to boost the bass somewhere in the amplifier system. Take it easy, though, to avoid getting punched in the stomach with a heavy drum beat —bass isn't supposed to be that strong. It helps to attend a live music performance every now and then to see whether you have things adjusted for high fidelity to the original and not high fealty to the highs and lows you think you ought to hear. Remember that the best records to test your system aren't drums and cymbals—they are records of ordinary musical selections with full orchestras or pianos or string quartets. If you just want to hear the thuds of the drums or the tinkles of the triangles, it's really much cheaper to go to a music store and buy the real thing. No lessons required either!

Who Makes What

To conclude this special loudspeaker section of AUDIO we list and briefly describe a number of loudspeaker products made by foremost manufacturers in the field. This is for your guidance. We do not list everything made by each manufacturer, only the products we feel are more interesting and appropriate. Further literature and details can be obtained by writing directly to the companies.

Altec Lansing Corp., 9356 Santa Monica Blvd., Beverly Hills, Calif. Altec's 604C coaxial speaker with multicellular high-frequency horn is shown in *Fig.* 18. A complete crossover network, neatly packaged as most of them are, goes with it. The 820A speaker system shown in *Fig.* 35 is another high-quality Altec staple. It consists of two 15-inch tweeters and a multicellular-horn tweeter, with crossover at 800 cps. The corner enclosure is a combination of direct radiating horn and bass-reflex. The drawing of *Fig.* 36 shows how the inside walls of the enclosure form the rudimentary horn, which the room walls continue.

Beam Instruments Corp., 350 Fifth Avenue, New York 1, N. Y. Beam's Stentorian line of speakers is manufactured by a British firm, Whitely Electrical Radio Co., joining the growing influx of British-made sound products. Most Stentorians are coaxial units, like the 12-inch unit of *Fig.* 37, and come with crossover network. Single tweeters like the T12 in *Fig.* 38 can be bought separately. Low-frequency cutoff is 2,000 cps and power rating 15 watts The much less expensive T10 is rated at 5 watts, with response to 16,000 rather than 20,000 cps, an insignificant frequency difference unless you are a dog. Beam also offers bass-reflex enclosures.

R. T. Bozak Co., 114 Manhattan Street, Stamford, Conn. Bozak makes

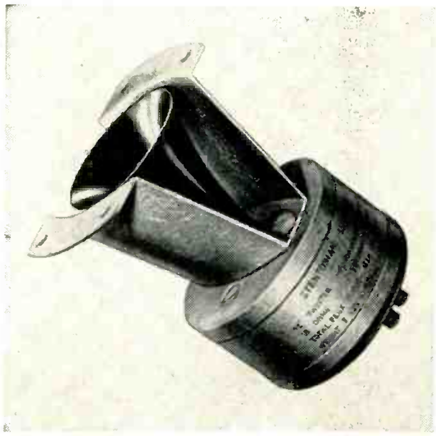


Fig. 38. This Beam Stentorian tweeter is the same one used in the coaxial assembly of Fig. 37, but is available separately with the mounting shown allowing it to be placed on any cabinet wall over a hole of suitable size. It is rated at 15 watts and is used with a 2,000-cps crossover.

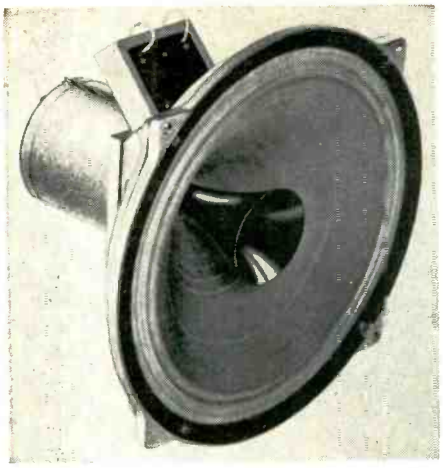


Fig. 37. A complete Stentorian coaxial speaker system, which comes with crossover network and is made by the British Whitely firm.

woofers, tweeters, and three-way systems. The tweeters are small cone speakers as shown in *Fig. 7*. Their most elaborate system is the B-310 shown in *Fig.* 39, containing four woofers, a mid-range cone speaker, and eight of the small cone-type tweeters. They claim to be the only company offering a tweeter system with 180-degree dispersion. Bozak is notable as one of the few manufacturers definitely recommending infinite baffles rather than ported or horn enclosures. They feel, undoubtedly with a good deal of justification, that infinite baffles damp resonances and avoid introducing extraneous peaks best of any type. To make up for relative bass inefficiency they tend toward multiple woofers, a practice formerly in vogue but now largely abandoned by the gadgeteers—mainly because of the cost.

British Industries Corp., 164 Duane Street, New York 13, N. Y. is the U. S.

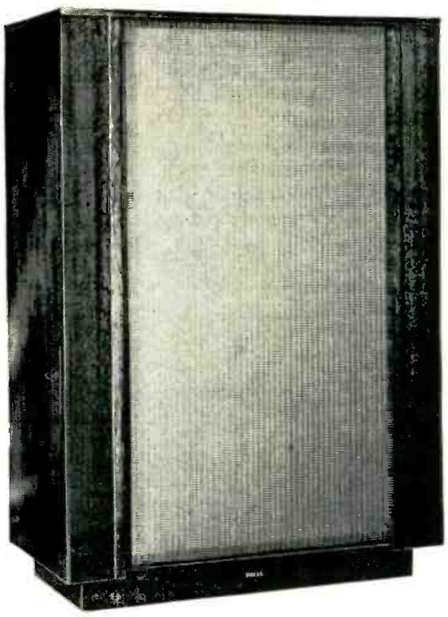


Fig. 39. Bozak systems employ the small cone speakers as tweeters. This photograph shows the most elaborate Bozak system, a 3-way job containing four woofers, one midrange speaker, and eight of the tweeters. The multiple woofers aid bass in the recommended infinite baffles.

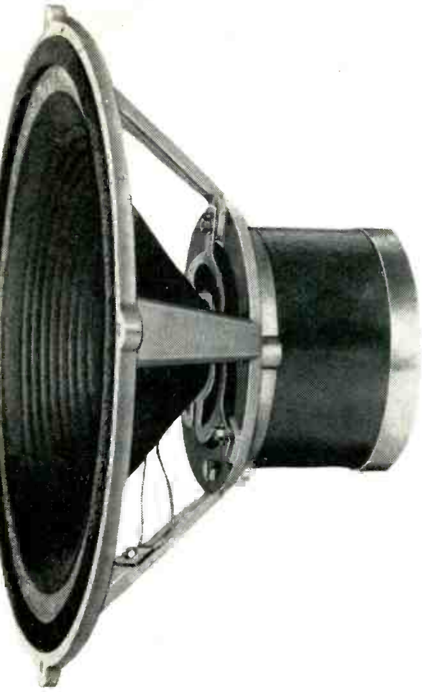


Fig. 40. The Wharfedale 12/CS/AL speaker. This is a full-range speaker which may be used without tweeters. Its cone is relatively stiff for good high-frequency piston action, but its edges are fastened to the outer edge of the frame with rather limp cloth so that it can make large excursions as required at bass frequencies.

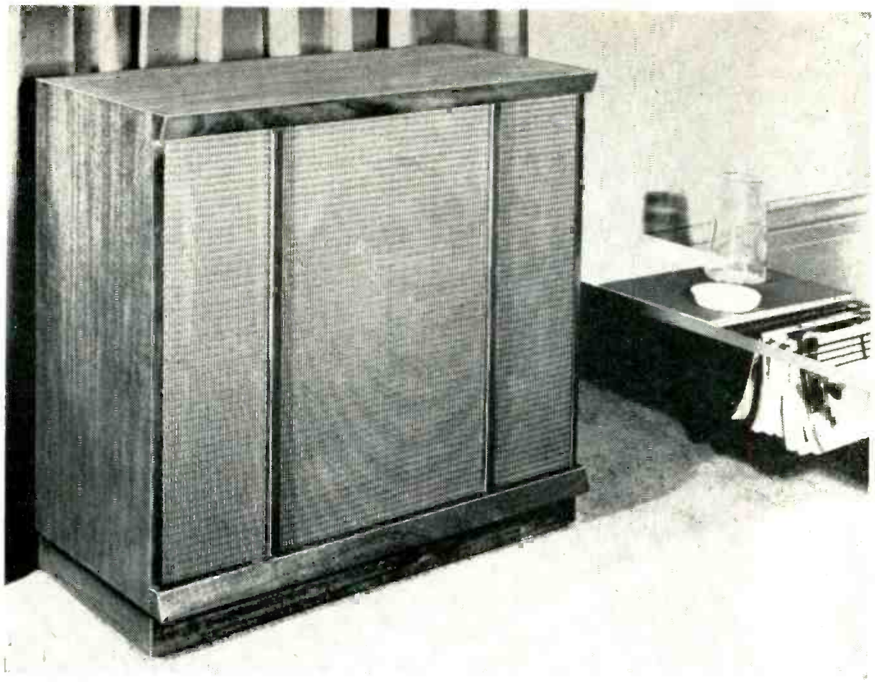


Fig. 43. Cabinart enclosures are made by G. & H. Wood Products, one of the fe firms making enclosures only, no speakers. This model is the KR-4-W, a Klipsch-type folded horn which can be obtained ready-built and finished or in kit form for home assembly.

importer and distributor of Wharfedale speakers, which are made in Britain. *Figure* 40 is a photo of the Wharfedale Super 12/CS/AL, a full-range unit often used without tweeters. Like most of this maker's line, it has a cloth suspension holding the outer edge of the cone to the frame, giving it greater compliance and lower-frequency bass resonance. Wharfedale also makes interesting treble speakers which look like ordinary cone units but which have a "bakelized" cone for greater stiffness.

Brociner Electronics Laboratory, 344 East 32nd Street, New York 16, N. Y.

Brociner was among the first licensees of Klipsch for the folded horn. They still specialize in horns, one of which, the 4M, is shown in *Fig.* 41. *Figure* 42 is a clear and simplified drawing of a vertical cross-section of the horn, with arrows showing the sound path. Notice how the area of the horn increases. The

Fig. 41. The Brociner type 4M horn. Similar units are made in a variety of styles and finishes, all on the same principle.

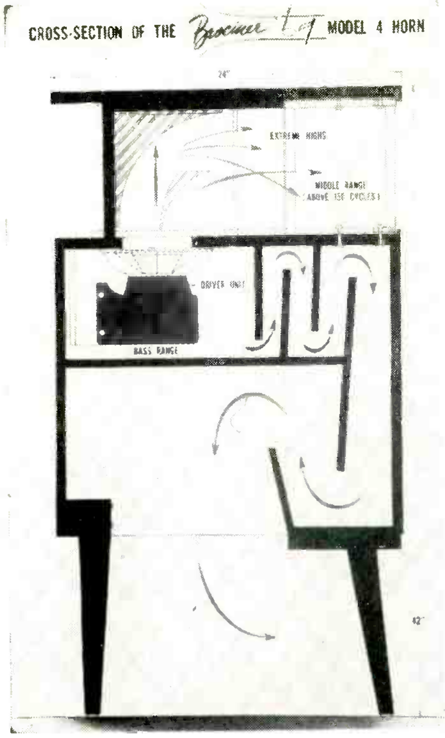


Fig. 42. Brociner was one of the first licensees to make Klipsch-type horn enclosures. This drawing shows an inside view of the cabinet of Fig. 41. The special British Lowther speaker, designed for this use, has two cones, one for high and the other for low frequencies, with mechanical crossover.

special British Lowther PM-2 speaker used by Brociner is unique in that it is designed exclusively for horn use, with two cones, one within the other. The inner cone is light and is used for the highs, being coupled to two rudimentary horns at the cabinet top. The larger cone feeds the big horn from its back. The crossover is mechanical, there being a coupling between the cones which is too compliant to allow the low-frequency cone to vibrate at frequencies above about 4,000 cps. The low-frequency cone is only about 6 inches in diameter, allowing a small horn apex and large horn length (7 feet), and giving smoother frequency response.

Cabinart is the trade name used by *G. & H. Wood Products, 75 North 11th Street, Brooklyn 11, N. Y.* The company makes only enclosures—no speakers. A top item of the line is the KR-4-W Klipsch-licensed horn shown in *Fig.* 43. Prices are moderate throughout, but a special attraction is the corner horn available in kit form at less than $50. A newer design, model KR-5, is somewhat smaller, being only 20 in. high, but is equipped for either 8- or 12-in. speakers.

DeMars Engineering and Mfg. Co., 360 Merrimac St., Lawrence, Mass., headed by Paul deMars, is the maker of the Styrocone speaker illustrated in *Fig.* 44. This interesting new idea is based on the thought that woofers are relatively inefficient because they don't push enough air—the reason why people ordinarily buy a 15-inch woofer in preference to one sized at 12 inches or less if they can. The big board seen in the photo is made of Styrofoam, a rather rigid material very similar to foam rubber in that it is mostly air and conse-

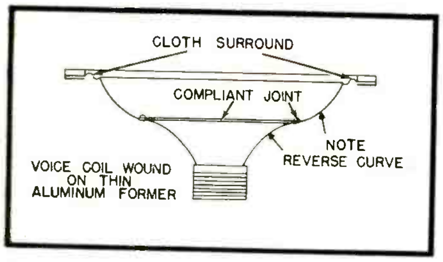


Fig. 51. The complete Hartley 215 speaker shown in cross-section in Fig. 50. This 10-inch model has a cone composed of two separate sections with a compliant joint between them and a reverse curve to the outer section which provides a stiffer structure at low frequencies without raising the resonant frequency unduly.

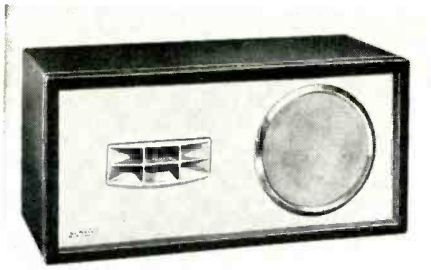


Fig. 52. The Jensen Duette, shown above, is a contribution to the cause of smaller speaker enclosures for portable use or for those who want portability, for which latter purpose it is also available in a portable case.

Jensen Mfg. Co., 6601 Laramie Avenue, Chicago 38, Ill. Jensen is really an old hand in the loudspeaker business and was the first to bring out bass-reflex cabinets, which was a registered name until only a few years ago, when Jensen released it to the public domain. The newest big cabinets are called Bass-Ultraflex and contain 2- and 3-way systems. The G-610 speaker shown in *Fig.* 19 is a top Jensen unit, with its three separate coaxial reproducers. A recent Jensen offering is the Duette of *Fig.* 52, a contribution to the modern desire for very small enclosures. The Duette is a two-way system with 8-inch woofer and multicellular-horn tweeter, with crossover network. Over-all dimensions are 11 inches high, 10 inches deep, and 23½ inches long. The bass speaker is in a tuned enclosure, but Jensen recommends some amplifier bass boost for best results, which is not surprising with an enclosure this size. The Duette is also available in a portable case with front cover and handle, which might be useful to those who do jobs in the field with good tape recorders.

Karlson Associates, Inc., 1379 East 15th Street, Brooklyn, 30, N. Y. The Karlson enclosure is shown in *Fig.* 53, and *Fig.* 54 gives a glimpse of the inside. We always think of a swallowtail coat when we see it. The principal is different from most others and is based on the broadbanding effect of an air column coupled to the outside through a slot of exponential shape. An article explaining it in detail appeared in the September 1952 issue of this magazine. Construction kits are also available to those who prefer building to buying.

Fig. 53. This is the enclosure made by Karlson Associates of Brooklyn, N. Y. It is not a horn but derives its effect from coupling an inside air column to the outer air through a slot of an exponential shape.

The Kelton Co., Inc., 55 Amory Street, Boston 19, Mass. Kelton appears to be chiefly the maker of one of the new "high-fidelity" small phonographs and incidentally to have come up with an interesting addition to the small-enclosure field. The enclosure looks like *Fig.* 55 in its high-class-finish model and like *Fig.* 56 for less money. The drawing of *Fig.* 57 shows its size and what's inside—principally two speakers, an 8-inch and a 6-inch unit. The upper speaker is for the highs only, from 200 to 12,000 cps and is enclosed in a simple, small absorptive chamber. The lower speaker is for the bass. The lower chamber is resonant and the separate holes which can be seen in *Fig.* 56 provide what GE calls a "dis-

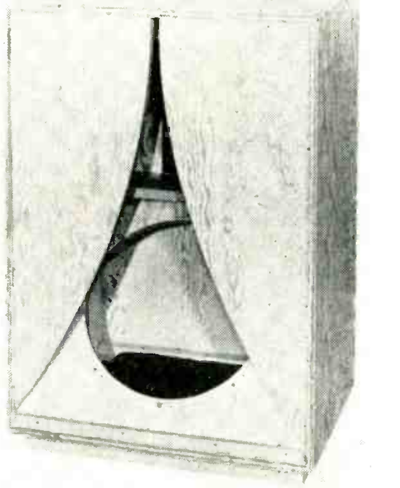


Fig. 54. This illustration gives an idea of what is behind the swallowtail opening in the Karlson speaker enclosure. The enclosure can be purchased already built and finished, but it is also available in the form of a complete kit for those who like home assembly.

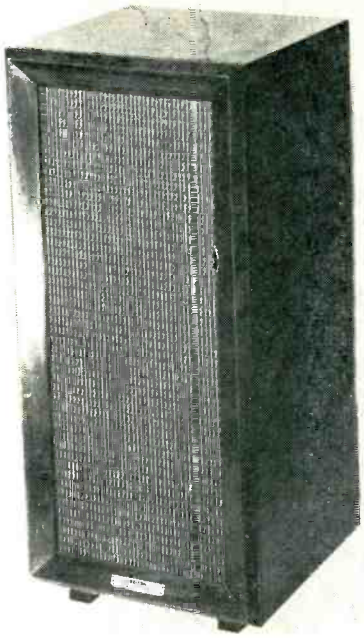


Fig. 55. This is the external appearance of the best enclosure made by The Kelton Co., another unit in the small-size category. An enclosure of this kind was included in a complete phonograph made by the company, now comes separately.

tributed port" (used differently). The special 8-inch speaker used provides larger than normal cone excursion, which is probably responsible for the fact that the system is claimed to be flat down to 50 cps.

Klipsch and Associates, Hope, Ark. It is difficult to say anything about Klipsch that has not already been said. He is the inventor of the most popular type of folded horn and licenses a good many companies as well as making a small line of his own, one of which was shown in *Fig.* 30.

Fig. 56. For a lower price, you can buy this version of the Kelton enclosure, which you can then finish in any way you like, though ordinary paint is probably best because of the fact that plywood is used in the construction.

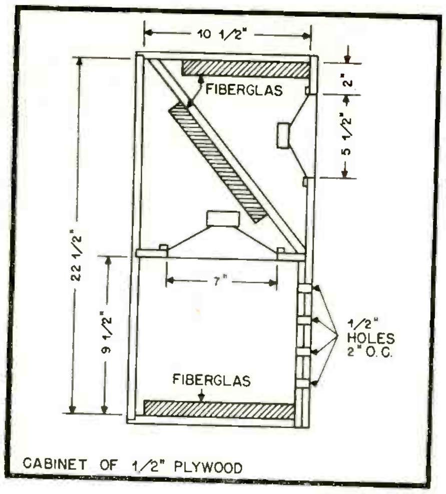


Fig. 57. The vertical cross-section of the Kelton enclosure, above, shows what's inside. A 6-inch speaker furnishes treble at the top and an 8-inch speaker, mounted face down in the center, provides bass. The bass compartment is enclosed except for the holes of a "distributed port."

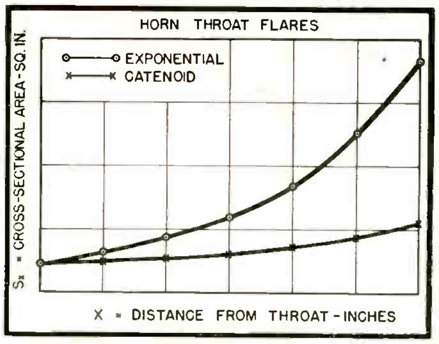


Fig. 58. The LEE speaker enclosure, illustrated earlier, is a horn using a catenoidal rather than exponential curve for the horn flare. Just for the record the two curves above compare the rates of exponential and catenoidal flares to give an idea of the difference between the two.

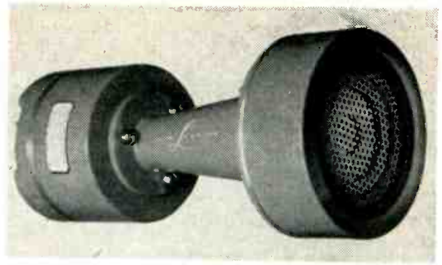


Fig. 59. Lansing's thesis is that multicellular tweeter horns create as many virtual sound sources as horn openings, which interfere and give nonlinear frequency effects. The Koustical Lens shown here is the answer.

LEE speaker enclosures are made by *Laboratory of Electronic Engineering, Inc., 413 L Street, Washington 1, D. C.* The principle of their enclosures, shown in *Figs.* 32, 33, and 34, is a horn with a catenary rather than exponential curve. Just for the record, *Fig.* 58 shows the difference between the two. It is claimed that the compromises necessary in any folded-horn design are more acceptable and less critical when the curve to be matched is catenoidal. LEE systems are three-way, with crossovers at 300 and 6,500 cps.

James B. Lansing Sound, Inc., 2439 Fletcher Drive, Los Angeles 39, Calif., makes the Jim Lansing line of speakers of all high-quality kinds and enclosures for them. One of their most interesting products, shown in *Fig.* 59, is the 175DLH tweeter unit with an acoustical lens or Koustical Lens, as the company's advertising puts it. The thesis is that in multicellular horns, usually used to disperse the normally highly directional output of high-frequency speakers, the mouth of each horn tends to act as a separate source of sound and phase interference effects occur. In the Lansing unit the sound is dispersed by the acoustical analog of an optical diffusing lens, made up of 14 perforated screens in the mouth of the horn, with holes sized and placed to achieve the 90-deg. dispersion effect illustrated by *Fig.* 60. By this means sound is uniform at all frequencies anywhere in the 90-deg. sector.

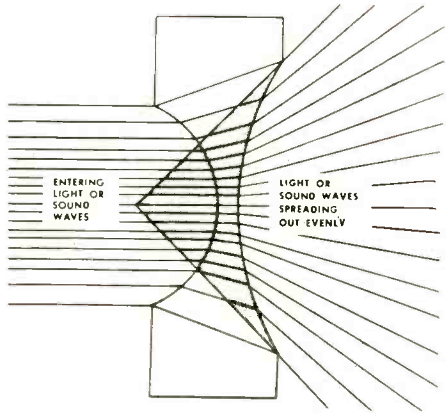


Fig. 60. This diagram shows that the high-frequency sound from the Lansing Koustical Lens tweeter is dispersed over a full 90-degree angle, with elimination of the spotty frequency effects caused by interferences in multiple horns.

The German-made Lorenz speakers and enclosures are distributed by *Kingdom Products Ltd., 23 Park Place, New York 7, N. Y.* The line includes some very well made speakers and some conventional-type enclosures, plus the unique and interesting Sound Corner illustrated in *Fig.* 61. The Sound Corner is a flat piece of wood of the shape shown, which mounts in a corner of the room about 13 inches from the ceiling. Because of its shape the enclosed volume increases from the speaker upward, forming the beginning of a horn, while the three joined walls at the ceiling level continue the horn effect and allow bass reproduction down to a surprisingly low frequency (the manufacturer claims 31 cps). The unit, meanwhile, is out of the way and inconspicuous. Rubber gasketing around the board edges makes a good seal to the walls and the little triangular bottom piece is spring-hinged so it closes up things nicely there. The LP-215 bass speaker is about an 8-inch unit (215 mm), and the little cone-type LP-65 tweeter has a 2½-inch (65 mm) cone. The LP-215 has a free-air resonance in

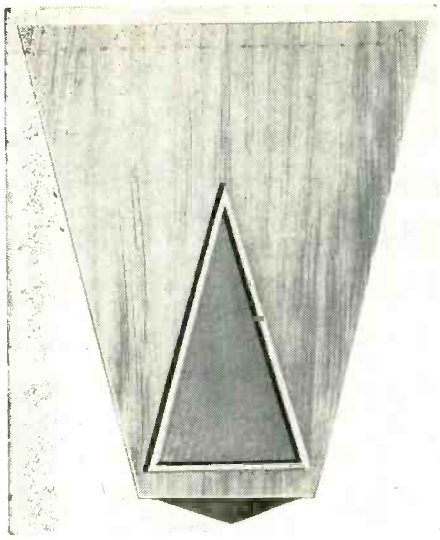


Fig. 61. The Lorenz Sound Corner, an ingenious answer to the question of what to do for a speaker enclosure when space is at a premium.

the vicinity of 70 cps, which is low for a small-diameter cone. The tweeter has a solid frame so it can be mounted in an enclosed cabinet—such as a bass reflex—without any chance of damage from the high internal pressures at low frequencies in such a box. The only crossover used is a small capacitor in series with the tweeter. Price of the Sound Corner is pleasantly low, as you might expect with such simple "enclosure" construction.

Lowell Manufacturing Company, 3030 Laclede Station Road, St. Louis 17, Mo. makes a complete line of speaker baffles for installations where loudspeakers are to be built in—particularly for homes where many rooms are to have speakers, and for offices, clubs, railroad cars, and so on. Their line includes a variety of units for ceiling installation as well as flush-type wall baffles. *Figure* 62 is one example of the ceiling type. These units provide excellent dispersion, and are constructed to be rattle-free. Also available are prefabricated grille units for home or office use, speaker housings combined with Circline fluorescent lamps, and most accessories required in making the installation.

Permoflux Corp., 4900 West Grand Avenue, Chicago 39, Ill., is another old-timer in the speaker business. They make a line of cone-type speakers including tweeters, plus a miniature system, the Diminuette of *Fig.* 63, and a full-size unit, the Fortissimo. The Largo is a third system soon to be seen.

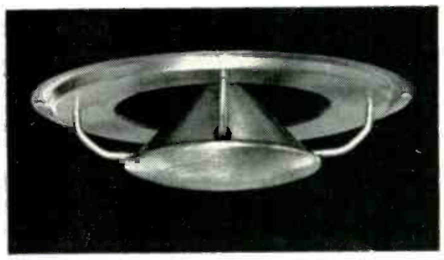


Fig. 62. Dispersion of high frequencies plus a degree of decorative cover-up for the speaker opening are afforded by this Lowell baffle for use with ceiling- or wall-mounted speakers.

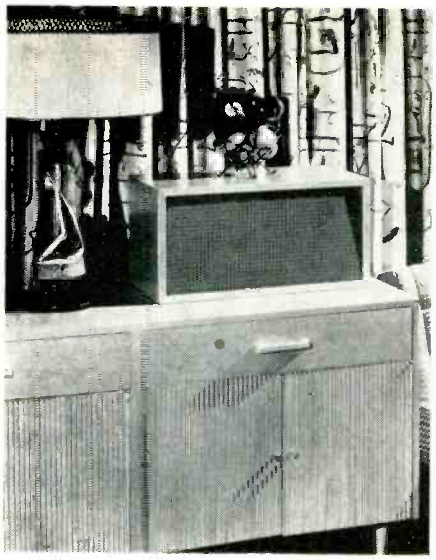


Fig. 63. Permoflux has returned to the hi-fi market with the Fortissimo and this Diminuette.

R-J Audio Products, Inc., 164 Duane Street, New York 13, N. Y. is the seller of the R-J enclosure, the first and most widely publicized of the undersized enclosures. The inside is built to act as a Helmholtz resonator at a frequency somewhat lower than the speaker's resonant frequency to extend the bass, and an area is provided to load the speaker acoustically at resonance so as to flatten the peak and prevent boom. The newest model, the "Wharfedale," shown in *Fig.* 64, is only 11 in. high by 10 in. deep by 23½ in. long, yet claims a frequency response of 50 to 16,000 cps. It utilizes a special 8-in. Wharfedale speaker unit which incorporates an aluminum voice coil and cloth suspension as well as a felt "buffer" between the cone and frame. The front grille front is made of "Expamet"—an expanded gold-anodized aluminum—which is of one piece construction to preclude the possibility of rattle.

Stephens Mfg. Corp., 8538 Warner Drive, Culver City, Calif. Stephens, which uses the Tru-Sonic trademark, used to be in the cinema-speaker field in a big way and still is, but in recent years they have made some excellent home-music-system units, as well as one of the few commercially available output-transformerless amplifiers. Earlier we illustrated one of their big, theatre-type tweeter horns just to impress you. The 617 system of two 15-inch woofers and a big high-frequency unit pictured in *Fig.* 65 is uniquely professional-looking but is used in many homes for quality reasons. The top home unit is the 628 Continental of *Fig.* 66, a rear-horn-loaded enclosure with two 15-inch woofers plus two drivers and higher-frequency horns to make up a three-way system with four speakers. Single-unit speakers of all kinds are made by Stephens, including high-frequency drivers and horns which can be bought separately instead of taking the one horn that comes with the driver. Separate crossover networks are also available, with crossovers from 400 to 5,000 cps.

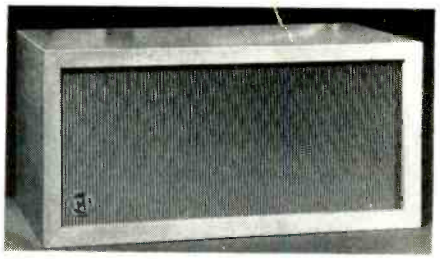


Fig. 64. R-J enclosures came early in the diminutive era. This one is designed for Wharfedales.

Tannoy Ltd., Norwood Road, West Norwood, London S. E. 27, England, is an old-established firm which became famous during the war among seagoing Britishers for the intercoms and ships' p. a. systems they made—so much so that one was instructed to call someone on the Tannoy much as we would say "Kodak" when referring to any camera or "Frigidaire" for any mechanical refrigerator. A principal product is the dual concentric shown in *Fig.* 67, with

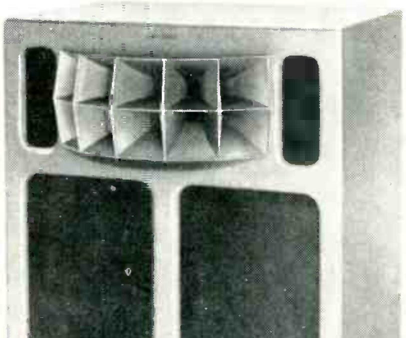


Fig. 65 (left). The Stephens 617 system's appearance shows traces of the company's cinema work. Fig. 66 (right). This is the 628 system, less professional in appearance but more decorative.

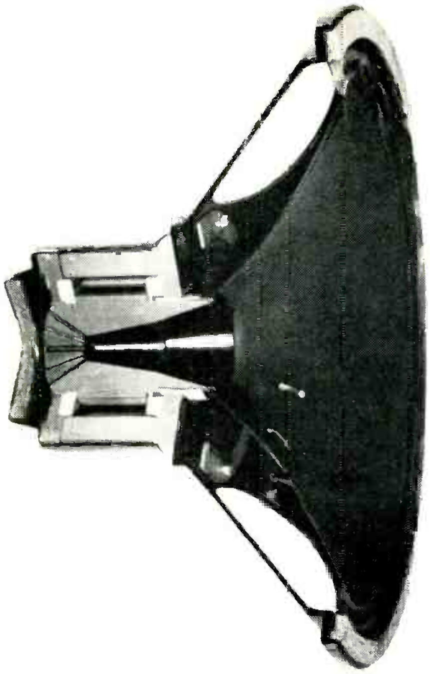


Fig. 67. The Tannoy dual concentric has a single field magnet but two entirely separate voice coils.

a woofer and concentric diaphragm tweeter. Both use the same 2-gap magnet but have separate voice coils. Tannoy also has a group of bass-reflex enclosures designed to go with the concentric speaker, elegantly called the Windsor, Edinburgh, and Balmoral. Tannoy products are brought to this continent by *Tannoy (Canada), Limited, 36 Wellington Street East, Toronto, Ont., Canada.*

Stromberg-Carlson Co., 1225 Clifford Avenue, Rochester 3, N. Y. Stromberg makes a line of speakers including both single and coaxial units. Their top coaxial is the RF-475, *Fig.* 68. The high-frequency horn contains 10 plastic plates, one behind the other, each with many holes punched in it. This is an acoustic lens, as in the James B. Lansing line, acting to disperse the sound. The speaker is used in the RL-486 labyrinth enclosure, *Fig.* 69, though the enclosure is also sold without the speaker. It comes in kit form.

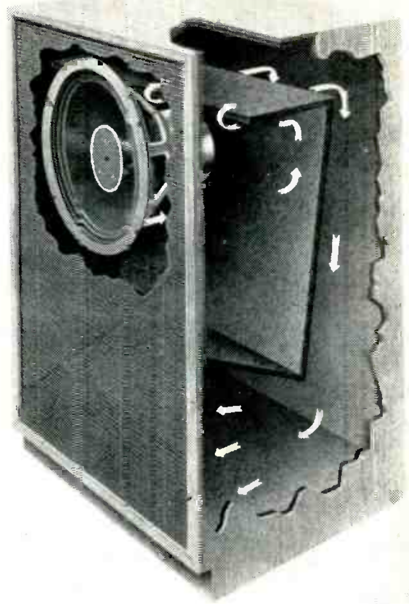


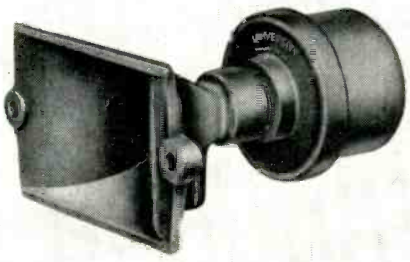
Fig. 68 (above). Best of the Stromberg-Carlson coaxials is the RF-475, which uses an acoustical lens like Lansing's. Fig. 69 (right). The RL-486 labyrinth enclosure comes in kit form with or without speaker.

University Loudspeakers, Inc., 80 S. Kensico Avenue, White Plains, N. Y. One of the earliest low-priced tweeters you could buy separately and add to your old speaker was made by University and still is, though now in several models. Single and dual versions are shown in *Figs.* 70 and 71. University makes a full line of speakers, including both P. A. and high-quality types, including the Diffusicone units of which *Fig.* 72 is typical and which have a high-frequency radiator at the center to disperse treble. University has available suitable enclosures for their speakers, most of which are horn types. The Companion, shown in *Fig.* 73 is a shelf-type miniature enclosure measuring 9⅝ × 24 × 10 inches and containing a combination woofer and midrange speaker and a tweeter. It also has a unique feature—a clock which turns things on and off.

Epilogue

We wish we could say, "That's all there is, there isn't any more!" Unfortunately, in addition to being a boon to music-loving humanity (as well as, in many cases, a thing of beauty and a joy forever) a loudspeaker and its appurtenances are products of the Age of Science. Science is never done trying, and in this case is ready to admit that better (we hope not bigger) speakers will be invented and offered to you in future—to say nothing of perhaps some methods of setting air in motion other than the present concept.

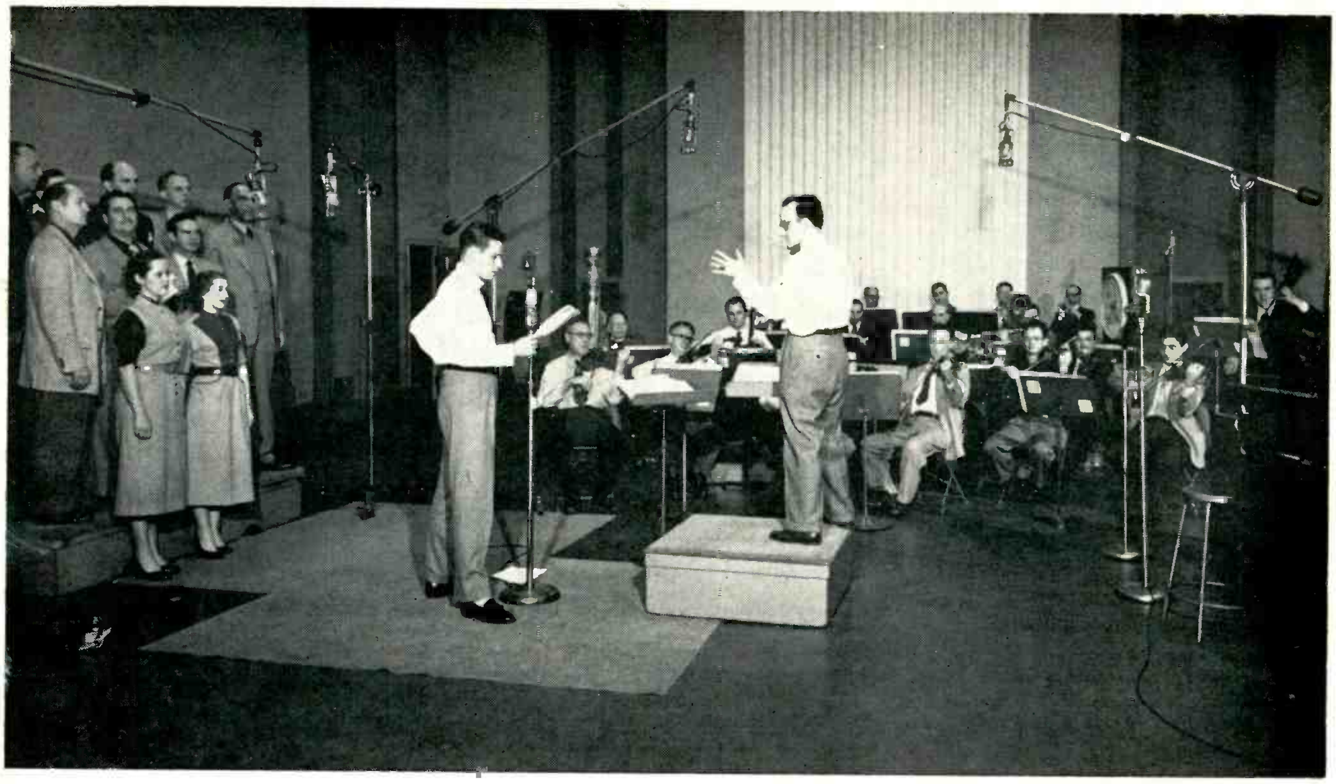
Meanwhile, there is some mighty fine sound awaiting you. While, as we said earlier, there is no real substitute for the procedure of just listening to discover the speaker system best suited to your ears and loving heart, we hope that this section of AUDIO has been of some help as a useful guide to the basics of a subject not entirely simple. You may think you're feeling well, but knowing how to use a thermometer either gives you confirmation or an urge to look yourself over a little more thoroughly. We hope we've supplied the thermometer!



Figs. 70 (top) and 71 (bottom). Single- and dual-driver versions of the famous University tweeters, often added later to existing systems.

Fig. 72. University's Diffusicone has a high-frequency dispersion radiator at center.

Fig. 73. Another miniature enclosure, the Companion, is made by University. It can be separate or mounted in a bookcase. The clock is one of the robot variety.



Actor Charleton Heston with conductor Richard Shores in a dramatic reading on "The Northeners." Heston's microphone is fed to both channels monaurally.

Stereophonic Listener Tests

JAMES CUNNINGHAM* and ROBERT OAKES JORDAN**

Practical experience in the field is the only way to evaluate the techniques and to observe the acceptance of a service which could contribute greatly to the enjoyment of music listening.

STEREOPHONIC SOUND has, in a relatively short time, become a significant factor in almost every medium of mass communication. Motion pictures, radio, disc and magnetic recording, and now television have caught the stereophonic fever. Rapid growth in these media is a certainty, especially the last named, since many more people own television and AM receivers than separate AM and FM receivers. From NBC in Chicago we broadcast two hours a week of regularly scheduled stereophonic shows. One show is a simulcast over WNBQ-TV and WMAQ (AM) with the TV and FM transmitters tied together so those with separate AM and FM receivers can still enjoy the stereophonic sound without television.

When program manager George Heinemann instituted the first regular stereophonic broadcast series in May, 1953, using the NBC Chicago orchestra, there were the usual gloomy predictions by the skeptics. After the first show it became by far the most popular live music show ever presented on the station, and it has remained so.

With all this activity one would expect the technique of stereophonics to be developed somewhat beyond the experimental stage. A perusal of the technical literature reveals a great deal of theoretical speculation but virtually no practical work which would be of help to anyone entering the field. The experimenting which has been done has produced some excellent results though few of them have been documented and are therefore of little help to those caught in a maze of misconceptions. An example of one misconception is the attempt to move a sound across stage by cross fading between channels. The authors feel that additional work is needed to bridge the gap between theory and practice.

One bit of theory that arises from time to time is the use of a third microphone bridged across th. traditional two microphones employed in a stereophonic pickup. In 1934, Bell Telephone Laboratories[1] investigated this quite thoroughly in relation to the localization of a speaking voice on stage. More recently[2] the theory has been advanced that this third or center microphone would enhance the reproduction of music by filling in the "hole" or "dead spot" noticeable between the two speakers. The theory is logical since monaural sound (supplied by this center microphone) is always observed to originate exactly between two properly phased speakers, and a combination of the two systems should fill in the so-called "hole."

The "hole" is not imaginary; reference to the work of Camras[3] shows that blindfolded listeners can locate the exact position of each speaker if the speakers are located in the corners or against one wall of a room.

* *National Broadcasting Company, Merchandise Mart, Chicago 54, Ill.*
** *Robert Oakes Jordan & Associates, 929 Marion Ave., Highland Park, Ill.*

[1] *Electrical Engineering*, Jan. 1934, six articles.

[2] O. C. Bixler, "A commercial binaural recorder," *J. S.M.P.E.*, Aug 1952, 109.

[3] M. Camras, "A stereophonic magnetic recorder," *Proc. I.R.E.*, April 1949, 442.

Joseph Gallicchio conducting the NBC Chicago Orchestra on the TV-FM-AM simulcast.

The authors put this theory to a series of listener preference tests, since in the final analysis it is the listener we are trying to please. Unfortunately, several listener tests of the past[4, 5] smack more of sorcery than of science because of faulty interpretation and too many variables.[6, 7, 8, 9]

Many preliminary tests were made by the authors in order to reduce the variables to a minimum. First, the best position for two microphones had to be found since the center channel will not, of course, increase the stereophonic effect, but merely supply the missing sound at the center. Microphone as well as orchestra placement is determined almost completely by the acoustics of the recording studio or hall, and no attempt will be made to enter into a discussion of this vastly complex subject. Here, in fact, is a wide-open field for researchers desiring to close the breach between theory and practice.

In lieu of this research, the cut and try method seems the only possible course—that is, to make many different stereophonic recordings, each with different microphone and orchestra placement. Assuming proper balance of the orchestra, an absolute reference standard is needed to judge the worth of the stereophonic effect obtained with two microphones.

Comparison Techniques

This can be supplied by A-B switching between monaural sound over two speakers and the usual stereophonic sound over two speakers. When the authors first tried this method a remarkable phenomenon was noticed which set the pattern of the listener preference tests. This phenomenon is that during a passage of music of varying character an A-B switch will at certain times show a vast difference, and at other times show scarcely any difference. Evidently the character of the music itself is a variable in regard to the stereophonic effect. Therefore, before the worth of the recording can be judged an extended passage of music must be subjected to this A-B switching and the total difference mentally averaged. This procedure is not difficult and is an extremely realistic over-all test.

Since this method is impractical to use in group listener preference tests, it was decided to use approximately 30-second samples of music and simply ask the listeners to write down which sample sounded the most realistic to them. Three musical selections were recorded by the NBC Chicago orchestra, the three selections being chosen for their diversity: first, the beginning of the scherzo movement from Dvorak's *Fifth Symphony;* second, a passage from *The Walk to Paradise Garden* by Delius; third, *Plink, Plank, Plunk* by Anderson. The first selection illustrates dynamics, the second a full orchestra sound, and the third pizzicato.

No tricks were used in the tests such as grouping the strings around one microphone and brass around the other. This detracts from the music and makes the listeners look as though they were at a tennis match. The normal antiphonic qualities of the music are enough to give the direction sense. At an actual concert it is hardly necessary to know the exact position of the second flute; direction is of little importance. The most important effect of stereophonics is the transmission of the spatial feeling which in turn transmits the fullness and timbre of the instruments.

Once we felt we had obtained the best results with two microphones the center channel was added, using isolating amplifiers to avoid any cross feed between the outside channels. This center channel was, of course, fed equally to both channels. Using the A-B switching method described, a slight loss in stereophonic effect was noticed when the center channel was 5 db higher than the outside

TABLE 1

Listener Preference Tests

Level of center channel	Selection #1	Selection #2	Selection #3	Averaged Totals
+2 db	30.0%	14.9%	32.1%	25.7%
−1 db	16.5%	34.5%	23.3%	24.7%
−4 db	15.9%	23.2%	19.6%	19.4%
out	23.5%	20.2%	10.7%	18.3%
Filters in, monaural	14.1%	7.2%	14.3%	11.9%

Selection No. 1: Beginning Scherzo Movement, Dvorak *Symphony No. 5* (Illustrates dynamic range)

Selection No. 2: Section *The Walk to Paradise Garden*, Delius (Illustrates full orchestra sound)

Selection No. 3: Section *Plink, Plank, Plunk*, Anderson (Illustrates pizzicato)

[4] H. A. Chinn and P. Eisenberg, "Tonal-range and sound-intensity preferences of broadcast listeners," *Proc. I.R.E.*, Sept. 1945, 571.

[5] N. D. Webster and F. C. McPeak, "Experiments in listening," *Electronics*, April 1947, 90.

[6] S. E. Stuntz, "The effect of sound intensity level on judgment of 'tonal range' and 'volume level'," AUDIO ENGINEERING, June 1951, 17.

[7] H. F. Olsen, "Frequency-range preference for speech and music," *Electronics*, Aug. 1947, 80.

[8] N. M. Haynes, (letter), "Listening Tests," *Electronics*, June 1947, 268.

[9] C. J. LeBel, "Psycho-Acoustical Aspects of Listener Preference Tests," AUDIO ENGINEERING, Aug. 1947, 9.

channels. Thus four recordings were made with the center channel at levels, relative to the outside channels, of +2 −1, −4, and off. Three decibels may seem a small change in level, but the resultant change was noticeable, as the listener tests will show.

There has been so much interest of late in pseudo-stereophonics[10]—one method being to insert a bass boost, treble attenuation filter in one speaker line and the inverse filter in the other speaker line, giving a certain direction sense to an ordinary monaural recording—that the authors decided to put one of these into the listener tests. For this a monaural recording was made and filters switched in at the proper time.

Table I shows the results from a total of fifteen recordings, five of each musical selection. The five were played in a scrambled order, the scrambling being different for each of the three selections. The purpose of this was to avoid the possibility of any of the five being in a favored position. The averaged totals, which also represent an average symphonic selection in regard to diversity of orchestral color, show that the listeners prefer the higher settings of the center channel. A cautious conclusion drawn from these results would be that the center channel is certainly worthy of a trial in any two-channel stereophonic set-up where an orchestra of perhaps ten or more musicians is involved.

The listeners chosen for these preference tests were primarily music lovers, musicians, and music students. Through more discriminating than a cross-section of an average audience, they were chosen because the audiences for the stereophonic programs on WMAQ are without a doubt made up of music lovers whose judgment would more likely agree with that of this selected group. There is ample evidence in the tremendous mail response to these programs to the fact that what would please an "average" audience would not please them. A total of 177 persons participated in the tests at various times, and the same preparation was given at each test.

The authors are greatly indebted for the suggestions of Ralph Knowles, director of stereophonic programs at WMAQ, without whose help these experiments would have been impossible; to Stanley Stoltz, assistant director of Robert Oakes Jordan and Associates; and to Joseph Gallicchio and the NBC Chicago Orchestra for their brilliant performance of some difficult repetition work.

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R. H. Snyder, "History and development of stereophonic recording," *J. Aud. Eng. Soc.*, 1.2; April, 1953.

E. Cook, "Recording binaural sound on discs," *Tele-Tech;* Nov. 1952.

[10] P. C. Goldmark, "The Columbia 360," AUDIO ENGINEERING, March 1953, 28.

The Northwestern University Orchestra showing experimental microphone placement.

Author Cunningham at the specially built three-channel amplifier for Magnecord Tape Mechanism.

COMING EVENTS

Aug. 25–27—Western Electronic Show and Convention. Ambassador Hotel, Los Angeles, Calif.

Aug 27–29.—Dixie Audio Festival, Henry Grady Hotel, Atlanta, Ga. Open to public 28–29, 1:00 p.m. to 10:00 p.m.

Sept. 30, Oct. 1–2—1954 High-Fidelity Show, International Sight and Sound Exposition. Palmer House, Chicago, Ill.

Oct. 4–6—National Electronics Conference, Hotel Sherman, Chicago. Papers are solicited on all electronics subjects, and the program chairman would appreciate sugestions for titles and authors of suitable papers. Write George E. Anner, Elec. Engrg. Dept., University of Illinois, Urbana, Ill.

Oct. 13–17—1954 Annual Convention, Audio Engineering Society. Hotel New Yorker, New York City.

Oct. 14–17—The Audio Fair, Hotel New Yorker, New York City.

Nov. 18–19—Sixth Annual Electronics Conference sponsored by the Kansas City Section of the I.R.E., Hotel President, Kansas City, Mo.

Equipment Report

Bogen DB15G Amplifier and R640 AM-FM Tuner Miracord "Magic Wand" Record Changer

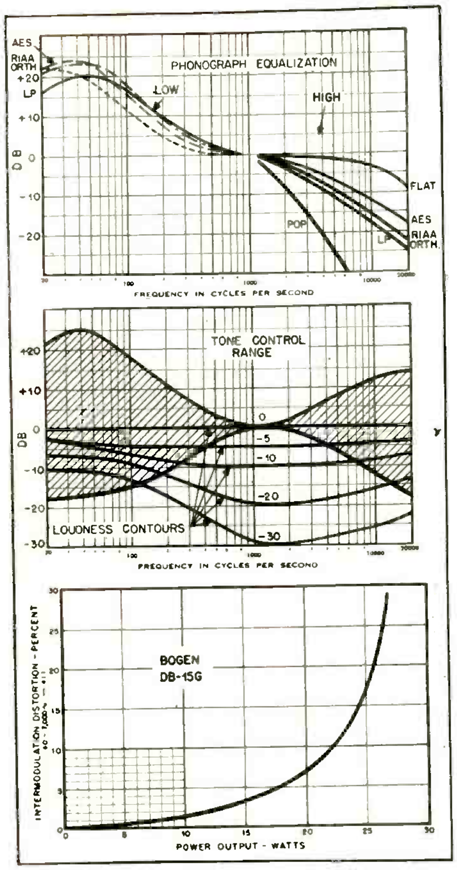

Fig. 1. Performance curves for the Bogen DB15 amplifier.

Two of the latest additions to the Bogen line of high fidelity equipment are in themselves all of the electronic equipment required for a complete home music system—add a speaker and enclosure and you can play radio programs with comfort and fidelity, and add a record changer and pickup and you have recorded music when you want it.

While the DB15G (in a metal cabinet, but available uncased as DB15) is small in size, its performance and flexibility put it in a class with many more expensive amplifiers. Among its attractive features is provision for feeding inputs from high- or low-output magnetic pickups, tape recorder, tuner, *and* microphone, and for feeding to a tape recorder an output which is not affected by either tone or volume controls. In addition, the compensation available for phono pickups is so arranged that both low- and high-frequency characteristics are separately controllable, with one knob on the panel marked to provide LP, AES, ORTHO-RIAA, and EUROPEAN low end, while another provides the LP-NAB, AES, ORTHO-RIAA, POP, and FLAT high end. This gives considerable flexibility to the user, and enables the most particular record enthusiast to obtain the curve he thinks is best. The upper section of *Fig.* 1 shows the available characteristics.

The tone controls used are of the increasingly popular Baxendall type, and provide low-and high-frequency boosts of 25 and 14 db respectively, and cuts of approximately 17 db at both ends of the audible spectrum. The Loudness Contour selector, available as an accessory to the Bogen R701 tuner described in these pages in September, 1953, is built in to this amplifier and permits the listener to adjust the loudness compensation to suit his average listening level with no further need to equalize input levels from various sources, or even from different phonograph records. This arrangement seems to offer one solution to the continuing controversy between loudness control backers and those who wouldn't use one under any circumstances. Tone control and loudness contours are shown in the second section of *Fig.* 1.

The amplifier is built in a singly mounted unit, although it is actually composed of two separate sections. Because of this construction, it would be easy to service, should the need arise. The output stage uses a tertiary winding on the transformer for the cathode returns of the 6L6GA output tubes, and the IM distortion is held satisfactorily low, with a curve more similar to that obtained from triodes than that normally exhibited by tetrodes with feedback.

Because of its small size—the amplifier

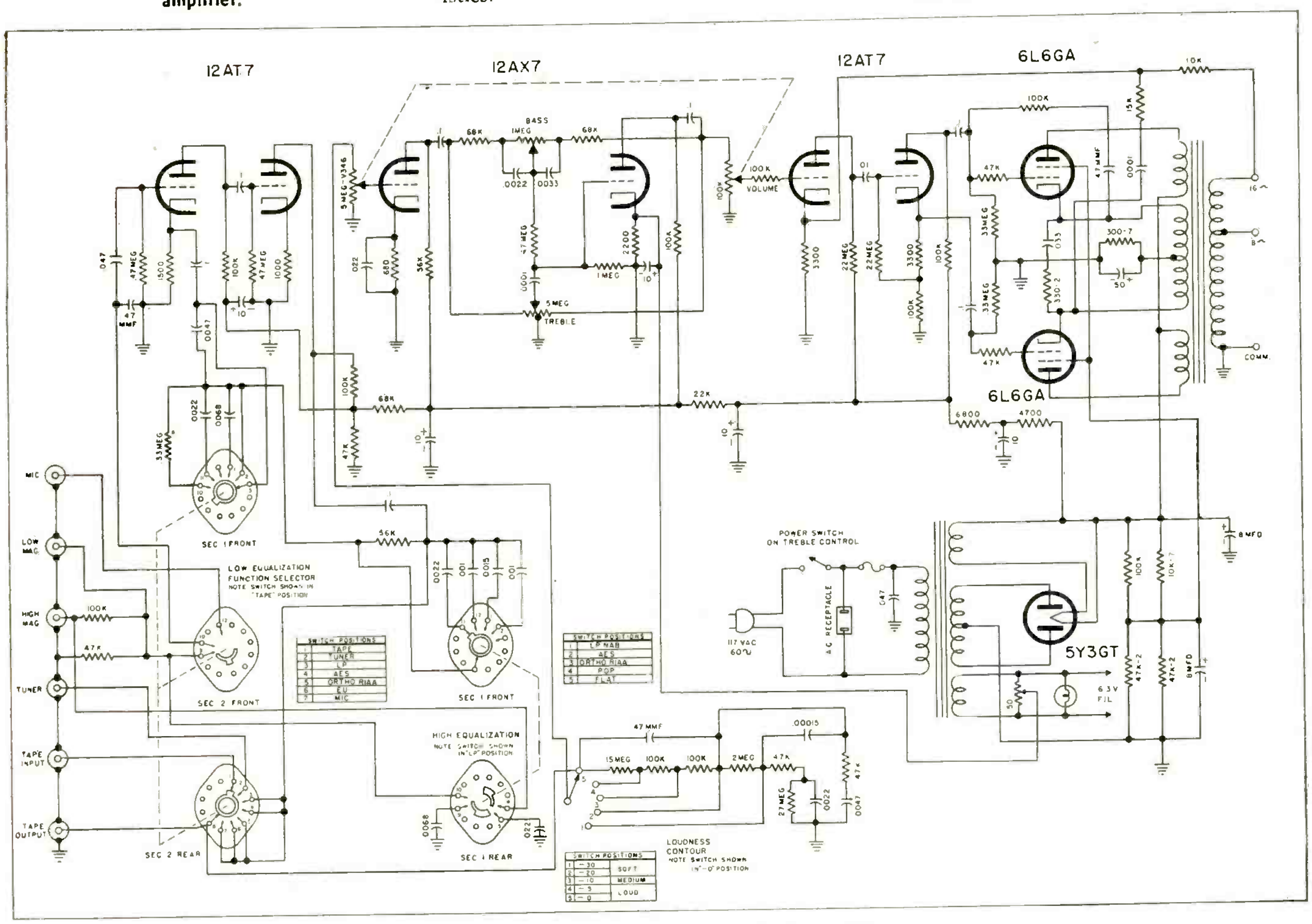

Fig. 2. Complete schematic for the DB15 amplifier.

Fig. 3. The Bogen DB15 amplifier. Functions of the two center controls are different on production run from this photo—third knob from left is selector and low-end equalization and fourth from left is high-end equalization, using a 5-position switch.

proper is only 4⅝ in. high—both units can be mounted in a space only 12 in. high. This makes for a relatively small over-all equipment installation.

With the volume control off, the hum and noise (essentially of the output stage) is 74 db below 1-watt output; with volume control at maximum, hum and noise is 60 db below 1 watt at the tuner and tape jacks, and 33 db below 1 watt at HI-MAG jack. It is improbable, however, that the volume control would be at the maximum position at any time. Sensitivity at the various inputs is, for 1-watt output: TUNER and TAPE, .083 v; LOW MAG, 1.75 mv; HI MAG, 6.6 mv; and for the MICrophone input, 2.6 mv. Voltage available at the TAPE OUTPUT jack is .083 v. for 1-watt output, which is the same as tuner and tape inputs. The amplifier is fused, and a switch-controlled a.c. receptacle for other items of equipment is provided on the rear.

R640 Tuner

The AM-FM tuner is similar in size to the DB15 amplifier, and has only two controls—one combining the power switch with the AM-FM selector and the AFC defeat switch, and the other a tuning control. The defeat switch is a convenience because it allows the AFC to be cut off momentarily while you are tuning in the station, and when you release the knob the AFC comes into action again. Drift—with AFC on—is negligible; after playing for 12 hours, a weak station was tuned in sharply with AFC defeated and the set was turned off. After 12 hours, the set was turned on again, and the same station was still tuned in accurately.

The R640 and the DB15 are especially suitable for installation where space is limited, but performance is certainly adequate for normal-sized systems.

Fig. 4. The Bogen R640 AM-FM tuner.

MIRACORD "MAGIC WAND" RECORD CHANGER

Newly imported German-made changer exhibits novel features which result in a unit that performs as well as advance publicity claimed

IT IS PROBABLE that the designer of a record changer considers that amplifiers are "Rube Goldbergs" compared to his simple products, but most of us who claim to understand amplifiers are amazed by almost any record changer. The amazing new Miracord has a number of features which are new to the changer field, but which result in an interesting and efficient unit.

Foremost of the features is the "Magic Wand"—a device which handles 78's and LP's in the same manner as the 45's are handled, from the center hole. And it works. This device, which readily lifts out from the hole in the center of the turntable, takes a stack of ten to twelve ordinary records, holds them carefully on its small spindle, and drops them as required one at a time; and when there are no more to drop, it turns off the motor. The Wand is actuated from below, and is held in place only during a change cycle; any other time it can be removed to facilitate taking a played stack off the turntable. It takes 10- or 12-in. records, intermixed as desired, and plays them all properly—the one requirement being that all are of the same speed and stylus type.

If, however, you should not want it to operate as a changer, you simply remove the Magic Wand and replace it with a short spindle, and the changer loses most of its automatic qualities and functions almost like a single-play turntable in that you can move the pickup arm at will, although it still stops at the completion of the record.

When you want to repeat a record a number of times (this applies only to 10-in. discs) you turn the short spindle over and replace it in the hole; then the device plays the record over and over as long as you leave it turned on.

For 45's, a large center spindle works in the conventional manner, the arm dropping at the 7-in. diameter so long as the speed knob is set to 45 r.p.m.

Push-Button Controls

The operation is controlled by four push buttons—START, REPEAT, FILTER, and PAUSE. The START button commences the operations once you have placed the records on the spindle and set the speed control knob to the correct r.p.m. The REPEAT button inaugurates a change cycle without causing the record to change. The FILTER button places a load resistor across the pickup to lower high-frequency response. But the PAUSE button—that's another story. As you depress this button, the number in a small hole back of the buttons changes. In the "O" position, the changer takes from 5 to 12 seconds to go through a change cycle—5 seconds for 78's, 12 seconds for LP's. When the number is "4" for example, the interval between selections is 328 seconds on LP's, 140 on 78's. Thus you can adjust the interval between records to as much as 5½ minutes with LP's, a convenience when listening to background music.

The changer is quite compact, and is mounted by means of soft rubber buffers, being held in place by a spring clip which can be removed readily with the fingers. The turntable runs on a ball suspension, and rumble was measured at less than 3 db higher than a comparison single-play unit. No wow was detectable on 78-r.p.m. piano records played at 33 1/3 r.p.m.—a test which appears to be quite indicative of any wow in a turntable.

The pickup is carried in a plug-in head, and needle force can be adjusted by means of a knurled nut. Pickup output is shorted during the change cycle. Needle force on the bottom record measured within 1 gram of the needle force on the top of a stack of ten 10-in. records. The connections are so arranged that a shielded pair can be used to connect to the pickup, if desired, to reduce hum pickup in the leads over that from a shielded single wire.

The changer is attractively finished, with maroon enamel and white push buttons and speed control knob, and the molded white rubber turntable cover which has round projections on its underside to engage in holes in the platter, and molded notches on the top to engage the large 45-r.p.m. spindle, as well as concentric grooving to hold the record without any slippage.

The four-pole motor is mounted well away from the pickup location, and is rubber mounted to minimize rumble. It is started and stopped by a switch in the pickup arm rest, and a spring clip holds the arm against the rest to protect the pickup when the unit is being moved.

SOUND DIFFUSION

(from page 19)

3. *Line irregularity.* The steady-state sound pressure irregularity is characterized by three factors: (a) the level difference between crests and troughs, (b) the number of crests and troughs in a given frequency band, and (c) the sharpness of the peaks.

The frequency irregularity, as defined above, does not take into consideration the width of the peaks. Somerville and Ward[3], introduced the term "line irregularity" in connection with a small model of a room which they used for their tests. The factor was obtained by measuring the total length of the record trace in a given frequency interval. As the minimum length of trace is the horizontal distance between the two extreme frequencies defining the frequency interval along the frequency axis of the graphic level recorder paper, this distance was subtracted from the length of the total trace to produce the quantity "line irregularity," which has a value of zero when there is no irregularity. Its units are db per cycle, the same as those of frequency irregularity. Note that the quantity does not recognize directly and explicitly the number of peaks in the steady-state sound pressure irregularity.

4. *Operating diffusion.* In order to include the mean crest height in the equation of frequency irregularity, Furrer and Lauer[4] divided the equation of the frequency irregularity by the number of crests in the measured frequency interval. They applied, vaguely, the term "operating diffusion" (Betrieb's diffusion) to this term. It is by given by

$$D = \frac{\Sigma P - \Sigma p}{n \Delta f}$$

where again P = pressure maxmimum, in db,
p = pressure minimum, in db,
n = number of crests, and
Δf = frequency interval in cps.

Evaluation of the numerator was done in the manner suggested by Bolt and Roop, in that a certain manner of averaging of the basic measurements was carried out in four 25-cps bands in the frequency regions of 375 and 1650 cps. Tests were made at six different positions in every room to achieve a good mean. They learned that for good "diffusion" in a room, the quantity obtained in the calculations should be less than a certain value for a room of given volume. This value decreases with increasing volume of enclosure, as *Fig.* 2 shows, and is approximately half as large for the 1650-cycle region as it is for the 375-cycle region.

The following tabulation gives a summary of the results obtained by the various investigations noted above.

Transmission irregularity decreases with increasing absorption in a room.

A diffuse room will show little if any major frequency-irregularity peaks when the results are plotted so that the data is normalized and uses a single set of generalized parameters.

The presence of diffusing elements on a plane wall of an enclosure reduces the line irregularity of the steady-state frequency characteristic. Diffusing elements produce some effect when the depth of the elements is of the order of one-seventh of the wave length.

Of the wall contours studied, rectangles produce maximum reduction in steady-state irregularity. More cylindrical elements are required to produce a reduction comparable with that given by rectangles. Diffusing elements reduce the irregularity of short-pulse decays and in general, reduce the space variation of the character of the pulse decay.

Of the elements studied, rectangles produce the greatest reduction in pulse irregularity. Walls covered with cylindrical and triangular shapes are approximately equivalent in their effect on both steady-state and transient phenomena. In a diffuse acoustically satisfactory room the value of diffusion should be below a certain value for a room of given size. The value decreases with increasing volume of enclosure, and is approximately half as large for the 1650-cps region as it is for the 375-cps region.

It is not clear to what extent steady-state sound transmission measurements provide a reliable index of the amount of diffusion in a room. It has been noted that the introduction of cylindrical elements has little effect on the sound-transmission measurements; yet their acoustical excellence is well established. In particular, it has been noted that such wall contours make for a smoother sound-decay curve, and that they have even been installed in reverberation chambers used for absorption measurements. Hence we may say that sound diffusion in a room has its greatest effect on transient sounds and may be determined, in a measure, from the character of the sound-decay curve. Somerville[5] has considered this condition, and used it as one of the factors for his "empirical acoustic criterion." He did this by determining the ratio of S/OB, where S is the shaded area on both sides of the mean slope of the sound decay curve shown in *Fig.* 3, and OB is the length of the line on the time axis on which the decay is measured For an absolutely straight decay curve, the ratio obviously will be zero because the area will be zero. He determined this ratio for a relatively large number of frequencies, then formed the mean of the ratios, and added it to some other quantity to arrive at a single-number index of the acoustic quality of an enclosure. In this manner, however, the specific is mingled with the general, and it becomes difficult to arrive at some figure of merit for an acoustic condition such as diffusion.

To clarify the concept, this writer has used a velocity microphone and has taken sound decay curves along the three orthogonal axes at a point in the room. These curves were not the same, and when evaluated somewhat in the manner shown by Somerville may provide a more critical measure of diffusion. However, considerably more work is in order before definite conclusions can be formed.

[3] T. Somerville and F. L. Ward, "Investigation of Sound Diffusion in Rooms by Means of a Model," *Akustica*, V. 1, No. 1, 1951, p. 40.

[4] W. Furrer and A. Lauer, "Die Diffusion in der Raumakustik," *Akustica*, V. 2, No. 6, 1952, p. 251.

[5] T. Somerville, "An Empirical Acoustic Criterion," *Akustica*, V. 3, No. 6, 1953, p. 365.

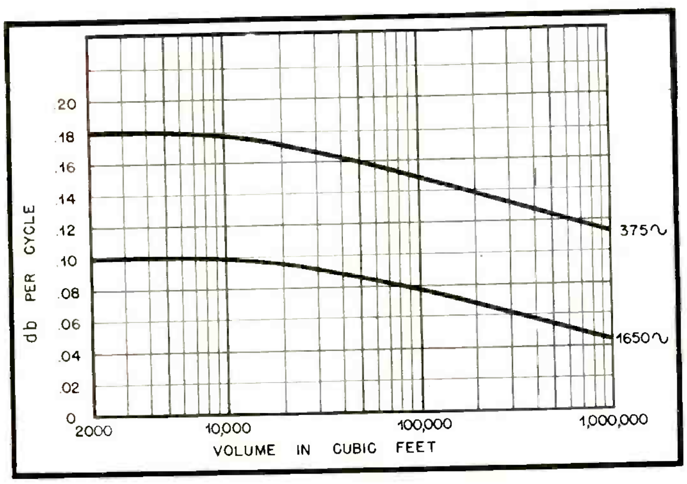

Fig. 2. "Operating diffusion" curves taken at two frequencies in a typical room.

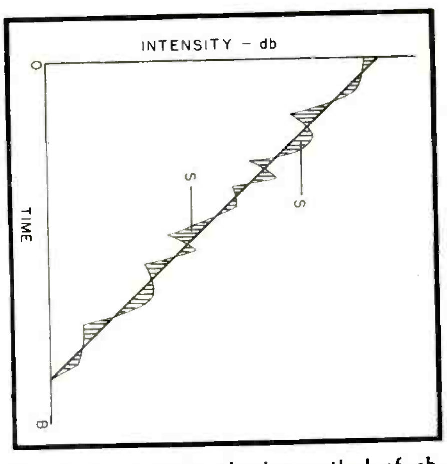

Fig. 3. Decay curve, showing method of obtaining "empirical acoustic criterion" by determining ratio of response above and below mean slope.

AUDIO ETC.

Edward Tatnall Canby

Tips and Tapes

A SHORT PLUG is in order for a short book just out, by my friend and competitor ("High Fidelity Simplified") Harold D. Weiler, called "The Wear and Care of Records and Styli" (Climax Publ. Co.). To begin with, anyone who can come up with a title like that must be pretty hot, and any pamphlet of the paper-bound type that sells, as this does, for one dollar and still is both useful, concise, orginal and comprehensive is really something. Finally, when you know that Mr. Weiler is one of the most persuasive super-salesmen in the greater-audio field, coming up with manufactured products every other minute—from the Nova Music Wall to K-33 record shampoo—you will be the more amazed at the objectivity and enthusiasm of this study of stylus wear.

The first section of the booklet, a sort of preparatory section, is an excellent account of the record itself and its manufacture, the nature of grooves, tracking and so on, replete with pictures; this continues into a nicely organized discussion of the forces at work, the development of "flats" on the stylus, the effect of dust on the record, what happens as a stylus wears away. But Weiler is a persistent so-and-so—and this is no rewrite job. His hobby, I would guess, is testing, of the more diabolically ingenious sort, and the rest of this booklet offers the fruits of some of the darndest testing I've run into outside of a full-fledged research lab.

How long should a stylus last? You'll find plenty of fine estimates I've made 'em myself, in carefully guarded words, leaving me a safe out. I gave a shrewd (I hope) guess on the basis of my experience and much second-hand information. Probably useful. But Weiler, instead, set out to make a Survey in two parts. First, a questionnaire to some 3000 people, asking how long they used their needles (styli) and how old was the one now in their machine? Much variation, but it averaged out four months. The Great American Needle, then, was four months old and a sapphire. Next, separately as I get it, Weiler investigated the hours per day and days per month of phonograph use. Again, averaging, the answer was that Mr. Average Hi-Fi American uses his phonograph around 18 hours a month. Half-hour a day, more or less. Quite reasonable.

Put the two together, for what it's worth, and you discover that the average sapphire needle is *already 72 hours old,* in terms of use. Four months at 18 hours a month. Now, admittedly, this is a sort of Dr. Kinsey-Gallup Poll approach, but read on: nobody knows *just* how long a sapphire should last and Weiler lists the complicating factors as every serious writer must; however, as he says, the various "experts" in the field judge that from *ten to thirty* hours is enough for any sapphire.

According to the Weiler Survey, then, the *average stylus now in use has already played from two to seven times as long as safety allows.*

Now I haven't run any organized test or survey, but I've run into a lot of people and their players these last years and I can only suggest that Weiler is probably righter than Kinsey. After all, many people still think that a "precious metal" point is permanent; they were sold as permanent (and even permanently built-in) for several years after the war. Again and again I have heard home machines whose needles were so obviously worn to a frazzle that my hair stood on end, though the owners of the machine hadn't yet begun to feel uneasy. I agree entirely. Taking all phonographs together, hi-fi, "hi-fi," and nondescript, we may assume that the average stylus is hopelessly worn out, *right now,* everywhere. Just as the average auto has about fifty vital things wrong with it and the average person hasn't been to his dentist for more months than he dares remember.

No, you can't prove this sort of thing exactly—and that is just where an intelligently managed survey or an opinion-poll or test sampling can be most useful. When well calculated, with eyes and ears open, so to speak, with a very careful sense of *all* the values involved, psychological as well as scientific, a survey can do a lot to clarify what is variously obvious to a lot of people, though undocumented. And this —to get back to the subject—is the sort of thing that goes on in the rest of Mr. Weiler's book, too.

For he didn't stop here. That devilish light in the Weiler eye, the gleam of high strategy, must have been positively atomic as he thought, *why not collect old needles?* So the man did. Picked up some 600 worn styli that were left in "Needle Graveyards" in record stores; and more than 300 were sapphires. How to determine their age? Mr. Weiler, armed with his super-binocular microscope, proceeded to wear new needles down until he had a time scale which he could apply to the Graveyard corpses. (Yep, he worked out a special microscope positioner and scale to measure by eye to a ten-thousandth of an inch. Leave nothing to chance!) . . . but I don't intend to tell all of this story.

Suffice it to say that Mr. Weiler went on to playing records, which, as you can now imagine, meant playing hundreds and hundreds of them hundreds and hundreds of times. Nothing was left unthought of. Three standard types of cartridge, a judicious mixture of brand new and somewhat used records (since new records alone would have distorted his test results) and a quantity of sheer patience that is diabolical. A changer and a manual player, just to be sure. Three brands of sapphire styli. Charts kept of everything, but everything, so that even the difference in compliance between makes of cartridge could be accounted for. The tests were done over and over until a safe average had apparently been achieved; then a new set of tests was done backwards—that is, new styli were used for a fixed number of hours to see whether they would wear according to the expectations of the earlier charts.

I gather that Mr. Weiler's chief assistant, Mrs. W., spent most of her time for a year tending record players, hour after hour. They didn't play out loud; but that faint, tantalizing hint of music from the stylus needle talk would have driven me quite nuts in a week I'm sure. Not she, nor he.

And so I leave you to read Weiler for yourself and find his conclusions. I'll only sneak in one. By far the biggest cause for stylus wear is airborne dust and dust ground from the stylus itself; clean, static-free records may last *60 per cent longer,* as an average, than uncleaned discs with static charge on them during playing. That is from a Weiler test, but I'm delighted to find it confirms the Canby hunch, as of our June issue.

You'll find "The Wear and Care of Records and Styli" quite breathtakingly interesting, what with all these diabolical ingenuities, and I heartily recommend it as some sort of Last Word.

I gather that Weiler is now tossing off a booklet on tape recording in the home and if it measures up to "Wear and Care" in reader interest it oughta be something. Last I heard he was off on a little deal that involved a jaunt into the Florida swamps to record alligators. Or was it crocodiles? We'll hear about that next year, no doubt.

Pre-recorded Tape

Funny thing the way an evaluative writer, if I may coin a phrase for myself, must change his tune to fit his audience. I've been doing some preliminary looking into the imminent new boom in pre-recorded magnetic tape that seems ready to bust out big, and, aside from having to remark immediately that this little subject will be found neatly covered in AUDIO ENGINEERING, August 1951, in this department (Record Revue, then) and that a large part of what appeared common sense in that article still applies now—I do feel that a boom in pre-recorded tape may have important advantages for many readers of this magazine, whereas the same boom will be a snare and a delusion to plenty of readers of a less audiophile frame of mind.

It all depends on who you are and what you like. For some of us, the advantages of taped music in technical quality are supremely important; we'll do anything to have the stuff, and have been doing it, as far as is possible, these last years. I know several two-Ampex people and plenty of Magnecorder owners and Concertone enthusiasts (not even mentioning the possessors of less expensive machines) who have been taking vast mileages of tapes off the air for years. Many of them tote their machines to as many "live" concerts as the union will allow, too. I heard some fantastic hi-fi tapes last year in St. Louis and the fellow who made 'em managed to get himself and his carload of equipment (including Stevens mikes) into just about every public event I attended, concert or otherwise. What energy and persistence! That man wouldn't listen to a commercial record on a bet.

Back in the disc days, even, there were

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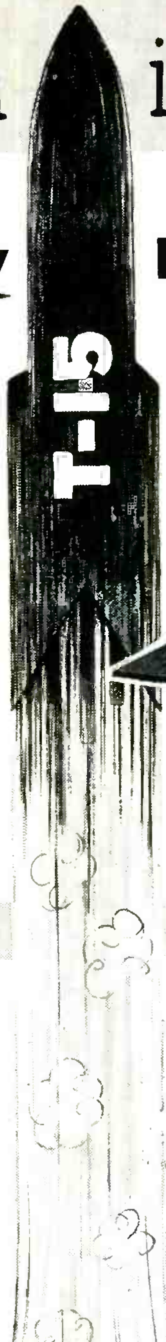

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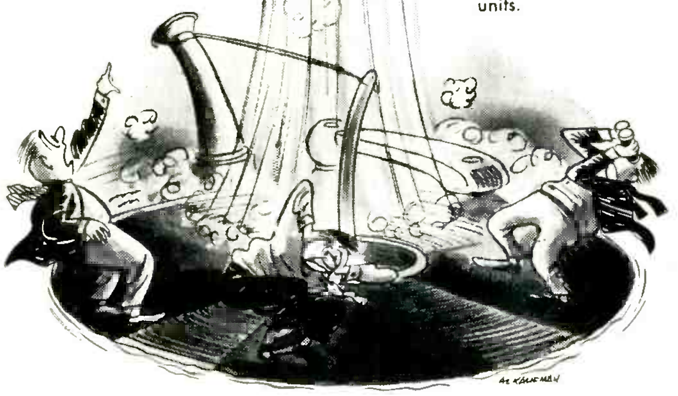

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fabulous fanatics, like H. Vose Greenough, who has (had) 16-inch transcriptions, 33 rpm., of just about every Boston concert of importance for a decade or more before tape was ever heard of. An incredible collection and I trust that some day it will be given to a permanent library—copied onto tape, of course. I bet a nickel you couldn't tell most of those pre-war discs right now from a brand new professional tape.

But with the best will in the world very few of us can afford to hire a symphony orchestra, let alone a string quartet. FM air tapes aside, there hasn't been any very good way for us to get real music on tape, to compete musically with the perfectly enormous selection available on disc records. For people who (a) want top quality sound, but also (b) want it to mean something, want it to be good as to content as well as quality, pre-recorded tape is important. I'm all for it, though I'll immediately say that I am not one of these people—yet.

In my own personal balance, the extra performance of a taped recording as compared to its LP commercial equivalent, while very evident to my ears on an AB basis (and I *should* be able to tell the difference, after all by this time) is just plain not of any real importance in view of the advantages inherent in present-day disc records. The LP catalogue, first—with a hundred thousand times the choice available on tape—so far. The ease of operation, second, especially the ease with which one picks out inner passages or movements on a disc, the quickness of starting and stopping and removing, the absence of rewind, threading, and so on. Yes, I've even copied some valuable older records on tape, splicing the record breaks and everything, but I don't play them often. Too much trouble.

Moreover, I have a very accurate idea of the degree of tonal advantage to be expected in good pre-recorded tape as compared with the same on equivalent good LP. I am aware as are many AUDIO readers, of the cost that this precise difference means, first in the tape itself, and, second, in terms of good playing equipment that can equal fine LP record equipment. It's not going to be cheap—nowhere near as cheap as LP. Most pre-recorded tapes, at 7½ inches (with potentially good quality, wide range) will cost from two to three or four times as much as the same music on disc.[1]

To our readers who can judge these factors in the balance, along with the catalogue of available tapes as compared to LP availabilities, I heartily recommend pre-recorded tape as a fine hobby-to-come and a new line of development for the home music system. The best of it will be sensationally good. (The worst, natch, will be a bit better then punk, as in every field. . . .) There is an undeniable "tape sound" unattainable on any disc, which we can credit directly, I think, to the complete absence of mechanical wave motion and the consequent distortions due to mass, inertia, compliance problems, transient difficulties, etc. in the stylus, that are a double-jeopardy in disc records, once in the cutting of the master and once again in the playing. Granted, the best LP (microgroove 78) recordings, played on top quality equipment, reduce this mechanical distortion to very low points. But in tape it does not exist at all, and that is an irreducible difference that makes tape automatically the preferred medium for the ardent audiophile. Knowing all this, balancing the factors of equipment, convenience, expense and repertory, a well-versed hi-fi man can benefit immeasurably from pre-recorded tape, if he is so inclined in his personal make-up, and this for "serious" music, music to be listened to directly in the foreground, a kind of music that has been notably lacking on tape until now.

Background

Background tape? Well, of course—there's nothing like it and most tape libraries, so far, have been essentially of this sort. Even the classical selections have been pretty much the conventional stuff, strung together in "programs" that offend the dignity of any music lover who has the guts to know what he wants and when. If you want background music on tape you can get it in anything from a little roll of medium-fi double-track 3.75-ips tape up to the huge continuous play commercial installations that serve up muted—very muted—hi-fi in airport terminals and restaurants and what-not. In this field, in spite of the Seeburg Selectomatic disc changer, tape reigns supreme, or will except for the old (and high quality) disc transcription libraries which, I gather, are still convenient and useful for radio stations.

Frankly, most of this background stuff gives me the shivers. Personal reaction, but worth mentioning as a warning to those who may have ideas of grandeur when it comes to taped music. Give me any sort of jazz, blues, folk music, any day, in preference to any background music on earth. There's a certain sound—in any kind of recorded music, even Bach and Mozart—that says *"background"* just as clearly as an announcer's tone of voice says *"commercial."* There isn't much doubt that commercials on the air are a big success and it's an equally sure thing that taped background music is what a lot of people want. But not this cookie. So, if you happen to be anti-background, choose your pre-recorded tape carefully, especially when it is classical in nature.

Foreground—The Dual Release

But background music, though it makes up the bulk of present tape (if we include a pretty wide variety of semi-pops, vocal music, light classicals, chime music and so on in that category), is not the big news of the presently expected expansion. Instead, it is the dual release of recordings in two commercial forms, on disc and on tape. Simultaneous release, we might call it.

Several ventures into tape release of material originally taped for disc use have been announced, though not much of a dent has been made until now in the larger record business. A-V libraries has the edge on others with a quite extensive listing of classical tapes which, as far as I can make out, are all to be found in the Remington LP catalogue. An A-V-Remington tie-up. I did not see or hear any of the Magnecord-Vox tapes announced a year or so ago, nor have a rumored set of Westminster recordings on Concertone-sponsored tape cropped up yet. But RCA Victor's entrance into this field within its own organization makes the trend official and it would seem highly likely that most major record companies will be out with their own limited "reissue" tapes, from the vast library of master tapes now existing. If not their own, then in connection with some other tape firm, as in the case of A-V.

These dual-issue recordings, as things now stand, are bound to set a far higher musical standard than the original tape-only issues, and for the simplest of economic reasons: the recordings—and the recording companies, complete with handling facilities, distribution, and, of course, artist contracts signed and sealed—are already there. No new overhead, a minimum of outlay, and the sale of the disc form of the recordings will cover the expense of really good artistry, allowing the tape venture to seek its own level and develop as it can. What more could be asked for?

We can therefore deduce a number of likelihoods. The disc-company tape offerings will at first be on a very limited scale—because it is quite economically possible to do so, under the dual arrangement. But, for the same reasons, the material will be top ranking, from the musical and performance angle. The companies can afford to offer their best. The RCA opening catalogue follows these precepts exactly. A relatively tiny list of Red Seal and Blue Bird recordings that are also on LP, but the artists include Toscanini. That list will grow exactly as fast as demand develops, and, remember, if conditions warranted, RCA could rush hundreds and hundreds of tapes on the market in a matter of months. So could most other firms, big and little.

BUT they won't. Not the way things are starting. Remember the ball point pen, at $15? One begins with a high price and small production! RCA's tapes cost well over twice as much as the same music on RCA disc. A-V tapes are in proportion, though the Remington discs are lower in price to begin with (but with generally a greater loss in quality from master tape to disc than in the standard-line LP's). I'll bet another nickel that it'll be quite awhile before pre-recorded tape prices come down significantly. Nobody wants to price himself and his discs out of the market. As long as discs hold their own as the mass medium for music, they'll be the cheaper form to buy. Tapes will be held above the disc price, even artificially, as long as there is a need to "protect" the disc market.

And, for that matter, it's not easy to see how tapes could ever entirely replace discs. Too clumsy, unwieldy, inconvenient, and—to go out on a limb—no cartridge or self-threading or self-rewinding device that I can imagine will replace the quick and convenient disc playing system. Tapes, then, will supplement discs for a good time to come, as a kind of de luxe recording, trading disc's lower cost and ease of handling for the advantages of higher sound quality. That is as it should be—and only a dope will throw out his turntable and diamond points and rush for the nearest tape player.

It is for this reason that in another magazine this very month I've written what will seem a rather pessimistic article on this same subject. The audience in that case is a wide and general one, most of whom are only vaguely aware of the technicalities of hi-fi and audio. That audience, unlike the present readership of AUDIO, is the one which must be warned of the pitfalls in the coming flood of joyous advertising—which will, needless to say, be of the sort which launched nation-wide "hi-fi" last year! And so I wrote pessimistically and played down pre-recorded tape, where in this article, I trust, all audiofans will understand that I am fully aware of pre-recorded tape's basic and unequivocal superiority to any disc in quality. Indeed, it will be quite exciting to have the major record companies issuing taped material and we can look forward to some really sensational new music-and-sound combinations.

Equalization and Curves?

I can't stop without voicing one very big IF. All will be well IF the tape engineers can get together on some sort of uniform tape equalization. Right now, this factor is far more seriously chaotic than it ever was with discs. Nobody gives a hoot about anybody else's tape curves (recording and playback) and the differences between the major outfits are fantastic.

(Continued on page 71)

[1] Exception: tape reissues of some older tape masters.

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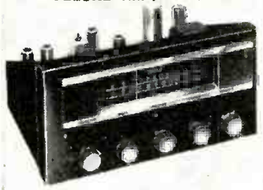

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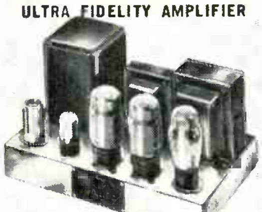

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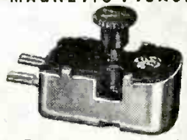

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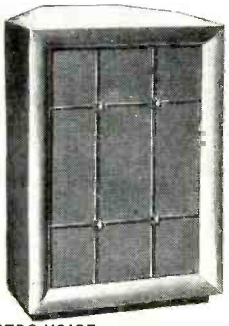

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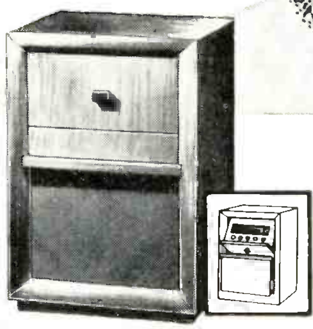

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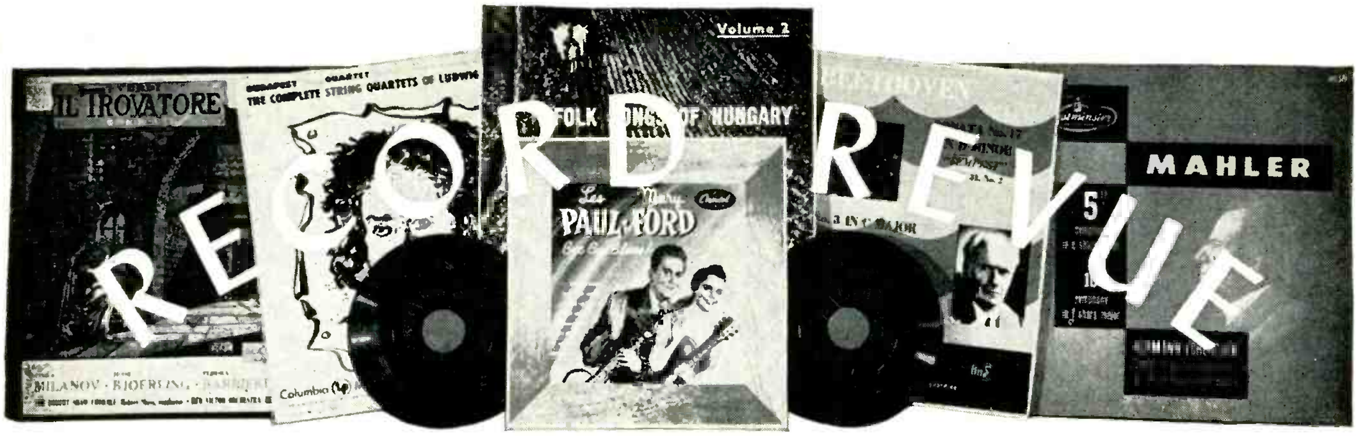

EDWARD TATNALL CANBY*

Fairly Epic

† Fauré Requiem. Orch. des Concerts Lamoureux, Choeur E. Brasseur, Soloists, Fournet. **Epic LC 3044**

This is definitely an epic release—for it is musically the first "Fauré Requiem" that really hits the somewhat pastel-tinted Fauré nail on the head! To my ear (and I've sung in the Requiem several times) this version is far ahead of any of the others on records—so far that it has been an intense pleasure to hear. That oddly restrained Catholic eloquence, Romantic but very French, not unlike the more familiar feelings of Cesar Franck's music, is extremely difficult to catch in performance. This one gets it to perfection and the music fairly jumps alive at you. Wonderful boy's voices, choral and solo.

Recording is good, though no hi-fi miracle.

Schubert: "Trout" Quintet. Amsterdam Piano Quintet. **Epic LC 3046.**

Aha! This is one of the best Epics yet, technically; evidently the problems involved in adapting the Epic originals to our LP are beginning to be solved. Still a slight nasal tone in the louder fiddle portions but on the whole this has an excellent open sound, very natural and musical. There is an odd emphasis on the violin (there's only one) which seems closer and louder than the other strings; but this is within the musically permissible variation in mike pickup and serves to accent the nice play between the constant high piano octaves and the fiddle part which Schubert wrote into the music.

The special quality of this score is its unusual combination—sharp, glittering, light-hearted piano and solid double bass, with violin, viola, and cello in the middle. The single double bass, very rarely used in so-called "chamber music," adds a peculiarly nice bottom to the sound not unlike the same double bass effect in a small jazz ensemble. Jazz specialists please note and compare.

This is a very musical though quiet performance, not as electrically charged as Serkin's piano might make it or the Budapest Quartet's version with another pianist, Horszowski, nor as heavily Romantic as the versions done in Austria. (Badura-Skoda saves the Westminster recording from over-heaviness.) A good record.

x Mozart: Piano Concerti #18 in B Flat; #19 in F. Hans Henkelmans: Vienna Symphony, Pritchard. **Epic LC 3047.**

Technically another substandard LP, for reasons not easy to fathom. The string sound, even at full (NARTB) roll-off, is nasal and thin with considerable distortion. (Won't show on limited-range home machines.) The piano, however, is good, wise mike placement making up evidently for technical imperfections. The piano has few high highs and wouldn't show up distortion of the sort that appears here in the strings.

Musically a good disc, the pianist sensitive and careful, the orchestra straightforward and reasonably kind to the music. (Many a Mozart concerto is ruined, even by the biggest musical name performers, by insensitive, harsh, unphrased, or patronizing playing.)

**780 Greenwich St., New York 14, New York.*

X Lalo: Overture "Le Roi d'ys"; Norwegian Rhapsody; Namouna Ballet Suite #1. Orch. des Conc. Lamoureux, Fournet. **Epic LC 3049**

Same orchestra as in the Fauré Requiem above; but here the "fi" is of more importance—the music is pleasant French-Romantic stuff, not too far from Gounod and Saint-Saens, relatively speaking. It is of the sort that needs real hi-fi sound to be worthwhile on discs for the non-specialist. The softer parts are fine, but louder passages are not clean here and that persistent slightly nasal string tone is still there. So-so.

KEY

- * Outstanding recorded sound for the type of material
- † Unusually fine performance
- B Big bass: heavy percussion, fine transients
- bb Bass end a bit thin. High turnover?
- c Close-to, sharp-edged, but in good liveness
- d Distant, overall miking, good liveness and presence
- D Recorded with deadish acoustics
- L Big, blown-up liveness
- s Solo part rather close and loud
- x Some distortion
- X Some distortion in louder passages

†X Mozart: Cassations in G, K. 63, B Flat, K. 99. Vienna Symphony, Sacher. **Epic LC 3043**

A Cassation is similar to a Divertimento—an informal small-size symphony with many movements, short and nicely contrasted for easy listening. These early Mozarts are beautifully written as only that incredible genius could do at 14. The playing is lovely, especially in the slow, singing movements. Technically similar to the Mozart Piano Concerti above, though a thinner instrumental ensemble makes distortion less noticeable. Could this series be from disc originals? Sounds that way to me. Substandard as to engineering, even if super-standard musically; Mozart lovers will thoroughly enjoy this anyhow.

bb Brahms: Vars. and Fugue on a Theme by Handel; Paganini Variations. Abbey Simon, piano. **Epic LC 3050**

A Romantic-style interpretation of these two works, legato and heavily pedalled, unbrilliant, technically not entirely at ease (these are fiendish pieces to play) but musical; I find this kind of playing much preferable to the hard, shiny, super-virtuoso stuff we hear so much nowadays. Good piano recording, with a natural sound and big liveness, though the treble is a little uneven. Bass can stand a bit of boost over U.S. average playback.

Vox—Many Keyboards

With this company's numerous releases I'm just beginning to catch up, after a too-long hiatus. The Vox recorded sound rates on the average in the higher hi-fi brackets; the company has developed some extremely effective milking techniques, though not with as much acclaim as some other concerns.

c Schubert: Sonata in C Minor; Sonata in B Minor. Friedrich Wuhrer, piano. **Vox PL 8420**

d Schubert: Sonata in G Major; Sonata in A Major. Friedrich Wuhrer, piano. **Vox PL 8590**

Four superb sonatas, part of a recorded series that will cover all Schubert's piano works; the first two here are remarkably seldom heard, the other two are more familiar. Schubert's piano sonatas aren't too often played in concert because of their great lengths, relative to other similar music. Concert audiences can't easily take so much concentrated stuff; they lose track of the sense and continuity and mood and begin to fidget. But on records it's altogether another story—for we can listen again and again, and Schubert's music has a most astonishing way of penetrating and sticking, on repetition, so full is it of wonderfully catchy tunes and motives and harmonic progressions and rhythms.

On records you can let these slip by effortlessly the first time or two, without fidgets; then along about the third time the miracle will begin to happen and you'll never recover as long as you live. A lifetime investment, musically speaking.

Friedrich Wuhrer is without doubt a big pianist and a Schubert man to reckon with. His hands are like steel and the piano fairly shivers in the louder passages—the music can take it too, since Wuhrer also knows his lyric contrasts and can soften his tone beautifully between tempests. He has a fine sense for the complex and beautiful shifting harmonies in these works, where many pianists, among them the biggest, fail dismally to "get them over" in their proper ultra-dramatic meanings. The Wuhrer finger technique is only so-so and the fast passages are on the rough and uneven side; the whole tone of the music is rather severe. But good musicianship, a true ear, and a fine feeling for the musical architecture combine to make his Schubert highly listenable.

The recorded piano sound varies here. The first disc seems a bit close to the mike, with a very slightly thin and wiry sound, rather percussive; the second disc, perhaps made on a different piano or in a different hall, seems on the contrary somewhat too far away with a blurring of the high tone into the general liveness. Both rate as good piano recording, if not tops.

†*c Stravinsky: Les Noces; Mass; Pater Noster; Ave Maria. Soloists, N.Y. Concert Choir, Orch., Margaret Hillis.

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Vox PL 8630

†* **Orff: Carmina Catulli.** Soloists, Wiener Kammerchor, Hollreiser.

Vox PL 8640

Here are two most unusual works, both scored for an odd combination of four pianos and percussion plus solo and chorus voices—and there couldn't be two pieces more different. (The Stravinsky Mass, Pater Noster and Ave Maria are dividends in the "Les Noces" record.) Both are stunning recordings technically, but one is European and the other American, the difference in acoustical situation being exactly what you would expect, a close, dry super-realism in the American recording and a more luminous, live sound in the European.

"Les Noces," begun in 1914 and finally orchestrated in 1923, is a work right out of the "Sacre du Printemps" (Rite of Spring) period and it'll be enough to suggest that if you like the "Sacre," you'll he able to catch onto it fairly quickly. Same semi-barbaric violent march-like structure, same semi-Russian background, this time a Russian peasant wedding instead of some pagan sacrifice; the idiom is largely transferred to voices here, with the pianos et al as percussive reinforcement.

"Les Noces" is sung in English, but you'd never know it. Maybe it's too much to say that Stravinsky ignores words. Rather, he has always made a point of putting aside just about every trace of the normal word accent, the ups and downs of phrasing and the longs and shorts of speech that make for spoken intelligibility. His words are merely sounds that accompany a strictly instrumental vocal part.

Choral singers and vocalists often resent this, and I do myself, for I feel somehow that if you use words for music, you should treat them as though they existed, so to speak. But I admit the validity of Stravinsky's very positive system. It is not a weakness—rather a strength of a sort that happens to conflict with other ideas of how words should be used. You may follow the printed English text here, word for word, and it helps—but you're likely to get lost at any moment, so hopelessly obscured are the original word-shapes. But once you forget the words and begin to hear these voices on instruments of the ensemble, you'll be amazed at the effectiveness of the whole. This is a piece as strong and as wild (and as disciplined) as the "Sacre" itself. Hi-fi effects throughout, too.

Carl Orff's Songs of Catullus (setting the Latin text of Catullus, which in large sections is much too hot to be translated) is of a much later date, 1943, and a wholly different texture. This joyously and tragically abandoned love poetry is set outwardly for the same kind of arrangement. But Orff, a South German composer in his late fifties, has an impeccable sense of word-shape and an enormous respect for the words themselves. His settings use a very fast, pagan-style chanting with a peculiar device of rapid repetition of each phrase over and over, for emphasis, to a jazzy and catchy rythm. Indeed, this whirlwind of easy Latin suggests the flippant sung Italian of Rossini (and the patter-English of Gilbert and Sullivan.)! Some passages are shouted, spoken, or just plain mumbled.

Musically "Carmina Catulli," for all the publicity it has had, doesn't come within miles of Stravinsky; it is, for that matter, harmonically static, it marks time energetically, with powerful rhythms but without very much musical structure to it. Large sections go along on only two notes, repeated ad infinitum. But it's fun to listen to and incredibly skillfully performed, as is "Les Noces" under the Hillis Concert Choir, of New York. Two superb records.

P.S. The Stravinsky Mass is a major recent work, rather antique and classic in its structure but continuing with modifications the Stravinsky way of setting words without their natural rythm and shape. Not too easy listening but good stuff. The other two are short items composed for Russian Orthodox services, short direct settings of the texts combining the rich traditional ("Don Cossack") Russian church music sound with a characteristically Stravinsky irregular rhythm and static, broken-motive harmony. Nice.

LS **Bach: Concerti in D minor, C major for Three Keyboards; Concerto in A minor for Four Keyboards (after Vivaldi).** Soloists, Pro Musical Orch. of Stuttgart, Reinhardt.

Vox PL 8670

If ever true stereophonic sound were needed it is right here—but the "binaural" promotors are more interested in hi-fi sound-effects. These works for three and four simultaneous keyboards are dutifully performed quite often because the music itself is intrinsically wonderful stuff. But on four pianos the mixture of sound, without overtone coloration, is a hopelessly confused mess even on the concert stage (the halls we use are too big so that the angle of difference between the instruments is far too narrow) and of course even worse in the non-directional monaural recording.

The music was intended for harpsichords. Their bright, shiny tone makes it possible to hear the content of the whole batch of keyboards together without heaviness, where on the piano the sound is knee deep in mud. This beautifully done recording will quickly prove that—even though the instruments, to my mind, are too loud in the balance and so on the heavy side (except, paradoxically, the four-keyboard work, which is better).

But what was the real original intention? No less than a 3-D perspective, wide-angle! In those times the listening rooms were much smaller than our concert halls, the audience was often very close to or even grouped around the performers. The instruments could easily be distinguished by both eye and ear together, as the performers played. The music was highly visual.

On a cinemascope screen (or Cinerama) this effect could be duplicated marvellously—if anyone were interested. But even minus the picture, a similar wonderful clarity and directional play of sound, exactly as intended by Bach, could be achieved via a stereophonic many-channel pickup—or a direct, "point-source" multichannel pickup with one channel and one mike devoted to each soloist, the orchestra spread through all channels.

The Pentron Company (perhaps inspired by a little account by me of such a "point-source" recording in the April 1953 issue of Audio, p. 52) is now featuring an "All-Electronic Orchestra" recording. In its six channels each of six instruments is picked up directly on a separate mike and recording channel, to be reproduced from its own single speaker "on the spot."

This technique (I used two channels for a dramatic dialogue between two actors, each with his own mike, spaced fifty feet apart) could be used to enormous advantage in these Bach works: three, or four, channels for the separate instruments, one apiece, the remaining channels for the orchestra and background liveness. It would bring out the whole built-in directional sense that is a part of this music.

Note also that Bach's Brandenburg Concerto #3 divides the string orchestra itself into about a dozen separate small groups, two or three men to each, and tosses its thematic material around in echo fashion from one to another in a startlingly directional manner, totally lost in a monaural recording where, of course, you hear merely a lot of repetition all from the same spot, as though the same instruments were merely repeating themselves. Far from it.

There is a famous "visual" passage in this concerto, almost humorous, in which a certain rather tricky little idea involving violent bow action (an arpeggio figure) proceeds straight across the orchestra through each group one after the other from top to bottom (that is, from left to right on the stage), becoming more difficult for each group, until it hits the celli and basses who practically do somersaults to play it. The sudden violent flurry of arm-sawing that hits each group as the theme reaches it is a delightful visual effect and surely was intended exactly that way by Bach himself. This is "3-D music par excellence, then, both as to sound and sight, but especially the sound. Pentron please note!

(The Vox monaural sound is excellent, with a huge liveness to the orchestral strings. Performance, as often in Stuttgart, is musical but a bit on the ponderous side.)

Remington

Still a lower priced label, its musical material now a bit more orthodox; there have always been remarkably good buys among the Remingtons. Surfaces are almost up to average, but the recorded sound quality tends to be a bit fuzzy in the loud passages, a kind of breaking-up that would seem to be in the cutting and pressing cycle, rather than in the original tapes. Not serious except to the super-hi-fi man.

* **Debussy: La Boite a Joujoux (Toy Box), orch. Caplet.** RIAS Symphony, Perlea.

Remington R-199-59

An excellent recording and nice performance of an unusual bit of Debussy, a ballet score that was left unperformed due to the first World War. Debussy had finished it in piano form; after his death it was scored by a friend of his and staged. The orchestration is good, entirely in the Debussy tradition. Humorous music, nominally for children but better for grown-ups who like clowns and what-not, an excellent ballet music disc to add to your collection of colorful orchestral music.

Note that Menahem Pressler plays the original piano version superbly on M-G-M E3042. A very interesting comparison.

* **Spanish Gypsy Airs and Rhythms.** Carlos Montoya and his Ensemble, Jose Moreno, Flamenco Singer. **Remington R-199-171**

A first rate Spanish guitar record, very well recorded. (But keep the volume down to normal guitar level.) This is the great Spanish folk tradition, unwritten, improvised, extremely strict as to its characteristic musical and rhythmic and vocal traits. Some numbers have castanets and (apparently) dancing feet as well as guitar—a bit too much of the dancing feet noise. Several have the typical quavering, elaborate vocals of the Flamenco style. (Jose Moreno's mike technique is evidently absent-minded. He seems to be in motion, towards and away from the mike, as he sings, perhaps dancing. This must be more or less of an "on the spot" pickup—which makes it doubly authentic.)

†* **Alec Templeton improvises on Offenbach and Strauss.** Alec Templeton, two pianos. **Remington R-199-158**

An extrordinary record, musically and technically. The art of improvising is the oldest in music—all music was, after a fashion, improvised before notation began. For a blind man, notation does not exist to this day and all musical re-creation must be by ear. A musician with Templeton's extraordinary gifts combines a fabulously exact memory (not only for tunes but for the entire harmony and fabric of a piece, which he must learn by hearing alone) an equally ready gift for re-composing, arranging, combining this material, re-fashioning it, in his own way but in its own style and language, and finally, a tremendous playing technique that can put musical thoughts instantly into action.

There aren't many improvisers today, except in jazz and folk music. (There are a lot who "improvise" noisy or tricky music that is really so much hash.) A real improviser "speaks" his musical languarge as we do our spoken language —quoting, making up, patching this and that, re-phrasing, all with impeccable grammar, according to the language in hand. As Templeton says, the music of Offenbach and Strauss is so clearly made and straightforward in its harmony that its parts can "be fancifully interwoven and intermingled with ease and effect." Yes, if you know how!

Thus the Strauss on one LP side and the Offenbach items on the other here fall easily into four-movement structures, two "little symphonies," easily in accordance with the character of the musical material. You'll hear all sorts of familiar things—but you will not, unless your ear is very quick, be able to tell how and where Templeton has done the patching and the refashioning, for the whole is absolutely impeccable musically. An absolutely remarkable feat of sheer musicianship, even though clearly there was a good deal of planning in the head before the recording itself. (The head and the memory are Templeton's only planning department.)

Templeton is, in a light way, a successor to Bach, who also improvised constantly and impeccably upon many well known tunes of his time, with the same knack of combining them in tricky ways. But there is a significant difference that is a commentary on our times. Bach improvised in a style that was of his day, or at least of his times. He could write his greatest music in the same style and often did, writing out his improvisations. (So, too, did Mozart, who could improvise and memorize note-for-note simultaneously.)

But Templeton plays and thinks in a musical style of a hundred years ago—and it is wholly intelligible to the mass of our people. We do not yet, except in our popular music and jazz, have a style that is well enough defined and understood to lend itself to good improvising—which requires by its very nature a thoroughly developed and quickly understood language, without confusion of meaning.

Note that this is two-piano music, and about the best-composed two-piano music I've heard in its clear use of the advantage of twenty fingers and four sets of arm-muscles. Templeton, you see, plays both pianos, and has improvised both parts, playing the second one by ear to the first one as he hears it played back. What a superb musical feat of pure thinking-out-loud! Bach would have loved to try it.

I recommend the disc not only as wonderful light entertainment, good piano recording (loud parts a bit peaky) but as a very serious lesson in musicianship and composition to all who are interested in the essence of the musical language. Just you try it, Mr. or Miss Famous Pianist.

Oddities

* **Tchaikowsky Waltzes, arr. Four pianos.** Manhattan Piano Quartet.
M-G-M E3100

Add this to the multi-keyboard department (above). Four pianos have been quite the rage these last years, partly due to the visual symmetry of dress tails on a stage and the sheer impressiveness of four huge black instrumental monsters all operating together. But aurally, four pianos offer a certain broadening of tone color and a thickening of counterpoint and decoration, at the expense of freedom and flexibility. Timing must be split-second, and tempi are apt to be on the rigidly controlled side.

These four skilled players carry on with a certain steely persistence, whanging away brightly to the last cadence; the sound is not unmusical but it does give a certain impression of having been wound up like a mechanical music box, to unwind very neatly and precisely. Good piano recording—four pianos always sound good, my theory is that with four of them the mikes usually are further away and the room is necessarily bigger, which gives almost automatically good pickup. Can't go wrong.

*D **Weill: Threepenny Opera.** Lyrics transl. Marc Blitzstein. Theatre de Lys (N. Y.) production.
M-G-M E3121

This is the New York revival in English that was a notable off-Broadway hit during the past New York season. (All New York sensations now arrive sooner or later in recorded forms; I just sit back and wait for them.) The opera, from 1925, is a sort of earlier counterpart of "Porgy and Bess," more startling in its day and, as of now, more dated in sound, in a very pleasant way—the instrumentation is pure 'twenties, with strumming banjo and the batty sound of the oldest jazz records—and, of course, of Stravinsky's "Histoire due Soldat" of 1918.

As pure music Weill's best work (I'd say) is still not great shakes, unless you want to line it up with a thousand or so musical comedies of later date. It's not in a class with pure Mozart or pure Ravel or pure Copland. The tunes are sometimes good, sometimes just banal, the arranging similarly is piquant in its old fashioned jazzy tone color but a lot of it just doesn't say very much.

However as a show this work picks up immensely—even on records. It is a show, and perhaps it's unfair to judge it any other way. The raucous singing and the slangy words (in the Blitzstein translation) are amusing and effective and very easy to follow—as easy as a good musical comedy. Nope, this is no imperishable masterpiece, but it is a strongly period piece in a style that is increasingly interesting to us as we move away from the twenties.

Excellent hi-fi recording, dry and sharp but highly suitable to the music. (A big golden liveness would be deadly.)

Bc **Beethoven: "Battle" Symphony, Op. 91; King Stephan Overture, Op. 117; Eleven Viennese Dances.** L'Orch. Radio-Symphonique de Paris, Leibowitz.
Oceanic OC #34

You've probably heard that the "Battle" Symphony was a famous Beethoven potboiler. It was, having been originally commissioned for a mechanical "orchestra," the Philharmonicon! It never got played that way, but Beethoven scored it for flesh-and-blood orchestra and made hay with it—since it celebrated the end of the Napoleonic power and the restoration of Beethoven's own Austria to national self respect. As a famous man, Beethoven could and did do well with such a piece in this worthy cause.

It's a noisy, padded-out work (though B. himself professed to think much of it, I hear) and the battle-proper is pretty sad, what with a soggy bass drum (beautifully recorded) as the cannon-power and much musical playing-about with "The Bear goes over the Mountain"—an air originally French and representing Napoleon's forces—and "My Country 'Tis of Thee," more properly "God Save the (British) King." But by all means do not overlook the final section, a triumphant treatment of parts of "God Save the King" in the very best late-middle Beethoven manner with strong suggestions of both the Fifth and Ninth Symphonies. Worth the whole record.

"King Stephan" is a late overture, a part of the "Ruins of Athens" incidental music; a good small overture if not a great one and well worth knowing. The Viennese dances are late ones, pleasant and melodious though repetitive, in the pre-Strauss Austrian manner.

Oh yes—the recording is rather terrifyingly hi-fi, with immense drum beats and crashing cymbals, an ultra-realistic definition of instrumental timbres almost uncomfortably revealing in detail (at the expense of good blending), the whole in an impressive liveness. Performance is on the squeaky side—all French playing of German heavyweight music seems that way—and rather rough and uneven throughout, but on the whole intelligent. Typical of Leibowitz, who seems to spend much of his time doing quick conducting commissions in Paris for various U. S. companies, probably with a couple hours' advance notice and practically no rehearsal! My guess, anyhow.

Arias Sung and Acted. Albanese, Bjoerling, Merrill, Warren; Deborah Kerr, Joseph Cotten, Dennis King; RCA Victor Orch., Cellini.
RCA Victor LM 1801

Here's another of RCA's current experiments with operatic arias; this time it's newly recorded ones only, but with a special slant—the actors in the recording first "act" an English translation of the scene, then the singers take over and sing it in the original language.

It's an interesting experiment, if not exactly as RCA bills it. What hits this listener most forcibly, hearing the libretto "acted" by professional actors, is the extraordinary difference between opera and spoken drama, and the unsuitability of the opera libretto to any sort of straight spoken reading.

Opera is strong but delicately balanced stuff; it can get over a kind of drama and excitement via its highly stylized and artificial musical medium, that no stage drama can cope with, short of the ridiculous. Even though a couple of these arias come from librettos actually adapted from stage works, the stuff of them is so much emotional poppycock, when it's spoken minus music. "They call me Mimi," indeed! Sounds far worse than any soap opera. Yet no one will deny that in its operatic form this aria is an effective bit of stage drama. So it goes with the others too.

One spoken scene is actually from the original play, "Camille" ("La Dame aux Camélias") by Dumas, on which the opera "La Traviata" is based. What then? Well, things fare no better—and perhaps the reason is an added thought-provoker. The modern stage style affected by this group of well known actors (above) is to my ear as out of place and as forced, in this delicate bit of late Romantic realism, "Camille," as it is in the flowery stuff of actual operatic libretto. One cannot die of consumption, giving up one's lover heroically at the behest of his father, in any ordinary modern stage language—except, as I say, in modern soap opera, where the tragic Marguerite (Violetta in the opera) would be entirely at home, consumption and all!

Students of drama and opera and the stage in general would do well to study this record in these respects. Good class room material. (Performances? Strictly routine professional, OK but hardly scintillating. Too many arias sung and acted by the same voices; it gets monotonous.)

AUDIO ETC.

(*from page 64*)

I have been struggling all spring trying to find a means of recording Ampex-curve tapes on a Magnecorder and, thanks to Jan Syrjala (of my "Pipeless Organ" article) have succeeded within a few db, but only after endless and agonizing tinkering—and the alterations in my machine are not suitable to a standard Magnecorder, which mine is not.

Magnecorder tapes played on an Ampex come out with a perfectly horrendous droop in the highs, 20 or 30 db or so and an unpleasantly boomy bass. Ampex tapes played on a Magnecorder are the opposite—high hiss, screetchy treble and thin bass. Radio station engineers evidently ignore these things with that blithe indifference to actual sound that only a man working with at least three speakers blaring simultaneously all day long in his ears can develop! Nobody bothers to equalize—they just blame the sound, if they listen, upon the maker of the tape.

Pre-recorded tapes are likely to follow several drastically different policies here and I predict a dismal mess if somebody doesn't do something quick. (A) They may take the Ampex playback curve as a practical near-standard, which would be a good idea as far as I can tell. (B) They may produce a unique curve of their own, to be "different," or perhaps to suit their own "unique" equipment, just a bit different from anybody else's. Thus the tapes will sound best on their own equipment. Very short sighted, I say. (C) Worst of all, the new tape records may be recorded with a tricked-up curve designed to get the most out of low-fidelity home machines. This, I gather, has already been done in at least one of the well-advertised new tape releases, though I will name no names until I've had a chance to try for myself. An *extremely* short-sighted policy, this, one that we have been through and long since put aside for good in the disc record field.

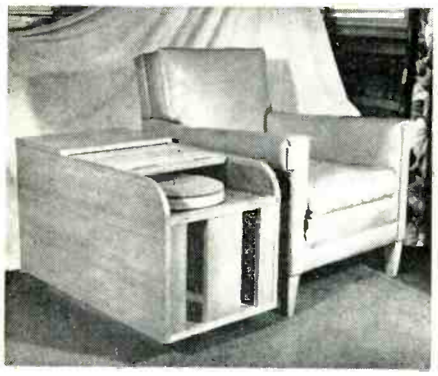
For Pete's sake, let's assume, first, that tapes, like records, are supposed to be interchangeable and playable on "all" machines. And second let's assume that the machines must be made to fit the tapes, not vice versa. We play wide-range, top-quality discs on cheap machines now and nobody any more makes any public pretense of producing records "especially suited" to small phonographs. (They're all rated as "hi-fi" phonographs anyway, so it wouldn't make much sense!)

I hope and pray that we are not going to go through the whole ten years' battle of recording curves all over again with tapes! But as things stand right now, the prognosis is not good at all. Indeed, most people don't even know that there is a problem of equalization in tape, though in the disc field a pretty good general idea of equalization has got around among the record collectors. I have not yet seen the slightest printed indication of recording curve—or even of a preferred playback machine—on any sample of pre-recorded tape so far to come my way. Just put it on and play it, is the implication. I understand that among the home tape recorders there is also a fine anarchy of recording curves and playback curves (they complement each other, of course), so that this easy-going direction is just set and bound to make trouble.

How about some committee work in the professional organizations on this newly pressing problem of standardization—before the chaos gets started in earnest?

NEW PRODUCTS

• **End-Table Cabinet.** Adequate space for turntable, amplifier, and record storage is afforded by the new end-table enclosure

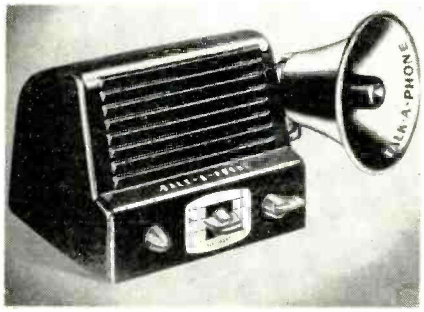


recently introduced by Components Corporation, Denville, N. J., for housing the company's professional model three-speed turntable. A convenient roll top permits the record to be covered while playing, thus reducing needle talk as well as protecting the disc against dust. Ample space is available for mounting any standard tone arm. The cabinet may be purchased in either blonde or mahogany.

• **High Fidelity Amplifier.** Rated power output of 12 watts and frequency response of 20 to 40,000 cps within 0.5 db are among the characteristics of the new Regency Model HF-150 audio amplifier. Intermodulation is less than two per cent at rated output. A single-chassis unit,

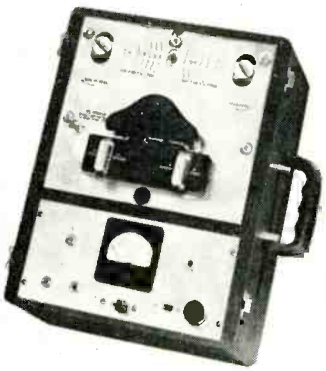
the HF-150 affords complete command of reproduction through five operating controls: bass, treble, loudness, level control, and combination record compensator-input selector. Three high-impedance inputs are supplied, one equalized for magnetic cartridge. Bass control ranges from 17 db boost to 25 db cut at 30 cps; treble control ranges from 13 db boost to 20 db cut at 20,000 cps. Noise level is 60 db below full output in phono position. Chassis and control panel are finished in gold-anodized aluminum. Further information will be supplied by Regency Division, I.D.E.A., 7900 Pendleton Pike, Indianapolis 26, Ind.

• **High-Level Intercom System.** Designed primarily for use in large areas and to overcome high noise levels, the new Talk-A-Phone master selective system has ten times the volume level of conventional in-



tercoms. It is available in two models, AC-5406 and AC-5411, differing only in the fact that former has a capacity of five substations while the latter will handle ten. Only the master station plugs into an electrical outlet. Both systems have an optional feature whereby substations can be connected privately or non-privately and still originate calls to the master station. Once a call has been initiated to a non-private substation, persons at other substations need operate no controls and can reply from a distance. Where the system is connected privately, the master unit cannot listen in on substations. Talk-A-Phone Company, Chicago, Ill.

• **High Fidelity Tape Recorder.** Many professional features are incorporated in the new Pentron PMD-1 tape recorder. Consisting of the new Pentron deluxe pre-

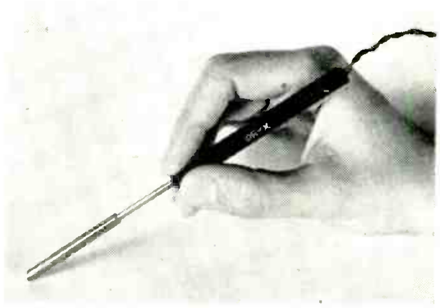


amplifier Model HFP-1 intermatched with the Model 9T-3M tape transport mechanism, the unit is supplied ready to plug into any high-fidelity music system. Features of the recorder include frequency response from 50 to 12,000 cps within ±3 db, illuminated VU recording meter, two tape speeds (3¾ and 7½ ips), and up to two hours recording or playback time. The motor used is a four-shaded-pole induction type. Recording heads are specially designed with removable pole pieces. The PMD-1 is attractively designed with brushed copper panel and contrasting jet black controls. Literature and complete details may be obtained from The Pentron Corporation, 221 E. Cullerton, Chicago 16, Ill.

• **Tape Playback-Amplifier.** The Rhein "Powertape" is a complete public address system, comprising a 30- or 50-watt power amplifier and a tape playback mechanism mounted on a compact, portable chassis

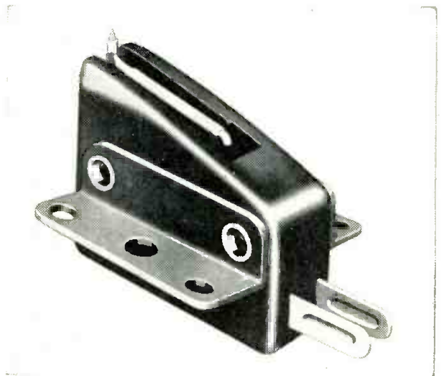
assembly. Inputs and controls are provided for two microphones, tape, and external phono. Tone compensation is afforded by separate bass and treble controls. Frequency response is 40 to 12,000 cps for microphone and phono channels, and 50 to 8000 cps for tape reproduction at operating speed of 7.5 ips. The tape unit is of the two-speed type which may also be operated at 3.75 ips when desired. Microphone gain is 120 db; phono channel, 76 db. The "Powertape" is of exceptional interest where music, speaking or entertainment can be provided by pre-recorded tapes. Descriptive sheet available from Rhein Sound Systems, Inc., Orlando, Fla.

• **Tiny Soldering Iron.** Weighing only one-half ounce, the new Oryx sub-miniature soldering iron, imported from England, offers remarkable advantages to engineers and hobbyists who service or construct delicate electronic devices. Only six inches in length, the units require but 30 seconds to heat. A variety of tips and operating



voltages is available, including six- and twelve-volt types for use with mobile equipment. Details and prices may be obtained by writing the manufacturer's U. S. representative: Television Accessories Co., Dept. 25, Box 6001, Arlington 6, Va.

• **High Fidelity Ceramic Cartridge.** An exceptionally flat frequency response from 30 to 15,000 cps and compatibility with the recording characteristics of modern records are among the more prominent features of the new Astatic Model 51-1-J pickup cartridge. Low mass of moving parts, coupled with high vertical and lateral compliance, results in extremely low distortion. The same factors also afford reduced needle talk and excellent tracking at low stylus pressures. Output of the unit is 0.70 volts, nominal, when used with the RCA 12-5-31V test record

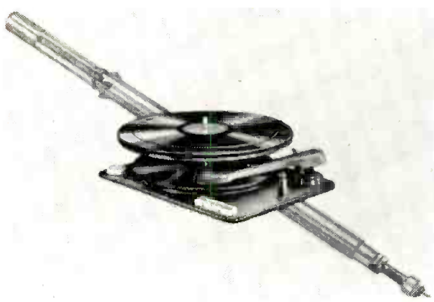


at 1000 cps. The tiny cartridge fits all standard tone arms with standard RETMA half-inch mounting holes, including plug-in head types. Stylus is replaceable and is available with either one-mil or three-mil tip, sapphire or diamond. The Astatic Corporation, Conneaut, Ohio.

• **High-Power Ultra-Linear Amplifier.** Power output of 60 watts with less than one per cent intermodulation is a guaranteed performance standard of the new Sonex Model UL-601 ultra-linear amplifier. The output stage is made up of four KT-66's operated in push-pull parallel. The amplifier is rated to deliver full out-

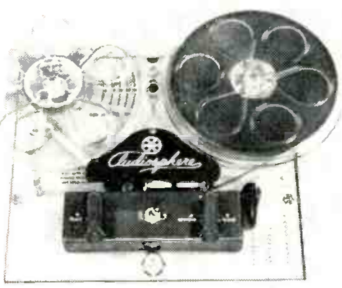
put within ±1 db with less than 0.5 per cent distortion at any frequency from 20 to 20,000 cps. Conservatively rated power components and 600-volt capacitors assure operating stability and long life. For convenience of installation, the UL-601 has a.c. receptacles, power take-off for preamplifier, and provision for remote on-off switching. Complete data may be obtained from Sonex, Inc., 245 Sanson St., Upper Darby, Pa.

• **Record Changer.** Many unique constructional and operating features are incorporated in the new Miracord Model XA100 three-speed changer which is being introduced by Audiogersch Corporation, 254 Grand Ave., New Haven 13, Conn. Designed primarily for high-fidelity operation, the changer is driven by a four-pole motor which is totally screened to prevent hum radiation. A new type of spindle known as the "Magic Wand" houses the changing mechanism. In operation, rec-



ords to be played are stacked on the spindle which drops them in the desired sequence as with the familiar 45-r.p.m. drop mechanisms. The XA100 is push-button controlled with switches provided for governing the following functions: (1) Start-reject: (2) Variation of pause time between records. With LP's this time interval can be varied from 12 to 328 seconds. (3) Filter. This is a "scratch eliminator" for playing old or worn records. (4) Repeat. Pressing this switch will repeat any desired record. Ten- and twelve records of the same speed may be intermixed at will. Plug-in head permits use of any standard hi-fi cartridge. Tone arm is ball-bearing mounted. The changer is finished in dark maroon with rubber-matted turntable and operating controls in contrasting white.

• **"Bi-Fi" Tape Playback Unit.** Designated "Bi-Fi" because it handles both conventional and binaural tape recordings, the new Audiosphere tape playback unit is being distributed as a complementary addition to the library of pre-recorded tapes



manufactured by Audiosphere, Inc., Livingston, N. J. Basically, the playback consists of a transport mechanism, a two-channel preamplifier, and an attractive cabinet which is available in either blond or mahogany. Equipped with a selector switch which allows three different playback conditions, the unit makes possible the building of a binaural as well as a monaural tape library, inasmuch as the binaural tapes can be internally mixed and delivered to an external amplifier as a monaural program. Thus binaural tapes can be played monaurally until a second channel is available. The playback affords two operating speeds, 7.5 and 3.75 ips, and the output level of each channel is approximately 0.3 volt. Distribution will be through regular hi-fi jobbers and distributors.

AUDIO AUDITIES

SAUL J. WHITE

The Autobiography of a Sound Wave

My LIFE IS FAST but short. I am quickly born and quickly die. I have many moods and forms, from the terrifying clap of thunder to the whimper of a baby. I am the growl of a wild beast, the parting of a glacier, the eruption of a volcano, the song of a bird, the sweetness of the soft touch of a string. I am a train-whistle across the night air, the squeak of new shoes and the cling of silver and gold. I have brought man out of the primordial mud, brought man down from the tree, out of the cave. Because of me man can teach his young, can be alerted to dangers, can woo his mate, can locate the ocean bottom or craft in the sky. The communication I offer builds friendship, intelligence, and culture. Of course I also give credit to the basilar membrane of humans, who of all creatures know and enjoy me most.

Scientists have given names to my various forms. When I am repetitious I am known as pitch and have musical quality. When my pitch is smoothly undulating I am told I am pure and sinusoidal, but when my undulations have slight quivers, jerks or sharp edges, I am complex and full of harmonics. When I lack repetitiousness I am merely a noise. I am affected by every vagary of the atmosphere. I am sensitive to heat and cold, dry air and moist, altitude, density, terrain, wind, and every invisible wistful current that circulates over the face of the earth.

Man has uncovered most of my routine habits with mathematical precision, or so he thinks, yet let him try to explain how I produce the sounds of the surf. I am forever creating new sounds, as who outside of recent years has experienced "jet thunderclap," that window-shattering force as I strike the earth from the pull-up of a jet aircraft dive. At one time during your school days you were told I behave much as a stone dropped into still water, spreading out in beautiful concentric waves. The truth is, I am nothing like that. I do not enjoy such simple serenity, because in my outward travels I encounter many obstacles. I am tossed about, diverted from my course, parts of me are torn off, or destroyed. It is seldom I arrive with the form and vitality with which I started. Sometimes my travels are filled with adventure and drama.

Let me tell you of one experiment I undertook with our good friend Lord Rayleigh, who perhaps more than any other scholar has delved deepest into my secrets. Although Rayleigh, along with the ancient Greeks, understood that a diverging wave will loose its intensity as the square of the distance covered, yet this inverse-square law could not account for my much faster decay which common experience informed him exceeds the inverse square rate. The anticipated loss of pressure, amounting to one quarter each time the distance was doubled, was always exceeded. Rayleigh knew these losses were due to other characteristics in the atmosphere beside the propagation effect. He understood that the threshold of hearing was of the order of .000,000,000,000,000,1 watt per square centimeter, and from this he readily calculated that a sound possessing an energy of approximately 25,000 acoustic watts could travel once around the earth in one direction if there were no losses except those due to the dispersion effect of an expanding hemisphere.

Rayleigh was a modest and conservative scientist. Most of his findings were based on mathematical analysis with only a minimum of experimentation. However, this is one of the rare cases where Rayleigh determined to set up elaborate experimental equipment to check on this propagation characteristic. Accordingly, he and his friends dragged a 600-hp steam engine to the top of a Scottish mountain. This drove a powerful mechanical siren. Several years before this experiment Rayleigh had uncovered the laws involving other characteristics of sound propagation through the atmosphere and had learned that many factors act to attenuate sound transmission. He selected a tone that would fall in the area of the ears' maximum sensitivity. It was here that I had my first chuckle, for almost 500,000 mechanical watts were to be expanded in order to set me in motion. On the basis of the inverse square law only, I should have been capable of traveling six times around the earth before passing below the threshold of audibility. But I held a surprise for Rayleigh and his assistants on that hot summer day.

Observers were stationed on surrounding mountain tops and in valleys both near and far. In fact, some official listeners were so remote that their findings were to be reported by telegram and letter. But alas and alack, no one outside of an hour's easy walk heard me. Upwind in a valley ¾ mile away I was inaudible, but on a mountain top three miles away I was just audible. Downwind in the valley I was detected as much as four miles from the source, but on a mountain peak only two miles away, with the wind blowing towards it, I was again inaudible. Those who heard the siren reported that its intensity fluctuated and that the character of the tone was altered.

Although the workers in the region of the siren were compelled to wear ear plugs, it appears that my maximum transmission was just over four miles!

Why was I heard as well on a mountain top against the wind, as in a valley with the wind? Why did my intensity vary second by second? Why, with such enormous

82 Elm St., New Rochelle, N. Y.

power, was I inaudible beyond four miles under the best condition? Because of the buffeting I received from every small variation in atmospheric conditions, and because of my struggles to remain on a straight course and to avoid being shoved and bent aside, sometimes reached a listener by several different paths with different time intervals.

The atmosphere twisted me into a distorted shapeless mass, absorbed and beat me down, robbed me of my vitality, took away my high-frequency components within a few hundred yards of my birth. In short, I exhibited a spectacular failure.

To overcome these meteorological obstacles, I must be propelled, not by *watts*, but by tons of high explosives. Even the energy in a bolt of lightning is insignificant and has never sent me hurling for more than five or six miles. But when I am released in the company of hundreds of tons of TNT, I can then furnish a measure of useful activity to scientists.

For instance, I can reveal the height and temperature of the stratosphere. I can ascend far beyond the highest sounding balloon and show up again on the earth hundreds of miles from my starting point. In fact, until recently I have been the only means for revealing conditions in the upper atmosphere, which in these days of projected rockets and guided missles is of extreme importance.

ABOUT MUSIC

HAROLD LAWRENCE

AIR from other Planets

IN PROGRAM NOTES on contemporary works, the terms "atonality," "dissonance," "polytonality" and "dodecaphonic" crop up as persistently as fireflies on a humid summer night. To the average reader, they are just as elusive. The only word that seems to evoke an almost universal response is "dissonance." On this score the listener needs no further elaboration; obviously it refers to sounds that are not "harmonious." As for the other expressions, they merely represent the jargon employed by the trained musician to disguise the fact that his music is intellectual, mathematical, etc.—hence, dissonant. Arnold Schönberg's vocal line in his F Sharp Minor Quartet, "air from other planets," might, to laymen's ears, apply just as well to great quantities of "modern" music.

The words "dissonant" and "radical" have at least two things in common: 1) their meanings depend upon whims of fashion, and 2) they reflect social and cultural development. Sir Humphrey Davy's discovery in 1800 of "laughing gas" was ridiculed or ignored for nearly half a century. Six years after Davy's experiments, a Viennese critic attacked Beethoven's Overture to *Fidelio* as a work in which "the most piercing dissonances clash in a really atrocious harmony." Both scientist and composer were ahead of their time.

The relativity of the term "dissonant," however, can be more dramatically illustrated in an example or two from the history of western music. Today we regard thirds and sixths as about as harmonious a blend of tones as a combination of two primary colors. In the Middle Ages, they were considered dissonant intervals. Even in Bach's day, a vestige of this harmonic stage remained in the attitude towards the minor third. Book I of the *Well-Tempered Clavier*, for instance, contains 24 pieces in minor keys, but only one concludes on a minor chord. All the others are resolved on the major, or "tierce de Picardie." To modern ears, the opening of Beethoven's First Symphony is a mellifluous set of chords preceding an extended lyrical theme. They provide about the same function as the dimming of lights and the raising of the curtain on a theatrical play; in each case, the audience's attention is the main objective. Yet in Beethoven's day, the idea of launching a symphony on a totally unexpected chord was disturbing. Tradition had it that the composer would begin his offering on the tonic. Thus, Beethoven's series of dominant sevenths threw his audience momentarily off balance.

For about two centuries, the harmonic language was that of the major-minor tonality. Even today, the music lover accepts this as an almost divine right. The majority of etudes and pieces in the standard repertoire are firmly constructed on this harmonic system. Our popular music also seldom strays far from the compound. Meantime, for over seventy years now, composers have been hacking away at the tonal structure.

Richard Wagner was one of the first to start the ball rolling. While never fundamentally challenging existing tonality, his chromatic melodies—some of which might be described as *glissandi* in slow motion—paved the way for twelve-tone writing. Debussy's revolt against traditional harmony turned his attention to, among other things, the whole-tone scale. Described as a "perfectly flat and featureless desert," Debussy's "organ tuner's" scale did away with the feeling of the "home" note; you can begin with, and end on, any note in the scale. Although Debussy rarely limited himself exclusively to this scale, he used it when it served his purpose. In *Voiles,* one of the piano preludes, he was aiming for precisely that effect suggested by the title—a sort of vague mist that comes out of nowhere and evaporates as mysteriously as it appeared.

Spearheaded by Glinka's use of "oriental" scales in his *Russlan and Ludmilla* (MGM 3053), the Mighty Five—and particularly Rimsky-Korsakov and Borodin—dipped into the reservoir of Russian folk music for melodies and scales that were then novel and refreshing sounds—and still are, for that matter.

Aside from chord enrichment and scale experimentation, another important development was that of polytonality. This involves writing in two or more keys simultaneously. In the allied arts, it might be compared to a painting in which two or more perspectives are presented at the same time, either side by side or superimposed. The effect, if not handled skillfully, is not unlike watching a 3-D film without proper lenses. One of the strongest advocates of polytonality is Darius Milhaud who, on occasion, has written passages in five or even six keys at the same time. An English musicologist once described polytonal counterpoint found in Milhaud's music as sounding like "the souls of politicians wailing in Hades," which is merely a Brittanic way of saying that it sounds like h—. A score like *Maximilien* (Westminster 5051), for which Milhaud was "abused, torn to tatters, and dragged in the dust" by the critics, requires a certain amount of concentration to follow its linear texture. Probably the most familiar example of polytonality in the standard symphonic repertoire is the combination of two major triads in Stravinsky's *Petrouchka* when the ghost of the puppet, portrayed by a pair of trumpets in C and F Sharp respectively, scares the wits out of the Charlatan in the concluding measures of the ballet score.

Shortly after the turn of the century, a young German composer named Josef Matthias Hauer delivered the *coup de grâce* to, what was to him, a tottering exhausted and outmoded system of harmony. He is the inventor of the "twelve-tone row." A definition, I think, is in order. Let's take one by the critic, Gerald Abraham: "The composer takes the twelve notes of the chromatic scale and arranges them in any order he likes. . . . This 'tone row' then provides the sole basis of the particular movement or composition. The intervals may be inverted—played backwards—or even transposed. But all the notes in the row must be used in the settled order each time before the row may be repeated."

Musical scrabble, you might say? Hardly. This theory, as developed later by Arnold Schönberg, was the practical result of approximately twelve years of work—a note a year?—whose aim was to "free music from the limitations of the major and minor system." In the latter system, the notes in the scale have varying degrees of importance: dominant, tonic, subdominant, etc., each performing specific functions. In the twelve-tone system, on the other hand, all the notes are equal; reminiscences of tonality are carefully suppressed; and a sort of tonal democracy emerges. (Soviet Russia, by the way, has officially outlawed twelve-tone writing as "bourgeois formalism.") Two words repeatedly associated with Schönberg's music are "atonality" and "dissonance." The composer disowned both terms. "For me," he once said, "no dissonances exist. . . . The title 'atonal' is meaningless and can only signify something that does not correspond to the nature of tone." He preferred the designation, "pantonal."

Atonal or pantonal, over forty years of twelve-tone composition have come up with only a handful of works that have been recognized by more than the tiny group of adherents to this musical philosophy. The most popular work by Arnold Schönberg himself is *Verklärte Nacht* (Transfigured Night), which exudes the lush fragrance of Wagnerian harmonies. In spite of his revolutionary teachings, Schönberg was a child of post-Romantic Germany who worshipped Brahms, Wagner, and Mahler. His new harmonic system failed to obscure the fact of this heritage, a clear example of which is the monodrama, *Erwartung* (Columbia ML 4524). Worthy of the Grand Guignol, the story concerns a woman who has a nocturnal rendezvous in the woods with her lover. She stumbles over his bloody corpse and, in a semi-hysterical state, makes love to it. A less exhausting introduction to Schönberg's twelve-tone music is the excellent Mercury recording of the *Five Orchestral Pieces* (Mg 50024). These orchestral miniatures are imaginatively scored and given evocative titles: *Premonitions, Yesteryears, Summer morning by a lake*, etc. You might just happen to find the water not as icy and forbidding as you thought.

Schönberg may have been the pioneer in a new harmonic and melodic language, but his disciple, Alban Berg, really put the twelve-tone system on the musical map. And, curiously enough, he did so by breaking many of the rules of dodecaphonic composition. For instance, he indulged in note-repetition, slipped in tones out of order, and —horrors!—even used a C Major triad in his opera, *Wozzeck*. No twelve-tone "ascetic," Berg took to heart Schönberg's philosophy: "When I compose, I try to forget all theories and I continue composing only after having freed my mind of them." In the case of *Wozzeck*, the *Violin Concerto*, and the *Lyric Suite*, the listener too soon forgets about theory and concentrates on what the composer has to say. An overpowering experience is in store for the discophile who has not yet heard the magnificent recording of Berg's operatic masterpiece, *Wozzeck* (Columbia SL 118). Proof of the work's enduring qualities is the fact that audiences of all strata of musical understanding and tastes seem to be equally affected.

The libretto of *Wozzeck* is based on a "drama fragment" by the nineteenth-century German poet Georg Büchner. The plot is simple: "Wozzeck, an orderly, is tormented by his superior, the Captain; by a physician to whom he surrenders himself for medical experiments that he may be able to support his beloved Marie and her child; and by visions rising out of his fantastic reveries. Marie is seduced by the Drum-Major. When Wozzeck, after torturing uncertainty, has convinced himself of her infidelity, he stabs his beloved and drowns himself."

In evaluating the extraordinary appeal of Berg's *Wozzeck*, Ernest Newman wrote: "The non-technical listener to the opera finds himself perhaps for the first time in his life, taking a vast amount of non-tonal music and not merely not wincing at it but being engrossed by it. That simple fact is the true measure of Berg's achievement; whether the listener can account for his interest or not the fact remains that he is interested in *Wozzeck* throughout, that he feels the music to be not only 'right' for the subject but the only musical equivalent conceivable for it."

In his book, *Schönberg and his School*, the French conductor and dodecaphonic composer René Liebowitz said that from Schönberg's theories will "radiate the death-rays that will finally pierce the inner shell of the tonal system." Schönberg himself took a much less warlike view of the musical world with his own forward-looking and constructive appraisal: "Tonality has not been destroyed; its boundaries have been greatly enlarged."

LOW-DISTORTION AMPLIFIER

(*from page 18*)

low-frequency signal that the speaker cannot handle.

In many cases the low-frequency coupling afforded by supplying the preamplifier plate power from the output amplifier is sufficient to make the whole system just marginally stable, or even unstable, so the power supply for this unit is self contained. This independent power supply also makes the whole unit much more versatile. The power transformer is a 40- or 50-ma shielded unit designed for use in TV boosters. This transformer is operated into a voltage doubler rectifier and an RC filter.

The photos show the appearance of the finished unit, which is built in a 7 × 12 × 3 chassis. A sub-chassis of 3/32 aluminum is fastened inside the chassis to hold the tube sockets and power supply components.

The IM distortion *vs.* volts output is shown in *Fig.* 5. This curve was taken with 5 feet of shielded cable connected to the output. Although a very long cable may be used without affecting the high-

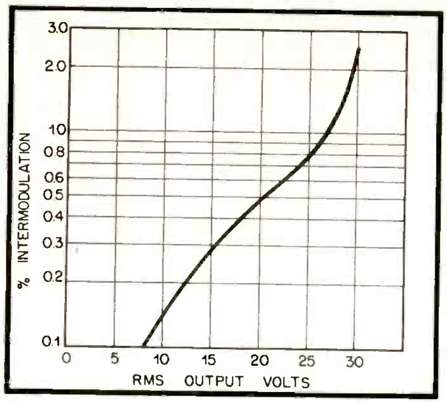


Fig. 5. IM distortion curve for outputs in excess of 8 volts rms. Distortion below this value is almost unmeasurable.

frequency response the plate current required from the output tube to produce a given high-frequency voltage is increased and with the increase in current comes an increase in distortion. Since the power of high frequencies in the usual program material is quite low the greatest difficulty in the use of a very long output cable will likely be hum voltages.

This article is intended to be a general discussion of the principles behind the design of a class of circuits and not a step by step constructional article so no dimensional drawings are included. It is thought that the amplifier pictured will fill the usual requirements found in high-fidelity home music systems and that the general principles outlined will be useful in a wider field.

BOOK REVIEW

THE ELECTRONIC MUSICAL INSTRUMENT MANUAL, second edition, by Alan Douglas. New York: Pitman Publishing Corp., 1954. 221 pages, $6.00.

Mr. Douglas has been concerned in both this and the older edition principally with theory and principle. While descriptions of commercial instruments are presented, they occupy only about 66 pages and are rather sketchy.

In the realm of principle, however, the author has furnished some excellent background information. He starts with a chapter on the nature of sound and defines the qualities and quantities of it. Next comes a chapter on music and noise which defines the scales and shows the principles of conventional mechano-acoustical tone generation—strings, reed, pipes, etc. This is followed by a brief description of ordinary polyphonic or multinote instruments.

Production and handling of electronic oscillations takes place, with interesting and informative discussions and illustrations on simple and complex wave generation from both practical and theoretical angles. A chapter on amplifiers, speakers, and tone controls gives good but standard data not particularly related to electronic musical instruments, and this is followed by the chapter on commercial instruments. Most of the principal instruments are given sufficient description to show the principles of operation.

In the last chapter Mr. Douglas speculates on possible future methods of attacking instrument design in a form calculated to stimulate readers in development work.

Employment Register

Personnel may be listed here at no charge to industry or to members of the Audio Engineering Society. For insertion in this column, brief announcements should be sent to Chairman, Employment Register Committee, P. O. Box 629, Mineola, N. Y. before the fifth of the month preceding the date of issue.

★ Positions Open • Positions Wanted

• **Audio Technician**—Likes show business. 10 years' experience as broadcast engineer with studio construction, operation, disc and tape recording, all-round maintenance. Now self-employed in recording. Would like career in recording, TV, or film. Box 801, Audio.

• **Audio Sales Engineer**—Experienced in high-fidelity component and complete installation sales. Wide experience from electronic-acoustic problems to marketing techniques. Desires executive position with progressive company. Available New York City. Make offer. Box 802, Audio.

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Industry People...

Meyer Berger, feature writer for The New York Times, has become unofficial header-upper of the hi-fi contingent among Manhattan members of The Fourth Estate; his music system and record collection are the envy of all—including even the music critics. (What greater accomplishment hath man to perform?) . . . **Charles Sinclair,** senior editor of Sponsor magazine and another hi-fi hobbyist from the field of journalism, reports many letters of approval for his satirical essay, "Hi-Fi-Manship," which appeared in the May issue of AUDIO. He is already at work on another gentle prod—one which will chide visitors to the New York Audio Fair; it is scheduled for the October issue.

Leon Wortman has been promoted by RCA to advertising and sales promotion manager for tape recorders and intermatched high-fidelity components . . . **H. S. Morris,** product sales manager for Altec Lansing Corporation, chose Chicago convention of NAMM to introduce the "Melodist," a miniature hi-fi system designed essentially as a demonstration unit for record dealers . . . **Ed Liberg,** assistant manager of electronic sales for Graybar Electric Company, New York, has been named to fill the newly-created post of manager of sound equipment sales . . . **Ed Straw** has been appointed national sales and advertising manager for Collaro record changers by Rockbar Corporation—has been associated with Collaro distribution program since its inception in 1951 . . . **Arthur E. Hafstad** is newly-elected assistant treasurer of Reeves Soundcraft Corporation—was formerly controller of General Foods Corporation plant at Houston, Tex. . . . **Gardiner G. Greene,** president of Browning Laboratories, Inc., has been chosen to head a non-profit corporation formed to conduct the first annual New England High Fidelity Music Show, slated for Boston's Hotel Touraine on October 22, 23 and 24. Other show officials include: **Lynn Eaton** of The National Company, vice-president; **Ted Jones,** manager of Station WCRB, treasurer; **Michael Scott,** manufacturers' representative, secretary. **Harry N. Reizes,** managing director of The Audio Fairs, has been engaged as consultant.

Jack Wilson, product manager of the magnetic memories division of The National Company, **Dick Gentry,** sales agent for metropolitan New York, and **Ben Ballard,** field coordinator, found great optimism among dealers toward hi-fi business this Fall during recent field trip . . . **Jules J. Bressler,** factory representative, has been appointed to handle sales for the Regency line of hi-fi equipment and television accessories in the metropolitan New York area . . .

William M. Cagney, formerly sales manager of the Link Radio Corporation, has been appointed regional supervisor for the American activities of Pye, Ltd., Cambridge, Eng. . . . **Allen J. Dusault** has taken over as sales manager for the transistor division of CBS-Hytron . . . **Anthony J. Ruscito** has joined the field engineering staff of Sprague Electric Company . . . **G. Leonard Werner,** formerly sales manager for Mark Simpson Mfg. Co., has been appointed general sales manager for The Astatic Corporation . . . **Jerry Minter,** president of Components Corporation, Denville, N. J., will announce shortly a revolutionary development in disc recording—a cutting table which entirely eliminates mechanical coupling between driving and driven units.

LEONARD RADIO, INC.
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ADVERTISING INDEX

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