audiocraft

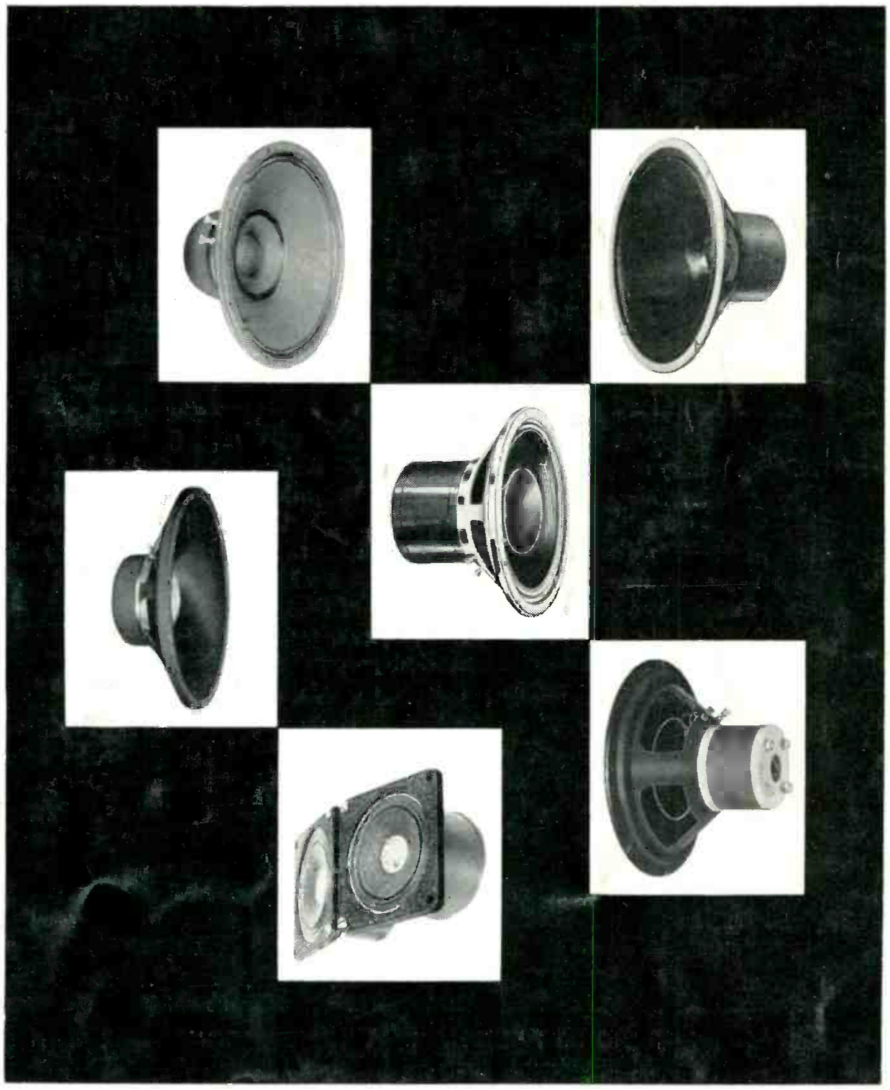
AUGUST 1956 — THE HOW-TO-DO-IT MAGAZINE OF HOME SOUND REPRODUCTION — 35 CENTS

THIS MONTH

Report on the Heath model HD-1 harmonic distortion meter kit.

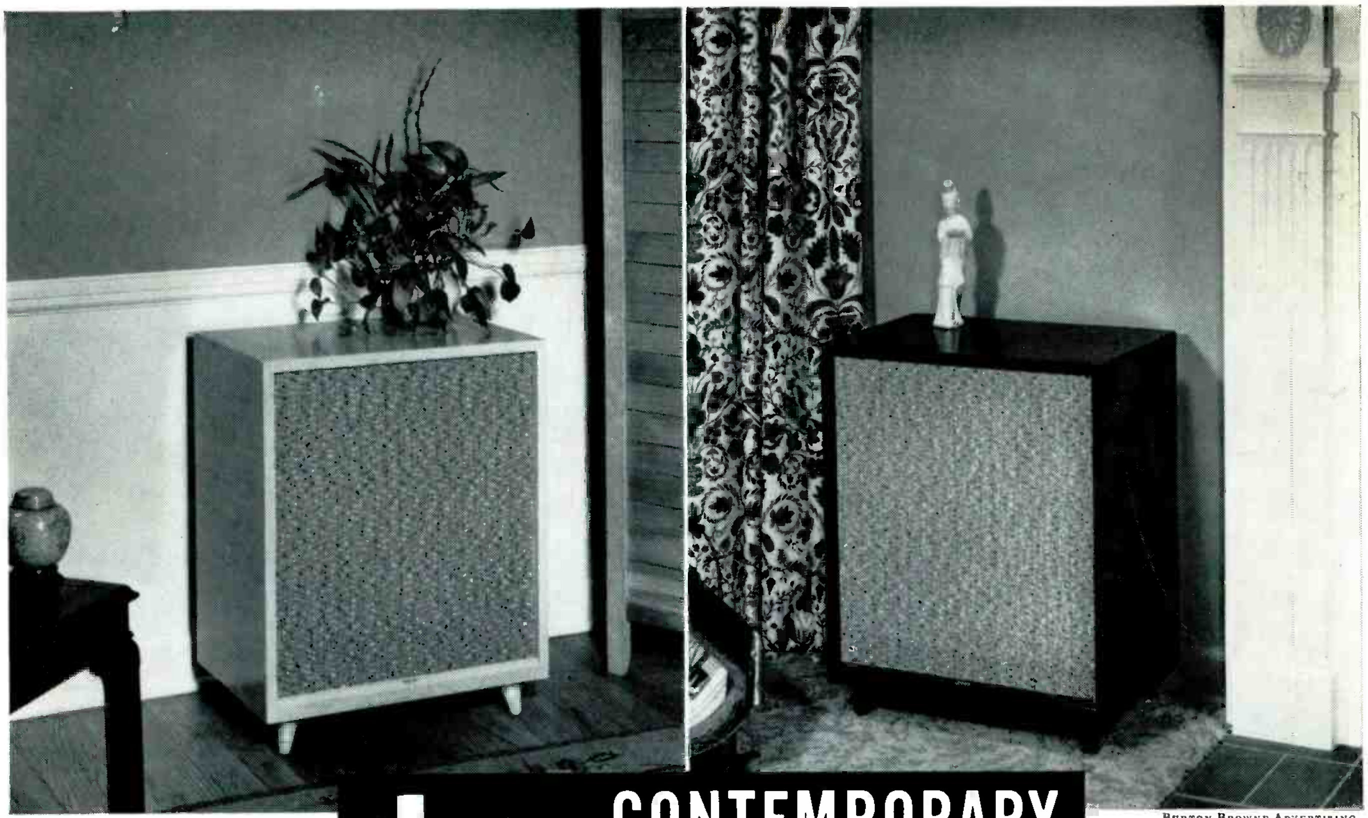
A sand-filled enclosure: sound is pure, construction simple.

How to make a power supply for the Huff TV sound amplifier.



LOUDSPEAKERS AND ENCLOSURES

Designers of loudspeakers and enclosures seldom agree on the best route to perfection in reproduced sound. This leads to great differences in construction that are explained in a new article series.



Burton Browne Advertising

Jensen CONTEMPORARY

NEW 2-WAY and 3-WAY LOUDSPEAKER SYSTEMS

These new Contemporary reproducers will thrill you with their elegant simplicity of line, executed in fine crafted genuine hardwood veneers, combined with performance which sets a new high standard in modestly dimensioned cabinetry. Neither too small nor too large for a room of any size, with classically simple lines that make them at home in a wide range of decor schemes. Available in hand rubbed Selected Mahogany or Blonde Oak. Textured cloth grille has excellent acoustical properties.

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CN-82 CONTEMPORARY 2-WAY SYSTEM

An authentic high fidelity two-way system with heavy duty 8″ woofer and compression driver horn loaded tweeter. Reproduction is especially smooth, with high response approaching the limits of audibility.

Makes an excellent modestly priced "starter" system which is easily converted later to a three-way system with the KTX-1 Range Extender Kit without cabinet modifications.

Selected Mahogany $88.75 Blonde Oak $89.75

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For those who want the ultimate response range in the Contemporary design, the three-way version is offered. Similar to the CN-82, except range from 4000 cycles to beyond limits of audibility is smoothly reproduced by means of an RP-302 Supertweeter in conjunction with an A-402 full Crossover Network. M-F and H-F balance controls in rear of unit.

Selected Mahogany $128.75 Blonde Oak $129.75

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You can build an authentic hi-fi speaker system of your own selection with this complete manual of do-it-yourself designs. Complete instructions for building free-standing or built-in single speaker and two-way and three-way speaker systems. Simplified detail drawings, parts lists, speaker data, circuit diagrams and wiring instructions.

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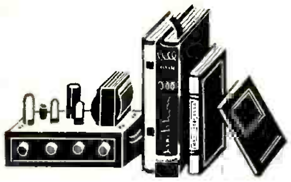

audiophile's bookshelf

Now available in this country

AUDIO HANDBOOKS

By Norman H. Crowhurst

213 **AMPLIFIERS**

Will help solve the problems that often arise when equipment has been built, but won't work. The "Trouble Tracing Guide" gives a table of symptoms, possible causes, and references to appropriate chapters which offer aid in spotting the trouble. 38 diagrams. Index. **$1.00**

214 **FEEDBACK**

Explains clearly what feedback can do, where and how to use it, and what its limitations are. The theory of closed loops is presented in easy-to-understand mathematics; design charts use a simplified approach; and the numerous examples illustrate practical applications to circuits. 40 diagrams. Index. **$1.00**

215 **THE USE OF A.F. TRANSFORMERS**

Shows in clear detail how circuit values affect a.f. transformer performance. With the aid of a detachable special ruler, the charts included make it effortless to find a circuit suitable for any a.f. transformer. 44 diagrams. Index. **$1.00**

216 **PUBLIC ADDRESS**

Covers all aspects of this rapidly expanding subject — selecting appropriate equipment, overcoming poor acoustics, and obtaining the best results for every kind of installation and program. Diagrams and half tones. Index. **$1.25**

217 **THE QUEST FOR QUALITY**

Discusses the basic requirements for fidelity and the various methods of achieving it. Shows how to design equalizers, how to build circuits for expanding dynamic range, how to check equipment. Also takes up various kinds of stereophony, and the arrangement of a listening room for the greatest pleasure. Diagrams and half tones. Index. **$1.50**

Each of these handbooks is concise in treatment, clearly written, and without involved mathematics. They are imported from England.

HIGH FIDELITY CIRCUIT DESIGN 218

By Norman H. Crowhurst and George Fletcher Cooper. Two of England's top high fidelity writers give complete information on high-quality audio circuit design procedures. The authors discuss circuit components, derive design formulas, describe testing methods and test equipment, and give examples of how to use design data for the improvement of existing audio systems. Numerous diagrams and charts. Index. **$5.95**

HOW TO SELECT AND USE YOUR TAPE RECORDER 219

By David Mark. An excellent book written for those who have little or no formal training in electronics and who either own or plan to own a tape recorder. The "Buyer's Guide" section gives specifications and prices for dozens of current models. Pictures illustrate actual setups for the many different applications of tape recorders. Illustrated. **$2.95**

HOW TO INSTALL TV ANTENNAS 162

By Samuel L. Marshall. A completely practical, illustrated "Antenna Bible". Tells you everything you need to know about installing TV antennas: safety precautions, putting up masts and towers, getting the best reception in fringe areas, etc. In short — how to do the job *right* — quickly, safely, economically. **$2.50**

HIGH FIDELITY RECORD ANNUAL, 1955 201

Edited by Roland Gelatt. Reviews of classical music and the spoken word which appeared in HIGH FIDELITY Magazine from July 1954 through June 1955. Comparisons with recordings of previous years emphasized. A performer index is given at the back of the book. **$4.95**

BUILDING YOUR RECORD LIBRARY 208

Edited by Roy H. Hoopes, Jr. Seventeen qualified experts help you to build a well-balanced record library, custom-tailored to *your* individual taste. **$3.95**

BINDERS FOR AUDIOCRAFT 223

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213	216	219
214	217	162
215	218	201
208	223	

August 1956

Volume 1 Number 10

audiocraft

THE HOW-TO-DO-IT MAGAZINE OF HOME SOUND REPRODUCTION

Authors new in this issue, in order as they appear at the right:

George L. Augspurger lives in Phoenix, Arizona, and has the odd distinction of being the only Arizona member of the Audio Engineering Society (at last count). He has done a lot of work on industrial and television sound systems, as well as home custom sound installations. His name has appeared as author of better than a dozen hi-fi articles in other magazines. Mr. Augspurger says he will answer inquiries related to his present series in AUDIOCRAFT, provided they are accompanied by stamped self-addressed envelopes.

Lee Beeder has also had some experience in writing for publication, but it has been in general-circulation magazines. His last article was on race horses; now, he informs us, he admires a thoroughbred audio system as much as a thoroughbred horse. Mr. Beeder lives in Los Angeles.

Arthur I. Zabriskie has been a Certified Public Accountant since 1940, and has a home in Provo, Utah. He describes himself as a pseudo-carpenter with all 10 fingers still attached, and a dedicated fisherman who knows the name and summer address of every trout within 100 miles of Provo. His interest in sound systems began, he claims, with an attempt to record a taxpayer's scream of horror when told the amount of the government's yearly bite.

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The Grounded Ear

by Joseph Marshall

Better Hi Fi Promotion

Hearing that in a nearby city of 50,000 population there would be a high-fidelity demonstration sponsored by the local section of the IRE, I drove down to see what would happen. I discovered some most surprising things. Although the demonstration was not publicized very widely (on the day it was held the local newspaper made no mention of it) some 500 men, women, and teen-age youngsters thronged the auditorium. The equipment was all supplied by a single manufacturers' agent; the cost was met by him except that for two vice-presidents of manufacturing firms, who gave talks. Presumably, they paid their way there. The biggest surprise to me was that, at the time this demonstration was held, there was nobody in town selling high-fidelity equipment. Probably the sight of those hundreds of people avid to learn about high fidelity spelled out an opportunity to someone, and by this time he may be prepared to start cultivating this very promising ground.

I'm not sure how worth while the audience found the demonstration. The people who produced it obviously did the best they could with the limited means at their disposal, and they deserve nothing but commendation for attempting the thing at all. There was no admission charge, so no one had a right to complain. But, although I admire the initiative of the people who put it on, I could not help thinking that here was an opportunity only fractionally utilized. Just a little more planning and preparation, and a little more money, could have produced 50 times the impact and, very likely, attracted at least five times the audience.

Some inquiries I have made since indicate that this was by no means an exceptional situation. The manufacturers' agent informed me that in his territory there are scores of cities with populations up to 500,000 in which the interest in high fidelity is virtually unexploited. He has put on other similar demonstrations, and he has received an enthusiastic response everywhere. He concedes frankly, though, that he cannot put on the kind of show singlehandedly that would exploit the possibilities in a really worthwhile fashion. I am told that the same thing is true of other areas. It is obvious to me that the industry is missing the boat badly here, and permitting the mass-production fakers of high fidelity to harvest the crop by default.

It would not take much effort or capital to provide the means to take advantage of this situation. The industry already has a sponsoring organization in the High Fidelity Institute. I, for one, think that it would be well advised to devote a larger part of its promotion to the uncultivated hinterlands, even if that would mean less for the big audio shows in the metropolises. It would require more intelligence than money; some well-thought-out planning could replace a lot of the outlay.

It is obviously impractical to try to bring full-scale audio shows to the smaller cities, if for no other reason than that the manufacturers could not afford the money and time to participate in many more shows. But it would be simple enough to set up four or five miniature traveling shows, each having a representative selection of equipment loaned by manufacturers. The equipment could be arranged and carried in packing boxes which would open out to produce the exhibit backdrops. These backdrops should include eye-catching charts, diagrams, and pictures explaining high-fidelity sound, how a system is assembled, the meaning of specifications, and so on. One section should incorporate a complete system; other sections should be devoted to classes of components — one for amplifiers; another for turntables, pickups, and cartridges; still another for control units and tuners; and one for enclosures and speakers. Finally, the exhibit should include a demonstration and lecture program on tape, which would furnish dramatic and effective demonstration of high-fidelity features and, possibly, some helpful advice on components and systems.

If the brains and resources of the industry were directed at producing these shows co-operatively, I'm sure it could be done with small investments in time and money for individual manufacturers; these shows would, probably, be far more effective than the big ones in selling the high-fidelity idea. Each unit could be allocated to one section of the country where it could be put to use (again, co-operatively) by manufacturers' agents, technical societies, and dealers. They could be moved from city to city as needed, and stored in warehouses between shows. A good show would justify a small admission charge and this would pay most of the expenses of moving and setting up the show.

Some manufacturers have been complaining that competition in the big metropolitan markets is getting too sharp and competitive. It seems to me that

And this is 20 cycles!

everyone would profit if, instead of fighting for the most saturated markets, the industry began to devote some time and attention to practically virgin territory. All it will take to open that territory up is some active promotion; I hope that, whatever it may think of the specific suggestions I have made, the industry will wake up to this opportunity soon.

Those Overcut Records

I wrote a few months ago about the recent improvement in recording techniques. Too high a proportion of records are still afflicted with a flaw that seems completely without excuse. Unfortunately, some of the best-selling labels are most guilty of it. This is a ridiculous continuance of overrecording on tape and overcutting on discs. Time after time, as I review a new record, my pleasure is diluted and often nullified by portions so heavily recorded or cut that the resulting distortion makes that part of the recording almost unbearable. Recordings of popular music continue to be worst in this respect, but a very high proportion of classical discs is also flawed.

There was some excuse for this in the days when inferior plastics produced high noise levels, and thereby reduced the available dynamic range; high-rumble turntables, low-level pickups, and low-gain equalizers required that the record output be kept as high as possible; and speaker systems, generating far more distortion than the record, masked the recorded distortion. None of this is true today, and I see no excuse whatever for recordings with audible distortion anywhere except possibly on some of the extraordinary crescendos which, unfortunately, invariably come on the innermost grooves.

To be sure, it would be impossible to guarantee complete freedom from distortion. Some pickups take peaks much better than others, and inferior or defective pickups will generate distortion on perfect records. But there are enough consumers today who have first-class equipment, and who buy a high enough proportion of records that, surely, they ought to provide the criterion market. This is especially true because perfect records will still sound pretty good on cheap equipment and will, therefore, be as good for that portion of the market as present "loud" records.

A few recording companies have proved that the slight reduction in output level and decrease in signal-to-noise ratio, which results from recording at a lower level on tape and cutting at a lower level on discs, are a small price to pay for better over-all quality. I do believe that the larger companies would find that the same procedure would produce a far higher proportion of satisfied than dissatisfied customers.

Audionews

HEATHKIT ELECTRONIC CROSSOVER KIT

The new Heathkit *Model XO-1* electronic crossover system is designed to operate ahead of the main power amplifier, instead of between the amplifier and the speakers. It consists of two independent electronic filters, one high-

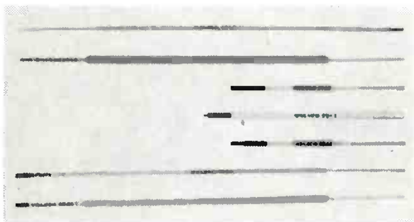
Heath variable electronic crossover unit.

pass and one low-pass, each furnished with a rotary switch for selecting the cut-off frequency. A single input is divided so that high-frequency and low-frequency portions of the spectrum are available at the outputs to feed separate amplifiers. The XO-1 is said to offer many advantages over conventional crossover systems.

Crossover frequencies are 100, 200, 400, 700, 1,200, 2,000, and 3,500 cps. Attenuation is said to be 12 db per octave with a sharp "knee" at the cut-off frequency, rather than a rounded slope as obtained with the usual RC filter.

ALIGNMENT TOOLS FOR COLOR TV

Seven new alignment tools for quicker servicing of both color and late-model black-and-white television sets have re-



Walsco alignment tool assortment.

cently been introduced by Walsco Electronics Corporation.

Among the new tools are molded-nylon, iron-core aligners carrying Walsco catalogue numbers 2541 through 2547; they are double-ended and have hex end diameters ranging from .075 to 0.125 in. for fitting all currently used slug openings. The ends of these tools are undercut on one side to enable servicemen to reach and align bottom slugs. Alignment tool No. 2541 is specially designed for TV sets using IF cans with smaller-than-standard openings. To reach difficult spots where ordinary 5- or 6-inch tools are too short, tools No. 2544 through No. 2547 were made 11 in. long to permit alignment without undesirable hand capacitance.

Further information about Walsco products is available on request.

TRANSISTOR BROCHURE

Lafayette Radio has just released a 32-page transistor brochure featuring a complete line of miniaturized parts and listing practically every transistor on the market. Pages are devoted to transistor specifications.

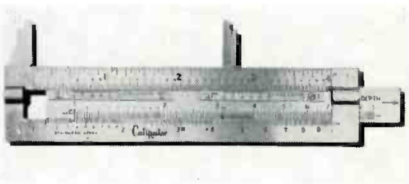
The brochure may be obtained at no cost. Request Brochure No. T4-56.

CALIPUTER

A new and improved version of the Caliputer, a precision measuring and calculating instrument, is now on the market. The instrument combines the functions of a slide rule vernier caliper, vernier depth gauge, and computer in a single unit.

One of the new features is an adjustable vernier scale which allows for easy maintenance and retention of accuracy. Purple enamel is used in the etched scale lines for better light reflection and easier reading.

The pocket-sized Caliputer is made of satin-finish type 302 stainless steel, and measures 4¾ in. long, 1½ in. wide, and ⅛ in. thick. Combining several commonly used devices into a single unit, the Caliputer provides direct readings of circumferences and cross-section areas. The slide rule will multiply, divide, find squares and square



Combined slide rule and scale.

roots. A machinist, for instance, can read directly the correct lathe speed setting in rpm by setting the rule to the diameter of stock used and by supplying the proper feed rate.

Caliputer readings are accurate to a thousandth of an inch, according to the manufacturer. The instrument retails at $8.95.

NEW TAPAK RECORDERS

Three new 1956-model, spring-powered walkie tape recorders are offered by Broadcast Equipment Specialties Corp. The Duplex model provides not only broadcast-quality recording but also, instead of only the usual earphone play-

New completely-portable tape recorders.

back, a built-in loudspeaker for reproduction to room-size audiences. A subminiature basic amplifier and transistorized bias oscillator make possible this advance without sacrificing battery life. Another model incorporating the addition of a VU meter and 600-ohm zero-

FOR MORE INFORMATION

For more information about any of the products mentioned in Audionews, we suggest that you make use of the Product Information Cards bound in at the back of the magazine. Simply fill out the card, giving the name of the product in which you're interested, the manufacturer's name, and the page reference. Be sure to put down your name and address too. Send the cards to us and we'll send them along to the manufacturers. Use this service; save postage and the trouble of making individual inquiries to a number of different addresses.

the Triplex. An economy model, the Simplex, consists of the basic recorder/reproducer without loudspeaker.

PELL-I-CAN OILER

Pell-i-can, manufactured by the Lofthouse Company of Binghamton, N.Y., is a pen-sized oiler with a long, narrow

Pocket-clip oiler for precision machines.

"beak", to fill small out-of-the-way oil holes in movie cameras and projectors, tape recorders, and other appliances.

The oiler features a visible oil supply and a handy pocket clip. It avoids spilling by delivering its oil one drop at a time. The Pell-i-can is said to be leakproof and unbreakable in normal use. It is priced at $1.49.

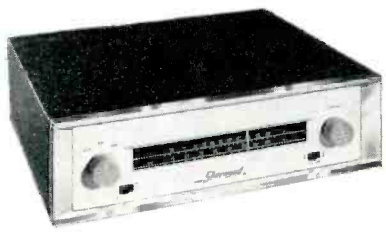
SHERWOOD AM-FM TUNER

A new AM-FM tuner, the *Model S-2000*, has been announced by Sherwood Electronic Laboratories.

This tuner, according to the manufacturer, has FM sensitivity of 1.2 μv for 20 db quieting. Automatic frequency control with 16 db correction is said to simplify tuning and to hold stations accurately in tune.

Special attention has been given to the AM section to assure fidelity. There is a choice of a 16-Kc wide "hi-fi" band-pass or a selective 5 Kc band-width for listening to weaker stations with minimum interference.

The Model S-2000 has flywheel tuning and a directable ferrite rod AM



Sherwood makes this AM-FM tuner.

antenna. The unit is compact, measuring 14 in. by 10½ in. by 4 in. high. The tuner is available in a variety of cabinets to match the Sherwood S-1000 amplifier. These include a black and gold stippled finish and gold-tooled mahogany, tan, black, or white leatherette.

FREE BOOKLET ABOUT SANDPAPER

A new, 16-page, pocket-size booklet offers help to the home craftsman in choosing and correctly using the proper sandpaper for finishing wood, metals, and plastics.

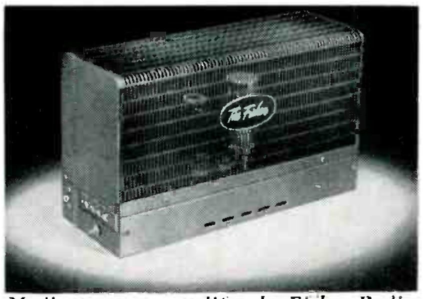
Titled *Sandpaper — How To Choose It*, the booklet identifies the five kinds of grain available and describes their properties. Useful tables recommend sandpapers and grit sizes for floor sanding, and for hand and machine sanding or polishing. Straightforward text is illustrated with more than a score of photographs and drawings. The booklet is published by Behr-Manning and is free on request.

FISHER STANDARD AMPLIFIER

Fisher Radio Corporation has announced the addition of a Standard Amplifier, *Model 20-A*, to its line of high-fidelity components.

The output of the Standard Amplifier is said to be constant within 1 db from 15 to 30,000 cps. It has less than 0.7% distortion at 15 watts and a hum and noise level 90 db below full rated output, according to the manufacturer.

The Model 20-A can be used with any preamplifier-equalizer and, in addition, is designed to operate with com-



Medium-power amplifier by Fisher Radio.

ponents that lack their own power supply. An octal socket provides all necessary AC and DC operating voltages, plus connections for switching the amplifier on and off from a remote position. Output impedances are 4, 8, and 16 ohms.

The Model 20-A is fully shielded and has a decorative metal cage for protection and ventilation. It is priced at $59.50 (slightly higher in the West).

HIGH-QUALITY SPEAKER SYSTEM

The *Premiere*, a high-quality, folded-corner-horn speaker system, has recently been introduced by United Speaker Systems.

Frequencies between 30 and 800 cps are reproduced by a 15-inch low-frequency driver coupled to a dual-throat folded horn with an approximate volume of 9 cu. ft. Frequencies between 800 and 22,000 cps are handled by a heavy compression driver mounted to a large, rigid sectoral horn. Both speakers were designed specifically for theater use.

Crossover at 800 cps is accomplished by a specially designed, 12-db, constant-resistance, parallel-type network using air-core coils and oil-filled capacitors.

Enclosures are 39 in. high, 33 in. wide, and 28¼ in. deep, and are custom

The Premiere two-way corner horn system.

built from genuine walnut, mahogany, and korina woods. Grille cloths are made from loosely woven decorator fabrics and are available in a variety of shades.

Additional specifications and other information about the Premiere will be furnished on request.

BASE FOR RONDINE TURNTABLES

A high-fashion turntable base styled to harmonize with either modern or traditional decor has been announced by the Rek-O-Kut Company, creator of the Rondine turntables and turntable arms.

Specifically designed to accommodate the Rondine or Rondine de Luxe turntables, the Rondine base is available in selected-stock American walnut or natural korina, hand rubbed to a satin finish. Choice ⅞ in. thick solid woods are used throughout. The turntable deck fits

Turntable base is of finished hardwood.

on a built-in ledge and can be mounted in less than five minutes with only a screw driver.

The base rests on rubber-ball feet which absorb external shock and vibration. The feet are also adjustable for leveling.

Dimensions of the Rondine Base are 16½ in. wide by 17 in. deep by 6 in. high. The price is $26.95.

TIPS FOR THE WOODCRAFTER

by George Bowe

The Router

For versatility in a power tool, the portable router is hard to beat. It's all in a day's work for a router to cut rabbets, dadoes, and grooves; dovetail, mortise-and-tenon, spline, and tongue-and-groove joints; decorative edges, scrollwork, wood inlays, carving, fluting, and beading. If there is time still left in the day, there are other chores the router can perform — provided the operator is up to it. Not only is it one of the safest

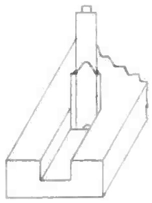
COURTESY THE STANLEY WORKS

Fig. 1. Basic portable electric router.

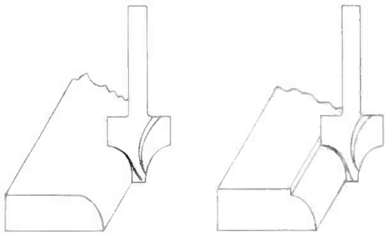
power tools to use, but it is also a tremendous work saver, turning tedious hand jobs into pleasant operations and producing topnotch craftsmanship.

The portable router is simply a small, powerful motor with a chuck attached to the spindle for holding the various router bits which do the cutting. The motor housing screws into a base with two hand knobs for guiding the router through the work (Fig. 1). A guide which fastens to the side of the base is designed for use in making cuts along straight and curved edges. Router bits are numerous and varied in shape to perform specific work on the surface or the edge of the stock; the side of a bit, rather than the end, provides the cutting edge. Here are some bits that will find frequent use in general cabinet construction:

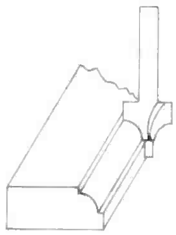
The *Straight Bit* is used for routing out hte background of a carving, for routing strip inlays, and for making rabbets,



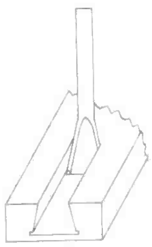
Straight Bit



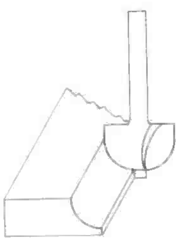
Rounding-Over Bit



Beading Bit



Dovetail Bit



Cove Bit

grooves, dadoes, and mortises. This bit must be used with a guide.

The *Rounding-Over Bit* is excellent to round the edges of any wood surface. The pilot tip on the bottom of the cutter acts as a guide along the edge of the work. Note that the same bit can be used to cut varied shapes by raising or lowering the cutter to the edge of the work.

The *Beading Bit* is designed to make corner beads and to cut beading along the edges of furniture. The pilot tip makes it easy to follow irregular shapes.

The *Dovetail Bit*, as the name implies, cuts dovetail joints and is used with a special dovetail jig. Dovetail joints are used especially in drawer construction.

The *Cove Bit* is used for cove cuts on the leaf of a drop leaf table, on table tops, and panel rails.

Now let's see how some of these bits can be put to work to make several of the most frequently used cuts. The procedure for cutting a groove (a channel cut *with* the grain of the wood) and a dado (a channel cut *across* the grain) is identical.

Groove and Dado

1. Fasten a straight router bit in the chuck of the router. Choice of the bit

COURTESY THE STANLEY WORKS

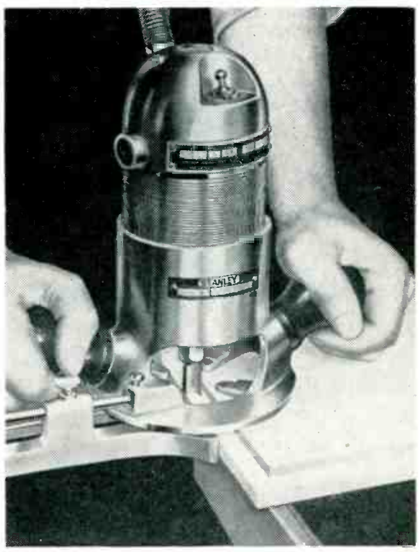


Fig. 2. Rabbetting with a straight bit.

depends on the width of the groove or dado.

2. Screw the motor into the base until the bit is set at the desired depth.

3. Lock the motor to the base by

means of the screw attached to the base.

4. Attach a guide to the dase anb set it to position the cut the correct distance from the edge of the work.

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Fig. 3. Router and dovetail attachments.

5. Start on one side or the end and move the router to make the cut (Fig. 2).

For cutting decorative edges there are many shapes of cutters, most of which use a pilot tip which does not cut but rides the edge of the uncut wood. Here is a routine to follow when shaping edges with a portable router:

Decorative Edges

1. Attach the desired bit in the chuck.
2. Adjust for correct depth by threading the motor into the base, then tighten the lock screw.
3. Make a test cut by running the router through a piece of scrap wood of the same thickness as the stock to be worked. If the cut is not the one you wish, it can be changed by raising or lowering the motor.
4. When the correct cut is set, proceed with the regular stock, holding the router base firmly against the top of the wood and moving the cutter into the edge.

To the amateur craftsman who dreads making drawers the portable router is a blessing, for it can perform all the necessary cuts after the initial sawing. It is particularly appreciated for its precision dovetailing of either flush or lipped drawer fronts. This is accomplished with a dovetail bit, a special dovetail jig, and a template guide tip attached to the base of the router. Simultaneously, the two pieces of wood to be joined are given a dovetail cut; one piece is cut with dovetail pins and the other with the dovetail sockets to receive them (Fig. 3). Follow the manufacturer's directions exactly for a perfect result. Many are the other functions that the hand router can perform, but, regardless of the specific task, here are some tips that apply generally to its use:

1. Be sure the plug is disconnected when changing bits. The switch on the router could be turned on accidentally.
2. Guide the cutting edge of the bit into the wood so that it "bites" its way into the work. The opposite way would push the cutter away from the stock.

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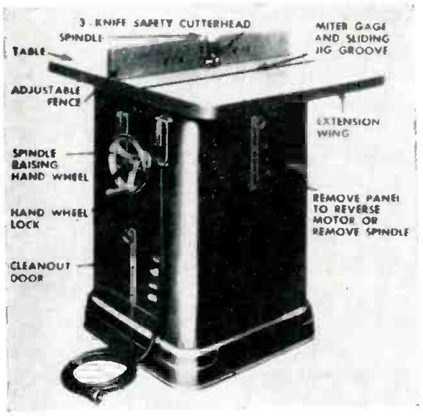

Fig. 4. A shaper; parts are identified.

3. Always check the bit adjustment by a test cut on a piece of scrap wood.
4. Don't force the router nor guide it too slowly. Experience will soon tell you the correct speed.
5. Always start by cutting across the grain, then cut with the grain. In this way, if the end grain splinters, the final

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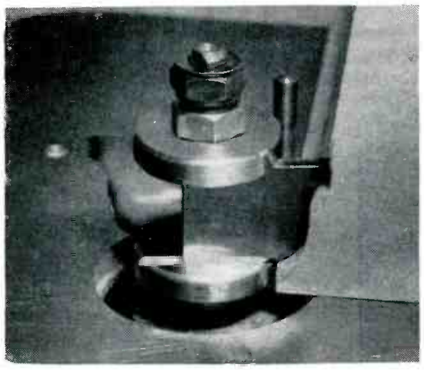

Fig. 5. Clamp-type cutter for a shaper.

cut *with* the grain will dig slightly past the splintered corner and leave a clean finish.

The Shaper

A fixed machine very similar to the router in output is the shaper (Fig. 4). Actually it is a router turned upside down with the spindle projecting above a metal table. The shaper is used for cutting moldings, edge designs, various types of joints, grooving, fluting, and reeding. If used correctly, the shaper can be one of the safest power tools in your shop. However, if carelessness prevails this machine becomes a dangerous piece of equipment. It's a tool to treat with respect. Like the router, the shaper will provide an added touch of craftsmanship to the job at hand. The size of a shaper is determined by the diameter of the spindle and by the size of the table. While industrial-type shapers frequently have two spindles, the average home-workshop model does very nicely with a single spindle. With the spindle operating at speeds between 5,000 and 10,000 rpm, it's easy to understand why the cutters produce such smooth work. Usually the spindle can be raised and lowered to make adjustments. To guide the work through the cutters, an adjustable fence is fastened to the table which is also grooved to take a miter gauge. In normal use the shaper should operate in a counterclockwise direction with the work being fed from right to left. In this fashion the cutter blade grips the wood as it cuts into it and does not throw it outward from the spindle. For jobs requiring the shaper to operate in a clockwise direction, the situation is reversed and the work is fed from left to right.

There are two types of cutters which a shaper uses: the *clamp-type* with removable blades (Fig. 5) and the *formed-type* which is a solid unit that fits over the spindle (Fig. 6). The formed-type cutters are much the safer since the two separate blades of the clamp-type can become loose in operation.

For his own protection, a beginner should confine his work to jobs requiring the use of the fence and the guard. When shaping with a fence, first loosen the spindle lock and adjust the spindle to the proper exposed length; then lock the spindle in position. When starting the machine stand slightly to one side, never directly in back of the cutter. Hold the work firmly against the fence and table with the left hand, and feed the work slowly into the revolving cutter, with the right hand applying the forward pressure. Never allow the fingers to get near the cutter. Never attempt to shape a piece of wood shorter than 8 in. without using a temporary jig to hold the wood at a safe distance while the edge is being shaped.

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DELTA POWER TOOL DIV

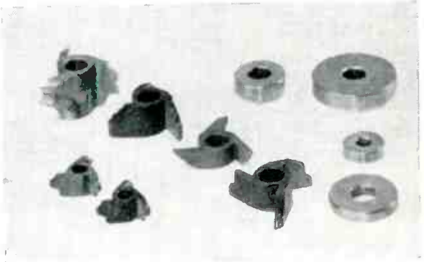


Fig. 6. Several formed shaper cutters.

Caution is the key word with all shaper operations. It is an excellent piece of equipment designed to make your work easier and better. Operate it correctly and it will serve you safely.

SOUND SERVICING

by Irving M. Fried

Loudspeakers, Part 3

Now that you have smoothed out your woofer and your tweeter (and your "squawker" too, if need be) you can begin to re-evaluate these improved drivers as they work together. One of the important things in getting them to work toward the common objective of reproducing better music is to get them in better phase with each other.

Everyone who has ever heard about a two-way system is interested in the problem of phasing, of getting the units to work together with the least amount of clashing. There is a school of purists which insists that a two-way system is always out of phase, no matter what you do with it. And there are those of tangentially opposite view point who insist that, in a living room, the ear is completely unable to detect any speaker phase differences. But we will avoid this controversy for the moment, and discuss methods of phasing.

Phasing is both an electrical and a mechanical problem, the two quite interrelated. If two drivers working in the same frequency range operate in unison, theory says no one will be conscious of crossover problems. Since one driver is lagging at the selected frequency (because of phase shift in the crossover network), and the other driver is ahead, the crossover itself creates a phase problem. After the electrical phasing, then, it is sometimes necessary to push one unit back and forth in relation to the other, so that sound waves from the sources are propagated into the room (or the greater portion of it) evenly.

Therefore, you should get the units into electrical phase first. This is easy if you can watch the voice coils or diaphragms of the two units (i.e., if you are using cone speakers for both ranges). Using a low-voltage DC source such as a flashlight battery, put a signal into the crossover network and watch the two cones. They are electrically in phase when both move in the same direction when you apply the battery.

Most manufacturers mark their own units for proper electrical phasing when used together. Yet I have found many factory-assembled systems hooked up backwards, inadvertently I am sure. Even if you buy a completely wired speaker system it is a good idea to check its phasing.

If you aren't able to tell by this method whether your units are phased electrically, you can use the methods following, all of which pertain more directly to mechanical phasing. In each case, you will find that proper electrical phasing will create a dramatic difference, after which mechanical phasing will be a refining process.

Expert listeners often prefer the listening method of phasing, as follows: A system in phase will have a smoother, sweeter, more integrated sound, with less masking of typical solo voices of the orchestra, such as cello, oboe, and clarinet, and of the human voice. I have often used the listening method, in living rooms, for final analysis and setup. I prefer to use a human voice, singing. If the crossover is in the 300 to 500-cps region, I use a baritone; if the crossover is between 800 and 1,200 cps, I use a contralto or soprano; if the crossover is well up, I use the highest soprano voice I can find in the collection.

Listen for continuity of sound, for absence of in-and-out effects as the voice moves up and down the register. Listen also for sweetness, and for a sense of smoothness and integration. As you listen, move the high-frequency unit back and forth, until you find a position at which everything seems most nearly to be in proper perspective (or, if you haven't adjusted the electrical phasing yet, listen, while reversing it, for smoothest effect).

If you just don't trust your ears, or aren't quite sure what to listen for, you will need an audio oscillator, a microphone (quality not too important), and an output meter (an ordinary VTVM will do, since you are interested only in relative rather than absolute loudness measurements). Sweep the crossover region for an octave on each side (if the crossover is 500 cps, this means between 250 and 1,000 cps); set up the microphone in various room positions. Choose as final that combination of electrical and mechanical phasing which produces the highest average readings. At that combination, of course, you are getting the least interference, and the highest amount of energy into the room.

Alternatively, you may want to try the theater method of phasing. For this you will need a small additional speaker, which you suspend half way between the woofer and the tweeter, as close to each as is possible. Choose on the oscillator the crossover frequency, and feed this into the small speaker. Measure the output of the crossover network (after you have disconnected the amplifier from the input terminals of the network you put the voltmeter across these terminals) while phasing electrically and mechanically. When the output voltage is maximum, the speakers are in best phase.

That should take care of phasing *per se*. But what can you do if your system still seems to be divided, if the drivers won't blend as they should?

You may find that increasing the rate of cutoff in the crossover network will help. Because of the complicated nature of the operation, I can only suggest that you purchase a standard unit with higher cutoff rates. This seems to help most when the woofer and the tweeter have varying types of propagation — if one is a horn and one a direct radiator.

Sometimes using a slower rate of crossover will help. More overlap helps

to fill in "holes" in the response; this measure is likely to be more successful when the drivers are compatible (both either horns or direct radiators).

Biamplifier driving systems, with electronic crossover filters, may help, particularly if the drivers have erratic impedance characteristics. An amplifier connected directly to the voice coil tends to smooth out the over-all effect.

Other possible helps: the ear is always less conscious of crossover phasing problems if the high-frequency unit is placed vertically over the low-frequency unit. The best (not necessarily the most practical) design for a four-way system would be each unit stacked close to and above the driver just below it in frequency range.

The ears may be fooled if the units can be deflected away from the line of sight. Get the offending units above or below ear level. Sometimes you can minimize phase problems, and get better blending, by directing the middle-range and high-frequency units toward a reflecting surface, such as the corner and/or the ceiling. If you use reflecting surfaces to break up direct beams, you will be less disturbed by tonal discontinuities.

AUDIO AIDS

We'll pay $5.00 or more for usable Audio Aids. *See page 33 for details.*

TAPE NEWS & VIEWS

by J. Gordon Holt

HARTFORD, Connecticut, is the site of the annual spring choral festival of New England secondary schools, which each year involves a chorus of about 400 voices, several soloists, and a sizable orchestra. Months of preparation go into the final single performance that is held in Bushnell Memorial Hall. For the past three years the Publishing House has been asked to record the performance; the sponsors then have records made for the participants.

As a recording venture, the Hartford Music Festival is both a challenge and a source of incipient ulcers: the former because it isn't very often that amateur recordists get the opportunity to take a crack at a group as large or as fine as this one, and the latter because the whole project is done with the gnawing realization that a single mishap anywhere along the line will mean a complete recording failure. There is only one performance, so this means it must be successfully committed to tape the first time; there isn't any second chance, or any opportunity to do retakes that could later be spliced in to make up a complete recording.

What could go wrong? Well, starting at the top of the list, any one of the several hundred components in the recording equipment could give out at a crucial moment. Then there might be trouble with a bad roll of tape producing drop-outs during the recording, or a cable connector might decide that it had had enough flexing and twisting and begin acting up, making and breaking one of the input circuits. Or, perhaps, the microphone placement might not sound as good through a loudspeaker as it did through the headphones, so the entire recording could have some gross aberration in balance or liveness or timbre.

We are devoting this month's column to a description of how we handled the job — not because we believe it to be the best possible way, but because it represented a fairly unique problem and the details may be helpful to others faced with unusual recording circumstances. As a matter of fact, two previous efforts to record the Hartford Festival were dogged with equipment failures, and it was only through the ingenuity and quick thinking of the recording teams that successful recordings were made. On both occasions two recorders were run simultaneously, just in case, and the case arose each time. The final tapes turned out to be half-and-half jobs, with half the program recorded on one recorder and the other half on the second recorder, then dubbed together. The records that were finally made attested to the success of this devious production, but the headaches involved were sufficiently severe to warrant planning this year's session with ultimate dependability as the first requirement.

Accordingly, we planned to use two recorders and two sets of microphones, with both machines kept running throughout the entire performance as constant insurance against each other's defection. Previous single-miked recordings had been quite successful in picking up the soloists from the front of the group, without the use of a solo microphone, but the initial choice of single-mike pickup may have been influenced by the fact that no one could dig up an input mixer in time for the performances. This time a mixer was at hand*, so there was at least a choice of single or multi-miking.

A few phone calls to Philadelphia brought an agreement with William G. Phillips (of the Dyna Company) for him to supply a full-track Ampex 401 recorder, an Altec condenser microphone, a lot of interconnecting cable, a pair of Permoflux dynamic headphones, and his own capable services. To this aggregation I was able to add a half-track Ampex 600, another Altec 21B microphone (borrowed from our publisher), a Telefunken condenser mike, a pair of Brush bass-boost headphones, and more interconnecting cable.

The last frenzied moments before the 29th were spent checking out the recorders, demagnetizing heads, and organizing the contents of the gadget cases that contained cables, headphones, gooseneck extensions, and all the other bits of addenda that a recordist carries around because he might need them sometime.

First stop in Hartford was at radio station WTIC, where we had arranged to borrow two mammoth microphone stands that would get the mikes high over the performers without having to resort to sky hooks in the auditorium. Then on to Bushnell, where the next half hour was spent wandering around gazing in awe at the size of the place and trying to figure out how we could best cover the huge performing group with a paucity of microphones. The performers were to be arranged in the conventional large-group manner, with the orchestra at the front of the stage and the chorus seated in tiers at the rear. There were four soloists located at the right of the conductor. The entire program was to be Mendelssohn's *Elijah*, which requires a big, full sound from the chorus and orchestra, and a slightly more intimate pickup of soloists.

Eventually, we decided to use one of the Altecs at the front and center of the stage, to cover the entire group, while the Telefunken was set for cardioid pickup pattern and placed in front of the soloists. It was also directed slightly away from the orchestra to minimize pickup of anything other than the soloists themselves.

During the afternoon rehearsal, we ran off a few test recordings, and wandered on and off stage pushing microphones around and raising and lowering stands with what must have appeared to the performers to be a rather off-hand approach to this serious undertaking. As a matter of fact, we were thrown into a temporary fit of indecision by the fact that our off-stage recording "studio" was separated from the orchestra only by a heavy curtain, so we were hearing about half the signal through the headphones and half directly from the performers. Since the cellos were on our side of the stage we kept hearing them above everything else, and it took a lot of fiddling before we finally got what struck us (through the headphones) as being pretty good balance and a nice sense of openness to the sound.

The final setup had the Ampex 401 being fed by the Telefunken and one Altec, with the mixer between to allow

*Holt, J. Gordon, "A High-Quality Microphone Mixer", AUDIOCRAFT, December 1955.

individual control of the microphones. The Ampex 600, then, was used with the second Altec, in the hope that if something went wrong with the main system we could get adequate pickup of the entire group with the single mike, as in preceding years.

We had taken the opportunity to note volume-control settings during the rehearsal, so there was very little adjustment needed for the performance. The level was set so that it was below the tape overload point on the loudest passages, and no further manipulations were made. The recording, then, came out with the entire volume range of the chorus and orchestra on it, and the hiss level from the recorders was low enough that the signal did not disappear at any time.

The actual performance started out on the wrong foot when the soloists began to address their voices to the audience. We had apparently failed to notice that during the rehearsal they had faced the conductor at all times, and when we found them coming in feebly we had compensated for it by turning the Telefunken around and raising its level slightly. Then came the performance, the soloists turned away from the microphone and toward the audience, and the solo mike placement was wrong. Adjusting the level of the solo mike helped, but the voices still had a certain degree of remoteness. As it turned out — fortunately — this sounded quite good on the final tape.

From that point on, it was smooth sailing until after the intermission. Then the functional troubles started. Shortly into the opening section of the second half of the program, the 401 started acting up. Any Ampex recorder has an A-B monitor switch which gives an instantaneous comparison between what is coming into the recorder and what is actually going on the tape. Flipping the switch this time showed that the input signal was fine, but the output from the tape was appallingly lacking in highs, and its volume had dropped to practically nothing. It didn't sound like electrical trouble, since even the hiss level from the tape seemed to be varying, and the over-all level fluctuated from time to time.

Then, amid much tearing of hair and unreportable editorial comment, the dropout or whatever it was suddenly disappeared. Everything seemed to indicate that something had lodged on one of the head surfaces, lifting the tape clear of the pole pieces at that point, but its seriousness would depend upon whether it had lodged on the record or playback head. Either one would have given us the kind of sound we were getting on playback — but if it had been on the playback head, we would still have a good tape. If it was on the record head, though, the tape would be ruined at that point. And we had no way of knowing which was the case until we got around to playing the tape afterwards. Meanwhile, we just let the recorder run for the rest of the performance anyway, hoping against hope that we would be able to use the tape. Then the Ampex 600 ran out of tape, 5 minutes short of the end of the performance, so *that* tape was incomplete.

We didn't listen to the tapes until the next day, and if we had had any optimism about the 401 drop-out, it was dispelled then. A 45-second period during a soprano and orchestra section was hopelessly muffled and low in volume, and two other short spots which we had overlooked before were similarly ruined. The recording was otherwise very good, albeit a little too closely-miked.

The 600's tape turned out to be somewhat less than good. It seems that, despite preliminary spot checks, the bias current on that recorder had been set a

little too low, so the high end on the tapes was slightly tipped up, accentuating sibilants from the voices and adding a steely edge to the strings; and the whole thing was slightly muddy because of the low bias current. Also, the microphone placement had lost a great deal of projection from the soloists, so they tended to blend *too* well with the chorus and orchestra, and they also sounded too far off-mike and much too echoey.

It was felt, however, that we *could* use the 600's take of the solo number that got lost on the 401, by dubbing a full-track copy from the half-track 600 tape onto the 401. I was about to go ahead and do this when a stroke of dumb luck changed the picture.

I had been using the 600 for editing, and was playing the 401's tape on it when I came to the place where the drop-out occurred. But this time, instead of muffling and a loss of volume, the signal disappeared completely; there was only silence, with a barely detectable mutter of something indefinable in the background. It took a minute for the significance of this to sink in, but finally the light dawned: the particle of dirt that had lodged on the record head had been at one *edge* of the tape, so the other edge might still be all right. I switched the reels on the 600 and ran the full-track tape backward through the section where the drop-out had occurred. We were in luck. That side of the tape was perfect, and not a trace of high-frequency or volume loss was evident.

The remainder of the restoration was easy. The tape was switched around again so that it was playing in the right direction, and was run through the section immediately preceding the drop-out, where there were two powerful crescendo passages, and the volume indication on the 600 was carefully noted. Then the Ampex 401 was fed by the output from the 600, and its record level was set to give the same reading on the same two passages.

The tape on the 600 was then run through again in the forward direction. A careful stop-watch timing was made of the drop-out period, and then the tape was allowed to run for exactly 30 seconds past the point where the signal came back in again. Then the reels on the 600 were switched around again, the 401 was loaded with blank tape, and the stop watch and 600 were started simultaneously, with the tape this time playing backwards. After 20 seconds (which was about the time needed to walk leisurely across the room and start the 401 recording), the 401's record level was brought up to its pre-determined setting and a full-track duplicate of the reversed tape was run off. Figuring a full minute of recording time from then, to allow plenty of leeway at the end of the dead section so it could be easily spliced into the original, the recorders were stopped; I had a full-track copy in reverse of the good side of the dropped-out tape, recorded at the same level as the original.

It was then a simple matter to reverse the duplicated length of tape, and splice it in to replace the defective length of the original. Repeating the same procedure for the two other brief drop-outs remedied them, too.

The only thing that remained to do now was to edit the tape for disc mastering. This meant eliminating as many as possible of the distracting sounds that occur between sections of a musical performance, and the "pacing" of pauses between sections so the music would maintain its logical flow and organization.

This was initially a somewhat hit-and-miss proposition, and became a matter of inserting a six-second period of silence between two sections and then listening to it to see whether the pause seemed too short or painfully long. It finally turned out that five seconds seemed appropriate between changes in the mood of the music, and periods as short as one second for places where the composer had indicated a pause in the score only to give the performers time to turn pages, sit down, or tune their violins.

This business of inserting quiet pauses in the program turned out to be a more difficult undertaking than I had expected. We have no idea how much noise there is in a "quiet" auditorium until we try splicing from it into a stretch of erased tape. Even during those rare periods of silence when the audience is hushed in expectation of the opening chord, and

Continued on page 35

Readers' Forum

Gentlemen:
Have just read page 3 of your May issue. I particularly refer to the statement: "His development of the Klipschorn—the first speaker system employing a folded bass horn that utilized a room corner as a horn extension — marked a turning point in home sound reproduction."

The speaker system that utilizes the walls of a room corner as part of a horn or baffle was patented by Mr. Maximilian Weil, Patent No. 1,820,996 (1931).

About a year ago, this matter brought on quite a discussion in one of the electronic magazines and, finally, it was Mr. Klipsch himself who wrote to the publication: "My first paper on corner horns cites 'Weil, U.S. Patent 1,820,996 (1931)', and I might add this patent was applied for in 1925."

Also, your editorial in the May issue speaks of live and recorded sound compared directly. At the New York Audio Fair, and at a repeat performance in Los Angeles, back in 1952, Mr. Weil had artists perform in person. Several times during the performance the artists and the recorded version alternated in rendering the composition. This detail is mentioned because in years past live talent had been used to sing, for example, *simultaneously* with the recorded version. Under such conditions, the artist's rendition easily covered up the "crime" of the recorded version.

G.V. Sullivan
Vice President
Audak Company
New York, N.Y.

Gentlemen:
My thanks to Joseph Marshall for his honest comment on my future activities in speaker design (AUDIOCRAFT, May 1956, "Grounded Ear"). I should like to think that I am wrong in supposing a large cone is necessary for commercial success, but experience suggests I am not. By good fortune three experts come together in one issue of AUDIOCRAFT: Joe Marshall, Irving Fried, and Dave Hafler. I know and like them all, perhaps because their opinions can't be bought. Fried runs a very good audio store in Philadelphia and his stated personal preference is for the Hartley 215; but he has told me bluntly that he can't sell it against 12- and 15-inch speakers which to his ears aren't any better. He and Hafler have measured lots of speakers and they tell me the 215 has the flattest impedance curve they

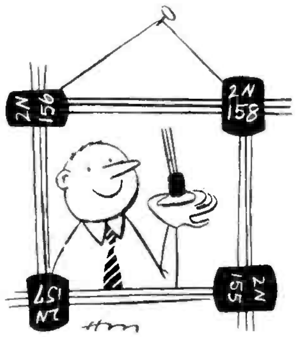
Continued on page 39

EDITORIAL

On a very warm day in the middle of June, which is when this is being written, it can be difficult for a writer of editorials to find something vital, interesting, and timely to discuss. It may not be quite proper to admit this, but it is so nevertheless.

At this season an editor is most concerned about procuring material for the fall and winter issues. Fall brings with its bracing temperatures increased hi-fi activity on everyone's part; home listeners and equipment constructors, back from vacations and weekends at the beach, spend more time indoors building and listening; manufacturers excitedly announce their newest models which, somehow, manage to be even better than last year's ultimate versions; audio shows are held in the big cities, one immediately following another; advertising jumps sharply, and so does the size of audio magazines.

We are so pleased with the results of our efforts to obtain especially good material for the busiest seasons that we



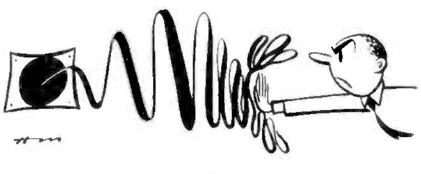
think you might like to know about it. First, and probably of greatest general interest, will be a series of articles by Paul Penfield, Jr., on the design of transistor circuits. He will begin with an explanation of how transistors work, continue with a comprehensive discussion of their operating characteristics, describe power-supply and bias requirements, impedance matching and mismatching, distortion and frequency-response performance, and the procedure of circuit design for transistors. This will be on a completely practical level; mathematics will be limited to simple algebra, although all information necessary for successful transistor use will be given. The aim of the series is to make readers as familiar with transistors, and as comfortable working with them, as they now are with vacuum tubes. It will be the first time (to our knowledge) that this has been done in any magazine, and we expect it will be a classic series. The first article will appear in October.

Our kit reports have turned out to be among the most well-received of all articles, and we offer thanks to all who've told us so. Perhaps we were too exhilarated upon being told so enthusiastically; perhaps it represented a challenge we couldn't pass up; or maybe we were caught in a lightheaded moment. Anyway, whatever the reason, we have agreed to do a kit report on an Artisan electronic organ kit. Anyone who has looked inside an electronic organ knows that this will be no small undertaking. But it should be a fascinating job, and we hope it will make interesting reading. Look for it in the late fall.

There are at least a dozen reports scheduled or in process of preparation on tuners, test instruments, amplifiers and preamplifiers, and kit loudspeaker systems and enclosures. Reports on the Triad HF-12 preamp/amplifier and the Sherwood Forester speaker system, for instance, will be in the September issue.

Philip C. Geraci, of the University of Maryland's Photographic Section, spent several hours and about $80 on modifications of his Pentron PMD-1 tape recorder. He says that it will now outperform recorders in the $500 class, and has certified laboratory test results to prove it. The modifications are generally adaptable to other low-cost recorders; Mr. Geraci will describe them in another article tentatively scheduled for November.

One reason a new high-fidelity enthusiast so often grows dissatisfied with his first sound system, we suspect, is that when selecting the equipment he didn't know what to listen for. It is unfortunate that, by the time he learns through experience, it is too late. The sound system that sounds most impressive and exciting in the dealer's showroom, alas, may become irritating and intolerable after being lived with for



some time. And because sound-memory is so unreliable, the man who attends performances of live music rarely or only occasionally has no standard of comparison. Joseph Marshall tries to give him one in a three-part article about judging hi-fi quality by listening — a masterful job of writing on a difficult subject, which we hope to be able to begin in the September issue. — R.A.

by George L. Augspurger

and LOUDSPEAKERS ENCLOSURES

I. General Loudspeaker Principles

LAST week I had a chance to examine a radio manufacturer's newest "high-fidelity" phonograph. It had a console cabinet made of wood-grain printed fiberboard, large gold-trimmed knobs, and a two-way speaker system. The bass speaker was the cheapest kind of bargain-line junk; a stamped metal frame and no visible magnet at all.

Any audiophile would react as I did with a mixture of amusement and disgust. We all know that a real hi-fi speaker is a husky affair held together by a massive cast-iron frame, and mounted in some sort of baffle utilizing various ducts, vents, or labyrinths. But precisely what is it that makes the difference? Why is the design of a quality loudspeaker such an esoteric process compared to the straightforward techniques used for tuners and amplifiers?

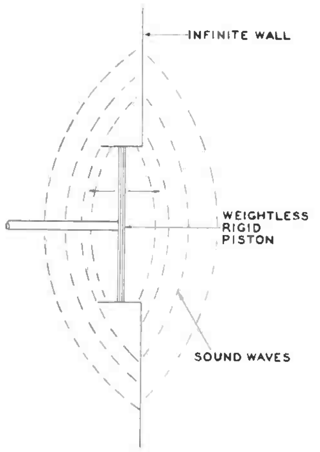


Fig. 1. Sound from an ideal propagator.

First of all, let's stop and consider just what a loudspeaker is supposed to do. The well-read hi-fi salesman says that an ideal speaker will produce sound waves which are accurate representations of the electrical signals driving the speaker. It would seem to follow that, if the rest of the system is as good as the speaker, the sound will be indistinguishable from a live performance. Unfortunately, this isn't necessarily true.

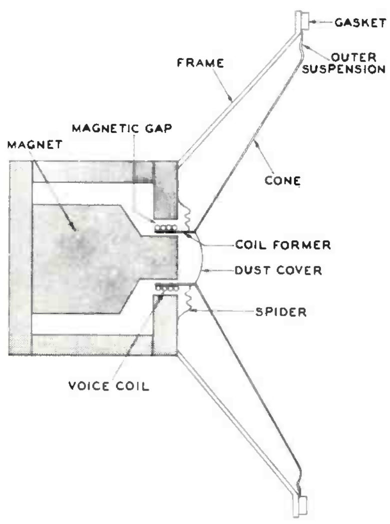


Fig. 2. Practical speaker construction.

I'll go into that later, but for the time being let's say a loudspeaker is a device that produces sound when connected to a suitable electrical signal, and the kind of sound it makes is related to the signal wave form.

The most common type of speaker consists of a diaphragm vibrated by a small motor. The diaphragm is, of course, the speaker cone, and the motor is a voice coil positioned in a strong magnetic field. The cone alternately compresses and rarefies the surrounding air to generate sound waves. In ribbon loudspeakers the diaphragm itself may be the moving element of the motor.

This isn't the only way to generate a satisfactory sound. Air molecules can be given an electrical charge, for instance, and set vibrating in an electrostatic field. This is the principle of the French *Ionophone* which caused a stir about three years ago. One trouble with that system is that there is only a tiny margin of adjustment between giving air molecules an electric charge and blowing all the fuses in one grand high-voltage arc!

You can also heat and cool air alternately in a confined space; as it expands and contracts it will generate sound waves. Dr. Lee DeForest holds several patents based on variations of this notion. The main disadvantage is the difficulty of heating and cooling something 60,000 or more times a minute.

Some clever inventors have built a sort of mechanical human voice. Compressed air is forced through a valve which is regulated by audio signal voltage. The result is an awesome thunder of sound indeed. So far, the system has proved practical only for voice reproduction.

With the exception of some special units, almost all high-fidelity speakers today use a cone or diaphragm driven by a voice coil. Every text dealing with loudspeakers begins with the classic concept of a vibrating source; the simplest source of sound waves is usually

COURTESY JAMES B. LANSING SOUND, INC.

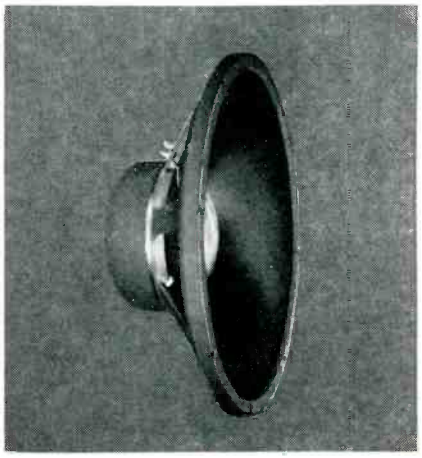


Fig. 3. James B. Lansing wide-range unit.

pictured as a piston moving back and forth in a wall of infinite dimensions. When the piston moves forward, Fig. 1, the air is compressed in front and rarefied in back. On the backward stroke the air is compressed in back of the piston and rarefied in front of it. If the piston is jogging back and forth, producing alternate compressions and rarefactions at a rate between 20 and 20,000 times per second, sound waves will be produced. The piston is, of course, an impractical one. It is assumed to have no weight and to be perfectly rigid. And the infinite wall is assumed to be rigid as well as infinite, transmitting no sound waves at all from one side to the other. Nevertheless, all sorts of useful calculations can be made from these idealized elements.

For example, here are three statements which prove to be extremely important in designing real loudspeakers:

1) The sound waves on the two sides of the wall are *out of phase*. The air is being compressed on one side at the same time a partial vacuum is formed on the other.

2) To do the same amount of work, the piston must move in and out much farther at low frequencies than at high frequencies.

3) To generate sound waves efficiently, the diaphragm must be larger to produce low frequencies than to produce high frequencies. Let's go into each of these in a little more detail.

If the wall were not there, and the piston were vibrating all by itself, the air would simply tend to slosh around from one side to the other as the diaphragm moved through it. If the piston were a foot in diameter and vibrating 40 times a second, nothing much would happen because the air would have plenty of time to slip around the edge instead of initiating a series of sound waves. But if the same piston vibrates at 1,000 cycles per second, things happen so fast that sound waves are produced before the air has a chance to short-circuit itself. The wall acts as an isolating barrier to prevent low frequencies from being lost by cancellation. To generate a 40-cps tone without the wall present, the piston would have to be at least 10 ft. in diameter.

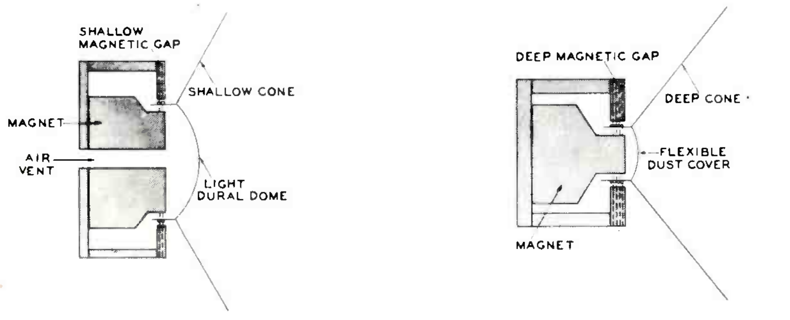


Fig. 4. Comparison of construction for speakers with large voice-coil diameter, at left, and that for speakers having normal voice-coil diameter, which is shown at the right.

Now to the statement that the piston has to move farther at low frequencies. Air is a viscous material, resisting displacement just as water resists a canoe paddle. This resistance is directly proportional to the velocity of movement; it is easier to move air slowly than at high velocity, as you know if you have ever held your hand out the window of a moving automobile. Work is the product of force multiplied by distance, so a slow-moving speaker that meets little resistance must move farther to accomplish as much work as it does at a higher frequency, for which air resistance is greater.

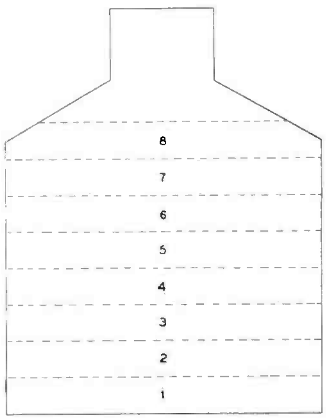


Fig. 5. Total flux is limited by magnet.

We've already discovered that, without a wall to separate its two sides, the piston must be very big to produce low frequencies. The same thing is true *with* a wall, but it's not quite so hopeless. If you were mixing a batch of cement, you could do a pretty thorough job with either a tablespoon or a large paddle. Of course, a few strokes of the paddle would do the trick, while you'd have to stir quite a while with the tablespoon. If you moved the tablespoon as slowly as you do the paddle, nothing much would happen.

Our piston has the same trouble — air is even more thin and fluid than concrete mix. Radiation of low frequencies drops off unless the piston is really big, or unless it moves in and out even farther than implied by the second general statement. If the piston's diameter is much smaller than a wave length, amplitude has to be increased to get much sound output.

So much for the standard rules. They form a basis for speaker design, but now let's consider some problems that arise with real speakers. Ignoring effects of acoustic loads and enclosures, there are new difficulties that didn't show up in the theoretical system:

1) A real cone has mass.

2) A real cone is not perfectly rigid.

3) A real cone has to be suspended somehow, and the suspension will be springy.

At this point things become a little discouraging. The mass of a real diaphragm requires force simply to overcome its own inertia, as well as to move air. If we decrease cone mass to increase efficiency, the cone becomes so limp that it buckles and flaps about. To make a rigid cone we must make a reasonably massive cone, and the fact that this mass is effectively suspended from a series of springs means that the whole thing has a natural frequency of resonance. At driving frequencies above resonance, we waste amplifier power overcoming the inertia of the cone. Below resonance we waste power pushing against the tension of the suspension.

Moreover, if the cone is made big enough to do a good job at low frequencies, it will be too heavy to reproduce high frequencies at all. This isn't all, either — since the cone can never be *perfectly* rigid at higher frequencies it breaks up into all sorts of queer sub-vibration patterns. Spurious vibrations ripple across the surface accentuating some frequencies, cancelling others, and

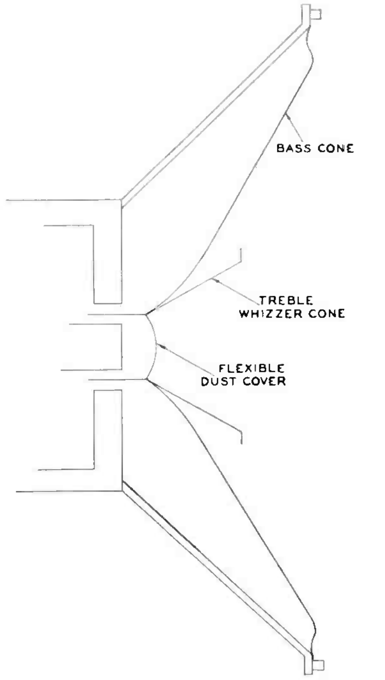


Fig. 6. Separate cone for treble range.

on certain notes creating buzzes and bad distortion.

It is no wonder that designers have conflicting opinions as to which are the most important considerations and compromises in dealing with these conflicting physical requirements. The basic assembly of cone, frame, voice coil, and magnet, however, is about the same in most speakers. Fig. 2 shows a cross section of a simplified dynamic speaker.

COURTESY ELECTRO-VOICE, INC.

Fig. 7. Electro-Voice SP-12 loudspeaker.

The cone is suspended at its outer edge either by a series of built-in corrugations or by a ring of some other material. Another ring of springy stuff, called the spider, keeps the cone centered at its apex and prevents it from wobbling sidewise. A small spool-shaped coil is wound on a conical former attached to the cone at its apex. The coil consists of several turns of copper wire or aluminum ribbon; the coil former may be from ¾ in. to 4 in. in diameter.

The voice coil is centered in a ring-shaped gap between the two poles of a strong magnet. When an alternating current is applied to the coil, it moves in the field under the same impulse that makes a motor turn. Vibrating longitudinally in the gap, it moves the cone along with it. The whole affair is kept in precise alignment by a heavy outer frame or *basket*. The voice coil must be as light as possible, yet have negligible electrical resistance. If it is wound of very light wire, amplifier power will be wasted in heat. If the wire is of heavy gauge to reduce the DC resistance, it begins to add weight to the cone assembly.

Note also that the magnetic field is concentrated in a small gap. If the cone moves too far, the coil will move part way out of the gap into a region where the magnetic field is much weaker. This results in severe distortion. But if the gap is made longer to keep the flux density uniform, then the field at any given point is not so strong, and the speaker's efficiency drops.

This explains why many expensive speakers with magnets weighing several pounds are not as efficient as the dime-store units used in television sets. If the cone suspension is made very loose to extend bass response, then the gap must be long so that the voice coil can swing a long way back and forth. If bass response is cut off by raising the cone resonance to about 200 cps the suspension can be quite stiff, and all the magnetic flux can be concentrated in a very short gap. This is precisely what is done in radio replacement and public-address speakers, where cost and power-handling ability are important.

I still haven't mentioned *all* the things which make speaker design logically impossible, but the ones covered are all important. Different companies use different approaches to these problems, and the buyer has to develop a nice balance between his ears and his sense of engineering propriety to decide which philosophy is the most convincing.

James B. Lansing speakers are interesting, for example, because they use larger voice coils than most others. The 12- and 15-inch Lansing speakers, Fig. 3, have voice coils 4 in. in diameter. It takes precise workmanship to wind a lightweight coil 4 in. across and perfectly circular.

Fig. 4 gives a rough comparison between the cross section of a Lansing

COURTESY H. A. HARTLEY CO., INC.

Fig. 8. The Hartley model 215 speaker.

speaker and one of more conventional proportions. There are two reasons for the larger diameter voice coil. In the first place, the cone is driven from a distributed source instead of a small area and is consequently controlled more rigidly. Second, the large coil permits a much shallower cone and, consequently, smoother distribution of high frequencies. A deep cone tends to concentrate the highs into a narrow piercing beam.

Since fewer turns of wire are used on a large coil it takes up less space lengthwise in the gap. Jim Lansing people claim that this enables them to use shallower gaps without sacrificing bass power-handling ability. This sounds quite logical, and one would suppose that the difficulty in making a 4-inch gap and voice-coil assembly is all that discourages other companies from following the Lansing design.

The opposing camp claims, however, that there are very good reasons why a smaller voice coil and a deep cone are good engineering. Stephens and Electro-Voice, among others, are champions of the "long-throw" voice coil. Howard Souther of the latter company explains why:

"In a 5-pound magnet there are just so many lines of flux available. Think of the magnet as a bottle containing magnetic flux. The dotted lines in Fig. 5 are flux units, eight in all, for purposes of illustration. Fig. 4a shows the magnet structure with a 4-inch voice coil. Steel of a certain thickness carries just so many lines of flux; two lines are shown. Because the area is large in diameter, we have used all the available flux in a shallow gap of only two lines.

"Fig. 4b shows a 2½-inch voice coil using the same 5-pound magnet. The gap is now twice as deep and carries four flux lines. With just so many flux lines available from the magnet, the smaller diameter allows a deeper gap for the voice coil to travel in. This means that bass tones can be generated with purity at high power, with the coil immersed in flux lines. In a 4-inch coil, therefore, bass tones must be limited in power or distortion occurs because the coil leaves the region of flux."

In an emphatic summation which leaves no doubt as to his own feelings in the matter, Mr. Souther concludes, "Smaller 2½-inch voice coils deliver full bass, eliminate peaked response, do away with heavy, efficiency-sapping coil support forms, are stronger and less susceptible to warping, prevent the listening distortion caused by 4-inch coils, and allow twice the cone excursion and permit double the amplifier power to be employed over a 4-inch coil."

Mr. Souther did not mention Electro-Voice's answer to the problem of high-frequency dispersion. Since a heavy cone designed for good bass will not reproduce highs satisfactorily, why not use a separate smaller cone to propagate

Continued on page 34

COURTESY ALTEC LANSING CORP.

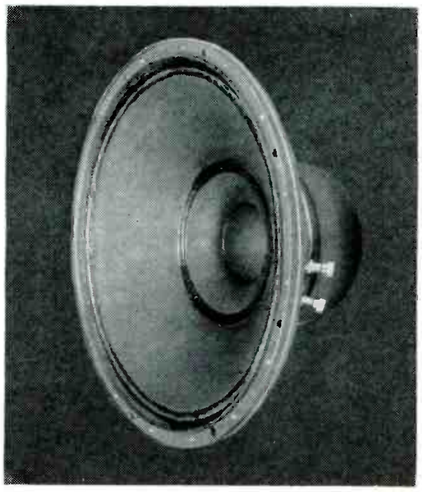


Fig. 9. An Altec-Lansing Biflex model.

Heath HD-1 Distortion Meter

An Audiocraft *kit report*

THERE are two methods of distortion testing in common use. In one, two sine-wave signals differing in frequency (and usually in amplitude) are applied to the equipment under test, and the modulation products are measured after the original signals are filtered out. This is intermodulation or IM testing.

In the other method a single sine-wave signal is used. After going through the equipment being tested, the original or fundamental signal is filtered out and the harmonic products generated by the equipment are measured. In either case, distortion is expressed as a per cent relation between the amplitude of the residues after filtering — the intermodulation or harmonic products — and that of the input signal.

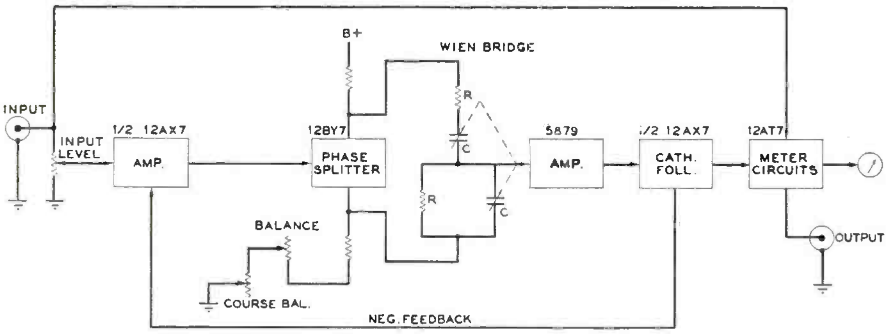
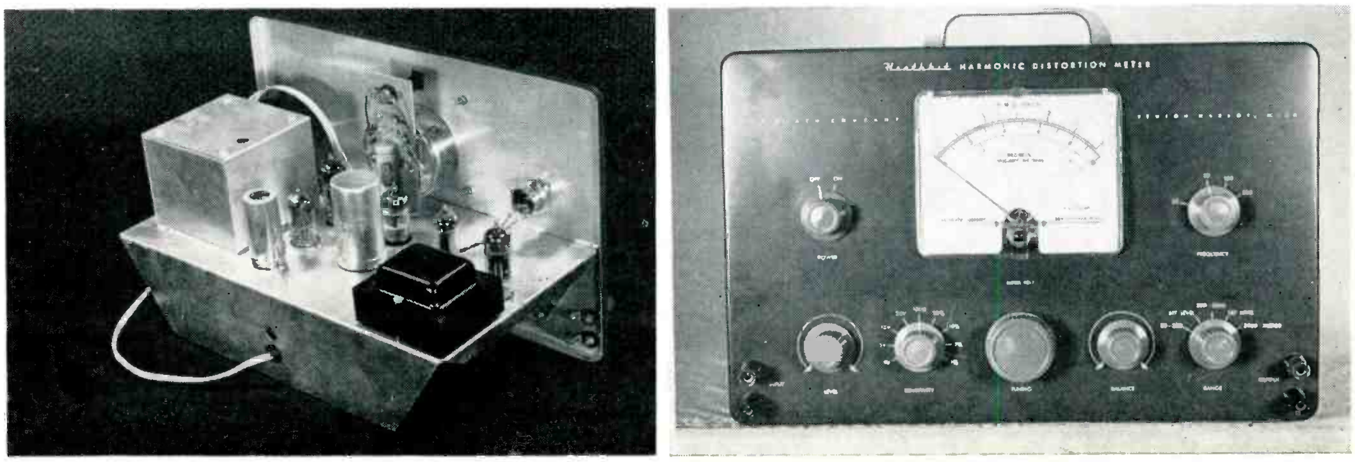


Fig. 1. Simplified circuit of the HD-1 kit. Fundamental is removed by Wien bridge.

Since both harmonic and intermodulation distortion are results of the same defect (non-linearity) in the equipment being tested, some correlation should be expected between distortion figures obtained by the two test methods. So there is, but it is a rough correlation at best. Factors which may throw off normal correlation are the frequencies chosen for testing, the frequency response of the tested equipment, how this frequency response was obtained, and many others. Measured IM distortion is often about four times the measured harmonic distortion at equivalent single-frequency power levels, but this ratio varies widely. Advertised equipment specifications often contain harmonic-distortion figures only, since they are smaller; if the type of distortion is not stated it is safe to assume that it is

Fig. 2. At left, chassis removed from its case; note shield on the tuning capacitor gang. At right, view of controls on front panel.



harmonic. Either figure can be misleading, however, unless the specific conditions are given.

It would be pointless to argue for or against either method of testing in this article. Each has its merits and disadvantages in certain applications. IM tests are certainly more sensitive in determining very small amounts of distortion, and it seems plausible that they are likely to indicate more directly the unpleasant effects of distortion. But for testing pickups and loudspeakers, harmonic distortion tests are very much easier because only one frequency is involved. Inexpensive IM test instruments are usually limited to one low frequency and one or two high frequencies, unless separate audio oscillators are used; and, if the built-in signal sources are employed, it is quite inconvenient to test amplifiers which do not have one output terminal connected to ground. There are many test records available with sine-wave tones suitable for use with a harmonic distortion meter, but very few for IM testing. To cover the full audio-frequency range with a low-distortion signal source, an external signal generator is needed with either instrument, so total costs are about the same. In short: with comparable investments in equipment available today, IM testing is probably more precise and more indicative of true listening quality in circumstances under which it can be used, but harmonic distortion testing is far simpler for the non-professional and is more easily adaptable to all circumstances.

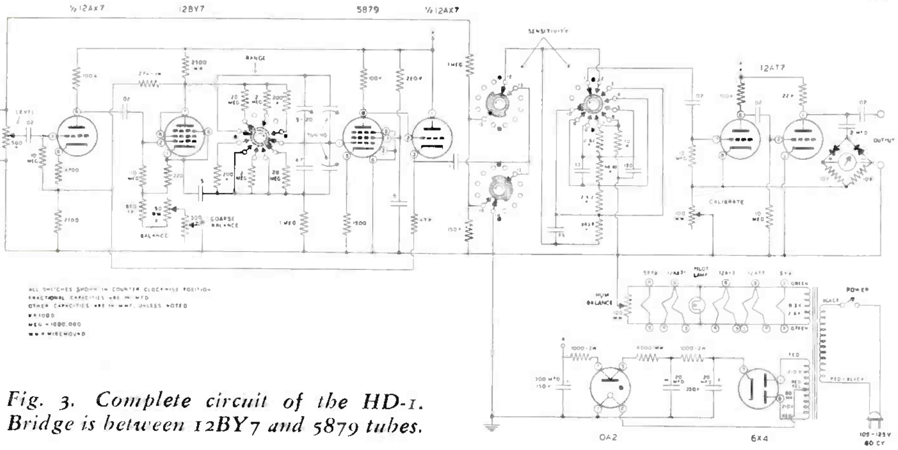


Fig. 3. Complete circuit of the HD-1. Bridge is between 12BY7 and 5879 tubes.

Circuit Description

Most IM tests are made by the SMPTE method, in which two signals widely spaced in frequency are employed. Surprising as it may seem, a circuit to measure intermodulation products by the SMPTE method is less complex and critical in component requirements than a circuit to measure harmonic distortion. This is because the filters in an IM tester are fixed-tuned, not critical in cut-off frequency, and need not have particularly sharp cut-off rates. The filter in a harmonic distortion meter, however, must be tunable (to accommodate any test-signal frequency being used); and it must have an extremely sharp cut-off rate — ideally, it should reject the fundamental frequency completely while passing all harmonics, including the second, without attenuation. Those are pretty tough design requirements.

They are met in the Heath HD-1 by means of a circuit centered around a Wien bridge. This is a resistance-capacitance bridge having one series RC arm and one parallel RC arm, as shown in the block diagram in Fig. 1. The resistances must be equal and the capacitors equal in value. Such a bridge, when fed AC voltages of opposite phase and of a 2:1 amplitude ratio, will have a null (zero output) at a frequency

$$f = \frac{1}{2\pi RC}.$$

The output rises for frequencies on both sides of the null frequency. It is still attenuated considerably, though, at the second harmonic frequency, so negative feedback was applied around the bridge in the HD-1. This effectively narrows the notch or null frequency range, making the bridge response almost ideal.

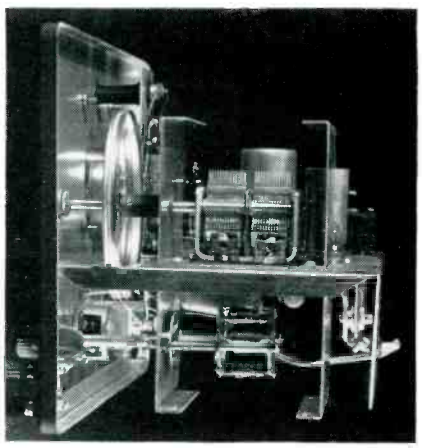
Fig. 2 is a close-up view of the front panel and operating controls; it should be referred to along with the block diagram for the rest of this discussion on the HD-1 circuit. A complete schematic diagram is given in Fig. 3.

The test signal, applied to the input terminals of the instrument, may be at any voltage level (a minimum of 0.3 volt is required). In order to establish the proper reference level, therefore, an input level control is necessary; it is located at the lower left on the front panel. From the level control the signal is fed to a triode amplifying stage, and thence to a pentode phase-splitter stage. This furnishes the two out-of-phase signals required by the Wien bridge. The 2:1 voltage ratio for the bridge is obtained by making the plate load of the phase splitter roughly twice as large as the cathode load. Precise control of this ratio, and compensation for inaccuracies in the bridge components, is obtained by a coarse balance potentiometer (located above the chassis) and a fine balance control (on the front panel), both in the phase-splitter cathode circuit.

Tuning of the bridge is accomplished by a range switch, at the lower right on the panel, and the tuning control, which is the large knob centered below the meter. Resistors in each arm of the bridge are changed simultaneously by the range switch; range positions are 20 to 200, 200 to 2,000, and 2,000 to 20,000 cps. A double-gang variable capacitor is controlled by the tuning knob to adjust both capacitive elements of the bridge simultaneously, thus tuning it precisely to any frequency within the selected range. This control is coupled to the frequency indicator also, located at the right of the meter in Fig. 2. It is an indicator only, and should not be used for tuning purposes.

The output of the bridge goes to another pentode amplifier, and then to a triode cathode follower which feeds the meter circuits. These utilize a twin triode amplifier driving a half-bridge rectifier. Negative feedback is employed to improve linearity and adjust sensitivity; this section in itself is a fine vacuum-tube voltmeter. Signals amplified by the meter amplifier are fed also to a pair of output terminals, so that whatever is being measured by the meter can be monitored visually on an oscilloscope.

It will be noticed that the range switch has two intermediate positions marked SET LEVEL. To operate the instrument, a signal from the equipment under test is connected to the input terminals. The range switch is turned to one of its SET LEVEL positions, which opens one arm of the Wien bridge and shorts out the other arm; this section of the circuit then



Tuning gang and range switch uncovered.

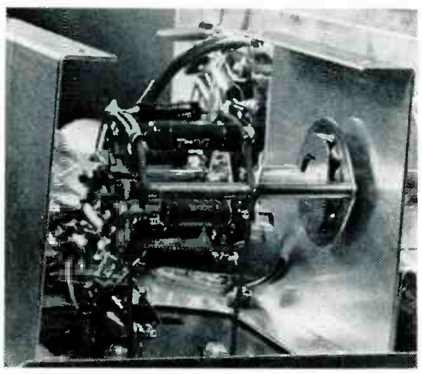
operates as an equivalent amplifier without the bridge null characteristic. With the sensitivity switch set in the 100% position, the level control is adjusted for full-scale indication. The reference level for the measurement is now set up. Next, the range switch is turned to the appropriate position for the test frequency being used. The tuning control is turned to give a minimum indication on the meter; if the signal has relatively low distortion, the meter indication may be unreadable at the null. This can be corrected by turning the sensitivity switch downward to the 30%, 10%, or a lower position. Then the balance control is adjusted for minimum meter reading, and the tuning control is readjusted to find the absolute null. At that point the fundamental has been completely removed, and the meter is measuring the harmonics and whatever noise is present. Simply read the meter scale appropriate for the sensitivity switch position used, and the measurement is made. If the meter reads half-scale on the 3% position of the sensitivity switch, for instance, the distortion indication would be 1.5%. Before doing anything else, then, return the sensitivity switch to the 100% position — moving any other control may bang the needle up against the full-scale stop and, possibly, damage it.

There are four other positions on the sensitivity switch marked 1, 3, 10, and 30 v. In these positions the input signal is connected directly to the meter circuit, and the meter section can be used to measure input signal amplitude. This is a nice convenience, because the operating level of the equipment under test can be set without using an external meter. And, with the range switch in a SET LEVEL position, the entire instrument can be used as an AC VTVM of much higher sensitivity, according to the positions of the level control and the sensitivity switch. For instance: if the level control were set to give full-scale indication for a 1-volt signal with the sensitivity switch in the 100% position, the full-scale sensitivity would be 0.3 volt in the 30% position, 0.1 volt in the 10% position, .03 volt in the 3% position, and .01 volt in the 1% position. The calibration would be valid only for that particular level-control setting, of course.

This basic circuit is unusually susceptible to hum pickup because of the very high impedances in the Wien bridge circuit, and because the tuning capacitor gang must be completely isolated from the chassis. It is particularly unfortunate because any extraneous voltage pickup, such as hum, reduces the accuracy of indication. Special measures taken to reduce hum and noise include shields for the tuning capacitors and range-switch components, an insulated tuning shaft, a balance adjustment for the filament wiring, a low-noise pentode following the bridge, a voltage-regulated power supply, and optimal lead dress and component orientation. Even an isolated power switch has been used.

Construction Notes

As is the usual practice in Heathkits, every part is furnished, including hardware and hookup wire. The builder

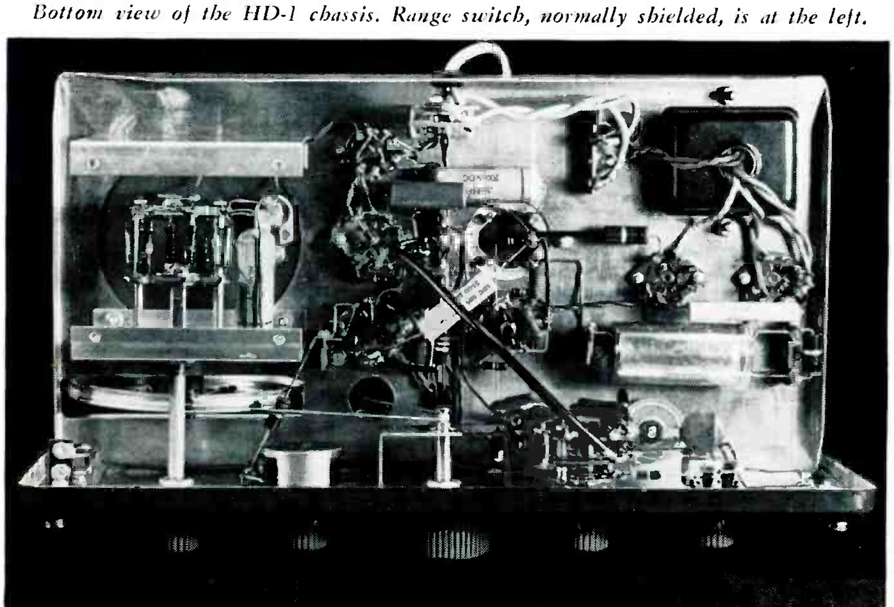


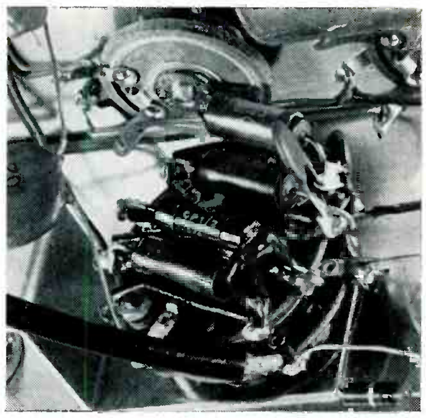
A close-up of the range switch assembly.

needs only a few simple tools — screw driver, knife or insulation stripper, needle-nose pliers, nut driver or small adjustable wrench, soldering iron, and rosin-core solder.

The instruction book is complete, well illustrated, and written unambiguously. Information is given that enables the most inexperienced constructor to identify each part, and step-by-step assembly

Bottom view of the HD-1 chassis. Range switch, normally shielded, is at the left.





Wiring of the meter sensitivity switch.

and wiring instructions are furnished. A section is devoted to test, calibration, and practical applications of the instrument.

We found no place where a reasonably careful builder could go wrong on the HD-1; accordingly, our comments in this section must be restricted to reiteration of our admiration for the fine work done by Heath's instruction book writers. It took us 11½ hours to check parts and wire the kit. Price of the HD-1 is $49.50.

AUDIOCRAFT Test Results

Specifications for the HD-1 include the following:

Frequency coverage. 20 to 20,000 cps, in three ranges.

Distortion ranges. 0 to 1%, 3%, 10%, 30%, and 100% full-scale. Voltmeter ranges as given in text.

Input resistance. 300 K.

Input voltage. 0.3 volt minimum for distortion readings.

Output voltage. 2.5 volts at full-scale meter reading.

Accuracy. Voltmeter section, ±5% of full-scale reading. For distortion readings, ±5% of full-scale reading +0.1%.

In terms of the most sensitive distortion scale (1% full-scale), where the tolerance is largest, the accuracy specification would imply that distortion could not be determined reliably with the HD-1 closer than 0.15%. That isn't very good accuracy when working with many of the new amplifiers and preamps, which have far less distortion than this at normal operating levels. The specification is quite misleading, fortunately.

Let's consider it in two parts. The "±5% of full-scale" is an outside accuracy tolerance for the meter section. Normally it will be much closer than that anyway; even if it isn't, the error becomes progressively smaller as the low end of the scale is approached. For readings at the upper end of the scale — near 1%, still considering the lowest range — the maximum error of .05% isn't going to make much difference. If you are working with very low distortion indications, at the low end of the scale,

Continued on page 34

Hi Fi Pure and Simple

by Lee Beeder

SURROUNDED by piles of wood, sand, resistors, capacitors, and tubes, my thoughts were mixed. I regarded the impending task (that of doing-it-myself and assembling a complete audio system) with ambivalence. On the one hand I was anxious to own a good sound system, but on the other hand I was beginning to perceive that I might have chosen a hard way to acquire one.

As stated in the title, the sound is pure and the work is simple.

I would like to offer the following details of the setup that I assembled. The bass enclosure, to my ear, sounds about as good as any enclosure extant, regardless of price. That this one can be constructed for about $20 can be regarded as a lagniappe. The crossover network is a full 12-db-per-octave

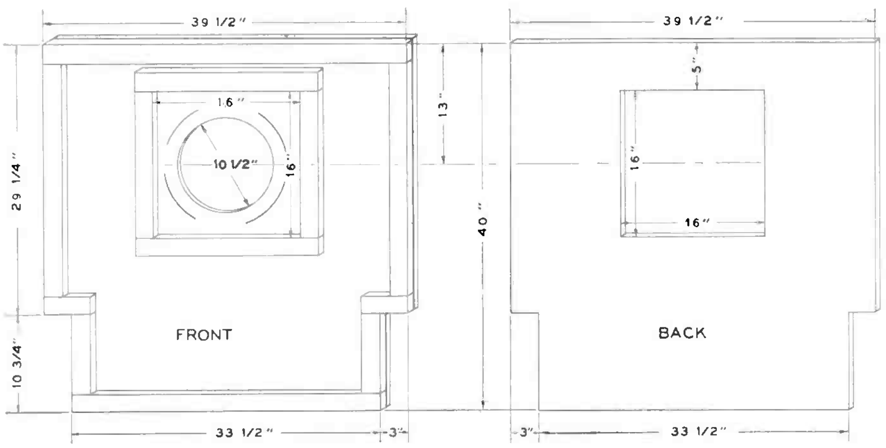


Fig. 1. These two sections, separated by dry sand, make main part of woofer baffle.

There seemed to be no turning back, however. An analogy suggested itself as I looked around me.

I thought of the time I had stood at the summit of a fairly steep and ice-covered hill, attached to the first pair of skis I'd ever even seen. As I hesitated, some helpful oaf (also standing on the mountain top wearing skis) said, "It don't take nothin' but nerve." Several seconds later I was zooming down the mountain and through the trees, in what I now recall (when I can bear to think of it at all) as a rather exhilarating descent.

In a sense, I was in the same position with the aforementioned raw materials. I wasn't worried about the theory involved, but the prospect of practical hand-assembly gave me pause. Happily, I had no real cause for concern. The work is all done now, and the resultant system sounds just fine. As a matter of fact, the whole job went so swimmingly that I'd like to urge any hesitant do-it-yourselfer to take the plunge without any more procrastination. You can save quite a bit of money, have a lot of fun, and acquire a really excellent rig.

parallel type, with air-core coils and oil-filled capacitors, such as are used in highest quality speaker systems. The price (about $10) is certainly very reasonable.

The woofer baffle is a slightly large version of the very well-known Briggs sand-filled corner baffle. Its panels are non-resonant and, with a good woofer, it produces very clean, undistorted low bass. Being in the corner and virtually part of the room, it produces a wonderfully realistic wide source of sound.

In his book* Briggs gives panel dimensions for a 7½-cubic-foot enclosure. I changed the dimensions to attain the magic 9 cu. ft. that is purportedly optimum. The two side cutouts, necessary to clear the baseboards of the room, also tune the enclosure to approximately 40 cps. Panel dimensions are given in Fig. 1, and the finished enclosure is shown in Fig. 2.

Materials required are:

*G. A. Briggs, *Sound Reproduction*, Wharfedale, London. Distributed in U. S. A. by British Industries Corp., Port Washington, N. Y.

1 sheet ½-inch plywood, 4 × 4 ft.
2 sheets ⅜-inch plywood, 4 × 4 ft.
100-pound sack of sand.
20 ft. of 1- or 1¼-inch × 1¾- or 2-inch wood cleating.
1 qt. gun-type calking compound or some wood glue.
1- or 1¼-inch screws or nails.

The actual construction is very simple. First, the two plywood boards are cut as shown (the speaker cutout being on the ½-inch plywood panel), and the side cleating added completely around the panel. This should be at least 1 in. thick, preferably 1¼ in. thick. The cleating around the speaker cutout should clear the speaker by about ¾ in. on each side.

If you have a 12-inch speaker I'd advise drawing a circle, as shown, large enough to accommodate a 15-inch speaker. Then, if you decide in the future to get a larger speaker, you will have planned for such a contingency.

Next, put a little calking compound (gun-type is best) all around the cleating and screw or nail it securely to the panel. If you feel that it's easier to glue the cleating to the panel and omit the calking compound, then by all means proceed accordingly. If you choose the calking-compound method,

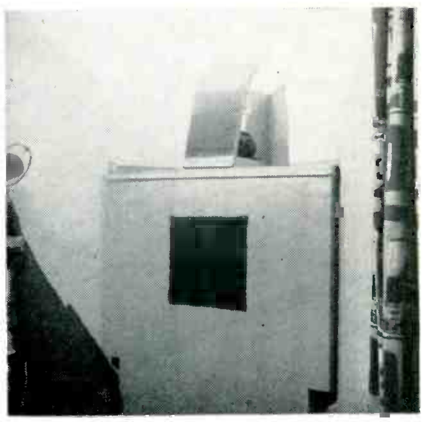


Fig. 2. Baffle installed in room corner.

as I did, you'll find that it can be applied easily with a knife blade. After this, pour in the sand, which should be very dry. The sand that was delivered to me was wet so I had to put it in some large pans and dry it in the oven. If you have to do the same thing and the wife gets inquisitive, tell her you're hatching turtle eggs.

Then, when the sand has been poured in and packed down smoothly, put a little more calking compound or glue all around the cleating and put the ⅜-inch plywood top on the sandwich of sand. Then screw or nail it on tightly. That's all there is to it; you now have a solidly built and fairly heavy corner baffle. It will stand securely in the corner while you fit on a top.

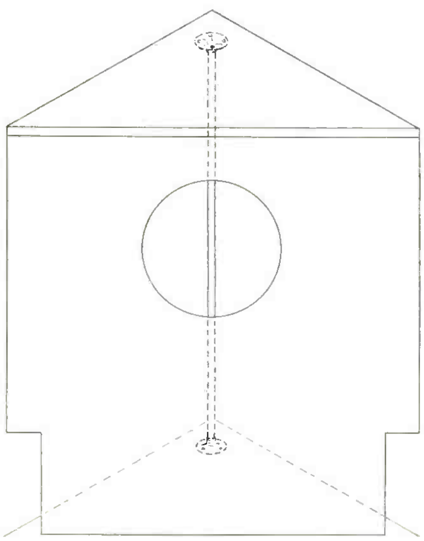
I made a thinner sandwich of sand for my top and supported it with a 40-inch shower-curtain rod. Briggs doesn't mention his way of securing a top to the enclosure, simply stating with typical British reserve, "Fit a top to the enclosure." It might be that there are many better ways of securing the top than the one I used, but mine works nicely and will support quite a bit of weight if necessary.

I attached one of the two flanges supplied with the rod to the top, at a point several inches from the apex, as in Fig. 3. The other flange was attached to a piece of wood (to protect the rug) and placed on the floor. The shower-curtain rod was then sawed to the proper length and inserted in the two flanges.

Here I'd like to mention that, though I no longer have the original invoices for material used, I am sure that the total cost was around $20, possibly less. I consider this quite reasonable for a bass enclosure having the characteristics of this one. With a good low-resonance woofer it is matched by only a very few top-quality commercial speaker systems.

Atop the bass enclosure I placed a horn-loaded mid-range driver and a horn-loaded tweeter. They were mounted as shown in Figs. 4 and 5. This is so simple that I won't spend much time on it, other than to say that the wooden base is weighted with a small sandwich of sand.

Fig. 3. How the top panel is supported.



Crossover Network

The network I used has a 16-ohm input and crosses over at 700 cps. It uses two 5.1-mh coils and two 10-µfd capacitors, connected as shown in Fig. 6. This is for a two-way system in which the tweeter range extends below 700 cps, or one in which the middle and upper-range reproducers are already equipped with their network. For a separate high-range tweeter crossing over a little above 5,000 cps, you can use the three-way network illustrated in Fig. 7; here a 0.5-mh coil and a 2-µfd capacitor are added. Again this is for an all-16-ohm system.

If you have 8-ohm speakers, you'll need two 2.5-mh coils and two 20-µfd capacitors for the 700-cps crossover point. For a third speaker in this network you'll need a 0.25-mh coil and a 4-µfd capacitor.

The 2.5-mh and 5.1-mh coils are wound with No. 17 single cotton-covered wire on wooden dowels 1¼ in. long and 1¼ in. diameter. Smaller coils are wound with the same wire on forms ¾ in. long and 1¼ in. in diameter. The wire, if locally unobtainable, is available from mail-order radio parts distributors.

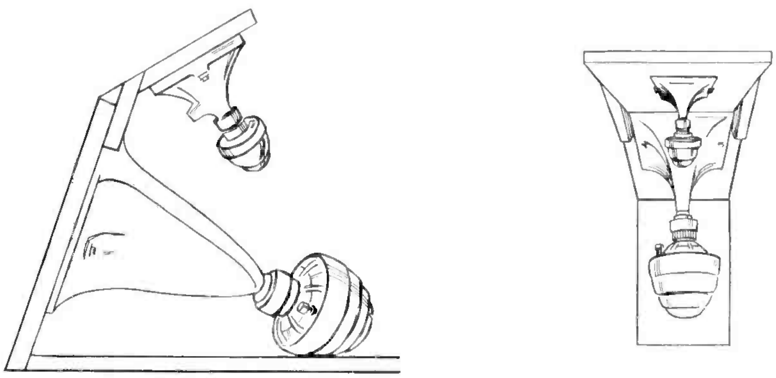


Fig. 4. One way to mount the middle-range and high-frequency reproducing units.

End pieces of the coil forms are either wood or hardboard, and are attached to the wooden core with solid brass screws; See Fig. 8.

The turns of wire necessary for these inductance values are as follows:

2.5 mh — 278 turns
5.1 mh — 382 turns
0.25 mh — 80 turns
0.5 mh — 113 turns

You needn't worry about any special winding technique; just put on the correct number of turns. When you've used all the wire on one spool and need more wire to complete the coil, make a good splice, cover the splice with tape, and continue winding. You'll need about 1½ spools of wire to make the 2.5-mh coils and somewhat more for the larger ones.

The oil-filled capacitors can be obtained quite inexpensively, even though a 10-µfd oil-filled capacitor normally sells for around $10 net. Many surplus outlets, as well as several radio and hi-fi

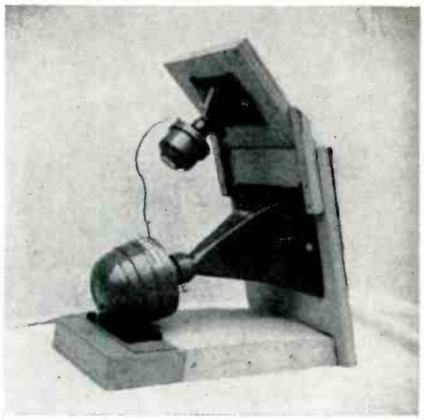


Fig. 5. Upper-range horns with drivers.

dealers, sell them for less than a half-dollar to a couple of dollars each, depending on size. I paid 49¢ for my 1-µfd oil-filled capacitor, and $1.39 each for the 10-µfd ones. The total cost of the network, figuring five spools of wire at $1.40 a spool, was $10.27. That will also be the approximate cost of an 8-ohm network, for, though you'll need only four spools of wire, the 20-µfd capacitors cost 40¢ or 50¢ more each than the 10-µfd ones.

On completion, components for the low-frequency network were installed on two 6-by-12-inch pieces of wood that were screwed together (at right angles) with solid brass screws. The capacitors were mounted in the center, and the coils at right angles to each other at the ends of the pieces of wood. I used solid brass screws to attach the coils to the pieces of wood. Fig. 9 shows this part of the network.

The two components making up the 5,000-cps network were wired directly to the baffle holding the tweeter, though they could have been assembled in the same way as the others, and put inside the woofer enclosure.

This completed the speaker setup, at a cost of about $30 for a superb, natural-sounding bass enclosure and a distortionless, low-loss half-section network. You could spend very much more and not do nearly so well. The cost of the speakers and drivers can vary widely, of course, depending on your preferences.

Continued on page 35

POWER SUPPLY

for the

TV Sound Amplifier

by Arthur I. Zabriskie

THE TV sound amplifier designed by John J. Huff* was built as specified by the writer. It was tested thoroughly before installation by substituting it for the amplifier in a high-quality sound system. The test results were very satisfactory.

Unfortunately, the television set in which the amplifier later was installed is a type having a series heater system, so it was impossible to obtain operating power from the TV. The audio output tube has 323 volts at the plate and 143 volts at the cathode. These operating potentials of the set dictated either a modification of the amplifier or the construction of a separate power supply.

It was decided that a separate power supply would be advantageous, because the builder would then be able to use the amplifier either for TV sound improvement, or as an excellent amplifier in a hi-fi sound system.

The amplifier requires approximately 250 volts DC at 37 ma, and 6.3 volts AC at less than 1 amp. In a power supply this permits the use of a 6X4 rectifier using a 6.3-volt heater at 0.6 amp, and eliminates the need for a transformer with a 5-volt secondary.

Because of these modest voltage and current requirements, construction of the power supply requires only a few hours of time and costs approximately $14 for parts.

The power supply, described in this article, furnishes the required output voltage with ample current to meet the requirements of the amplifier. The supply and the amplifier were tested together in a hi-fi installation, driving the University Classic system (discussed in AUDIOCRAFT for February 1956), and it responded very well. It handled transients quite satisfactorily. There was no ringing nor audible hum.

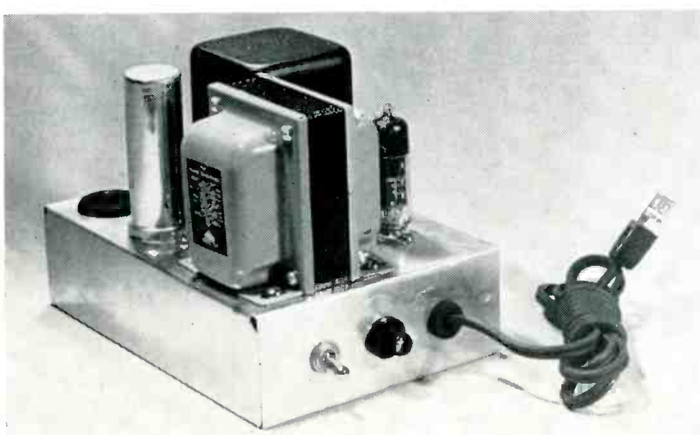
An aluminum chassis 2 by 5 by 7 in. furnishes ample room for the power-supply components. A layout diagram is given in Fig. 1; this can be cut out and used to make centerpunch marks on the chassis directly. Parts orientation is shown in Figs. 2 and 3, and in the pictorial wiring diagram, Fig. 4. When assembling parts on the chassis, note that lug terminal strips are fastened with several transformer and choke mounting bolts. Note also that the fuse post is oriented in such a way that one of its lugs is bent down to touch the lug on one single-lug terminal strip, and that one of the power cord conductors is soldered to both lugs.

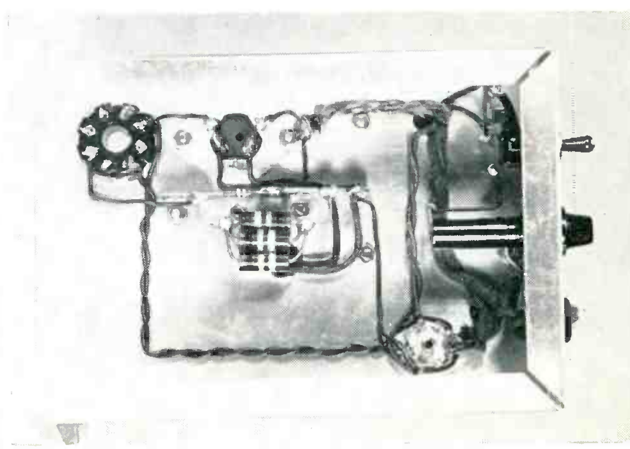
The transformer leads will have to be cut to proper length, since they are much too long. Some will be cut short enough that you will remove all the tinned length. Before trying to solder or tin these, you must scrape off the enamel insulation for half an inch at least. It chips or scrapes right off with a knife, but it is impossible to solder the wire until the enamel is removed.

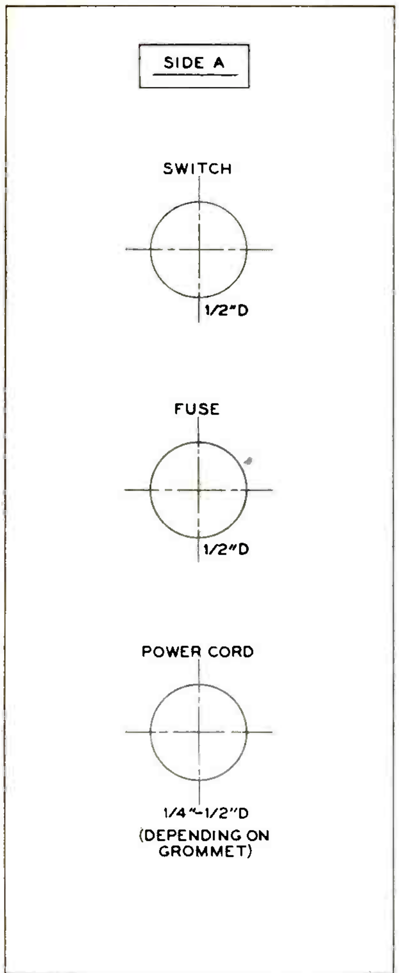
Both fiber and aluminum mounting wafers are furnished with the electrolytic capacitor. Use the one of aluminum. The capacitor lugs are inserted through the wafer slots and twisted a quarter turn with needle-nose pliers. Two of these lugs have holes in the center, so that it is easy to solder wires to them; they are used for ground connections, as Figs. 3

*"TV Can Sound Better", AUDIOCRAFT, December 1955.

Figs. 2 and 3. Two views of power supply. Four 2-watt resistors shown at center may be replaced by single 10-watt unit specified.







and 4 show. A standard ground lug is mounted under one of the wafer mounting nuts also.

The schematic diagram of the power supply appears in Fig. 5. A capacitor-input double-pi filter is used, with the choke as the first series element and a 2.5-K resistor as the second. The final capacitor section of this filter is in the amplifier proper. If the power supply were to be used with some other equipment, it would be a good idea to add another capacitor section across the output. The series resistor can be adjusted in value to obtain the desired voltage with other loads. Not more than 65 ma at high voltage should be taken from this supply, however. A 220-K resistor is also connected across the B supply to serve as a bleeder.

To use the amplifier and power supply for TV sound, it would be best to eliminate the shielded signal lead from the amplifier terminal strip to audio plug pin 5. Then a grounded phono jack could be installed on the amplifier chassis and the signal lead run from the jack to the terminal strip lug to which is attached the 10-K grid resistor of the 12AX7.

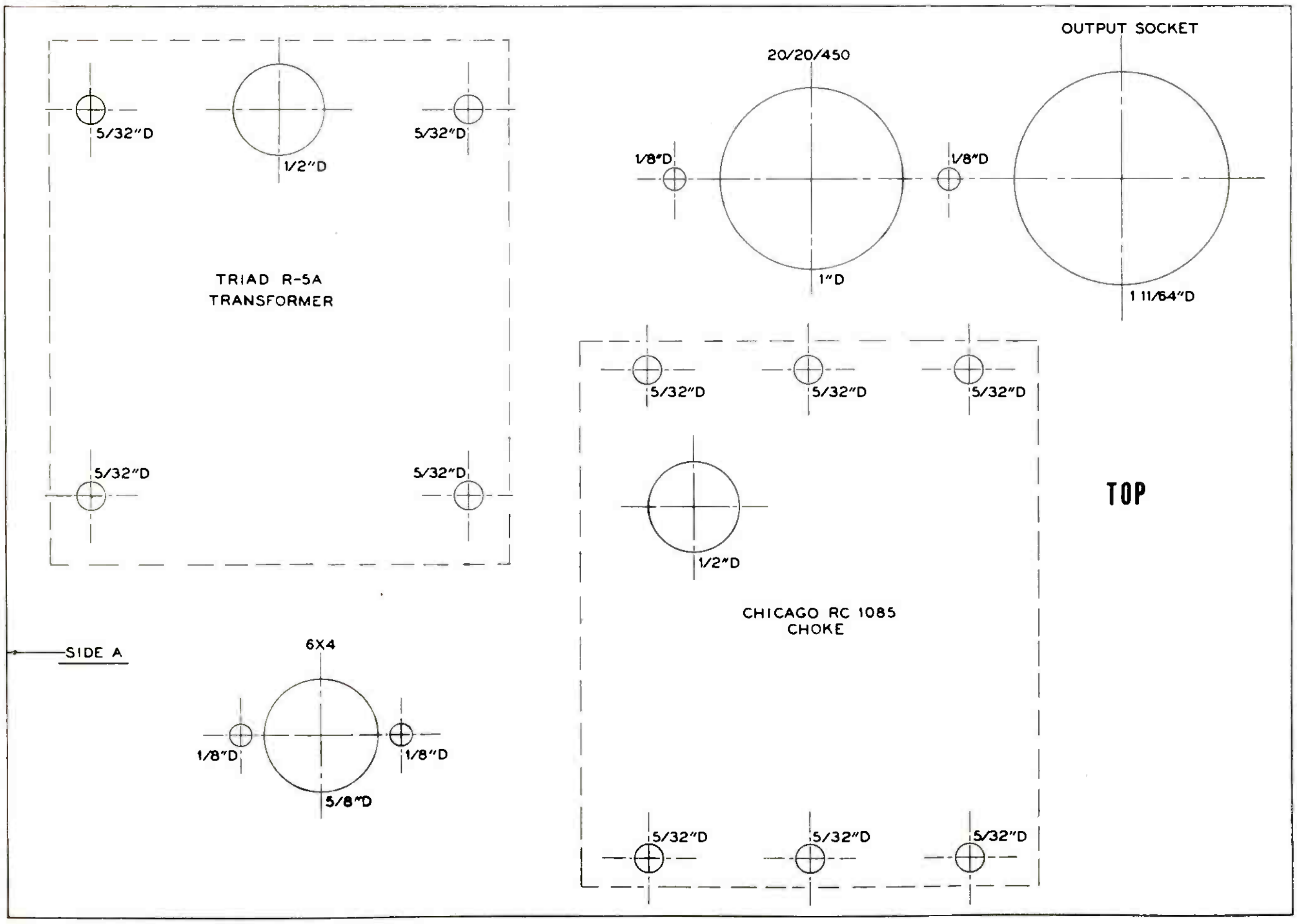
The TV signal is then picked up by a shielded lead attached, through a 0.1-μfd 600-volt paper capacitor, to the plate pin of the first audio tube. The shield of the signal lead is connected in the TV set to the shield of the lead from the high side of the volume control, through a 16-μfd 150-volt electrolytic capacitor. The other end of the hot signal lead is soldered in a phono plug, and the other end of the shield to the phono plug shell.

The SPST switch indicated in the parts list should be of the rotary type with a shaft long enough to reach through the TV cabinet, so that the amplifier's power supply can be operated from outside the cabinet as a separate unit. An IRC "Q" control may be used by purchasing the attachable switch. If the amplifier is to be used outside the cabinet an ordinary toggle switch will serve.

As stated by Mr. Huff, it is necessary to use an 8- or 16-ohm speaker with the TV amplifier. He also suggests that further refinement of the sound quality can be obtained by using a different speaker system. Since the speaker in most television receivers has a 3.2-ohm voice coil, and it is, therefore, necessary to replace it anyway, the construction of a bass-reflex cabinet for a table-model TV is worth while. The added expense for a cabinet would be approximately $6.

The cabinet is built with its horizontal dimensions the same as the TV receiver's base, and high enough to provide approximately 3 cu. ft. inside. The use of ½-inch or ¾-inch plywood, and plenty of

Fig. 1. Chassis hole-punching template, good only if the parts specified are employed.



glue and screws, is recommended. The speaker mounting panel should be tilted 1½ in. back at the top, so the sound will be projected upward toward the listener.

There are many excellent speakers of 8- or 10-inch size which provide good sound at low cost. Among such 8-inch speakers are the Quam 8A10X and the Permoflux 8T-8-1. I used the Quam 8A10X, which costs about $11, and found that a port opening of 30 sq. in. was best with a 3-cubic-foot cabinet. The outside of the cabinet was covered with leatherette. The legs of the old TV base were used to support the new speaker cabinet and the TV receiver above it.

There are further advantages in the use of a separate power supply, other than its usefulness in both TV and hi-fi sound reproduction with the amplifier. Among them are:

1) With a volume control wired into the amplifier, the existing TV speaker and the bass-reflex speaker can be used simultaneously or alternately. I installed a 1-MΩ IRCQ 13-137 volume control for this purpose.

2) There is no need for a voltage-dropping resistor in the B+ lead to the amplifier.

3) Picking up the signal from the first audio-tube plate pin, rather than at the output tube, reduces the possibility of having to provide further filtering of the television B+ supply in order to minimize hum.

4) The new TV sound system may be used, if desired, only for those programs which are enhanced by wide-range reproduction.

The disadvantages of the new system, and the precautions to be observed, are:

1) The new TV sound (if a live broadcast) is likely to contain more floor squeaks, paper rustles, and audience coughs than seems possible.

2) Unless a pilot light is installed you may forget to turn off the TV amplifier, because there is no hum to act as a reminder. Leaving the amplifier on provides a good test of operating temperatures of the amplifier components, but is not recommended. Mine has been tested this way for 24 hours on two different occasions. There was no overheating, but a pilot light will be installed.

3) If you decide to use only the new TV amplifier and power supply, do not remove the TV audio output tube. Let it burn, otherwise you may find the TV voltage divider unbalanced. To eliminate all sound from the TV's old audio section, connect a shorting wire across the output transformer primary winding.

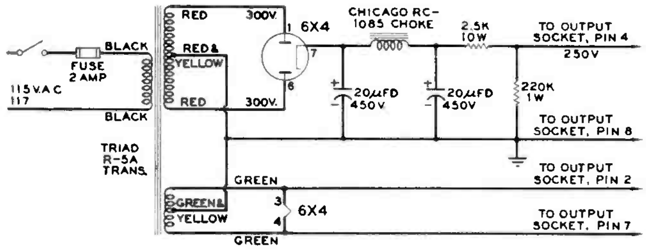


Fig. 5. Schematic diagram of the TV amplifier power supply. Note heavy filtering.

Parts List

Transformer, 600 vct @65 ma, 6.3 vct @2.7 amp: Triad R-5A or equivalent.

Choke, 10 h @85 ma: Chicago RC-1085 or equivalent.

Electrolytic capacitor, 20/20 μfd @450 volts: Sprague TVL-2755 or equivalent.

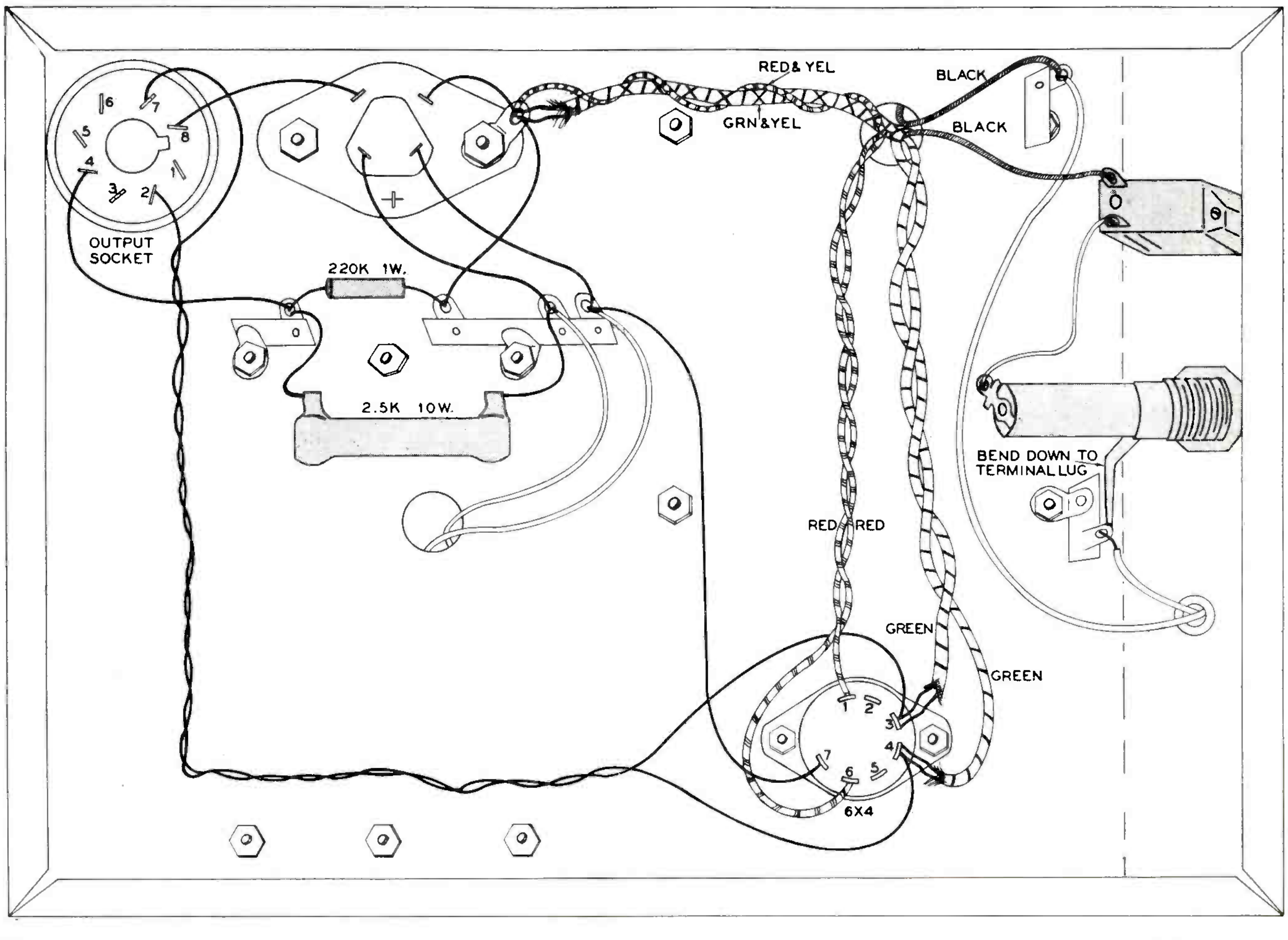
6X4 tube.

2.5-K, 10-watt resistor.

220-K, 1-watt resistor.

Continued on page 37

Fig. 4. Wiring diagram. Tie-lug terminal strips and ground lug are secured on capacitor, transformer, and choke mounting bolts.



BASIC ELECTRONICS

by Roy F. Allison

X: The Sine Wave

IN the preceding chapter of this series, the concept of alternating current was introduced. A square wave form was used for illustration of the terms *period, frequency, peak amplitude, peak-to-peak amplitude*, and *effective value*, which are important characteristics of all regular AC wave forms.

The most important and probably most common AC wave form, however, is not the square wave but the sine wave.

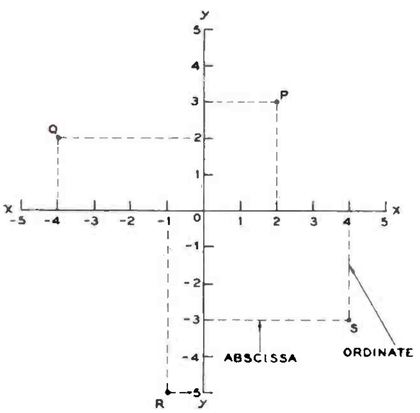


Fig. 1. Rectangular co-ordinate scales.

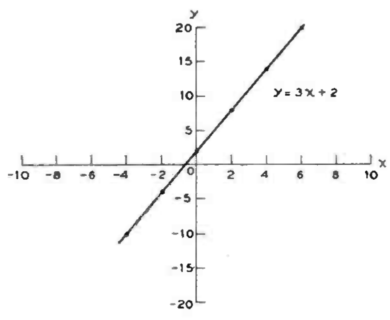
Indeed, all other repetitive wave forms can be obtained from combinations of sine waves having various frequency and amplitude relationships to one another. To understand the sine wave it is necessary to touch lightly on some basic geometric and trigonemetric figures.

Rectangular Co-ordinates

Anyone who has been subjected to high-school algebra will be familiar with the Cartesian or rectangular co-ordinate system. This system permits solution of problems in geometry by the use of algebraic equations; and the simplification of such equations, which express the relationships among two or more quantities, by their visual interpretation as lines, curves, or surfaces. It will be necessary here to consider only two-variable relationships and, therefore, only two-dimensional co-ordinates.

These are plotted on a chart that consists of two mutually perpendicular scale lines, as in Fig. 1. The horizontal scale is called the *x*-axis; the vertical scale is the *y*-axis. They cross at the zero-value point for both, which is called the origin. Values that are more than zero — positive values — are to the right of the origin on the *x*-axis and above the origin on the *y*-axis. Negative values are to the left of the origin on the *x*-axis and below the origin on the *y*-axis. An *x*-value of a point is called its *abscissa*, and the *y*-value its *ordinate*. Together they are known as the co-ordinates of the point. Co-ordinates are usually given with the abscissa listed first. Thus, the co-ordinates of point P (Fig. 1) are 2,3; those of point Q are —4,2; those of point R are —1, —5; and those of point S are 4, —3.

Equations in algebra can be looked on as describing precisely in what manner the value of one quantity is dependent on the value of another quantity. To obtain a visual indication of this relationship, we may ascribe arbitrary values to one of the quantities and, for each given value, solve the equation to obtain one or more values for the other quantity. Each pair of corresponding values thus obtained can then be plotted with rectangular co-ordinates on a scale similar to that in Fig. 1, and a smooth line or curve drawn connecting the plotted points. The independent variable is usually plotted with the *x*-axis scale, and the dependent variable with the *y*-axis scale. Fig. 2 shows a plot of the equation $y = 3x + 2$. This is a linear equation, so its plot is a straight line. It will be noted that the value of the dependent



x	-4	-2	0	2	4	6
y	-10	-4	2	8	14	20

Fig. 2. How an equation can be plotted.

variable, *y*, can be read from the chart for any value of *x*, even those not used to plot the line.

Trigonometric Functions

Trigonometry is the branch of mathematics concerned with the interrelations between the angles and sides of triangles, and the application of these relationships to the solutions of problems. We need be concerned with only a few special applications of this fascinating study.

The compass, as everyone knows, is marked off into 360 divisions called degrees. Any circle can be similarly divided around its circumference. Now, the intersection of two straight lines forms four angles, as shown in Fig. 3. If we draw a circle with its center at the intersection, and mark the circumference off in degrees, the intersecting lines will cut the circle at certain points along the circumference. The size of the circle doesn't matter; if it were larger or smaller, the lines would cross the circle at the same degree marks. Moreover, if the circle were rotated so that the lines cut it at different degree divisions, there would be the same *number* of divisions

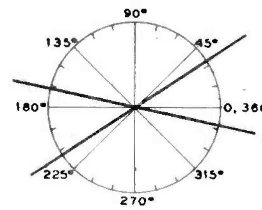


Fig. 3. Angles mark off circular degrees.

between the crossing points. It is evident, therefore, that angles can be measured in circular degrees, and so they are. Fig. 4 shows some angles of various sizes. Comparison with the circular scale in Fig. 3 will confirm the indicated degree marks.

A triangle always has three included angles, and the sum of these angles is always 180°. Fig. 5 shows a special type called a right triangle. It is so known because one of the included angles is 90°, or a right angle. The sum of the other two angles, then, must be 90°.

In a right triangle the side opposite the right angle is the hypotenuse. Let this be labeled side A, Fig. 5, and the other sides B and C. For the angle θ, side B is opposite and side C is adjacent. There are six ways in which the three sides can be combined as fractions; these are known as the trigonometric functions. They are the sine, cosine, and tangent, with their reciprocals the cosecant, secant and cotangent, respectively. Their definitions and abbreviations follow:

$$\sin\theta = \frac{B}{A} = \text{opp. side} \div \text{hyp.}$$

$$\cos\theta = \frac{C}{A} = \text{adj. side} \div \text{hyp.}$$

$$\tan\theta = \frac{B}{C} = \text{opp. side} \div \text{adj. side.}$$

$$\csc\theta = \frac{A}{B} = \text{hyp.} \div \text{opp. side.}$$

$$\sec\theta = \frac{A}{C} = \text{hyp.} \div \text{adj. side.}$$

$$\cot\theta = \frac{C}{B} = \text{adj. side} \div \text{opp. side.}$$

There are a number of interesting relationships that might be investigated in a mathematics text. It is obvious, for instance, that the sine of the angle formed by sides A and B is identical to the cosine of θ, and vice versa. But here the discussion must be limited to sin θ. In Fig. 6 is drawn a series of triangles with the hypotenuse of equal length in each, showing how the lengths of the other sides vary as θ is increased from zero to 90°. Let us say that the hypotenuse is 1 unit in length; since sin θ is the opposite side divided by the hypotenuse, then sin θ will be numerically equal to the length of the opposite side. At Fig. 6A θ is zero, the opposite side has no length at all, and sin θ is therefore zero. In Fig. 6B θ is 30°; the opposite side is ½ the hypotenuse in length, and sin θ is accordingly 0.5. When θ is 45°, Fig. 6C, the length of the opposite side (and sin θ) is 0.707. This increases to 0.866 for θ = 60°, Fig. 6D. Finally, at Fig. 6E, when θ reaches 90°, the opposite side is as long as the hypotenuse. Sin θ is then 1.

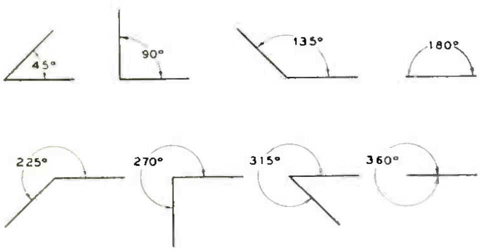


Fig. 4. Some angles of typical sizes.

From these representative values it can be seen that sin θ varies in magnitude from zero to 1, but can never be more than 1. Further, the increase in sin θ is not linear with increasing values of θ; that is, a direct proportion does not exist between θ and sin θ. The other functions of θ, listed previously, all change in magnitude also as θ is varied, although in different ways. Tables are available which give exact values of the trigonometric functions for small increments in θ. With them, and with a few simple formulas, all sides and all angles of any triangular construction can be found when only two sides, two angles, or one side and one angle are known.

The Sine Wave

To see exactly how sin θ varies with θ from zero to 360°, let us employ the construction diagrammed in Fig. 7. This is a plot in rectangular co-ordinates of the formula $x^2+y^2 = 1$. The "curve" obtained is a circle whose radius is 1 and with its center at the origin. Angular degrees are marked off around the circle at intervals of 15°; the right-hand section of the x-axis coincides with the zero and 360° marks. A radius is drawn from the origin to each division on the circle.

The length of this radius is constant, always equal to 1. Each position of the radius corresponds to the hypotenuse of a right triangle, and θ is the angle made by the hypotenuse with the right half of the x-axis. *The opposite side of the triangle is, then, the ordinate of the point on the chart at which the radius touches the circle.* The construction is darkened for illustration at θ = 45°, for which the ordinate is 0.707, and again at θ = 240°, for which the ordinate is —0.866. Now, since the hypotenuse is 1, and the sine of θ is the opposite side divided by the hypotenuse, and the opposite side is always the ordinate of the point of contact between the radius and the circle, it follows that sin θ is also equal to the ordinate of that point. This is merely an extension of Fig. 6.

The next step, as shown in Fig. 8, is to plot sin θ as a variable dependent on θ. Beginning with zero at the origin, the x-axis is scaled into 15° divisions just as the circle was, representing the successive positions of the radius, or θ. The y-axis is calibrated to +1 and —1, and will be used to plot the amplitude of sin θ. At zero degrees sin θ is zero, so the first point plotted is 0,0. At 15° sin θ is 0.259, so the second point to be plotted is 15,0.259. At 30° sin θ is 0.5; the third point is, therefore, 30,0.5. So on up to 90°, when sin θ reaches its maximum of 1. Beyond 90° it decreases, reaching zero again at 180°. It continues downward into the minus region; all ordinates between 180° and 360° have negative signs and, since the radius is assumed to be always positive, the values of sine functions for these angles are negative. The maximum negative value, —1, is reached at 270°, after which sin θ becomes less negative, reaching zero at 360°. When these plotted points are joined as shown, a sine wave is produced.

Obviously, if this process were to be repeated — if we were to plot sin θ for another revolution of the radius or hypotenuse — we should obtain exactly the same curve repeated. Here one revolution, or one cycle, of the radius corresponds exactly to one cycle of a sine wave. Now, suppose that the radius rotated continuously at a constant rate; say, 3,600 rpm. This would produce 60 complete rotations or cycles every second and the frequency of this sine wave would be 60 cps. Its period would be 1/60 second, or .0167 second.

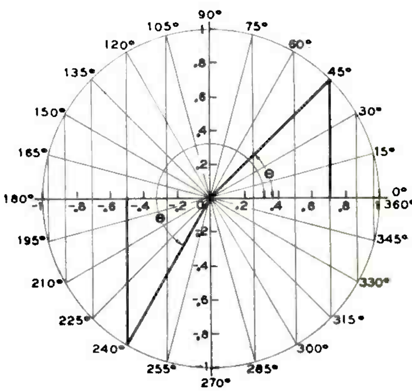


Fig. 7. How sine value varies with angle.

What has this to do with basic electronics? Quite a lot. For now, visualize a coil rotating in a uniform flux field produced by an electromagnet. During that part of the coil's rotation at which it lies in a plane perpendicular to the flux field, it would be cutting no flux lines, and the output voltage would be zero. Let us call that point zero electrical degrees. As it travels on it begins to cut flux lines at a faster and faster rate until, when it lies in a plane parallel to the flux lines, it is cutting them at maximum rate. At that point, corresponding to 90°, maximum output voltage in one direction is achieved. Beyond that point the output voltage begins to decrease, because the flux-cutting rate decreases, until at 180° travel the coil is again perpendicular to the flux; output again is zero. Now, beyond that point, the output voltage again begins to build up — but the coil is cutting flux in the opposite direction than before, so the voltage produced at the coil terminals is of opposite polarity. This negative

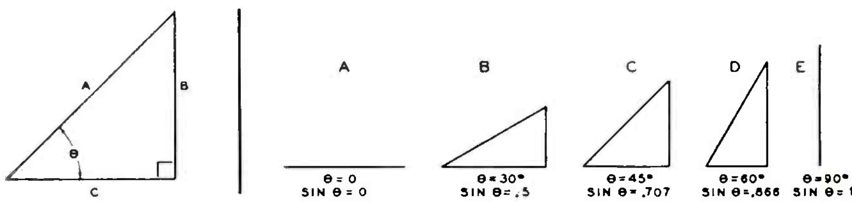
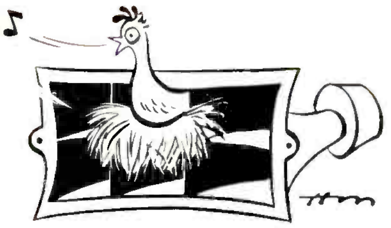


Fig. 5, left, and Fig. 6, right. Right triangles. See text for full discussion.

Continued on page 37



Sound-Fanciers' Guide

by R. D. DARRELL

IF YOUR eyes are as sharp as your ears should be, you must have noticed the neglect of a popular category of demonstration and display recordings in last month's survey of percussion, ultra-brilliant symphonic, and various queer-sound discs and tapes. Like most so-called long-hair reviewers, I completely ignored the poor man's substitute for an orchestra — the theater organ, or Mighty Wurlitzer, with its Plenty of Traps, Bells, etc., which usually contributes a lion's share of the decibels at audio shows. For that matter, I also passed over the more serious manifestations of the King of Instruments — the baroque, classical, and modern symphonic organ recordings prized by many audiophiles not only for their strictly musical interest, but also for the ability of their incidental 32-foot pedal tones, distinctive-stop timbres, and spacious sonorities to demonstrate the versatile powers of any truly wide-range home sound system.

I, at least, was conscious of these omissions, and of my own general disregard of the extensive recorded organ repertory in the past, and I resolved to amend matters promptly. For, although I'm by no means an organ devotee and long have deliberately closed my ears to whatever siren charms the public apparently finds in theater organs in particular, I realize that I may have been missing a great deal of special audio significance. Since I pride myself on my aural catholicity (liking to paraphrase Terence's famous "*Homo sum*," etc., as "I am a listener; and nothing sonic is alien to me!"), I have doggedly spent a good part of the last month investigating a wide variety of organs, organists, and organ music as represented on current and fairly recent discs and tapes.

It's been a hard but revelatory workout, for both my electronic and neural equipment! I must admit that my capacities for aesthetic if not strictly aural tolerance were sorely stretched by my first direct home encounters with theater organs in full blast. In fact, the *Mighty Wurlitzer* of Gordon Kibbee's Vol. 1 (Omegatape 7009; also on Starlite LP, ST 7002) offered the plodding, tasteless, popular-tune performances I feared, enlivened by little of the thunderous or exotic sounds I hoped for. The recording itself is unquestionably brilliant, perhaps even exaggerating the highs, but the arrangements and registrations here reveal no boldly imaginative attempts to exploit it fully.

It was only when I turned to George Wright's *Encores* on another Wurlitzer (Hi-Fi Tape; also Hi-Fi Record R 702) that I realized what the shouting is all about. Again, the recording is ultra-crisp and clean, but here it is given a real chance to show what it can do with a fancy variety of tinkles, booms, and other novelty sound effects — not excluding a real, if monstrously blown-up, whippoorwill. Moreover, while Wright is no less an aesthetic vulgarian, he at least frankly exults in his vulgarity and displays it with uninhibited verve and assurance. I still don't like it. But I certainly can't deny that it exerts a lively if horrid fascination — and that considered only as sound, rather than as music, it makes a glorious racket which the best of home systems will have to work hard to reproduce without distortion, color dilution, or rounding-off of steep wavefronts.

Still Lower Depths

An injection of Wright's jauntiness and confidence might help a supposedly more serious organist, Richard Purvis, who

presently seems to equate seriousness with stodginess in two recital programs on the San Francisco Grace Cathedral organ (Hi-Fi Tapes, either single channel or stereo; also on Hi-Fi Records R 703-4) His instrument, a modern Aeolian-Skinner, displays some admirably big sonorities and wide dynamic ranges in a recording which has more glitter than warmth. But his thickly registered tonal mixtures seldom exhibit any pure or highly distinctive colors, and while he plays a bit of everything, from Bach (arranged) and Purcell, through Dupre and Widor, to a number of his own pieces or settings, the best of it is dull and the worst pretentious. Hearing the first volume in its stereo version made some of the more strident qualities far more tolerable and enhanced the dynamic contrasts, but also made Purvis's own sentimentalities even more obvious.

I'm afraid even stereo wouldn't help Virgil Fox's *Bach Program* on the New York City Riverside Church organ (RCA Victor LM 1963). The rich sound of this rather romantic instrument hardly could be more beautifully recorded than it is now (in single channel), and no engineering possibly could ameliorate Fox's mincing rhythms, exaggerated dramatics, and propensity for schmaltzy effects. Some of the little pieces and transcriptions actually are brought down to funeral-parlor background-music levels; the big ones (*Toccata and Fugue in D minor*, *Prelude and Fugue in D major*) are misconceived when they are not utterly travestied.

But this marked the nadir of my organological investigations. Turning to Kurt Rapf's recital on the Vienna Piaristenkirche organ (Audiosphere 8, or on two stereo reels, Audiosphere 711-2 BN), I found far less "virtuoso" playing on an obviously antiquated and noisy-action instrument. Yet, while his performances are orthodox enough (unsteady and faltering in the great Bach *Passacaglia*, as a matter of fact), he is obviously a sincere musician and elicits some delightfully attractive tonal qualities from the many quite distinctive stops of an extremely interesting (sonically, at least) organ. His two Mendelssohn sonatas are able representations of what to me is music of no great character, but his Franck *E major Choral* has fine lyrical warmth and a truly radiant triumphal conclusion.

I heard the stereo tapes of this recital and, while I want to defer any general notes on stereo to a later column where it can be treated more fully, I should contradict immediately the general misconception — which I had expected to share — that stereo is not particularly effective with the organ. To be sure, it makes little difference as far as the actual organ tones themselves are concerned, but it does notably enhance the acoustical setting of these tones, and gives the whole performance a far more open, spacious, and gracious naturalness.

Around Europe in 23 Organs

Then, my attention whetted by the intriguing qualities of many of the Piaristenkirche organ stops, I set off with E. Power Biggs to visit some 20 different historically or otherwise famous European instruments in *The Art of the Organ* (Columbia SL 219), and 11 of these, plus 3 others, in Bach's *Toccata in D minor* (Columbia ML 5032). I have not greatly admired Mr. Bigg's highly popular releases of recent years, but these revive much of the fresh, substantial appeal of his long-ago first ones, and indeed warrant some kind of gold-medal award for an inspired notion, ably executed: a major contribution to listeners' better understanding not only of organ characteristics themselves, but also of the intimate relationship of these to specific church acoustics, particular musical selections, and individual interpretative styles. I can recommend highly every earnest sound-student's attentive, repeated hearing of these discs and reading of Edward Tatnall Canby's illuminating notes for SL 219, which follow Briggs's own pleasantly informal and attractively illustrated, but far less usefully informative, travel diary.

What we have here is, in the album set, an arresting variety of instruments and acoustical environments, more-or-less (but mostly more) suited to the great early periods of organ writing, heard in appropriate musical examples largely by Sweelinck (Dutch organs), Pachelbel (Southern Germany), and Buxtehude (Northern Germany and Scandinavia). Both music and combined organ-and-environment qualities range through a whole gamut of greater and lesser attractions, of course, but at their best they are unqualifiedly magnificent. Personally, I responded with perhaps greatest enthusiasm to the enchanting Schnitger organs at Neuenfelde (1682) and Steinkirchen (1540-1685), the deliciously delicate and ripe tones of the 1725 Amsterdam organ, and the quaint breathy stops of the 1730 Drottningholm organ; but I also was profoundly impressed by such more recent instruments which recapture much of the essence of baroque tonal styles as effectively as the 1951 Amstelveen organ, the 1929 Trondeim and 1636-1945 Luebeck instruments with their massive sonorities (and superb pedal stops), and the 1952 Nuernberg organ. My one really adverse criticism is that my own favorite of all organs, the nontempered 1612 Compenius instrument at FrederiksborgCastle in Denmark, is not represented by such fascinatingly raw colorations as in the memorable Videro 78's of some years back.

Yet by concentrating on a single well-known piece of music, Bach's *Toccata in D minor*, ML 5032 enables us to analyze the stops and acoustics of individual organs even more closely and accurately. While this is an audio text book record, rather than one for sheer pleasure, I can't imagine any more enjoyable (or more effective) medium for enlightening sonic study. The recording itself is competent rather than extraordinary, but that is all the better in some ways, for it focuses our primary attention on the sound qualities and the ways in which they are further colored and enriched by the auditoriums for which they were specially designed.

Completing the Course

The next step in the sonic education of any audiophile who desires a better theoretical as well as practical understanding of organ qualities is to go back to — if he doesn't know it already — the first volume in the Aeolian-Skinner *King of Instruments* series. The seven later releases are mostly orthodox recital discs of considerable but hardly unique merits, but No. 1 is a sonically illustrated lecture by G. Donald Harrison on "The American Classic Organ" which compresses a whole course in organ-stop identification and description into two LP sides. It is done in truly masterly

fashion, and as a bonus for wide-range hi-fi fans it includes a demonstration scale (on the Boston Symphony Hall organ) which ranges from an almost toneless pulsating 16 cps up to an 8,000-cps fundamental with harmonics far above that. How well does *your* home sound system handle this?

The final step is to bring your now enormously expanded knowledge of organ stops, techniques, and acoustics to what are for me the most impressive examples of organ recordings achieved to date. These represent, too, almost the extremes of both instrumental design and interpretative approach, for one is a Bach program played with dazzling virtuosity by Clair Coci on an immense modern "symphonic" organ, that of the West Point Cadet Chapel, the world's fourth largest and, like Miss Coci, heard here for the first time on records (Vox DL 210; also Phonotapes-Sonore PM 140 in preparation); and the other two are also Bach programs, performed in no less brilliant but more rigorous and less rhapsodic style by Carl Weinrich, on the baroque-type reconstructed organ of the Skaenninge, Sweden, Church of Our Lady organ (Sonotape SW 1011 and SW 5002; also partially available, in different couplings, on Westminster W-LAB 7023 and WN 18148).

Apart from the common links of instrumental genus and composer (and superb executant and electro-acoustic techniques), no greater contrasts in tonal qualities and treatments could be imagined which would still fall within the bounds of high artistry. Miss Coci is boldly and freely expressive, and the West Point organ is extraordinarily rich, mellow, and big-toned; while Weinrich achieves even greater dramatic impact by more rigidly restrained means, and his Swedish organ is characterized by the quintessence of baroque color rawness, roughness, and italicized differentiations.

In the elaborate booklet for the Coci disc, I sketch some of my own justifications for hearing Bach's organ works on modern instruments as well as on those of the general baroque type for which he specifically composed. But individual listener ears are the final justification — if, indeed, any is needed. Hear for yourself the mighty D minor *Toccata and Fugue*, C minor *Passacaglia*, and A minor *Prelude and Fugue* in the golden sonorities and broad climaxes of the Coci-West Point versions . . . Then hear the *Toccata and Fugue* again, the "St. Ann" *Prelude and Fugue in E flat*, C major *Toccata, Adagio, and Fugue*, and other works in the brisker, more incisive, and even harshly powerful Weinrich-Skaenninge versions. Either experience is a memorable one; the two in contrast provide incomparable sonic and aesthetic revelations.

Which you prefer is almost beside the point. And anyway, don't be too sure, until you have lived with both for some time, that your original choice is likely to retain its primacy. The Weinrich approach, simply because it may be more startling and unfamiliar to orthodox present-day ears, perhaps will be the more difficult for many listeners, but personally I feel it offers the more profound, and certainly the more electrifying rewards. But I won't argue the point, for it depends too much on individual aural training and temperamental affinities. The main thing is that *both* approaches be studied closely and sympathetically, for if that is done I can guarantee that you will gain from these works (as I have) a vastly enhanced respect for the inexhaustible potentialities of organs and organ music, as well as for the overwhelming clarity and brilliance with which the best of these qualities can be captured on discs and tapes by the finest present-day engineering techniques.

Organic . . . Pathological . . .

The sound of organs can be of vital import also to the custodians of human health. And as connoisseurs of all kinds of sonic phenomena, we too can find aural fascinations when we turn from music to listening-in, via a stethoscope, to the sometimes sadly arrhythmic beating, wheezing, and growling of hearts, lungs, and intestines! It's a startling, if a rather unnerving, experience to hear them in both diseased and normal states in *Sounds of Medicine* (Folkways FPX 127), which also contains a sonically less interesting but otherwise absorbing sound-scene of a surgical operation. Obviously of immense documentary and training value for medical students, the stethoscope recordings also comprise a rare contribution to the complete audiophile's ever-expanding library of odd sounds.

The ingeniously compiled Folkways Science Series also ventures outside legitimate bounds to explore the pseudo or bogus sciences. Its *Vox Humana* (FPX 123) reaches into the same cloud-cuckoo realms as those in which the devotees of Bridey Murphy, flying saucers, and ESP rapturously disport, to exhibit Alfred Wolfsohn's zany experiments in stretching normal vocal ranges both to the ultrasonic heights of an Yma Sumac or Erna Sack and to subsonic depths even below those plumbed by Russian *basso profundos*. Seven, eight, or more octaves? Nothing to it, says Wolfsohn — just listen to what my pupils can do! Well, I've listened, and what I hear is mostly hiccups at the low end and peanut-whistles at the high end. This is singing? Wolfsohn goes on to somewhat (but not much) more orthodox demonstrations of vocal "coloration" and instrument imitation, which are just plain rather than fancy awful. These exhibits must be heard to be believed — and then I don't. But they are not to be missed, if only for the accompanying booklet, in which a quite sensible plea by Henry Cowell for further study of human-voice potentialities is followed by disciple Leslie Shepard's paean to Maestro Wolfsohn's theories and accomplishments. And this wins, hands down, the Darrell Award for Supreme Obscurantism in Record Annotations. I bitterly regret having to pass over the memorable achievements of Wanda C. Von Rudolph-Ronty and other notable practitioners of the mystic art of verbal obfuscation, but Shephard is strictly *ne plus ultra*. As he (or she) pertinently inquires: "Where will it end? And what does it mean?"

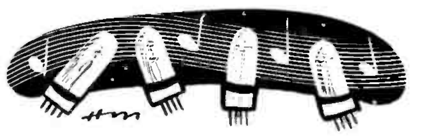
. . . and Synthetic

When engineers rather than metaphysicians begin dreaming, the results are much more substantial, but should be subject to no less dubious scrutiny. At an IRE Convention over a year ago I heard Dr. Harry F. Olson lecture on and demonstrate, via recordings, the RCA Electronic Music Synthesizer — and deemed it (to borrow the words of a not-entirely-awed fellow listener) one of the most "fascinating monstrosities" I'd ever heard. The substance of that lecture is repeated, by John Preston, and the musical examples are displayed on RCA Victor LM 1922, to permit you to judge the new invention's potentialities for yourself. To my mind it has genuine musical possibilities, which are scarcely hinted at here; probably best are the pieces "in the style of" a clavichord, hill-billy ensemble, and (decidedly corny) dance band, where some of the percussive sounds in particular seem well worth further development. Curiously enough, the main thing wrong is closely akin to the failings of Wolfsohn's *Vox Humana* ensemble performances: completely amateurish direction in a musical sense. Given better aesthetic supervision, the synthesizer at least might become a truly valuable contributor to our expressive sound-producing means; meanwhile, it remains little more than a singular, if astronomically elaborate and expensive, toy.

Nevertheless, this disc is still of genuine value to audiophiles. The illustrated lecture on Side 1 is one of the best and most informative brief analyses of sound characteristics I have yet come across. No one who wants to learn more about the nature of sound and how its objective factors stimulate our subjective aural perceptions can afford to pass it by.

Well-Earned Relaxation

If you're still with me after all this concatenation of controversial man- and machine-made sounds, we both deserve an aural rest — and nothing could be more relaxing than to settle back in an easy chair and absorb, rather than listen to, the soothing murmurs of falling or tide-borne water. But the skies are clear, you protest, or you live far from the seashore? No matter. Emory Cook is right on hand with No. 002 in his White Label series of Microfusion pressings: some 25 uninterrupted minutes each of *Rain and Surf*. Just trickles, drips, gurgles, and surges — but they're so real you'll want to raise an umbrella over your turntable and you'll distinctly scent salt breezes emanating from your speaker!



Or let the versatile Emory and his equally indefatigable peripatetic pal Sam Eskin transport you to Yemenite Israel, Trinidad, and Cuba for three authentic dance-and-musical *Rituals* (Cook 1043). If they're just too voodooishly exotic for your tastes, join a different *Cook's* (Thos. & Son, not Emory) *Tour of Latin America* (Vox 25110), with Jose Valdes y los Embajadores beating out vivacious Latin rhythms. And for kindredly infectious toe-tickling materials, spin with the reels of Omegatapes 5004 and 5006 to Carlos Molina's *Exotic Cha Cha Cha* (some also on Alphatapes 4 and 8) and Marco Rizo's *Piano Havana*. All these are first-rate recordings, but you're an even more fanatical audiophile than I am if you can concentrate on their technical qualities at the expense of their light-weight but irresistible musical charms. After all, does listening always *have* to be wholly serious or hard work?

OTHER REVIEWS

Mozart fanciers are surely getting more breathless by the minute trying to keep abreast of the vast quantities of the master's works presently (and happily) flooding the market. This is a particularly tough problem to the opera buyer from purely economic considerations (it costs less to replace a single-record symphony than a multi-record opera!). Fortunately, there is at least one Mozart opera now so definitively recorded that it seems extremely unlikely to be surpassed soon, musically or sonically. The new Decca *Magic Flute* has typically first-rate Deutsche Grammophon sound (close-up, shimmering strings; singers a little breathy) which seems a perfect complement to the shining artistic merits of the album. Vocal honors must go first to Rita Streich for her stunning coloratura pyrotechnics as Queen of the Night. Dietrich Fischer-Dieskau sings an engagingly masculine and aggressively good-natured Papageno. Maria Stader's Pamina sounds limpid and is histrionically secure. The Tamino of Ernst Hafliger is a bit thin but of the desired lyric quality. Josef Greindl's Sarastro is well thought out, but he, unfortunately, does not have sufficient range to sing the part easily (not an uncommon complaint). Inclusion of the spoken dialogue is really necessary to a completely satisfactory performance; this is the only recorded version having it. Ferenc Fricsay leads the whole affair in a strong, deft, light-hearted reading making, in the final analysis, the major contribution to this album. — W.B.S.

One of the most ear-pleasing records I have ever heard, bar none, is Mercury MG40015: *British Band Classics*. Frederick Fennell conducts the Eastman Symphonic Wind Ensemble in Gustav Holst's *Suite No. 1 in E-flat* and *Suite No. 2 in F*, and Vaughan William's *Toccata Marziale* and *Folk Song Suite*. This is witty, captivating music in excellent taste that can be enjoyed fully without undue mental effort. It is performed beautifully and recorded flawlessly here. The record is well worth having simply for the educational value it offers; most listeners not familiar with concert band music will be surprised at the wide range of tonal colors and harmonic textures revealed in this disc. But it will be played again and again for the wonderful spirit of the performance, the really exceptional quality of the recording, and most of all for the lovely, lighthearted music. — R.F.A.

Handle Bars for Phono Plugs

Until some manufacturer comes to the rescue, here's a way to tame the intractable phono plug.

Remove one inch of outer insulation from the phono cable and unbraid the ground shield. Divide the strands into two equal parts on opposite sides of the cable and twist them into two tight ropes. Remove ¾ in. of insulation from the center wires, tin the wires quite heavily, insert in the phono plug, and solder them securely.

The ropes of shielding should be bent back against themselves to form two handle bars ½ in. long. Then twist them tightly in the reverse direction from the first twist, just as rope is laid up. There should be from ¼ to ⅜ in. of the single twists lying against the outer shell of the phono plug. Solder these to the shell, and soak the entire handlebar construction with solder. This will give a nice finger grip and, because of its solidity, will prevent the fragile wires in the center of the cable from breaking off with continued flexing.

Carl F. Propson
Lumberton, N. J.

Speaker-Cabinet Lining

It is a well-known fact that the better the cabinet, the better the sound. While better means bigger (up to a point), it means also rigidity of construction and virtual elimination of panel resonance. If you can feel the sound when you put the flat of your hand on a panel, panel resonance is present. A near-zero resonance will result in much cleaner sound.

Near-zero resonance is possible without recourse to brick and concrete or even sand-filled panels. Such installations are fine when built permanently into new construction where the room is acoustically designed, but they are not for the mass of audiophiles. A mastic composed of asphalt and asbestos, either roofing cement or automobile undercoating, liberally sprayed over the interior paneling will kill the panel resonance. Automobile undercoating may be obtained for a very nominal sum at any garage that undercoats cars. Having the coating sprayed on is better than applying it yourself with a brush or trowel. The coating should be about ¼ in. thick. Allow it two days for drying; then cover it with Kimsul. The improvement in the quality of sound will be more than worth the effort.

Albert Sadler
San Diego, Calif.

Cleaning Tape Recorders

Although most owners of tape recorders know that the heads, tape guides, and rubber capstan roller should be free from an accumulation of tape oxide, many recorders are not cleaned because some dismantling of the machine must be done in order to reach the heads and capstan.

Most drug stores sell a cotton swab with the trade name of "Q"-Tip. "Q"-Tips are small enough to pass through the tape loading slot of most tape recorders; the heads, etc., can thus be cleaned without any dismantling.

A "Q"-Tip slightly moistened with Audio Devices Head Cleaner will clean a rubber capstan roller faster and better than a lot of alcohol and a lot of scrubbing. One word of caution: head cleaner may mar a plastic finish, so be careful when using it.

Herbert Friedman
Brooklyn, N. Y.

60-Cycle Stroboscope

Used in conjunction with a strobe card (available from most hi-fi shops), a 60-cycle stroboscope will insure that a turntable is operating at the correct speed.

Construction of a stroboscope is very simple. A Ne-O-Lite, or similar neon electrical tester, is connected to a 6-foot length of line cord. The other end of the cord is fitted with a male plug and inserted into any AC outlet. The Ne-O-Lite indicator is held over the turntable on which the strobe card has been placed, and the turntable speed control is adjusted to make the bars on the strobe disc appear stationary at the speed in use. The Ne-O-Lite electrical tester is made by the Ne-O-Lite Company of Rockford, Illinois, and may be purchased at most hardware stores for about 50¢.

George T. Mitchell, W7ZQX
Seattle, Wash.

Record Brush

Some time ago, I purchased a brush at a camera store to keep my camera lens clean. Then I tried using this same brush on my records and was amazed at the large quantities of dust the brush picked up from the record grooves.

The brush is small, about the size of an ordinary lipstick tube, and works on the same principle, having a retractable inner tube and a cover for the protection of the hair when the brush is not in use. To remove dust from a record, hold the brush perpendicular to the record and allow the hair to ride in the grooves as the record revolves on the turntable. Move the brush slowly and lightly across the record from the outside to the center, letting the hair do the work of picking up the dust. Do not apply too much pressure.

Once the factory seal has been opened it is never possible to remove *all* the dust from a record. But this brush method used once or twice before every play will do much to minimize record wear and stylus damage.

R. D. Hildebrand
Schenectady, N. Y.

Control-Knob Marker

Most control knobs on hi-fi equipment today are round, having only a small, almost inconspicuous dot to show their position. It takes a good light to make this dot visible.

By drilling out the marker dots and putting a small, round-headed escutcheon nail in the hole, it is possible to tell exactly the position of the knob by feeling. Moreover, the round brass head of the nail detracts in no way from the appearance of the knob.

R. H. Shevenell
Ottawa, Ont.

Head Demagnetizer

Many soldering guns, such as the Weller *Junior*, radiate from their tips a fairly strong AC magnetic field which can be used to demagnetize the record or playback head of a tape recorder.

The tip of the soldering gun should be cleaned thoroughly to prevent solder from getting on the heads and pads. Then the gun is switched on and moved about the area of the head as close as possible (within 1/4 in.). The gun, still operating, is then slowly withdrawn from the head for a distance of about a foot and switched off.

Jon G. Montgomery
Eldorado, Ill.

Machine Screw Sizes

On page 41 of the March issue of AUDIOCRAFT Mr. Fowler went into considerable detail about the diameters of machine screws. However, he left us in the dark as to what the actual relation is between the numbers and the sizes.

Having worked with these screws for many years, I thought that perhaps the following information might be of use. The basic size to remember is #5, 0.125 in., or exactly 1/8 in. Each size higher or lower than that is plus or minus just .013 in. So we have:

Size		
0	.060 in.	
1	.073 in.	
2	.086 in.	
3	.099 in.	
4	0.112 in.	
5	0.125 in.	1/8 in.
6	0.138 in.	9/64 in. approx.
7*	0.151 in.	
8	0.164 in.	5/32 in.+
9*	0.177 in.	
10	0.190 in.	3/16 in.+
11*	0.203 in.	
12	0.216 in.	7/32 in. —
13*	0.229 in.	
14	0.242 in.	1/4 in. —

*Not in commercial use.

Leslie F. Garrett
Derry, N. H.

Finish for Woodwork

All woodworkers dream of constructing a device that not only operates well, but also looks nice. A speaker cabinet may be the latest thing as far as design is concerned, but the chances are that the lady of the house won't even listen to it unless it's good-looking and fits in with the decorative scheme of the home. This goes for all other home-made woodwork as well — it must fit in with the decor.

Often this means that the device must be made of mahogany, walnut, oak, or maple — all woods notoriously difficult to work with. Second-grade white or yellow pine may do the job very well, and be easy to work with and cheap besides; but they are taboo if looks are to be considered.

Fortunately, there is a simple and inexpensive solution to this problem. Make the speaker enclosure or other piece of equipment out of some soft, easily worked wood, and then apply a decalcomania over it to make it resemble the wood of your choice.

These decalcomanias are actually high-quality photographs of good specimens of wood, made under ideal conditions. The photograph is reproduced on a tough, pliable, plastic sheet that can be glued to the finished product with a minimum of effort. If done well, the result is a finish that can't be distinguished from the real thing.

There are a few such decals on the market; the most popular is *Plastic Veneer*, made by the Meyercord Company of Chicago, Illinois, and distributed in department and do-it-yourself stores over the country. Many types of woods, in different shades, are available along with a few odd items such as cowhide and marble. Complete instructions, including a number of tips for covering odd-shaped structures, come with each box of the material.

It should be stated, however, that to get good results it is absolutely necessary that a perfectly smooth surface be obtained prior to applying the Plastic Veneer. This means — and there's no getting around it — elbow grease and sandpaper. Applying Plastic Veneer is more work than merely staining the wood used in construction and polishing it. It is necessary to apply a coat or two of varnish to seal the wood, and then a coat or two *over* the Plastic Veneer for a first-class surface. However, the results are startling.

The wood-working audiophile should not miss out on the opportunities offered by this material.

Paul Penfield, Jr.
Birmingham, Mich.

Pilot Light for Heathkit Preamp

Many owners of the Heathkit WA-P2 preamplifier wish it were equipped with a pilot light to indicate whether the power is on or off; particularly is this true when the main amplifier is mounted out of sight.

Adding a pilot light is not necessarily as simple as it may seem at first. The hum level of the WA-P2 is extremely low and one wants to keep it that way. Since the main power leads to the AC switch are completely shielded, it would be difficult to tap into them without increasing hum. A similar risk would arise if the 6.3-volt AC filament leads were tapped to power a pilot light.

My solution is both attractive and relatively easy. It requires no shielding, wire twisting, or dressing wires close to the chassis. Four holes are drilled through the front panel (and the clamp behind it), and a #110 *Flushlite* neon light installed above the treble control with two 6/32 brass bolts. One lead goes to the terminal marked with a rectangular oblong on the filter condenser can, providing 140 volts DC. The other lead is connected to the ground terminal of the hum balance control. All joints should be soldered. Be sure to put sleeving over the joints where the lamp terminals go through the panel.

The pilot light functions perfectly without raising the hum level of the unit. Because it draws less than 3 ma current, it does not appreciably load the DC power supply. An interesting effect arises from the fact that the NE-2 bulb in the Flushlite does not fire under about 90 volts DC; it does not start glowing

Apperance of the Flushlite pilot light.

until the rectifier tube in the power supply warms up and the filter capacitors have become fully charged. With my W-5 amplifier, which has an extremely well filtered power supply, the lamp does not light until 15 seconds or so after the power is switched on, and it keeps glowing for about 7 seconds after the power is turned off.

D. H. Brooks
Trenton, Ont.

Twisting AC Heater Leads

In constructing amplifiers and preamplifiers, as well as test equipment, AC heater leads should be twisted and tucked into the corners of the chassis to avoid hum induction in other parts of the circuit. When this is done by hand the leads are often loosely wound, and some of the usefulness of the winding in cancelling possible contaminating currents is negated. Loosely wound leads also present a sloppy appearance.

In order to wind the leads uniformly and tightly, cut off a length of wire about three times the required length. Use solid wire. Bend the wire in half and place the end with the bend in it in the jaws of your electric or hand drill. Clamp the two loose wires at the other end in a vise. Start the drill and let it run until the wire is uniformly twisted. Don't overtwist or the wire will break.

When this procedure is properly done, the result is a very uniformly twisted wire which will present a professional appearance, can be bent as needed, and will offer the circuit a minimum of AC contamination.

Ben Zale
Bronx, N. Y.

AUDIO AIDS WANTED

That's right — we'll pay $5.00 or more for any short cut, suggestion, or new idea that may make life easier for other AUDIOCRAFT readers, and which gets published in our Audio Aids department. Entries should be at least 75 words in length, and addressed to Audio Aids editor. No limit on the number of entries.

LOUDSPEAKERS

Continued from page 18

the high frequencies? Fig. 6 is a sketch of such an arrangement in cross section. The Electro-Voice SP-12 is shown in Fig. 7. The inner "whizzer" cone is driven by the same voice coil as the woofer section, but the mechanical constants are arranged to separate the highs into the little cone and the lows into the big one.

The idea is not original. P.G.A.H. Voigt designed an excellent twin-cone speaker over 20 years ago, and a modifi-

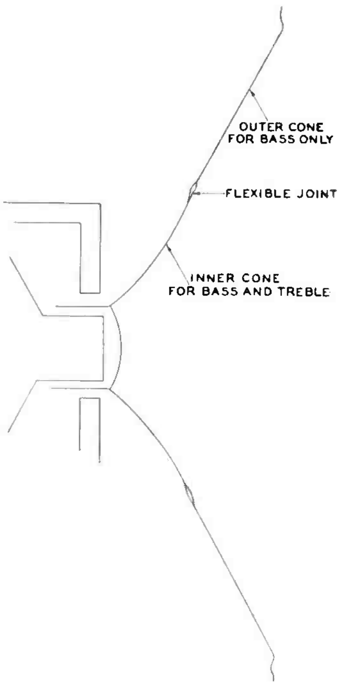


Fig. 10. Construction of the Biflex cone.

cation of the original Voigt speaker is now built by Lowther in Britain and used in Brociner speaker systems in this country. Another British firm, Goodmans, has used the auxiliary cone in high-quality speakers for years, but it was not until after World War II that Electro-Voice adopted the idea in the United States.

There are pros and cons to the merits of the whizzer cone too. H.A. Hartley, one of the pioneers in quality speaker design, has this to say about the twin cone: "The main objection to this device is that the outer edge of the small cone is free to resonate at a frequency depending on the material of the cone and its diameter; more generally, the effect resembles that of modulation noise in tape recording, wherein a high-frequency modulation of the wave form produces a 'zizzy' coloration of high frequencies, particularly of stringed instruments."

Hartley's own speaker (Fig. 8) is the original version of the Altec Lansing "Biflex" which has just been introduced. The Biflex design is shown in Fig. 9. The idea, stated quite simply, is that at low frequencies the whole cone moves together as a piston, but at higher frequencies only the inner cone vibrates, while the mass of the outer section keeps itself stationary. A cross-section of the general design is shown in Fig. 10. The action of this type of segmented cone speaker might be described as "controlled cone breakup."

The cone shape, type of suspension, method of introducing proper compliance, and so forth is quite different in the two speakers pictured. The question of optimum design for this type of wide-range speaker has already been the subject of an exchange of comments between Mr. Hartley and Alexis Badmaieff of Altec Lansing. Harry F. Olson has also done extensive theoretical work on the segmented-cone speaker and his findings are incorporated in RCA's line of "Duo-Cone" speakers.

In the midst of this discussion about which portion of the speaker cone is going to reproduce what, the logical query seems to be, "Why not design 2 or 3 speakers for specific frequency ranges, and then use electrical networks to divide the audio spectrum?" This, indeed, seems to be the most popular approach. Even the greenest audiophile, when inspecting a hi-fi installation, will immediately ask if it has a coaxial speaker.

Although most cheap "hi-fi" sets operate on the theory that it's far easier to sell a man three worthless speakers than one adequate one, there are difficulties in a multi-speaker arrangement even when quality units are used:

1) Highs and lows may seem to be coming from different places.

2) Variations in design of the individual speakers may make highs and lows sound different in tonal quality.

3) Acoustic cancellation and electrical phase shift can introduce objectionable distortion at crossover frequencies.

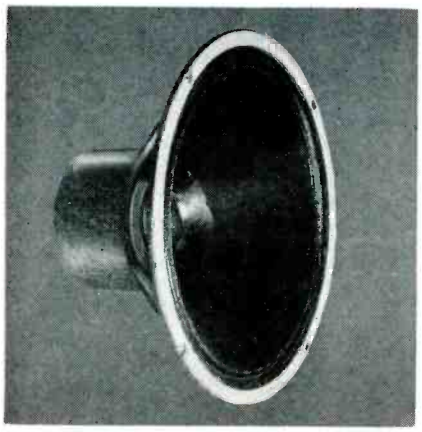
These problems are very real, and they multiply as the number of speakers is increased. Thus some manufacturers think 2-way systems are all right but frown on more elaborate arrangements. Others design ultra systems with 4 or 5 channels. Such companies claim that a multi-channel system is necessary to reduce intermodulation distortion to a reasonable level, and to give adequate response at the ends of the frequency spectrum.

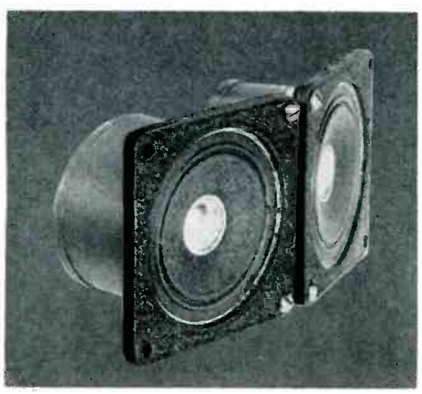
Most such systems use horn or electrostatic speakers for some of the channels, and I will save the discussion of these variations for a later article. Bozak, however, uses three cone speakers as the basic units of 2- and 3-way systems involving as many as 12 separate speakers! All Bozak speakers are paper-cone, direct-radiator types. Fig. 11 shows Bozak's woofer and tweeter. In larger systems more than one of each unit is used.

Bozak says that, with gentle crossover networks and similar design in all three speakers, the objections to 3-way systems disappear. Bozak is also quite specific about how his speakers are to be mounted — the trick, it seems, is the use of an infinite baffle.

At this point you may think that the whole subject of speaker design is just one grand infinite baffle, but perhaps future articles in this series will clear up a few of the questions.

Fig. 11. Bozak woofer and dual tweeter. Middle-range unit also has same cone type.





COURTESY R. T. BOZAK CO

HEATH HD-1

Continued from page 21

that error will be reduced to somewhere around .01%. Remember, that is *maximum* meter error; it may well be less. In any case, it isn't anything to worry about.

The second part of the accuracy specification, "+0.1%", is an allowance (again, maximum) for hum and noise pickup in the instrument itself. After calibration of our completed instrument, we were dismayed to find that the residual indication with no signal input was about 1.5% in the 20 to 200-cps position of the range switch. It is in this position that the highest impedances exist in the Wien bridge circuit, and the instrument is most sensitive to hum pickup. Obviously, something was wrong — 1.5% is a long way from the 0.1% maximum we should have obtained. Replacement of the 5879 tube was the answer. When this was done we found that the residual reading was less than .03% anywhere in the 20 to 20,000-cps range, and over most of the range it was less than .025%. You can use this figure to replace the 0.1% part of the accuracy specification, so in our HD-1, which may be considered typical, low values of distortion can be measured meaningfully down to about .05%.

The frequency-indicator knob is not closely calibrated, but we found it to be very accurate. It can be used to obtain a rough but reliable indication of an unknown incoming frequency, which is a convenient feature. By tuning the instrument to harmonics of a known badly distorted frequency, and watching the meter for the amount of dip at each harmonic, an estimate can be made of the relative amplitudes of the individual harmonics.

We should mention again the matter of proper procedure after a measurement is made. Remember that the total incoming signal is adjusted originally with the level control for a full-scale reading on the least sensitive meter scale. When the fundamental is removed by tuning the bridge, the operator usually adjusts the meter sensitivity upward several steps to obtain a readable indication. Moving the tuning, balance, or range controls then disturbs the null adjustment and, with the meter's greatly increased sensitivity, the fundamental signal now getting through will cause the meter to pin violently. Upon completing a distortion measurement *turn the sensitivity switch to 100% or turn the level control off* before doing anything else. The voltmeter circuit is designed so that maximum meter overload is about 400%, which isn't supposed to be enough to cause damage. Still, it doesn't do any harm to play it safe.

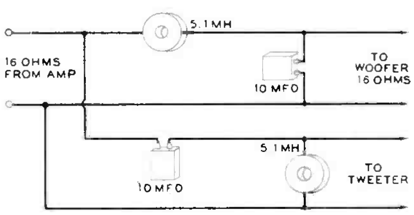
The operator soon becomes familiar with proper use of the controls, and can make a complete measurement in a surprisingly short time. The HD-1 is a fine instrument, well-designed and made with dependable components, and will be a real asset to any serious sound enthusiast or service shop.

PURE AND SIMPLE

Continued from page 23

I might add that if you prefer cone speakers for the middle and high frequencies, you will be surprised at the natural sound you get if you don't mount them inside an enclosure. You needn't worry about a large baffle for the middle-range unit; a wave length at the crossover frequency is only about 18 in. The speakers atop the bass enclosure may be turned straight up, faced directly into the listening area, or turned into the corner.

Fig. 6. 16-ohm 700-cps dividing network.



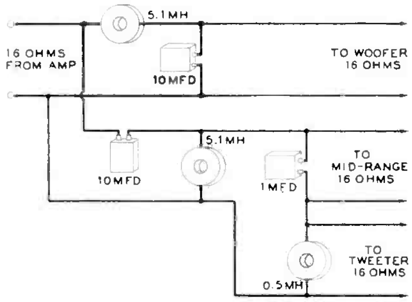


Fig. 7. A 700- and 5,000-cps network.

When I mounted the woofer I put small pieces of masking tape around the rim to seal out any possible air leaks. The tape doesn't leave any residue at all and is quite inexpensive, costing only about a quarter for a roll 1 in. wide and about 30 ft. long. It's obtainable anywhere painting supplies are sold. Also, if you wish, you can make a tape seal along the sides and top of the enclosure. This will assure you that there is not a trace of air leaking out to mar the very low bass response, though I believe Briggs states it isn't necessary to seal the enclosure joints completely.

If, after having completed the bass enclosure and network, you decide to assemble and wire one of the many amplifier kits available, I believe you'll be surprised at the performance and economy such a kit affords. I chose a 20-watt kit having the preamp and power amplifier on one chassis. During the construction I made a few notes that might be helpful.

Before doing any work at all I read the instruction book carefully. Everything seemed very clear and I didn't anticipate any difficulty at all. All operations were treated in a clear, concise fashion and arranged so that one operation, having been completed, could be checked off before continuing. Also, the accompanying pictorial diagrams were crystal clear.

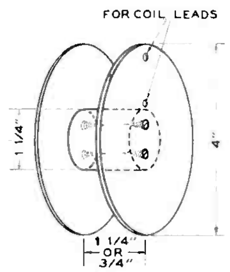


Fig. 8. Form for winding coils; see text.

I kept a time sheet on the project and on completion found that I had expended 22 hours, and I worked in a steady though leisurely fashion. When I checked the amplifier I found that a piece of solder was grounding the signal at the grid of the voltage-amplifier tube. This was cleared up in about 10 minutes, and the amplifier was connected to the speaker system. I attached an FM tuner and it operated perfectly. In fact, I think it performs better than a factory-assembled model I have that is, of course, more costly. For one thing, the output transformer of the kit amplifier is quite large, which is often important. The highs are smooth and the bass is full and excellent.

There is no compensated loudness control on the kit, a fact that I regard as a selling point although others may not. And, although there is a microphone input (which I wanted to handle a hi-fi, low-gain AM tuner), the magnetic phono equalization is quite skimpy. In my case that wasn't a consideration.

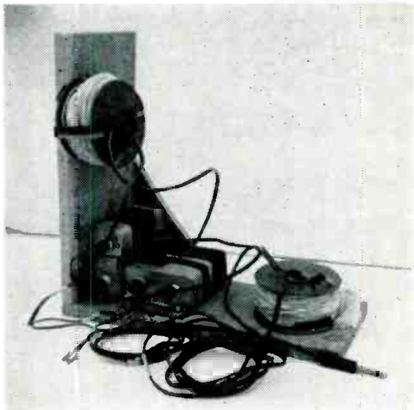


Fig. 9. The low-frequency network parts.

Others, however, might consider it a drawback.

When it comes to servicing hi-fi equipment, I feel it is definitely an advantage to have built your own. You'll be personally familiar with quite a few connections and components, so much so that, when one starts acting up, discovering its location won't be too difficult.

I don't want this to sound too easy, because there is a certain amount of concentration required. I do feel, however, that any average audio fan, not firmly committed to the idea that he is technically inept, can duplicate the system I've outlined without much trouble at all.

TAPE NEWS

Continued from page 13

the orchestra is waiting for the conductor's signal, there is much low rumbling and shuffling noise. Any attempt to splice directly from this into a length of blank tape results in a very

Continued on page 37

AUDIOCRAFT Sound Sales Directory

Following is a list of dealers who state that they carry the products specified.

KEY TO PRODUCTS HANDLED

1 Audio system components
2 Speakers and enclosures
3 Records and record accessories
4 Tape recorders
5 Pre-recorded tape
6 Radio hardware
7 Tools, wood
8 Audio parts
9 Microphones
10 Books
11 Test equipment

A series of items numbered consecutively is identified by a hyphen between the first and last numbers. Thus, 1-6 indicates 1 through 6; 8-11 indicates 8, 9, 10, and 11.

CALIFORNIA

Beverly Hills
Minthorne Music of Beverly Hills
230 N. Beverly Dr.
BR. 2-7676; CR. 6-4793 — 1-5, 9

Culver City
Bar-Shel's Hi-Fi Supply
5512 S. Sepulveda Blvd.
EX. 8-6747 — 1, 2, 4-6, 8-11

Hollywood
Hollywood Electronics
7460 Melrose Ave.
Webster 3-8208 — 1, 2, 4, 5

Los Angeles
Crenshaw Hi-Fi Center
107 Santa Barbara Plaza
Axminster 3-8201 — 1-5, 9

Menlo Park
Science in Sound
777 Santa Cruz Ave.
DA. 2-1611 — 1-11

Santa Barbara
Pacific Audio Supply
2919 De La Vina St.
Woodland 5-6422 — 1-5, 9

CONNECTICUT

West Hartford
Audio Workshop, Inc.
1 South Main St.
Adams 3-5041 — 1-5, 9, 10

DELAWARE

Newark
Delaware Music House
20 Academy St.
Endicott 8-3258 — 1-5, 8, 9

FLORIDA

Miami
High Fidelity Assoc.
3888 Biscayne Blvd.
82-8401 — 1-5, 8-10

GEORGIA

Atlanta
High Fidelity Sight & Sound Sys.
606 Peachtree St., N. E.
Vernon 6534 — 1-10

ILLINOIS

Champaign
The New Sound
35 E. Springfield Ave.
6-9119 — 1-5, 8-11

Chicago
Allied Radio Corp.
100 N. Western Ave.
Haymarket 1-6800 — 1-11

ILLINOIS — *Chicago (continued)*

Allied Radio Corp.
2025 W. 95th St.
Beverly 8-1067 — 1-11

Musicraft
48 East Oak St. — 1-11

Voice and Vision, Inc.
Rush and Walton Place
Whitehall 3-1166 — 1-6, 8-10

Evanston
Allied Radio Corp.
602 Davis St.
Davis 8-8822 — 1-11

LOUISIANA

New Orleans
The Music Shop, Inc.
4215 South Claiborne Ave.
TW. 1-5871 — 1-5

MASSACHUSETTS

Boston
The Listening Post, Inc.
161 Newbury St.
COpley 7-7530 — 1, 2, 4, 5, 8-10

MICHIGAN

Detroit
Hi-Fi Studios
8300 Fenkell
Diamond 1-0894 — 1-6, 8-10

MISSISSIPPI

Pass Christian
The Music Box
121 Davis Ave.
185 or 301W — 1-5, 8, 9

MISSOURI

Kansas City
David Beatty Hi Fi Sound & TV
1616 West 43rd (Westport Rd.)
Jefferson 1-3110 — 1-10

St. Louis
The High Fidelity Showroom
6383 Clayton Rd.
Parkview 1-6500 — 1-5, 9

NEW YORK

Albany
Hi-Fidelity Center, Inc.
324 Central Ave.
3-1167 — 1-5, 8-10

Amsterdam
Adirondack Radio Supply
185-191 West Main St.
VI. 2-8350 — 1, 2, 4, 8

Buffalo
Frontier Electronics, Inc.
1507 Main St.
GA. 5727 — 1-5, 8-10

New York City
Arrow Audio Center
65 Cortlandt St.
Digby 9-4730 — 1, 2, 4-6, 8-11

Grand Central Radio, Inc.
124 East 44th St. at Lexington Ave.
MUrray Hill 2-3869-70 — 1-6, 8-11

Yonkers
Westlab Electronics, Inc.
2475 Central Ave.
SP. 9-6400 — 1-6, 8-10

OHIO

Dayton
Custom Electronics, Inc.
1000 S. Main St.
AD. 3158 — 1-11

PENNSYLVANIA

Allentown
Baker Communications
9 South 12th St.
Hemloch 3-3326 — 1-11

Bethlehem
Audio Laboratories, Inc.
808 Mohican St.
UNiversity 7-3909 — 1-5, 9

Philadelphia
Almo Radio Co.
509 Arch St.
WAlnut 2-5153 — 1-4, 6-11

Danby Radio Corp.
19 South 21st St.
Rittenhouse 6-5686 — 1-5, 8-10

Radio Electric Ser. Co. of Pa., Inc.
7th & Arch Sts.
LO. 3-5840 — 1-11

Ten Cate Associates
6128 Morton St.
Germantown 8-5448 — 1-5, 8-10

SOUTH CAROLINA

Columbia
HI-FI Sound & Records Co.
621-23 Harden St.
6-3538 — 1-6, 8-10

TENNESSEE

Kingsport
Radio Electric Supply Co.
245 East Market St.
Cl. 5-1471; 5-2651 — 1, 2, 4, 6, 8-11

TEXAS

San Antonio
Bill Case — Records & Sound
4206 Broadway
TAylor 2-1341 — 1-5, 8-11

VERMONT

Rutland
Fleetwood Sound Studio
285 S. Main St. (Rt. 7)
Prospect 3-2312 — 1-5, 8, 9

VIRGINIA

Arlington
Audio Services
3131 N. 10th St.
Jackson 7-2552 — 1, 2, 4, 5, 8, 9

WASHINGTON

Seattle
High Fidelity Headquarters
603 Broadway North
CApitol 2266 — 1-4, 8, 10

WEST VIRGINIA

Huntington
Singer Electronics
1555 Third Ave.
3-7439 — 1-5, 8-9

WISCONSIN

Milwaukee
The Hi-Fi Center, Inc.
4236 W. Capitol Drive
Uptown 1-2113 — 1-5, 9, 10

CANADA

Montreal
Payette Radio Limited
730 St. James West
UNiversity 6-6681 — 1-11

Toronto
Electro-Voice Sound Systems
141 Dundas St., W.
EM. 8-7750-8301-8302 — 1-11

TAPE NEWS

Continued from page 35

audible change in ambient noise. On the other hand, leaving the noise in constitutes a source of annoyance, not so much because of the coughs and tuning-ups, but because half the noises are unrecognizable and seem to have no resemblance to anything earthly. Out they must go.

The first step, then, was to cut into the tape precisely at the beginning of the opening note of each section, so that by the time the music became audible it was overriding the background noise. But editing the blank tape into the sections *following* each number proved to be more difficult, because there is always a certain amount of echo following the end of any sound, and cutting into this results in a shocking cut-off. On the other hand, the farther the echo is allowed to die away before cutting into the tape, the more audible is the background noise at that point. The trick was to catch the echo at just that critical point where it was at the threshhold of audibility, so the final cessation of the echo occurred when it might ordinarily be expected to in the concert hall. A final chord then would diminish almost to the background level, and then would be run out altogether in the angled splice. The result, while not ideal, was at least comfortable to listen to. While a trained splice-hunter would undoubtedly be able to hear the cut-in/cut-out spots, these are not the source of agitation or shock that they might have been.

Finally there came the technical session, which consisted of running off a 10-second 1,000-cps signal from a test oscillator, recorded at zero level. Like most other custom processors, Columbia Records require this at the beginning of each reel of tape, so that they can set the recording level of their cutting amplifiers, and adding these signals plus the long plastic leaders at the beginning and ends of the tapes

constituted just about the last step. Leaders had also to be added mid-way through each reel, to indicate the transition between record sides, and the reels were finally labeled with red grease pencil, listing sides and record numbers.

The rest of this project is in the hands of Columbia Records' Custom Processing Department, and we're still waiting to receive our complimentary copies. The recordings may not put Westminster out of the Laboratory-Series business, but if anyone were to ask us how much fidelity is lost between a tape and a disc, we should have an answer and a ready comparison.

POWER SUPPLY

Continued from page 26

2 by 5 by 7-inch aluminum chassis base.

7-pin miniature tube socket for 5/8 in. mounting hole.

Snap-ring octal socket for 1 11/64 in. mounting hole.

Type 3AG fuse post and 2-amp 3AG fuse.

Single-pole single-throw switch (see text).

Line cord.

Miscellaneous: 3 single-lug terminal strips, 1 three-lug terminal strip, 1 ground lug, 1 rubber grommet, hookup wire, machine screws and nuts.

AUDIOCRAFT Test Results

This is a power supply of conservative design and it uses high-quality parts. It should give long, trouble-free service.

Drilling and punching the chassis will be the most difficult part of this construction job for most builders. It isn't unusually complicated, but any chassis-cutting operation is tedious unless you have proper tools. We have learned through bitter experience that a good set of chassis punches (for holes larger than 1/2 in. in diameter), expensive as it may be, is still a very good investment. Punches last indefinitely, with proper care, and save a great deal of labor every time they are used. For holes up to 1/2 in. metal-cutting bits are available for manual brace drills; for holes of 1/4 in. and smaller standard metal drills are used.

Without chassis punches it is necessary to make large holes by filing or reaming a 1/2-inch hole up to the required size or, alternatively, drilling a lot of small holes close together inside the outline of the large hole. The jagged edge left is then smoothed with a half-round file.

Since we had the required punches, the total job consumed just under 3 1/2 hours. When connected to the Huff amplifier the B+ voltage was 246 volts — quite close enough to the nominal 250. Performance of the power supply was perfectly satisfactory; no overheating occurred and no hum was measurable with ordinary test instruments at the amplifier output.

Cost of parts has recently gone up slightly, so the total for all parts was just about $15.

BASIC ELECTRONICS

Continued from page 28

voltage becomes maximum at 270°, when the flux-cutting rate reaches its peak, and then decreases toward zero again as the coil rotation approaches 360°. If the output voltage were plotted on a time base (assuming a constant rotary speed for the coil) the result would be a sine wave.

The device just described is a simple alternator: a generator of alternating current. In slightly more complex form

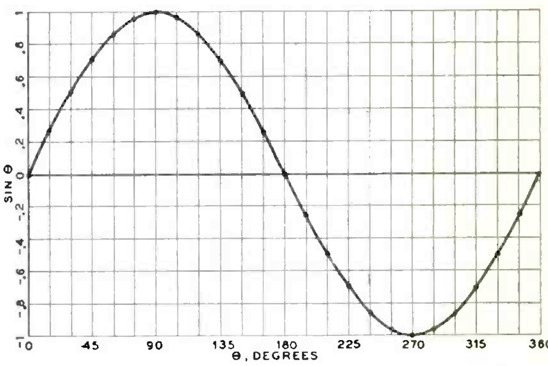


Fig. 8. Plotting sine values for regular angular increments yields a sine wave.

it is used at electrical power stations to generate the current that reaches us over AC power lines.

A sine wave is the basic alternating wave form, the fundamental pattern of repetition. The motion of a pendulum, or of any resonant system in natural vibration, describes a sine wave when measured on a time basis. The wave form of a pure tone is that of a sine wave. Any complex wave form, so long as it is repetitive, can be shown to consist of a sine wave of the fundamental frequency combined with sine waves at multiple frequencies of the fundamental. It follows that a sine wave is the only wave form consisting of one single frequency.

Sine Wave Properties

A few other properties of sine waves should be mentioned at this point. Two complete cycles of a sine wave are drawn in Fig. 9, of 50-cps frequency. During each cycle the voltage rises to a maximum in one direction, falls through zero to a maximum in the other direction, and returns to zero again. As with the square wave used as an example in the preceding chapter, the peak voltage is the maximum value in either direction from the zero line, and the peak-to-peak voltage is that between the opposite swings.

It was pointed out that the logical way to correlate AC and DC voltages and currents was in the power delivered to a given load. Since the direction of current makes no difference to a resistive load, and the voltage across the load

Continued on page 39

Abbreviations

Following is a list of terms commonly used in this magazine, and their abbreviations. The list is arranged in alphabetical order.

alternating current	AC
ampere, amperes	amp, amps
amplitude modulation	AM
audio frequency	AF
automatic frequency control	AFC
automatic gain control	AGC
automatic volume control	AVC
capacitance	C
cathode ray tube	CRT
characteristic impedance	Z_0
current	I
cycles per second	cps
decibel	db
decibels referred to 1 milliwatt	dbm
decibels referred to 1 volt	dbv
decibels referred to 1 watt	dbw
direct current	DC
foot, feet	ft.
frequency	*f*
frequency modulation	FM
henry	h
high frequency	HF
impedance	Z
inch, inches	in.
inches per second	ips
inductance	L
inductance-capacitance	LC
intermediate frequency	IF
intermodulation	IM
kilocycles (thousands of cycles) per second	Kc
kilohms (thousands of ohms)	K
kilovolts (thousands of volts)	KV
kilowatts (thousands of watts)	KW
low frequency	LF
medium frequency	MF
megacycles (millions of cycles) per second	Mc
megohms (millions of ohms)	MΩ
microampere (millionth of an ampere)	μa
microfarad (millionth of a farad)	μfd
microhenry (millionth of a henry)	μh
micromicrofarad	μμfd
microvolt (millionth of a volt)	μv
microwatt (millionth of a watt)	μw
milliampere (thousandth of an ampere)	ma
millihenry (thousandth of a henry)	mh
millivolt (thousandth of a volt)	mv
milliwatt (thousandth of a watt)	mw
ohm	Ω
permanent magnet	PM
potentiometer	pot
radio frequency	RF
resistance	R
resistance-capacitance	RC
resistance-inductance	RL
revolutions per minute	rpm
root-mean-square; effective value	RMS
synchronous, synchronizing	sync
television	TV
ultra high frequency (radio)	UHF
vacuum-tube voltmeter (multipurpose)	VTVM
vacuum-tube voltmeter for AC measurements only	AC VTVM
variable reluctance	VR
very high frequency (radio)	VHF
volt	v
volt-ampere	va
voltage, or potential difference	E
volts, center-tapped	vct
watt	w

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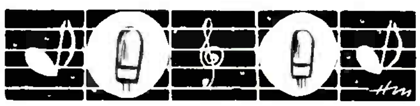

BASIC ELECTRONICS

Continued from page 37

from a square-wave AC source is always the peak voltage except for the instants during which it changes direction, then the effective voltage of a square wave is the peak voltage. This is not true for a sine wave. The direction of the voltage at any given time makes no difference; for purposes of power computation, both half-cycles could be considered above the zero line. But the maximum or peak voltage of a sine wave is approached gradually and recedes gradually, not instantaneously as in a square wave. Accordingly, the average power must be considerably less than that obtained at the peak of the wave form.

Power delivered to a resistive load at any instant (instantaneous power) is determined with the conventional power formula $p = e^2/R$, where p is instantaneous power, e is the voltage at that instant, and R is the load resistance. The average power over a cycle can be found by calculating the instantaneous power at a great many equally-spaced parts of the cycle, and then averaging the figures obtained. This average power figure can then be used in the formula $E = \sqrt{PR}$ to find the equivalent DC voltage that would produce the same power in the load. Such a voltage is the *root of the mean of the squares* of the sample instantaneous voltages used to find the average power, and is abbreviated RMS voltage. The RMS (effective) voltage of a sine wave is found, by this process, to be 0.707 times the peak voltage. Thus, the effective voltage of the wave form in Fig. 9 is 7.07 volts; the peak voltage is 10 volts, and the peak-to-peak voltage is 20 volts. It would deliver the same average power to a load as would 7.07 volts DC.

When the type of measurement is not specified for an AC voltage or current, it can be assumed that it is RMS or

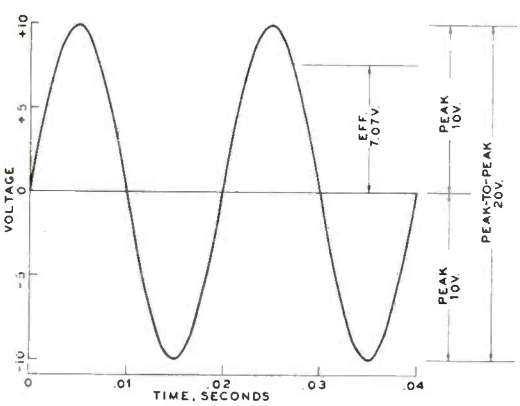

Fig. 9. Sine wave peak and RMS values.

effective value. To convert this to peak value, it must be multiplied by the reciprocal of 0.707, which is 1.414. A common power-line voltage is 117 volts; this is about 165.4 volts peak and 330.9 volts peak-to-peak. It is interesting to note also that the peak instantaneous power is exactly twice the average power over a cycle.

The matter of phase is also important in AC calculations. It will be recalled that the individual parts of a sine wave

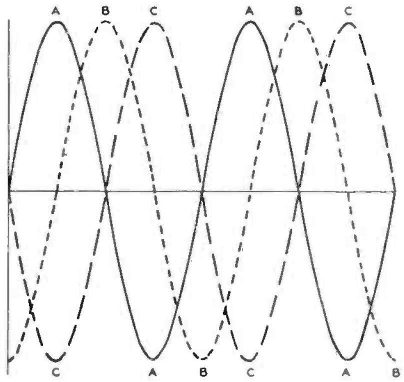

Fig. 10. Sine wave phase relationships.

are closely associated with electrical degrees of rotation; indeed, they can be measured in this way. Referring to Fig. 8, it can be seen that a half-cycle occupies 180°, a quarter-cycle 90°, and a full cycle 360°. If two sine waves are going through the same part of their cyclic form at any instant, they are said to be *in phase* at that instant; if not, they are *out of phase*. If sine wave A is going through its positive maximum at the moment sine wave B is going through zero in the positive direction, then A either *leads* B by 90° or A *lags* B by 270°. If A is going through its maximum positive peak while B is going through its maximum negative peak, then A leads or lags B by 180°; A and B are said to be in opposite phase. In Fig. 10 wave forms A and C are 180° out of phase, or of opposite phase. Wave form B is 90° out of phase with both A and C; B lags A and leads C.

READERS' FORUM

Continued from page 15

ever measured. But that doesn't sell the speaker.

I have been designing and producing speakers for 30 years. I have tried every conceivable size, shape, and kind of cone; single and multi-channel systems; electrostatics large and small. I never had an ax to grind; I just wanted to build good speakers, and what I put into the 215 gave the nearest approach to realism I could manage. Sales are consistent to discriminating music lovers, but according to the ordinary audiophile in the street the speaker hasn't got what it takes. I don't say this is right or wrong; I just note that is what happens. Am I, therefore, to stick to what I know is best and lose sales, or produce some-

Continued on next page

READERS' FORUM

Continued from preceding page

thing that will sell easily? My dealers to a man say they like the 215 immensely, but, as dealers, they have to live, and beg me to produce something the customer will take away under his arm. They point out that if the customer wants floor-shaking bass at high volume two 215's in simple series will do the trick at reasonable cost. But it isn't cost; the argument is that if one unit won't make as big a row as a large speaker it can't be any good, and nothing will convince them otherwise. Similarly, a less sensitive speaker is rejected for one that bangs out plenty of acoustic watts, whatever the quality. Edgar Villchur (*Audio*, April 1956) points out that it is as useless to compare speakers without making adjustments for equal sound levels as to compare different pickups without adjusting amplifier gain.

It's a hard life for the musical engineer conscientiously doing his best.

H.A. Hartley
London, England

Gentlemen:

You've done it again! What I mean is, there has hardly been an issue of AUDIOCRAFT that has not helped me improve my sound system. This (June) issue it

was the article on the Heath preamp, regarding the correct setting of the volume control.

Recently the articles on amplifier stability resulted in my making some changes in my Heath W-3M amplifier that, apparently, improved its stability. The articles on the use of the vacuum-tube voltmeter enabled me to locate a defective tube. In other words, I think your magazine is tops.

I cannot suggest any way to improve AUDIOCRAFT. You seem to anticipate very successfully what guys like me want to know. Just keep going as you are.

I write an occasional audio column for a weekly newspaper, where I am employed as a printer. My main purpose (aside from seeing my name in print) is to do my small bit to popularize high fidelity.

I wish you continued success.

Virgil McCarley
Spring Valley, Calif.

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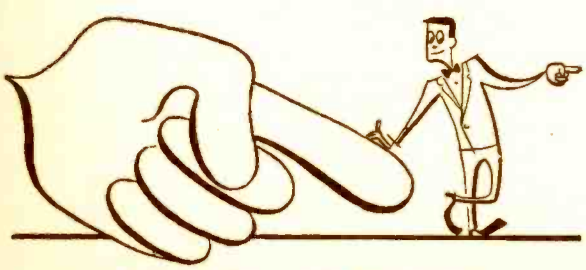

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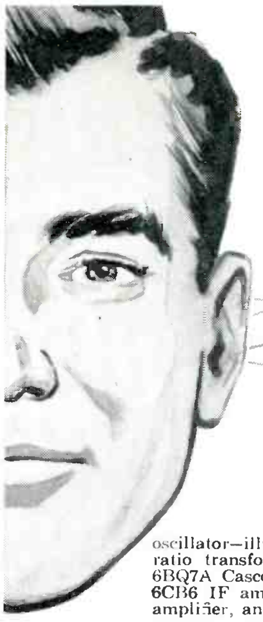

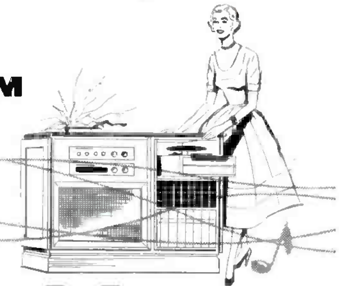

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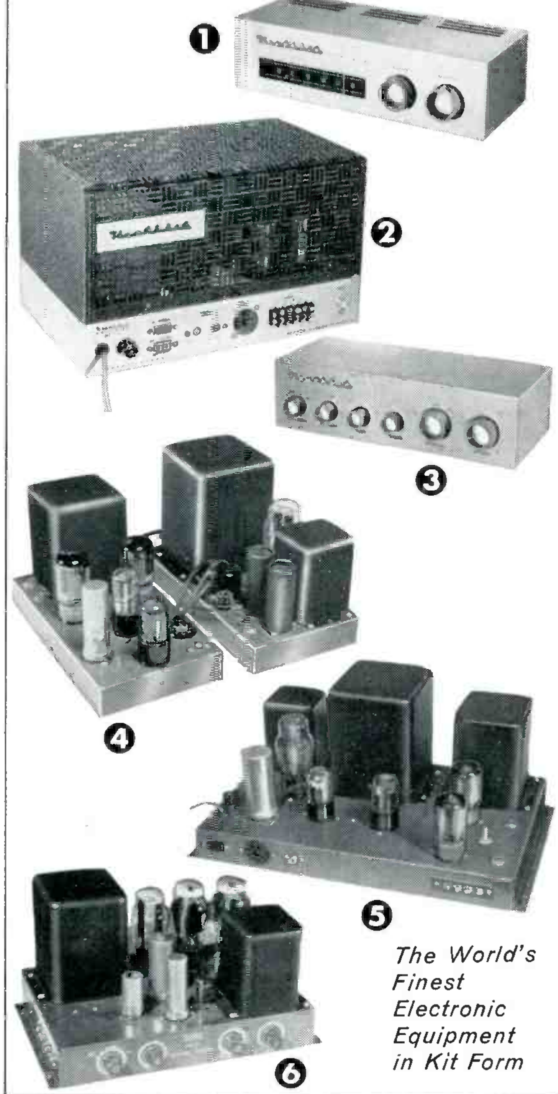

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