audiocraft

MARCH

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

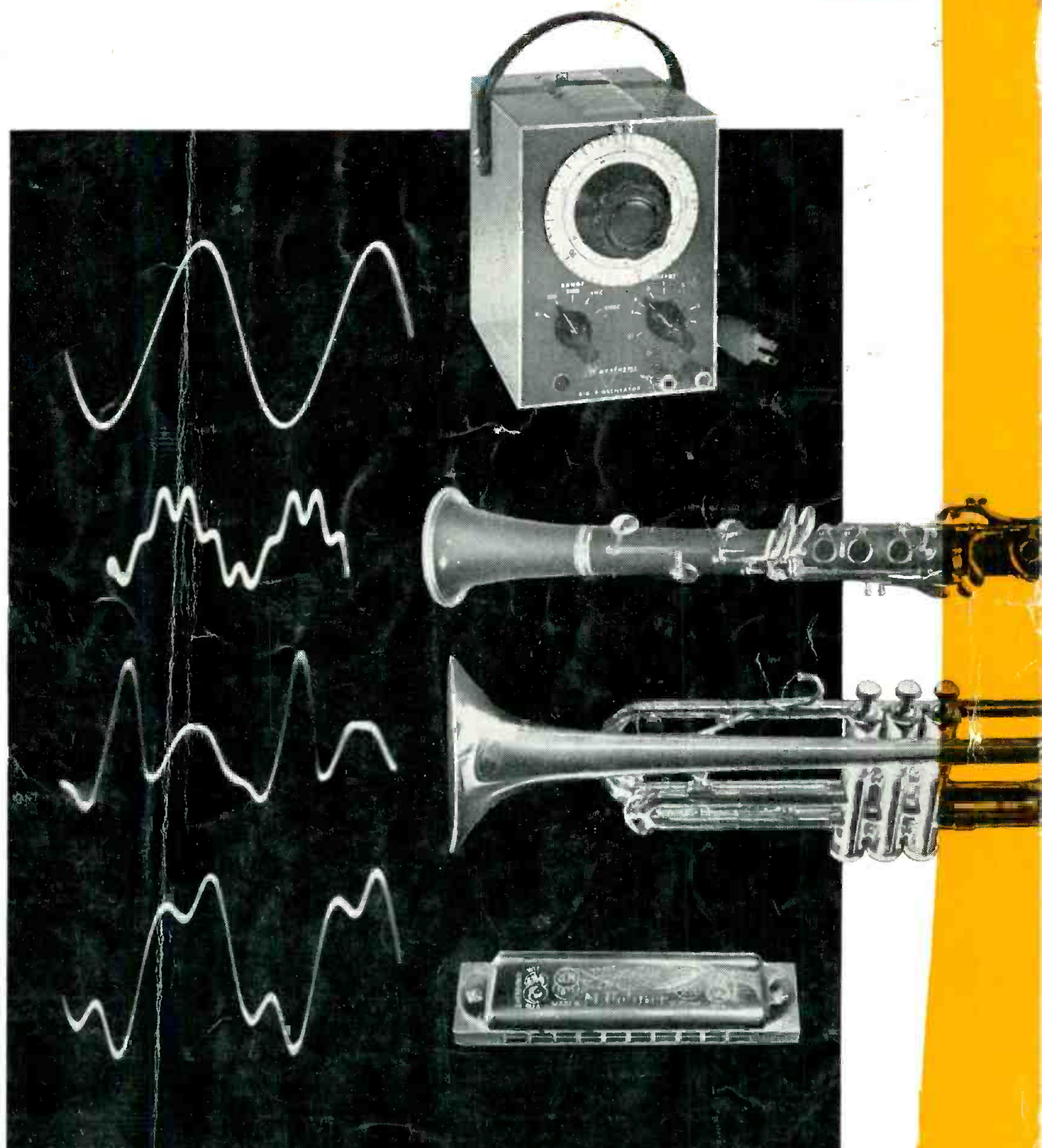
35 CENTS

THIS MONTH

How to work with the new printed circuits found in modern kits.

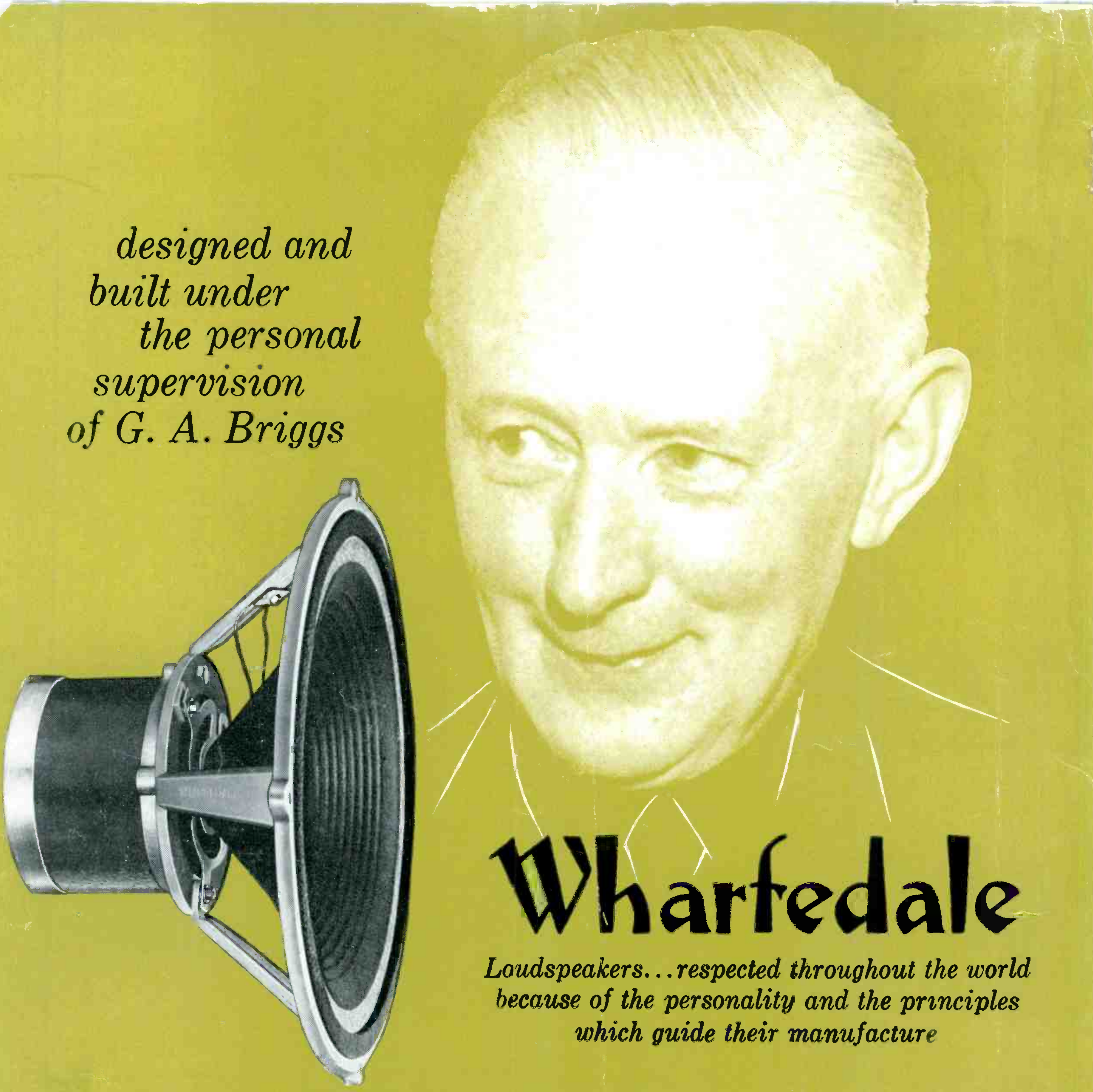
Tips on audio system troubleshooting with no test instruments.

Beginning an article series on designing your own amplifier.



DISTORTION AND ITS EFFECTS

A non-mathematical discussion of the various kinds of amplifier distortion, their causes and audible effects, how they can be measured and evaluated, and what can be done to minimize them.



by h.h. Scott

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The 311 FM Tuner, $99.95*

There are NO weak stations with this new tuner

■ Terrific 3-microvolt sensitivity makes distant stations sound as clear and strong as those nearby.

■ New wide-band FM design gives super-selectivity, to separate stations so close together you would ordinarily pass right over them.

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TECHNICAL SPECIFICATIONS

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*Slightly higher west of Rockies.

310 FM BROADCAST MONITOR TUNER

For perfectionists and connoisseurs, H. H. Scott offers the 310 FM tuner. High Fidelity Magazine says: The 310 ". . . is a tuner that seems as close to perfection as is practical at this time." Price, including case $149.95 East Coast; $157.45 West Coast.

by h.h. Scott

The Greatest Amplifier Buy You Have Ever Seen

99-B Transcription Amplifier $99.95*

Imagine! 22 watts – complete controls – only $99.95

■ The famous "99", a complete amplifier, now with twice the power — a brilliant 22 watts.

■ Complete equalizer-preamplifier with five-position record compensator. Equalizes virtually all records.

■ New adjustable rumble filter and record scratch filter reduce record noise and rumble.

■ Two magnetic inputs, switched on panel, allow use of both changer and turntable.

■ Special provisions for playback of pre-recorded tape through your 99-B.

■ Continuously variable LOUDNESS compensation, with volume-loudness switch, gives perfect tonal balance at all listening levels.

TECHNICAL SPECIFICATIONS

Input selector switch for two magnetic pickups, crystal or constant amplitude pickup, three high-level inputs, and NARTB tape playback — frequency response flat from 20 cps to 30 kc — hum better than 80 db below maximum output — harmonic distortion less than 0.8% — first-order difference-tone intermodulation less than 0.3% — class A circuits throughout — easy panel mounting — beautiful accessory case $9.95* *Slightly higher west of Rockies.

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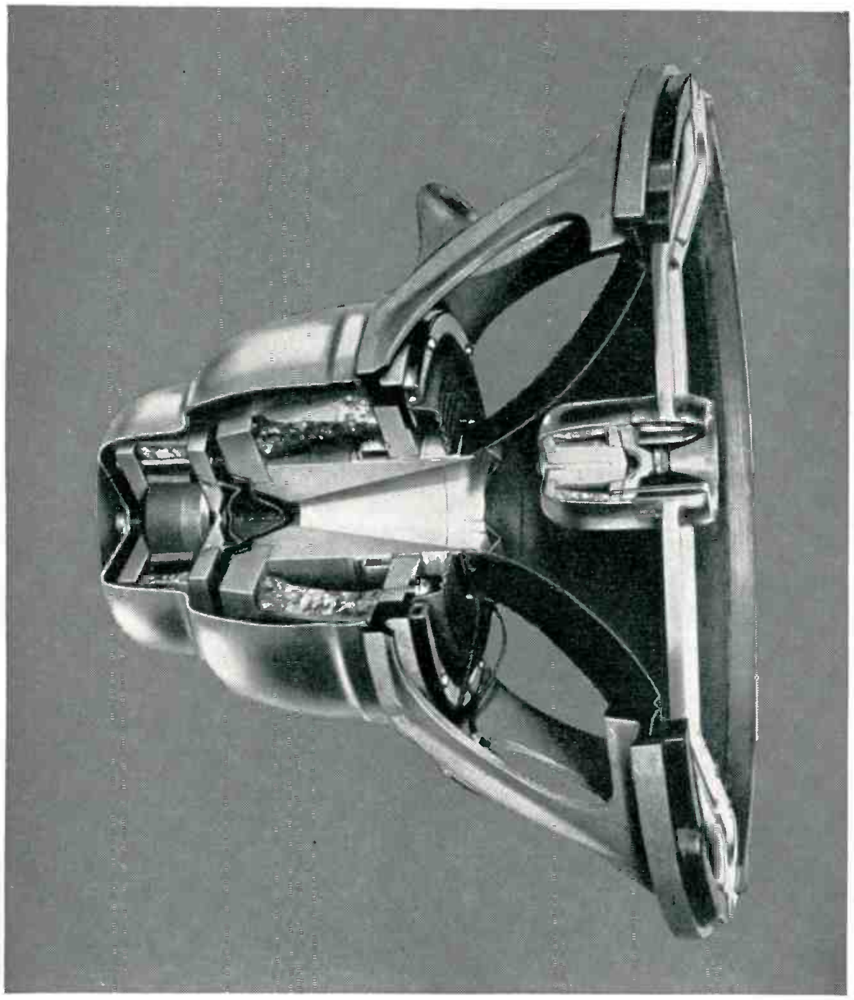

March 1956

Volume 1 Number 5

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

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

Scott J. Saunders is a man handy with a hammer, as the pictures illustrating his article (beginning on page 16) will attest. Typical of do-it-yourselfers, his versatility is seemingly boundless; fortunately, it extends to mastery of the typewriter too. He makes his home in Plainview, N. Y.

Richard D. Keller is another author with wide interests. He holds degrees in both business and electrical engineering from the University of Colorado; while in the naval air force he was a charter member and treasurer of the Hawaiian Audio Society; he is an avid skier and amateur motion-picture photographer; and he is working at transistor engineering for GE at Electronics Park, Syracuse, N.Y.

Charles Fowler will need no introduction to most readers. Publisher of HIGH FIDELITY and AUDIOCRAFT Magazines, he is well known also as an authoritative and engagingly enthusiastic writer of articles on all aspects of sound and its reproduction. McGraw-Hill will publish soon his book *High Fidelity: A Practical Guide.*

Norman H. Crowhurst, recently arrived from England, is author of the famous *Audio Handbook* series. The list of magazines to which he has contributed includes just about every technical and non-technical publication dealing with audio, radio, and electronics. Currently free-lancing in Whitestone, N. Y., Mr. Crowhurst has prepared a lengthy series of articles for us; it begins with this issue.

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The Grounded Ear

by Joseph Marshall

Modern Design

It is no great problem today to design and manufacture commercial equipment which fully meets, or even exceeds, standards of high fidelity performance that a brief 5 years ago were considered to be achievable only in the laboratory. One result is that for certain components — like amplifiers — performance is so uniformly high that there is little to choose between various units in the same (or even in different) price ranges. It is characteristic of our American merchandising system that when this state of affairs is reached in any field, manufacturers begin to exploit secondary qualities such as appearance, convenience, or compactness to achieve a sales advantage over competitors. Most of the mass consumer products — automobiles or refrigerators, for example — provide very little choice in utilitarian performance characteristics; they are designed with an emphasis on eye-appeal and groomed for the newly discovered "impulse-buying instinct". It is not surprising, therefore, that in the now highly competitive high fidelity field, manufacturers have taken the same cue. Now the industrial designer is becoming as important as the engineer.

There is, of course, no reason whatever why high fidelity equipment should not appeal to the eye as well as to the ear, and decorate the living room as well as fill it with gorgeous sound. So long as "modern design" is kept in its place as a secondary consideration, and the principal emphasis continues to be put on high performance, any improvement in appearance is all to the good and strictly a dividend to the consumer. Unfortunately, this has not been so with automobiles and other consumer goods. Up to 3 or 4 years ago the appearance of American automobiles was attributed such importance that the technical quality and utilitarian features were far behind those of European motor cars; the American motorist was offered more pounds of chrome trim per dollar, and less genuine performance, than his Continental counterpart. For that matter, even the eye-appeal designers became so immersed in considerations of chrome trim, rather than basic form, that the American industry eventually lost (to the Italians) leadership in eye-appeal design — to such an extent that today's American cars owe not only many of their technical qualities but even a good deal of their eye-appeal to Continental sources. In other words, so long as modern design is the handmaiden to performance, the consumer profits; but once it becomes mistress, the tendency is for performance to take a back seat. Technical development is retarded and eventually becomes stagnant. I suppose everyone who is concerned about the welfare of the high fidelity industry would deplore any tendency in that direction.

So far, the part played by "modern designers" in high fidelity has been very largely to the good, but there is evidence that, in this field, too, some have acquired dominance over the engineers. I know of several products in which good engineering was nullified by modern design. Fortunately, in each of these cases the harm was largely undone by the comments of magazine critics, as well as complaints of distributors and buyers, which moved the manufacturers to make corrections. On the whole, however, design has been kept in its proper place. I hope this excellent state of affairs will continue, but I have some misgivings.

For example, there is the "pancake" design for amplifiers and tuners. There is no denying its attractiveness. For some reason or other it is a form which appeals to our modern temperament and taste. For that matter, the form presents some advantages in layout for both the design engineer and the production department. But it has also disadvantages. To begin with, it is not really compact: it looks small but, when used in a practical household, it takes up more room than the old-fashioned chassis layout. True, if you could stack the pancake units, it might be possible to put both an amplifier and a tuner on one shelf. Unfortunately, this cannot be recommended in all cases. The closed, compact form reduces heat dissipation, and the pancakes run hot. If one of them is a tuner, subject to drift with changes of temperature, it is a poor idea to stack it above or below an amplifier that also runs hot, unless it has been designed (as some have been) with that possibility in mind. Still, stacking isn't a good idea. And when you put anything on a shelf it occupies the entire height of that shelf — unless the shelves are tailored to order. The normal bookshelf is 8 or 9 in. high; the normal pancake tuner or amplifier is 4 to 5. One therefore occupies a space on the shelf equivalent to its length, and the low height actually provides little or no advantage. Indeed, if the same gadget were built in a case 8 in. high it would occupy only half the shelf space of the pancake unit.

Second, the pancake design raises problems of heat dissipation even when not stacked. This is better dealt with in later pancake gadgets than in earlier ones, but they still tend to run hot. To compound this, the engineer must often use a much smaller power transformer than he would like to, because the pancake case has room for only a small one. The smaller transformer, with smaller wire, will generate more heat and, at the same time, is less able to withstand the higher temperature inside the smaller case. Trouble may develop not only in the transformer but in other components. The more compact form requires smaller resistors and capacitors, but the higher ambient temperature calls for larger ones. All too often the engineer, against his better judgment, is forced to use smaller parts than are necessary for safe ratings.

This should not be interpreted as condemnation of the pancake design. It probably has as many points genuinely in its favor as the older design and, properly used, produces a more attractive product. I merely point out that it is less tolerant to over-heating and at the same time invites it; and I urge manufacturers to keep durability, serviceability, and high performance in mind — indeed, to give them priority, even when the sales departments ask for still more compact, still more enclosed forms. Before following the trend to an abyss of uniformity and mediocrity, it should be recalled that the old chassis design has a third of a century of reliability to recommend it and can be dolled up very nicely, too.

Some New Products

If you are producing or servicing hi-fi equipment and have to make many frequency runs every day, the new HEATHKIT AUDIO GENERATOR will save its cost in a brief month or less. Instead of the normal tuning knob it has 3 decading switches to select frequency to 2 significant figures, and multipliers which cover the whole range from 9 to 110,000 cps. You can run a sweep of the entire audio spectrum by units to 100, tens to 1,000, hundreds to 10,000, and thousands to 100,000 cps in a small fraction of the time needed with the continuously tuning types, and with far better overall accuracy. The generator also has an output meter and a pad-type attenuator, which provides output as low as 1 μv and as high as 10 volts.

Both ALLIED and RAULAND have accessory TV sound tuners for their AM-FM tuners. These make possible the reception of TV sound with a fidelity and freedom from distortion far beyond that of even the 630-type TV receivers. In fringe areas, you can use the TV sound tuner to bring in the sound and adjust the TV receiver for picture reception only, and thus often obtain good reception of both sight and sound, which is otherwise impossible. Furthermore, in our family we have found that TV sound communicates almost all the enjoyment of TV shows and is far less distracting to other activities.

CABINART has a simple handful of accessories for a turntable that may help obtain better and more trouble-free phono reproduction. It consists of a bubble spirit level, like that on simple sextants and transits, to be mounted on the motor board of the turntable. This indicates at a glance whether the turntable is level. If it is not, the package also has 4 screw-type padded "legs" to put on the turntable base; these can be adjusted individually to take up 3/8 in. or more of turntable slant.

Audionews

HEATHKIT SPEAKER SYSTEM

A new product of the Heath Company is the Heathkit *Model SS-1* speaker system. It is claimed that response of the system is essentially flat and is within ±5 db from 50 to 12,000 cps.

The enclosure measures only 11½ in. high by 23 in. wide by 11¾ in. deep. The cabinet comes in knocked-down form and is made of furniture-grade plywood suitable for either a light or dark finish. The picture-frame molding (which is factory assembled) makes it suitable for use with any decorating scheme.

Kit speaker system by Heath Company.

The cabinet is a ducted-port, bass-reflex type, and the system features 2 separate speakers for high and low frequencies. A balance control allows level adjustment on the high-frequency speaker. The speakers are manufactured by Jensen and come ready for installation. It is necessary only to assemble the cabinet, wire in the speakers and crossover network, and apply the finish of your choice.

While the SS-1 is an excellent reproducer in its own right, it is also the first step toward an outstanding new speaker system. Heath's future plans call for an enclosure that contains an additional woofer and a super tweeter, with crossover, for a complete 4-way high fidelity speaker system.

Additional information will be furnished on request.

CATALOGUE OF PRE-RECORDED TAPE

A new catalogue of pre-recorded tape is now available. Known as the *Harrison Catalog of Recorded Tapes,* it lists all currently available pre-recorded tapes. Listings are arranged by type of music, composer and artist, and company of manufacture.

Copies of the Harrison Catalog are available free upon request to Dept. AC, Neu Tape Center, 2233 W. Roosevelt Drive, Milwaukee 9, Wis.

ALL-PURPOSE MULTIMETER

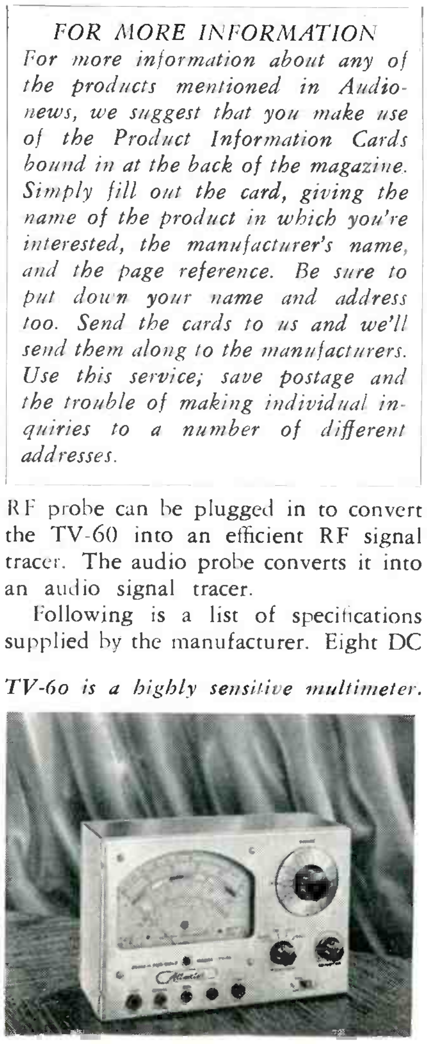
Superior Instrument Company has announced the *Model TV-60 Allmeter,* a complete all-purpose, 20,000 ohms per volt multimeter. The unit has a recessed, 6½-inch, 40-microampere meter with a mirrored scale to assure accurate and easy reading.

Among the features of the TV-60 are an accurate direct-reading capacitance meter, a kilovoltmeter, an RF signal tracer, and an audio signal tracer. The RF probe can be plugged in to convert the TV-60 into an efficient RF signal tracer. The audio probe converts it into an audio signal tracer.

> *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.*

Following is a list of specifications supplied by the manufacturer. Eight DC voltage ranges (at a sensitivity of 20,000 ohms per volt): 0 to 15/75/150/300/750/1,500/7,500/30,000 volts. Seven AC voltage ranges (at a sensitivity of 5,000 ohms per volt): 0 to 15/75/150/300/750/1,500/7,500 volts. Three resistance ranges: 0 to 2,000/200,000 ohms, 0 to 20 megohms. Two capacitance ranges: .00025 μfd to 0.3 μfd and .05 μfd to 30 μfd. Five DC current ranges: 0 to 75 μa, 0 to 7.5/75/750 ma, 0 to 15 amp. Three db ranges: −6 db to +18 db, +14 db to +38 db, +34 db to +58 db.

TV-60 is a highly sensitive multimeter.



FAIRCHILD AMPLIFIER

Fairchild Recording Equipment Co. have announced the latest addition to their line of high fidelity equipment, the *Model 275 High Power Amplifier.*

Designed to deliver 75 watts continuously, and capable of peaks up to 150 watts, the Model 275 is rated at 65 watts with IM distortion less than 0.5% for any combination of frequencies within the audio spectrum, according to the manufacturer.

Fairchild's new high-power amplifier.

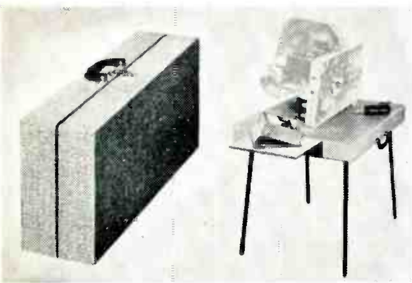
Dimensions of the amplifier are 12½ in. by 7 in. by 7 in.; its weight is 32 lb. Styling is by Raymond Loewy Associates, and matches other high fidelity equipment in the Fairchild line.

Taps are provided on the primary of the power transformer to produce full rated power output at any line voltage from 105 to 125 v. Also included are readily accessible adjustments for dynamic balance of the 6550 output tubes, bias adjustment, and DC plate current balance, the latter acting also as a final adjustment for minute amounts of residual hum due to DC unbalance. Three AC power outlets are provided for tuner, tape recorder, turntable, etc., along with a master gain control. Output impedances are 4, 8, and 16 ohms. Input signal for full rated output is 0.8 v (RMS).

QUICKIE BENCH

A new portable work bench, the *Quickie Bench,* has been introduced by the Argos Products Company.

When folded for carrying, the Quickie Bench resembles a small suit-



Argos portable work bench.

case 26 in. by 11 by 6¼. The bench can be set up in less than 35 seconds, according to the manufacturer, and provides a work surface 26 in. by 22 in., about 25 in. high. In addition, there is a small extension shelf 9 in. by 12 in. on one end for a service meter or tube checker.

The Quickie Bench has a masonite work surface and wedge-type legs which provide great strength and wobble-free support. The sides of the bench are covered with heavy tweed-gray fabric.

The address of the manufacturer is Argos Products Company, Genoa, Ill.

JUNIOR TAPE SPLICER

Robins Industries Corp. are now producing a new-model Gibson Girl tape splicer, the *Model TS-4 Jr.* This new model is smaller, lighter, and lower in cost than other Gibson Girl models. Gibson Girl tape splicers cut tape ends diagonally and trim the tape edges with-

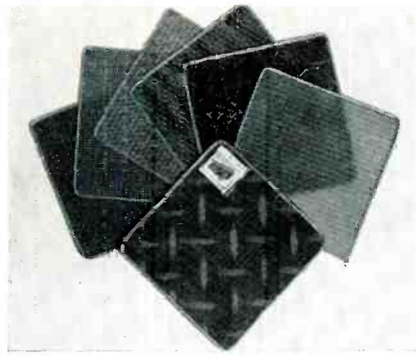
TS-4 Jr. Gibson-Girl splicer.

out the use of scissors or razor blades, and have the unique feature of producing a slightly narrow waist at the splice to prevent contact of the adhesive with recorder parts.

MELLOTONE GRILLE FABRIC

Wendell Plastic Fabrics Corp., manufacturers of *Mellotone* grille fabric, are making a special offer for a limited time to consumers.

If you are interested in grille fabric,



Mellotone plastic speaker-grille fabrics.

write to the manufacturer, stating your color requirements. Upon receipt of your letter, a free sample swatch of an appropriate Mellotone pattern will be mailed to you promptly. The address to which your request should be sent is Wendell Plastic Fabrics Corp., 17 West 17th St., New York 11. N,Y.

CABINETS FOR HI-FI COMPONENTS

King-sized cabinets for hi-fi components are additions to the Cabinart 1956 series of furniture and speaker enclosures for high fidelity installations.

A full 60 in. wide, the *Model 65* will accommodate a radio tuner, amplifier, record changer or player, manual player or professional turntable, tape recorder, and speaker or speaker system. A companion unit, the *Model 65 D,* offers phonograph-record or tape storage in lieu of a speaker compartment. The 65

Continued on page 44

Appropriate Settings: No. 2

HOW would you like to see *your* television set in this setting? The set here belongs to Sylvania Electric Products, Inc., and is pictured surrounded by Jeanne Kessey, "Miss San Francisco" of 1955, and $5 million. The purpose of this setting is to show graphically how much money Sylvania invested in their 1956 television sets. Sylvania officials said that $5 million had been spent in engineering, designing and tooling up for the new *Cabinet of Light* and *Halolight* sets, and in the purchase of raw materials.

The money in the picture was supplied by the Bank of America.

SOUND SERVICING

by Irving M. Fried

Treble Amplifier Stability

Last issue, methods of stabilizing power amplifiers at low frequencies were discussed. This month the more difficult subject of high-frequency stability will be treated — how to increase it, and why it should be increased.

First, why? It can be said, unequivocally, that complete stability at high frequencies will 1) help prevent your tweeter or super-tweeter from being blown out on sharp transients or by parasitic oscillations; 2) produce smoother sound, with less record noise and distortion; and 3) help clean up low-frequency interaction effects — many cases of low-frequency instability are cured when the top end is stabilized.

Equipment required for high-frequency stabilization is an oscilloscope, some form of square-wave generator, a resistive load, assorted small-value capacitors, and resistors of various values.

The first step is to look at the shape of your generator's square wave by connecting it directly to the 'scope. Use a 5-Kc square wave, and fix its shape firmly in your mind. Then connect the generator to your amplifier, and that to the 'scope, using a resistive load across the amplifier output terminals. Now observe the general shape of the square wave — is it still about the same? Or have the smooth top and bottom horizontal lines been replaced by successions of high-amplitude wiggles, which indicate instability? No power amplifier passes a perfect square wave at 5 Kc, but a well-stabilized amplifier should produce wiggles of a very small amplitude compared to the height of the square wave itself. Typical well-designed units have wiggles only 1 or 2% of the square-wave height.

Before you start tearing everything apart, check the wiggles using your loudspeaker as a load. Some amplifiers have larger ripples with a speaker load; a few of them behave better on loudspeaker loads than on resistive terminations. Then, if you don't like the performance on loudspeaker load, proceed with some of the following techniques — using the same speaker as a load.

Often, by bridging the feedback resistor with various values of capacitance, you can clean up a bad square wave. It is suggested that you disconnect the capacitor in the feedback loop (if a capacitor is used) and try replacing it with others having values from half to twice that of the original. A brief explanation of what you are doing: the capacitor increases feedback at higher frequencies, and you are trying to find a value that will increase feedback — and thus decrease overall amplifier gain — just enough to take out the high-frequency peak that occurs when amplifier phase shift approaches 180°. However, if the capacitor is too large in value, feedback is increased beyond this just-right point, and the amplifier will really take off into continuous high-frequency oscillations.

Therefore, if you start with a capacitor of half the value of the one formerly in the amplifier and increase the capacitance in gradual steps, you should have bigger wiggles at first; they should get

smaller as you increase capacitance, and then they should suddenly increase in size. Use the capacitor that gives the smallest ripples on the square wave. If there is no such capacitor in the amplifier originally, try adding various values beginning at about 50 $\mu\mu$fd.

If you can't do much that way, try a gimmick based on the idea of balancing the drive. In the driver stage you can often add a capacitor of such value that, when attached from one or the other plate (either side of a push-pull driver arrangement) back to the cathode of the input voltage-amplifier stage, will balance the high-frequency signal and increase the stability margin. One popular Williamson-type circuit uses a 100-$\mu\mu$fd capacitor from the plate of that driver-section stage that is fed by the cathode side of the split-load inverter. You can try other values, and other placements.

For instance, you might find that a capacitor from plate to grid of the driver will help; try either or both sides. Or a small capacitor from plate to grid of the input stage sometimes works.

Another remedy, used by Williamson in his original circuit, is to parallel the plate-load resistor of the first stage with a combination of resistor and capacitor in series, of such value that you roll off high-frequency response in the stage at the critical frequency by just a few db. You might want to try this, too, though it will help here to have a wide-band audio oscillator. With this you can scan the upper-frequency range (from 50 to 200 Kc, roughly) and find where you have your peak. The resistor-capacitor combination is then chosen to roll off the response just a bit at that frequency.

All these methods are tedious and time-consuming — unless you happen to be lucky. High-frequency stabilization is not an easy job. If it were, I suppose there wouldn't be so many amplifiers on the market today that need stabilization. If you go too far with any of the methods listed, you can get into trouble either by making the amplifier even more unstable or by rolling off the high-end response too much, adversely affecting the sense of "liveness" in listening.

If you can't seem to make the wiggles come down by any of these methods, you can always replace the output transformer with one having better bandpass and phase characteristics. Cheap output transformers and stable heavy-feedback amplifiers don't go together. At the expense of increased distortion you can always reduce feedback 6 db by doubling the value of the feedback resistor. If that doesn't make your amplifier stable, you had better give up or get rid of it.

As a final check after stabilization, try putting across the output terminals both your speaker and a fairly moderate capacitance, say .003 μfd. This is to simulate the loading effects of long speaker-cable lengths. If the amplifier's square wave is still good, you have reason to feel quite comfortable about its high-frequency stability.

TIPS FOR THE WOODCRAFTER

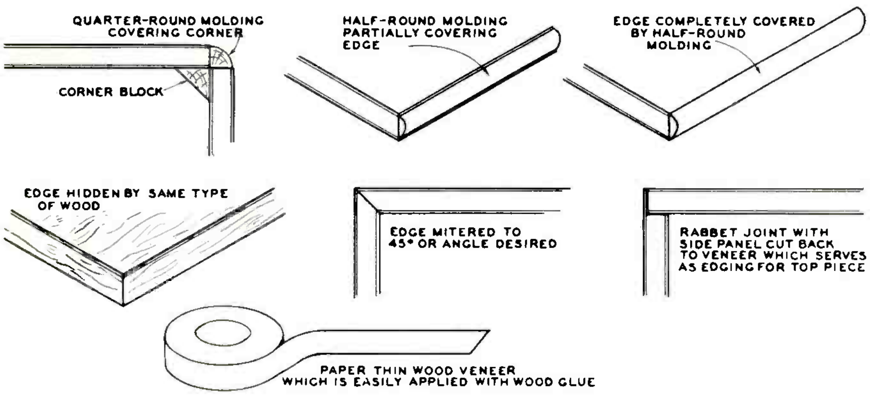
by George Bowe

Plywood

Plywood means many things to many men. The weekend fisherman will tell you it's what his boat is made of; the ping-pong player will point to his table and paddles; the home builder starts with plywood concrete forms and builds the product into the floors, walls, cabinets, counters, doors — in fact, almost any part of the house. And what a boon plywood is to the constructor of high fidelity systems!

Knowing something of the plywood story will give the home craftsman a better understanding of the ways in which this versatile product may be used. Plywood, as we know it, was first produced in 1905, although the art of veneering was known to the early Egyptians. The difference between veneer and plywood is simply this: veneer refers to the very thin sheets of wood that are sawed or peeled from logs. These sheets may be glued to pieces of solid wood for decorative effect in the process known as veneering, or they may be bonded to other sheets of veneer to form plywood. Thus plywood is built up of layers of veneer with the grain of adjacent sheets (or plies) running at right angles. Alternating the direction of the grain provides great strength and rigidity.

Plywood is made from "peeler" logs especially chosen because of their size and quality. Diameters of 4, 5, and 6 ft. are common and, occasionally, giant logs 9 and 10 ft. in breadth come into the plywood mill. The usual lengths are 8, 10, and 12 ft. After the bark has been removed, the logs are placed in a gigantic lathe which turns them against a long, razor-sharp knife. The thin, continuous sheet of wood that is peeled from the log looks very much like wrapping paper being pulled from a roll. As it passes a long conveyor table, the veneer is clipped to specific widths, eliminating any defects, and is sorted according to grade. These sheets of green veneer are then put through long automatic driers where the moisture content of the wood is greatly reduced. Next, glue or synthetic resins are applied to the veneer and the sheets are stacked with the grain of each at right angles to the sheets immediately above and below it. The freshly glued sheets are placed in huge hydraulic presses where the adhesive sets under heat and pressure. When the panels emerge from this process, they are sanded to specified thickness and trimmed to specified lengths and widths. This would include the very popular 4 by 8-foot size panel with which most of us are familiar. The better grades are also turned out in widths of 30, 36, 42, and 48 in. The lengths vary from 5 to 12 ft. in intervals of 1 ft. Other widths and lengths are manufactured for special applications.



Methods of finishing exposed edges of plywood. Explanations given in text.

The glues used in plywood are stronger than the wood itself. In general there are 2 types employed: moisture resistant and moisture proof. Moisture-resistant glue is used to produce plywood for interior use, and includes protein-type adhesives which are principally synthetic resins. The moisture-proof glue is used on exterior plywood to withstand exposure to the elements. Some of these glues are so waterproof that the plywood can undergo repeated cycles of boiling and drying without damage; nor are they affected by salt water. A third type, known as dry bond, is satisfactory when there will be no exposure to dampness or high humidity.

The most commonly used plywood is made of fir, although there is a wide choice of other woods including pine, knotty pine, redwood, and a good selection of hardwood veneers. Various surface styles offer interesting effects in finishing: striated, V-grooved, accented grain, brushed, and smooth. For most cabinet work the smooth-surface plywood is most desirable. If the finish is to be natural wood, give thought to one of the hardwood-faced plywoods such as mahogany, cherry, walnut, birch, ash, sweet gum, birdseye maple, elm, red oak, and Philippine red lauan.

All fir plywood is not the same grade. Since the face of the panel and the back might vary in quality, it is possible to have several different grades. Perhaps you've heard the phrase "good one side" or "good 2 sides" referring to plywood. The former simply means one side is free of imperfections and the other side is not. "Good 2 sides" means both sides are good quality. The use determines the grade to select for the job. For instance, paneling on a wall would require "good one side" only, since the back of the panel would never be seen. In cabinet work where both sides would show, "good 2 sides" would be the choice. Actually, fir plywood is graded according to the following scale:

A — high standard.
B — smooth, paintable.
C — knotholes and splits.
D — knotholes, used for inner plies of plywood.

The grading of the individual panel would comprise 2 of the letters above, one for the face and the other for the back. A-A would indicate top quality on both sides, A-B would mean one side suitable for a natural finish and the other where a smooth surface is desired, etc. In hi-fi construction, grade A-A plywood is essential for cabinet doors and other areas where both sides will be exposed; cabinet sides, tops, bottoms, and panels will do very nicely, in most cases, with grade A-B.

Hardwood-veneered plywood has its own system of grading:

Custom — no knots or patches.

Good — tight, smoothly cut veneer, grain matched at joints.

Sound — no open defects, veneer not matched, possible stain and mineral streaks.

Utility — discolorations visible.

Reject — large knotholes, splits.

Once the grade of plywood is decided, it's the thickness that gauges the price. In other words 1/8-, 1/4-, 3/8-, 1/2-, 5/8-, and 3/4-inch plywood has a definite price already established according to the thickness. All that remains is to find the square footage. For example, suppose we want to find the price of a piece of plywood 3/4 in. by 4 ft. by 8 ft. We simply find the square footage by multiplying width times length, which is 4 ft. by 8 ft. = 32 sq. ft. The price of this grade of plywood at 3/4-inch thickness is established at 40¢ per sq. ft. Thus this panel costs 4 × 8 × 0.40 = $12.80; or 32 sq. ft. × 40¢ = $12.80.

When buying plywood, you'll save money by buying the whole panel or half a panel. Let me explain. If you ask the local lumber yard or cabinet shop for a piece of plywood 3/4 in. by 2 1/2 ft. by 4 ft., here's what frequently happens. The plywood is cut to the size you request, but you are charged for a complete half panel, 3/4 in. by 4 ft. by 4 ft., because the other 18 in. is simply scrap to the shop. They add it to their accumulation and you go home without it after paying for it. So always order a half panel (or a full panel). Ask the dealer to cut it to the desired measurements, but also take the remaining piece home with you. A strip of plywood 18 in. wide and 4 ft. long could easily make a couple of fine instrument or control panels.

While we're on the subject of economy, its a good idea always to keep in mind standard plywood sizes when planning hi-fi cabinet work. It not only saves money; it saves time and work. Since a standard-size plywood panel is 4 ft. by 8 ft., try to plan your work areas to fit those dimensions, such as 2 ft. by 4 ft. or 1 ft. by 2 ft. or any other measurement that makes use of all the plywood without leaving any waste. This method also applies when buying molding or other wood because it comes in standard lengths of 8, 10, 12, 14, and 16 ft. Just a word of caution! Never forget that the saw you're using always steals a bit of wood. So when you cut a 24-inch piece of plywood in half, you won't get two 12-inch pieces. Each will be a trifle short. Sandpapering will make a further reduc-

Continued on page 42

Tape News and Views

by J. GORDON HOLT

Frequency Response and Bias

To anyone who has been fiddling with tape recorders since the late 1940's, the advertising claims made for some of today's inexpensive units might be likely to bring a patronizing smirk of skepticism or an occasional choleric outburst about how ethical standards and integrity in the high fidelity business are all going to pot.

But when you've been weaned on an iron-bound thumb rule to the effect that a recorder's top response is in a direct kilocycle relationship to the tape speed, giving a maximum of, say, 7,500 cps for 7.5 ips, it may be somewhat difficult to take seriously a manufacturer's claim of 2 db down at 15,000 cps, *at 7.5 ips,* for a non-professional tape machine.

Actually, though, these seeming flights of promotional fancy are usually based on fact, as anyone with an audio oscillator and an AC voltmeter can readily determine for himself. Most of the new home recorders will, indeed, produce measured flat response out to their rated limit, with a few units even surpassing the advertising claims. The main reason for the ever-increasing high-frequency limit is the ever-decreasing gap widths in modern recorder playback heads.

I'm not saying that improvements in recording tapes haven't contributed something to this rather remarkable state of affairs, but the tape factor is a relatively minor one so far as response is concerned. A couple of rolls of Scotch tape that I've been using for almost 8 years have almost as good high-frequency response as the best of today's supertapes. The acetate backing may be pretty much desiccated, and the coating may be starting to fall off here and there, but the oxide formulation of the earlier tapes doesn't seem to be too unlike what we're getting in 1956.

So the provable superiority of later model recorders is mainly attributable to the heads, plus one or 2 other tricks.

The ultrasonic bias current in a recorder has a tremendous influence on high-frequency response, distortion, and signal-to-noise ratio of tape recordings. Purity of the bias waveform affects the noise characteristics, but the *amount* of bias current flowing through the record head largely determines how the machine will perform with respect to distortion and treble response. Too little bias current will give increased high-frequency response, a reduction in output from the tape, and massive amounts of distortion. Too much current will drop the distortion to an insignificant level, but can cause saturation of the steel around the record head gap, increasing the effective gap width and so pulling down the high-end response.

Some recorder manufacturers have shrewdly observed that tape recorder buyers generally tend to judge a unit by its high-frequency response, so they have set the bias current in their products accordingly. This slightly low setting raises the distortion, but it produces beautiful response ratings, and since it has been standard practice for most home recorder manufacturers to ignore such petty matters as distortion ratings, the buyer is not always aware that he is sacrificing truth for beauty when he carts one of these units home.

For professional tape recorders, there *do* seem to be some standards set up for tape distortion at certain recording levels. One manufacturer, for instance, rates his units for a maximum of 1% total harmonic distortion at 400 cps at "Normal Operating Level", which is zero db on the record meter. The frequency-response ratings are, presumably, for the bias setting that gives this distortion level, so a prospective purchaser knows where he stands when he gets a professional recorder.

With home recorders, it is generally a safe policy to choose a unit having slightly less-phenomenal high-frequency ratings, unless the specifications also state distortion at a certain recording level.

This may sound like heresy, coming from someone who so much likes the additional definition that goes with every little extension of high-frequency range, but it must be admitted in all truthfulness that the audible difference between 2 db down at 10,000 cps and the same drop at 15,000 cps is often slight. It gives you a nice cozy feeling to know that the treble response on your recorder goes out so far it makes your speaker look sick, but when this extension is achieved at the expense of higher distortion (which is audible through the whole reproduced frequency range) it is likely to be little comfort.

On the other hand, it is not uncommon for a recordist to find that his recorder with the 13,000-cps rating sounds great and has crisp tinkling highs, but exhibits appallingly poor high-frequency response on bench tests. I have heard of a number of cases like this: users were completely satisfied with their recorders until they were seized with the impulse to *test* them. A number of these users were so outraged to find the highs 12 db down at a frequency for which they were supposed to be 2 db down that they immediately sent back the recorders and took to writing poisonous letters to manufacturers and magazines.

In most cases, though, there was really nothing the matter with the recorders; they were perfectly in adjustment and within specifications. The test procedure was at fault.

In a tape recorder, the signal fed to the tape must be at a level representing a compromise between tape hiss, which determines the minimum recording level, and tape overload, which sets the volume limit. But to overcome the inherent high-frequency losses in record and playback heads, the highs must be boosted while recording, and if the natural harmonics in speech and music were produced with the same intensity as the fundamental sounds, this high-frequency boost would certainly cause tape overload.

In practice, though, the frequency distribution of natural sounds is not a series of equally intense harmonics all the way out to our limit of hearing. There is a rapid tapering of intensity as the frequency rises, so the 10,000-cps component of a violin sound is far weaker than the fundamental. The high-frequency boost, because of this fact, does not approach the tape's maximum recording level.

But an audio signal generator obviously isn't a violin. Its output is, or at least we expect it to be, uniform all the way out to its limit. If we try to feed 10,000 cps to the tape at the indicated maximum level, the tape will be driven into overload because of the built-in high-frequency boost network. It simply cannot be magnetized to the intensity of the signal being fed into it. The result is an apparent collapse of the high end on playback.

Whenever making frequency-response tests on a tape recorder, then, the level at 400 or 1,000 cycles must be set so that it is between 15 and 20 db *below* normal recording level, and the entire response check run at that level. Normal recording level for a machine that lacks a VU meter can be considered as the point at which a "magic eye" completely closes, or at which a neon glow lamp just begins to register overload. As read on an AC voltmeter, 15 db down is just shy of 1/5 this "normal" recording level. Under these test conditions, any recorder should be expected to come up to its manufacturer's frequency-response specifications. If it doesn't, it either needs adjustment or the manufacturer has stretched things a little bit in his promotional copy.

When running response checks on a recorder, it is a good idea to remember that different brands of tape vary considerably with respect to *optimum* bias current. The differences in apparent treble response are very often directly tied in with bias requirements, as was pointed out before. A tape that operates at its best with a slightly lower bias level than that for which the recorder is set will invariably give worse high-frequency response than one with a higher bias requirement.

There are also one or 2 tapes on the market that seem to generate unusually high intermodulation distortion when fed by a signal that is too low for the recorder to reproduce. This can only be checked by ear, but the effect is quite obvious as a swishing sound on playback of a medium-level (— 6 db) recorded 15- or 20-cps note. This sound, as a matter of fact, bears some resemblance to that of a speaker that is doubling furiously due to overload from a very low-frequency signal.

It should be emphasized that a tape recorder can never be expected to equal the performance of a good amplifier. This should be quite obvious from the nature of the tape medium itself, but it still isn't uncommon to hear perfectionists beefing because their recorders have a slight, broad dip in the middle range and a gentle collapse at both ends of the rated range. As long as a recorder covers its *rated* frequency range, within the specified decibel limits, the owner has nothing to complain about. On the other hand, a recorder which is advertised as simply having a certain frequency range, with no decibel limits specified, may be expected to be down at both ends anywhere from 4 to 20 db. This can't exactly be held against the manufacturer, either, since there is still some disagreement in the industry as to just what constitutes "cutoff" in a response curve. The figure usually accepted is 4 db down, but then, as I said, there's not much agreement, particularly on the part of manufacturers of budget recorders.

The most common cause of poor high-frequency response in a tape recorder is wear on the playback and/or record heads. Recorder heads are so constructed that when they start to wear to some appreciable degree, the exposed gap becomes wider, reducing the high-frequency response. The only remedy is to replace the worn head, which will then have to be re-aligned either with a commercial alignment tape or by ear, using a pre-recorded tape of music. The latter expedient isn't very accurate, but it will serve the purpose for users who aren't too concerned about the last octave at the high end.

Since head wear is just as real a problem with tape recorders as is stylus wear in a phonograph, albeit lacking in the destructive aspects, it will pay a recordist to consider it when buying a new machine. Indications are, so far, that the useful life of a good record or playback head, operated at moderate tape tensions, is over 2,500 hours; but misuse or poor design of a recorder can greatly shorten its life expectancy.

For instance, it is evident that a recorder which passes the tape across the heads under tension during the rewind and fast-forward functions will materially shorten head life. Similarly, a recorder which uses felt pads to force the tape against the playback head may

also be expected to show increased head wear above one that maintains its tape contact only by drawing the tape at a slight angle across the pole pieces and keeping it there by moderate drag braking on the supply reel.

One answer to the problem is the replaceable pole pieces that are to be found in some heads. While such heads are a little more likely to get out of alignment with use than are other types, they drastically reduce the cost of worn head replacement.

Another precaution that the user may observe, with any recorder which lacks provision for removing the tape from the heads during rewind, is simply to pull the tape from the head assembly and run it directly between the reels during the high-speed functions. This is a minor operational inconvenience compared with the effort and expense involved in replacing a prematurely worn record or playback head.

We'll have more to say about bias settings at a later date.

Meanwhile, I've had more than enough opportunity to test thoroughly the Fen-Tone B&O-50 microphone that was given favorable mention in this column some time ago.

Later recording ventures with this mike indicate that it does indeed have an extremely smooth frequency response, but its' my impression that its entire high end rolls off gently out to its specified limit. I may be mistaken about this, but in direct comparisons with an Altec 21-B and a Telefunken U-47M, the B&O-50 was found to have a slightly more subdued high end. As a spot mike, for close pickup of voice or instruments, it is extraordinarily good; for pickup of larger groups, however, it lacks the crispness of either the Altec or Telefunken.

Admittedly, this is an unfair comparison, since the mikes it was compared with cost at least 4 times as much, but it seems to be so much better than any I've run across in its own price class that I was curious to see how it would stand up to some of the top microphones currently available. It does remarkably well. But it is not a unit that can be recommended for use with recorders that are a little weak at the high end. It performs at its best when the whole system is very flat and wide-range; but it also does an excellent job when a user's loudspeaker tends to be peaked in the "presence" range. A bright-sounding microphone can give painfully hard sound from a "brilliant" loudspeaker system, while a smooth unit adds nothing on its own, letting the speaker add presence.

In case you're thinking of running right out and buying one, the B&O-50 is a low-impedance mike, which means that it will not operate with most home recorders without an additional input-matching transformer. Its impedance is 50 ohms; output on MUSIC position is 55 db below 1 volt per microbar; directional pattern, bi-directional; and generating element, ribbon.

On each of its "live" pickup sides, it exhibits maximum high-frequency response for sounds striking the front or rear from straight-on, at right-angle incidence. As the sound source moves around toward the dead sides, treble response begins to fall off slightly more rapidly than the rest of the range, but no more so than in any other ribbon velocity microphone.

Also, like other ribbon mikes, it may tend to be rather fragile, and will not take well to being dropped or blown into. It's not a toy; it is strictly a top-quality unit, and I still find the price a little hard to believe. But there it is!

READERS' FORUM

Gentlemen:
I am very interested in your article "TV Can Sound Better" in the December issue. Can you tell me where I can purchase the Triad S-31A output transformer?

Robert Erb
Logansport, Ind.

Although the Triad S-31A output transformer is not listed in some of the catalogues, most parts distributors have it in stock and can ship it immediately upon order. — Ed.

Gentlemen:
I have noticed in "Readers' Forum" a request for an article concerning the Theremin. I would like to endorse this request, as I, too, am interested in building a Theremin but cannot seem to find much information on them.

Leon Florschuetz
Champaign, Ill.

Mr. Florschuetz is one of many who have expressed interest in a Theremin article. We have contracted for what promises to be a very good one. It should appear within the next few months. — Ed.

Gentlemen:
I have read the editorial on page 15 of your January issue with much interest.

I am sure that you will be pleased to hear that in a number of the top Magnavox High Fidelity radio-phonographs both input and output jacks are provided for tape recording or other external sources. These are at low impedance, which you mentioned, and are tied into the circuit in such a way as to assure the absolute minimum of signal distortion. We have been providing such convenience input and output jacks for some time and have not found it necessary to increase our prices as a result.

From my own experience, I have a considerable number of complete operas recorded off the air and have found them extremely satisfactory.

R. H. G. Mathews
Director of Public Relations
The Magnavox Company
Fort Wayne, Ind.

EDITORIALS

We think it's about time something was said on the subject of instructions supplied (or not supplied) with sound-system components. Before going after the guilty ones, though, we ought to make it clear that many manufacturers furnish accurate and complete specifications, installation and operating instructions, and sometimes maintenance tips with their equipments. They are to be congratulated, not only for providing this valuable assistance to buyers, but for their astuteness in recognizing that it builds good will for themselves and for the entire audio industry.

Unfortunately, however, such completeness is an exception to the rule. A few manufacturers furnish hopelessly inadequate or inaccurate information; even the specifications are nebulous and imprecise. Not long ago we worked with a power amplifier having what looked like an AC outlet on the chassis. The instructions didn't explain what it was for, or didn't explain it clearly; at any rate, we spent a frustrating 20 minutes trying to get the tubes to light up before we discovered that the receptacle was for a remote on-off switch. The buyer had to wire up the switch himself, or he could use the amplifier normally by inserting a shorted AC plug into the receptacle — but this wasn't explained, and who, without tracing the circuit, would conceive of putting a shorted plug into a receptacle that had all the earmarks of being hot with 115 volts AC? And suppose the buyer couldn't trace the circuit? This same "instruction" manual omitted performance specifications entirely, limiting itself to saying smugly that the amplifier was so good that it should be used only with the finest associated equipment. Too bad, for it *was* a superb amplifier.

That was a case in which the manufacturer simply didn't realize the need for instructions; perhaps he was so familiar with the product himself that he thought its workings would be known to everyone. In other instances manufacturers still gear their instruction manuals to the professional market, to which they sold before home high fidelity became commonplace. Most of their sales are now to home consumers, but they don't seem to be aware of that. As a case in point, take a fine transcription pickup arm that we recommended to a friend recently. He bought one, took it home, tried for hours to install it, and finally called us that night. Having a limited knowledge of technical matters, he couldn't translate the instructions *for broadcast station use* into hi-fi how-to-do-it. There were 3 wires (for a balanced input and ground) coming from the arm base; how was he to know, without being told, how to connect those 3 wires for an unbalanced input to his preamplifier? We've often wondered how many others had the same difficulty. It would have taken 1 or 2 sentences and a small line drawing on the instruction sheet to save us a late-evening trip and some embarrassment.

We could give many other illustrations, but perhaps some constructive suggestions would be more appreciated:

First, complete information on the product itself (and how it works) should be furnished. This would include a well-planned schematic diagram and, if mechanical parts are involved (as in a record changer or tape recorder), clear drawings of the parts showing their physical relationships with one another. Normal-performance specifications are essential. A brief outline of the functions of the various stages and parts should be given, with parts numbers, values, and ordering information.

Second, installation and operating instructions should be detailed and written as simply as possible. This is the section that will be of most immediate concern to the buyer, and it must be assumed that he is 1) unfamiliar with the product, 2) going to install the unit himself, and 3) possessed of only the most rudimentary knowledge of sound equipment and its interconnection. Drawings are helpful and should be used liberally. Full-size mounting and escutcheon templates are more useful than is generally realized. The purpose of every knob, receptacle, and control should be explained fully. Important precautions should certainly be listed. Matters such as position or orientation, terminating impedances, and ventilation ought to be explained. To some laymen the significance of these things is completely unknown, and that fact ought to be kept in mind by writers of instruction manuals.

Third, maintenance information should be supplied in as great detail as possible. For electronic gear, normal-voltage readings for each stage are invaluable in finding troubles. Complete lubrication information on mechanical equipments is needed. Adjustments that should be made from time to time, or when parts are replaced, ought to be specified and information given on how to make them. For tape recorders particularly, proper methods of performance testing should be outlined. — R.A.



First operations in making the built-in cabinet are laying the platform, and nailing 1-by-4-inch boards to wall for top supports.

BUILD IN YOUR HI-FI

by SCOTT J. SAUNDERS

THE design of cabinets to house hi-fi components is usually a matter of individual taste. In large rooms, several separated units can be accommodated easily without crowding. But for small rooms (such as mine, which is 8 by 10 ft.) it is better to put all the components in a single wall-to-wall cabinet that is functional as well as decorative. In an arrangement of this type there is space enough for the addition of new equipment also.

Dimensions and materials of the unit described here can be varied to suit your needs and budget. Placement of the amplifier, record changer, speaker, and other components if used can be changed around. The purpose of this article is to describe the construction of a practical and good-looking wall unit that is readily adaptable to any normal needs; specific details can be modified as desired.

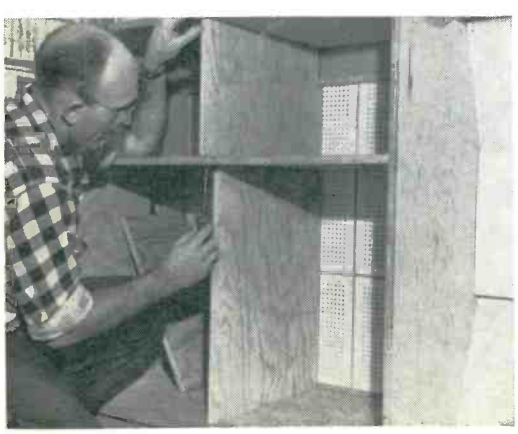
This cabinet is 92 in. long, 39¾ in. high, and 21 in. deep. It is made mostly of fir plywood, since this wood is economical, easy to work, durable, and it lends itelf to a variety of finishes — stain and varnish, paint, formica covering, or simply shellac and wax. Most lumber yards will cut plywood panels without charge to the sizes you want.

Construction Procedure

The first step in construction is the platform, or bottom frame (92 by 18½ in.) put together with 1-by-4-inch common pine and flushed to the 3 walls (see Fig. 1.) Nail the short spacers to the front and back framing members with 2-inch brads, coating the edges with glue before assembly. The cabinet floor is a single piece of ¾-inch plywood (90 by 20 in.) nailed to the frame, the front extending over the platform by 1½ in. After the floor is in place, nail or screw the cleats for the shelves to the ¾-inch plywood side panels and 2 center uprights (20 by 34½ in.). The side panels may then be nailed or screwed to the side walls, preferably into wall studs. Two 1 by 4 by 36½-inch boards are nailed to the back wall 36 in. from the cabinet floor to the top edge. This will act as a support for the cabinet top (92 by 20 in.) which is nailed down.

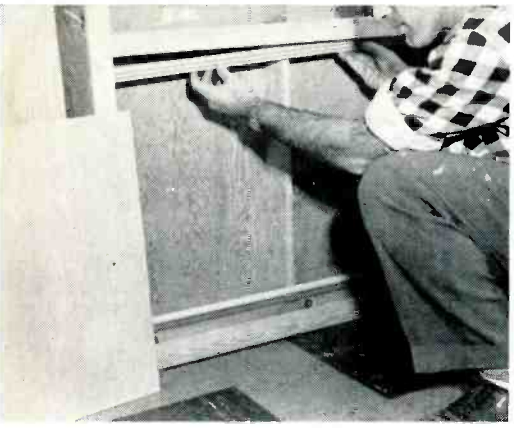
If a wall-to-wall unit is not desired, the side panels should be attached to the top and bottom with extra care, since they will have to provide the rigidity formerly given by the side walls of the room. This can be done by screwing 1-by-1-inch bracing pieces along the

The shelves and partitions are installed.



1-by-2-inch facing strips are then added.

Fitting the amplifier control front panel.



Plastic tracks for the sliding doors are nailed in place, and the record changer or turntable slides are installed on upper shelf.

top and bottom edges of the side panels. Set the panels in place and then screw the top and bottom members of the cabinet to the braces. Placing the 2 center uprights in position will help to support the top. A back covering will be necessary, of course, and for this you can use 1/8-inch plywood or 1/8-inch hardboard, attaching it to the plywood frame with 1-inch nails.

All nail heads exposed to view should be driven below the wood surface with a nail set, and the holes filled with plastic wood.

The 2 uprights in the center (20 by 34½ in.) support the top and become the sides of the loudspeaker enclosure; this is 20 in. deep by 36 in. high by 16 in. wide, or an approximate volume of 6 cu. ft. That is the recommended volume for the University 6201 dual-range 12-inch speaker, which I used. Other dimensions can be used for other required volumes. The uprights are glued at the top and bottom, and nailed in place. Additional rigidity can be obtained by using dowels, but this is tricky cabinet work and is recommended only for the experienced perfectionist.

The top left-hand section of the cabinet houses the amplifier and record changer, which are separated by a ¾-inch plywood partition. Drill a 1-inch hole at the bottom rear edge of this partition before assembly, to allow for later passage of hook-up wires and leads.

Directly below are 2 shelves and 2 record storage areas. Make the dividing partition here no more than 18 in. deep, since room will be needed for the sliding doors and their tracks.

The shelf areas on the right-hand side of the cabinet were not finished, but will be when the tuner, tape recorder, and other equipment are purchased. If you already have them, follow the same instructions given for the amplifier and record-changer compartments.

After all the divider partitions have been joined firmly, nail 1-by-2 facing strips to the front edges of the cabinet on both sides, being sure to leave the bottom strip off until the sliding doors have been inserted. The edges of the speaker enclosure are covered with ¾-inch square pieces that are nailed flush with top and bottom surfaces.

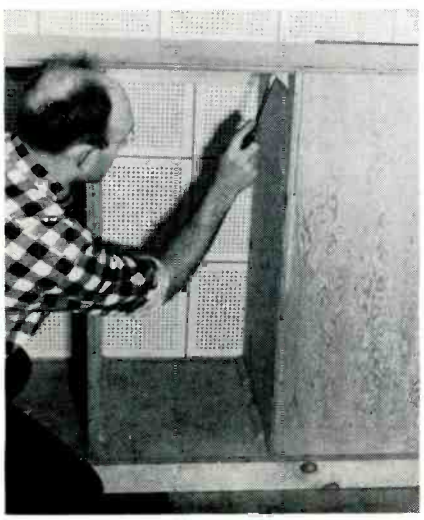
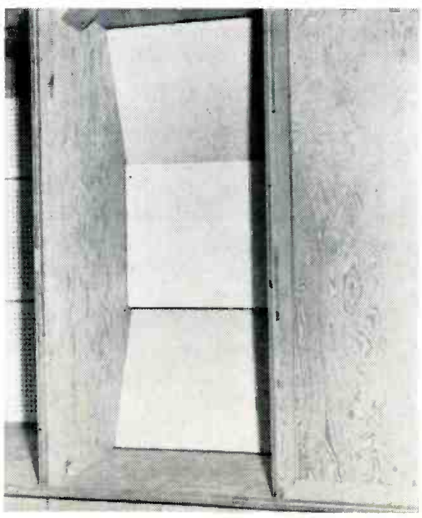
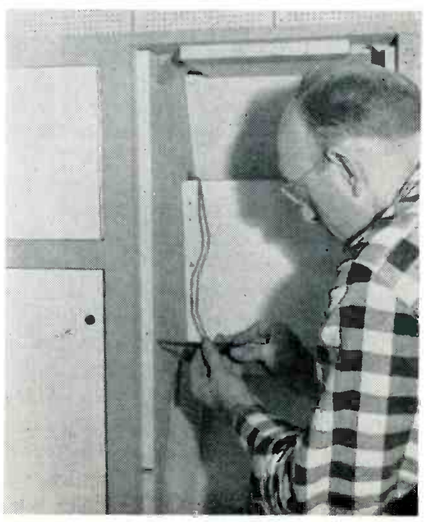
The front panel of the amplifier compartment is a piece of ½-inch plywood cut to fit flush with the sides and bottom, but with a 1-inch space at the top to provide for necessary ventilation. The panel is held in place by screwing it into two 1-by-1 strip supports inside.

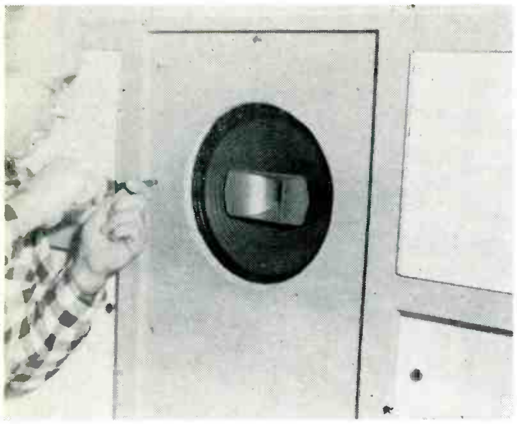
Drill the holes for the amplifier control knobs to conform with measurements taken from the template supplied by the manufacturer. The illustrations show a Bogen DB-110 amplifier, which is held rigidly in place by 2 screws that are brought up through the shelf and inserted into holes supplied for this purpose in the chassis base.

A simple way of mounting the record changer so that it will slide in and out of the cabinet is to attach 2 ball-bearing slide arms to the holes in the bottom of the changer base. In the case of a V-M changer, it is only necessary to remove the 4 screws that hold the small rubber feet to the base and screw the narrow part of the slide arms into the same holes. Positions of holes in the slide arms and changer base are identical.

Adjust the slides on the cabinet shelf so that at least half the changer can be pulled out of the drawer. Mark this position and attach the wide bottom rails by screwing the forward ends to the shelf. Pull the changer all the way out until it stops and screw down the rear of the slides. This eliminates the

Cleats and corner blocks are used in the speaker compartment to make it strong and to screw on baffle boards and speaker panel.



Speaker panel, held by screws in cleats.

need for a special mounting board if you already have a changer with a base. A little grease on the ball bearings of the slide arms will keep them rolling easily.

The top doors on both sides are ¾-inch plywood cut with care to allow no more than 1/16 in. clearance on all sides. Piano-type hinges were used because they are easier to install than standard types and present a neater appearance. After cutting the hinge to the correct length, screw it first to the door edge, with about ⅛ in. of the hinge protruding beyond the edge. Set the other half of the hinge on the door frame with a ⅛-inch overhang and spot the hole positions. When properly hung, the doors should swing freely without binding.

The sliding doors below are cut from striated plywood and should slip easily into the upper and lower plastic tracks. These tracks are available in 6-foot lengths and are easily sawed and nailed. They should be flush with the inside edges of the facing strips. Drill small holes about 8 in. apart in the tracks to simplify nailing; 1-inch or smaller finishing nails are adequate.

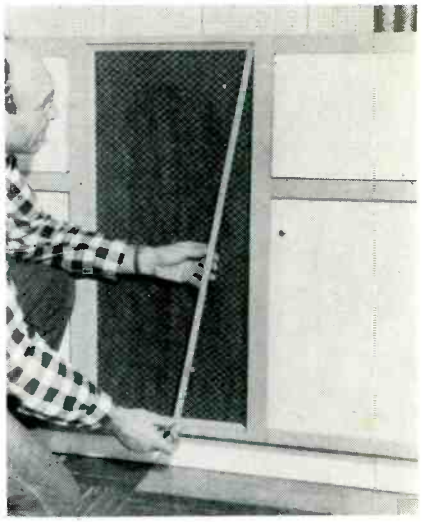
Before inserting the striated doors, plane or file bevels on both top and bottom edges so that they will slide easily in the tracks. This is necessary because the thickness of the striated board is slightly greater than the track width. For doors that will be easier to install, use ¼-inch tempered hardboard. After you have attached the small button-type chromium door pulls (2 in. from either side), sand and finish the doors and then put them in the tracks. Follow this by nailing on the bottom facing strips.

LIST OF MATERIALS

Top and bottom	¾" plywood	(2) 20" x 92"
Uprights	¾" plywood	(2) 20" x 34½"
Side panels	¾" plywood	(2) 20" x 34½"
Main shelves	¾" plywood	(2) 20" x 36½"
Base (front & back)	1" x 4" pine	(1) 92", (1) 90½" long
Spacers	1" x 4" pine	(3) 17", (2) 17¾" long
Hinged doors	¾" plywood	(4) 11" x 17¼"
Sliding doors	striated ply	(4) 19" x 21½"
Facing strips (vertical)	1" x 2" pine	(4) 36" long
Facing strips (horizontal)	1" x 2" pine	(6) 34½" long
Partition dividers (upper)	¾" plywood	(2) 12" x 20"
Partition dividers (lower)	¾" plywood	(2) 18" x 23½"
Speaker panel	¾" plywood	16" x 34½"
Amplifier panel	½" plywood	10" x 17⅞"
Shelves	½" plywood	as needed
Cleats for shelves	1" x 1" pine	as needed
Side supports (speaker panel)	1" x 2" pine	as needed
Side supports (acoustical material)	1" x 1" pine	as needed
Molding (over grille cloth)	¾" cove	cut to fit, miter
Molding (outside of cabinet)	1" cove	cut to fit, miter

Miscellaneous: — Piano hinges, Celotex, perforated acoustical tile, 1" knobs for hinged doors, ½" knobs for sliding doors, plastic door tracks, record changer slide arms, grille cloth, 1" brads, 2½" finishing nails, 1¼" wood screws, glue, plastic wood, sandpaper.

Piano-hinged doors cover top sections.



Molding is added to speaker panel.

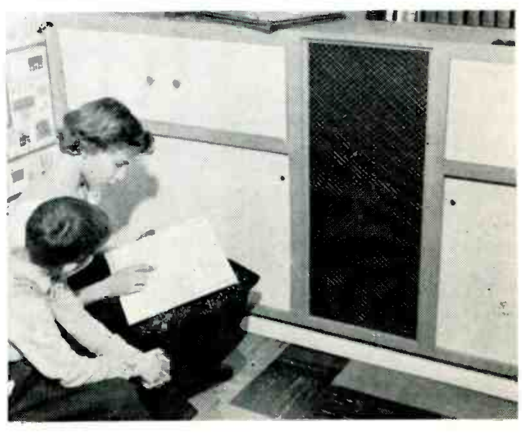
Special attention should be given the loudspeaker enclosure. All joints must be well reinforced with glued and screwed cleats (quarter-round molding will do well), and the sides and side supports constructed similarly. Note that the wall behind the cabinet is faced with perforated acoustical tile, a precaution to keep some of the speaker sound from penetrating to the room on the opposite side.

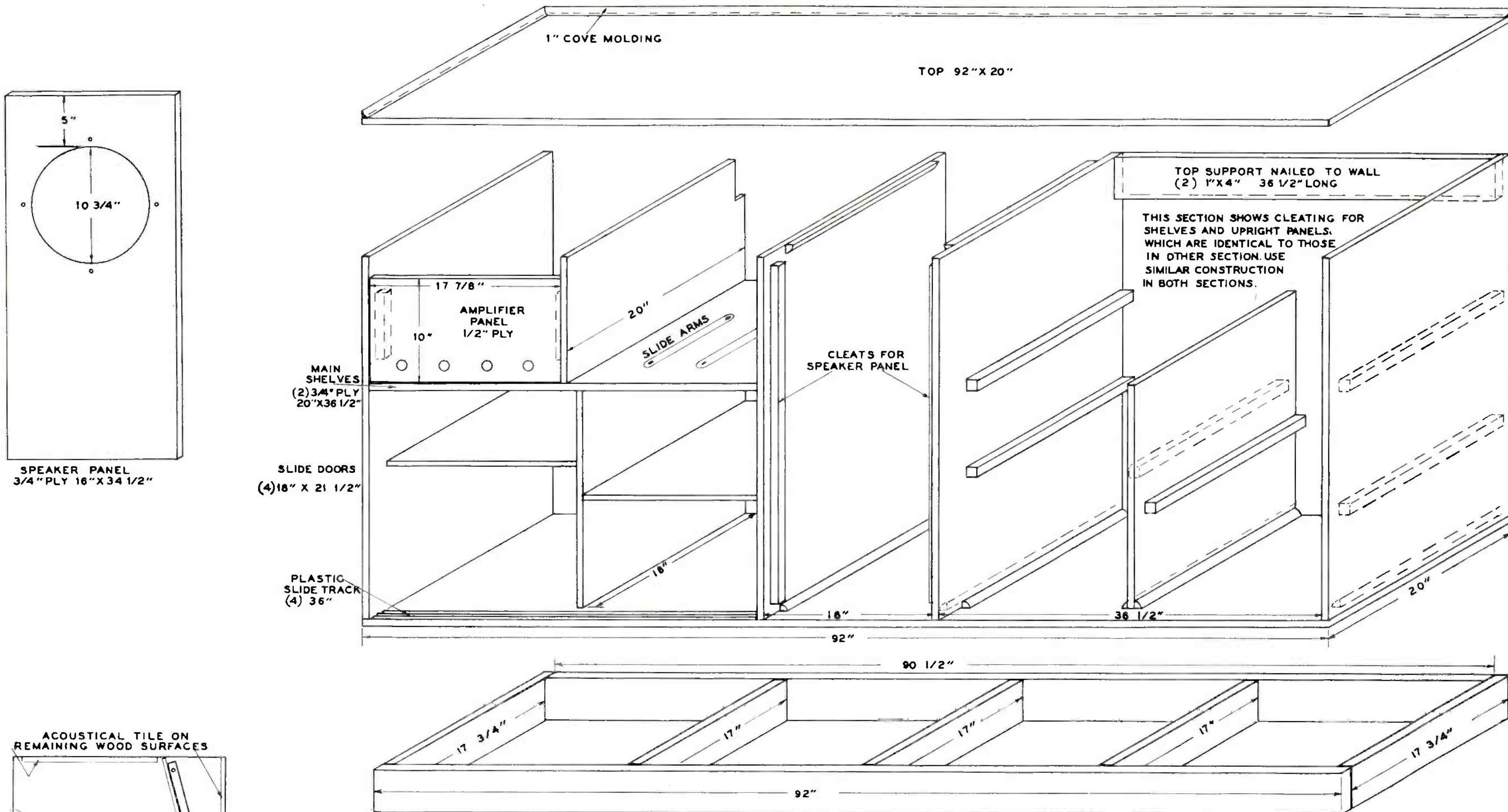
To reduce the possibility of booming and to provide for an infinite-type baffle, 2 successive layers of sound-proofing material were placed 3 in. apart, one in front of the other. Three pieces of Celotex (14 by 16 in.) are nailed to side supports and overlap from top to bottom. The second layer is a single piece of the same material positioned in the exact center of the enclosure. At least 50% of the remaining plywood surfaces were covered with the acoustical tile. The brilliance control supplied with the speaker was attached on the right side of the enclosure and concealed behind one of the doors.

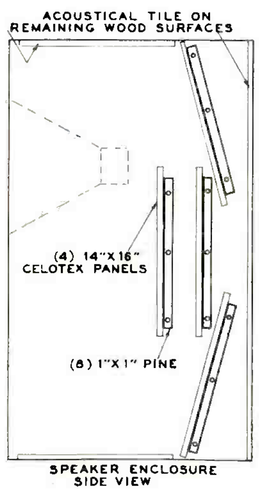
The hole for the speaker is 10¾ in. in diameter; the speaker is attached with machine screws and nuts. The speaker panel is secured by 12 wood screws into the side supports. Speaker

Continued on page 44

Complete, painted wall-to-wall unit.







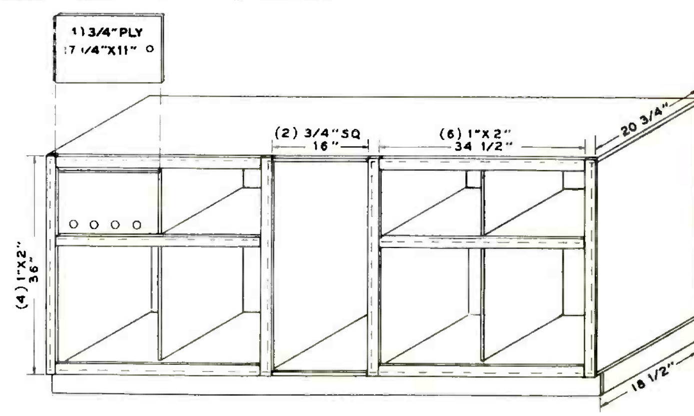


Fig. 1. Cabinet that can be built to fill entire room wall or only part of a wall. Space is furnished for all components of a home sound system, as well as for record and tape storage.

Designing Your Own Amplifier

by Norman H. Crowhurst

Part I: Voltage amplifier stages

HOW do you go about designing an amplifier? Reading the more advanced textbooks on the subject, you may get the impression that it is impossible to design an amplifier without a knowledge of higher mathematics. One book will start by giving a complete analysis of network theory, involving simultaneous equations in a large number of unknowns; another will approach the matter from the viewpoint of tube characteristics and give a general equation for the law relating plate current to plate voltage, applied grid voltage, and so on, in terms of a power series. The best this can do is tell you how to find the performance of an amplifier using some kind of ideal tube that never exists in practice. There are practical ways to design amplifiers, however, which are quite easy to follow, taken in simple stages; these are what will be discussed in this series.

In this article I shall discuss the design of a simple voltage-amplification stage, using various tube types. In subsequent articles we shall go on to other parts of the amplifier, including the phase splitter, the power-output stage, application of feedback, and so on.

Using Load Lines

The best way to examine the gain and operating conditions of a simple resistance-capacitance coupled stage, of the type shown in Fig. 1 for a triode tube, is to use the plate current-plate voltage characteristics published for the tube type. We can find out fairly easily all we want to know about the operation of the tube. The characteristics for a triode tube are shown in Fig. 2. *Each curve* represents the variations of plate current as plate voltage is changed, with a fixed value of applied grid voltage. Usually, the curves are plotted for uniform intervals of grid voltage, which makes them convenient to use for our purpose.

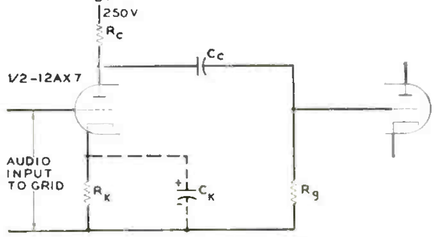


Fig. 1. Triode voltage-amplifier circuit.

Suppose the B+ (power supply) voltage is 250 and we use a plate-load resistor (Rc) of 100 K. Then, if the plate is short-circuited to ground, this plate resistor will pass 2.5 ma because it will have 250 volts across it. (The easy way to figure this is to remember that current in ma, multiplied by resistance in kilohms, gives voltage drop in volts.) If the plate potential is 100 volts, the drop across the resistor will be 150 volts so the current through it will be 1.5 ma. If the plate voltage is 150 volts the drop across the resistor will be 100 volts, so the current through the resistor will be just 1 ma.

If these plate-voltage and current values, determined by the value of resistors chosen, are joined together on the diagram of Fig. 2, we get a straight line passing through the B+ voltage of 250 (at zero current) and the plate current of 2.5 ma at zero plate voltage. This is the *load line* for this tube type with a plate-load resistor of 100 K and B+ of 250 volts.

How is the operating point fixed? We must select a suitable grid bias of such value that excursions away from this central point on the grid characteristics will give us suitable amplification. If we use a negative bias of 1 volt, the operating point will be at 142 volts on the plate with a current of 1.08 ma; this is where the —1-volt curve crosses the load line. For the moment let us assume that we have chosen this point.

Now, to get a suitable grid bias we must select a cathode resistor Rk of such value that with 1.08 ma passing through it the bias is 1 volt. With a 1-K resistor (assuming the value to be precise, which it is unlikely to be using 10% resistors) 1.08 ma will produce 1.08 volts for bias. This will get us as near to the operating point we have chosen as is practical. We could go right on to calculate the output of the stage for a given input, and then obtain a figure for stage gain, except that we have not yet taken into account the effect of the grid resistor (Rg) in the following stage.

Assume Rg to be 470 K. Then, so far as the audio signal is concerned, 470 K will be coupled in parallel with the 100-K plate load resistor. This is because the coupling capacitor is large

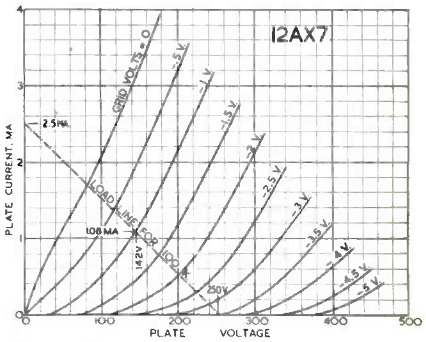


Fig. 2. Plate current-plate voltage characteristic curves for 12AX7 tube, used to evaluate circuit elements in Fig. 1.

enough that it does not allow the charge across it to change during the audio fluctuation, and accordingly the audio signal voltage at the top end of the grid resistor swings with the voltage at the plate of the stage we are considering.

So the effective plate load, for AC signals, is 100 K in parallel with 470 K, which works out to about 82 K. The 100-K resistor fixes the operating point, because the DC plate feed to the tube passes only through this resistor. But the dynamic load line, as it is called, representing the condition of the tube when an audio signal is being amplified, must be obtained by using a value of 82 K.

As a result we must draw a load line whose *slope* represents 82 K, passing through the operating point we have chosen at —1 volt grid bias and a plate voltage of 142. We use the same means of finding the new slope as for the old.

A conveniently close approximation would be taken by assuming 3 ma passing through 82 K (with the plate grounded). This will produce a voltage of 246 volts. We draw a couple of light markers at the 3-ma point on the plate-current scale and the 246-volt point on the voltage scale. These determine the new slope; with a parallel rule we draw a line with the same slope through the old operating point. This is shown in Fig. 3.

Now, let's see what this load line can tell us.

Gain. For a swing of grid voltage 1 volt either side of the bias point, which means from zero to —2, the plate voltage will swing (from its central point of 142) down to 85 and up to 195. This means that for 2 volts change on the grid there is a 110-volt change at the plate, or a gain of 55.

Distortion. The excursion of plate voltage for 1 volt positive on the grid — that is, from —1 to zero — is from 142 volts down to 85, a change of 57 volts. For an opposite grid swing of 1 volt, the plate voltage changes from 142 to 195, or 53 volts. We assume that the cathode resistor is bypassed by Ck, Fig. 1, so that cathode voltage is constant. This difference between plate swings on opposite grid swings shows there will be some second harmonic distortion in the output waveform. We can work out approximately how much by seeing how much the center of the waveform is offset. The midpoint between 85 volts and 195 volts is $\frac{85 + 195}{2}$, or 140 volts. This is 2 volts off center. The ratio of peak-to-peak amplitude of second harmonic to peak-to-peak amplitude of fundamental is $\frac{2}{195 - 85}$, or 1.8%

Effect of Cathode Bypass. If we do *not* bypass the 1-K cathode-bias resistor it will help to reduce harmonic distortion, but it will also reduce the calculated gain of 55. This can also be determined with the curves. At the quiescent operating point, the 1.08-ma normal current will produce a bias voltage drop of 1.08 volts across the 1-K cathode resistor. When the grid is made 2 volts negative with respect to the cathode, the plate current drops to 0.48 ma, which means that the voltage drop across the bias resistor decreases to 0.48 volt. This, in turn, means that the grid signal required to obtain —2 volts from grid to cathode will have to be 1.6 volts. In the opposite direction, the 1-volt grid-to-cathode swing changes the current from 1.08 ma to 1.78 ma, so the grid voltage swing must be 1.7 volts in this direction.

Now see what this does to the amplification. The negative-going grid excursion of 1.6 volts gets amplified to a

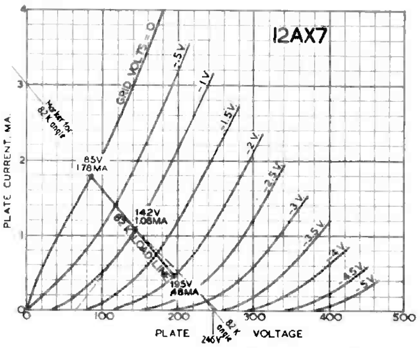


Fig. 3. Curves for 12AX7 with addition of a dynamic load line, showing change in slope caused by following stage grid resistor. Markers indicate the new slope.

plate excursion of 53 volts, a gain of 33.1. In the positive-going excursion the applied grid voltage swing of 1.7 volts gets amplified to 57 volts, a gain on this half of the applied waveform of 33.5. The *average* amplification throughout the whole applied waveform is now 33.3 in place of the original 55.

But we have an off-center effect of 0.2 part in 33.1 + 33.5, or 66.6, which means the second harmonic distortion has now been reduced to about $\frac{0.2}{66.6}$, or 0.3%

Applying a nice large capacitor, like 50 μfd, across the 1-K bias resistor will bring the gain up to the 55 figure by making the 1-volt swing either way from the bias point give the full plate swing. This happens since the cathode voltage does not have time to change by the values just calculated, because of the large charge stored in the capacitor.

Maximum Level. The voltage swing we have given is in each case that of the peak value. Ordinarily the instruments we use measure RMS values, which are 0.707 times the peak values. This means that the 1-volt peak input voltage, assuming the cathode to be bypassed, will be 0.707 volt RMS, while the RMS plate output voltage will be 55 × 0.707, or 39 volts.

Changing Circuit Values

Any variety of load lines can be drawn to represent different values of the plate coupling resistor and the following stage grid resistor. Following this method appropriate values of the cathode resistor can be evaluated, together with the available gain and voltage output, and an estimate of distortion. In general, with the 12AX7 curves we have just used, if a larger output swing or slightly greater gain is required, it can be obtained by using higher values of resistance all around.

For example, if you care to try the scheme yourself: Using a plate resistor Rc of 470 K, a grid resistor Rg of 1 megohm for the following stage, and a cathode resistor of 5.2 K, the grid bias is 1.5 volts, the voltage gain 61, and the peak-to-peak output voltage about 182, representing an RMS value of about 64 volts. The static plate voltage is 120, and the negative and positive voltage swings go to 20 and 202 volts respectively. This means that the 120 is approximately 9 volts off center, which, for 182 volts excursion, represents a distortion of about 5%.

It will be realized from these few figures that this tube is a fairly good voltage-amplifier triode because it is not very critical of the values with which it works, and 5% second harmonic distortion for a tube with that large an output swing is quite good. A stage requiring an output voltage that large will usually be operated in push-pull, so as to minimize the distortion.

But there is something further we want to find out about this stage before we pass on to other types of circuits. This is its frequency response.

Frequency Response

A typical resistance-coupled amplifying stage is maximally effective over a certain range of applied frequencies. At both ends of this range the stage gain decreases progressively for higher and lower frequencies.

The low-frequency rolloff is determined by the relation of the coupling capacitor reactance to the resistances in the circuit, which are the source resistance apparent at the plate of the stage we are considering, and the following stage grid resistor Rg. See Fig. 4. It will be realized that transposing the series elements, consisting of the source resistance and the coupling capacitor, will not affect the response of the arrangement in any way. When we arrange it as at Fig. 4B, it is quite evident that the frequency characteristics of the voltage appearing at the junction between the source resistance and Rg will be the same as the frequency characteristics at the top end of the source resistance, because this

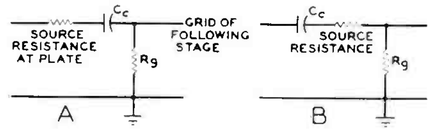


Fig. 4. Total resistances on both sides of the coupling capacitor determine the low frequency attenuation characteristics.

is merely a straightforward resistance potentiometer and cannot have any influence on frequency response.

Now we must calculate the source resistance at the plate. This involves the AC plate resistance of the tube, which we shall discuss further in a minute, but for the moment we consider it as an AC resistance between the plate of the tube and ground. If we use a very high B+ voltage and an extremely high plate-load resistor to go with it, of many megohms, the source resistance would simply be the plate resistance of the tube. Taking the tube manual value of 62.5 K for the 12AX7, if we put a load on the output of 62.5 K we should halve the available voltage swing.

In point of fact, the 100-K plate-load resistor that we used here also reduces the available voltage swing, because the tube has an amplification factor of 100 and we have only managed to get a gain of 55 from it. We have already shunted down the plate resistance with 100 K to B+, so any further loading effects applied to the plate, through the coupling capacitor, will be applied across the parallel combination consisting of the plate resistance and the 100-K plate load shunting it. The source resistance we consider in Fig. 4, then, is the parallel combination of the effective plate resistance with the plate-load resistor.

The value for plate resistance given in the tube manual, 62.5 K, is not taken for the operating conditions that we have assumed. To get an accurate result we should take the value of plate resistance at the chosen operating point. This, it will be recalled, was a bias of 1 volt, 142 volts on the plate, and 1.08 ma plate current. The plate resistance at this operating point is the controlling factor in frequency response. Actually, the response will change slightly throughout the waveform because the plate resistance varies up and down the load line, but we cannot take this into account too readily.

As far as the load line is concerned, it is unimportant that the characteristics are curved in the regions on either side of it. The only part of interest is the slope of the section immediately adjacent to it. So, drawing a tangent to the —1-volt grid curve where the load line crosses it, we get an ideal version of the tube characteristic. And using the same method we employed to construct the load line, we can calculate the value of the resistance. This tangent line passes through zero ma at 65 volts on the plate-current scale, and 1 ma at 140 volts on the plate-current scale, a change of 75 volts for 1 ma. The effective plate resistance is 75 K. The parallel combination of the 100-K plate load

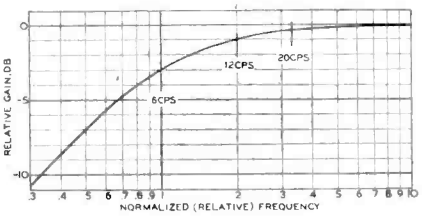


Fig. 5. Normalized low-frequency rolloff curve for one RC circuit. Point at which response is down 3 db is labeled "1", and attenuation in db can be determined for multiple and submultiple frequencies.

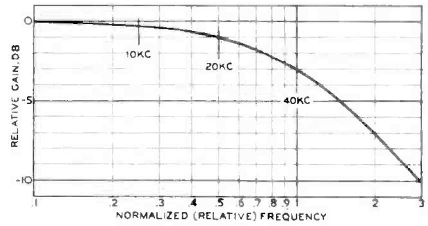


Fig. 6. Normalized high-frequency rolloff curve. It is used in same way as Fig. 5.

with a 75-K plate resistance gives a 43-K source resistance. In this case our careful calculation of plate resistance hasn't affected very much the total circuit resistance (Fig. 4) because 43 K has to be added to 470 K, which adds up to 513 K.

Use of a reactance chart shows that .05 μfd gives a reactance of about 500 K at a frequency of 6 cps. This means that the response would be 3 db down at 6 cps, about 1 db down at 12 cps, and 0.4 db down at 20 cps. These figures can be obtained from the normalized response curve, Fig. 5. Response at the low-frequency end seems to be adequate.

At the other end of the response curve the loss is due to stray capacitance shunting the total *parallel* impedance of the circuit. This is made up again of the plate resistance of the tube, the plate-load resistor, and the grid resistor. We have already calculated the effect of 100 K in parallel with 470 K — about 82 K. The combination of the 75-K plate resistance in parallel with the 82-K effective plate-load resistance is approximately 39 K.

If the following stage is another half of a 12AX7, which has a grid-to-cathode capacitance (Cgk) of 1.6 $\mu\mu$fd and a grid-to-plate capacitance (Cgp) of 1.7 $\mu\mu$fd, and if it is also working at a gain of 55, the Miller effect will raise the *effective* grid-to-ground capacitance markedly. Effective tube capacitance from grid to ground is Cgk + Cgp (1 + A), where A is stage gain. In this case it is 1.6 + 1.7 (1 + 55), or approximately 97 $\mu\mu$fd. If we allow 7 $\mu\mu$fd for various stray effects owing to the wiring, this makes a total of 104 $\mu\mu$fd. From a reactance chart 100 $\mu\mu$fd has a reactance of about 40 K at 40 Kc, which means the gain of this stage will be 3 db down at 40 Kc and about 1 db down at 20 Kc. Loss of gain can be determined from the normalized response curve, Fig. 6.

Now it is obvious why we need to know the plate resistance more accurately. It is the lowest resistance value of those that determine high-frequency rolloff, so that variation of the plate resistance will effect the resultant more than small variations of either of the other values in the circuit.

With a plate-load resistor of 470 K and a grid resistor of 1 megohm, the plate resistance is 85 K. The total parallel resistance works out to be about 67 K, which means that the 3-db down point will move from about 40 to 23 Kc.

Just how important these figures will turn out to be depends on the rest of the circuit and how we are designing the amplifier, which we shall come back to later in the series. We may be using feedback, in which case we can allow rolloff at frequencies within the audible range and straighten the whole thing up to some degree by feedback. Or we may be designing a stage which doesn't have feedback, or which is required to have an extremely good response in spite of the fact that feedback is being used. But before these factors can be discussed it is necessary to know more about the rest of the problem.

Tube Manual Tables

An alternative method of design, if it is not convenient to use the curves available, is to follow the tables published in tube manuals which give information similar to that shown (right) for the 12AX7. Such tables give a limited number of suitable operating conditions, but usually no information is given about distortion. Neither is it possible to see

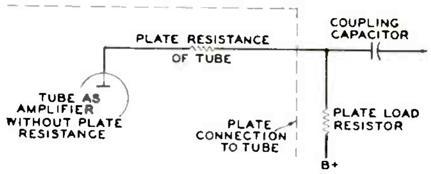


Fig. 7. Equivalent circuit used as basis for calculation of stage gain from plate resistance and the amplification factor.

what effect different bias values have on the total available swing, or on distortion.

In general, if the tube is being used for maximum gain, but is not required to have maximum voltage-handling capacity, a lower value of bias can be used to achieve better linearity and lower distortion by keeping well away from the curvature of the characteristic.

If not as much information as this is available, one has to be content with the tube parameters issued and the simple gain formula. The tube parameters given for the 12AX7 operating at 250 volts on the plate are: amplification factor, 100; plate resistance, 62.5 K; plate current, 1.2 ma for a bias of —2 volts. This is rather limited information. Moreover, the plate voltage here does not have the same meaning as the B+ voltage we have been working with. Here it means voltage *on the plate;* it does not allow for the drop in the plate-load resistor. The only way we can obtain a figure for a bias resistor is to divide the plate current into the bias voltage given, which in this case is 1.2

12AX7, Plate Supply 100 volts

Plate-load resistor:	0.10		0.22		0.47 MΩ	
Grid resistor, next stage:	0.22	0.47	0.22	0.47	0.47	1 MΩ
Cathode resistor:	4,700	4,800	7,000	7,400	12,000	13,000 Ω
Max. output voltage:	6	8	6	9	9	11 RMS
Voltage gain:	35	41	39	45	48	52

12AX7, Plate Supply 250 volts

Plate-load resistor:	0.10		0.22		0.47 MΩ	
Grid resistor, next stage:	0.22	0.47	0.22	0.47	0.47	1 MΩ
Cathode resistor:	1,500	1,700	2,200	2,800	4,300	5,200 Ω
Max. output voltage:	47	55	45	57	51	64 RMS
Voltage gain:	43	47	49	54	57	61

ma with a bias of 2 volts. This would require about 1,660 ohms. We shall need less than 2 volts, because the working plate voltage will be less than 250, so that we must guess that 1,200 or 1,500 ohms would be all right.

If the tube were operated from a *very* high B+ voltage, and fed through a large load resistor so as to give 250 volts on the plate, an amplification of 100 would be approached. But, since we are using a practical load of 82 K, we are effectively picking off a fraction of this 100 amplification factor across the 82 K, which is in series with the plate resistance quoted as 62.5 K. This is shown in Fig. 7. We obtain an actual amplification of $\frac{100 \times 82}{82 + 62.5} = 57$, which is a little optimistic compared to the figure we determined more accurately a while ago.

But it will be seen to give a figure which is pretty close, provided not much other information is needed. We cannot tell how much grid swing we can use, nor what plate swing we can get out of the stage, nor how much distortion to expect. But for some circumstances this much information could be very useful; it would enable us to compute, within limits, the gain of a stage.

So much for designing a voltage-gain amplifier around a triode. What about using a pentode, which gives considerably more gain?

Pentode Voltage Amplifiers

Fig. 8 shows the plate current-plate voltage characteristics for a 6BR7, a comparatively new, low-microphony pentode

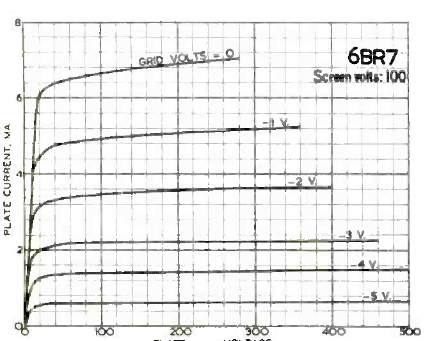


Fig. 8. Plate current-plate voltage characteristic curves for the 6BR7 pentode.

for audio work. It is possible to use these characteristics to set up the conditions and calculate the gain of the tube under any given operating conditions *provided the screen voltage is 100.* But the usual practice in this type of stage is to series-feed the screen, as shown in Fig. 9, rather than apply to it a stabilized 100 volts. This type of circuit gives a more stable gain figure and more consistent performance than one with a fixed screen voltage and automatic cathode bias, because the grid bias and the screen voltages tend to adjust to compensate one another.

But, since it is much more difficult to compute distortion figures from pentode characteristics for the reason that they do not produce primarily second harmonics, like triodes, there is not too much to be gained by going through the full procedure. The best way to design a simple pentode amplifier stage is to use tabular information given in tube manuals. The table below gives the published operating conditions for a 6BR7.

6BR7 Operating Characteristics

Plate voltage:	100	250	volts
Plate current:	2.0	2.1	ma
Screen voltage:	100	100	volts
Screen current:	0.7	0.6	ma
Grid voltage:	—3	—3	volts
Plate resistance:	1.5	2.3	MΩ
Mutual conductance:	1,100	1,250	μmhos

As Resistance-Coupled Amplifier

Plate & screen supply:	100	200	300	volts
Plate-load resistor:	0.25	0.25	0.25	MΩ
Screen series resistor:	1.0	1.0	1.2	MΩ
Cathode bias resistor:	2,500	1,500	1,200	Ω
Peak output voltage:	35	70	100	RMS
Voltage gain:	90	120	140	

Selecting the middle condition, for 200 volts plate and screen supply, we will get a voltage gain of 120 with a peak output (an RMS figure) of 70 volts. The thing we don't know about this is how much distortion is produced, but if we want an output well inside 70 volts we can be assured that this tube will give a reasonable waveform compared with other tubes.

In this case the plate resistance is quoted as 2.3 megohms for 250 volts on the plate, or 1.5 megohms for 100 volts on the plate. We shall be fairly safe in taking an intermediate figure of 2 megohms. In calculating response we have to use the effective resistance of the plate-load resistor, given as 250 K, in parallel with the plate resistance of 2 megohms, which comes out to about 230 K. If we use a compromise of 220 K for the plate load, which is a preferred resistor value, the resultant works out to 200 K for the parallel combination, and we can expect the gain to be about 105 with the 220-K load instead of the quoted 120.

In this case, however, the data do not include the value of the following grid resistor, as in some tables. It is to be assumed that the grid resistor will also shunt down the gain proportionately because, as with the triode, it is effectively in parallel with the plate load for AC signals. Thus, the gain with a 220-K load would be reduced to about 90 in practice. It may be better to use a plate-load resistor of 270 K in this case, with a 1 megohm resistor in the following grid. That way, for calculating frequency

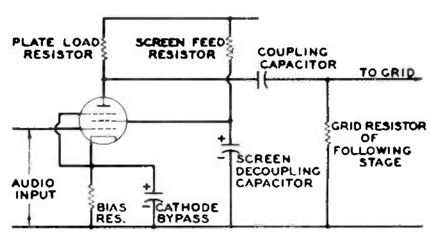


Fig. 9. Pentode voltage-amplifier stage.

response, the parallel combination of the plate resistance with the 270-K resistor works out to about 240 K, while the effective AC load for the tube is 1 megohm in parallel with 270 K, which is about 210 K and which should give a gain of just about 100.

Now for the frequency response. The value we must use with the coupling capacitor to determine low-frequency rolloff is the total series resistance, or 1 megohm in series with 240 K, a total of 1.24 megohms. For the high-frequency end the reactance of the stray capacitance has to be compared with the total parallel circuit resistance: 1 megohm in parallel with 270 K in parallel with 2 megohms plate resistance, which works out to be about 200 K.

If the following stage has an effective input capacitance of 100 μμfd the stage gain will be about down 3 db at 8 Kc, which is not too good. It may be better to reduce the value of the plate-load re-

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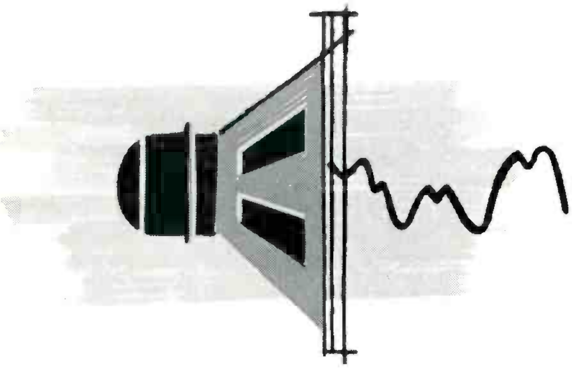


DISTORTION:
*Causes and Effects**

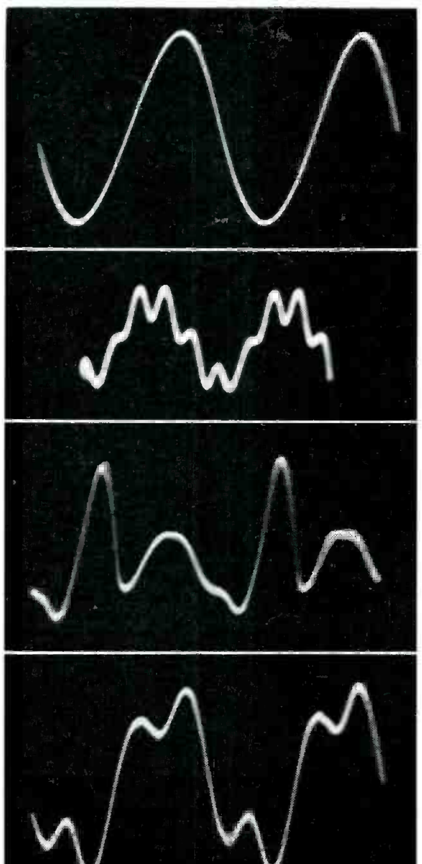
by RICHARD D. KELLER

THE conveyance with perfect realism of sounds and tonal structures, from concert and music halls through the dimensions of time and space and into our homes, has long been a goal of science. In recent years, that goal has been approached more closely by the introduction of frequency modulation broadcasting, and improved recording and reproduction techniques such as magnetic tape and vinylite microgroove records. These have all helped to reduce distortion and increase realism in sound reproduction. Reproduction equipment in the home often creates more distortion than the recording or broadcasting process. And because distortions deteriorate sound quality and actually create nervous tensions and irritations, it is important to recognize them and know how to minimize them.

What is Sound?

To understand distortion it is necessary to know what sound itself is. A quick glance at any physics textbook reveals that it is simply a physiological sensation produced in the auditory center of the brain. Its source is air waves of alternate compressions and rarefactions, pulsating at frequencies within the range of approximately 20 to 20,000 cycles per second, which cause vibration of the ear drums.

However, sound is usually not merely a simple compressive wave train. In fact, the only waveform consisting of one single frequency is a pure sine wave, such as that emitted from a tuning fork (see Fig. 1A). Whenever a wave differs from this pure form, it does so because additional frequencies are present. Nearly all the single-source sounds we hear are actually waves consisting of a sine wave fundamental plus many accompanying overtones or harmonics of the fundamental, all combined into one complex waveform and heard simultaneously. Note that individual harmonics are also pure sine waves, but are vibrating faster at integer multiples of the fundamental and are consequently higher in pitch. Also, they are vibrating at different intensities or loudnesses than is the fundamental. In fact, it is the respective proportions of these overtones which provide distinctiveness, timbre, and character to sounds emitted from various vibrating sources. Thus, a particular note played on a clarinet may be distinguished from the same note played on a trumpet or harmonica because of the different inherent vibratory characteristics of the instrument materials, as illustrated by the actual waveforms in Figs. 1B, 1C, and 1D. All instruments were playing middle A (440 cycles per second).



Figs. 1A through 1D, top to bottom: pure 440-cps sine wave; same tone played by clarinet, trumpet, and harmonica.

To illustrate this (Fig. 2) the harmonica tone in Fig. 1D has been separated into its frequency components to show that the vibrating brass reed has a 440-cps fundamental and a very prominent 1,320-cps third harmonic which is about half as intense as the fundamental vibration. Similarly, the sound of a clarinet or trumpet or any other waveform, no matter how complex, can be analyzed or broken down into its simple sine wave components.

In the recording process, the original sound air waves are converted by a microphone into electrical energy and then amplified many times, finally being impressed magnetically or engraved in some more-or-less permanent physical form. For playback purposes, the tiny impressions are reconverted into electric signals that are amplified and sent to a transducer (loudspeaker) which converts the electrical power back into vibratory air wave pulses similar to the original sounds.

How does distortion enter the picture? Well, a sound becomes distorted whenever its *original waveform* is altered. In the process above, there are obviously many places where such alterations of the original sound waves can occur. Distortions may arise mechanically in microphones, recording and pickup equipment, and loudspeakers, or electrically in poorly designed electronic or magnetic circuits.

Distortions from *any* of these causes will modify the resulting output waveform. Consequently, the entire system is no better than its weakest link, and at the center of the chain is the electronic amplifying circuitry discussed in this article. Now, amplification is ideally a process in which vibratory power is increased without disturbing the time sequence and pattern of its oscillations. Since this condition is never perfectly realized, the amplifier has a certain amount of measurable distortion, which, if excessive, deteriorates the audio quality and becomes annoying to the listener.

*This article appeared also in more technical form in the June 1954 issue of *Electrical Engineering*.

Electronic Distortions

There are 3 principal sources of electronic distortion: tube nonlinearity, circuit reactances, and random interference.

Tube nonlinearity is usually subdivided into 1) harmonic and 2) intermodulation distortion; *circuit reactances,* into 3) frequency, 4) phase, and 5) transient distortions; and *random interference,* into 6) noise and hum. These 6 effects will be discussed in order.

Tube Nonlinearity: Harmonic distortion arises when a sinusoidal (single-frequency) input voltage comes out of the amplifier as a non-sinusoidal wave (consequently, one containing new and added frequency components). This results from nonlinear tube dynamic characteristics. Intermodulation results from interactions between 2 or more input frequencies which give rise to sum and difference components or "beats" in the output. Like harmonic distortion, it is caused by nonlinear tube characteristics and *always* accompanies the latter when *more* than one input frequency is involved (which is nearly all the time, since music, as we have seen, consists of complex and not simple one-frequency waves).

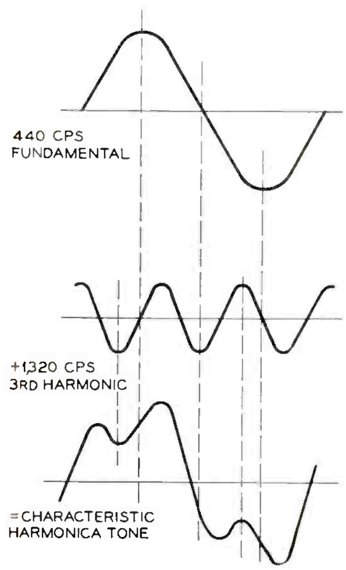


Fig. 2. How harmonica tone is made of fundamental and third harmonic.

Circuit Reactance: Frequency distortion occurs when the amplifier gain is not equal for all frequencies. It is caused by circuit or tube reactances which change with frequency. Phase distortion results when the relative phase differences between the various frequencies are not the same in the output as in the input. The same tube and circuit reactances which give rise to frequency distortion cause phase shifts. Transient distortion occurs when the amplifier is unable to handle the sudden stops and starts of percussion instruments and the like. Reactances are also the cause of this type of distortion.

Random Interference: Noise and hum result from a variety of causes such as electron thermal agitation in small-signal high-gain tubes, induced stray voltage, and poor power-supply filtering.

Now, let's look at each of these types of distortion individually. Measurement techniques and effects upon reproduced sounds will then be considered.

Harmonic Distortion

For an electronic tube to amplify sound properly, it should be operated only over the straightest portion of its dynamic transfer characteristic. This means merely that a given change in the grid voltage input (e_g) should always produce a particular given change in the plate current output (i_b). The input and output waveshapes, therefore, should always be identical, the only difference being the larger amplitude of the output, as shown in Fig. 3A.

If, on the other hand, operation takes place partly over the curved bottom portion (Fig. 3B), the output waveshape will obviously be blunted on the bottom half and no longer a perfect replica of the input. As discussed before, any repetitive wave which is not a perfect sine wave can be analyzed into a certain combination of pure sine waves all harmonically related to the basic or fundamental wave. Thus, the output waveform in Fig. 3B can be considered as a fundamental plus its second harmonic, as shown. Since the newly created second harmonic was not present in the pure input waveshape, it could be said that an unwanted harmonic has been added. Thus the name, "harmonic distortion".

Now, the area of operation of an electron tube is determined by the amount of negative grid bias or DC voltage applied between its cathode and grid. This is shown as e_{cc} in Figs. 3A and 3B. In Fig. 3A, the correct bias operates the tube over its most linear portion, whereas the excessive amount of bias in Fig. 3B has caused harmonic distortion. Similarly, too little bias would have distorted the top half of the output by moving the DC operating point too high on the dynamic transfer characteristic. Also, a larger input signal (e_g) in Fig. 3A would have caused distortion by driving both peaks into curved non-linear regions.

As for measurement, the ear is perhaps the most sensitive and critical of all means of measuring distortion. In the final analysis, it is the listener who subjectively determines whether a music amplifier system is tolerable.

Still, scientific standards are necessary for specifications and comparisons. The simplest test for the presence of distortion in a Class A amplifier stage can be made by applying a signal-level voltage to the input and inspecting the circuit for DC grid current and DC plate current fluctuation, both of which are abnormal conditions.

For more comprehensive measurements, a good-quality oscilloscope, sine- and square-wave generator, and an AC vacuum-tube voltmeter are required.

Harmonic distortion can be detected by noting the output waveform quality on an oscilloscope when a sine wave is being fed into the amplifier. The waves should appear as in Figure 3A. Distortion analyzers are used for more sensitive measurements, since it is difficult to detect less than 5% distortion visually on an oscilloscope. Such analyzers filter out the fundamental frequency and pass only the residual harmonics for measurement.

Nonlinear distortion is most likely to occur in the power-output stages, where the grid-voltage swing is considerable and the tubes are more easily driven into nonlinear regions. By designing the output in a balanced push-pull arrangement, even-harmonic distortion generated in one of the tubes is cancelled by its out-of-phase counterpart in the other. This eliminates one of the largest distortion components in most output stages.

Intermodulation Distortion

Since intermodulation arises from the same nonlinear tube operation as does

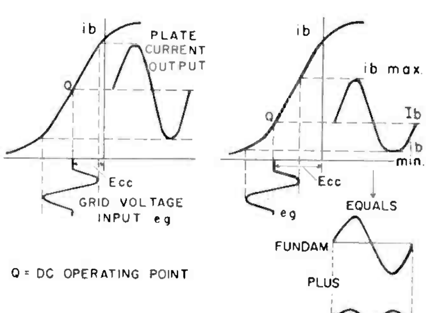


Fig. 3. Formation of spurious harmonics.

harmonic distortion, the same analysis may be used. We are now, however, dealing with the interaction between 2 input frequencies, so not only are the previous simple harmonic distortion components of each of the input waves present, but new sum-and-difference beat frequency products are also present in the output. Although these new frequencies may have small amplitude, they are quite noticeable as a dissonant background noise or chatter that bears no definite musical or harmonic relationship to the original tones and which appears to vary with the loudness of the signal. This is much more objectionable than harmonic products which are, at least, harmonically related to the input signal and not necessarily displeasing even though they do alter the original tonal quality to some degree.

Tests for intermodulation are made by mixing 2 widely separated frequencies, such as 60 and 3,000 cps, in a 4-to-1 amplitude ratio so that the high frequency modulates the lower one. This is shown as the undistorted input, Fig. 4B. The waveform is applied to the amplifier input, and is detected and measured as it issues from the output.

Normally the amplifier output is applied to a load resistor (R_L) of the speaker's rated impedance (usually 8 ohms) and size (5 watts or so) rather than to a normal loudspeaker load, in order to eliminate varying reflected speaker impedances and other external factors. Also, a vacuum-tube voltmeter is generally used for measuring the output amplitude, as shown in Fig. 4A, since it is more accurate and has less effect on the circuit being measured than does a simple voltmeter.

If nonlinearity is present, the high frequency will be unevenly amplified on the positive and negative extremes of the large low-frequency swing, as can be seen easily in Fig. 4B. If the low-frequency component is then filtered out in the intermodulation analyzer, the high frequency will appear as an amplitude-modulated signal with a modulating frequency of 60 or 120 cps, depending on whether the flattening occurs on one or both peaks of the low-frequency swing. Fig. 4E shows flattening only on the negative low-frequency peak.

The modulating signal is next detected in the analyzer to remove the high-frequency component, and the resultant AC voltage, Fig. 4F, actuates a meter which indicates the percentage of distortion. Obviously, there would be zero IM distortion if there were no modulation or flattening of the waveform and consequently no AC voltage to the meter. This is the case, of course, with the injected signal which is shown detected in Fig. 4D.

An increase in power level invariably leads to greater harmonic and IM distortion, since the power-output tubes are being driven further into their nonlinear regions. Thus, a typical home amplifier with less than 2% IM at 8 watts may measure nearly 8% at 14 watts, as shown in Fig. 4G.

Although the test equipment is rather expensive, IM testing is probably one of the most significant and revealing of amplifier distortion tests, since intermodulation background "hash" is directly related to listener fatigue and irritability. Results usually indicate 2 to 4 times as much intermodulation as harmonic distortion; consequently, manufacturers are often reluctant to publish IM figures. The better amplifiers today, however, will generate less than 4% IM distortion and the best amplifiers, less than 1% at rated power output.

Frequency Distortion

Frequency distortion occurs because circuit reactances, such as coupling capacitors and tube interelectrode capacitances, vary with frequency.

As an example, let the signal voltage from tube 1 be E_{in} (Fig. 5A). This is applied to tube 2 across R_g after passing through coupling capacitor C, inserted to keep DC off the second grid. C and R_g, however, form a voltage divider as shown in Fig. 5B, so part of the signal is dropped across X_c and the remainder appears across R_g.

At low frequencies, the capacitor shows a high reactance; only a small portion of the signal is left to appear across R_g and be amplified by tube 2. Consequently, the lower frequencies are reduced in intensity compared to the higher frequencies, for which the capacitive reactance is much less. If R_g and C are both made quite large, this effect does not become noticeable until the frequency is very low. The value of C is limited, however, by leakage, physical size, and cost; and that of R_g by the grid-ion bombardment of tube 2.

Frequency discrimination of this sort is used advantageously and purposely in tone-control circuits where circuit reactances are used to accentuate certain frequency ranges and subdue others. In the design of amplifiers, it is best to furnish a wide-range, flat response and then provide the listener with flexible tone controls to compensate for varying mechanical and acoustic conditions according to his own taste. In one sense, of course, this amounts to controlled distortion but, since it doesn't introduce new or strident sounds not present in the original performance, but rather merely blends those original sounds in different proportions, it can be used to increase rather than decrease listener enjoyment. And tone controls may be considered as a facility to restore balance lost elsewhere in the chain.

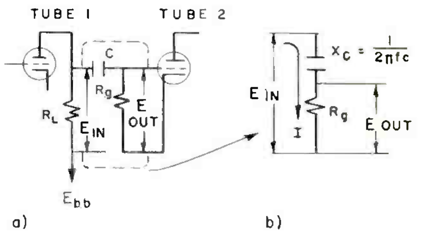


Fig. 5. Grid coupling network acts as a voltage divider at very low frequencies.

Frequency-response measurements can be made by injecting one sine wave frequency at a time into the amplifier and recording each respective input and output amplitude as read on the AC meter. (Tone controls, of course, should be bypassed or set in the "flat" position.) The resulting gain ratio is plotted point by point to give a curve which would ideally be a perfectly straight line, but which in reality droops at the frequency range extremes. A voltage divider equal to the approximate amplifier gain ratio is used at the input, as shown in Fig. 6, so that the VTVM need not be continually shifted in range when switched from input to output.

The input is read and adjusted before each output reading in order to compensate for the possibility that the audio-generator output and meter responses may not be constant. Also, the output leads may be connected across the vertical input of a cathode-ray oscilloscope, both for visual monitoring purposes and to assure that readings are taken below distortion levels where the

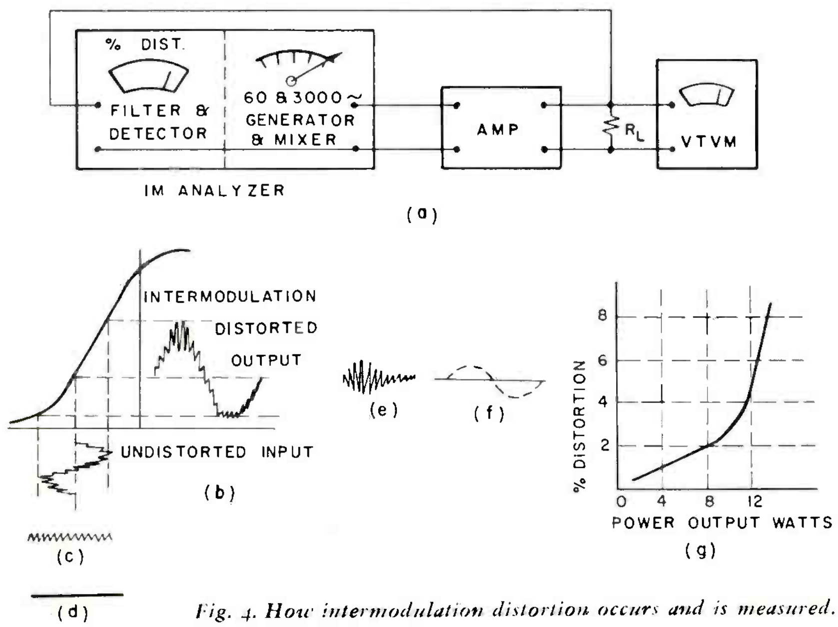


Fig. 4. How intermodulation distortion occurs and is measured.

peaks would begin to flatten.

By careful circuit design, response variation no greater than ±0.1 db over the entire audio range from 20 to 20,000 cps is achieved in many current amplifiers, whereas a range of 10 to 60,000 cps or more is available at room listening-levels from several truly outstanding amplifiers. The reason for desiring as wide a range as possible is to eliminate feedback-loop problems and to increase the "presence" effect discussed under transient distortions.

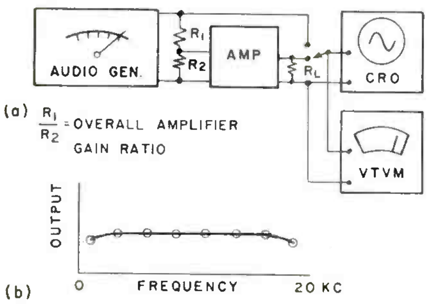


Fig. 6. Tests for frequency response.

Phase Distortion

Phase distortion has the same origin as frequency distortion — circuit reactances. If, in Fig. 5B, a frequency E_{in} is applied so that the capactive reactance X_c equals R_g, it can be shown that E_{out} will be reduced to 0.707 the value of E_i. In fact, not only will the amplitude of the voltage be reduced to 71% of its true relative value, but the wave will be shifted ahead 45° in phase relative to its original position (see Fig. 7).

At the higher frequencies, distributed wiring and tube interelectrode capacitances begin to enter the picture, acting as shunt paths to ground and shifting the phases of these frequencies.

Since it is apparently impossible to do away with these various circuit reactances, some phase distortion must always be present in a wide-range amplifier in some portion of its frequency range. Fortunately, the human ear seems to tolerate phase distortion* without any noticeably unpleasant effects. It is important, nonetheless, that the phase shift not approach 180° for any frequency at which the gain is unity or more, in an amplifier section around which feedback is applied, since this would actually produce positive or regenerative feedback at such frequencies and cause oscillations and instability.

*Some authorities believe that phase shift — particularly rapid phase shift — is more important to realism in reproduced sound than is generally realized. — ED.

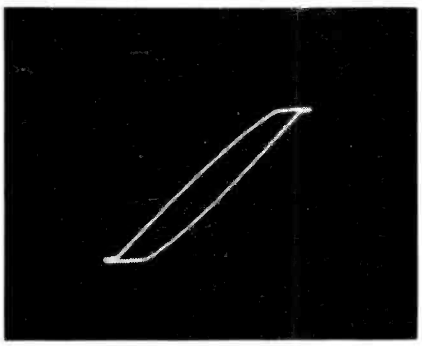


Fig. 9. Scope pattern with nonlinearity.

Phase shift can be measured by connecting the amplifier output to the vertical input terminals of an oscilloscope and the signal source to the horizontal scope input (as well as to the amplifier input through a voltage divider). This setup is diagrammed in Fig. 8. The signal source is, therefore, used as the sweep frequency base in place of the internal sawtooth sweep of the oscilloscope.

Since the vertical and sweep frequencies are the same, the pattern should be a stable one, and it will appear as a diagonal line, ellipse, or circle, depending upon the phase relationship of the 2 voltages (assuming that no nonlinear distortion is being introduced within the amplifier, in which case there may be kinking or flattening of the pattern, as shown in Fig. 9).

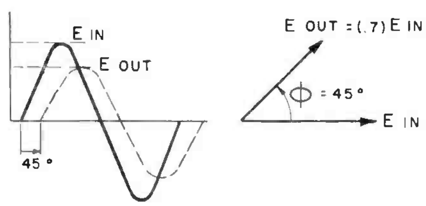


Fig. 7. Phase shift at a low frequency.

At any given frequency, the distance X from the center to the point at which the pattern crosses its vertical axis, divided by the distance Y from its horizontal axis to the greatest vertical amplitude (Fig. 8B), will equal sine phi, where phi is the approximate phase shift angle at that frequency. Phi may be computed for frequencies throughout the audio range in this manner and plotted as a frequency vs. phase-shift response curve, Fig. 8C, similar to the gain vs. frequency curve. Recent amplifiers have a phase shift no greater than ±15° over the entire 20 to 20,000-cps audio range.

Transient Distortion

When a piano note of a certain frequency is suddenly struck, signals of many times that frequency are momentarily generated by the initial steep wavefront. If these extremely high frequencies are attenuated by the amplifier, the reproduced sound will lack the sharpness or incisiveness — the illusive "presence" effect — of the original.

For this reason, an amplifier with a frequency limit of 5,000 cps or so is incapable of re-creating the full impact

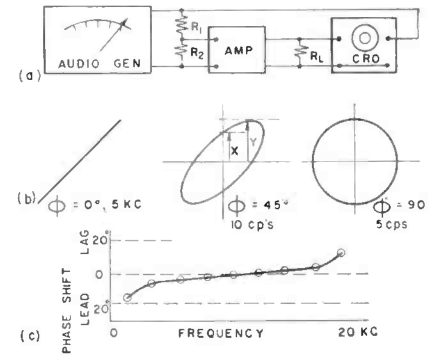


Fig. 8. A phase shift measurement setup.

of a live musical performance, even though the musical instruments themselves generate fundamental tones no higher than about 4,600 cps (the highest note on a piccolo). This is another reason for the trend toward amplifier bandpasses to 20,000 cps and more.

A particularly sharp transient is obviously generated by the leading edge of a square wave. An analysis shows that a square wave is actually composed of a fundamental, plus an infinite number of odd harmonics in decreasing proportions. Therefore, it actually requires many frequencies combined in proper phase to give what we see as a square wave with very sharply rising edges. If such a wave is applied to an amplifier, and internal-circuit and tube reactances limit its ability to amplify all these frequencies equally well, the resulting output waveform will no longer be sharply squared. This illustrates the close correlation between transient response and a wide, flat bandpass.

Transient response is best evaluated, then, with square-wave analysis. The oscillographs shown in Fig. 10 illustrate the true multi-frequency nature of square waves by showing the development of a square wave from a fundamental sine wave (Fig. 10A) of unity amplitude. The effect of the addition of a third harmonic of ⅓ amplitude, Fig. 10B, and a fifth harmonic of 1/5 amplitude, Fig. 10C, is clearly shown. Addition of still higher odd harmonics squares up the wave and straightens out the peaks until, finally, the clean square wave of Fig. 10D is formed. When the corners of the resultant wave are sharply squared, it can be assumed that at least the first 11 harmonics are present in proper strength. Square waves generally are obtained in practice, however, by over-amplifying a sine wave and then clipping off most of both peaks, or by means of multivibrator on-off switching circuits.

Now, if the leading edge of a 500-cps square wave input emerges from an amplifier sharply squared without any rounding, the amplifier may then be considered flat up to the 11th harmonic, or 5,500 cps. Similarly, sharp

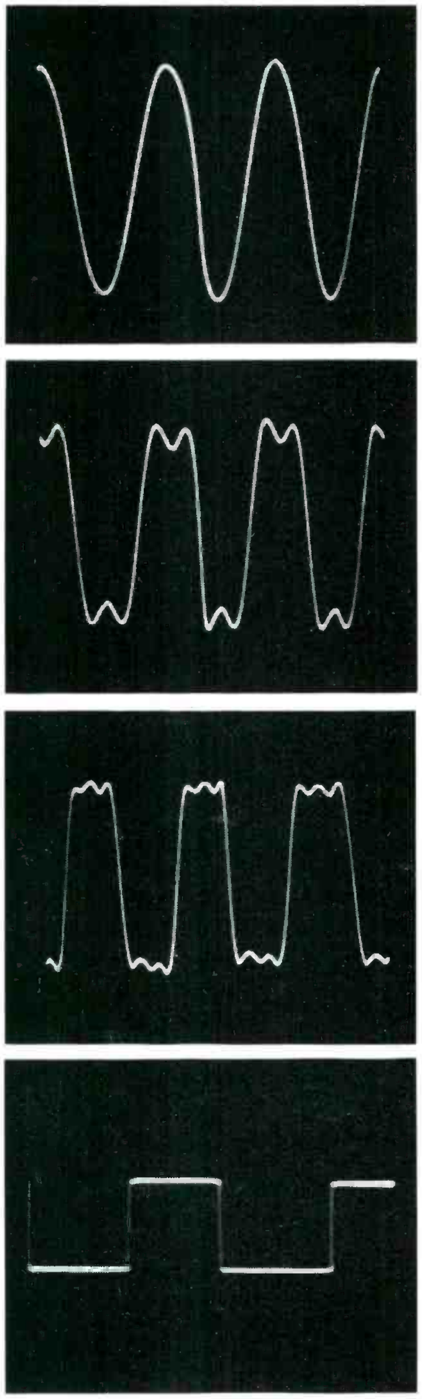


Fig. 10. Formation of a square wave from sine wave and its odd harmonics.

response to a square wave with a fundamental repetition rate of 2,000 cps will indicate an amplifier bandpass beyond 20,000 cps.

If, however, the leading edge is rounded, this is an indication that the higher multiple frequencies are attenuated, resulting in poor transient response. Also, if the top line of the output wave tilts upward, a lagging low-frequency phase shift is indicated; if the tilt is downward, the lower frequencies present in the square wave have been shifted forward in phase (and probably attenuated).

The extreme low frequencies may be checked by feeding in a square wave of only 20 to 50 cps. A pronounced concavity of the originally flat top and bottom peaks, as shown in Fig. 11, indicates attenuation of the fundamental frequency.

Tiny parasitic oscillations or ringings caused by distributed wiring reactances or high-frequency feedback troubles may also be detected visually by square-wave testing, appearing as damped ripples on the peaks. Accentuated high-frequency response results in differentiation or "spiking" of the leading edges of square waves.

It is obvious, then, that square-wave testing can be used to determine the approximate frequency and phase limitations of an amplifier, as well as its transient response, without resorting to tedious point-by-point frequency and phase readings. It also portrays actual dynamic operating characteristics more closely than do the static tests. A good amplifier should pass crisp, clean square waves of 20 or 2,000 cps, and an excellent unit should pass waves of 10 and 6,000-cps basic repetition rates.

Noise and Hum

Any output voltage present when the signal voltage is removed, if not due to feedback troubles, may be attributed to hum and noise components. Such extraneous noises become particularly noticeable and objectionable at full gain.

Noise originates primarily in low-level stages and manifests itself as a background hissing or rushing sound which appears as a general blur on the oscilloscope, since its frequency components are highly random.

Noise results from such things as leaky capacitors, "noisy" carbon resistors and potentiometers, microphonic tubes, and thermal causes, and may be reduced by using non-inductive, wire-wound resistors and special low-noise, non-microphonic preamplifier tubes.

Thermal noise is caused by the random motions of free electrons in conductors such as tubes and resistors. These random currents cause a potential difference across the terminals of the conductor whose amplitude is proportional to the temperature, resistance, and bandpass range of the conductor. The higher any of these factors, the higher the thermal noise.

Hum is caused by insufficient power-line filtering and by electromagnetic and electrostatic pickup, especially in low-level grid leads. It is important to keep these leads away from any AC sources and to use special care in grounding these early stages.

The origin of hum may be largely determined from its oscilloscope pattern. Straight capacitive pickup by high-impedance grid circuits and heater-cathode leakage shows as a slightly rough 60-cps wave, whereas inductive hum picked up by transformers is rich in harmonics (yet free from sharp peaks and spikes). Nearby flourescent light interference is characterized by spikes and hash superimposed over the 60-cps wave. Power-supply ripple due to insufficient filtering, however, is a very smooth 120-cps wave.

Careful amplifier construction and wiring techniques will eliminate most hum problems. Orientation of transformers for minimum inductive pickup, shielding and shock-mounting of extremely low-level tubes and circuits, adequate power-supply and AC-input line filtering, isolation of grid circuits from AC wiring, twisting the filament wiring and centertapping the filament winding to ground, and the use of a single bus-wire ground system attached to the chassis at only one point, are some of the precautions which are taken to give some current amplifiers the very low hum and noise level of 90 db below rated output. 70 db is usually considered adequate.

In conclusion then, consider this: A large symphony orchestra generates its fabric of tone from 100 or more sources of vibrating strings, reeds, diaphragms, and such, over an area of perhaps 500 sq. ft. This extremely complex wave pattern is picked from the air, fed to a recording system, and permanently inscribed.

A home sound system is then called upon to re-create the impact and beauty of the original performance by amplifying the inscribed signal thousands of times, applying the resultant electrical power to a small voice coil, and reproducing the original sound waves from a cone about a square foot in area.

With the innumerable sources of distortion possible in the journey of the sound patterns from concert hall to home, it is truly remarkable that this process may be accomplished today with such realistic fidelity. Furthermore, new developments such as stereophonic reproduction, with re-creation of the original spatial as well as structural patterns and relationships, have been introduced and hold promise for even greater realism in home sound reproduction in the future.

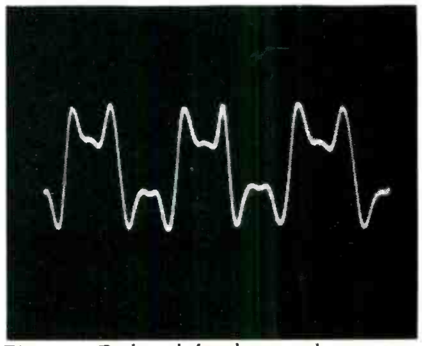


Fig. 11. Reduced fundamental response.

TESTING WITHOUT INSTRUMENTS

by Glen Southworth

YOU'VE just soldered the last connection of an amplifier kit, hooked up the loudspeaker and record changer and, while the family stands in breathless anticipation, you place Sibelius' Symphony No. 2 on the spindle in expectation of an auditory reward for your electronic labors. At this point you may find yourself joining an unfortunate minority that have had the distressing experience of listening to high fidelity needle talk and nothing else, lest it be the snide remarks of the audience. Or the expectant hush may be terminated by a wall-shaking hum, or a fire-siren howl, that quickly disperses listeners before the AC plug can be pulled from the wall.

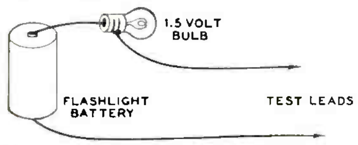


Fig. 1. Simple continuity tester.

If you are a sensitive individual you'll probably try to avoid dramatic situations like these. It can be done by 1) being reasonably lucky; or 2) pretesting the equipment in solitude and remedying any defects that may show up. If you fall in the first category, wonderful — my congratulations — just go on to the next article, please. But if you're a little shaky in your relations with good fortune and electronics, read on: this is for you.

Let's take as an example a Williamson amplifier kit. A number of them are on the market today that offer good performance, simple construction, and appreciable savings over a ready-made unit. Pre-punched chassis and pictorial diagrams make assembly a matter of a few hours work, even if you're just a beginner. The only real problem occurs when you've made some mistake in the wiring, or some defective tube or other component has accidentally found its way into the parts complement. These things can happen, too, in ready-made equipment; in either case, difficulties may not show up until you've had the system in operation for some time.

If you have troubles you might call in an expert, provided you can find one, and let him straighten things out. This is likely to cost money as well as prestige. Along with others, it is a good reason why you'd do well to acquire a little extra audio information and fix it yourself. Test instruments are undeniably helpful in home trouble shooting — as a matter of fact, you can go only so far without them. But there are many simple tests you can make without meters and, until you can get started on your home laboratory, they are often successful.

Localizing the Trouble

Glancing at the complicated mass of resistors, capacitors, and other parts, finding a trouble looks like an impossible task to the beginner; his first reaction might be to disconnect the speaker and record player and try out another amplifier chassis. This isn't really a bad idea, because it will tell you whether or not there is something wrong with the speaker or pickup. One of these, not the amplifier, might be defective.

Isolation of defects is greatly simplified once it is understood how the various components and circuits depend on

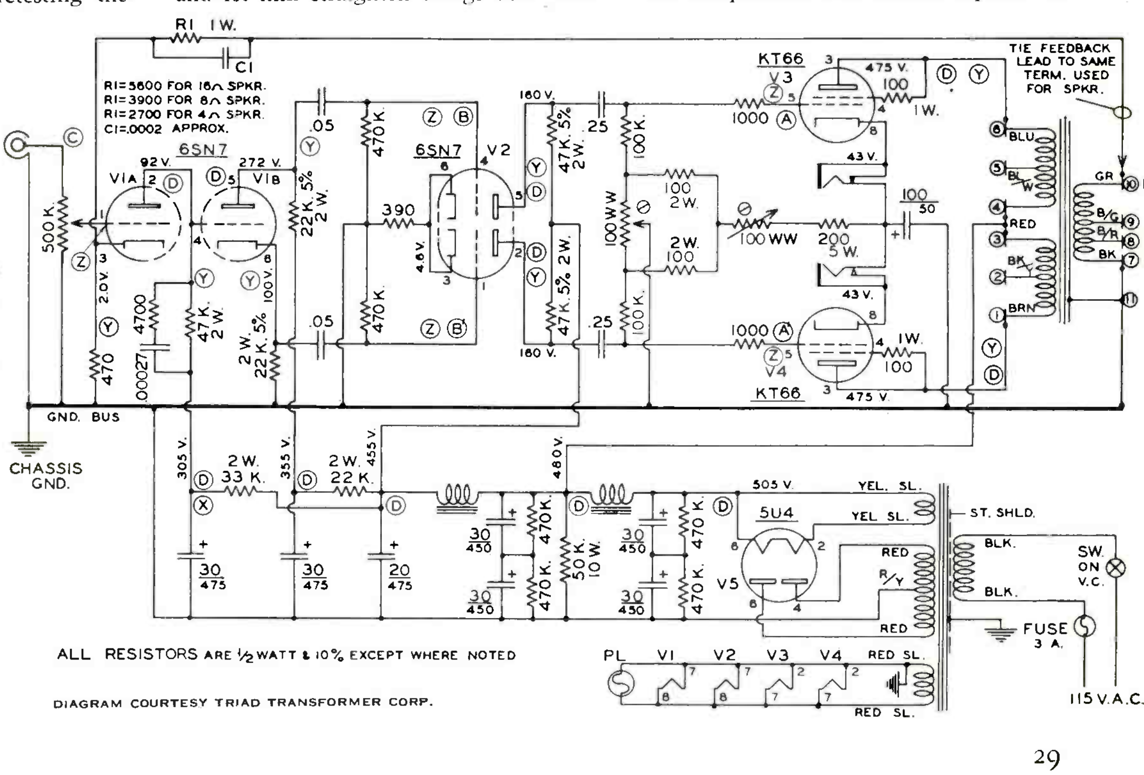


Fig. 2. Schematic diagram of a typical power amplifier, showing test points that are referred to in this article.

one another. The amplifier isn't going to function if it isn't plugged into a *live* receptacle, if the AC switch is off, if the fuse has been left out or has blown, or if the equipment is mis-wired so that the tube filaments don't light properly. Remember that the operation of the whole amplifier is dependent first of all on the power supply; be sure that all the tubes light up and that the power-supply rectifier tube isn't turning bright red because of a short circuit somewhere. If it is, remove power quickly.

Here is where a little ingenuity is helpful, since it is necessary to establish whether the trouble is in the circuit or in the tube. A simple continuity checker may be helpful; one can be made for about 20¢. Buy a fresh new flashlight battery and a bulb to match it. Wire them in series as shown in Fig. 1, leaving the circuit open with 2 fairly long leads. Attach one lead to the amplifier chassis, Fig. 2, and (with the AC power off) touch the other lead to all the high-voltage points marked D in succession. If the bulb glows even faintly at any of the test points it indicates that the resistance from that point to ground is too low; there is a shorted connection or a faulty component somewhere nearby. Check to see that no wires or component leads are touching that shouldn't, and that a blob of solder hasn't made an unwanted ground connection. If the bulb doesn't light at all you must assume (temporarily, anyway) that the trouble is in the rectifier tube. Take it out to be checked by a radio service shop and, while you're at it, take along the other tubes and have them tested too.

Suppose the amplifier lights up properly and nothing seems to be burning up or arcing over, but nothing comes from the speaker. The first step then is to disconnect the speaker leads from the amplifier and touch them across the terminals of the battery. If a click, pop, or scratch results, then you may assume that the speaker is working. Connect the speaker to the amplifier again, turn the amplifier chassis upside down, and locate the socket connection for the control grid of one of the output tubes. (Usually this is pin No. 5, counting clockwise from the keyway). Clip one terminal of the flashlight battery to the chassis and momentarily brush the other against the grid terminal. This should result in noise from the loudspeaker if that half of the output stage and the power supply are both working correctly. An equally loud pop should be heard when the other output tube's grid is touched similarly. An alternative method is simply to pull the output tubes out of the socket one at a time and listen for a pop from the speaker. This is known as the circuit-disturbance method of trouble shooting, while use of the flashlight battery is called signal injection. Both are effective, and easy to use, requiring only the employment of the ear and a bit of simple reasoning.

To go back a bit, let's take a look at Fig. 3, which shows a block diagram of an audio system and the manner in which a signal passes through it. Obviously, the signal can be interrupted at any point along the way by circuit failure, while if a signal is injected into the system further on something will be heard from the loudspeaker. Fig. 2 is the same amplifier in schematic form, a typical Williamson amplifier circuit. Points A and A′ are the control grids of the output tubes; momentary connection of the flashlight battery to either of these points should cause a pop in the loudspeaker. B and B′ are the control grids of the driver tubes, through which the signal must pass to reach the output stage. A much louder click or pop should be heard from the speaker on touching the battery to either of these points if this part of the circuit is working. Note, however, that there is a slight possibility of damaging the speaker when making some of these tests, especially if it has a sensitive tweeter unit. I recommend the temporary use of a cheap replacement speaker for these tests. Also be sure that you have *some* speaker connected at all times when you are operating the amplifier.

If output is obtained from applying the battery to B and B′, then you should go back to C, the grid of the input stage. If this stage and the inverter are working C will probably be sensitive enough that you can check it by merely touching it with a finger, which should produce hum in the loudspeaker. In the event that you still have output when applying a signal to C, you'd better start checking your input connections, preamplifier, or pickup, because the amplifier is in some kind of working order, anyway.

The point of all of the foregoing is that by applying the signal from stage to stage you can isolate the part of the system that isn't working. For example, if you obtain output with a signal applied to point B, and nothing from point C, it's pretty well settled that the input stages are at fault. In turn, this means that all you have to worry about is why one tube and a few resistors and capacitors don't work as they should.

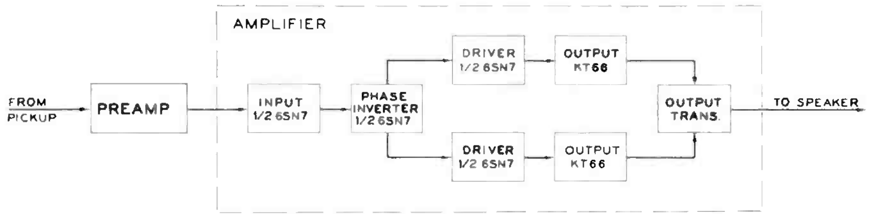


Fig. 3. Block diagram of an amplifying system, showing the function of each stage.

Once you've localized the difficulty, you can turn off the power and check the wiring for mistakes. The continuity checker is often helpful here too — the bulb should light when the test leads are touched to 2 points that should be interconnected, and it should *not* light when the test leads are touched to 2 points that ought to have appreciable resistance between them. For instance, there should not be even the faintest glow when the leads are connected to any grid and ground, or across a coupling capacitor. If this doesn't locate the trouble, you're ready to begin substituting components, beginning with the tube, then capacitors, and finally resistors. Sometimes you can find the trouble simply by connecting another resistor or capacitor across the suspected one, if the bad part happens to be open-circuited. On the other hand, if you have an internally shorted component, you'll have to remove the old part before the replacement will work.

At this point circuit-disturbance testing with the power on may simplify location of faults even more, and you may graduate to the ranks of the screw driver mechanics through use of that wonderful test instrument, the screw driver. This technique is based on the fact that certain voltages should be present in the amplifier circuits, and when these voltages are interrupted by short-circuiting them to ground, a signal may be heard from the speaker or — in the case of high-voltage B+ — sparks can be seen.

The screw driver has the advantage that it's a lot harder to burn out than a volt-ohm-milliammeter and costs a lot less, too. However, be careful what part of the circuit you use it in, and keep it away from the 110-volt AC wiring, the rectifier tube, the filament wiring, and the power transformer, because at these points there is sufficient power to do damage if a short circuit occurs. Also, some painful shocks can result if you get your fingers where they don't belong.

Fig. 3 shows a number of "test points" for the screw driver. Momentarily shorting point X to the chassis should cause a flash, indicating that the power-supply system is working properly, and that B+ is being supplied to the input stages. Temporarily shorting those points marked Y to the chassis should cause loud noises in the speaker and sparks, while shorting those marked Z should cause nothing to happen (except a slight pop in the speaker, maybe) unless a faulty capacitor, tube, or wiring causes voltage to be where it shouldn't. It's important to note that these shorts should be just quick jabs with the screw driver and should not occur for more than a second, or some of the circuit components may become overloaded. This is the primary disadvantage of the screw-driver technique: it *can* cause more damage than a sensitive multi-tester, which scarcely affects the circuits being tested.

As an example of this technique's application, let's assume that in the Williamson circuit the input stage failed to pass a signal. Shorting the plate of the input tube to ground causes a loud pop from the speaker, indicating that B+ is available, the plate resistor isn't open, and the phase inverter half of the tube is working. Next, shorting the cathode of the input tube also causes a pop, showing that the tube is conducting and the probable trouble is a short in the input grid circuit.

Actually, simply pulling the tube out of the socket would be a nearly equivalent test for this trouble. But if no sound were heard it might mean that the cathode circuit was open when the previous test with the screw driver was made and this was the actual difficulty, rather than a shorted grid. Some of this ambiguity can be reduced by feeding a signal into the input jack of the amplifier at the same time you make disturbance tests or are substituting parts. This will let you know the instant the amplifier starts working. If an open cathode circuit were the trouble you'd hear the signal over the speaker when you shorted the cathode to ground.

Up to now I've made a number of assumptions, the primary one being that your amplifier didn't work at all to begin with. However, this is only one case out of many, and you not only want the equipment to work, but to work well. Once the system is in operation you might be faced with any or all of the following problems: hum or other noise, distortion, inadequate volume, howling sounds. These difficulties can be isolated by much the same means that were used before to find out what part of the amplifier wasn't working.

For example, let's suppose that there is serious hum coming from the loudspeaker when the equipment is turned on. First, pull out the driver tube; if the hum remains, it is originating somewhere in the output stage. Then the output tubes can be removed and replaced, one at a time, to see if this localizes the trouble even more. On the other hand, if removing the driver tube stops the hum, we may assume that it is originating either in this stage or a previous one, so the next logical step is to replace the driver tube and remove the phase-inverter tube which precedes it. If hum is once again present, it's likely to originate in the driver stage; but if things are quiet, then the trouble is in the phase inverter or someplace before. In fact, the trouble may be outside of the amplifier, and result from the fact that you got your preamplifier connections on wrong.

While pulling out tubes is probably the simplest way to localize hum problems, you may prefer to go under the chassis and interrupt the signal path via the screw-driver method. This is done by first shorting the grids of the output stage, then those of the driver stage, phase inverter, and finally the input stage. It should be noted that you may have to short both output-tube grids simultaneously in order to kill the signal. This is because of the push-pull circuit design; the same thing applies to the driver grids, since shorting only one will still allow the signal to be fed through on the other.

Noise can be located in a manner almost identical to that used in finding hum, as they are both spurious signals originating in some part of the circuit. Probable sources of hum are faulty wiring[1], bad electrolytic filter capacitors, and gassy or shorted tubes. Probable sources of noise, hissing, frying, popping, and so on are tubes, resistors, poor connections, and oscillations[2]. Parts substitution is the easiest way to cure most noises, once it has been determined that they do not result from poorly soldered joints. Tubes, of course, are most suspect. High-frequency oscillation usually must be treated by a service organization, although it is often cured by connecting a 50-ohm resistor permanently across the output terminals or by using matched parts in the push-pull stages.

A third aurally unpleasant condition may arise from *audible* whistles or howls caused by positive feedback. The most likely source of trouble to the amateur is the fact that, with most modern amplifier designs, the reversal of 2 leads on the output transformer can cause ear-splitting sounds to issue from the speaker the moment that the equipment warms up. The first step is to make sure that all the leads from the output transformer are going where they are supposed to. Next, have all the tubes checked, and try interchanging those of the same kind. Then try replacing the feedback resistor and the input-stage cathode resistor; either could cause oscillation if radically off value. Finally, try replacing components in the phase-inverter, driver, or output stages — or, probably less expensive in the long run, get your amplifier to a competent service outfit or to the manufacturer. Again: things to look for with oscillation troubles are wrong wiring, bad tubes, and faulty or wrong-value components.

Howl resulting from microphonic tubes is easily found by tapping gently on the tube envelopes. The closer the tube is to the input stage of the amplifier the more free from mechanical vibration it must be; sometimes you can save time and money by trading tubes of the same type around in the amplifier. A slightly microphonic tube won't be nearly as objectionable in the driver stages as it would be in the input stage of the amplifier.

High-pitched whistling or even inaudible oscillations may occur when input and output leads to the amplifier are too close together, particularly if long speaker lines are used and a high-impedance, low-output magnetic cartridge is employed. So be careful of the manner in which you interconnect your equipment, and be sure to use shielded wire between the pickup and preamplifier and between the preamplifier and power amplifier.

Distortion is a subject that I hate to bring up, because it can be rather a heartbreaking matter. Chances are good that you're going to encounter it, however, either in your own pet project or in that ready-made beauty, after you've been running it 17 hours a day for the last year or so. Anyway, if you don't like the sounds that finally are coming from the loudspeaker, here are some of the things you can try before calling in an expert.

First, cross-check the different parts of your system to ascertain that it isn't something besides the amplifier at fault. For example, record reproduction may be terrific but the sound from the FM tuner miserable: this is a sure sign that the tuner has defected, not the amplifier. Be certain that the loudspeaker is connected to the proper output tap on the amplifier and, if possible, substitute another speaker in order to be sure that the first one isn't malfunctioning in some way.

Once you've pretty well decided that the amplifier is producing the distortion, the classic procedure is to change the tubes. One might be shorted, gassy, or just plain weak. If you still don't have any luck, turn the chassis upside down and start looking once again. We'll assume that you have found all of the wiring mistakes by now, but how about component values? Putting in a 4.7-K resistor in place of a 470-K unit could have a lot to do with your troubles. Or perhaps you slipped a little and put in a .001-μfd capacitor where the circuit called for a 0.1-μfd value, thus upsetting the push-pull balance.

Next to tubes, capacitors seem to be the components most likely to fail in

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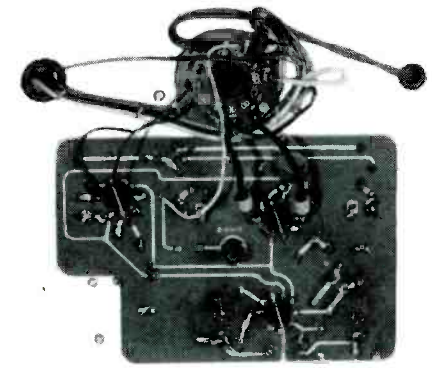
[1]See "Chassis Layout and Wiring", by Glen Southworth, AUDIOCRAFT, December 1955.
[2]See "Sound Servicing", by Irving M. Fried, AUDIOCRAFT, January 1956.



Construction of Modern Kits

by CHARLES FOWLER

IF the editor of AUDIOCRAFT were more free with his editorial space, he might grant my request that this entire page contain only the words (in bold capitals) IT WORKED! ! And, as everyone who has ever assembled a kit knows only too well, even such a gesture of editorial sympathy would convey only about 1% of the emotional thrill we kit-builders experience when we finally screw up our courage to turn on the AC switch for the first time . . . and the kit then does whatever it is supposed to do. The moment of do-it-or-die courage and the aftermath of thrill (or shattering disappointment) are great enough even for the simplest kit; they are almost unbearable when the kit involves, as did the one I have just completed:

229 parts, excluding nuts, bolts, and washers.

278 operations to be check-marked as done.

379 soldered joints.

After all this work, I feel as if I had wired the latest IBM electronic brain, but actually I have put together a Heath Laboratory Oscilloscope, their Model 0-10 which — if I may say so* — is a beauty. Of course, I may be prejudiced; I wired it, and it worked, the first time!

Well, enough of my excitement: if you've assembled and wired a piece of equipment you'll understand; if you haven't, there is something in store for you. This article is really about some of the lessons learned from my latest assembly project, and about the temptations carefully shunned. First, though, is the project of building this 'scope worth undertaking? I'd say yes, definitely. An oscilloscope is certainly not the first piece of equipment that should be acquired for an audio amateur's test bench; things like multimeters and sensitive AC voltmeters should come first. Nor would it be wise to choose something as difficult as this for a first project; the fundamentals of wiring, soldering, and even of following detailed instructions should be learned by putting together something simpler. I think, however, that it is possible for anyone with some soldering experience and a willingness to follow step-by-step instructions with painstaking caution to come out at the right end of this job.

From the strictly financial point of view, assembling and wiring your own equipment is nearly always advantageous. When an experienced technician assembles a piece of electronic equipment he knows all the danger spots, and can be depended on to deliver a product which will give optimum performance. That is the advantage of buying an already-assembled piece of equipment. But if a kit is well designed and the instructions are adequate and accurate, and if the builder does exactly as instructed, then the end results should be comparable — and the money which a manufacturer would have to pay to his assembly line technicians can go into your pocket (or, rather, can stay in your pocket). In the case of this oscilloscope, the Heath 0-10 kit costs $69.50; comparable quality and performance in a completely assembled 'scope might cost 3 to 5 times as much. But the ultimate responsibility for its performance rests on the constructor, so it pays to be extremely careful and painstaking. Let me illustrate this warning by telling about my own experience. About a year ago I built a simpler Heath oscilloscope kit. It worked too . . . for the first 15 minutes. Then it quit; the trace jumped off the screen. As usual, it was late at night, so I turned off the AC switch and came back to the job the next evening. Presto! It worked . . . for 15 minutes. To make a long and agonizing story short, I spent hours trying to locate the trouble. It was intermittent; the 'scope would work for a while, then quit, then start again. The trouble was finally traced to a bad soldering job. I never did discover why it would hold for a while and then break apart (maybe heat had something to do with it) but, if it took 15 hours to build the unit, it took 50 to make it work. And when you have 379 soldering jobs ahead of you, as on the 0-10, you had better be careful as you go!

There's no doubt about the usefulness of an oscilloscope as part of a complete audio test or experimenting setup. It's almost a "must" piece of equipment, to enable you to see just what is going on inside electronic circuits. And I'd say buy or build the best one you can afford. The 0-10 has a 5-inch tube and exceptionally fine frequency response, which is important when you are worrying about oscillations, ringing, and response at ultrasonic frequencies.

Circuit Boards

Construction of audio and test equipment, both on assembly lines and in the home, is being made much simpler these days by the use of printed circuit

*That you may say. — ED.

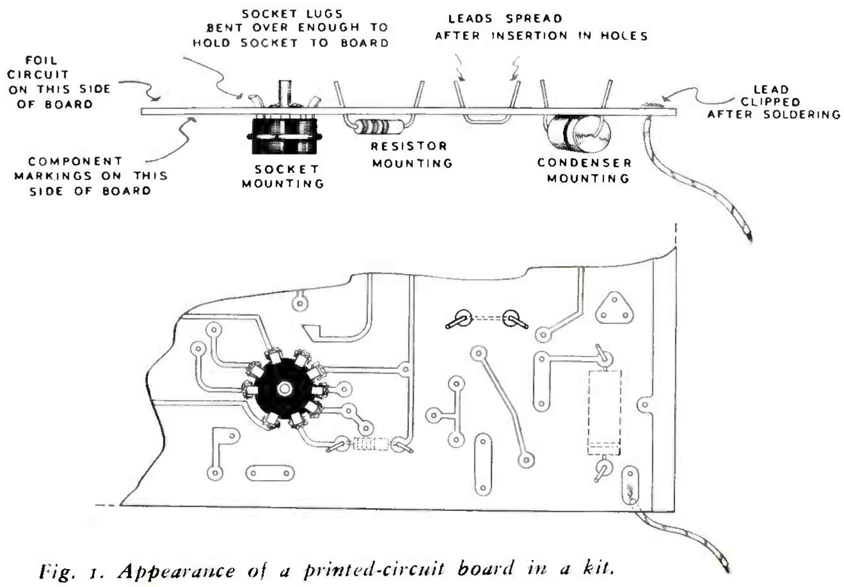


Fig. 1. Appearance of a printed-circuit board in a kit.

boards. These are thin sheets of phenolic material with holes punched through for wires and even tube sockets. Components are mounted on one side, with their leads sticking through to the other. The leads are connected by very thin lines of silvered copper, about 1/8-inch wide, which replace conventional point-to-point wiring. All you have to do is put the resistors, capacitors, and other components through the right holes and solder their leads to the etched lines. Then you snip off the excess wires. The drawings from the instruction manual, Fig. 1, will give you an idea of how a printed circuit board looks.

This technique saves lots of time, makes a neat job, and assures optimum positioning of critical components and wires. But it brings with it some hazards not encountered in old-fashioned wiring, the major one being soldering. The instruction manual warns that if you use a soldering gun you must avoid excessive heat. I would like to amend that to read: don't use a soldering gun under any circumstances. Guns are too hard to handle delicately and accurately. A soldering pencil is fine; I used a 25-watt miniature iron. The problem is that the printed lines are delicate; if too much heat is applied, you may melt the phenolic base and then the wafer-thin strip of silvered copper may melt loose, or the phenolic may flow over the top of the printed line and form a most effective layer of insulation.

There are 3 types of components that must be soldered to a board (at least on the 0-10, and these suggestions will apply to other pieces of equipment): leads from resistors, capacitors, and other components; wires from other sections of the equipment; and tube socket lugs. I had good luck with the wires and leads if I applied the tip of the iron to one side of the wire about 1/16 in. above the circuit board. When the wire was hot enough to melt solder held to the side opposite the iron, I slid the iron down the wire until the tip of the iron touched the printed circuit line and held it there gently for about 2 seconds, while rotating the tip around the wire so it would heat the entire circle of printed circuit which surrounds each hole. This way, a minimum of heat was applied to the board itself; the rotating procedure spread the solder neatly around the wire.

Be very, very careful to use as little solder as possible. If you use too much, the blob will jump from one circuit line to an adjacent one, and that is just about fatal. It creates, of course, a short circuit, and then you have the job of getting the solder off and the 2 printed circuit lines separated, all without melting the phenolic base.

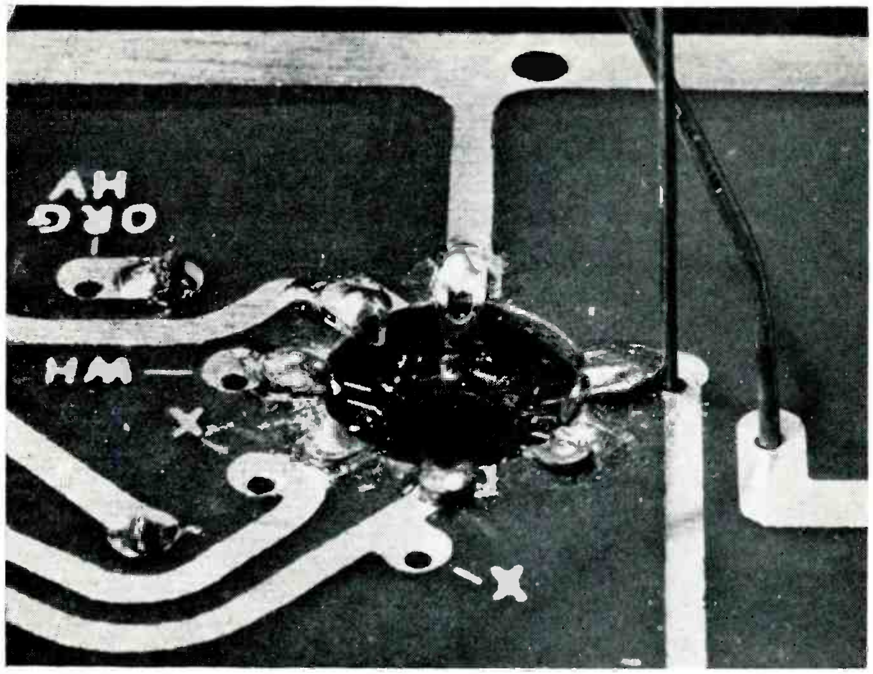


Fig. 2. Tube socket connections on PC board. Some solder joints are good, some bad.

Nasty job; it's better to use too little solder and add a smidgin if necessary.

Incidentally, standard radio solders seem to have too much resin in them; the melted resin often flows from one circuit line to the next. It's advisable to scrape it off, gently, with a slightly dull knife, after it has cooled and hardened. It flakes off easily.

Tube sockets are another problem. The socket, and the lugs, slip through a hole in the circuit board. The lugs are then bent over, gently but firmly, and soldered to the proper printed circuit lines. Once the lugs are soldered, the sockets are held in place quite firmly. I found the lug-soldering job trickier than wire soldering; after soldering 66 lugs my technique improved! (The 2 circuit boards in the 0-10 kit hold 8 tubes and 76 other components.) The best method seems to be to hold the tip of the iron just about at the bend in the lug, where it has been folded over, until the lug gets hot enough to melt solder held at its extreme end. There is about 3/32 in. between the soldering iron tip and the end of the lug. When the solder starts to melt, the iron should be slid along toward the end of the lug and down onto the printed circuit line. Do not press down on the lug to hold it against the circuit board! This is one of those temptations to be shunned rigorously. It seems like the logical thing to do, but here's the trouble: no matter how you try, you cannot always bend down the unsoldered lugs so that they stay pressed firmly against the circuit board. If you hold them down with your soldering iron, they'll snap back up when you remove the iron to let the solder cool. The motion may be so slight that you cannot see it, but snap back up they will — and you may have a bad solder joint.

That, as a matter of fact, was what caused my trouble with the 'scope made a year or so ago. One of the lugs on a tube socket looked perfect, but it wasn't.

While we're talking about tube sockets in these circuit boards, another word of warning: when you get around, much later, to inserting the tubes, do so as gently as possible and hold your thumb or finger firmly underneath the socket. Miniature tubes require a surprising amount of force to seat in their sockets; I pushed one in too hard and snapped a printed circuit line (which was soundly soldered to a lug) off the phenolic base. Fortunately, I heard the snap and suspected what had happened, so I soldered the circuit line back together again. The main precaution seems to be to press firmly against the bottom of the socket when you push the tubes in; then, after the tubes are in, wiggle them very slightly while watching the lugs to make sure all the joints are still tight.

The problem of soldering tube-socket lugs is illustrated in Fig. 2. The lug nearest the bottom (at 6 o'clock) is properly soldered. You can see that the lug was not forced down with the soldering iron and, therefore, the solder has flowed under it. The number 2 lug, at about 4 o'clock, caught, but not too well. The number 1 lug, at 3 o'clock, is a mess! In the first place, the dark shadow under it indicates an absence of solder; furthermore, as is obvious, the solder didn't flow around the wire at the end of the lug.

So examine all your soldered joints on a circuit board very carefully, even using a magnifying glass if necessary.

Since each wire has to be snipped off, check its joint before snipping. If you have missed a connection, leave the wire standing to indicate "no solder". The wire at the right in Fig. 2 is typical.

Incidentally, the holes in the circuit board in the illustration are for wires still to be run. Some soldering is done

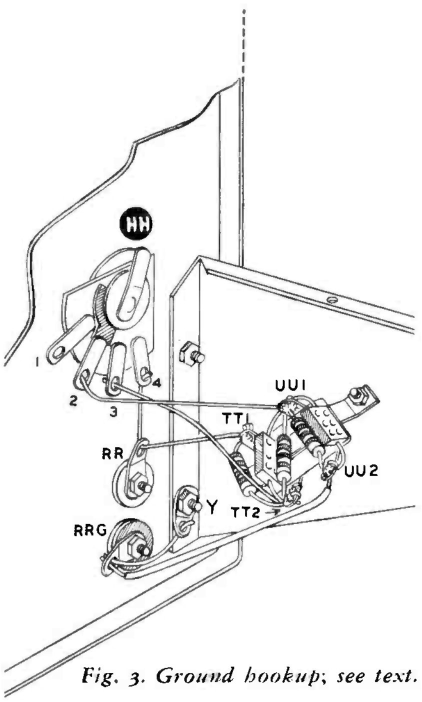


Fig. 3. Ground hookup; see text.

after the circuit boards have been mounted; when such connections have to be made, hold the instrument or whatever you are building in such a way that the circuit board is levelly horizontal. Soldering will be easier, and there'll be less danger of the solder running downhill and skipping from one printed circuit line to the next.

Another hint in connection with circuit boards: when you start pushing resistor and other component leads through them, prop up the board on a couple of pieces of wood or something so that the leads can dangle beneath. There's less danger that way of a component being pushed out of the board again. If you try to hold everything in your hands or work with the board on the bench, it gets too complicated.

Still one more hint for soldering circuit boards. Note the empty hole to the left of lug number 5 in Fig. 2, the one at about 9 o'clock. If you're too free and easy with the solder, you can fill up that hole. This isn't fatal, but if you do a neat job of filling it up, you may have trouble later trying to find the hole for the wire which was supposed to go there.

Short Cuts

One of the problems in putting together a kit is to stick to the instructions. There always seem to be a number of short cuts which make so much sense — but the chances are one in a thousand that they actually do make sense. For example, in putting together the 'scope, I had to attach a one-lug terminal strip to a point about halfway between the front and back of the chassis. Later, a wire was required from this terminal strip to one of the circuit boards near the back of the cabinet. Still later, I had to run a wire from the terminal strip to a potentiometer on the front panel. And where did the wire go when it entered the circuit board? To ground — so why not short-cut all this nonsense and run a wire from the potentiometer to the front panel, an inch away, which was also ground? Ask this question of an engineer and he will look at you sagely and say, "Ground loop."

Quite rightly, ground loops are a problem; they can introduce all sorts of odd effects and are the reason behind a number of wiring "idiosyncrasies". Take, for example, the bit of wiring shown in Fig. 3. RRG is ground; Y is ground; so why not pass the wire from UU2 through Y and to RRG? Ground loops. Further, in this case, we find that later there's a wire going from RRG to a potentiometer about an inch away, so we could have run a single wire from UU2 through Y and through RRG to the potentiometer. Much simpler. But — ground loops.

The firm rule here is: unless you are perfectly positive of yourself, follow the instructions precisely. I puzzled over the arrangement in Fig. 4 for some time; why wouldn't it have been easier and simpler to have mounted the left-hand lug facing to the right, so that it would have contacted the right-hand lug? That would have saved cutting and soldering "a piece of bare wire 3 in. long" and, later, discovering it was 2 in. too long and had to be twisted back on itself. The answer here is not (for once) ground loops, but making sure of a good connection. The bracket that angles off the edge of the picture is a major supporting member; it could wiggle or flex and if it did, it would flex the bolt and the lug. So if the 2 lugs were soldered together, such flexing might break the connection. Close your eyes, think not, just follow the instructions!

Miscellany

Finally, some passing comments. Sorting out resistors is a problem. If you are thoroughly familiar with color codes you can, like most skilled technicians, reach into a jumbled pile and almost automatically pick out the right one. Just watching such a procedure makes me nervous, so I have worked out the following classification method. First, I get several sheets of ordinary white paper, say 8½ by 11 in. At about one-inch intervals down the left side I write the values of resistors called for in the parts list. If three 100-K ½-watt resistors are called for, I write down 100 K three times. I make separate sheets for different wattage ratings. When all resistors are listed, I start on the pile, checking color coding very carefully. A device such as the IRC "Resist-O-Guide" is helpful and reassuring at this stage. As each resistor is identified, I insert one lead twice through the paper, Fig. 5, opposite the matching value. Finally, if I am at all worried about values, I run over the sheets checking each value with an ohmmeter. This method provides an automatic check on the parts list and 99% certainty that a 100-K resistor won't land where a 10-K one was required.

Condensers (or capacitors) are sometimes a problem to identify. The values are printed in English on the bodies of many capacitors, but some rely on colored dots. Checking against the parts list and using a process of elimination is helpful here. This method is also useful when parts substitutions are made by the manufacturer, as is occasionally necessitated by tight supplies. When substitutions are made, and last-minute correction sheets included with instruction manuals, both the parts and

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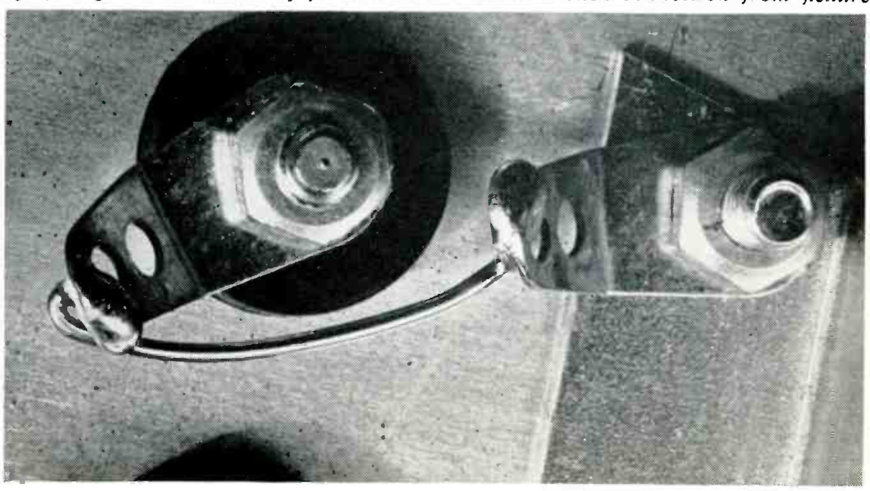
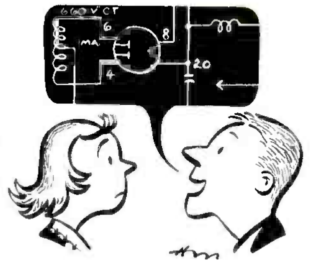
Fig. 4. Lugs are connected by flexible wire to avoid broken connection from flexure.



by JOSEPH MARSHALL

Practical Audio Design



Part III: Decoupling networks and voltage regulation.

EVERY amplifier or amplifier stage must have a load in order to achieve either amplification or power delivery. The load should ideally be coupled from the output of the tube directly to ground, so that no part of the signal could be developed in an impedance between the load and ground. Unfortunately, except for cathode followers, the load has to be returned to a source of voltage. And almost all power supplies, even batteries, have some internal resistance which represents an impedance to the signal frequency.

If several stages are used in cascade or series, and the main loads returned to the same point in the power supply, some voltage from each stage would be developed across the supply because there is impedance between this point and ground. Parts of the late-stage signals would feed back to previous stages. When there are only 2 stages the plate currents are out of phase; the signal fed back would result in degeneration — some loss of gain — and except for this, no great harm would result. But when there are 3 stages the signal currents of the first and third stages are in phase. The feedback would be positive and, if of sufficient amplitude, instability or oscillation could occur. This instability or oscillation almost always happens at a very low frequency because power-supply impedance increases as frequency decreases. The effect is familiarly called "motorboating" because it sounds exactly like the putt-putt of an outboard motorboat.

The condition for motorboating through a common impedance in the plate return circuits is diagrammed in Fig. 1. This is a 3-stage amplifier in which all the loads are returned to the same point in the power supply and therefore have a common impedance. There is usually a filter capacitor at this point to ground; there is also a series impedance consisting of the reactance of chokes and the resistance of filter resistors, and the rectifier. At most frequencies the reactance of the shunt capacitor is very low compared with either the resistance of the power supply or the resistance of the individual stage loads. Most of the signal at point X is, therefore, bypassed through the capacitor to ground. But at low frequencies the capacitor reactance begins to approach that of the power supply and/or the loads. Then part of the output signal will travel along the paths indicated by arrows back to the load of the input tube, producing positive feedback and low-frequency instability.

The solution to the problem has 3 steps: 1) arrange things so that the loads are returned to impedances which are not common to the several stages; 2) provide capacitors of very low reactance in front of these independent impedances so as to bypass the signal to ground; 3) reduce the impedance of the power supply to the lowest possible value.

This can be done by changing the circuit in Fig. 1 to that in Fig. 2. Here we have added some additional resistors (marked R_d) and capacitors (marked C_d) in the plate-supply lines. This accomplishes steps 1 and 2 specified in the previous paragraph. Each load now looks into an independent impedance, and at each point there is, so to speak, an escape path for the signal frequencies through the capacitor. There is a fairly high impedance between each stage, and a low impedance to ground from each stage.

It is obvious that the added resistances cause reduction of B+ voltages. To hold down this drop and thereby hold up the gain, the decoupling resistors should be no greater in value than necessary. In most audio amplifiers that have a power-output stage the voltage at point X is 100 volts or more higher than it need be to obtain adequate gain from the early stages; therefore you don't need to worry much about holding the decoupling resistances down except for another circumstance, illustrated in Fig. 3. Here one of the stages is redrawn. R_L represents the normal plate load, R_d the decoupling resistor, and C_d the bypass capacitor. The actual load consists primarily of R_L in series with the parallel combination of R_d and the reactance of C_d. At frequencies for which the reactance of C_d is very low, the effect of the decoupling network on the load is very slight. But as the frequency is lowered the reactance of C_d increases and, after a certain point, becomes high enough so that the combination of R_d and C_d in parallel produces a total impedance which is a considerable fraction of R_L. At that point, the low frequencies will be

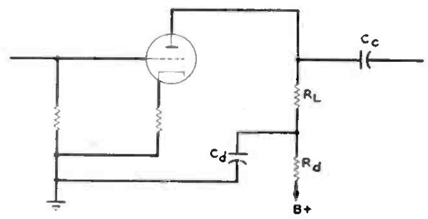
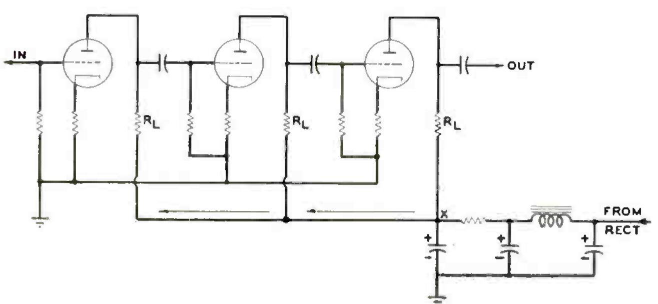
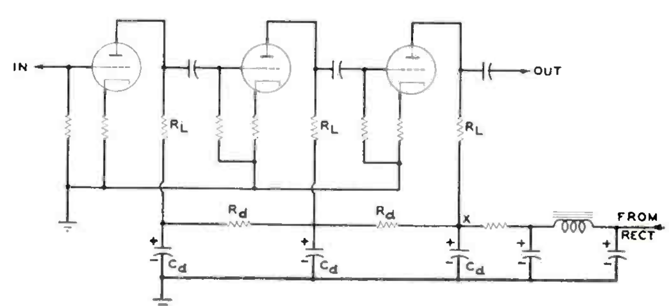


Fig. 3. Illustrating low-frequency boost.

Fig. 1, left: Condition for motorboating; plate loads connected to common power-supply impedance. Fig. 2, right: decoupled loads.





working into a greater load than the higher frequencies, the gain will increase, and the result is a boost of lows. This effect is sometimes employed in amplifiers to make up for low-frequency losses in the coupling capacitor (C_c) between the stages. It is quite possible so to proportion the decoupling and interstage coupling capacitors as to break even, as it were, and achieve a flat curve down to some much lower frequency than would be possible with-

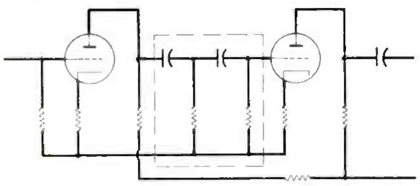


Fig. 4. Multi-section interstage network.

out the decoupling network. But this is rather a risky business. As might be expected, the boost in the low end means that there is more voltage which might be fed back to cause instability, and it is very seldom that one can be sure of obtaining both a better bass response and better stability in this way.

A good rule is to make R_d equal to 1/5 R_L in late stages, where the signal level is high; a higher value may be used in earlier low-level stages. The value must not be so high, however, that the voltage drop through it would result in applying too low a voltage to a preceding stage. Ordinarily, the rule given will take care of this, but, if the gain of some stage is reduced more than is desirable, it may be well to reduce the decoupling resistors.

The time constant of R_d and C_d (R in megohms times C in μfd) should not be less than .01 for any kind of audio amplifier (except possibly a speech-frequency ham or telephone amplifier); for high-quality amplifiers it should not be less than 0.1. The simplest way to improve the time constant is to increase the capacitance. Electrolytics are universally employed today and they are good enough to cause little trouble. Occasionally, however, one is excessively leaky, and sometimes one is noisy and

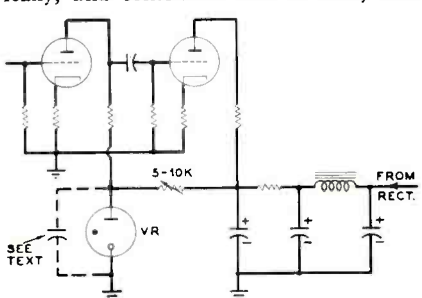


Fig. 5. VR tube employed as decoupler.

will be troublesome in low-level stages. It is not a good idea to use decoupling resistors greater than 50 K with electrolytic capacitors, and lower values are preferred. Electrolytics sometimes have high reactance at high frequencies and, because of this, it is possible to have feedback through the power-supply impedances at high frequencies. This can be minimized or corrected by wiring paper capacitors of from .01 to .05 μfd in parallel with the electrolytics. Stubborn cases of parasitic feedback at high frequencies can sometimes be cured by this method.

The decoupling network, then, is actually a part of the hum-filter network. Indeed, both hum filtering and decoupling are achieved by the same components. Ordinarily the need for progressive hum filtering discussed in part 2 of this series dictates the use of a hum decoupling element between every stage in a multiple-stage amplifier. However, if hum isn't a serious consideration, and decoupling with economy is, a saving can be made by using decoupling filters only between odd-numbered stages. That is, the second and third stage might be fed from the same point in the power supply, while the first stage had a decoupling filter. When the number of stages exceeds 3, the decoupling problem be-

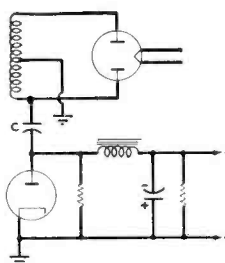


Fig. 6. A diode of conventional type can be used for bias supply.

comes severe; even extensive decoupling may not prevent motorboating. One reason for this is that there are other paths for feedback: the plate-grid capacitances of the individual tubes; coupling by adjacent spacing of tubes or wiring; plus, in most modern amplifiers, a deliberate negative feedback loop. With so many paths for feedback, even a little feedback through the common plate impedances may be sufficient to cause trouble. It is not wise, therefore, to attempt to use the same power supply for more than 4 stages at the most, and much safer to use one power supply for the power amplifier plus its driver and voltage-amplifying stages, and another power supply for the tone-controls, mixers, and preamplifier stages.

But if it is absolutely essential to use a single supply for many stages, it is often possible to prevent motorboating by putting sharp cutoff filters between the stages. This is achieved simply by adding another capacitor and resistor to the grid-coupling network, as in Fig. 4. Using this at several points may produce acceptable performance of multiple-stage units. However, this applies only if there is no negative feedback loop around the whole amplifier; if there is, the addition of the sharper cutoffs will only accentuate phase shift in the feedback loop and lead to more serious instability. I find the double RC coupling network most useful between the input to the power

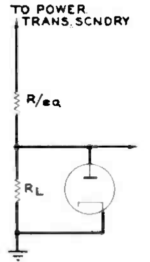


Fig. 7. An equivalent circuit for the bias supply in Fig. 6. See text for explanation.

amplifier — just ahead of the stage to which overall feedback is applied — and the preceding control unit or preamplifier.

Voltage Regulation

It has been pointed out that lowering the internal impedance of the power supply reduces the probability of instability. A power supply with voltage regulation has extremely low impedance. If vacuum-tube regulation is used, the common impedance may be so low that no decoupling network at all is needed. Furthermore, a voltage regulator provides extremely effective hum filtering; a device capable of bypassing the very slow irregularities of a DC current can offer little reactance to audio frequencies.

I have had good results with gas-type voltage regulators of the OA2 and VR-150 types as decoupling and hum filtering elements. As many readers know, I have incorporated them in several of my designs. They furnish a simple and effective way to obtain very good decoupling and hum filtering. Since one VR tube costs less than just about any combination of choke or resistor and capacitor, it is also an inexpensive way.

The application of VR tubes for decoupling requires very few precautions. First, the current flowing through a VR tube must be sufficient to ignite it and

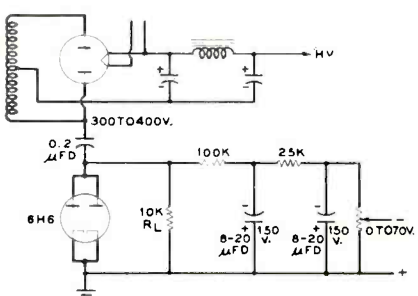


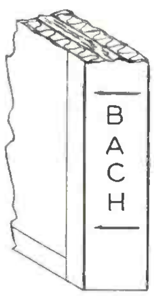
Fig. 8. Practical variable bias circuit.

keep it ignited. This can be insured by using a variable series resistor between the VR tube and the power sup-

Continued on page 41

Filing LP's

Long-playing records usually come packed between 12 by 12-inch pieces of corrugated cardboard. Convenient marker-dividers can be made by taking 3 of these cardboard squares, reversing the grain on the center one, and binding them together with 1½-inch gummed tape. The edges will be about ½ in. wide. Choose the smoothest edge and on it print vertically the classification of the record grouping to be identified by the marker. If the records are at the left of the marker, arrows pointing to the left may be drawn above and below the identification as shown in the illustration. Records can be filed by composer, period, or in any other way the owner wishes to have them.



LP file marker made of corrugated record mailing reinforcers.

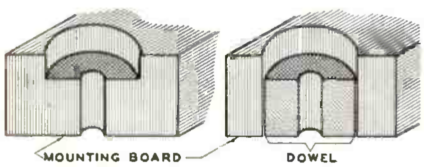
Edward H. Marsh
Purcellville, Va.

Turntable Mounting

Although mainplate suspension systems vary greatly in detail, many require drilling concentric holes of different depths, Fig. 1, in the turntable mounting board — a nasty job, particularly in plywood, when the object is to get the turntable level. In the case of some units, the depth of the larger hole may be different for each mounting hole.

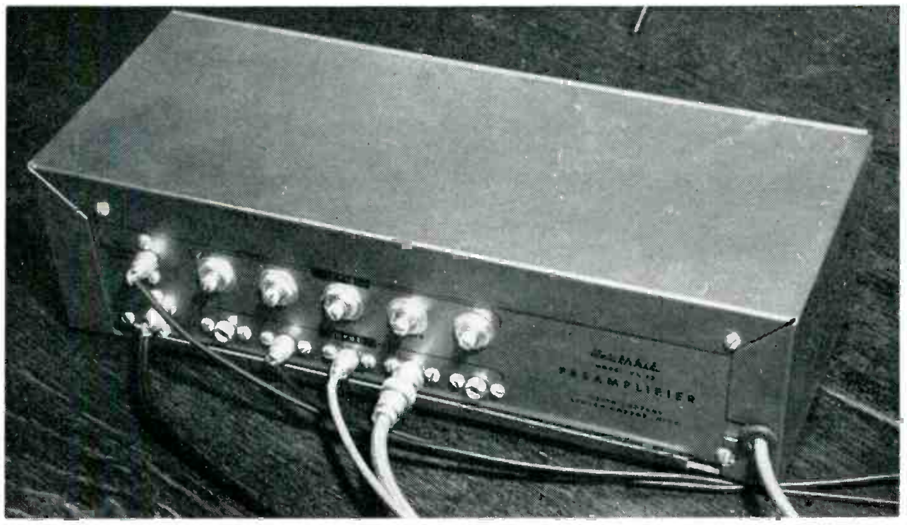
One solution is to drill the larger hole all the way through the mounting board. Then take a dowel of such size that it makes a snug fit in this hole and, proceeding from the underside of the mounting board, push the dowel (liberally coated with a good wood glue) up into the hole until the required depth is obtained, Fig. 2. Allow the glue to dry thoroughly; then trim the dowel even with the underside of the mounting board and drill the smaller hole in the "plug". Using this method it is possible to establish the depth of the larger hole with great accuracy.

Fig. 1 *Fig. 2*



Paul E. Soherr, 2nd Lt.
Dover AFB, Del.

Noise-Free Connectors Reduce Hum



STANDARD pin-tip phono plugs and connectors are quite inexpensive, but that is about all that can be said in their favor. They are unhandy, inclined to be fragile, and often make a very poor ground connection to the socket. This does not normally cause any great difficulty, except when working with a low signal level — as, for example, with a Ferranti or Electro-Sonic pickup. Under such conditions hum becomes a great problem. Rotating the plug in the socket may make the hum fluctuate 20 db or more, and it won't stay low if you try to pick a best position.

With a low-level pickup one should also use a well-shielded cable for the same reason, hum reduction; most phono cables have poor shields. The Ferranti people recommend RG-59/U and RG-71/U cables, the latter a double-shielded coax with the lowest available capacitance. I am using 71, both for hum reduction and for minimum attenuation of highs with a long cable.

For both these cables, the Amphenol coaxial connector series is made to order. The socket is UG-290/U, and the plug is UG-260/U for the 70-ohm cables, or UG-88/U for any smaller diameter 50-ohm cable. The ground shields on these plugs are spring loaded, a great virtue. They therefore snap on and off with just a twist of the wrist, and also make a good ground. They are easy to assemble — the shield of the cable is simply clamped, not soldered. Unfortunately, they cost over $1.00 a unit. One only needs a set on the phono input of the preamp, however — not on the high-level inputs.

The socket can be added easily in place of a standard phono connector simply by enlarging the main hole — the mounting screw spacing is the same as at least some makes of pin-tip phono connectors. My Heathkit preamp, so modified, is shown in the illustration.

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Rumble Cure

Flutter and rumble are usually caused by mechanical transmission of motor vibrations to the turntable. One of the recommended remedies is to tap the motor armature smartly with a wooden mallet. This, presumably, lets the motor shaft settle snugly in its bearings. I have come across several rumbling turntables which took kindly to this treatment, but I found it most irritating to dismantle the record player, tap the motor, put it back into position, and check it with a record (the clarinet is the best instrument to check flutter with), only to find that the job was not done completely, necessitating a repetition of the whole procedure. It is very helpful and saves time to use a microphone as a sound probe. Any part of the mike housing can pick up motor vibrations when it touches the spindle or other stationary part of the record player. If the microphone is connected with the amplifier and speaker while this is done, the motor noises can be heard in all detail, and the tapping of the armature can be followed immediately by an aural check with the microphone, until the quietest operation is achieved.

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BASIC ELECTRONICS

V: Magnetic effects.

by Roy F. Allison

Magnetism

Everyone is familiar with magnets and magnetism, and with some of the basic characteristics of magnetic structures. Iron, nickel, and cobalt, with most of their alloys and many of their compounds, are magnetic substances; that is, they are attracted by magnets or can become magnetized themselves.

A bar composed of one of these materials can be magnetized by stroking it with another magnet or by winding a current-carrying coil of wire around it. Some magnetic substances, notably hard steel, are difficult to magnetize but, once magnetization is accomplished, retain the magnetized state tenaciously. They are known as permanent magnets. Other substances — "soft" iron, for instance — become magnetized easily but do not hold their magnetized states well. These are most often used inside electrical coils to form electromagnets; when current exists in the coil a strong magnet is obtained. As soon as the current is interrupted, however, the bar loses most of its magnetization; the little that is left is called residual magnetization.

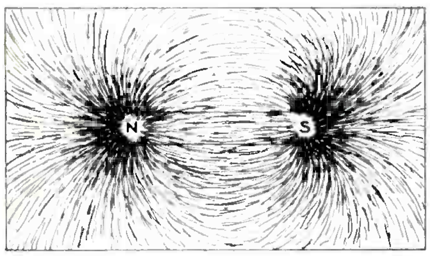


Fig. 1. Iron filings in a magnetic field.

Any magnet, no matter what its shape, has 2 poles or ends at which the magnetic strength is most concentrated. If the magnet is suspended so that it is free to move in any direction, and away from the influence of other magnets, it will invariably rotate so that its poles are aligned roughly north and south. Further, if this alignment is disturbed, the same end will always return to the north and the opposite end to the south orientation. A magnetic compass is nothing more than a magnetized needle with free suspension about its center. Evidently, the 2 poles of a magnet differ in some way; this difference in effect is known as polarity. The pole of a magnet that seeks north is called, logically, its north pole, and the opposite is the south pole. It is found invariably that the poles of a magnet, though different in polarity, are exactly equal in strength. Also, when 2 magnets are brought close together, the north poles repel one another and the 2 south poles act in the same way; one south pole attracts the other north pole, and vice versa. If the magnets are held in the hands this mutual attraction and repulsion can be felt easily — as the 2 are brought closer together a strong twisting force attempts to align the magnet poles north-south and south-north. In this respect, then, magnetic forces are similar to electrostatic forces: like magnetic poles are mutually repulsive, while unlike poles are mutually attractive.

As a sidelight, it is interesting to note that the earth itself is a huge magnet, which accounts for the alignment of compasses in the general north-south direction. But — since the north pole of a compass magnet points north, and the south pole south, then the "south" pole of the terrestrial magnet must be in the geographical north, and the "north" pole in the southern hemisphere!

There are at least 2 widely held theories to account for magnetism. In the one currently most satisfactory, electron motion in atoms is examined. Since electron movements represent electric current, and magnetic effects are closely tied to electric currents, the theory seems reasonable.

It will be recalled (from Chapter I) that an atom consists of a central core or nucleus, containing primarily neutrons and positive protons, and rings of planetary electrons which revolve about the nucleus. Besides this orbital revolution, the electrons spin about their own axes just as solar system planets do. In magnetic substances the crystal structures are combinations of atoms having electrons spinning predominantly in a given direction; that is, there is a spin direction taken by some of the electrons that is not compensated for by electrons spinning in other directions. Crystals having similar uncompensated spin directions are grouped in extremely minute regions (domains) throughout the substance. Each domain is permanently magnetized, and when the substance as a whole is unmagnetized the domains are not aligned nor polarized in any particular order; they cancel each other. When the substance is magnetized, however, the domains are organized or aligned to a degree depending on the strength of magnetization. The difficulty with which the uncompensated spin directions of the crystal electrons in a substance can be organized or disorganized permanently determines the permanency of magnetization. But any substance can be demagnetized by heating it excessively; the increased activity of the atoms caused by the heat results in loss of order in their arrangement, and the uncompensated spins return to random orientation. Demagnetization can be accomplished also by subjecting the magnetized object to a gradually decreasing alternating current field; more on that in another chapter.

As with electric charges, the force of attraction or repulsion between 2 magnetic poles is directly proportional to the product of the pole strengths and inversely proportional to the square of the distance between them:

$$F = \frac{m_1 m_2}{\mu r^2}$$

Here F is in dynes, r is separation in centimeters, and μ (permeability) is a constant determined by the medium

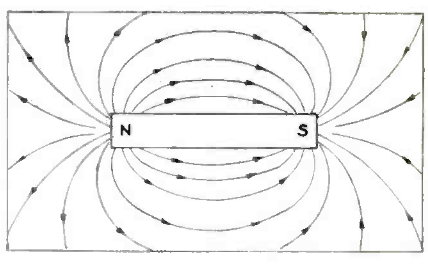


Fig. 2. Direction of field around magnet.

through which the force acts. Permeability is somewhat analagous to conductivity. For a vacuum the permeability is 1, and for air it is nearly 1. The equation defines the unit pole also: ***The unit pole is that strength of magnetic pole which, when separated 1 cm in a vacuum from another of the same strength, will exert a force of 1 dyne upon it.*** Thus, m_1 and m_2 in the formula are the strengths of the magnet poles expressed in unit poles. If the magnet poles are of the same polarity the force is one of repulsion; if of opposite polarity, one of attraction.

Readers having an uneasy feeling here will be glad to know that it is justified. We are discussing isolated poles and their effects on one another, yet not long ago the statement was made that any magnet has 2 poles of equal strength and opposite polarity. This must be taken into consideration when applying the formula; the forces acting between both pairs of opposite poles must be calculated and added, vectorially, and then the same process carried out for the similar pairs of pole. The difference is the total force between the magnets.

Yet the concept of an isolated pole is useful in other ways as well as in defining the unit of pole strength. From experience we know that the area of influence of a magnet is, in practicality, limited, although we cannot say exactly where it becomes negligible. This area of practical influence is called the magnetic field, and in it other magnets or magnetic substances will be subject to forces having both direction and magnitude. Assume a bar magnet long enough that one pole can be brought within the field of another magnet while the other remains outside that field. We will then have, effectively, an isolated magnet pole in a field. As it is moved within the field the amplitude and direction of the force acting on it change, of course, and because we have an isolated pole the following definition can be made: *The direction of the force acting on an isolated north pole at any point is the direction of the magnetic field at that point.*

The intensity of the field around a magnet also varies; the unit of intensity is the oersted, after the pioneer in electromagnetism, Hans Christian Oersted. *One oersted is that magnetic field strength at which the force on a unit pole is 1 dyne.* Since the force is directly proportional to the strength of the magnetic pole and that of the field which acts on it, $F = mH$, where H is the field intensity in oersteds and m is the strength of the isolated magnet pole in unit poles. This is usually written as

$$H = \frac{m}{F}$$

It follows from this and the formula for the force between 2 magnetic poles that the field intensity from one magnet pole is proportional to the strength of

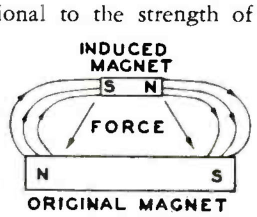


Fig. 3. How a magnet induces poles in other magnetic bits.

the pole in unit poles, and inversely proportional to the square of the distance in cm from the pole:

$$H = \frac{m}{\mu r^2}$$

The symbol μ represents, again, the permeability of the surrounding medium.

Now, suppose a sheet of paper is held over a bar magnet and iron filings are sprinkled on the paper, after which it is tapped gently. The filings will arrange themselves in a pattern of alignment with the magnetic field around the magnet; it will look like the rough drawing in Fig. 1. Near each pole the accumulation of filings is most dense, showing graphically the greater field intensities at those points. Moreover, the pattern has marked directionality, indicating the direction of the field at all points. Fig. 2 is a diagram representing more clearly the field around such a magnet. The lines show the direction of force that would act on an isolated north magnet pole placed in the field at any point; they are, accordingly,

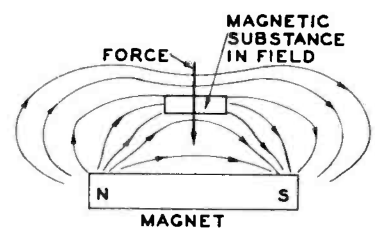


Fig. 4. Elastic force-line theory.

called magnetic lines of force. Because they indicate the direction of the field they cannot cross, but flow side by side in a smooth pattern from the north to the south magnet pole. If an unmagnetized bit of magnetic substance, such as iron or steel, is placed within the field it is attracted to the magnet. Why? Because the field of the magnet, passing through the substance, induces in it magnetic poles of such polarity that they are attracted to the corresponding opposite poles of the original magnet. This process, shown in Fig. 3, is called magnetic induction. Note that in the original magnet the lines of force flow internally from south to north; the same lines in its external field, flowing through a magnetic substance, must of necessity produce a south pole where they enter and a north pole where they leave the substance.

Another way to explain magnetic forces is to think of the lines of force as being elastic in nature and inclined to seek the shortest external path between poles, this tendency offset to some extent by mutual force-line repulsion. The balance determines the field configuration. The reason for this behavior is not clearly understood, but it is consistent with observed effects. For an illustration of this method of reasoning, see Fig. 4. Because the permeability of the magnetic substance inserted in the field is greater than that of air, the lines of force are distorted by its presence; they tend to converge and

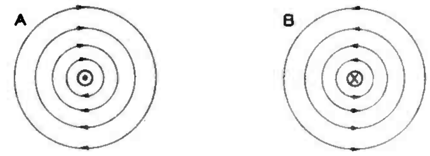


Fig. 5. Magnetic lines around conductors.

flow through the substance. But this makes them longer than before, and because of their "elastic" nature they tend to draw the object toward the magnet, thus decreasing their lengths again.

Electromagnetism

We have remarked before on the close relationship between electric current and magnetic fields. It should not be surprising, therefore, that an electric current is always accompanied by an electromagnetic field that is equivalent in its effects to a magnetic field. Electromagnetic fields around conductors are shown in Fig. 5; the conductors are viewed from the ends. In Fig. 5A the current direction is toward the viewer, shown by a dot representing an arrowhead, and in Fig. 5B the current is away from the viewer (into the page). This is indicated by a cross to represent the tail of an arrow. The field intensity is greatest nearest the wire in either case, but the direction of the lines of force depends on the current direction. Field direction for current-carrying conductors can be determined by the left-hand rule: if the thumb of the left hand is extended outward from the closed fist, and pointed in the direction of the current, the fingers curl in the direction of the field. If the conductor is long and relatively straight, the field intensity (H) in oersteds at any point r cm from the conductor is

$$H = \frac{I}{5r}.$$

I is, of course, the current in amperes.

Suppose the conductor is bent in the form of an arc, as in Fig. 6. Point P is at the center of the circle determined by the arc. Lines of force are drawn around the conductor at 4 places along the arc. It is easy to see that they will converge and reinforce at the center, so that the total field intensity at the radial distance P will be greater than it would be at the same distance if the wire were straight. The formula for field intensity at the center of the arc is

$$H = \frac{Is}{10r^2},$$

where I is current in amperes through the conductor, s is the linear length of the curved conductor, r is the distance from the conductor to the center of arc, and H is field intensity in oersteds.

If the arc is extended to a complete circle, forming a closed loop, the formula need be altered only by substituting for s the term for a circle's circumference referred to its radius: $2\pi r$. (The value of π is very close to 3.14.) This gives, then,

$$H = \frac{\pi I}{5r}.$$

Further, if the conductor is coiled again and again to form several turns lying close to each other — so long as the length of the coil is minute compared

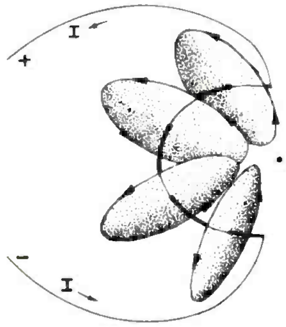


Fig. 6. Force patterns for a curved wire.

to its diameter — the total field intensity will be increased in direct proportion to the number of turns. Calling the number of turns N,

$$H = \frac{\pi NI}{5r}.$$

Now, assume that the wire is coiled in helical form to make a long cylindrical shape, as pictured in Fig. 7: this is called a solenoid. The field will be similar to that of a straight bar magnet. Intensity within the solenoid is given by

$$H = \frac{2\pi NI}{5l}.$$

The term l is the length of the solenoid in cm. Other terms are the same as given previously.

If a bar of some magnetic substance is placed within the solenoid it will be found that the total field is increased significantly. The field intensity in oersteds cannot, strictly speaking, have increased because, according to the formula, it is dependent only on the current and the number of turns per unit length. Still, additional lines are contributed by the inserted magnetic substance; they are attributed to magnetization of the substance by the original field. The original lines of forces are sometimes called the magnetizing field, as a matter of fact, and the added lines the induction field.

Taken together (since they are indistinguishable) the lines of force are called magnetic flux. *One line of magnetic flux is a maxwell,* a unit named for James Clerk Maxwell; the symbol for maxwells is the Greek letter Phi. Often, though, the total number of lines (maxwells) is not so important as the *concentration* of flux, or the flux density. The unit of flux density is the gauss (for Karl Gauss), and it is defined as follows: *a flux density of one maxwell per square centimeter is one gauss.* Its symbol is B. If a total flux of 10 maxwells passed perpendicularly through an area of 2 sq. cm, the flux density would be 5 gauss; if the area were 40 sq. cm, the flux density would be 0.25 gauss. Remember (when interpreting loudspeaker specifications, for example) that maxwells is the total flux in a magnetic circuit, and gauss is the flux density.

Since the area of the magnetic field in the solenoid did not change when

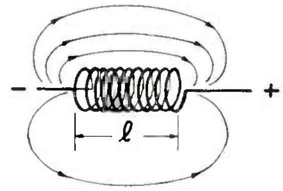


Fig. 7. Field around a solenoid.

the magnetic core was added, while the total flux increased, obviously the flux density must have increased too. The ratio between B (flux density with the magnetic core) and H (field intensity or density without the core) is the permeability (μ) of the material:

$$\mu = \frac{B}{H}.$$

Permeability of magnetic substances is often much greater than unity, but it is difficult to state fixed values for various substances, because they are dependent not only on the substance but on its state of previous magnetization and even on the intensity of the field it is used in.

WITHOUT INSTRUMENTS

Continued from page 31

service. A slightly leaky coupling capacitor can put a positive voltage on the following tube grid, with resultant distortion. Referring once again to Fig. 2, you'll remember that shorting those test points marked Z to ground should cause only a slight pop in the loudspeaker. If a loud noise does occur, it is probable that the interstage coupling capacitor is leaking or shorted, although in some instances it may indicate a gassy tube.

Another method of localizing the source of distortion is to use signal injection. This requires a clean audio signal with an amplitude of a volt or two, such as might be obtained from a tuner, preamplifier, or tape recorder. The output of the signal source, Fig. 4, is fed through a 0.1-μfd capacitor in series with a 100-K resistor, in order to protect the preamp or other source from possible damage by DC voltages in the amplifier, and to prevent circuit interactions.

The procedure to be followed is much like that used previously with the flashlight battery, except that in this case the sound should be listened to carefully for quality. Start out with points A and A', as shown in Fig. 2. Poor sound quality at either point may indicate bad output-tube unbalance, a gassy output tube, a leaky coupling capacitor, or some other trouble in the output stage. If the signal sounds clean, though weak, go back to points B and B'. Here the signal from

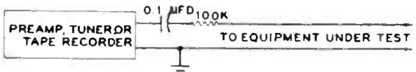


Fig. 4. Signal source for live testing.

the speaker should be much louder, due to the extra amplification of the driver stages, and sound quality should be the same from either point B or B'. Once this stage is checked out, go back to point C, where again you should obtain an increase in loudness because of the extra gain contributed by the input stage. If sound quality is clean throughout the amplifier at low volume levels, but not at higher levels, it's possible that the rectifier tube is weak and is not supplying high enough voltage. Of course the other tubes, particularly in the output and driver stages, are suspect too.

To reiterate, the main idea in trouble shooting is to isolate the difficulty to the point of origin. Then you have something relatively simple to experiment with and change. While solving the first troubles may be difficult, as your experience and self-confidence grow you should be able to keep the family hi-fi system in operating condition. But your critical ear will probably become more demanding as your experience increases, and will insist that your system be in *top* operating condition. This requires at least a minimum supply of test equipment, which is available in inexpensive kit form. Building it will not only supply an outlet for your creative urge but will provide you with tools for doing your own servicing more easily and completely.

The rules given for amplifier trouble shooting are valid also for preamplifiers, tape recorders, and other audio equipments, although these may have complicated and specialized circuits. Always try to get as complete service data as possible on the equipment you own and keep it on file; it will not only help you to locate faults and understand the proper operation of the system components, but will be of considerable aid to the serviceman in the event that you need to call in outside assistance.

PRACTICAL AUDIO DESIGN

Continued from page 36

ply, as shown in Fig. 5. The resistor can be adjusted so that, with the tubes all in the circuit, at least 10 ma current flows through the VR tube. Alternatively, if there are as many as 3 or 4

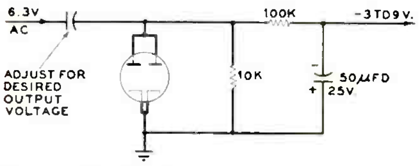


Fig. 9. Simple low-voltage bias supply.

high-mu triodes drawing from 5 to 10 ma beyond the VR tube, a series resistor of 7,500 ohms (at least 5 watts) will do the job.

Some VR tubes are noisy and produce a hash which might be audible in high-gain amplifiers. Some may also oscillate in the circuit. It is usually a good idea to bypass the tube with a paper capacitor of .05 μfd if either effect is noticed. Sometimes, though, adding this capacitor only makes matters worse! Trial-and-error changes of shunt capacitance, or tube replacement, is necessary in such cases.

Bias Supplies

The use of fixed bias on power-output tubes often makes possible a considerable increase in power output for the same voltage supplied to the plates. Bias required for modern power-output tubes ranges from —10 to —60 volts. There are a number of simple ways to obtain the required bias voltage.

One of the simplest is the use of a shunt rectifier, shown in Fig. 6. No additional winding on the power transformer is needed. If the circuit is redrawn as in Fig. 7, it becomes obvious that the reactance of the capacitor at 60 cps (indicated by R_{eq}) and the rectifier load (indicated by R_L) comprise a voltage divider that reduces the high voltage of the power transformer. Any voltage needed can be obtained simply by changing the value of the capacitor, or the value of the load resistor, or both. The current drawn depends on the load resistance — the larger the resistance, the lower the current. A few ma is sufficient, and values of R_L around 10 K are satisfactory. Fig.

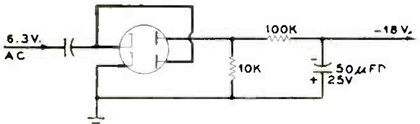


Fig. 10. Voltage doubler hookup, Fig. 9.

8 shows values for a practical supply delivering up to —70 volts, enough for just about any output tubes. Lower voltages can be obtained either by reducing the size of C or by using a pot across the bias supply and adjusting it for whatever bias value is needed.

Note that there is a high resistance between R_L and the filter capacitors. This resistance is essential; without it, the first filter capacitor would become the bottom half of the voltage divider, and excessive current would be drawn. The rectifier may be one or both sections of a 6AL5 or 12AL5 in parallel, a 6H6 or similar tube, a small triode connected as a diode (by tying grid to plate), or the independent diode of a multiple-section tube, provided the cathode can be grounded directly.

When a very low bias voltage is needed (—3 to —9 volts) the circuit drawn in Fig. 9 can furnish it with only the filament voltage as a source. Fig. 10 shows a voltage-doubling version which can supply —18 volts from a 6.3-volt filament string — enough for 6V6's, EL84's, and similar tubes.

Another way to obtain bias voltage,

Continued on next page

MODERN KITS

Continued from page 34

the corrections should be studied carefully to make sure everything jibes. In one kit assembled recently, I found that the features which would have identified a substitute part were incorrectly described; checking against the parts list cleared up the confusion since only one component was listed which had the value of the unidentifiable substitute.

Sometimes kit manufacturers, realizing that their customers are not all experienced machinists and do not know the difference between 6-32 × 3/8 and 8-32 × 3/8 machine screws, give life-size drawings of the hardware. But they often don't. Fortunately, I have never found a kit yet in which the quantities of machine screws of different sizes were the same, so the process of elimination can be used here too. Meantime, for your information, the first figure ("6" in 6-32) relates to the diameter of the machine screw, and the smaller the figure, the smaller the diameter. The second figure ("32" in 6-32) refers to the number of threads per inch; therefore the higher the figure, the finer the threads. Finally, the "3/8" refers to the length in inches. Thus the threads of a 6-32 and an 8-32 screw will match (you can lay one on top of the other and they will interlock or mesh), but the 6-32 screw will be thinner.

All this may make "assembling your own" sound forbidding. Don't be worried — too much. Be worried just enough to take it slowly, do exactly as the instructions say, take all the time you need . . . and then, what a feeling of satisfaction!

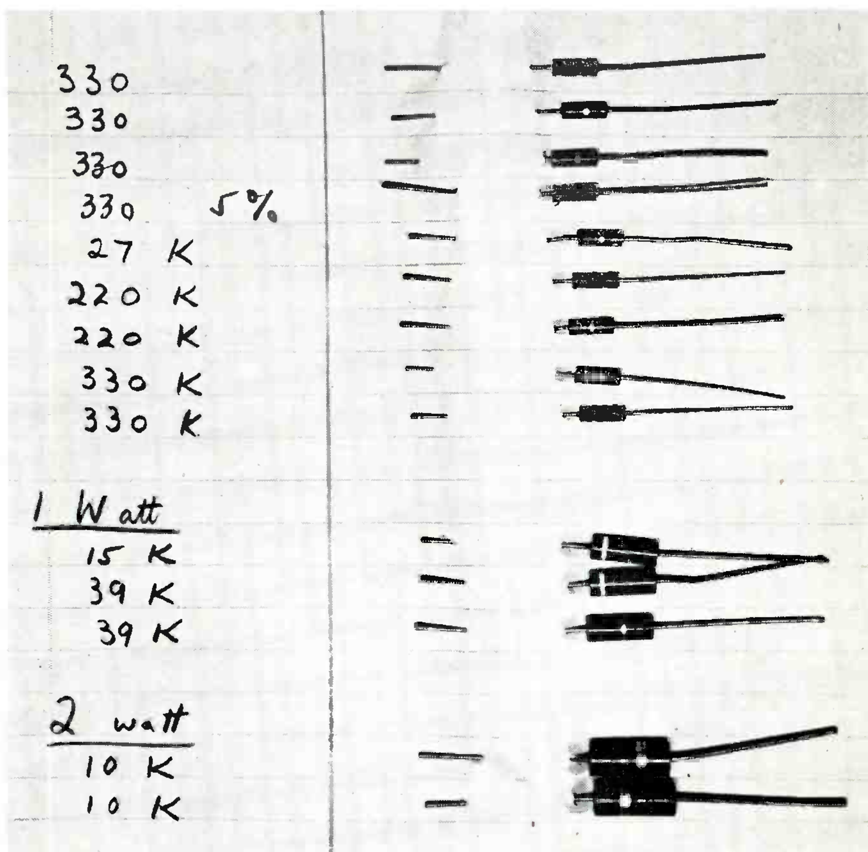


Fig. 5. An easy way to keep kit resistors sorted after they have been identified.

PRACTICAL AUDIO DESIGN

Continued from preceding page

especially when more than —100 volts is required (circuits, for instance, in which output tubes are directly coupled to cathode-follower drivers, and driver grid bias voltage is needed to buck out the cathode voltage) is to use a miniature 6.3-volt filament transformer wired in reverse, as in Fig. 10, and then rectify

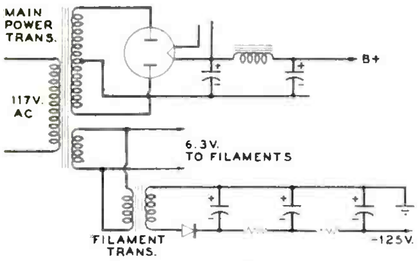


Fig. 11. High-voltage fixed bias circuit.

the 117-volt output. Inexpensive and compact selenium rectifiers are perfectly satisfactory for this purpose. If the load resistor is kept fairly high, the bleeder current is low and a high-resistance filter can be used to achieve adequate hum filtering.

The trouble with using fixed bias is that failure of the bias supply removes bias from the output tubes and may, in consequence, burn out the plates in a few seconds. There are at least 2 very simple devices to provide insurance against this, one of which will maintain operation in spite of a failure. In Fig. 11 is shown an output stage with fixed bias. Note that there is a bias resistor in the cathode circuit but the

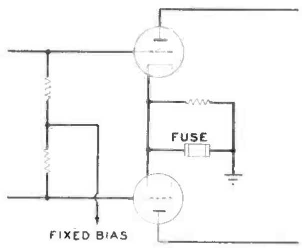
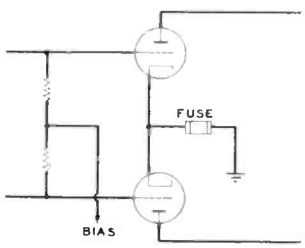


Fig. 12. Emergency bias circuit.

resistor is shorted by a 1/4-amp, 250-volt fuse. Failure of the bias supply would increase current violently, causing the fuse to blow. This would remove the short on the self-bias resistor until the fault was corrected.

In Fig. 12 there is the same fuse

Fig. 13. Simple protective device.



but no bias resistor. Failure of bias will blow the fuse but, in this case, the result is merely to open the cathode circuit and prevent flow of plate current. This is not quite safe; a tube can be damaged, though it is far less likely to be, when plate and filament voltages are applied with an open cathode. But it does give immediate indication of trouble. These measures cost so little that it is foolish not to include them.

WOODCRAFTER

Continued from page 11

tion so it's better to plan on making each piece 11¾ in. finished length.

Plywood has a tendency to splinter during sawing unless certain precautions are taken. Here are 3 proven methods; take your choice of one or a combination. 1) If you use a crosscut handsaw or a motorized bench saw, make sure the saw is sharp and keep the important side of the plywood facing up. If the saw is a portable electric saw, keep the good side of the plywood down. 2) With a sharp knife or chisel, score along the line of cut on both sides of the plywood before sawing. The scoring should penetrate the surface layer of veneer. 3) Cover the cutting line with either cellophane tape or masking tape before sawing.

Boring holes in plywood presents no more hazards than the same operation in ordinary lumber. In either case the possibility of splintering, as the bit or drill pushes through the opposite side, can be eliminated by clamping a piece of scrap wood to the back side of the work and boring through into the scrap.

Planing plywood edges can be tricky since the plies run counter to each other. Approach the job much as you would planing the end grain of regular lumber. Starting from one end plane toward the center; then reverse the process and plane in from the opposite end. This will prevent splintering the ends. The plane should be operated somewhat diagonally to the stock.

Usually plywood edges that are exposed to view require special treatment for a smoothly finished effect. Plan your construction for a minimum of exposed edges and then decide on a method of handling the edges that are to show. One of the simplest treatments is to cover the edges with strips of matching solid wood with mitered corners. Use glue and brads for attaching all wood facing. Shaped molding, available in numerous patterns, is effective as a frame for a speaker enclosure or along the edge of a cabinet top or shelves of a bookcase. Simple half- or quarter-round molding makes a smart trim and can be the same width as the edge or narrower, allowing a small shoulder of the edge to appear on each side of the molding. In some cases it is possible to miter the edge, taking care not to damage the scrap strip of plywood. Reverse the strip so the veneered side faces out and re-attach it in this fashion to the mitered edge, restoring the squareness with a new, smooth finish now exposed to view.

If a piece is to be painted or stained dark, the edges can be filled with plastic wood of the desired color and sanded thoroughly. In still another method, you can burn the edges by concentrated application of a power sander until the wood becomes discolored. The result will be a hand-glazed finish, ideal for painting or very attractive without touching a brush to it.

Plywood edging has become much simpler with 2 recent wood-veneer products designed for easy application. One type comes coated with adhesive which sets as a warm iron presses the veneer against the plywood edge. The other (Weldwood Wood-Trim by U. S. Plywood Corp.) is packaged in rolls 1 in. wide and 8 ft. long, retailing for about 79¢ per roll. This is applied with any good wood glue and offers a choice of woods: fir, oak, walnut, mahogany, and birch.

Finishing plywood surfaces is a subject we'll discuss in the very near future in an article devoted to various types of wood finishes. Careful cutting and assembling of plywood cabinet parts will assist greatly in giving the final finish a professional appearance.

YOUR OWN AMPLIFIER

Continued from page 23

sistor so as to improve the high-frequency response.

Alternatively, if we are considering an input stage, there are reasons why it would be better to keep a high plate-load resistor, with a fairly high value of cathode-bias resistor, and shunt the stage down by using a lower value of resistor in the grid of the following stage. Both procedures will, of course, reduce the gain too. These conditions we shall discuss later in the series.

There is not much problem at the low-frequency end. A .01-μfd coupling capacitor will have a reactance of 1.25 megohms at about 13 cps — and we certainly don't need to use as small a value as .01 μfd.

Triode or Pentode?

We can see right away why triode tubes are preferred for most applications in audio amplifiers, except the output stage, which we shall discuss in later articles. The operating conditions are easier to calculate; the tubes have more stable gain and more consistent distortion figures. If you try fitting load lines to the characteristics of the 6BR7 or any other pentode, you will find that a relatively small movement of the load line on the curves, representing slight changes in operating conditions (which can easily occur), will alter the proportioning of the spaces along the load line quite considerably. This is illustrated at Fig. 10.

If the load line gets far toward the "knee" of the zero grid voltage curve, the grid curves close up toward the top end as well as the bottom end of the line.

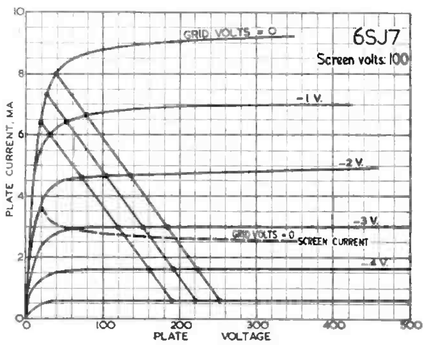


Fig. 10. Characteristic curves for 6SJ7 pentode. Position of the load line is critical for pentodes. Each load line is for resistance of 30 K; displacement is caused by variation of other parameters, and affects distortion. Center load line closes equally at ends, indicating mostly 3rd harmonic distortion. Other 2 lines indicate 2nd harmonics of opposite phase.

If the closing up is approximately similar at both ends of the load line, the resulting distortion produced is mostly third harmonic. But if the load line is moved further away the distortion contains second harmonics in opposite phase to those produced when the load line is too near the knee, so that the closing up is more toward the top end.

In short, without going into a lot of detailed work, pentode characteristics are fairly critical as to operating conditions for obtaining minimum harmonic distortion. Because in audio work it is important to achieve very low distortion, it is generally better to keep away from pentodes unless there is some good reason for using them.

AUDIO AIDS

We'll pay $5.00 for usable Audio Aids. See page 46 for details.

BUILD IN YOUR HI-FI

Continued from page 18

leads are brought into the enclosure by means of a 1-inch hole drilled in the side.

Grille cloth should be cut to such a size that it can be stretched slightly to prevent wrinkling and vibration against the speaker panel. Tack the cloth to the wood, leaving room for access to the panel screws, since you may have need to remove the speaker or get into the enclosure later.

Any small decorative molding, mitered at the corners, will supply the finishing touch to the enclosure and conceal the rough edges of the grille cloth.

The entire cabinet was first coated with a primer-sealer and then finished with a flat wall paint. Hinged doors and striated panels were painted in a lighter shade for contrast. Final steps in construction included the 1-inch cove molding around the outside edges of the cabinet, and the clamshell molding along the bottom frame. These should be painted before assembly.

AUDIONEWS

Continued from page 7

D storage section is finished inside and fitted with an adjustable shelf.

The practicability of the 2 new models is pointed up by a combination of hinged and sliding doors. Both extreme right and left doors hinge outward, while the center panel slides left or right. When the left and center sections are devoted to equipment storage, accessibility to equipment is ideal. The right section is available for speaker mounting or additional storage space.

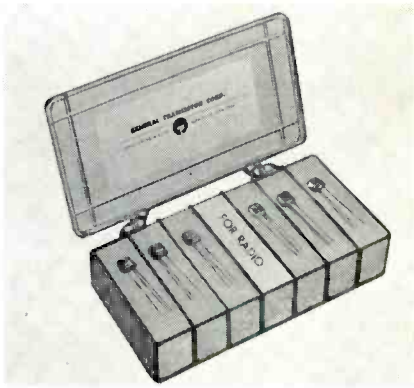
Cabinart King-Size component cabinet.

The Model 65 is contemporary in style. Detail is held to a minimum, including only a slight bevel molding, tiny brass door pulls, and wood or brass legs. Finishes are hand-rubbed mahogany, walnut, and korina veneers. Black lacquer, too, is available at no additional cost. Retail price is $210.00.

TRANSISTOR KIT

General Transistor Corp. of Jamaica, N.Y., has made available a kit of 6 diffused P-N-P junction transistors for all types of radio receivers.

Designed for experiments, engineers, and technicians, the kit includes 1 converter-oscillator transistor, 2 intermediate-frequency transistors, and 3 audio transistors. The transistors are packed in a lucite box. Users of the kit will find that, in building almost any type of radio, transistors can be substituted for tubes, thus making the set much more compact.



General Transistor kit for radios.

Kit No. 2, as it is known, is available for immediate delivery. The suggested retail price is $17.95.

PAPER-CAPACITOR CATALOGUE

Plastic Capacitors, Inc., have released a new catalogue covering their line of paper capacitors. Catalogue 155 includes all of this company's paper capacitors for AC and DC applications.

The catalogue contains data on mineral oil impregnated paper-dielectric capacitors in a variety of containers, synthetic oil impregnated paper-dielectric capacitors, and Aroclor-AC capacitors. Information includes charts, diagrams, technical information, and part numbers.

Copies of the catalogue are available on request.

STARLIGHT TURNTABLE

The Metzner Engineering Corporation of Hollywood, Calif., has just announced the development of a new center-drive turntable named the Metzner *Starlight*.

The new turntable will handle all size records from the smallest home recording to full-size professional transcriptions. A built-in "pop-up" hub provides for standard 45's. The drive motor is the conventional 4-pole type and is fully shielded to assure minimum induction hum when magnetic cartridges are used. An exclusive feature of the Star-

The Metzner Starlight turntable.

light is the manner in which the drive motor is fully encased in lead to eliminate possible vibration.

The turntable incorporates 4 speeds, including the new 16⅔ rpm for talking books. All speeds are adjustable to pitch with the use of an illuminated, built-in stroboscope. The turntable itself is machined of special aged aluminum and is padded with pure cork. Wow and flutter, according to the manufacturer, are held to less than 0.2% RMS. The entire unit is mounted on heavy-gauge aluminum plate with the vernier-type speed control conveniently located and clearly marked. Retail price, less pickup, is $49.50.

For complete information, write to the Metzner Engineering Corporation, Dept. S, 1041 North Sycamore Ave., Hollywood 38, Calif.

FISHER AM TUNER

The Fisher Radio Corporation has announced an AM tuner, the *Model AM-80.* The new tuner is the AM counterpart of the Fisher FM-80 FM tuner.

The AM-80 features a relative-sensitivity tuning meter and a 3-position

New deluxe Fisher AM-80 tuner.

adjustable bandwidth control. Frequency response of the audio section of the tuner is said by the manufacturer to be uniform from 20 to 20,000 cps with the bandwidth control in the broad position. The unit has 3 high-impedance inputs and a cathode-follower output which permits leads up to 200 ft.

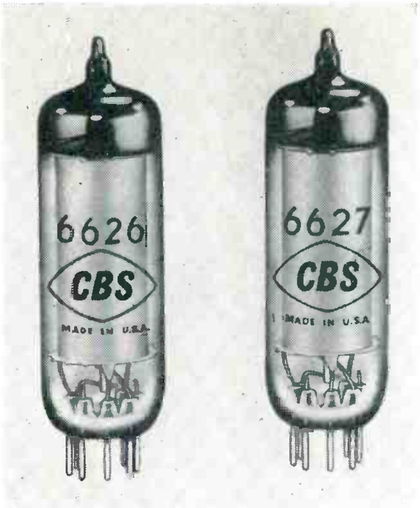
The price of the Fisher AM-80 is $139.50. Blond or mahogany cabinets are available at $14.95.

CBS-HYTRON VOLTAGE-REGULATOR TUBES

A pair of versatile gaseous voltage-regulator tubes featuring improved reliability has been developed by CBS-Hytron, a Division of Columbia Broadcasting System, Inc.

The new tubes, known as the 6626 and the 6627, have important applications in the military and high-quality commercial products fields. It is claimed that they offer longer life, improved dark starting, and elimination of voltage shifts. The tubes are recommended in such critical applications as voltage reference circuits where extreme stability and repeatability are vital factors.

The problem of starting when tubes are shielded from all sources of radia-



CBS-Hytron's new VR tubes.

tion is said to have been overcome in the 6626 and 6627 by incorporating a small amount of radioactive nickel in the starting electrode. The amount of radiation, while adequate to insure uniform starting voltages, is less than that obtained from an ordinary radium watch face.

Complete data are available on these new tube types in CBS-Hytron Bulletins E-253 and E-254.

VOLUME EXPANDER

A small accessory known as the *Hi-Fi Volume Expander* is being manufactured by Lincoln Records, Inc. The Volume Expander is said to create dramatic effects by restoring the original concert-hall volume range from whispered, soft passages to brilliant, powerful swells. The

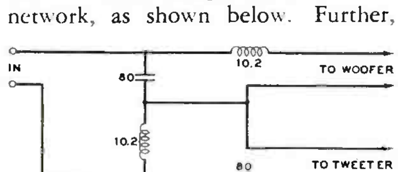
Continued on page 48

ERRATA

Errors seem to be inevitable in magazine articles, no matter how carefully they are checked and double checked. Here are several that have been brought to our attention:

In "A High-Quality Microphone Mixer", by J. Gordon Holt (December, page 24), input connections are incorrect as shown in the schematic diagram. In each case pin No. 1 should be grounded, *not* pin No. 2.

In "Carnegie Hall on Wheels", by Brooking Tatum (January, page 24), the schematic diagram for the crossover network is incorrect. Mr. Tatum uses a series-configuration, 16-ohm network, as shown below. Further,



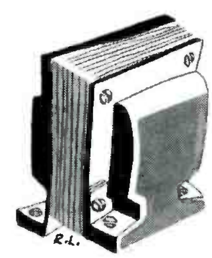
the bass enclosure isn't fastened to the wardrobe as stated.

Sorry!

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
cathode ray tube	CRT
capacitance	C
characteristic impedance	Z_o
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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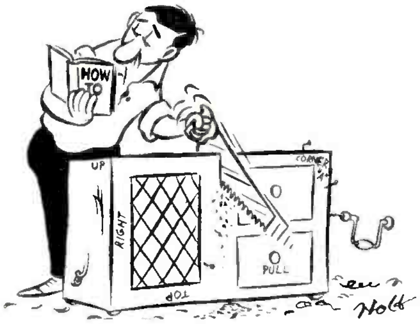
Audiophile's Bookshelf



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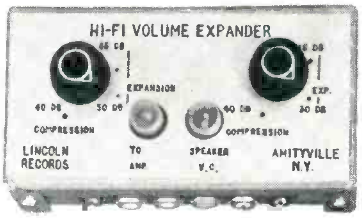
AUDIONEWS

Continued from page 45

manufacturer states that the Hi-Fi Volume Expander restores to recordings original volume range which has been compressed to prevent shattering and overmodulation.

This new accessory may be installed easily in any system by anyone, using



Lincoln volume compressor/expander.

only a screw driver. Since not all recordings contain the same amount of compression, provision is made to set volume expansion from 30 to 45 db. Once set, the Hi-Fi Volume Expander operates automatically.

In addition to its use as an expander, the unit may be used as a compressor so that, in home recording, the same method as is used in recording studios will prevent overly loud passages from spoiling the recording. The manufacturer states that volume compression up to 40 db can be obtained.

MUNSTON HI FI AMPLIFIER

A new 10-watt amplifier, the Munston *Maestro,* has been announced by Munston Manufacturing Co., Beech St., Islip, N. Y. The new amplifier is compact in design and features continuously variable equalization and calibrated tone controls. Manufacturer's specifications for the Maestro are as follows:

Harmonic Distortion: 0.5% at full rated output.

Frequency Response: 20 to 20,000 cps, ±½ db.

Hum & Noise: 55 db below full rated output.

Equalization: continuously variable.

Input Sensitivity: 10 mv and 1v

Maestro simplified amplifier-control.

on phono; ½v or more on tape, auxiliary, and tuner.

Phono Tone Control: Bass: —5 db to +27 db at 20 cycles; Treble: +7 db to —27 db at 20,000 cycles.

Tape Tone Control: Bass: —15 db to +13 db at 20 cycles; Treble: +10 db to —18 db at 20,000 cycles.

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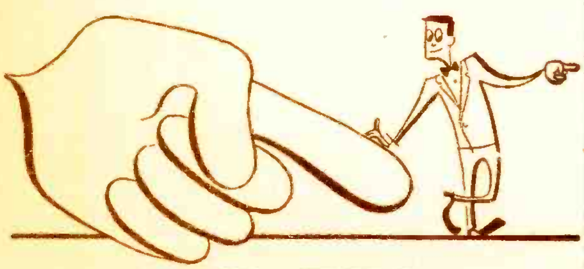

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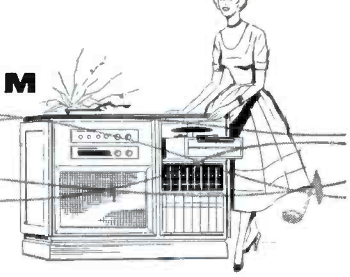

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❶ Heathkit FM TUNER KIT

Features brand new circuit and physical design. Matches WA-P2 Preamplifier. Modern tube line-up provides better than 10 uv. sensitivity for 20 db of quieting. Built-in power supply.

Incorporates automatic gain control—highly stabilized oscillator—illuminated tuning dial—pre-aligned IF and ratio transformers and front end tuning unit. Uses 6BQ7A Cascode RF stage, 6U8 oscillator—mixer, two 6CB6 IF amplifiers, 6AL5 ratio detector, 6C4 audio amplifier, and 6X4 rectifier. Shpg. Wt. 7 Lbs.

MODEL FM-3 **$24.50**

❷ Heathkit 25-Watt HIGH FIDELITY AMPLIFIER KIT

Features a new-design Peerless output transformer and KT66 output tubes. Frequency response within ±1 db from 5 cps to 160 Kc at 1 watt. Harmonic distortion only 1% at 25 watts, 20-20,000 cps. IM distortion only 1% at 20 watts. 4, 8, or 16 ohms output. Hum and noise, 99 db below rated output. Uses 2-12AU7's, 2-KT66's and 5R4GY. Attractive physical appearance harmonizes with WA-P2 Preamplifier. Kit combinations:

W-5M AMPLIFIER KIT: Consists of main amplifier and power supply, all on one chassis. Shpg. Wt. 31 Lbs. Express only. **$59.75**

W-5 COMBINATION AMPLIFIER KIT: Consists of W-5M amplifier kit *plus* Heathkit Model WA-P2 Preamplifier kit. Shpg. wt. 38 Lbs. Express only. **$79.50**

❸ Heathkit HIGH FIDELITY PREAMPLIFIER KIT

Designed specifically for use with the Williamson Type Amplifiers, the WA-P2 features 5 separate switch-selected input channels, each with its own input control—full record equalization with turnover and rolloff controls—separate bass and treble tone controls—and many other desirable features. Frequency response is within ±1 db from 25 to 30,000 cps. Beautiful satin-gold finish. Power requirements from the Heathkit Williamson Type Amplifier. Shpg. Wt. 7 Lbs.

MODEL WA-P2 **$19.75**

❹ Heathkit Williamson Type HIGH FIDELITY AMPLIFIER KIT

This amplifier employs the famous Acrosound TO-300 "Ultra Linear" output transformer, and has a frequency response within ±1 db from 6 cps to 150 Kc at 1 watt. Harmonic distortion only 1% at 21 watts. IM distortion at 20 watts only 1.3%. Power output 20 watts. 4, 8, or 16 ohms output. Hum and noise, 88 db below 20 watts. Uses 2-6SN7's, 2-5881's and 5V4G. Kit combinations:

W-3M AMPLIFIER KIT: Consists of main amplifier and power supply for separate chassis construction. Shpg. Wt. 29 lbs. Express only. **$49.75**

W-3 COMBINATION AMPLIFIER KIT: Consists of W-3M amplifier kit *plus* Heathkit Model WA-P2 Preamplifier kit. Shpg. Wt. 37 lbs. Express only. **$69.50**

❺ Heathkit Williamson Type HIGH FIDELITY AMPLIFIER KIT

This is the lowest price Williamson type amplifier ever offered in kit form, and yet it retains all the usual Williamson features. Employs Chicago output transformer. Frequency response, within ±1 db from 10 cps to 100 Kc at 1 watt. Harmonic distortion only 1.5% at 20 watts. IM distortion at rated output 2.7%. Power output 20 watts. 4, 8, or 16 ohms output. Hum and noise, 95 db below 20 watts, uses 2-6SN7's, 2-5881's, and 5V4G. An exceptional dollar value by any standard. Kit combinations:

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W-4A COMBINATION AMPLIFIER KIT: Consists of W-4AM amplifier kit *plus* Heathkit Model WA-P2 Preamplifier kit. Shpg. Wt. 35 lbs. Express only. **$59.50**

❻ Heathkit 20-Watt HIGH FIDELITY AMPLIFIER KIT

This model represents the least expensive route to high fidelity performance. Frequency response is ±1 db from 20-20,000 cps. Features full 20 watt output using push-pull 6L6's and has separate bass and treble tone controls. Preamplifier and main amplifier on same chassis. Four switch-selected inputs, and separate bass and treble tone controls provided. Employs miniature tube types for low hum and noise. Excellent for home or PA applications. Shpg. Wt. 23 Lbs.

MODEL A-9B **$35.50**

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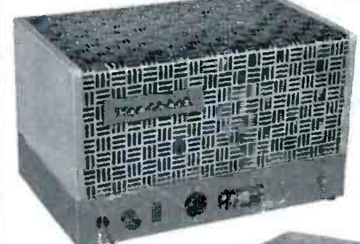

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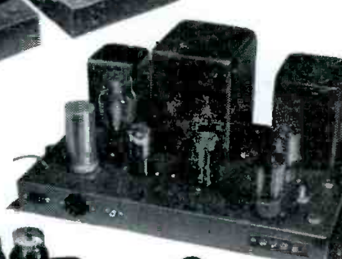

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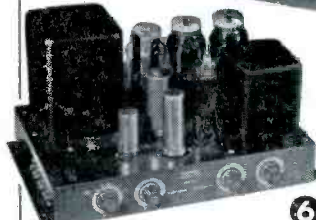

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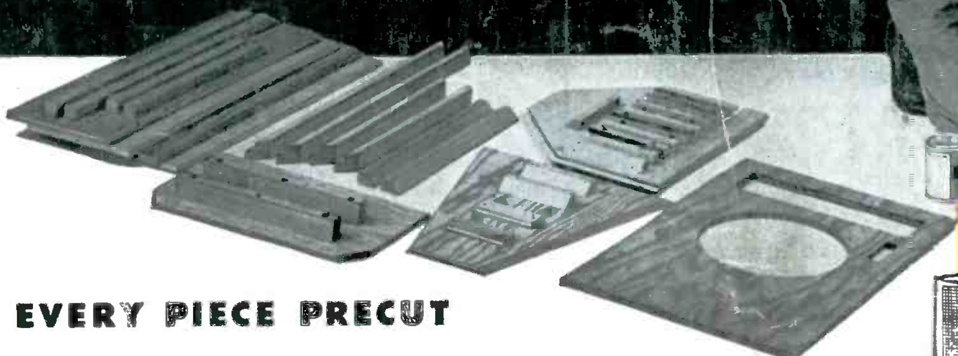

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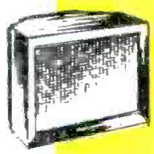
THE ARISTOCRAT KIT. Folded-horn corner enclosure designed for 12-in. speakers and separate 2 and 3-way systems. For use with Electro-Voice SP12 or SP12B coaxial speakers, 12TRX or 12TRXB triaxial reproducers, and 108, 111 2-way and 108A, 111A 3-way systems. Smooth reproduction down to 35 cps, with remarkable purity and efficiency. Finished size: 29⅝ in. high, 19 in. wide, 15¾ in. deep. Shpg. wt. 37 lb.

Model KD6 Net, $36.00

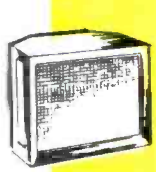
THE BARONET KIT. Phenomenal reproducer for such small size. This folded-horn corner enclosure is designed for use with E-V Model SP8B 8-in. Radax speaker. E-V T35 or T35B Super Sonax UHF driver can be added for a separate two-way system. Finished size: 23 in. high, 14 in. wide, 13 in. deep. Shpg. wt. 24 lb.

Model KD7 Net, $24.00



THE EMPIRE KIT. Economical, folded-horn enclosure for use in a corner or flat against one wall. Designed for 15-in. speakers and separate 2 and 3-way systems. Particularly effective when used with SP15B coaxial speaker, 15TRXB triaxial reproducer, or 116 2-way or 116A 3-way system. Recommended components for Regency kit may also be employed. Finished size: 29⅝ in. high, 32 in. wide, 16 in. deep. Shpg. wt. 45 lb.

Model KD5 Net, $48.00



THE REGENCY KIT. Most popular lowboy style folded-horn enclosure that can be used in corner or flat against one wall. Improves the bass range and response of any 15-in. speaker. Makes an outstandingly efficient reproducer when used with E-V SP15 coaxial speaker, 15TRX triaxial reproducer or 114A 2-way or 114B 3-way system. Finished size: 29⅝ in. high, 33½ in. wide, 19 in. deep. Shpg. wt. 70 lb.

Model KD4 Net, $69.00

THE PATRICIAN IV KIT. An interior assembly kit for those desiring the finest. This augmented design of the Klipsch corner folded-horn bass section delivers an added full octave of bass. Designed for use with E-V Model 103C Patrician IV four-way driver components. For built-in installations or to be decorated as you choose. Finished size: 57½ in. high, 34½ in. wide, 26⅞ in. deep. Shpg. wt. 150 lb.

Model KD1 Net, $89.00

THE GEORGIAN KIT. An interior assembly kit that creates authentic Klipsch indirect radiator type corner folded-horn bass section for 15 in. 4-way speaker system. Exceeded in range only by the Patrician IV. For use with E-V Model 105 or Model 117 package of 4-way driver components. For built-in installations or to be decorated as you choose. Finished size: 38½ in. high, 26¾ in. wide, 22½ in. deep. Shpg. wt. 88 lb.

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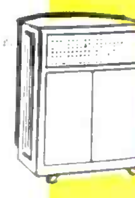

THE CENTURION KIT. Four-way system folded-horn, corner enclosure. Uses exclusive E-V "W" type single-path indirect radiator for propagation of extended bass. Sealed cavity behind 15 in. low-frequency driver cone promotes superlative transient response, subdues cone excursions, lowers distortion. For use with E-V Model 105 or Model 117 package of driver components. Finished size: 42⅛ in. high, 29 in. wide, 22½ in. deep. Shpg. wt. 75 lb.

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