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# audiocraft

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

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

**Abraham B. Cohen** is engineering manager for University Loudspeakers, Inc., which may explain part of his obvious enthusiasm for the Classic system. The rest springs from pardonable pride in the Classic's fine performance, in which Mr. Cohen had a hand also. He has written many articles for technical as well as non-technical publications, including HIGH FIDELITY Magazine.

**Herman Burstein** lives in Wantagh, N. Y. Among other activities he is a consistent contributor of articles to the technical press, usually as lucid and well-written as this one.

**Coming soon** is a series of articles on amplifier design by Norman Crowhurst, whose name is familiar to most readers. We have a few installments on hand now, and can say that this is going to be a series you can't afford to miss: it covers everything you'll need to know to design your own amplifiers from scratch, without resorting to higher algebra. We hope to follow it with a series on preamplifier-control unit design.

Also on the fire is an article by Charles Fowler on working with printed-circuit boards; another by Richard D. Keller on distortion in all its forms, and its audible effects; articles on transistor devices by Rufus Turner; and another contribution by Glen Southworth, on audio trouble shooting without test instruments.



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# The Grounded Ear

*by Joseph Marshall*

## Here We Go Again?

I suppose most of us had assumed that the war of record speeds was long over, and that we could settle down to enjoy whatever speed and size record we preferred without worrying about obsolescence. It now appears that any hope of things being settled permanently was quite premature. Columbia Records has announced that it has begun producing records revolving at 16⅔ rpm, with grooves playable by a 0.3-mil stylus at 2 or 3 grams. These records are now available only with the so-called Highway Hi-Fi, sold as an extra on Chrysler vehicles, but the presumption is that they will soon be offered to the general market.

There have been 16-rpm recordings for some time, but they are limited-fidelity, speech-frequency recordings of the "talking-book" type. They are cut with the same cutter used for microgroove records and can be played back with the normal 1-mil microgroove stylus, provided that the changer or turntable used has the slow speed. The frequency response of these talking-book records is limited at the high end by the forward groove velocity, which is too low for the cutting needle to make the required cuts in a given length of groove. This could be corrected by using a smaller cutting stylus capable of making more transverse cuts per lineal inch of groove, and by using a smaller playback needle capable of following the much finer undulations. Theoretically, if the styli sizes are reduced in proportion to the reduction in speed, the HF response can be as good at the lower speed as it is at 33⅓ rpm. The playback stylus force must also be reduced, however, to reduce damage by the needle to the finer grooves, and the moving mass must be low. So far, Columbia is claiming a response only to 10,000 cps, which is below that of the better 33⅓-rpm microgroove recordings.

The advantage is a large increase in the amount of program material that can be cut on a disc of a given size. With the lower speed and the smaller stylus (which permits even closer groove-spacing) the playback time can be more than doubled. In fact, Columbia is talking of 45 minutes per side on a 7-inch disc, which is about 50% more than the usual maximum on a 12-inch disc at 33⅓ rpm. Most of us would favor this in theory, if only because records could presumably be cheaper and take less space. Unfortunately, as I have remarked previously, this is a world in which everything must be paid for. The new 16-rpm disc has some disadvantages too. Wow and rumble problems will be much more serious at the slower speed. Tracking will be more critical. It will take better arms to track at the 2 or 3 gram pressures, and it will be more difficult to keep the needle in the groove unless special measures, such as viscous damping, are employed. (The Highway Hi-Fi does have a viscous-damped arm.) There will be new problems of cartridge compliance and tracing distortion. The mass of the cartridge and arm will be more critical. The narrower groove spacing will mean a reduced stylus swing, and cartridges will have to be more sensitive; if they are not, the hum problem will be more severe. With narrower grooves and finer cuts on them, the effects of dust, lint, and granular structure in plastics will become a more serious problem, and records will have to be kept cleaner — not only to keep the dirt from filling up the grooves, but also to keep it from accumulating on and around the stylus to the extent that it will no longer track. This is already a problem of some proportion with the 1-mil needle on microgroove records.

It is doubtful that present changers could be modified easily to produce good results with both the older speeds and older pressures, and the new speed and lower pressure. Very likely the first answer will be an improvisation offering 2 arms, one for microgroove and 78-rpm records, the other for the new slow-speed records.

In other words, now at least the problems posed by the new records are very much larger than the advantages they promise. To be sure, one can look at this as providing still another challenge to the ingenuity of the audio industry; there is probably as much reason to suppose that in the end we will have better reproduction as there is to assume that any improvement won't be worth the chaos. But I, for one, hope that the industry will not lose its collective head this time, and that we can look forward to an orderly evolution rather than another revolution.

## The ½-mil Microgroove Stylus

While we're on the subject, there has been some talk about reducing the size of the microgroove stylus to ½ mil; Pickering is in fact offering such a needle as an accessory to its Fluxvalve cartridge. Other manufacturers will probably follow suit soon.

The promised advantage of the smaller needle is a better high-frequency response and a reduction in translation loss at high frequencies. Cutting styli are chisel-shaped, rather than circular; cuts of the very high frequencies are accomplished mostly with the narrow side edge of the needle rather than with the broad front edge. At frequencies above 10,000 cps the size of the cut is smaller than the radius of the 1-mil playback stylus and, instead of tracing them sharply, the circular 1-mil stylus hits only the high spots rather like a hammer handle drawn over the teeth of a hand saw. This not only reduces the output amplitude but distorts the waveform, producing sawtooth rather than sine or complex waves. The ½-mil needle is able to follow these small cuts much more faithfully with a lower translation loss and, indeed, will probably permit response above the 18,000 cps which is the practical limit for the 1-mil stylus on the outermost grooves. Ferranti does much the same thing at 78 rpm by using a stylus of elliptical cross-section — 2.5 mils wide to fill the wide groove from wall to wall, but only about ½ mil front-to-back, to better follow the smallest undulations of the highest frequencies.

This, too, offers its problems. The smaller stylus should operate with lower pressure, but if it misfits the groove it may cause much greater damage when vibrating in it. Not all cartridges will prove suitable for the smaller stylus: in some, the compliance will not be high enough; in others, the lower stylus force will increase distortion. Tracking ability must be good and resonances controlled closely. The smaller stylus will fit records with a V groove (like Cook's) better than those with the normal spade groove — one reason why Emory Cook has been a vociferous proponent of the smaller needle. In my opinion, for those interested in the highest approximation of the ideal of perfect reproduction, the ½-mil stylus offers far more promise than the 16-rpm speed, and the problems will be more worth solving.

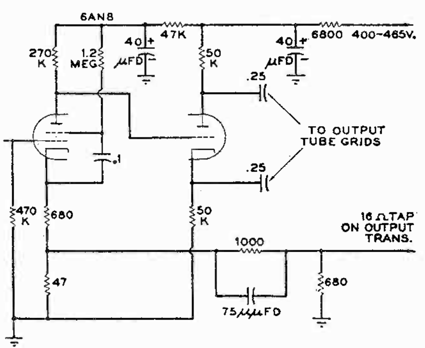
### Radioactive Pickups

Before leaving the subject of pickups, let me report that the Fenton Company, importer of various excellent European high fidelity and audio equipments, is offering the B&O magnetic and the PE crystal pickups with built-in strips of radioactive foil to neutralize the static charge on plastic records. This is so excellent an idea that I cannot understand why it took so long for somebody to do it. (Fenton, however, has patents on this; others should take this fact into account.) What's more, the idea works out fine. It takes a couple of plays to neutralize completely a highly charged record, but once this is done the charge is automatically held down.

### The Hafler Circuit

In the first issue of this series I made some kind comments about the Mullard circuit which, within certain limitations, had some most interesting possibilities. I'd like to report that Dave Hafler, co-developer of the ultra-linear circuit, and presently heading his own concern (the Dyna Company), has come up with a variation of the Williamson front end which possesses most of the virtues of the Mullard and is even simpler and cheaper to apply.

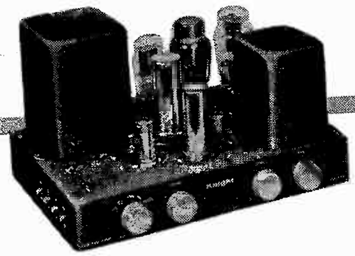
As the diagram shows, a single 6AN8 tube is used for the entire front-end circuitry, and to drive the output tubes directly. The 6AN8 has a medium-mu triode (rather like one section of a 12AU7) and a high-transconductance pentode. Here the pentode is used as the voltage amplifier and is direct-coupled to the split-load triode inverter. The combination produces as much gain as the



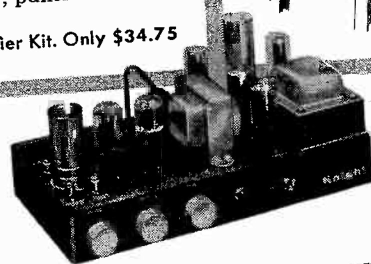

*Voltage-amplifier section of the Dynakit.*

*Continued on page 41*





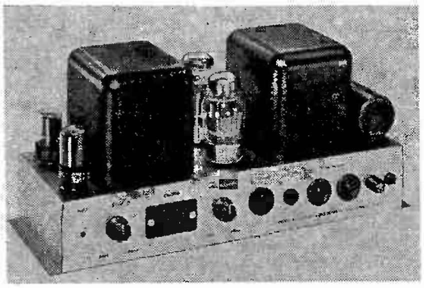




# Audionews

## NEW 100-WATT AMPLIFIER

The Interelectronics Corporation has just introduced a 100-watt amplifier known as the *Constellation.* This new unit uses only 4 tubes. Two 6550's, in a Class A circuit, are used as the power tubes.



*Interelectronics Constellation.*

Heavy-duty, metallic-plate rectifiers, with an estimated life of over 40,000 hours, are used in the power supply. All circuit components are encapsulated in Epoxy resin. A continuously variable loudspeaker damping control insures a full range of loudspeaker damping. More than 50 db of multiple negative feedback is used in the circuit, and distortion is stated to be less than 0.1% at 90 watts.

By means of a multiple-winding output transformer any loudspeaker impedance from 0.1 to 50 ohms can be matched exactly, according to the manufacturer. It is claimed that almost any number of loudspeakers of different impedances can be operated together with exact impedance matching and correct volume, and as many as 160 16-ohm loudspeakers can be operated in parallel without the use of line-matching transformers. For existing installations a 70-volt line connection is provided.

Additional information about the Constellation 100-watt amplifier may be obtained from the Interelectronics Corporation, 2432 Grand Concourse, New York 58, N. Y.

## SUBMINIATURE PAPER-DIELECTRIC CAPACITORS

Capcon announces a complete line of flat and round subminiature paper-dielectric capacitors. Type designations are *MF* for subminiature flats and *MR* for subminiature rounds. MF's and MR's are especially suited for the close stacking and assembly requirements of miniature electronic devices. Capcon flats, type MF, are ideally adapted for circuit applications where space is at an exceptional premium.

Both types MF and MR are non-inductively wound with the highest grade condenser-kraft paper (not metalized paper) and then thoroughly impregnated. The leads are securely anchored to the capacitor body. The manufacturer states that the units can be dipped into molten solder, as in modern printed-circuit assembly, without being affected in any way.

MF subminiatures include units from 3/8 in. in length, 1/16 in. in width, and 1/8 in. in thickness. MR subminiatures include units from 3/8 in. in length and 1/8 in. in diameter. The capacitance values for both range from .00005 $\mu$fd to 10.0 $\mu$fd. Working voltages range from 75 to 1,000 volts. Operating temperatures range from $-55°$ C. to $+100°$ C., without derating. Capacity tolerances available range from 0.5% to a standard tolerance of $\pm 20\%$. Capacitors are sealed to withstand any 95% relative humidity test. They will withstand severe mechanical and thermal shock and total immersion.

Complete information about type MF and MR subminiature capacitors are available from Capcon, Inc., 25 Willett St., New York 2, N. Y.

## BIFLEX SPEAKERS

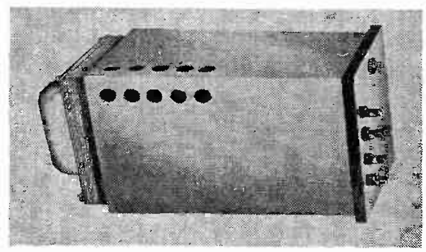
Altec Lansing Corporation has introduced 3 new loudspeakers known as *Biflex* loudspeakers. These speakers feature the use of multiple concentric compliances by means of which the cone has not only the compliance at the outer edge, but an additional compliance approximately midway down its slope. According to the manufacturer, this permits the entire cone to react to lower frequencies while the smaller area within the middle compliance can operate independently for the reproduction of higher frequencies. The 3 units with their manufacturer's guaranteed frequency range are the 8-inch 408A, 60 to 13,000 cps, $24.00; the 12-inch 412A, 40 to 13,000 cps, $45.00; and the 15-inch 415A, 30 to 13,000 cps, $60.00.

*Altec Lansing Biflex Speakers.*

Further information about Biflex speakers is available upon request from Altec Lansing Corporation, 161 Sixth Ave., New York 13, N. Y.

## NEW LINE OF VOLTAGE-REGULATED POWER SUPPLIES

Kemlite Laboratories have announced a new line of power supplies especially designed for the operation of radio tube filaments used in electronic equipment and experimental devices. Their voltage output varies inversely with the tube filament load so that filaments varying by 10% in resistance receive the same wattage. The output circuit is insulated from the case for 1,000 volts. The units use germanium rectifiers and are designed for chassis mounting.



*Kemlite power supply unit.*

Model 10: Output 6.3 v at 300 ma DC; regulation 3%; price $29.55.

Model 20: Output 5.6 v at 50 ma DC; regulation 3%; price $29.55.

Model 30: Output 6.3 v at 600 ma DC; regulation 2%; price $34.95.

Model 40: Output 18.9 v at 300 ma DC; regulation 3%; will operate three 6.3 v tubes in series; price $32.95.

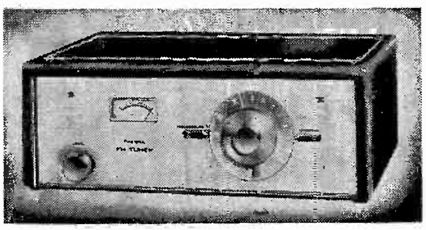
Model 50: Supplies high voltage for small amplifiers, photocells, and experimental devices. The unit contains a full-wave rectifier rated at 360 v and 20 ma. Regulation 3%. Voltage output from 360 to 100 volts is set by selection of the load resistor. Price $27.50.

Special units for either AC or DC output can be supplied with regulation up to 0.1% and are available with outputs up to 5 Kv.

For more detailed information, write to Kemlite Laboratories, 1813 North Ashland Ave., Chicago 22, Ill.

## NEW SCOTT UNITS

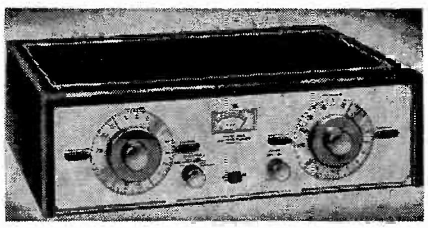
The new *Type 311* FM Tuner, announced recently by H. H. Scott, Inc., is a wide-band FM tuner designed to sell for under $100.00. Planetary drive tuning features an outer concentric knob which allows high-speed station selec-



*Scott Type 311 FM tuner.*

tion; an inner concentric knob has a vernier control for precise tuning to weak or distant stations. Technical specifications supplied by the manufacturer include: 3 $\mu$v sensitivity for 20 db quieting; 2 Mc wide-band detector and limiters; 80 db rejection of spurious cross-modulation responses by strong local signals; automatic gain control; low-impedance output permitting cables up to 70 ft. in length to the amplifier. The unit is equipped for multiplex. Consumer net price is $99.95.

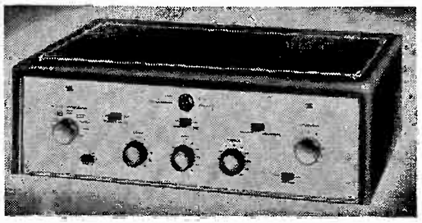
The *Type 330-A* is an AM-FM tuner. Design of the AM circuit permits reception of the full 10-Kc frequency range broadcast by better AM stations. Manu-



*Type 330-A AM-FM tuner.*

facturer's specifications include, for the FM section: 3 $\mu$v sensitivity for 20 db quieting, and 2 Mc wide-band detector. AM section specifications include: 1 $\mu$v sensitivity, 10 Kc whistle filter, extended frequency response to 10 Kc. The tuner incorporates special provisions for playback of pre-recorded tape and 2-speed planetary-drive tuning. Chassis design facilitates custom installation of the unit without removing the front panel or knobs, and without using an escutcheon. Consumer net price is $169.95.

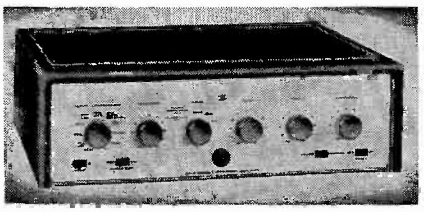
The *99-B Transcription Amplifier* is a new model of the 99, and offers a 22-watt power amplifier together with a flexible equalizer-preamplifier in one compact case. Among the controls on the unit are a 5-position record compensator and an adjustable rumble filter and



*Scott 99-B transcription amplifier.*

scratch filter. There are 2 magnetic inputs, switched on the panel, to allow the use of both record changer and turntable. Special provisions for playback of pre-recorded tape through the amplifier are provided. Separate bass and treble controls offer both boost and attenuation. Manufacturer's technical specifications include: 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%, and Class A circuitry throughout. Consumer net price of the 99-B is $99.95.

The *210-D "Dynaural" Laboratory Amplifier* is a complete 30-watt power amplifier with equalizer-preamplifier in



*210-D "Dynaural" laboratory amplifier.*

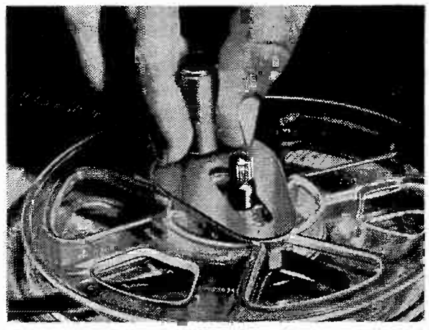
a single compact case. The amplifier features the "Dynaural" Dynamic Noise Suppressor and record distortion filter. The suppressor can be used with all inputs so that the reception of noisy AM or FM broadcasts may be improved. The unit has a 7-position equalization control. Unique features are provided for tape recording with 3 special inputs for recording and monitoring, and special provision for playback of pre-recorded tape through the amplifier. Two low-level magnetic inputs are provided with switching on the front panel. Continuously variable speaker damping control is provided together with an automatic loudness control with switching to remove loudness compensation. The 210-D is equipped with 3-channel treble and bass tone controls and a phono input-level adjustment. Low-level tube filaments are DC operated for minimum hum.

Manufacturer's technical specifications include: hum level 80 db below maximum output, power section is Class A throughout, harmonic distortion is less than 0.5%, first-order difference tone intermodulation less than 0.25%, frequency response is flat from 19 cps to 35 Kc. Consumer net price of the 210-D is $169.95.

Additional information about any of these Scott products may be obtained by writing to Hermon Hosmer Scott, Inc., 385 Putnam Ave., Cambridge 39, Mass.

## FLAHAN TAPE THREADER

The Flahan Tape Threader is a sturdy metal device which holds tape to the reel for the first few turns to get the winding started. After threading, it can be left in position to serve as a reel



*Flahan tape threader in action.*

crank or it can be slipped off. The threader can be used on any size reel up to 7 in. and fits all makes of recorders having standard reel spindles. It is also suitable for use on any standard 8-mm. movie reel.

The Flahan Tape Threader retails for 98¢. Additional information can be obtained by writing to the Flahan Company, 7517 Pelham Drive, Cleveland 29, Ohio.

## PORTABLE TAPE RECORDER

Amplifier Corporation of America has announced a new battery-operated, spring-motor, portable tape recorder known as the *Weathertite VU Magnemite.* A VU meter is incorporated in the unit to act as recording-level indicator, output-level indicator, and "A" and "B" battery meter. It also simplifies the correct setting of recording and playback levels to compensate for battery voltage changes. A high-impedance microphone input jack is placed on the front panel

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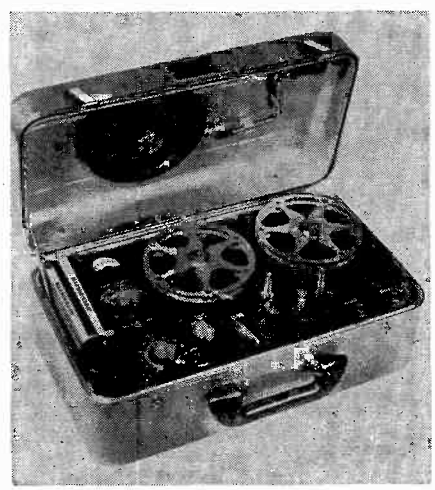


*Weathertite portable tape recorder.*

# TIPS FOR THE WOODCRAFTER

by George Bowe

## Joinery

There is no form of cabinetry that requires greater attention to joints than does high fidelity woodworking. Professional designers of enclosures emphasize that all joints must be accurately fitted and rigidly secured to assure the best possible sound reproduction. Not only do good joints make for better listening, but there is considerable satisfaction for the craftsman in creating them.

Joinery is one of the most treasured skills of an expert cabinetmaker. Derived from the word "join", it identifies the art of constructing by joining together pieces of wood. The finished project is no stronger than its assembled joints, but even a beginner can turn out rigid construction if he confines himself to the simpler types of joints which adequately meet the average requirements of hi-fi cabinet work. There are various power tools which are specifically designed for, or can be adapted to, the making of many types of joints. In subsequent articles we shall concern ourselves with their operation, but let's devote this discussion to the basics of joint construction with hand tools. A step-by-step description of 6 common joints in the making will include mention of the required tools.

*Butt Joints.* The simplest of all joints and the most frequently used is the plain

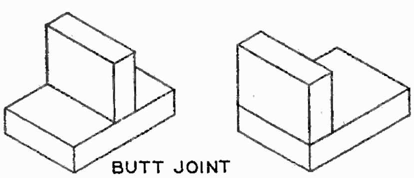



*Fig. 1*

butt joint in which the squared edge or end of one piece of wood is joined to the surface of another. This joint (Fig. 1) can be reinforced with nails, screws, glue, dowels, corner blocks, and with corrugated fasteners driven at an angle of 45°. Much of the interior construction in hi-fi enclosures can be accomplished with this joint. For a perfect fit the edge and surface to be joined must be perfectly square. Check carefully with a try square.

*Dado and Butt Joint.* The dado and butt joint, Fig. 2, is strong and neat in appearance. A dado is a channel cut *across* the grain of a piece of wood into

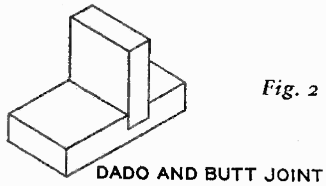



*Fig. 2*

which another piece is fitted. (Whenever a channel is cut *with* the grain it is called a groove.) This type of joint is generally used for shelves in cabinets or bookcases. The construction procedure is as follows:

1) Across the piece of wood to be dadoed, place the end of the piece to be housed in the dado.

2) With a knife, mark the width of the end; then, with a square, score lines on the face and edges of the wood where the dado will be cut.

3) Mark lines on edges to indicate the depth of the dado. (A marking gauge or a combination square can be used for this.)

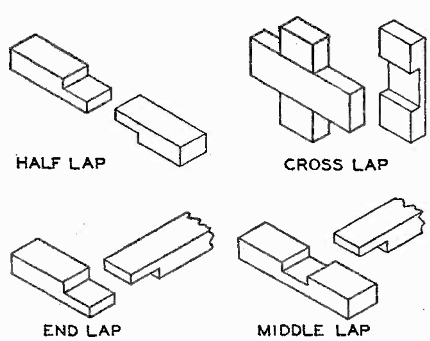
4) Place the wood in a vise and, with a back saw, cut to the marked depth.

5) Remove the waste wood between the cuts with a chisel.

*Lap Joints.* Whenever it is necessary to have 2 pieces of wood cross each other and still be flush on the surface, a lap joint is in order. Commonly used in making window screens, this joint has many other applications, including various types of framing and grill work. Essentially, there are 4 simple types of lap joint: half lap, cross lap, end lap, and middle lap; see Fig. 3. Construction methods of the half lap and end lap are the same. With the cross lap and middle lap, simply add the instructions for the dado cut. Here is the routine for making a half-lap joint:

1) Use squared stock of the same

*Fig. 3*




thickness, size, and shape.

2) Mark the stock so that the same amount of wood will be taken off both pieces.

3) A back saw can be used for removing the waste wood. If any extra trimming is necessary, finish with a chisel.

*Doweled Joint.* Dowels are round sticks of hardwood available in lengths of 3 ft. and in various diameters from 1/16 in. to as much as 3 in. They are used to reinforce butt joints, mitre joints, and other fittings. Sometimes they substitute for mortise and tenon joints, which is an indication of their strength. Accuracy is essential in making doweled joints and is readily achieved by using a "doweling jig" designed especial-

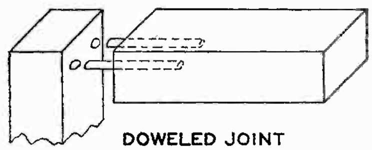



*Fig. 4*

ly for this work. The jig (a product of Stanley Tools) enables the user to bore dowel holes in the edge, end, or face of the wood with accuracy. It will take any thickness of material up to 3 in. When the jig is clamped to the work, the steel guide is automatically set to guide the bit properly while a bit gauge regulates the depth of the hole. The jig can also be used as a guide in mortising. A doweled joint is shown in Fig. 4.

Proceding *without* the doweling jig, here is a method for making a doweled joint:

1) Be sure both surfaces are true.

2) Clamp together side by side in a vise the 2 pieces of wood to be joined. The surfaces that are to be butted should be face up and flush with each other.

3) With a try square, mark lines across both pieces at the same time, to indicate locations of the dowels. On these lines locate the centers of the dowel holes by using a marking gauge.

4) Bore holes for the dowels. Be sure dowel and bit are of the same diameter and that the dowel is about ⅛ in. shorter than the combined depth of the 2 holes into which it will be glued.

5) The dowel should have a small groove running its entire length to permit excess glue to escape and thus pre-

vent splitting of the wood. The groove can be made with a saw.

6) Before applying the glue, make a trial assembly of the pieces to see if they fit. If the dowel holes do not align, fill the hole or holes out of alignment with a glued dowel. When it hardens, saw off the dowel flush, mark accurately, and bore a new hole.

7) When the work is ready for final assembly, apply glue to the edges to be joined and also inside the dowel holes.

8) Clamp the pieces together and leave until dry.

9) A brad hammered through the wood members across each end of the dowel will prevent the joint from coming apart.

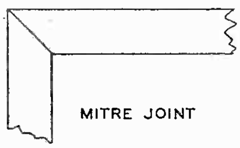



*Fig. 5*

*Mitre Joint.* The mitre joint, Fig. 5, is perhaps most known for its use in picture frames, but it has its own very definite place in hi-fi woodcrafting, too. The edges of cabinets, where exposed end grain would not be suitable, can be fitted together attractively with a mitre joint. It thus makes an ideal joint for building with plywood. Speaker grills can be outlined with a mitred frame, and cabinet doors can be treated the same way. A right-angle mitre joint is produced by making a 45° cut on each of the 2 joining pieces. However, frequently other shapes are desired and this governs the angle of cut. Among the reinforcements that can be used with a mitred joint are nails, screws, dowels, corrugated fasteners, and a spline. The accuracy of the angle cut determines the precision of the joint. For small work a back saw does nicely, especially if used with a mitre box. The mitre box can be a homemade affair or, better yet, a manufactured product of metal with lasting accuracy. If the work is large, it would be wise to have it mitred by a friend with power equipment or at a local woodworking shop.

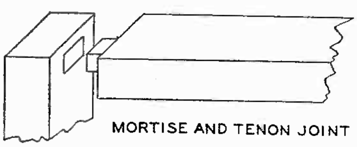



*Fig. 6*

*Mortise and Tenon Joint.* The mortise and tenon is one of the most commonly used joints in woodworking and finds special application in cabinets, tables, desks — in fact in any project which must withstand hard usage. The mortise is the hollowed-out portion, as shown in Fig. 6, and the tenon is the neck of

*Continued on page 39*

# SOUND SERVICING

by Irving M. Fried

## Amplifier Bass Stability

One of the most important — yet least understood — factors in audio quality is that of stability in the power amplifier. While it is true that certain preamplifier circuits are unstable, the percentage of unstable preamplifiers to unstable power amplifiers currently available is quite low.

The problem, of course, is intensified by feedback circuitry, and it is for this reason that many power amplifiers, which all use heavy feedback, are either absolutely unstable or conditionally unstable. A further complication is deterioration. An amplifier that, when new, is too close to instability — one that does not have wide phase safety margins to start with — will almost certainly become unstable as parts age and change value.

How can you tell whether that screech or muddy bass you have just noticed is caused by an unstable amplifier? In this issue we will consider the problems of bass instability, which are by far the easier problems to diagnose and cure.

Telltale symptoms of bass instability or marginal stability are the following: 1) excessive rumble from turntables that seem normal on other amplifiers; 2) putt-putt sounds in the speaker whenever a transient (as from a selector switch) occurs in the system; 3) slow weaving of low-frequency speaker cones even when no input signal is present; 4) ill-defined, muddy bass, that seems to drone on and on; 5) too much bass — you have to turn down the bass control for normal listening.

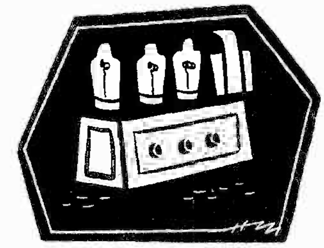
Cures require at least some technical knowledge and test equipment. First you must have a high-resistance AC voltmeter, or a multi-tester having high input impedance on the AC volts ranges. Then you can chase down the trouble using the "open-circuit" test, as follows:

Connect the AC voltmeter to your amplifier output, with no other load (i.e., no speaker or resistor connected) and no input signal. Turn it on, and wait several minutes for all parts to warm up. If the needle rises and stays up, you have high-frequency oscillation (more on this next month). If the needle of the voltmeter flicks up and down the scale, you have an amplifier that is completely unstable at the bass end — and a real problem.

If the needle remains at rest the next step is to "pulse" the amplifier, or give it a shock excitation. One method is to touch a screwdriver to the input plug for a brief moment. If your amplifier is absolutely stable in the bass range, the needle will flick up once, corresponding to the input pulse, and then return to zero. A conditionally stable amplifier will bounce twice, or more, then gradually come back to rest. But if the amplifier is a bit closer to complete instability, you may notice the needle flicking up and down continuously.

Why do you check "open circuit", instead of into rated impedance? The reason is quite simple: no loudspeaker has its rated impedance at the bass resonance frequency; most appear to the amplifier more like an open circuit — with an impedance many times the rated value. Therefore, an amplifier that "pulses" well on a resistive load may actually, under conditions of loudspeaker load, be quite unstable — and quite muddy.

What do you do next to correct the instability if it exists? The following suggestions may be of help. These are rough-and-ready cures, not always effective, and certainly not workable if the problem is either very bad original design, severe components deterioration, or a poor output transformer. A well-equipped laboratory, and a thorough knowledge of the mathematics of phase shift, feedback factor, Nyquist diagrams,



and so on would be even more helpful. But lacking all this, try the following:

First step, of course, is to check decoupling of your high-voltage (B+) supply, from stage to stage. Certain economy kits and ready-made amplifiers don't have enough resistors and capacitors, even when new. If the manufacturer has cut close to save money, and your capacitor values happen to be low, you can be in lots of trouble. For instance, if you have one of the popular Williamson kits in which the driver stage isn't decoupled from the output stage, put a 150-ohm, 5-watt resistor in series with the driver plate supply, with a 16-$\mu$fd, 500-volt capacitor to ground. You may find it much easier then to stabilize your putt-putts.

Also, using a 16-$\mu$fd, 500-volt capacitor with its minus lead grounded to the chassis, check stability by pulsing while moving the plus lead of the capacitor from decoupled point to decoupled point, working from the highest-voltage end of the power supply to the lowest. If at one point the amplifier suddenly becomes stable, solder in the extra capacitor and begin to enjoy better listening.

When you are fairly sure that your decoupling is as good as possible, move on to interstage coupling networks. What you are after here, to digress a bit, is to cut the low-frequency response of one coupling system between the plate of an early stage and the grid of the next, so that its response will be far down before the other stages shift phase enough to make the total combination regenerative. Let us take a typical Williamson-type circuit, and apply this in practice. Incidentally, your amplifier may be one of these even if it is not so labeled.

The typical Williamson circuit has 4 stages, with direct coupling between the first 2. There is no phase shift with direct coupling. The places where phase shift can be introduced are, then, the coupling capacitors between the split-load phase inverter and the 6SN7 driver, those between the driver and the output tubes, and in the output transformer. Since you have no control over the phase shift in the output transformer (except by changing to a better unit), and since the phase shift characteristics of the transformer vary under different conditions of load, you must design so that the other 2 sources of phase shift have cutoff frequencies as far away from each other as possible — without reducing bass response and/or increasing distortion.

I have found the following values of capacitor and grid resistor to work quite well in stabilizing otherwise bad units: Between driver and output stage, 0.25 or 0.5 $\mu$fd and 150 or 100 K; between phase splitter and driver stages, .02 $\mu$fd and 470 K. Anyone familiar with frequency discriminating networks will understand how these values tend to stagger the time constants, cutting off low-frequency response before the driver stage.

If this won't work, you might consider next replacing the output transformer. Certain units just won't maintain response at low enough frequencies for stable use in a feedback

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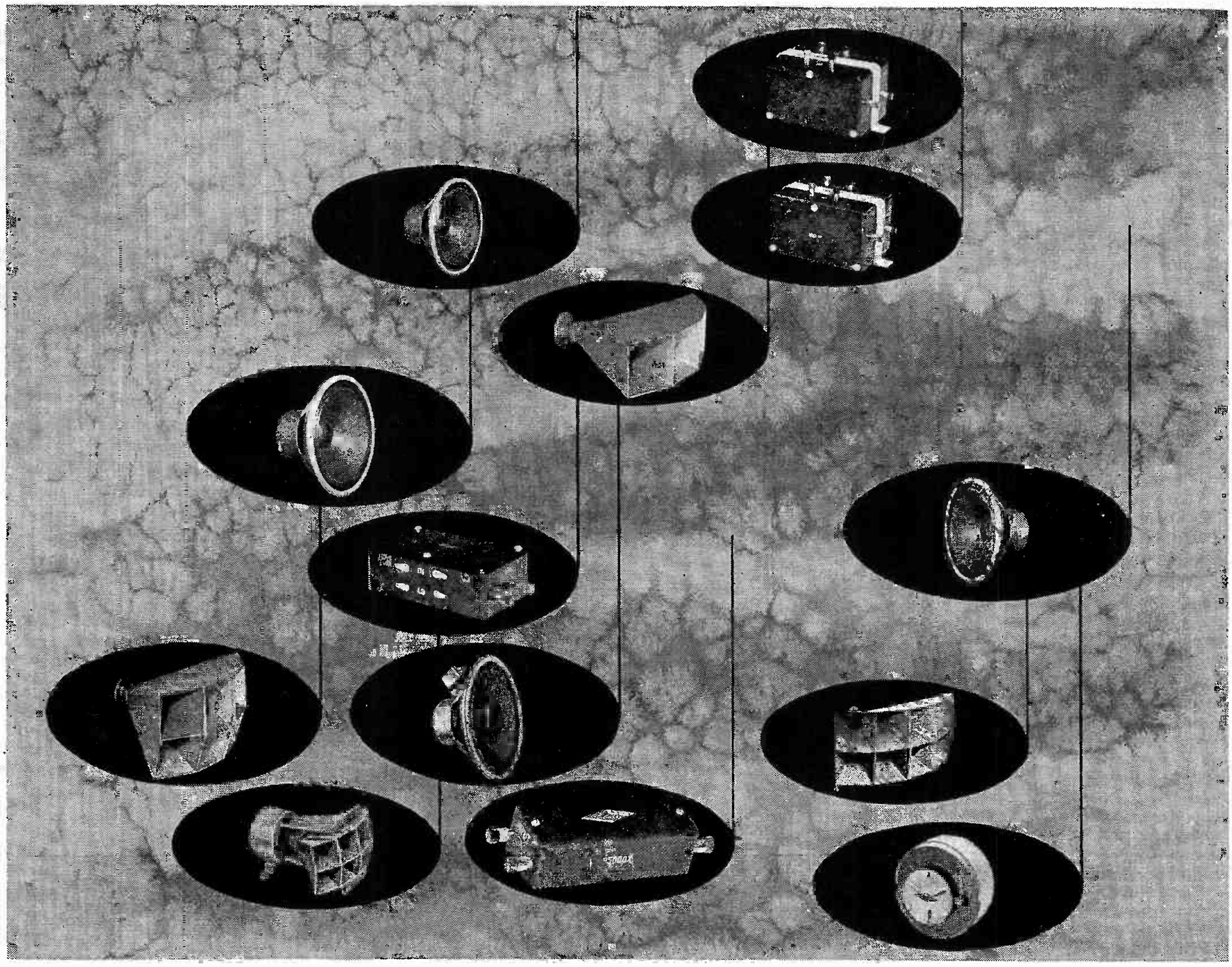

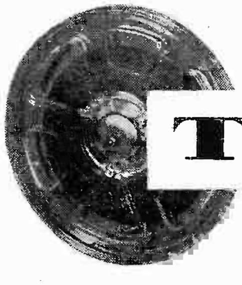

# Tape News and Views

by J. GORDON HOLT

## The Prospective Buyer

There still seems to be a great deal of confusion among high fidelity enthusiasts about just what constitutes the ideal qualifications of a tape recorder for home use. This was pointed up not long ago in a letter sent to Audio Forum, HIGH FIDELITY Magazine, in which the correspondent asked whether he should buy a stereophonic tape recorder or a standard monaural one. He added that the recommendation given him would determine what he purchased for his home.

This was putting Audio Forum on the spot, since such an inquiry was almost like asking whether he should buy a sedan or a convertible automobile. But to tell the correspondent simply that "it is up to him" wouldn't help to clarify things much, either. If he had known the factors that should determine his choice, he patently wouldn't have felt the need to inquire about it.

So what should he have been told? That stereophonic recording is all right for those who like it, but monaural recording is better for those who don't care to get too involved? Or that stereo recorders are usually more expensive than monaural recorders, except when they are of equivalent quality, in which case they are more so? Not much help, since he's probably never had the opportunity to use either type, and hasn't formed any preferences.

Actually, his query isn't too different from those of other prospective recordists: should I use 7.5 or 3.75 ips speed? Or should it be 15 ips *and* 30 ips? What about 30 ips? Twin-track or full track? Automatic reversal or manual changing of reels from one track to the other? Very high high-speed rewind or moderately fast rewind? How many operating adjustments should a recorder have? Bias frequency? Bias current? Record equalization? Playback equalization? Adjustable head alignment? Low impedance or high impedance? What about quality? How good is acceptable? Good? Professional-quality? How about signal-to-noise ratio? What frequency range? How much speed regulation? And so on.

Now who's confused?

This business of buying a tape recorder is really much more simple for the man-in-the-street with the tin ear and the pedestrian taste in reproduced sound. He buys a little box from a department store, takes it home, and he's in business. He has no specific requirements to speak of. He wants a recorder, so he buys a recorder.

But let's take a microscopic look at the shrewd, sonically-sensitive sound enthusiast with Requirements. Say he has a typical hi-fi setup: magnetic cartridge of moderate quality, with diamond stylus; a reasonably ambitious collection of records, both LP and 78, and a 3-speed record changer that revolves them at tolerably constant speed; an amplifier with tone controls and 1 or 2 equalizer knobs; a speaker that either has or has not "presence", depending upon how far away from a live orchestra he likes to sit; and maybe a tuner of some kind.

He has noticed a few discouraging facts: that records wear out at a rate which is directly proportional to the number of times he plays them; that some of the best sound that issues from his system comes from radio broadcasts; that he can't afford all the records he would like to own; and that half of his present record collection has been superseded by better recordings or better performances.

This gentleman is ready for a tape recorder. He wants a unit that is at least as good as the rest of his equipment, that will make it possible for him to dub from recordings to tape at reasonable cost, and that won't cripple his financial standing in the community when he goes to pay for it. Specifically, what are his requirements?

Tape prices being what they are, he is going to have to be able to record for a total of at least an hour on a single 7½-inch reel if he is to save money in dubbing from discs. This means either a full-track recorder operating at 3.75 ips, or a half-track one operating at 7.5 ips. The ratio of frequency range to tape speed for most current home recorders is about 1.6 Kc per ips tape speed. This means that at 3.75 ips he might expect a high-frequency limit of about 6,000 cps if he's lucky; no matter which way you slice it, that isn't good enough for high fidelity music reproduction. For good quality, then, he should choose a 7.5-ips machine (or a dual-speed 7.5- and 3.75-ips one if he wants to record operas off the air), with dual-track heads. This will give him a maximum of 48 minutes of uninterrupted recording time on one side of a reel of extra-play tape, or an hour on half-mil tape. Then he can flip the reels over and record for the same time on the other side.

The speed requirements of a recorder, then, will depend upon how much uninterrupted recording time the user wants, and upon what frequency range he will need for the recorder to be as good as the rest of his system. For speech recording, 3.75 ips will be quite adequate, but even speech quality suffers somewhat at this slow speed. Needless to say, 1.875 ips is useless for high fidelity applications, but what about the higher speeds? Despite what some Philistines claim, there *is* an improvement in sound quality when a good recorder is used at 15 ips. The difference between that and 7.5 ips may in some cases be very slight, but it exists. The 15-ips speed is used for professional recording purposes mainly because of its greatly increased ease of editing. At the higher speed, a word or a single syllable may be removed or transposed from a program with considerable ease, while at 30 ips, editing is simplified even further, and special equalization circuits may be added to the recorder to reduce tape hiss to a phenomenally low level. But consider the tape costs before running out and buying a 30-ips recorder.

A very popular recording platitude concerns the fact that you can't edit a half-track tape. Like most platitudes, this is only a half-truth (no pun intended), since the limiting factor is the second track. With both tracks recorded, the editing of one will obviously mutilate the other track — but if only one track is being used, editing is just as simple and straightforward as with a full-track recording. However, if you are inclined to fiddle with weird sounds and have a yen to play tapes backwards for the unearthly sounds they produce, get a full-track machine — you can't play a half-track one that way.

Getting back to our hypothetical audio enthusiast, who has decided so far on a dual-speed, half-track recorder for 3.75 and 7.5 ips, his next question concerns automatic versus manual tape switching. He has noticed that some recorders have an ingenious switching mechanism that changes the direction (and track) of a tape as soon as it reaches a small tab of metal foil attached near the end of the tape. This, he feels, would be excellent for recording full-length operas, non-stop, at top quality, but if his major interest is in orchestral music, which rarely lasts for more than 45 minutes, he may decide against it on the basis of the additional cost of such a unit compared with others of equivalent sound quality. So to the next consideration: shuttle speeds.

While this is largely a matter of personal taste, based on one's degree of patience with long-drawn-out and monotonous mechanical procedures, there does seem to be some evidence that an extremely high-speed rewind is bad for tape. It may tend to wind it too tightly, leaving it with large stresses between the layers that encourage print-through and longitudinal stretching. Any machine that takes 60 seconds or more to rewind a 1,200-foot reel of tape should be all right. Since most recorders now approximate this figure in rewind time, the choice is pretty wide.

The matter of available adjustments is rather a touchy one, since not all tape users are sufficiently well versed in the technical aspects of their machines to wish to fiddle with the screwdriver adjustments. As it works out, the number of screwdriver adjustments on a recorder is very often a measure of the unit's maximum performance capabilities. Tape coatings and component tolerances vary enough that, for absolute top quality, a recorder must be precisely adjusted to suit the tape being used with it. With any but the very best units the maximum quality is limited more by the design compromises and the component quality than by the tape; fine adjustments would then be wasted. On most home recorders it is next to impossible to tell the difference between one tape coating and another, unless one tape happens to be flagrantly bad, so minute differences simply don't exist as far as the user is concerned.

My own feeling is that there should be at least 2 adjustments on every recorder that is to be used in conjunction with a high fidelity system: an ultrasonic bias current adjustment, and provision for aligning the azimuth of the head gap.

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## TAPE NEWS

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Only a few recorders are still being made without variable head alignment, since a misaligned head will cause significant high-frequency losses from tapes that were not originally recorded on that particular machine. Bias adjustment, however, is something that isn't quite as common in home recorders, and since the bias current greatly influences the high-frequency response, distortion, and efficiency of erasure, it should be variable at least over a narrow range. At the other side of the ledger is the professional recorder which has practically every function adjustable and which takes a skilled technician to put in peak condition. As much as the perfectionist demands perfection in his tape recorder, I would be inclined to advise against his purchasing such a unit unless he has either a ready source of qualified, cut-rate maintenance (that's a tough combination to find) or a certain degree of technical proficiency himself.

Our hypothetical purchaser is next faced with the consideration of fitting his recorder into his system. If he wanted only a self-contained unit, with no frills like external high fidelity systems, he would not have to concern himself with impedance. But interconnecting cables are as much a part of a custom high fidelity system as are components, so the operating impedances of a recorder can't be ignored. The length of cable that can be used between a recorder and its associated equipment without loss of highs will depend upon how low are the impedances feeding the lines. A recorder's amplifier-output connection should have an effective source impedance of less than 10,000 ohms; otherwise, there will be highs losses with anything over 3 ft. of shielded cable. Ideally, the output should be from a cathode-follower or a terminated line transformer.

Finally, there is one small matter that has been quietly ignored up to the present. The question of choosing between a monaural and a stereophonic recorder was what prompted this whole discourse, but it is also the most difficult one to answer categorically. There is no doubt that good stereophonic recordings have an emotional impact and realism that is not often equalled by standard single-channel recordings. On the other hand, they have a lack of availability that isn't approached by standard sound sources either.

If a recordist has frequent access to stereophonic (also wrongly called "binaural" in this instance) broadcasts, or live musical performances that can be

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## Automatic Amplifier Shut-Off

The Garrard changer has built into it a switch which automatically shuts off the player at the completion of the last record; this switch can be used to turn off the amplifier as well. The only difficulty is that the amplifier, if tied in directly with the player motor, does not have sufficient time to warm up to reproduce the first part of the first record.

A simple single-pole, double-throw switch can be installed so that, in one position, it permits the amplifier to warm up while the player controls are being set and the records put in place. As soon as the player is turned on, the SPDT switch can be flipped to its other position so that the amplifier is supplied with power only through the Garrard auto-switch. When the last record is finished the Garrard switch will open, shutting off both player and amplifier.

The extra switch can be a plain toggle switch for manual operation, but if your player is mounted on drawer slides, or has a lid which is lifted and then shut while the records play, you should try using a micro-switch. This should be installed so that when the player is pulled out — or the lid lifted — the switch assumes the warm-up position. Then, once the player is started and is pushed back on its slides or the lid lowered, the switch automatically goes into the run position. With a micro-switch installed in this way, the auto shut-off for the amplifier becomes truly automatic.

Charles V. Thayer
Springfield, Vt.

## Testing a Home-Built Amplifier

Glen Southworth, in his article "Chassis Layout and Wiring", which appeared in the December issue of AUDIOCRAFT, tells of the hazards of trying out a newly wired home-built amplifier. For those who have access to a variable transformer, a safe way to make the initial test is to start out with zero volts on the power supply and then increase the voltage gradually to the normal 117. An AC ammeter in the circuit would be of value in showing up any dangerous trouble before the fireworks began. If an am-

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# EDITORIALS

CALIFORNIA, ever since the introduction of sound tracks for motion pictures, has been a leading center of interest in — and enthusiasm for — high quality sound reproduction. A rising star in the west-coast audio firmament is Oliver Berliner, currently doing business as Oberline, Inc. Mr. Berliner must be an excellent salesman as well as a fine engineer-author, for he has sold the Canadian Broadcasting Company a weekly program series entitled "Sound Thinking". He will conduct the programs himself; they will consist of discussions on high fidelity sound and interviews with well-known people in the industry.

CBC is to be commended also for being on its toes. The hi-fi spark has been lighted in Canada and is spreading in the same way as it did here some years ago. There is already a Canadian audio industry of considerable size which, as well as the listeners, will be grateful for the network show.

That brings up a puzzler. Why aren't there more programs of like nature available here? To be sure, many independent good-music stations and a few of the regional networks (very few) have hi-fi discussion and educational programs on a regular basis. But the major networks seem to make a principle of ignoring the subject. As a principle this would appear to be rather inexplicable, because the radio industry — sorely pressed by television — could only gain by an increased public awareness of the pleasure and relaxation that can be obtained from highly realistic sound *without* sight. Once this demand were achieved the nets would be in a better position to satisfy it than anyone else. And right now they are in a better position than anyone else to create the demand. Why don't they? It's claimed that the radio and television divisions of the networks (those that have both) are operated competitively, but this is hardly apparent in this case.

HI-FI discussion programs can, of course, give mis-information as well as information. A case in point is revealed in a recent letter from a reader. He credits a popular commentator on a good-music station with saying, "No one has yet proven that anything over 10,000 cycles is audible." The speaker went on to admit, reports our correspondent, that there *are* overtones above 10,000 cps, but that they are so weak they are drowned out by the lower frequencies.

Now, this is sheer nonsense, as anyone can tell you who has replaced a magnetic pickup cartridge cutting off at 10 Kc with one having smooth response to well beyond that. The range above 10 Kc heard by itself consists of mere gossamer whispers, usually of low intensity, that you would be justified in guessing wouldn't be missed among the relatively great masses of sound in frequencies below 10 Kc. And you would be right, in that a system with smooth response up that far can sound very good. When you make an A-B test, though, switching from one complete sound system that cuts off at 10 Kc to another one — identical otherwise — that doesn't, the difference is immediately obvious to anyone who doesn't have subnormal hearing. It isn't only changes in record surface noise or distortion that you hear, either; the test is even more dramatic with a high-quality microphone feeding the systems directly.

If the commentator had said that anything above 16, 17, or 18 Kc isn't audible in ordinary program material, he would have been on relatively safe ground. But even then he wouldn't have achieved invulnerability to attack from some quarters. There are many who insist that the *entire reproducing system*, from microphone to speaker, should ideally have response well outside the normal "audible" range. They base this opinion on either of 2 assumptions:

1) The ear, they say, is much more sensitive to phase shift within the hearing range than is commonly believed. In order to reduce phase shift within the range to insignificance it is necessary to have response substantially outside the range as a first requirement (there are other problems too, notably in achieving perfect phasing among the units of multi-speaker systems).

2) An alternative and unrelated argument is that the ear, being a non-linear device, serves as a detector of vibrations outside the normal range *when sounds within the range are being heard simultaneously*. In this way beat notes within the range are created which alter the character of normally heard sounds. If these accompanying ultrasonic harmonics are not reproduced they cannot, by ear difference-tone generation, be brought within the hearing range, and accordingly the reproduction cannot be completely realistic.

We'll take no stand on these theories, since we haven't had access to reproducing systems with the range required to make the necessary tests. But we think they are reasonable enough to warrant intelligent laboratory investigation. — R. A.

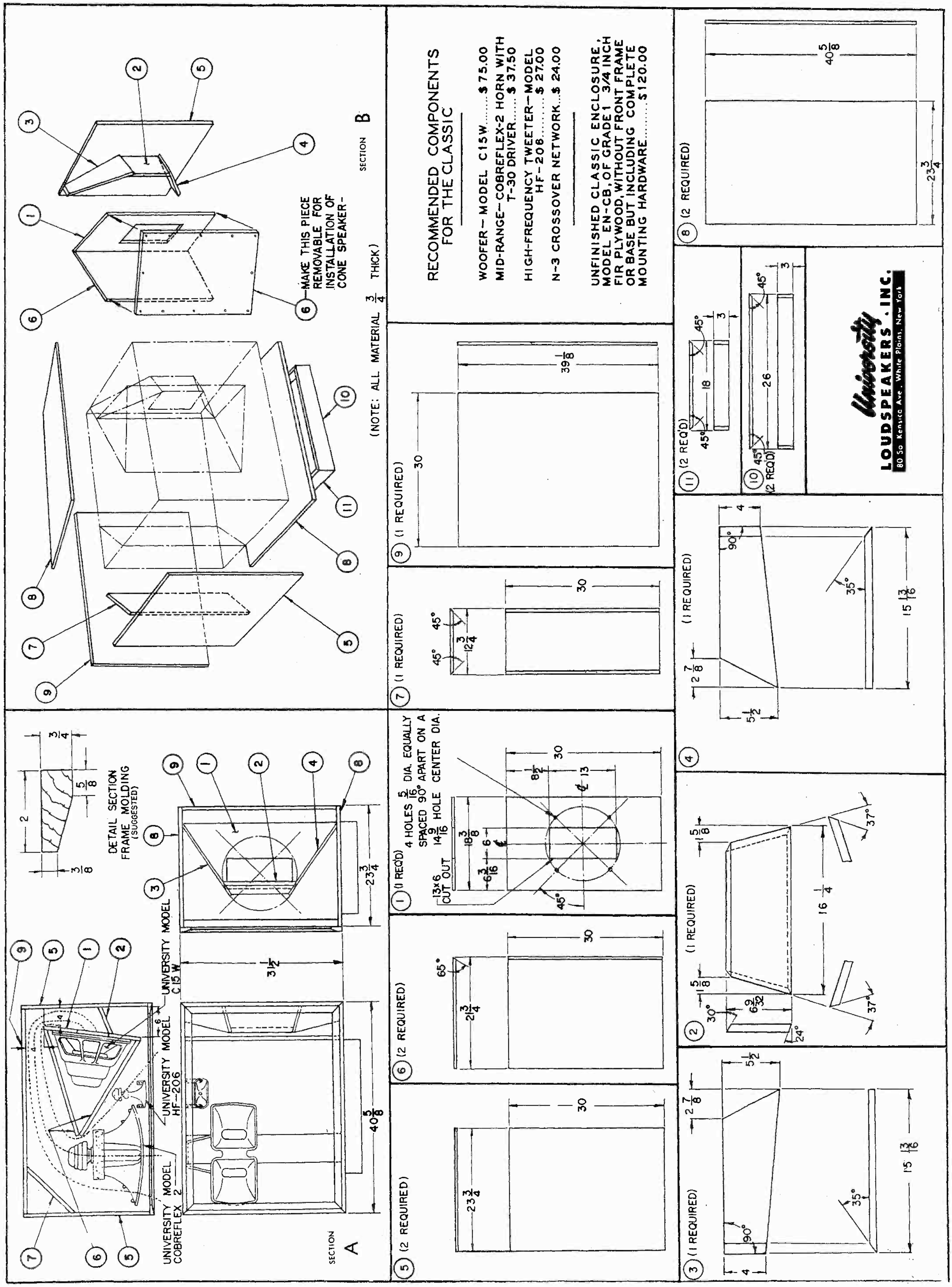
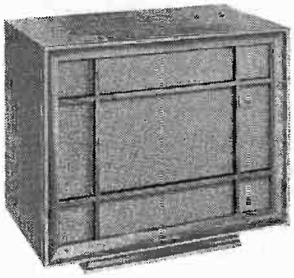



*Fig. 3. Dimensions of wood parts for building the Classic speaker system. Sections A and B show how these parts are assembled.*



# Making the University Classic

## by Abraham B. Cohen

THE University Classic is a true exponential folded-horn enclosure that provides for wide-range, high-efficiency operation. In addition, the Classic is designed for use with any type of room arrangement or decor. Although it is a true horn reproducer, it is not necessarily limited to corner placement. Like any enclosure it will work best if located in the corner of a room but, because of the integral side-wall construction by means of which the complete bass horn boundaries are built directly into the assembly, the Classic does not depend upon the walls of the room to complete its horn function. It may be used anywhere in the room, in the corner or against a flat wall, and still function as a true horn. Moreover, the enclosure is not limited to horizontal mounting; it can be used either as a "lo-boy" or as a "hi-boy". This is made possible by the unique method of mounting the enclosure base, as described later on. The whole assembly can be mounted either horizontally, as shown in Fig. 1, or vertically, as desired. Such adaptability in both attitude and position makes it ideally suited for use in any room.

Fig. 2 illustrates the internal construction of the folded bass horn as a self-contained integral unit. The high efficiency of this structure is obtained because the horn is energized by a woofer arranged as a compression driver; that is, the driver unit, a University model C15W 15-inch multi-impedance woofer, is mounted within a compression chamber. This is shown by the breakaway section, Fig. 2. The rear of the speaker is acoustically loaded to that optimum value which will bring the speaker resonance into the proper range to work with the natural horn cutoff frequency.

This differs considerably from the compromise in which only the rear of the speaker feeds a folded horn, and the front of the speaker is used for direct radiation. In the latter type of construction there must be some sacrifice in the low-frequency efficiency of the horn, because the loudspeaker driving the horn is not properly loaded on the back side; its efficiency must be kept down to match the directly radiated middle frequencies. In a true compression driver, the acoustic stiffness in back of the woofer serves to provide linearity of motion of the loudspeaker diaphragm and to improve the efficiency in the range of operation.

Because the Classic has a highly efficient bass horn, it is important to use compatible high-efficiency mid-range and tweeter units to balance the bass section. These compatible units, shown in Fig. 2 also, are the Cobreflex-2 horn with a T-30 driver unit and the HF-206 tweeter. Crossover points are 350 cps (between the woofer and mid-range horns) and 5,000 cps (between the Cobreflex horn and the high-frequency HF-206 tweeter). One very important requirement for treble reproduction is wide-angle dispersion, so as to diffuse high frequencies throughout the listening area. The patented reciprocating-flare horns used on both the Cobreflex-2 and the HF-206 are eminently successful in providing wide-angle dispersion of the middle and treble frequencies. Both are compression driven also, and have efficiencies comparable to that of the C15W in the bass horn.

It will be observed in Fig. 2 that the arrangement of the mid-range and tweeter units is suitable for horizontal mounting. If it is desired that the Classic be used vertically, they are secured to the mounting block shown on the left-hand wall of the enclosure. When the assembly is turned up on its shorter side, then, the flares of the Cobreflex and the HF-206 horns are in the proper horizontal direction.

Although the Classic's woofer horn may be considerably more complicated than a simple bass-reflex enclosure, the results obtained are fully deserving of the time and effort spent in correctly building and putting together the component parts. Fig. 3 shows in detail the various sections of the cabinet. Each element is individually dimensioned; when all are cut they can be easily fitted together. Section B is an exploded view showing the way in which the cabinet as a whole is assembled. In Section A

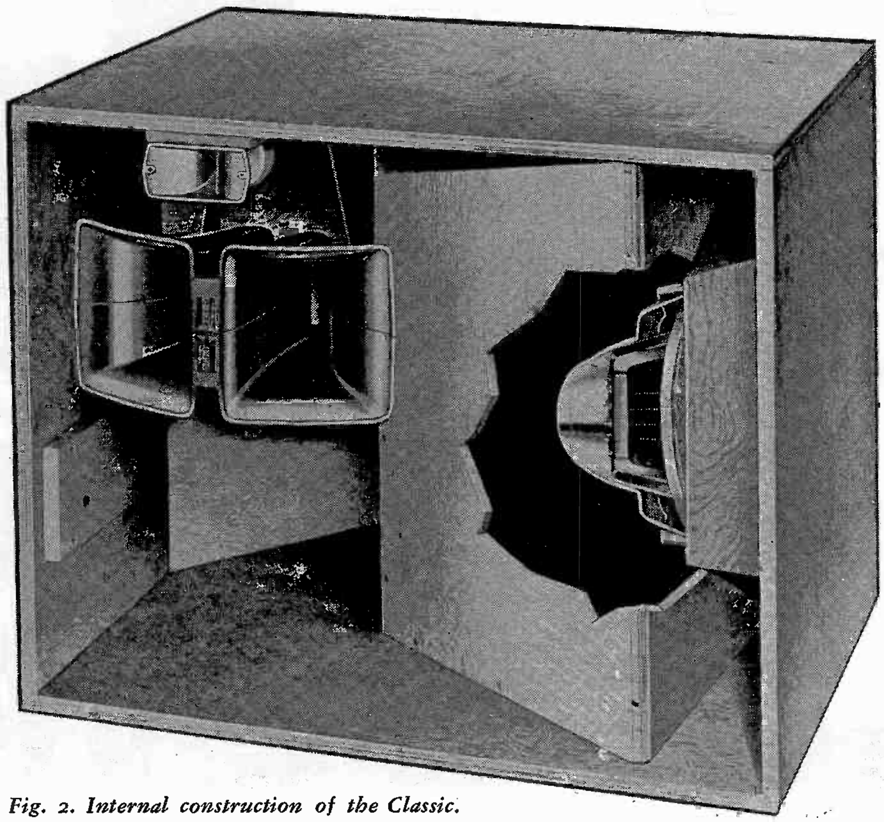


*Fig. 2. Internal construction of the Classic.*

the sound path for the woofer is indicated in dotted lines. Even though the enclosure itself is of moderate size, the acoustic length of the horn is actually considerably longer than the outside dimensions. A low-frequency horn may be folded on itself several times without any adverse effects on low-frequency reproduction. Folds will, of course, affect the high-frequency performance. Such effects are desirable, since they aid the electrical crossover network in separating lows from middles and highs — especially when these middles and highs are reproduced by other units.

The base of the cabinet is not screwed to the bottom of the enclosure but is fastened to the back, as shown in Fig. 4, so that the cabinet can be turned either way desired for lo-boy or hi-boy operation. The front panel of the compression-chamber panel, No. 6 (section A, Fig. 3), is made removable for providing access to the woofer mounting board (panel No. 1). Construction of this compression chamber is very important for proper operation of the loudspeaker. It should be made of plywood at least ¾ in. thick; it should be rigidly glued and screwed together and to the rest of the cabinet, so that there will be no acoustic leaks.

The cabinet as a whole should also be made of ¾-inch lumber, preferably plywood, and all seams and joints should be rigidly glued and screwed together so that there will be no loose edges to flutter or buzz from acoustic vibrations imparted to the panel. It is desirable in any horn construction to keep horn-panel vibration to a minimum; such vibrations deteriorate performance, re-

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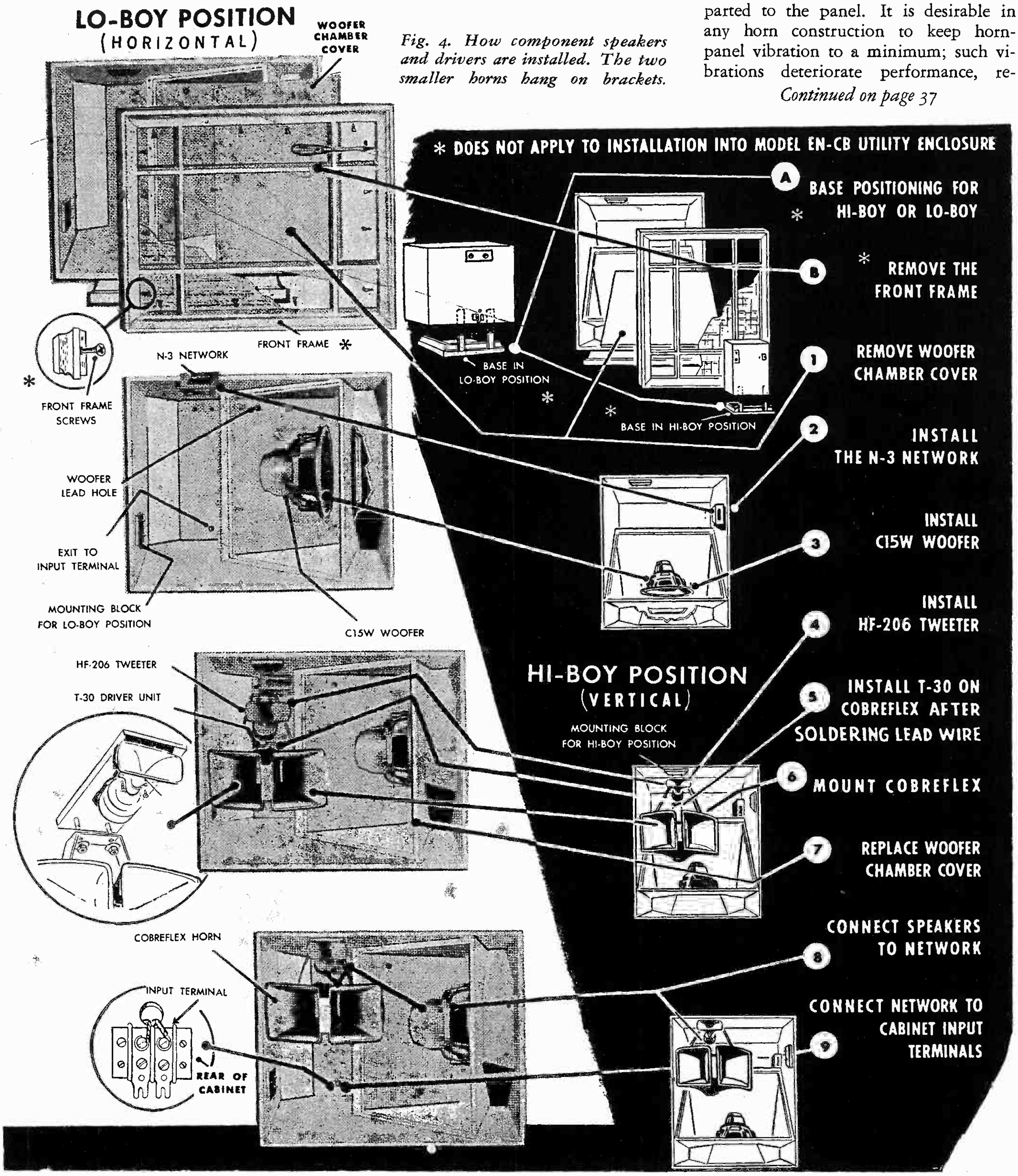



*Fig. 4. How component speakers and drivers are installed. The two smaller horns hang on brackets.*

# PLAIN and FANCY

by R. D. DARRELL

PERSONAL "expressiveness" and "style", the distinctive if not essential traits of musicians, painters, and poets, seldom are ranked among the more admirable attributes of technical craftsmen. Barely permissible in an inventive design engineer, they are generally regarded as dubious, and very likely dangerous, idiosyncrasies in an amateur home-equipment assembler or builder.

Yet, to deny or misjudge them is a more certainly dangerous miscalculation of the basic motivations of any home craftsman, intent not only on investing some of his leisure time in a fascinating hobby, but also on exercising what actually are powerful, if often subconscious, self-expressive urges. The vogue of doing it yourself has many attractions, and not the least of these is the obscure desire to make the "doing" at least a modest kind of *creating* and to identify "yourself" not merely as the anonymous

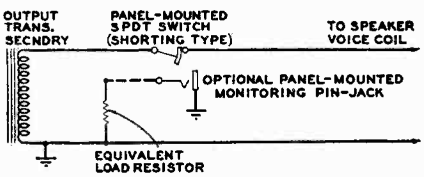



*Fig. 1. Switched amplifier load resistor.*

workman involved, but as an *executant-interpreter* who stamps the finished work with something of his own individuality.

Admitting the existence — and power — of this expressive stimulus is not, however, to ignore its genuine dangers nor to dull the painfully sharp horns of the dilemmas on which the inexperienced craftsman is sure to become impaled. Yielding to the temptation to depart from conventional procedures and to attempt more-or-less novel or original approaches has been the first step in almost every new invention, to be sure; but for the novice who has yet to master the basic techniques of his craft, reckless experimentation generally leads straight to disaster. And even for the more experienced, most departures from standard practices risk the waste of both time and money. Yet some such ventures *do* have a successful or at least a gratifying outcome. In any case, they are often an irresistible temptation.

Unfortunately, the audio hobbyist in particular seldom can find outside help in solving this troublesome problem which faces the self-expressive amateur in every field: how to chart — and steer — a safe course between the Scylla of rash innovation and the Charybdis of stultifying orthodoxy. There is plenty of useful information in the audio literature, all right, but except for sound advice on elementary construction practices[1], it exists mostly in scattered bits which can be unearthed and pieced together only by indefatigable searching. And even more unfortunately, the ideal source of guidance — the personal counsel of a master audio craftsman — seldom is easily and constantly accessible in most home workshops. Anyway, it is perhaps only an experiment-scarred veteran amateur who is foolish enough to rush in where even angels hesitate to tread . . . who can fully sympathize with an impatient, imaginative novice . . . and whose own case-history provides a wide range of off-the-beaten-path adventures which profitably might be emulated (or avoided).

Now, I probably flatter both the novice and myself here: the former in assuming that he really wants or is able to learn from anyone else's experience; myself in presuming that there are significant lessons to be drawn from the many grotesque failures and few very modest successes which have been experienced in my own workshop. Perhaps everyone must learn things the hard way by personal trial and error, and indeed most of my own lessons have been drubbed in by just such painful means. Yet I can't forget that at least a few (and the most profitable) of them came easier — thanks to the timely suggestions or warnings of friends who were far more experienced, or perhaps simply more gifted with plain common sense! I find a kind of rueful pleasure (perhaps only the psychological relief of confession) in reviewing the checkered record of my own departures from conventional amateur-audio practices — and I dare hope that publishing that record may, at worst, save some novices from being trapped in some of the dead-end alleys in which I've often found myself or, at best, perhaps stimulate a few others to share some of the rare, but incalculably rewarding, discoveries I've either stumbled on or been led to.

It strikes me now that most departures from strict amateur orthodoxy (in circuit design, component selection, construction practice, or whatever) represent movements either away from or toward greater complexity or stylishness — i.e., trends which conveniently can be christened (with due acknowledgments to the Broadway hit show) respectively "Plain" and "Fancy". In one direction the ideal goals are greater simplicity or economy of means; the obvious dangers are under-design and lowered performance quality or reliability. In the other direction the ideals are increased versatility, professional performance and appearance standards, and perhaps above all elegance; the corresponding pitfalls are needless over-design, excessive cost or labor, and superfluous — if not operationally risky — complications.

Let me illustrate the latter (Fancy) trend by 3 characteristic examples from

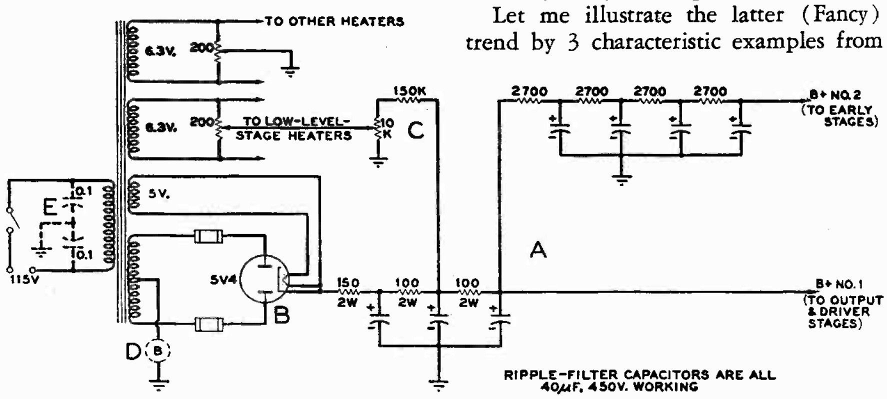



*Fig. 2. Diagram of amplifier power supply, illustrating several points in the text.*

my early audio-craftsman activities, on which the final verdicts turned out to range from Failure, through No Decision, to Success.

The first was an ambitious yet seemingly reasonable attempt to incorporate complete monitoring facilities in my main home sound system. This cost me the price of 3 good meters and several switches and pin-jacks, to say nothing of

[1] C. G. McProud: "Construction Practice," in *Audio Engineering*, April, May, and June 1950 (reprinted in *The Second Audio Anthology* 1953). Note: see also Chap. 24 of *The Radio Amateur's Handbook*, 32nd ed., 1955, for helpful advice on standard electronic workshop practices.

countless hours of (steel) panel-drilling and figuring out the elaborate switching circuit required, for current measurements especially. Well, the AC power-input voltmeter provided a convenient indication that the system was switched on — but was considerably more expensive than a pilot light, and not much more usefully informative. And after the first few weeks, I found that I never paid any attention to the DC voltmeter and milliammeter, in whatever circuit they happened to be switched, and I seldom made any use of the multitudinous audio-signal check points. If operation was off or impaired, I recognized it by ear before I did by eye, and if tube replacements didn't clear up the trouble immediately, I had to work over the chassis on my bench, where — turned

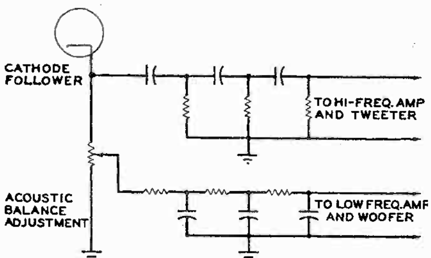



*Fig. 3. Pre-amplifier crossover network.*

over — they were just as easy to check directly from various key circuit points as via my fancy array of panel meters and pin-jacks.

However, I should add that this ill-fated venture wasn't completely fruitless. It sold me on aluminum rather than steel chassis and panels, once post-war stocks of the former became available. And at least one of my monitoring devices proved to be genuinely useful: permanently mounted, panel-switched, amplifier load resistors which could be readily substituted for the normal speaker loads (Fig. 1) whenever I wanted to measure power outputs, or — in the case of my Drisko-designed dual-channel amplifier — whenever I wanted to illustrate woofer or tweeter performance alone.

In the second (even Fancier) example, I grandly squandered some of my war-time savings on the installation, wherever possible, of "Red" or other deluxe "reliable" tubes — 5692's instead of 6SN7's, 1620's instead of 6J7's, and so on. All of them lived up to their exceptional long-life or low-noise ratings all right, but when I no longer could afford them, I soon discovered that by careful selection I could do just about as well with ordinary tubes — for about one-fourth the price! Strictly speaking, the actual saving is not as great as comparative prices might indicate, since a larger supply of ordinary tubes has to be maintained to permit effective selection and pair-matching. On the other hand, this oversupply provides also a useful stock of replacements and emergency spares.

Anyway, I've now come around to using ordinary tubes everywhere except in my main-amplifier output stages where it seems to me that 5881's are enough superior to 6L6's in both longevity and ease of matching to warrant their only slightly greater cost. Nevertheless, this particular case of Fanciness still strikes me as worth while under certain circumstances — perhaps especially for audiophiles who build only a single system and do not need to maintain a considerable tube stockpile.

In the third case, however, the verdict was neither adverse nor equivocal. The most satisfactory single audio investment I've ever made was that of a studio-type calibrated T attenuator. Even at its present price of around $18 (as against less than a dollar for an ordinary carbon pot), I still recommend it as no extravagance, both for its consistently smooth, quiet, and trouble-free operation and for the inestimable sense of gratification in owning a kind of "Jaguar" audio component. Once it has been tried at all (and mine has been in constant use since 1948) I can't imagine any audiophile remaining content with the ordinary jumpy, often noisy, and uncalibrated volume control. My main excuse for getting it in the first place was for use in a 500-ohm line between pickup preamp (or tuner) and main amplifier (which also enabled me to mount it on a long extension cable so that it always could be at hand no matter where I sat in my listening room). But it proved to be so satisfactory in other respects that nowadays I invariably use similar calibrated step-type potentiometers where T attenuators are not practical, as in high-impedance circuits.

I realize, of course, that the markedly higher cost of such professional attenuators or volume controls makes them impracticable for ordinary commercial use, yet even so I have never been able to understand why frankly deluxe, high-priced home systems fail to include a feature which so gratifyingly combines true luxury with topmost quality. The situation for the amateur audio craftsman, who builds only one or an occasional system for his own use, is obviously entirely different, for here the original cost is not subject to a lengthy series of further mark-ups. An occasional luxury component should not break even a limited budget and, if it does stretch that budget uncomfortably, the rewards in this case justify the additional strain.

Turning from Fancy to Plain departures from conventional amateur-audio practices, I find (with some embarrassment) that I can't illustrate so wide a range of successful, unsuccessful, and indecisive examples as those above, since practically all the comparatively few true simplifications I have tried have been uniformly successful! The best of these by far has been the consistent use of RC, rather than LC, power-supply filtering (Fig. 2A), but that's too big — and commonly unappreciated — a subject to discuss in detail here. Instead I might cite 2 others of less consequence, which also achieve considerable savings of different kinds.

One is the simplest means of protecting output stages from the shock of high-level audio signals before the output tubes themselves are fully warmed up. Instead of using separate switches for the amplifier heater and B+ voltages (always remembering to wait a while before switching on the latter), or installing a more-or-less elaborate time-delay relay in the B+ line, it is enough merely to substitute an indirectly heated or cathode type of main rectifier tube(s) in place of the conventional filament type (a 5V4 instead of a 5U4, for example — Fig. 2B). The 5V4's in my Drisko system take just about as long to heat up as do the 5881 (or 6L6) output tubes, with the result that just as the delayed first surge of B+ voltage starts to go above its normal operating level, the simultaneous rise to full current in the output tube drags it down to stabilization. Thus, the output stages are in normal operation by the time the first audio signal comes through the similarly warmed-up earlier stages.

The other is even simpler, if far less elegant. After some years of cursing and sweating to unsolder and disentangle tightly wrapped connections whenever I had to make circuit repairs or revisions, I deliberately abandoned my orthodox notions of constructional neatness. I use standard wire only for heater lines: everywhere else I not only use solid hook-up wire, but am careful always to leave a short bit of pig-tail standing clear after I have wrapped it around a

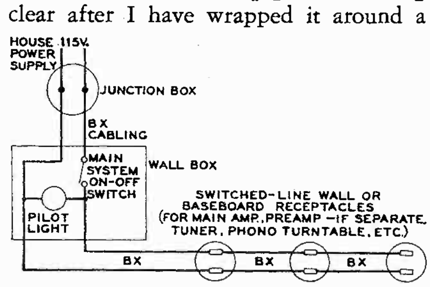



*Fig. 4. Separate sound system power line.*

terminal pin or hooked it through a tube-socket lug. It doesn't look pretty or even workmanlike, but whenever the connection has to be unsoldered it makes for a tremendous saving on one's dexterity and temper alike!

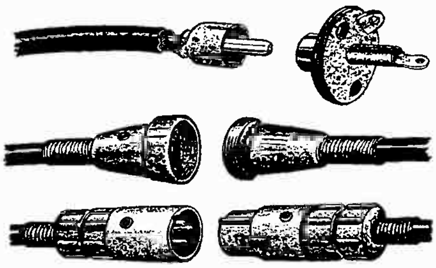
The last notion, obvious or dubious as it may seem, is one of my very few quasi-spontaneous departures from audio norms in the direction of Plainness. Apparently the natural bent of my own "expressive" temperament has an irresistable orientation toward Fanciness!

At any rate, it is others to whom I am indebted for most of my successful Plain ventures — above all to my earliest and most resourceful mentor of all, Benjamin B. Drisko, whose own circuits themselves rank among the finest exemplifications of economy of means in audio design. I strongly recommend the careful study of his preamp and main-amplifier circuits, which have been published elsewhere [2, 3] for many of their distinctive features have been not only extremely successful in these particular applications (as testified by extensive use-tests by many others besides myself), but are easily adapted for comparably successful utilization elsewhere.

Specifically, I am indebted to Drisko for persuasively arguing and demonstrating the virtues not only of the RC power-supply filtering and indirectly heated rectifier tubes mentioned above, but also of 2-stage-only (as opposed to multi-stage) feedback loops. In addition, I have profited greatly by his recommendation of larger-than-normal grid resistors (in voltage-amplifier stages only, of course), where a combination of conservative design, keeping well below tube maximum-dissipation ratings, and heavy feedback minimizes the otherwise omnipresent danger of grid-current troubles. With such 2.2-megohm grid resistors, it becomes possible to use coupling capacitors as small as .01 $\mu$fd without low-frequency performance deterioration. That, in turn, permits the use of mica rather than paper capacitors. I'm not sure whether this should be ranked as a Plain or Fancy departure from conventionality: micas may cost a little more, but on the other hand they take less space, look more distinguished, and certainly are superior performancewise.

Moreover, I am personally convinced of the performance superiority of dual-channel amplifiers (utilizing high-impedance RC frequency-dividing networks before the power amplifiers, Fig. 3) over single-channel amplifiers which require LC crossover networks between the output transformer and any dual or multiple speaker system. But that's too controversial a subject to argue here and, in any case, it's a far from clear-cut example of Plain-vs.-Fancy trends: it is Plain, perhaps, in its substitution of inexpensive, genuinely constant-impedance RC networks for the more costly and quite mythically "constant" - impedance LC types; it is definitely Plain in its simplification of the always tricky amplifier-output/speaker-input relationships; but it is undeniably Fancy in its doubling of the conventional single-amplifier requirements.

Such ambiguity, or Plain-in-some-respects/Fancy-in-others, is characteristic of numerous other departures from strict amateur-audio conventionality. Except for the use of DC-biased AC (rather than straight DC) heater supplies for low-level stages, Fig. 2C, which are now fairly common practice but which I first picked up from Drisko, I can't remember or trace the source of the ideas themselves. But wherever they came from, they've worked out as well or almost as well in the following instances: Scotch-taping a stroboscopic disc permanently to my early-model Rek-O-Kut turntable (no longer necessary in many current models of this and other high-quality turntables) . . . . Maintaining up-to-date schematics and voltage charts on every piece of equipment I build or revamp . . . . Never building a power supply for any individual piece of equipment until it has been fully tested in operation from an auxiliary bench power supply and the exact voltage and current requirements determined . . . . In some power supplies, avoiding the use of 2 regular fuses in the rectifier-plate leads, or one in the B+ line, by inserting a 6-volt, quarter-amp pilot lamp — which acts as a combination fuse and current-on indicator — between the power-transformer secondary centertap and ground (Fig. 2D) . . . . And, perhaps most notably of all, installing a switched 115-volt AC line for all system units in normal use, so that only a single on-off switch (with pilot light) is required for normal system operation, Fig. 4 — a real convenience and protection, especially when it's necessary to turn one's system over to the kind but clumsy hands of one's family and friends!



*Figs. 5A to C, from top: audio connectors.*

Not surprisingly, it has been in the unambiguously Fancy department that I have experimented most frequently and with the liveliest relish, even though such ventures generally turned out to be failures except for the lessons they taught me — the hard way!

For example, I feel that I have learned most about what *not* to do in the way of chassis size and shape selection and component layout from having once fully indulged a passion for compactness and miniaturization. It probably was my familiarity with wartime airborne equipment which prompted me to build a whole series of amplifiers, preamps, and tuners, using miniature tubes and whatever other dwarfed components I could lay my hands on, all crammed together in veritable Chinese puzzles of undersized chassis. They worked, sure — but not as hard as I did whenever repairs were needed or revisions wanted! Now, I'm more likely to go to the other extreme, but at least I can find my way around in any chassis and replace any individual components without completely disassembling the entire unit.

And it wasn't until I had spent many hours checking actual performance results that I became quite convinced that 2-stage-only feedback loops were far more effective and stable, as well as easier to handle, than multi-stage loops . . . that careful component matching, heavy feedback, and the omission of cathode and screen bypasses[4] made special push-pull balancing devices quite superfluous . . . that there are better ways of avoiding hum troubles than by excessive, indiscriminate shielding . . . that voltage-regulator tubes or fancy regulated power supplies are quite unjustified in any normal home sound systems embodying true Class-A amplifier operation . . . that by-passing the power-input lines, Fig. 2E — considered *de rigueur* when I first began building audio amplifiers — is pointless unless one lives in the neighborhood of a ham or broadcast station or some other source of RF interference. . . .

On the other hand, an occasional triumph of Fanciness has more than repaid me, not only for the costs directly involved, but also for all the indirect losses incurred in the complete or semi-failures. One of these, which ranks with the adoption of calibrated attenuators mentioned earlier, should have a special appeal to those home assemblers and builders who have complained so frequently (in the pages of HIGH FIDELITY and elsewhere) of dissatisfaction with the so-called "phono" type of audio signal line connectors, Fig. 5A. In addition to echoing their main gripe about the lack of proper finger-grip facilities, I have never been happy with these and the popular screw-type of shielded single-line connectors, Fig. 5B, on other grounds as well. Certainly they seldom seem to be either convenient or dependable when they are subject to frequent connection and disconnection.

At any rate, a happy chance introduction (first at a wartime-surplus counter) to the XL locking type of professional or microphone shielded twin-line plugs and jacks, Fig. 5C, sold me on these completely. For a good many years now I have been using them exclusively and they not only have been impeccably trouble-free but consistently easy and pleasurable to use — and even a delight just to look at! I recommend them par-

*Continued on page 37*

[2]Drisko, B. B., & St. George, P.: "Versatile Phonograph Preamplifier," *Audio Engineering*, March 1949 (reprinted in *The (First) Audio Anthology*, 1950).

[3]Drisko, B. B., & Darrell, R. D.: "40-DB Feedback Audio Amplifier," *Electronics*, March 1952.

[4]Drisko, B. B., & Darrell, R. D.: *Op. cit.*

by **HERMAN BURSTEIN**

# *Multiple-Speaker Switching Systems*

AFTER having installed an elaborate sound system in his living room, the owner may all too often find himself removed from listening range when chores or interests require his presence elsewhere in the house. The obvious remedy is to install additional speakers in various rooms, or to provide outlets for one of the portable speaker systems, including cabinet, that are available today. (If such outlets are used, the outlet jacks and speaker plug should ***not*** be of the type used for electrical wiring. Otherwise, the portable speaker might be plugged into the 117-volt house power line, with disastrous results.)

The following 3 principles apply to multiple speaker use in the home:

1) The ratio between the amplifier's speaker-tap impedance and the total impedance of all speakers connected to it should lie within the limits of 3:1 and 1:3*. When several speakers are used simultaneously in parallel, the total impedance is $\frac{1}{\frac{1}{Z_1}+\frac{1}{Z_2}+\cdots\frac{1}{Zn}}$, where $Z_1$, $Z_2$, etc. are the impedances of the individual speakers.

2) The ratio (damping factor) between speaker impedance and amplifier output impedance should in general not be less than 5:1. (Although lower damping factors may on occasion provide fuller bass, this is not true as a rule when speakers of good quality are used.)

3) The total power consumed by several speakers in parallel should be within the amplifier's rated power at low distortion. For example, if an amplifier can supply 10 watts before distortion becomes evident, and if 3 watts into each speaker is the maximum desired by the listener, then 3 similar speakers can be used simultaneously.

## Impedance Matching

Impedance matching between the amplifier's output transformer and the auxiliary speaker ordinarily should not be a serious problem. A 1:1 ratio between speaker-tap impedance and speaker impedance yields optimum results in terms of transferring power to the speaker at minimum distortion. Mismatches between 3:1 and 1:3, however, are generally tolerable for home purposes. High ratios approach a short circuit across the transformer, while low ratios approach an open circuit and are inefficient in transferring power from amplifier to speaker.

Damping factor is also affected by transformer-speaker impedance matching. Damping is the ability of the amplifier to prevent the speaker from vibrating independently after amplifier signals have ceased. Independent low-frequency vibrations are known as hangover, and high-frequency ones as ringing. Good damping results from a high ratio between speaker impedance and the amplifier's output impedance. Output impedance, or source impedance, is quite different from the speaker-tap impedance. Speaker-tap impedance is the proper load impedance for the amplifier; when a load (speaker) of 4 ohms is connected to the 4-ohm output terminals, the output stage is terminated so as to produce maximum power at minimum distortion. Output impedance is determined by the output stage type and the feedback used, and is ordinarily much lower than the speaker-tap impedance; their ratio is known as the damping factor. If an amplifier's damping factor is 10, for instance, any load connected to the 4-ohm speaker tap would "see" a 0.4-ohm source impedance; if connected to the 8-ohm tap it would "see" a 0.8-ohm source impedance, and so on. Low output impedance heavily loads (short-circuits) the voice coil currents produced by independent speaker vibrations, so that it becomes difficult for the speaker to make such excursions.

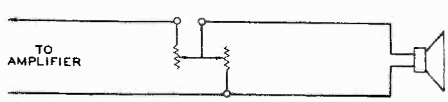



*Fig. 1. L-pad level control for speaker.*

A 5:1 ratio between speaker impedance and amplifier output impedance usually produces efficient damping. Lower ratios may be acceptable when the speaker cabinet is designed to produce good damping. Higher ratios are believed to accomplish relatively little improvement. Amplifiers with heavy feedback, such as the Williamson, have high damping factors, on the order of 10:1 or as high as 30:1, when connected to speakers of the proper impedance. Such amplifiers can tolerate considerable mismatch between speaker-tap impedance and speaker impedance without a noticeable effect upon damping.

Given a good amplifier, one auxiliary speaker and the main speaker can be operated singly or simultaneously without adverse results if both have the same impedance and are connected to a speaker tap of similar value. For example, if two 16-ohm speakers are on a 16-ohm output tap, total speaker impedance is only 8 ohms. The damping ratio is cut in half, which still leaves a high degree of damping for a Williamson or other quality amplifier. Also, the worst possible mismatch beween speaker-tap impedance and speaker is 2:1, which is within tolerance. If only one speaker is switched on, of course, there is no mismatch.

Use of an auxiliary speaker with different impedance than the main speaker may be troublesome if the amplifier has only one speaker tap. Assume the impedance of the main speaker is 16 ohms, that of the auxiliary speaker 4 ohms, and that of the output tap is 16 ohms. A mismatch of 4:1 occurs between output-tap impedance and speaker impedance when the auxiliary speaker is on by itself, and of nearly 5:1 when both speakers are on. However, if the transformer can be rewired for an 8-ohm speaker tap, the problem is no longer serious. Mismatch becomes only 2:1 or 1:2 with either speaker on, and 2.5:1 with both on.

Again, difficulties can arise if more than one auxiliary speaker is added. Assume 2 speakers of 8 ohms each are added to a 16-ohm system. If the 16-ohm output tap is used, a mismatch of 4:1 occurs with the 2 auxiliary speakers on, of 3.6:1 with the main speaker and one auxiliary speaker on, and of 5:1

*A 1:3 or 3:1 mismatch is considered by many to be quite severe; the maximum ratios are often quoted as 1:2 or 2:1. There is no disagreement, though, that it is best to approach a 1:1 match as closely as possible and, when a mismatch is necessary, a downward mismatch is preferable: that is, total speaker-load impedance less than amplifier speaker-tap impedance.—Ed.

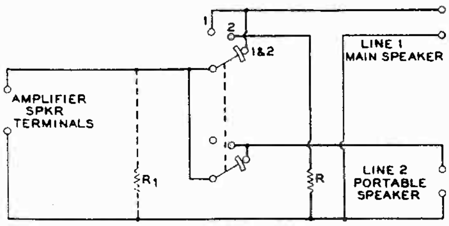



*Fig. 2. Two-line speaker switching setup.*

with all 3 on. But if only an 8-ohm output tap is available, the problem becomes relatively minor, for the mismatches are cut in half.

Use of speakers with different impedances not only produces impedance matching problems from the viewpoint of efficient power transfer, distortion, and damping, but also results in each speaker taking a different amount of power. When the speakers are on simultaneously, this difference in power level can be disturbing. If a 4-ohm speaker is used simultaneously with a 16-ohm one on the same tap, four-fifths of the available power is consumed by the 4-ohm speaker. Relative efficiencies of the 2 speakers may partly compensate for the power difference, or may make matters still worse.

Equal power transfer to speakers of different impedances is possible with some output transformers which have several impedance taps that can be used simultaneously. For example, assume the amplifier has 16-, 8-, and 4-ohm output taps that can be used at the same time. Suppose speakers of 8 and 16 ohms are to be used. Now, to the amplifier a speaker of 16 ohms on the 16-ohm tap appears as the same load as an 8-ohm speaker on the 8-ohm tap. If the 2 speakers are put on taps of their own impedance values the impedance match will be perfect when only one is used at a time, and this is the way they should be connected if both will be used at the same time only occasionally. But if they will be used together often, the 16-ohm speaker should be put on the 8-ohm tap and the 8-ohm speaker on the 4-ohm tap. From the transformer's viewpoint, the 16-ohm speaker on the 8-ohm tap looks like the same load as the 8-ohm speaker on the 4-ohm tap. Therefore when both speakers are on, it will be as though there were 4 ohms (two 8-ohm speakers in parallel) on the 4-ohm tap — again, a perfect match.

If more than 2 speakers are involved, say two 8-ohm speakers and one 16-ohm speaker, the maximum mismatch with all 3 switched on, but with each on a tap of its own impedance, is 3:1. This can be reduced, if the transformer has sufficient impedance taps, by shifting all speakers to the next lower-impedance speaker taps. The two 8-ohm speakers can be shifted to a 4-ohm tap, and the 16-ohm speaker to an 8-ohm tap. Then for any one speaker on there is a 1:2 mismatch, for both 8-ohm speakers on there is a 1:1 match, for one 8-ohm speaker and the 16-ohm speaker there is a 1:1.3 mismatch, and for all speakers on together there is a 1:1.5 mismatch.

## Individual Volume Controls

It is often necessary to equip a speaker with an L-pad, as shown in Fig. 1, of impedance equal or nearly equal to that of the speaker. This permits the volume of the speaker to be controlled independently without changing the impedance it presents to the output transformer and accordingly without changing the performance of other speakers which may be on at the same time.

L-pads are available in a variety of impedances for $2 to $3. The most common type is rated at 4 watts. This presents no limit to the power that can be fed to the speaker but only limits the power which can be safely absorbed by the pad without excess heating as volume is reduced by means of the pad. However, a 4-watt pad, with volume completely down, is adequate to absorb all the

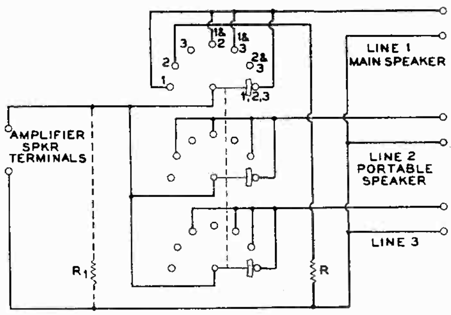



*Fig. 3. Switching system for 3 speakers.*

power generated by an amplifier at comfortable home listening levels.

It should be pointed out that L-pads have some disadvantages too. First, even when turned all the way up, they consume a certain amount of power; when you are operating several speakers from one amplifier, you don't usually have much power to spare. Second, when turned part-way down they insert resistance in series with the speaker, which tends to weaken the effectiveness of amplifier damping. They should not be used indiscriminately, then, and when used should be turned all the way up when possible. Their function should rightly be that of achieving balance when 2 or more speakers are used at the same time; they are not meant to replace the amplifier's volume control.

## Switching

A desirable feature of a multiple-speaker installation in the home is a switching system that feeds the amplifier into any one of a number of "lines". One line may consist of leads to the main speaker. A second line may consist of leads to the several rooms where a portable speaker is to be used. Uses may be found for third and even fourth lines.

A switching system is useful for at least the following reasons:

1) As a means for shutting off one speaker or another, it is easier and more satisfactory than pulling the speaker plug out of a jack on the line.

2) Centralized speaker control is obtained.

3) Power at minimum distortion can be supplied to a single speaker or a selected group of speakers. Additional speakers on the other lines, although silenced by means of their respective L-pads, draw power if the lines are not switched off, so that for a desired acoustic output the amplifier has to supply more power at higher distortion.

4) It can be seen at a glance what lines or speakers are connected to the amplifier.

5) Signaling through control of speakers at remote locations may be convenient.

For 2- and 3-line systems it is feasible to use rotary wafer switches costing less than $2, such as are made by Mallory and Centralab. Fig. 2 is a diagram of a 2-line switching arrangement, and Fig. 3 a 3-line arrangement. When 4 or more lines are involved, the combinations become too numerous for a single switch and it is necessary to employ a separate switch for each line as in Fig. 4.

It is advisable to use shorting-type (make-before-break) switches. In transferring from line 1 to line 2, as in Figs. 2 and 3, the switch would then contact line 2 before breaking contact with line 1. In Fig. 4, contact with resistor R (its purpose and value are discussed below) would be made before breaking contact with any line.

Shorting switches eliminate the possibility of injury to the amplifier's output transformer if, in the presence of a high signal level, the load (the speaker) on the transformer is temporarily removed in switching from one speaker to another. They also prevent clicks and thumps in switching. The possibility of injury to the output transformer because there is no speaker to load it occurs also if the amplifier feeds a line intended

*Continued on page 42*

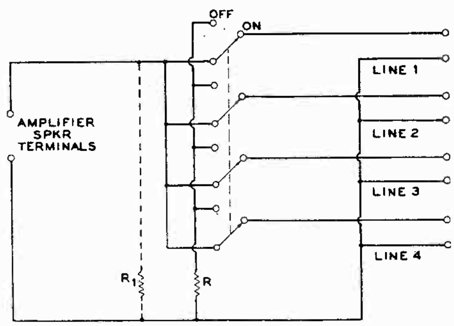
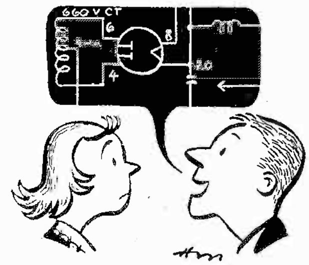



*Fig. 4. Individual switches for 4 lines.*




by JOSEPH MARSHALL

# Practical Audio Design

## Part II: Power-supply filters

IN Part 1 of this article we considered at some length the selection, application, and modification of power transformers. Choosing a suitable rectifier is ordinarily not much of a problem. The principle considerations are: 1) ability to handle the desired voltage and current; 2) space occupied; and 3) heating. Cost is usually not a factor because the smallest rectifiers cost very little less than the largest. In compact construction space saving may be important, and bantam or miniature rectifiers may be preferred. If the equipment is to be totally enclosed, heating may be a consideration, and rectifiers with lower filament current may be preferred to reduce heat.

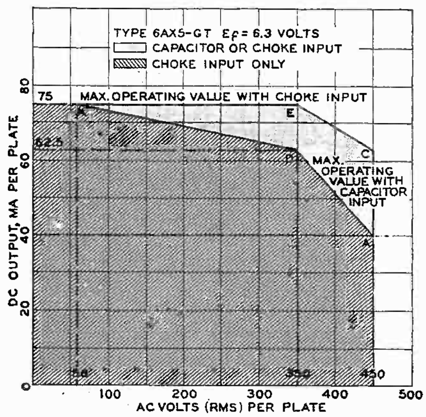



*Fig. 1. A ratiug chart for the 6AX5-GT.*

As noted before, the essential data for choosing a rectifier are presented in charts and curves of the tube manuals. Three charts are usually given. The first is a simple rating chart which tells at a glance the maximum parameters under which the tube will operate satisfactorily. In Figs. 1 and 2, the maximum rating charts for the 6AX5-GT and the 5U4-GB are presented. Please note, to begin with, that the boundary lines of operation for capacitor-input filters are not linear, but have 2 angles like a mansard roof; the curve is horizontal up to a point, then begins to slope at one angle, and finally, after a critical point, it slopes again at a much sharper angle. In the case of the 6AX5-GT this occurs at 350 volts per plate AC input. The tube will deliver 75 ma per plate with 58 volts AC per plate, and 62.5 ma with 350 volts per plate; but, if the voltage is raised beyond 350, the permissible current drops rapidly until at 450 volts AC per plate only 40 ma can be drawn. On the other hand, 450 is the critical voltage for the 5U4-GB. At that point it can deliver 137.5 ma per plate, but only 81 ma per plate at 550 vac input, which is the maximum rated input voltage. It is quite simple with these maximum rating charts to determine what tubes can do the job at hand.

The second and third sets of curves give various practical operating parameters with capacitor and choke inputs; from these, the exact performance can be determined for any given set of operating conditions. Capacitor-input curves, Fig. 3A, require no explanation.

### Choke-Input Filters

Choke-input curves, Fig. 3B, are somewhat more complicated in appearance, but are really just as simple to apply once a few things are clear. The inductance of a choke varies with the current through it, and falls very sharply when the current is high enough to saturate the core. There is, therefore, a *critical* inductance for any load; when the load changes, as it invariably does in equipment using choke-input power supplies, it is essential that the choke have critical inductance or more at both maximum and the minimum loads. If it does not, the reactance of the choke will be so low at some point in the operating range that it has very little or no effect—the rectifier will then appear to face the capacitor which follows the choke, and will behave as if the filter were a capacitor-input type. There will be a violent rise of voltage, and the regulation would be no better (and possibly worse) than if a capacitor-input filter *were* used.

The formula for critical inductance is quite simple. First, determine the load resistance by dividing the voltage by the current (in amps) the supply is to deliver and, to be accurate, add the series resistance of the filter. If the supply is to deliver 400 volts at 100 ma, and the choke resistance is 200 ohms, the load resistance is $400/0.1 + 200 = 4{,}200$ ohms. However, if the current falls to 40 ma, the load resistance rises to $400/.04 + 200 = 10{,}200$ ohms.

The critical inductance for a 120-cps full-wave circuit will be approximately

$$\frac{\text{Load resistance}}{1{,}000}.$$

In the example given, the critical inductance would be 4 h at the full load and 10 h at the minimum load. The *optimum* inductance is twice the critical inductance. So, for the above example, we would need a choke with an inductance at 40 ma of 20 h, and no less than 8 h at the 100-ma point.

The newer type of graph for choke-input filters eliminates the need for these calculations, and makes it possible to determine the behavior of a given rectifier with a choke of a given inductance or to choose the optimum inductance at a glance. It will be noted that Fig. 3B has, besides the familiar solid lines representing different AC input voltages, additional curves in broken lines. The long-dash lines radiating from the lower left corner are the load lines for different choke values. There is also a set of short-dash lines marked with values of inductance too, which meet and extend to the left the solid input-voltage lines; these show what happens to the output DC voltage with chokes of the given values when the current drops below certain points.

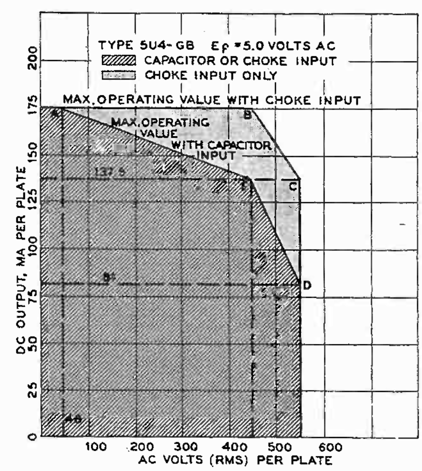
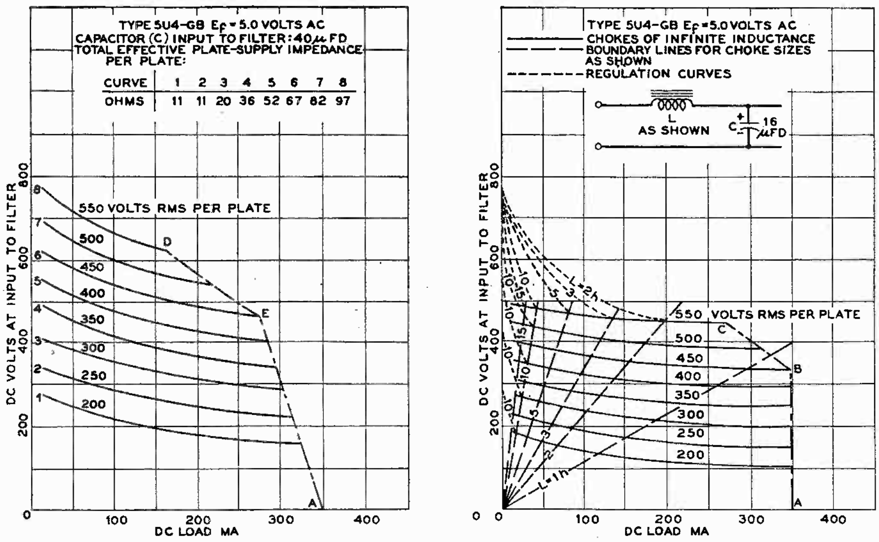



*Fig. 2. Maximum ratings for the 5U4-GB.*

For example, the long-dash curve marked 2 h and the short-dash curve also marked 2 h intersect the 550 volts AC solid-line curve at the 200-ma coordinate. This means that with an input




*Figs. 3A and 3B. Operating characteristics for capacitor- and choke-input filters.*

of 550 volts AC per plate, and a load of 200 ma, the 5U4-GB will deliver 450 volts DC into a choke of 2 h. Now let's see what happens as the load is changed. If the current falls to 100 ma we see, by following the short-dash line, that the DC voltage would rise to 550. An inductance of 2 h is less than the critical value at this load and is, in effect, a short circuit; the rectifier now works into the following capacitor and the DC rises to the peak value of the AC; the regulation is poor and no better than it would be with capacitor input. On the other hand, going the other way, we follow the solid line and see that we can maintain close to 450 volts DC output between 200 and 275 ma, the maximum rating for the tube. Obviously, then, with a choke of 2 h we must keep the current from falling below 200 ma if we want the regulation to be good. To put it another way, a 2-h choke is usable with this tube only when the minimum current does not fall much below 200 ma.

The general rule when designing a choke-input power supply is to choose operating conditions (or choke size) so that the desired DC output voltage and current drawn will always fall *below and to the right* of the long-dash line representing a given inductance.

## Tube Choice

Although there are dozens of rectifier tubes, 99% of audio applications will be served with the following most common types:

For maximum loads above 175 ma — 5U4-GB or 5R4GY.

For maximum loads between 125 and 175 ma — 5V4G.

For maximum loads up to 125 ma — 5Y3-GT or 6AX5-GT.

For maximum loads up to 75 ma — 6X4 or 6X5-GT.

When the extreme rating of the smaller of 2 possible tubes is approached it may be advantageous to use the next larger tube. When 70 ma is required, the maximum input for the 6X4 or 6X5-GT is 325 volts AC per plate, and the DC output would be 300 volts. But with the 5Y3-GT or 6AX5 the input could be raised to 450 volts per plate, and a higher output voltage could be obtained; or, with the same 325 volts per plate input, the 5Y3-GT could deliver 350 volts DC output.

Sometimes space is a consideration, and the miniature 6X4 would be preferred even with lower output voltage. Occasionally there is only a 6.3-volt filament winding on the power transformer, and in such cases the 6-volt series of rectifiers with cathodes must be used. Moreover, the use of a rectifier with an independent cathode prevents excess voltages from appearing across the filter capacitors during warm-up and it is then possible, if necessary, to use filter capacitors of more compact form and lower voltage rating.

It may be necessary or advisable in rare cases to increase current capacity by operating 2 rectifiers in parallel. There is some difference of opinion as to whether the 2 sections of a single rectifier should be paralleled or corresponding sections of 2 rectifiers. I prefer paralleling the 2 sections of one rectifier, because when one tube goes out the power supply will still operate (as a half-wave supply). But there isn't much choice because, in either case, the remaining tube will be seriously overloaded and may be damaged. Because of differences between individual tubes or sections of tubes, there will not be an equal division of the load with parallel operation. This can be corrected at least partially by connecting a resistance of 50 to 100 ohms in series with each plate. Of course, if the resistors are not reasonably matched there may be no improvement at all, so they should be paired with an ohmmeter for the best possible match. It is much more preferable to parallel identical tubes — but it *is* possible to use dissimilar tubes. Some TV sets, for example, use one 5U4-GB and one 5Y3-GT; one section of the 5U4 is paralleled with one section of the 5Y3. There is little to be gained from the use of dissimilar tubes, however, and their use may cause severe trouble in case of breakdown.

## Hum Filtering — General

The principle behind hum filtering is a rather simple one. At the output of the rectifier the current has 2 components: the desired DC component, and the unwanted AC component (or ripple). If we could devise a network that would let the DC through, but close the door on the AC, we could take the one and leave the other behind. We need something that, when placed in series with the load, would offer very little resistance to DC, but a high resistance (or, more accurately, reactance) to AC. This would provide the closed door or dam for the AC. Now, if we added another element from the rectifier output to ground that offered a very high resistance to DC but a very low reactance to AC,

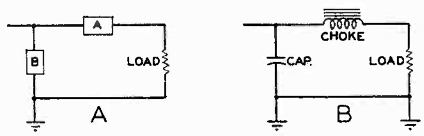



*Fig. 4. Power-supply filters; see text.*

the dammed-up AC would be bypassed to ground. In the block diagram, Fig. 4A, the element A has high resistance to AC but low resistance to DC; the element B has high resistance to DC but low reactance to AC. When they are given a choice of paths, all electrical currents divide in inverse proportion to the resistance (or reactance) offered by the several paths. They take the easiest way, and the path of least resistance (or reactance) will carry the higher load. An inductance has high reactance to AC but low resistance to DC, and so is suitable for the series element A. On the other hand, a capacitance has low reactance to AC but high resistance to DC, which are characteristics desired for the shunt element B. Working into such a crossroads most of the DC will take the easier road provided by the inductance, while most of the AC will take the easier road provided by the capacitance. At the output of the inductance, Fig. 4B, a current is obtained which is very much freer of AC than at the input.

Single-section filters often suffice when the power supply feeds a power amplifier. For low-level stages with high gain, though, the residual ripple would be amplified and would be audible at the output. Additional filter sections are therefore added to reduce ripple even further. And to reduce the need for large chokes, the filter can be made *progressive*; that is, the different stages in an amplifier are fed from different points in the filter — the high-level stages close to the rectifier, the low-level stages far from it. With such an arrangement, Fig. 5, only the first element of the filter has to carry the total current of the amplifier.

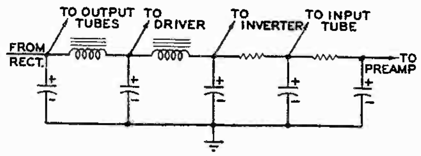



*Fig. 5. The progressive filter design.*

The first choke might have to be capable of carrying 150 ma, but the second choke would probably carry no more than 50 ma and could accordingly be much smaller and less expensive. High fidelity amplifiers may have as many as 8 sections of filter. Besides reduction of hum ripple, the additional sections also provide decoupling (more on that later) and so perform 2 functions simultaneously.

There are all sorts of complicated formulas for working out filters of any desired type or effectiveness. These are useful in special applications, such as figuring out the cheapest way to obtain filtering of a high-voltage supply for transmitters, or in a production amplifier. But for 99% of home audio applications, common sense in applying the information supplied by the tube manuals will be perfectly adequate.

A few points only need be kept in mind. First, the frequency of the ripple will depend on the type of rectification. A half-wave rectifier produces a 60-cps ripple (we assume 60-cps line current in all cases) but a full-wave rectifier produces 120-cps ripple. This is evident from the diagram in Fig. 6. Here A is the unrectified AC input to the power supply. B is the waveform of a half-wave rectifier working into a capacitor load, which smooths the output by charging on peaks and discharging in the nulls. The saw-tooth wave representing the ripple is of the same frequency as the original 60-cps AC. With a full-wave rectifier, however, the spaces between the positive peaks are filled by the inverted output of the second rectifier section, and the resulting waveform appears as in C. Here there are twice as many positive peaks, twice as many saw-tooth peaks, and accordingly the ripple frequency is 120 cps. It takes roughly twice as much inductance and capacitance with a half-wave supply to obtain filtering equivalent to that with a full-wave supply. It is easier and cheaper, therefore, to filter full-wave than half-wave rectifier supplies.

There is one exception. In low-fidelity equipment (such as table radios and small phonographs) the response is sloped sharply below 100 cycles, partly because the speaker baffling is limited, partly because a small and cheap output transformer is used, and partly because rolling off response below 100 cycles minimizes the danger of motorboating and other types of instability. It is then advantageous to use a half-wave supply for this reason: the slope in the response of the equipment will *itself* reduce the 60-cps hum, but it would have very little if any effect on 120-cps hum. In such a case the half-wave sup-

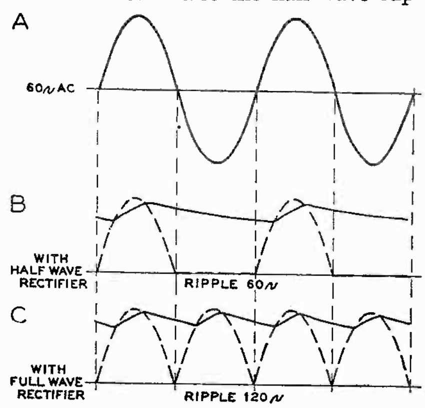



*Fig. 6. Full-wave ripple is at 120 cps.*

ply would require much less filtering than the full-wave, and the same hum level in the output of the radio could well be achieved at lower cost with a half-wave supply.

## Capacitor-Input Design

It has been noted that a capacitor-input filter produces highest DC voltage with a given AC input to the rectifier, and so is preferred whenever the current load is reasonably stable. The value of capacitance into which the rectifier works can be important. Up to a point, the higher the capacitance, the higher the DC output voltage. Fig. 7, for instance, is the graph for the 5T4 rectifier. You will note that raising the capacitance from 4 to 16 $\mu$fd produces an appreciable rise in voltage, but there is little additional increase as the capacitance is increased beyond 16 $\mu$fd. Also, larger input capacitors give better filtering action. On the other hand, a large capacitance may increase the *peak* initial charging current of the power supply beyond the rectifier's ratings and cause trouble. This is a consideration only when the power supply is expected to deliver close to the maximum possible current for the given rectifier. In such cases, it is wise not to exceed the maximum input capacitance specified in the tube manual. And when trouble is anticipated, it can be minimized by inserting current-limiting resistors of about 50 ohms in series with the rectifier plates, as discussed previously.

In a capacitor-input filter the capacitor closest to the rectifier carries the greatest part of the AC current. The ability of a capacitor to carry high AC currents depends on its capacitance, voltage rating, and type of dielectric. The higher the capacitance and/or voltage rating, the higher the permissible AC current. Plain-foil electrolytics can carry higher ripple currents than etched-foil units, and paper capacitors can carry a much higher current than electrolytics. For most audio uses it will suffice to use the following rule of thumb:

Etched-foil electrolytics can be used for loads up to 70 ma (DC) with no precautions whatever.

Plain-foil electrolytics can be used for DC loads up to 150 ma with no precautions.

When the load exceeds 150 ma, the best capacitors available with the highest working voltage and capacitances up to 40 $\mu$fd should be used.

If the current is much in excess of 150 ma and the greatest insurance against breakdown is desired, it may be wise to use 2 capacitors in parallel; they will divide the ripple current and each will carry only a portion of the total. When the current load exceeds 200 ma, the use of paper capacitors is worth while.

As in the case of power transformers, the rigid ratings are often disregarded in commercial practice today. Thus in TV receivers, the input capacitor as often as not carries ripple currents up to 100% greater than JAN ratings. Many of these operate for long periods without trouble.

Electrolytic capacitors invariably conduct some appreciable DC current be-

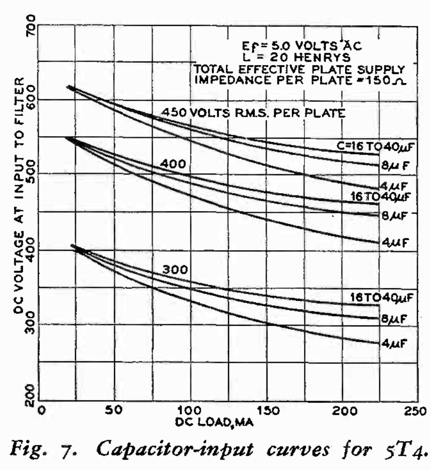



*Fig. 7. Capacitor-input curves for 5T4.*

cause the dielectric is imperfect. This DC leakage may be fairly high with high capacitance. It is wise to make sure that the leakage of the input capacitor is not excessive, by measuring the DC resistance *after* a period of use. Electrolytic capacitors must be formed by applying a DC load for a period of time; when first installed, they usually have high leakage, which is greatly reduced when the electrolyte is formed — *if* everything goes right.

When the power supply is called upon to deliver more than 400 volts or so, the working voltage comes uncomfortably close to the maximum ratings of electrolytic capacitors. When the highest reliability is desired the capacitors can be used in series. The individual sections should each be shunted with a high resistance, as indicated in Fig. 8, to equalize the load.

Aside from these considerations, capacitance of the filter capacitors is not very critical. In general, the higher the capacitance, the better the filtering; for audio amplifiers the individual sections should be at least 20 μfd and preferably 40 μfd. When making audio equipment for myself I use an 8 or 10-μfd paper unit for the input capacitor. This provides the highest insurance at the most critical point. I follow this with multiple-section or plug-in electrolytics of the highest practical capacitance and working voltage.

The inductance of the choke used in a capacitor-input filter is, to be sure, important; but the choice is limited to what

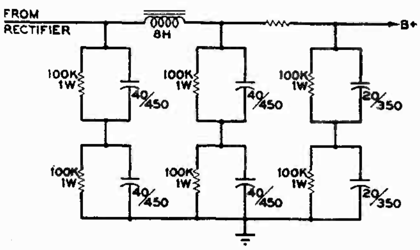



*Fig. 8. Filter for a high-voltage supply.*

is available. Transformer companies have standard designs that offer a good compromise of various characteristics. Actually, in choosing a choke from a catalogue the principal consideration will be the current-carrying capacity. There is some choice of inductance for a given current capacity, but not much, and the cost of a choke goes up drastically as the inductance or the current capacity increases. Furthermore, the DC resistance also increases with the inductance, and the drop in output voltage because of the greater resistance may be more important than the improvement in filtering efficiency. It is cheaper to increase capacitance than inductance, and it is not very material (in a capacitor-input filter) whether a larger LC product is achieved with higher inductance or higher capacitance. An inductance of 8 h or more is desirable in a power amplifier, and 8-h chokes are available for just about any current load up to 300 ma.

If you can afford it, it is nice to operate a choke well under its maximum rating — that is, to use a 200-ma choke for a load of only 125 ma. But, with the intermittent operation common for most audio equipment outside radio and recording studios, choke heating is practically never much of a consideration. There is no great disadvantage in operating chokes at their maximum ratings, nor much to be gained from operating at lower loads. However, exceeding the rating is another matter entirely because the inductance then drops severely and filtering action is greatly impaired.

Small chokes capable of handling between 50 and 100 ma were very common in pre-war radios, especially console models, and can be salvaged from them. Occasionally the deluxe console models will yield chokes of 120 to 150-ma rat-

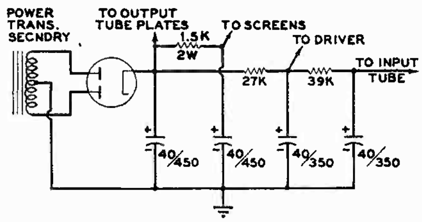



*Fig. 9. A chokeless power-supply filter.*

ing, and even larger ones can sometimes be found in old public-address amplifiers. Size and weight give a rough indication of current-carrying capacity. If they have been stored in basements or garages under poor humidity conditions, corrosion may have opened the soldered joints where the ends of the windings are connected to the leads. It is often possible to repair these by slitting open the paper cover over the windings to expose the connections, and resoldering them or installing new leads. Not all experimenters are aware that old output transformers can be used as chokes in applications calling for currents of 50 ma or less. One or 2 old output transformers and a 3-section 40-μfd capacitor can make an excellent filter for a preamplifier, control unit, or similar equipment. When using a transformer this way the primary is used for the choke; the secondary is left open.

Many fine high fidelity amplifiers today are powered by supplies using no choke at all. In these the voltage for the output tubes is taken directly from the output of the rectifier. This is possible because several conditions are met. First, the inductance of hi-fi output transformer primaries often exceeds 50 h. This inductance presents a very high reactance to the AC ripple and behaves as if it were a big choke, because the output stages of high fidelity amplifiers are push-pull and almost always have means for balancing the output tubes quite exactly. When the output stage is exactly balanced the hum in the output tranformer will cancel out because it is in phase in both halves. Under these circumstances the hum suppression is high enough to be completely satisfactory. This expedient, however, does not work nearly as well with ordinary low-inductance output transformers and output stages which do not have provisions for balancing the 2 sections. Fig. 9 is a diagram of a chokeless power supply suitable for such amplifiers.

## RC Filter Design

Resistance-capacitance (RC) filters are quite practical also for control units, preamplifiers, mixers, and such units wherein the current load is moderate. A real economy is possible by using a power transformer capable of delivering an excess voltage and then using relatively high resistors and capacitors in the filter. For example, if 300 volts at 50 ma is needed we can design an inexpensive RC progressive filter much more compact than one using one or more chokes. The only critical resistor is the first one, because it carries the total current. We can estimate the total current with the aid of the tube manual, or by the rule of thumb allowing 10 ma for each low-mu triode and 2 ma for each high-mu triode or pentode. We can use a transformer and rectifier chosen to deliver 400 volts at the input capacitor. We now have 100 volts we can throw away. The maximum resistance for the first resistor can be figured by dividing the desired or permissible voltage drop by the total current that will pass through the resistor. In the example this would be 100 volts divided by 0.05 amps (50 ma) which gives us 2,000 ohms. Large capacitors must be used; 40 μfd would be suitable. I find it convenient to use an adjustable power resistor of 10 watts with a resistance about 50% higher than that calculated. The slider can then be moved to provide the required voltage at the first takeoff point. Additional

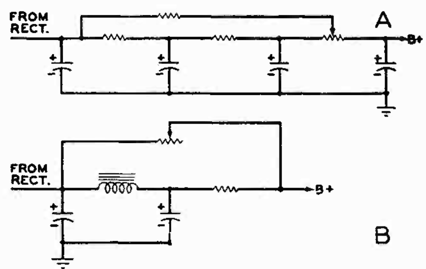



*Fig. 10. Feed-through hum cancellation.*

filtering is obtained from the decoupling network which follows, as we shall discuss later.

There is a useful trick for reducing ripple with RC (and one-choke) filters. This is shown in Fig. 10. You will see that a resistor is connected from the input of the filter to a point following the second section. This is a form of feed-through. The ripple is out of phase at the 2 points and, if the feed-through ripple is equal to the ripple existing at the point to which it is applied, cancellation occurs. The value of the feed-through resistor is roughly 10 times the value of the 2 resistors in series (or, if a choke is included, the sum of the reactance of the choke and the resistance of the resistor). Cancellation can be adjusted either by making the feed-through resistor a rheostat, or by adding a small pot at the end of the filter and adjusting it for a null in hum. This method can produce complete suppression of the fundamental ripple frequency; it is not as effective on the harmonics, but in most cases the harmonic ripple is no worse and usually a little better than that without feed-through.

## Choke-Input Design

It has been emphasized that the design of a choke-input filter is complicated and critical and that, unless you're very careful, it is easy to run into trouble. Fortunately, there is very little application in audio today for choke-input filters; the introduction of the 5U4-GB with its higher current capacity and lower internal resistance has removed most of the marginal needs for choke input. A few comments, however, will not be amiss in this elementary treatment.

A choke-input filter is desirable only when the current swing is very wide and the regulation must be very good. A Class B power amplifier of very high output (100 watts or more) is just about the only audio device that poses such a problem. I mentioned above that the 5U4-GB tube had eliminated the need for marginal applications. Consider, for example, the case of a Class AB2 amplifier using a pair of 6L6's or KT66's with fixed bias for an output of 50 watts. Plate voltage can be 375 to 510 volts, and the current swing (of the output tubes only) will be from 80 ma with no input to 175 ma with maximum input. If we use choke input and a 5U4-G we will need 550 volts AC per plate and we will get about 470 volts DC at the no-signal point and about 450 at the maximum-output point. This yields a swing of only 20 volts, which is very fine regulation indeed. But let us see just how much improvement if any this fine regulation offers over that possible with capacitor input.

With a capacitor-input filter the 5U4-G could deliver 510 volts at the zero-signal point and about 425 at the full-output point, with an input of only 425 volts AC per plate. Assuming an efficiency of roughly 60%, the amplifier would deliver about 47 watts with choke input and 44.5 watts with capacitor input. The 5U4-GB, however, reduces and in fact eliminates the disparity. With the same 425 volts per plate, it would deliver the same 510 volts DC at zero-signal point; but at the 175 ma full-output point the voltage would be around or just under 470 volts, for an output of close to 50 watts. Clearly, the additional complications of choke input are not justified. In such operation it is very much more important to regulate the voltage to the screens, and this problem is very little (if any) worse with capacitor input.

The problem teeters in favor of choke input, however, when more than 50 watts are needed. First, the plate voltage needed will exceed 500 volts, and this surpasses the maximum ratings of most electrolytic capacitors. With bigger tubes, the current swing may exceed 125 ma between no-signal and full-output conditions, so that here choke input may be worth while and possibly cheaper.

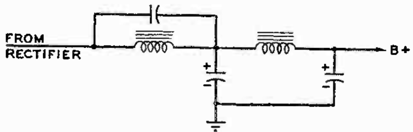



*Fig. 11. Tuning a choke to reduce ripple.*

Recently added charts in the tube manuals greatly simplify the design of choke-input filters. The inductance of the input choke is the most important consideration. We have already seen that the inductance must be larger than a certain critical value at both the minimum and maximum load conditions; this rules out the ordinary choke when the current load is high. Ordinary chokes are designed to hold the inductance fairly constant with variations in load. A choke which might have critical or optimum inductance at maximum load might easily have too little inductance at minimum load. To solve this problem many chokes are made with a large gap so that the inductance rises steeply as the current decreases. These are called swinging chokes, and are rated at 2 inductance values. For example, a 5- to 20-h swinging choke would have a minimum inductance of 5 h at the maximum rated current, and a minimum of 20 h at about one-third the maximum current.

The need for a swinging choke is illustrated by the example just considered. You will note that to obtain about 450 volts at the zero-signal, 80-ma load a minimum inductance of 5 h is needed. Suppose that we used a choke having only 3 h at that load. Then we would get about 450 volts at the 175-ma maximum-output point, but at the zero-signal point the voltage would rise to more than 510 volts. This would give us no better regulation than we could obtain with a capacitor-input filter, and there would be no point to using choke input since the same choke would work very well in a capacitor-input filter.

When designing a choke-input supply, therefore, the important thing is to choose a choke of sufficient inductance to keep the voltage from rising unduly at the minimum load point. The choke boundary lines on the graphs permit one to do this quite simply; it is only necessary to keep in mind that both the minimum- and maximum-load points must fall to the right of and below the dashed line for a given inductance. Or, conversely, choose a choke whose radial load line is always to the left of and above the operating point for minimum load. For example, if the minimum current does not fall below 200 ma, you can get up to 450 volts DC at about 280 ma with a 2-h choke. But suppose you want 300 volts with a current swing ranging between 50 ma (zero-signal) and 300 ma (maximum output). You can obtain this from two 5U4-GB's with an input of about 400 volts AC per plate, but you will need a choke higher than 5 h; otherwise the rise in voltage at the zero-signal point will be excessive.

It happens occasionally that the minimum load is lower than that permissible for a choke of a given inductance. You can fix things by terminating the filter with a bleeder resistor drawing enough current to raise the total current at zero signal above the safe minimum. Say, for example, you have a 5-h choke but at the minimum load the equipment will draw only 40 ma. This would lead to a violent rise of voltage. You can add a bleeder that will draw 25 to 40 ma, effectively raising the minimum load to 65 or 80 ma. This would get you by the critical knee and would provide good regulation.

One advantage of choke input is that the filter capacitors do not handle such high peak voltage or ripple currents. This is important when the supply voltage is close to the maximum rated working voltage of the capacitors. In a 450-volt supply, 450- or 500-volt electrolytic capacitors will be much safer with choke than with capacitor input. There is another trick, too, for improving the filtering of a choke-input filter. It consists of tuning the choke by inserting capacitance in parallel. For a ripple frequency of 120 cps (applicable to full-wave rectifiers) the product of L (in henries) to C (in $\mu$fd) necessary to achieve resonance is approximately 1.7. To find the approximate tuning capacitance for a given inductance, divide the inductance into 1.7. With a choke of 8 h, then, you would need a capacitance of approximately 0.2 $\mu$fd; with 10 h you would need about 0.17 $\mu$fd; with 2 h you would need a little over 0.8 $\mu$fd, and so on. You will usually have to adjust the capacitance by trial and error. Take a good-quality paper capacitor, Fig. 11, smaller than that calculated, then parallel it with even smaller capacitors until you obtain lowest hum. Again, this measure is effective on the fundamental ripple frequency, but not on the harmonics, though in most cases the harmonic ripple will be better with tuned than with untuned filters.

## *Yours for the Making . . .*

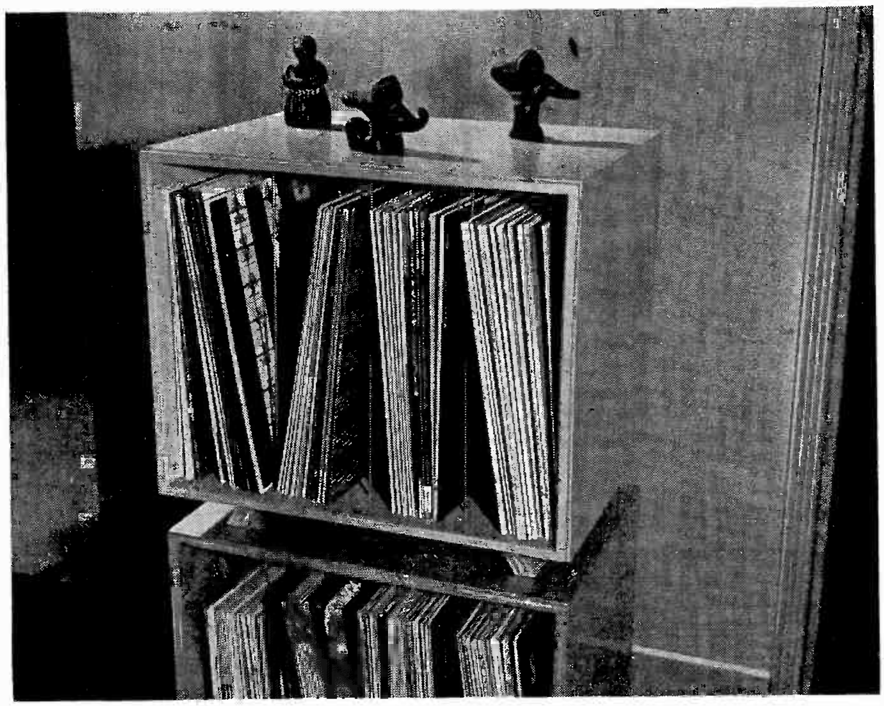
# Attractive Record-Storage Cabinet

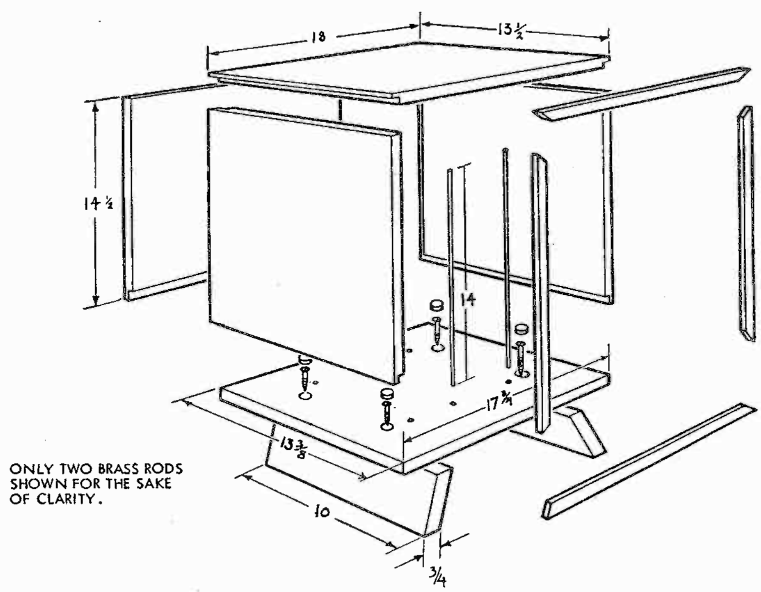
YOUR precious LP's will be protected against accidental damage and warpage by this handsome record storage cabinet, which is designed to hold 100 discs upright in small groups. Cut from ½-inch birch plywood, the cabinet is glued and clamped together; the only nails used are those through the bottom into the sides and back. Visible putty holes are thereby avoided. The feet are held in place with screws whose heads are hidden under glued plugs. Six holes ¼ in. deep are drilled inside the top and bottom pieces for the brass separator rods before the cabinet is assembled. The front grain-edge can be hidden with any suitable molding which is carefully mitred, glued, and held with clamps until dry.

After sanding inside and out, the cabinet is finished with clear lacquer to bring out the natural blond beauty of the wood. Wax is then applied for added luster and protection. Other finishing methods or other woods can, of course, be used.

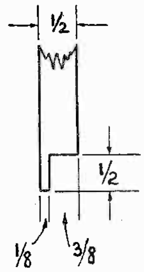
The separator rods are made of ⅛-inch brass wire stock which can be bought at most large hardware or metal stores. They are cut ⅜ in. longer than the inside dimension of the cabinet, polished with steel wool, and given a coat of clear lacquer to prevent tarnish. Bent slightly, they are installed in their holes after the cabinet is finished, and will straighten when released.

S. R. WILLIAMS






DETAIL OF ALL LAPS



by E. B. Mullings

# Using Test Instruments

## *Vacuum-Tube Voltmeters, Part III*

THIRD OF A SERIES OF ARTICLES ON TEST INSTRUMENTS AND HOW TO USE THEM.

VACUUM-TUBE voltmeters were introduced in Part I of this series. It was explained how the circuits within the VTVM function to make the instrument the valuable measuring tool it is.

Part II, which appeared in last month's issue, explained how the VTVM is used to make measurements. The actual mechanics of handling the test leads and controls of the instrument were described, and directions were given for making basic operating-potential measurements in electrical equipment. DC, AC, and resistance measurements were covered, and it was suggested that a chart of the normal operating voltages of your home sound equipment be prepared and retained for reference, so that new readings could be compared with the normal values in case some difficulty occurred.

Part III will be concerned with what can be done in the way of trouble shooting if one or more of the basic operating voltages is found far enough from normal to indicate trouble. While it is true that the results of such operating-potential tests are sign posts to indicate what is happening in the circuit, they are valuable only to the extent that you are able to interpret them and proceed with further tests to locate the defective component.

It is assumed that the reader is now familiar with the mechanics of operating a VTVM, and is thoroughly familiar with the normal precautions required for the protection of both the operator and the instrument. If you are unsure on either of these counts, reference should be made to the first 2 parts of this series.

### Power-Supply Tests

*No B+*: One of the basic tests for determining the operating condition of the power supply is to measure the B+ voltage. This is the high-voltage DC output of the supply measured between ground and the point marked "B+", Fig. 1. If no B+ voltage is present at this point, the first step is to be sure that primary power is reaching the power transformer. This includes checking to see that the wall socket is live, that the switch is on, and that the fuse (where applicable) has not blown.

If this procedure does not reveal the source of trouble, the next logical step is to have the rectifier tube in the power supply tested on a tube tester. Naturally, it should be replaced if it is found defective. If the rectifier tube is not the source of trouble, proceed to measure the AC voltage across each half of the high-voltage secondary winding. If no voltage is present on one half of the winding, remove power from the equipment under test and make a resistance measurement across the winding in question. The resistance reading will vary with the particular transformer being tested, but will probably be somewhere between 200 and 800 ohms. An "open" indication would indicate that the power transformer is defective and should be replaced. If, on the other hand, the high-voltage secondary winding is delivering its proper AC voltage, use the ohmmeter section of the VTVM to test the choke coil, because it may be open. Also, it may be that the rectifier is not getting filament voltage. (This would be obvious because the rectifier tube would not light up.) An AC voltage test across the filament terminals of the rectifier will indicate whether the tube is bad, or whether the AC voltage is not being delivered by the filament winding of the power transformer.

*No B+ and power transformer overheats*: The above test procedure is designed to locate the trouble when the symptom is merely a lack of B+ voltage output from the power supply. Should it be found that there is no B+ and, in addition, the power transformer is overheating or the fuse keeps blowing, a somewhat different procedure might be called for. While a lack of B+ suggests an open of some type, lack of B+ coupled with power transformer heat (or fuse-blowing) is suggestive of a short circuit. Have the rectifier tube tested to see if it is shorted, or test between elements with the ohmmeter section of the VTVM. Measure the resistance of the high-voltage secondary winding to see if either half is shorted. Finally, check the resistance of the input filter capacitor to see if it may be shorted or excessively leaky. This test requires that one lead of the capacitor be disconnected from the circuit so that a resistance reading will not be obtained through some other path. Polarity must be observed when checking electrolytic capacitors. The negative ohmmeter lead connects to the capacitor case, and the positive lead to the positive capacitor terminal. An electrolytic capacitor should normally measure higher than 100,000 ohms if it is in good condition. A lower resistance reading than this indicates that the capacitor is leaky, and perhaps completely shorted when power is applied, and the short is overloading the power transformer. The filter capacitor should be replaced.

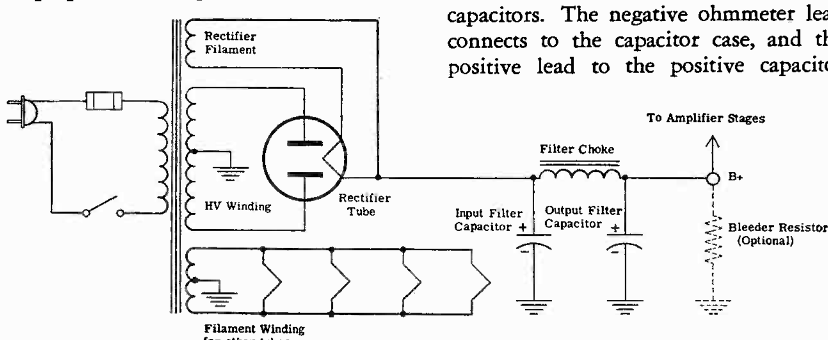



*Fig. 1. A typical amplifier power supply circuit. Defects in this section are common.*

*No B+, power transformer and choke overheat*: Should it be found that there is no B+ output from the power supply, the power transformer overheats and, in addition to this, the choke coil overheats, the first step would be to check the resistance of the *output* filter capacitor to see if it is shorted or leaky. The test procedure is the same as that for the input filter capacitor. If the capacitor is good, the trouble would most likely be caused by a short or ground external to the power supply; that is, in one of the amplifier stages connected to the B+ output of the power-supply circuit.

The best method of proceeding under these circumstances would be to try to isolate the short or ground to a particular section of the amplifier circuit. Inspection of the wiring of the circuit will undoubtedly reveal a junction point at which several B+ leads branch out to supply operating potentials to various stages in the circuit. If there is a short in the circuit, a resistance test between this junction and the chassis will reveal a lower-than-normal resistance reading (substantially less than the value of the bleeder resistor if there is one), or, if there is no bleeder, less than the resistance reading obtained across a good electrolytic capacitor. Start systematically disconnecting the leads branching out from this junction point one at a time and make a resistance test after each lead is disconnected. When the lead running to the circuit containing the defect is disconnected, the resistance reading at the junction point will rise to the normal value, thereby pointing out the defective section of the circuit. One can then repeat this procedure by following the defective lead through the circuit to the next junction point, and then systematically disconnecting one wire at a time until the defective one is located. This procedure will normally lead you directly to the defective part or connection.

*Low B+*: If the symptom being investigated is not a complete lack of B+, but merely lower-than-normal B+ voltage, the first test is to measure the line voltage to see if it is below normal. This could be the whole source of trouble. But if line voltage is satisfactory, have the rectifier tube tested to see if it is weak. If the rectifier tube is all right, test the high-voltage secondary winding by measuring the voltage across each half to see if it is up to normal. If the voltage is below normal at the high-voltage secondary winding, the winding may have shorted turns, reducing its voltage output. Remove all power and measure the resistance across each winding to verify this. Resistance readings will be lower than the normal value established previously if turns are shorted. Test both the input and the output filter capacitors for some degree of leakage. It may be that the leakage is sufficient to pull the voltage down without being great enough to overheat the power transformer or choke coil. If this series of tests fails to reveal the source of trouble, there is probably an indirect short between some point along the B+ supply to the tube plates and screens, and the chassis. Therefore, the trouble is external to the power-supply circuit, and the procedure for localizing the trouble by disconnecting leads at the junction points while making resistance measurements should be followed.

*High B+*: Higher-than-normal B+ voltage is very often caused by higher-than-normal line voltage. Therefore, this should be one of your first tests. Measure the AC line voltage at the wall socket with your VTVM. If the line voltage is normal, however, the bleeder resistor (if the circuit contains one) should be tested with the ohmmeter section of your VTVM. If the bleeder should become open, it will cause the B+ voltage to rise above normal. Also, an open in the B+ supply to one of the stages in the amplifier will reduce the load on the power supply and allow the B+ voltage to rise above normal. Resistance tests between B+ and the plate and screen of each tube in the circuit should be made. Reference to the schematic diagram of the circuit under test will enable you to compute the proper total resistance value for each measurement, and compare the actual result with what should be obtained according to the schematic.

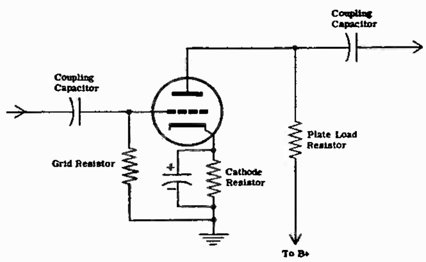



*Fig. 2. Triode voltage-amplifier stage.*

## AC Filament-Supply Tests

*No filament voltage*: The filament circuit is normally considered a part of the power supply, even though the circuit consists of the filament winding of the power supply and the wires leading out to the filaments of all the tubes in the amplifier circuit. This is a simple parallel circuit, as indicated in Fig. 1. If it is impossible to obtain an AC voltage reading across the tube filaments, test the filament voltage right at the transformer winding. If you still obtain no filament voltage, check the resistance of the winding by itself and it may be found open. If voltage is present at the filament winding, but not at the filament terminals of the tubes in the circuit, use the VTVM as an ohmmeter and check the continuity of the leads, looking for an open somewhere between the filament winding and the tube filament terminals. If only one tube is not lighting up, don't overlook the possibility that the tube may be burned out.

*No filament voltage and transformer hot*: If it is found that there is little or no filament voltage reaching the tubes, and at the same time the power transformer is excessively hot, it is much more likely that there is a short in the circuit rather than an open. Remove the tubes one at a time (but not the rectifier) and leave the test leads connected to measure the AC voltage across the filament winding of the transformer. If removing one of the tubes results in the filament voltage rising to its normal value, this particular tube is defective and should be replaced. However, if removing tubes does not alleviate the situation, remove the connections leading out to the tubes from the filament winding, and make the measurement across the winding itself. Then, if the measurement shows normal filament voltage, there is a short somewhere in the wiring to the tube filaments. If, when power is applied to the primary, you still do not get filament voltage, even with the circuit disconnected from the filament winding, the winding is apparently defective and the transformer should be replaced.

## Voltage-Amplifier Stage

*No plate voltage*: If no plate voltage is obtained in measuring the operating potentials of a voltage-amplifier stage, the first test would be to check the B+ output of the power supply. If none is available, the information contained in the previous paragraphs will lead to the source of the trouble. Assuming that the power supply is delivering B+ but no plate voltage is obtained, adjust the VTVM for making DC voltage measurements and, with the negative test clip on the chassis, connect the positive test probe to B+ and then move it along in the circuit in steps toward the plate of the voltage-amplifier tube to see where the voltage is lost. Somewhere in stepping along the circuit toward the plate you will reach a point where voltage is no longer obtained. At that point you will have just moved your test probe beyond an open in the circuit, or will have reached a junction where there is a short between the high voltage and the chassis. Perhaps this can be made more clear by reference to Fig. 2. The plate-load resistor in this circuit is labeled. If you are able to obtain B+ voltage when connecting the test lead just below this plate-load resistor but with the test lead on the upper side of the plate-load resistor *no* voltage is obtained, it would indicate either that the plate-load resistor is open or that there is a short between this part of the circuit and the chassis.

The next step would be to remove power and use the ohmmeter section of the VTVM to determine whether the trouble is due to an open or a short. Testing across the plate-load resistor with the ohmmeter and comparing the results with the color-coded value of the resistor will immediately indicate whether or not it is defective. Testing for a short can be done by measuring the resistance between the plate terminal of the tube and the chassis, to see that the resistance value is at least as high as the value obtained when measuring between B+ and the chassis. If a partial or complete short is present the resistance reading will be lower than normal. The trouble would probably be due to an actual physical short between leads, although the possibility of a shorted tube should not be overlooked.

*Low plate voltage*: If the symptom in this same amplifier stage is low plate voltage, rather than no plate voltage at all, the first step is to test the B+ voltage as described previously to see if it is up to normal. Assuming B+ to be normal, an ohmmeter test should be made of the plate-load resistor, or any other resistance elements that may be in series between the plate of the tube and B+. It is not unusual for a resistor to change value with time. If this is not the trouble, test the resistance between the plate terminal of the tube and chassis to see if there is a leakage path that may have developed across an insulated surface, or through a bypass capacitor or coupling capacitor, if the circuit uses them. If none of these seems to be the reason for the low plate voltage, it may be that the bias (the difference in DC potential between the cathode and control grid of the tube) is lower than normal, causing heavier-than-normal plate current. A leaky input coupling capacitor, or a short circuit between grid and cathode, would cause this. The excessive plate current would produce a larger voltage drop across the plate-load resistor, and the resulting plate voltage would be lower than normal.

*High plate voltage*: Another condition you might run into is higher-than-normal plate voltage. As in the previous cases, a measurement of the B+ voltage would be your first step in looking for the trouble. Assuming normal B+, higher-than-normal plate voltage usually suggests that there is little or no plate current flowing in the stage, and therefore little drop across the plate-load resistor. This results in a plate-voltage value that is practically equal to the B+ value, when it should be a good deal lower. There are 4 likely causes of trouble. One would be an open tube filament. This could be checked easily by measuring the resistance between the filament terminals of the tube with your ohmmeter. Also, it may be that the tube is not receiving filament voltage. This would prevent current from flowing through the tube. There may be an open between the cathode of the tube and the chassis or ground. A resistance test between these 2 points should give you a reading equal to the cathode resistor's value. A open cathode resistor is not an uncommon trouble. Finally, the tube may be defective because of decreased emission or an open internal connection other than the filament.

*Bias voltage too high or too low*: Should it be found that the DC bias voltage developed across the cathode bias resistor is either too high or too low, the trouble can probably be narrowed down to one of 4 possible conditions. The first of these would be that the tube itself is defective, since reduced plate current would result in low bias voltage. The proper bias value is dependent on the fact that the tube is supplied with the correct plate voltage. Therefore incorrect operating potentials would be another possible source of wrong bias. Third, the value of the cathode resistor used in the circuit may have changed, for reasons similar to those given for the plate-load resistor changing value. A simple resistance measurement between the cathode and the chassis, and a comparison of the resulting resistance reading with the color-coded value of the part would answer this question. Finally, the input coupling capacitor or bypass capacitor connected across the cathode resistor may have become leaky.

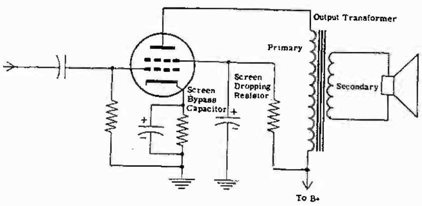


*Fig. 3. Single-ended power output stage.*

*No bias voltage*: In a case where no bias voltage at all is obtained, it's very likely that the tube itself is burned out, or that it isn't getting filament voltage. A complete lack of plate voltage would be a possible source of trouble, as well as a completely open cathode resistor, or a direct short in the cathode bypass capacitor. All these conditions can be checked with the VTVM.

## Power-Amplifier Stage

A typical power-amplifier stage is shown in Fig. 3. So far as measurements are concerned it differs only slightly from the voltage-amplifier stage. The main difference is in the function of the circuit. The voltage amplifier is, as its name implies, designed for voltage gain, while the power amplifier is designed primarily for high current and power gain, in order to drive a loudspeaker. Modern power amplifiers often use a beam power tube with a screen grid. Also, the plate circuit of the tube normally works into an output transformer rather than into a resistor. The various tests for locating trouble in the plate circuit would still be the same, however, and the procedures outlined for an amplifier using the plate-load resistor can be applied to a power amplifier in which the primary of an output transformer is used instead of a plate-load resistor. The DC resistance value of the transformer primary is considerably lower than the value of the plate-load resistor, so that normally the plate voltage is only slightly lower than the B+, but otherwise the plate circuit is the same. The screen grid in a power-amplifier stage is supplied with voltage either directly from B+, through a resistor, or through a portion of the output transformer from B+. In addition, the screen grid is often bypassed to ground with a capacitor.

*No screen-grid voltage*: No screen-grid voltage at all in the power-amplifier stage would suggest that the B+ voltage should be tested first, as described previously. Assuming that B+ is present, a shorted bypass condenser would be a likely source of trouble. A resistance test of the bypass capacitor with one lead of the capacitor disconnected from the circuit will indicate its condition. If it is a paper capacitor, the resistance reading should indicate nearly "infinity" on the VTVM ohmmeter. The normal resistance of a good paper capacitor is considerably higher than the 100,000-ohm minimum value given for electrolytic capacitors. Also, there could be leakage or a short to ground across an insulating surface, or a lead of the resistor may be touching a grounded terminal or wire. The screen-grid dropping resistor may be open, and a resistance test across the resistor would reveal this trouble. Finally, the tube itself may be bad in that the screen grid is shorted to some other tube element. Don't overlook a defective tube as a possible source of trouble.

*Low screen-grid voltage*: If screen-grid voltage is present, but is lower than it normally should be, the screen-grid dropping resistor may have changed value. It should be tested with your ohmmeter. Slight leakage may have developed in the bypass capacitor, or a leakage path may have developed across insulated surfaces from moisture or dust collecting between 2 terminals.

You will note that the circuit given in Fig. 3 is single ended, in that a single output tube is used to drive the loudspeaker. Most modern high fidelity amplifiers use push-pull output stages in which 2 tubes are used in push-pull operation to drive the speaker. Push-pull stages may be treated the same as

*Continued on page 41*

# BASIC ELECTRONICS

## *IV: Power; Series and parallel circuits.*

by Roy F. Allison

### Power

Power is a rate of doing work: it is work done per unit time. If 1 joule of work is done in one second, the power is 1 watt. The formula for power (*P*) expressed in one form is, then,

$$P = \frac{W}{t}.$$

Since work is done when electric charge moves from one potential to another, and current is the time rate of flow of electric charge, there is obviously an intimate relation among power, current, and voltage. This can be derived easily, as follows: The formula for voltage is

$$E = \frac{W}{Q}.$$

This can be restated by cross-multiplication as $W = EQ$. Substituting $EQ$ for $W$ in the power formula above, we obtain $P = \frac{EQ}{t}$. This is identical to

$$P = E\frac{Q}{t}.$$

Now, the formula for current is

$$I = \frac{Q}{t}.$$

Substituting $I$ for $\frac{Q}{t}$ in the formula above, we obtain $P = E\,(I)$ or $P = EI$. That is, power in a resistance is equal to the voltage across that resistance times the current through it. Again, by substi-

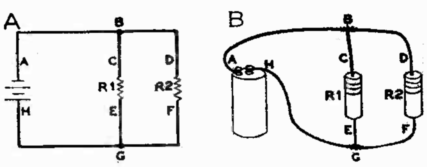



*Fig. 1. Schematic and physical drawings.*

tuting in this formula the relationships stated in the various versions of Ohm's Law, we obtain 2 other power formulas:

$$P = \frac{E^2}{R} \quad \text{and} \quad P = I^2R.$$

Power in a circuit or circuit element can, then, be found by multiplying the voltage by itself and dividing by the resistance; or by multiplying the current by itself and then by the resistance. If any 2 of the 3 Ohm's-Law quantities are known, the other can be found from them and power computed by the most convenient formula, or the power can be found directly by applying the pertinent power formula.

### Resistive Circuits

Amateur audio workers as well as professionals must use Ohm's Law and the power formulas at just about every turn, since they express the basic relationships among power, voltage, current, and resistance. Along with reactance (which will be discussed in subsequent chapters) these are the main tools of the circuit designer and experimenter. Their use, in fact, must become second nature to anyone who wants to be able to understand the operation of typical electronic circuits. For that reason the rest of this chapter will be devoted to an examination of simple resistive circuits and what goes on in them.

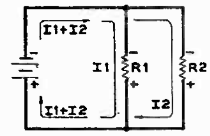
The matter of line resistance was mentioned previously only because it can be important in some circumstances — when the line must be very long, or when it must carry an extremely high current — and it would be dangerous to neglect it in those instances. But in home audio gear, particularly in electronic equipments such as amplifiers, tuners, and the like, wires connecting the circuit components are so short and the currents so relatively minute that their resistance can be considered negligible. Therefore voltage drop in the wires is quite insignificant, and we shall ignore this factor in subsequent calculations; interconnecting wires in circuits, shown as unbroken lines, will be considered to have no resistance and, accordingly, no voltage will be developed along them nor any power consumed by them.

To illustrate the application of this generality, consider Fig. 1A. This is a circuit consisting of 2 resistances connected across the terminals of a battery: schematic diagram notation is used, in which a resistance is represented by a zig-zag line. We are familiar with the symbol for a battery already. Fig. 1B is a drawing of what the corresponding physical hookup might look like. Similar letters indicate identical points on each diagram. Since there is negligible voltage drop in our connecting lines, obviously the points marked A, B, C, and D must all be at the same potential — must be at the same voltage with respect to point H, for instance. In the same way points E, F, G, and H must all be at the same potential. Each resistor, then, must have the entire voltage of the battery applied across it.

Two other generalities are often of use in circuit calculations. Known as Kirchhoff's Laws, after Gustav Kirchhoff, they are self-obvious also but have even more significant ramifications than the first. Here they are:

1) *At any junction point in a circuit, the sum of currents directed toward the junction is equal to the sum of currents directed away from the junction.* Fig. 2

*Fig. 2. Currents in circuit of Fig. 1.*




is Fig. 1A redrawn without all the letters, but with currents indicated. The battery produces current through both R1 and R2, since both are directly across its terminals, and the current directions are shown by the arrows. The negative battery terminal forces electrons outward and the positive terminal attracts them, so current flows downward through the resistors and the resultant voltage developed across them has the polarity shown. The total battery current entering the upper junction and leaving the lower junction is certainly the sum of the currents through R1 and R2.

2) *The algebraic sum of the voltages around any closed circuit is equal to zero.* We have seen already that, since line resistance can be ignored in most cases, resistors R1 and R2 are effectively across the battery terminals directly, and the full battery voltage is developed across them. Assume a battery of 10 volts. If we begin clockwise from the lower junction, we encounter a plus-to-minus 10 volts for the battery and then a minus-to-plus 10 volts for either R1 or R2, whichever path we take back. Plus 10 and minus 10 in either case is zero. Or, beginning at the lower junction and going first through R1 and then back through R2, the same voltages are encountered with the same

result. It works the other way around, too — try it! The rule holds true even if line resistance *isn't* negligible, of course; line resistance would be treated simply as another resistance, and the voltage drop across it would be of the same polarity as for any other external resistor.

There is one hidden resistance that cannot be ignored, though, in many cases. That is the internal resistance of the voltage source: in this case, the battery. Any producer of voltage — battery, generator, amplifier power supply, even a vacuum-tube amplifying stage — has its own resistance to the current that it may produce in a circuit. This is known as its *source resistance*. The more

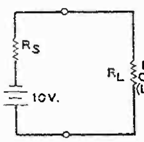



*Fig. 3. The source resistance shown in series with a load.*

current a circuit obtains from the source, the more current flows through the source resistance and the more voltage is lost across it; hence, the lower is the actual terminal voltage of the source. Paradoxically, maximum voltage at the terminals of a voltage source can be obtained only when no current is circulating! Source resistance (Rs) must be taken into account whenever it is large enough to be significant compared to the total resistance of the external circuit (the *load*). In such cases it is considered to be a resistance in series with the source terminals, as in Fig. 3, and the source voltage is assumed to be that obtainable at the terminals with no external circuit connected. When no source resistance is shown in diagrams used herein, it means that Rs is small enough to be ignored.

## Resistances in Series

Fig. 4A is a schematic diagram of 2 resistors in series across a voltage source; Fig. 4B is a drawing of the physical arrangement. All the battery current must flow through both resistors, since there is only one path in the circuit. The more resistors are wired in series, the greater the total resistance of the combination. It is obvious that *the total resistance value of resistances wired in series is the sum of the individual resistances.* (In Fig. 4, as in most schematic diagrams, symbols are used to indicate resistance values. The capital Greek letter Omega — Ω — is the symbol for ohms; the capital Roman letter K indicates thousands of ohms. Thus R1 is 500 ohms, and R2 is 1,500 ohms. For other symbols refer to the list of symbols and abbreviations on page 46.)

Now, suppose we wished to know the voltage across R1 and R2, the power consumed in each, and the total power consumed in the circuit. We know only their individual resistances and the battery voltage. To find the voltage across each, the current through them would be helpful; then, Ohm's Law (E=IR) could be used. Very well: all the current through the battery must go through each resistor, and since we know the battery voltage we could find the circuit current if the total circuit resistance were known too. This is the sum of R1 and R2, 500 + 1,500, or 2,000 ohms. Current is then

$$I = \frac{E}{R};\ I = \frac{10}{2{,}000};\ I = .005 \text{ amp or } 5 \text{ ma.}$$

This current flows through both resistors. So the voltage drop across R1 is E = IR; E = .005 × 500; E = 2.5 volts. And the voltage drop across R2 is E = IR; E = .005 × 1,500; E = 7.5 volts. We could have found the voltage across R2 by another, perhaps simpler means, using one of Kirchhoff's Laws. The sum of the voltages around a closed circuit must be equal to zero. If the battery voltage is 10, and that across R1 is 2.5, then the voltage across R2 must be equal to the difference, or 7.5 volts.

Here is another significant point: the voltages across the resistors divided in direct proportion to the ratios of the resistance values. Further, each resistor has a voltage drop in proportion to its ratio to the total resistance in the circuit. R1 has ⅓ the resistance of R2, so that it has ⅓ the voltage drop of R2. R1 has ¼ the total circuit resistance, and R2 has ¾ the total; accordingly, ¼ of the source voltage is dropped across R1 and ¾ across R2. These relations apply only in series circuits, of course, or for resistances in which the same amount of current flows.

Since all 3 values (resistance, voltage, and current) are now known for R1 and R2, any of the 3 power formulas can be used to find the power consumed in them. Perhaps most convenient at this point is P = EI. For R1, then:

P = EI; P = 2.5 × .005; P = .0125 watt or 12.5 mw.

And for R2:

P = EI; P = 7.5 × .005; P = .0375 watt or 37.5 mw.

For the entire load:

P = EI; P = 10 × .005; P = .05 watt or 50 mw.

Observation reveals that *the total power consumed in the circuit is equal to the sum of the powers consumed in the individual elements,* which is certainly reasonable. Further, the division of the

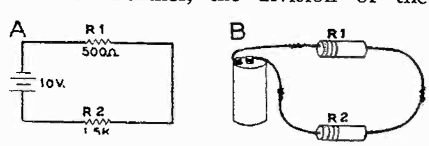



*Fig. 4. Series circuit drawn in 2 ways.*

total power is exactly in the same proportion as the individual voltage drops. Another general rule, then, is *for resistances in which equal currents flow, voltage drop and power consumption in individual resistances are directly proportional to the resistance values.* These rules are valid in any circuits, and are sometimes helpful in avoiding laborious calculations. Let's see how helpful they can be.

Fig. 5 is another series circuit containing 3 resistances. The values of all 3 are known, together with the voltage

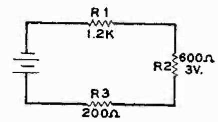



*Fig. 5. Schematic diagram of 3 resistances in series.*

drop across R2. What are the voltage drops across the other resistors, the battery voltage, and the power consumption of each resistor and in the entire circuit? First, we should ordinarily expect to find the current as a starting point, and we could do it because we know the voltage drop and resistance of R2. But it isn't necessary if we apply the proportion rules. Let's find the power in R2 instead:

$$P = \frac{E^2}{R};\ P = \frac{3 \times 3}{600};\ P = \frac{9}{600};$$

P = .015 watt or 15 mw.

The resistance of R1 is twice that of R2, so its voltage drop is 2 × 3, or 6 volts; the power consumed is 2 × 15, or 30 mw. The resistance of R3 is ⅓ that of R2, and its voltage drop is accordingly 3/3, or 1 volt; its power consumption is 15/3, or 5 mw. Now, applying the sum rule, the battery voltage must be 3 + 6 + 1, or 10 volts. The total power in the circuit is 15 + 30 + 5, or 50 mw.

One more series-circuit example. It is quite often desired to "tap off" part of the voltage from a source: that is, to reduce it to a lower amplitude suitable for the application at hand. Suppose we have a load of 500 K, for which we want only 50 volts (to make calculations easy), and a source with a terminal volt-

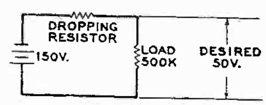



*Fig. 6. Simple potential-divider circuit.*

age of 150. The obvious procedure would be to put a resistance in series with the load, as in Fig. 6, so as to reduce the voltage applied to the load. But of what value should the dropping resistor be? With the desired 50 volts across the load, the current is

$$I = \frac{E}{R};\ I = \frac{50}{500{,}000};\ I = .0001 \text{ amp.}$$

The dropping resistor must take up 100 of the 150 volts at the source, so its value must be

$$R = \frac{E}{I};\ R = \frac{100}{.0001};\ R = 1{,}000{,}000$$

ohms, or 1 megohm.

Again, this might have been calculated

by the proportion rule; the dropping resistor had to absorb twice as much of the source voltage as the load resistance, so that its value should have been twice that of the load. This would be 2 × 500,000, or 1 megohm.

## Resistances in Parallel

Resistances connected as shown in Fig. 1 are said to be in parallel. They are joined at both ends and may be connected to the source directly across its terminals or, as a combination, in series with other resistances. To begin with, let's examine a simple parallel combina-

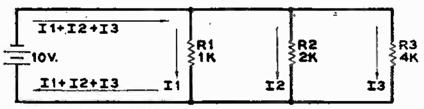



*Fig. 7. In a parallel circuit, separate currents flow in each of the resistances.*

tion of 3 resistances across a 10-volt battery, as in Fig. 7. R1, R2, and R3 are each across the battery terminals individually, so each has the battery's terminal voltage. It is obvious as well that each must have the *same* voltage across it as the others in the parallel combination, no matter how the combination is used with other resistances in the most complex circuit. So long as the resistances are joined at both ends there must be the same potential difference across them. Herein lies the reason for the basic differences in series and parallel circuits: resistances in series have a common current but different voltage drops; resistances in parallel have a common voltage drop but different currents.

If R1 alone were attached to the battery in Fig. 7, the current through the battery would be only that taken by R1 as the load.

$$I1 = \frac{E}{R1};\ I1 = \frac{10}{1{,}000};\ I1 = .01 \text{ amp.}$$

Adding R2 across the terminals also, in parallel with R1, obviously cannot have any effect on the current through R1, since the voltage across R1 has not changed. But R2 takes current of its own from the battery:

$$I2 = \frac{E}{R2};\ I2 = \frac{10}{2{,}000};\ I2 = .005 \text{ amp.}$$

So far as the battery is concerned, adding R2 did *not* increase the total resistance of the load. Since more current is drawn, the total load resistance has decreased. In the same manner, adding R3 decreases the total load resistance still further because more current is taken:

$$I3 = \frac{E}{R3};\ I3 = \frac{10}{4{,}000};\ I3 = .0025 \text{ amp.}$$

The total resistance of resistors in parallel, then, is always less than the resistance of any of them singly. In this case the total current in the battery is I1 + I2 + I3; .01 + .005 + .0025 is a load current of .0175 amp. Total load resistance is

$$R = \frac{E}{I};\ R = \frac{10}{.0175};\ R = 571 \text{ ohms.}$$

This value is called the *resultant*; it represents an equivalent resistance for a combination of resistors in parallel. The *lowest* resistance in the combination has most effect on the resultant, and the highest resistance least effect. This can be deduced by comparing the relative figures for I1, I2, and I3. R1, the lowest resistance value, accepts most current from the battery; R3, the highest resistance, accepts least. I1 makes up most of the total current that determines the resultant, and I3's contribution is relatively small. If still another resistor of, say, 100 K were added to the present combination, its effect on the total resistance would be negligible. On the other hand, if one of 300 ohms were added it would become the determining value in the resultant.

This method of computing the resultant of parallel resistances—adding the individual currents and using Ohm's Law with the total figure and the voltage across the combination—is fine when you *know* the voltage, and so can compute the individual currents directly. But very often you don't know this, particularly when the parallel combination is in series with other resistors or circuit elements. Then the relationship between the resultant and the other circuit resistances determines how much voltage appears across the parallel group; and to know the relationship you must know the resultant. So you must be able to calculate the resultant directly from the resistance values in the combination. This is the general formula for that purpose:

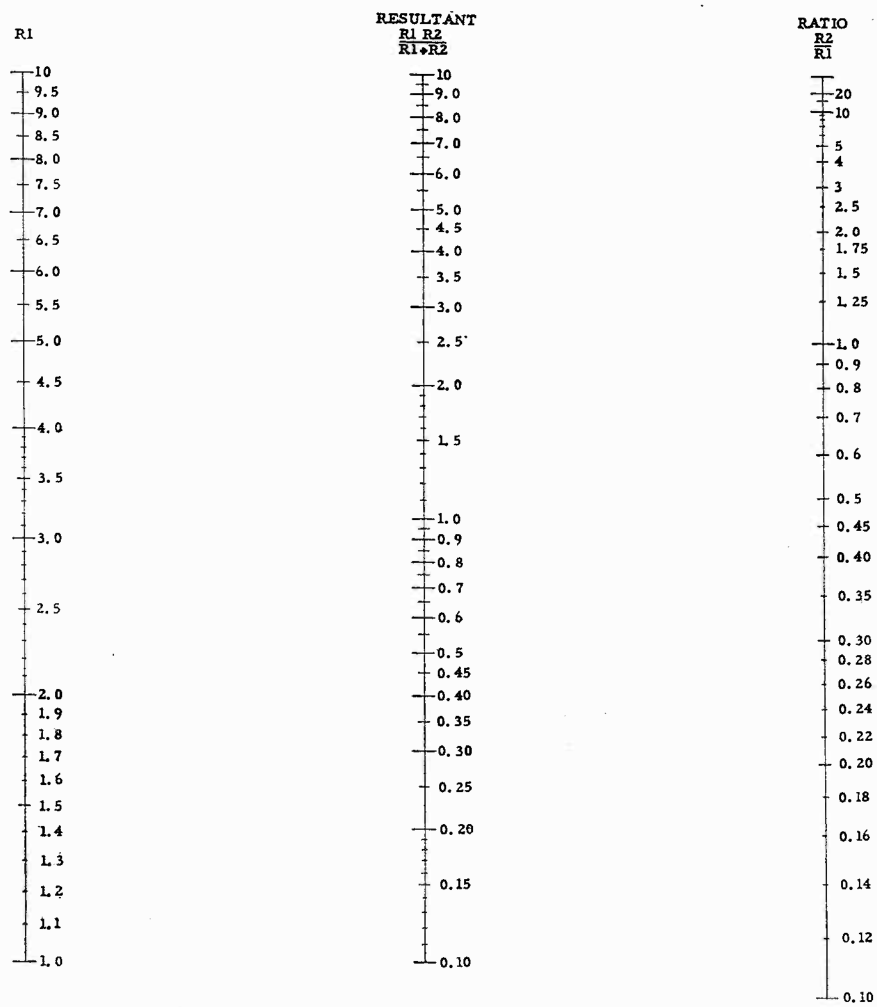
$$R = \frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \frac{1}{R4} + \ldots}$$

It's formidable, all right, but there's a way to avoid using it. When only 2 resistances are involved the formula reduces to

$$R = \frac{R1R2}{R1 + R2}.$$

Using the formula above we can calculate the resultant of the first 2 resistances in a parallel combination; then, considering the resultant as one resistor, compute a subsequent resultant with the next resistor in the combination, and so on. If there are 4 or fewer resistances to deal with it is easier to go at it this way than to use the general formula. Let's see how it works out for the resistances in Fig. 7:

*Fig. 8. A nomograph for resistances in parallel. Operation is explained in the text.*



$$R = \frac{R_1R_2}{R_1 + R_2}; \quad R = \frac{1{,}000 \times 2{,}000}{1{,}000 + 2{,}000};$$

$$R = \frac{2{,}000{,}000}{3{,}000}; \quad R = 667.$$

That's the first operation. Now, considering the resultant (667) as R1 and calling R3 (Fig. 7) R2 for the formula, we have

$$R = \frac{R_1R_2}{R_1 + R_2}; \quad R = \frac{667 \times 4{,}000}{667 + 4{,}000};$$

$$R = \frac{2{,}668{,}000}{4{,}667}; \quad R = 571.$$

This is the same answer obtained by the current-addition method; its advantage is that voltage across the parallel group need not be known.

In order to simplify these calculations even further, I've prepared the nomograph in Fig. 8. This is for use with 2 resistors at a time also. To use it, first divide R2 by R1. Find the ratio in the column at the right. Then find the first few significant figures for R1 in the column at the left. Connect the 2 points with a straightedge, and where it crosses the center column read off the resultant. The value is, of course, related to that of R1. For example: consider resistances of 60 K and 120 K; what is their parallel resultant? Call the 60-K resistor R1 and the 120-K resistor R2. The ratio of R2 to R1 is 2. Putting a straightedge between 2, column 3, and 6, column 1, column 2 is crossed at the value 4. The resultant is, then, 40 K. If you had called the 120-K resistor R1 and the other R2, the ratio would have been 0.5. Connecting 0.5 in column 3 with 1.2 in column 1, the rule would cross the center column at value 0.4, so that the answer would again have been 40 K. It works both ways; you can't go wrong as long as you keep the decimal places right.

In practical circuit calculations, tolerances are such that it's a waste of time to consider any parallel resistor that is 20 times or more larger than another. Simply take the value of the smaller as the resultant, and forget about the other.

The nomograph can be used in another way as well. Suppose you have a resistance of 10 K and want to parallel it with a resistor of such value as to produce a resultant of 6 K. A straightedge from 10 in column 1 through 6 in column 2 will cross column 3 at 1.5. The ratio of R2 to 10 K, then, is 1.5, and a 15-K resistor will do the job. The nomograph can be a real time-saver if you become proficient in its use, since only one multiplication or division is required for each operation. But the formula may be less confusing, and it's possible to carry it in your head — which you can't say for the nomograph.

Going back to Fig. 7, you'll recall that with 10 volts across the parallel group the 1-K resistor took .01 amp, the 2-K resistor .005 amp, and the 4-K resistor .0025 amp. *The currents are inversely proportional to the resistance values in a parallel group.* We should expect a 500-ohm resistor to take .02 amp if it were placed in the same group, and a 10-K resistor .001 amp, without bothering to compute it by Ohm's Law. This assumes that the voltage remains constant, of course, and it would in Fig. 7. Since voltage is the same for each resistance, and power consumed is a product of voltage times current ($P = EI$), it follows that power is inversely proportional along with the current: *The power consumed in individual resistances of a parallel group is inversely proportional to the resistance values.*

Now let's try out the rules and formulas on a circuit combining both series and parallel resistances, as in Fig. 9. Given are the battery current ($I_b$ = 0.1 amp), the voltage drop across R1, and the values of resistances R2, R3, R4, and R5. To find: battery voltage and voltage across R5; currents through R2, R3, and R4; power in each resistance and in the total load.

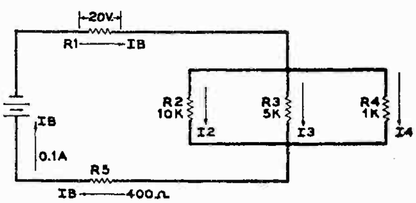



*Fig. 9. Series and parallel combinations.*

First, we know that the entire battery current flows through both R1 and R5. We can divide this into the voltage across R1 to find R1's value:

$$R_1 = \frac{E_1}{I_b}; \quad R_1 = \frac{20}{0.1}; \quad R_1 = 200 \text{ ohms}.$$

R5 is in series with R1, and is twice its value, or 400 ohms; consequently, there is twice as much voltage developed across R5: 2 × 20, or 40 volts. Next we must find the total resistance of the parallel group, either by formula or nomograph. By the 2-step formula:

$$R = \frac{R_2R_3}{R_2 + R_3}; \quad R = \frac{10{,}000 \times 5{,}000}{10{,}000 + 5{,}000};$$

$$R = \frac{50{,}000{,}000}{15{,}000}; \quad R = 3.333 \text{ K}.$$

$$\text{Resultant} = \frac{R\ R_4}{R + R_4};$$

$$\text{Res.} = \frac{3{,}333 \times 1{,}000}{3{,}333 + 1{,}000};$$

$$\text{Res.} = \frac{3{,}333{,}000}{4{,}333}; \quad \text{Res.} = 770 \text{ ohms}.$$

Knowing the current through this combination, we can find the voltage across it:

$E = I_b\text{Res.}$; $E = 0.1 \times 770$; $E = 77$ volts.

With this we can find the current through one branch — say I4:

$$I_4 = \frac{E}{R_4}; \quad I_4 = \frac{77}{1{,}000}; \quad I_4 = .077 \text{ amp}.$$

Since currents in a parallel group are inversely proportional to the resistances, and R3 is 5 times greater in value than R4, I3 must be 1/5 as large as I4, or .0153 amp. By the same reasoning I2 should be 1/10 I4, or .0077 amp. These should add up to the total battery current given as 0.1 amp; fortunately, they do.

To find the battery voltage we add the voltage drops around the circuit: 20 volts across R1, 77 volts across the parallel group, and 40 volts across R5. The sum is 137 volts. As a check, let's add the total resistance in the circuit (200, 770, and 400, or 1,370 ohms) and multiply it by the battery current: $E_b = I_bR_L$; $E_b = 0.1 \times 1{,}370$; $E_b = 137$ volts. Check.

Now for the power consumption. Power developed in R1 is $P = EI_b$; $P = 20 \times 0.1$; $P = 2$ watts. By the rule of proportionality the power in R5 is twice this, or 4 watts. Power in R4 is $P = EI_4$; $P = 77 \times .077$; $P = 5.92$ watts. By proportion again, power in R3 must be 1/5 that in R4, or 1.18 watts; that in R2 must be 1/10 that in R4, or 0.59 watt. Adding 2, 4, 5.92, 1.18, and 0.59 should give the total power consumption in the circuit; the sum is 13.69 watts. Checking this by multiplying total load voltage times total load current, $P = E_bI_b$; $P = 137 \times 0.1$; $P = 13.7$ watts. The difference is insignificant, and we can assume that it checks.

## Power Transfer

Consider at greater length the simple circuit in Fig. 3. Assume for convenience that the battery has an unloaded voltage of 10 volts, and that its source resistance is 10 ohms. The external circuit may consist of one or more resistances; what counts is the total external resistance, $R_L$, connected to the battery.

Suppose at first that we make the load 1,000 ohms. This is so much greater than Rs that the internal resistance can be ignored. Effectively, the full battery voltage is developed across the load, and the power in the load can be found easily:

$$P = \frac{E^2}{R}; \quad P = \frac{10 \times 10}{1{,}000};$$

$$P = \frac{100}{1{,}000}; \quad P = 0.1 \text{ watt}.$$

Now let us reduce the load to 100 ohms. Rs must be considered, since it is now 1/10 the load value, and in order to find the power it is necessary to calculate current in the circuit, because the entire battery voltage is not now dropped in the load.

Total circuit resistance is $R_s + R_L$, 10 + 100, or 110 ohms.

$$I = \frac{E}{R}; \quad I = \frac{10}{110}; \quad I = 0.91 \text{ amp}.$$

Power *in the load* is, then, $P = I^2R$, $P = .091 \times .091 \times 100$; $P = 0.83$ watt. Note that this is substantially more power than was obtained with a higher load resistance. Assume this time a load value (quite unrealistic, to be sure) of 1 ohm. By the same process given above the circuit current is found to be 0.91 amp, and power in the load is $P = 0.91 \times 0.91 \times 1$; $P = 0.83$ watt. Finally, let's make the load *equal* to the source resistance, 10 ohms. Now the current is:

$$I = \frac{E}{R};\ I = \frac{10}{20};\ I = 0.5 \text{ amp.}$$

And power in the load:

$P = I^2R$; $P = 0.5 \times 0.5 \times 10$; $P = 2.5$ watts.

What has been demonstrated here can be proved mathematically as well: *Maximum transfer of power to a load is achieved when the load resistance is equal to the source resistance.* This is the principle of "matching" loads to power sources; the greater the mis-match that exists between source and load, the less power is transferred to the load. When the mis-match is down — i.e., load less than source resistance — current is very high, and the source is said to be heavily loaded. Most of the source's voltage and power is developed across its own internal resistance. When the mis-match is up — the load is greater than the source resistance — most of the source's voltage and power is developed in the load.

This principle is true in AC as well as DC circuits. In any case, you match or mis-match loads according to what you're after; if you want maximum power transfer, as from a power amplifier to a loudspeaker load, a close match to the output stage is desirable. But if you want maximum efficiency or maximum voltage developed in the load, you will probably mis-match upward. There are other considerations that may govern, of course, but this discussion should convey an understanding of why so much emphasis is put on the matter of load matching.

## PLAIN AND FANCY

*Continued from page 21*

ticularly when audio signal lines have to be shifted even occasionally, both for their convenience and their sureness of firm contact — in the ground and low lines as well as the hot one. And for myself, I install them not only in all my main-system units but also in all my auxiliary and test equipments; this uniform audio-connector practice proves to be a time saver whenever any unit has to be set up on the bench for test and maintenance work.

Some other Fancy ventures which also have worked out well, if not as impressively so, include: Use of double-action, plier-type wire-strippers, which may not do a better job than simpler types, or even a knife, but certainly do it in a more elegant fashion. . . . Installation of a tuning meter or eye tube in any FM receiver which lacks it (although this might better be ranked as a necessity than an example of Fanciness) . . . . Use of miniature-tube pin straighteners and — when wiring sockets — dummy tubes (don't forget to remove the latter before switching on heater voltages!) . . . . Checking all resistor and capacitor values with a reliable ohmmeter and capacity checker *before* components are wired up and, in the case of some critical resistors, *after* they have been soldered in . . . . Never omitting anti-click resistors in any high-impedance audio signal line switching circuit, Fig. 6 . . . . And finally, the liberal utilization of turret-type sockets in equipment construction. For a long time I tried to combine the advantages of point-to-point wiring with those of resistor-board small-component mounting, but since the socket turrets came on the market I've found that resistor boards are seldom needed or wanted.

### Conclusion

Now, I fear that some of my more sophisticated readers may have found little here that strikes them as notably novel, or may hasten to point out that many of my Fancy ventures are nothing but borrowings from standard professional-studio practices. Quite true — but it's only too easy for experienced craftsmen to forget that to the *novice* every circuit or component or technique he hasn't encountered before is novel, and that one's first acquaintance with amateur-audio literature and the most easily accessible equipments, kits, and components designed for home markets provides for the most part a surprisingly conventional view of "normal" practices, sometimes encourages dabbling in unnecessary or risky complications, and

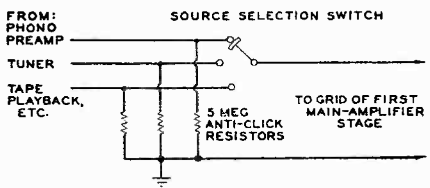



*Fig. 6. Quieting resistors for switches.*

seldom promotes a genuine appreciation of certain professional or laboratory procedures and components which are ideally suited to more widespread utilization or adaptation.

Anyway, many of the novices I have met or heard about strike me as either too smugly content with strict amateur-audio orthodoxy or too timorous about personal experimentation and occasional costly investments in really top-notch components. I have no illusions about the waste of time and money involved in most of my own Plain and Fancy ventures, yet I still feel that nothing which has helped me spend so many busy, hopeful, and instructive hours can have been entirely wasted. After all, one of the most appealing attractions of audiocraft is that there can be a lot of fun and mental stimulation even in achieving failures! And perhaps it is only against a background of failure that one can fully appreciate the rich gratification experienced every time one uses or looks back on an adventure in either Plainness or Fanciness which has turned out really well — and which, quite regardless of where the original stimulus came from, one has made (by purchase, adaptation, or labor) distinctively one's own.

## UNIVERSITY CLASSIC

*Continued from page 18*

sponse, and efficiency of the horn. It is, therefore, advisable to use as strong a panel structure as possible and to permit minimum opportunity for these panels to vibrate either by themselves or against one another. Rigid ¾-inch plywood, and glue-and-screw construction is recommended.

The 3-speaker system used in this enclosure works in conjunction with the University N-3 crossover network (The Acoustic Baton), which separates the amplifier output into 3 different frequency bands and feeds each range to the appropriate speaker. The N-3 is a 3-way, 6-db per octave network with crossovers at 350 and 5,000 cps, compatible with the 3 driver units of the system. Since this network operates in conjunction with horn-loaded speakers, which have their own natural cutoff frequencies, the operation of the network is aided by the speaker system itself. The N-3 network is furnished with presence and brilliance controls. By means of the presence control it is possible to raise or lower the level of the mid-range speaker and, with the brilliance control, to raise or lower the level of the tweeter. The system can be balanced with these controls to suit the user's needs and preferences. Fig. 4 gives a step-by-step procedure for installation of the Classic components.

When properly made, and when used with the recommended speakers, the University Classic provides well-balanced, high-efficiency reproduction of the entire audio spectrum.

# *The* RUMBLE *Seat*

Gentlemen:
I very much enjoyed the first issue of AUDIOCRAFT, and look forward to receiving it regularly. Certainly it fills a need for a type of high fidelity information not adequately supplied by previous publications.

However, I must wax a bit critical. Even making allowances for first-issue troubles, there seem to be quite a few mistakes in the magazine. For instance, in Gordon Holt's otherwise excellent microphone roundup, the RCA microphone prices are ridiculous. Our list from RCA here at the station shows the 44BX at $129.00 and the 77-D at $145.00. Other points about microphones: the RCA Starmaker is not listed, although they claim the best frequency response of all their mikes for this unit. Also, the Altec Saltshaker is listed in group II; personally, I'd put it in group I. We use original Western Electric versions of this model with wonderful results.

Moving along to page 40, we see a statement by Mr. Holt again dismissing staggered BN machines with the remark that "While . . . . , so we shall quietly ignore Fig. 5." Mr. Holt should know that staggered BN tapes can be played interchangeably on Magnecord, Livingston, Viking, and V-M machines as well as on especially set-up Berlants and Concertones. Also, I might point out that the staggered BN system has certain advantages over the more obvious stacked system. The heads do not interact; hence a BN machine such as the Magnecord is most versatile, being able to record BN, single half-track, or 2 unrelated signals. Also, when recording with this head configuration high-quality monaural tapes can be produced, since microphone mixing from 2 mikes may be done after the recording session — thus eliminating expensive retakes for microphone balance with highly paid talent.

It must be noted that staggered BN heads can be aligned separately for optimum performance. Also, such a machine will play back practically any tape, except a stacked binaural.

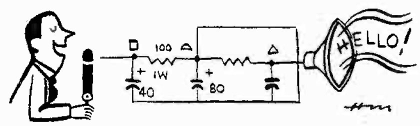
Richard L. Kaye
Program Director, WCRB
Boston, Mass.

*Reply:*
It was stated clearly at the head of the "Microphone Roundup" that the prices shown were manufacturers' catalogue prices "and may be subject to additional discounts". This is one of those cases.

Many equipment manufacturers maintain a dual price scale; one for professional users, such as broadcasting stations, and one for the so-called audiophile. The prices shown for the RCA mikes are, as far as the hobbyist is concerned, the standard prices, and if discounts can be obtained the actual prices may be lower than those shown.

As for the Starmaker microphone: the primary design consideration for this mike was, judging by the manufacturer's literature, small size and optimum performance under the conditions in which it would be used. The mike was intended, apparently, for inconspicuous use in television broadcasting, where it would be used for speech far more often than for any other type of program material. And it is an accepted fact that optimum results on speech are *not* obtained with a microphone having a flat, extended low-frequency response. A glance at the published response curve of the Starmaker will show why it was not included in the list of microphones suited for music recording. The Starmaker has a collapsing low-frequency end which, while ideal for speech, is far from desirable for high-quality music recording.

It was pointed out also at the head of the "Roundup" that groupings of the




listed microphones under various quality headings were based on *all* available published specifications for the mikes. If the manufacturer of a mike of dubious quality wished to claim exceptionally high quality for his mike in sufficiently vague terms that it was impossible to verify the claims, his mike would be put among others that are actually far better, and this could easily work the other way. Inevitably, on this basis, there will be some mikes for which *complete* information is supplied grouped in a lower category than others that are *unspecifically* claimed to be top-notch. Of a pair of mikes that are rated identically, one may be relegated to a lower group on the basis of its published response curve, whereas the other may have no such published curve available. This is what happened in the case of the Saltshaker.

Re my rude dismissal of staggered heads for bin*a*ural tape in the "Tape News and Views" column, Mr. Kaye is quite correct in his statement that staggered heads have certain advantages over stacked-pair heads. Such machines *are* somewhat more versatile, as Mr. Kaye pointed out, but their mutual compatibility is another matter. Since the stereophonic effect depends to a considerable extent on the phase differences between the same sounds appearing at 2 separate points, it is evident that the relative spacing between 2 heads playing these same signals must be precisely the same as the spacing between the record heads, or the phase differences will be thrown out of their original relationship. And while this extreme accuracy may be approached reasonably closely on a machine operating at 15 ips, the tolerances are halved when the speed is reduced to 7.5 ips.

The interaction that always exists to some extent between stacked heads does rule out the possibility of recording different information on the separate channels, but in stereophonic use the actual degree of interaction is so small, and overlaps over such similar signals, that it is quite insignificant in the final results.

J. Gordon Holt

Gentlemen:
Mr. Holt makes a number of points in reply to my letter. The explanation regarding microphone prices is fair enough, but misleading for a magazine such as AUDIOCRAFT which usually publishes "Audiophile Net" prices. These mike prices are full list and must be considered as "Audiophile List". The fact remains that anyone: broadcaster, recorder, PA man, or audiophile, can buy RCA microphones at the net prices (as can be seen in the catalogues of several large mail-order firms).

Apparently, with the Starmaker as with many other audio products, it depends on what you read and believe of the manufacturers' specs. Mr. Holt is basing his judgment on "Permits artist's or performer's face to be in full view" and "Frequency range . . . 70-15,000 cycles" — both quoted from RCA's spec sheet. I also quote: "inherent characteristic for producing 'naturalness' in its translation of voice and *music*". And I point out that the published curve indicates a response of ±4 db from 60 to 20,000 cps. By comparison, from the similar curves published, the 77-D is ±4 db from 40 to 15,000 in its best position (cardioid), and the 44BX is ±6 db from 40 to 15,000. The attenuation of about 4 db below average level at 60 cps of the Starmaker may not be as serious a drawback to some recordists as the 8 db (77-D) or 10 db (44BX) below average at 15 Kc of the mikes preferred by Mr. Holt. (At 15 Kc the Starmaker is not down at all.)

It is obvious I have not yet converted Mr. Holt to staggered-head stereophonic tape recording. I will agree with what Mr. Holt says about head-spacing and its effect on phase relationships — but I don't think this has anything to do with stereophonic reproduction (note that I am not talking about "binaural" in the technical sense, with 8- or 9-*inch* mike spacing — but about stereo, with inter-mike spacing of the order of 10 ft. or more, for loudspeaker reproduction). I am willing to allow that staggered head placement may be less precise than stacked. If we consider an error of 1/100 in. likely, we have an error of 1/750 second at 7½ ips in the time of arrival of the 2 tape tracks at the reproducing heads. Since the speed of sound is about 1,100 ft. per second, this is the same as a total error in the apparent spacing of the microphones and loudspeakers of about 1½ ft.

If you have worked with this type of sound, you will know that such movement has very little effect. Ten-foot inter-mike spacing sounds very much the same as 12 ft., and inter-speaker spacing rarely corresponds to the inter-mike spacing by anything as close as 1½ ft. Actually, we have been at this intensively at WCRB for about a year and a half. We have always used staggered-head machines, and we have most often used 7½ ips for tape economy. In this time, we have used about a dozen different Magnecord binaural pullers, including 2 which we changed to stereo from regular PT-6 AH models ourselves; at least 3 different Livingston or Viking playback units; and one converted V-M unit (and note that the conversion kit is something like $16.95). There has never been a problem of interchangeability; all gave excellent stereo effect. I might add that I seem to remember learning that the human ear is rather insensitive to phase relationships — Olson makes some confusing points on this, and I tended to credit phase with more importance in stereo reproduction myself until I tried to juxtapose Mr. Holt's correct logic with my own practical experience. The only resolution seems to be to go back to what Olson quotes as "Ohm's Auditory Law — The ear tends to analyze the compounds of a complex sound regardless of the phase relations". Thus, we are left with differences in intensity and reverberation reaching the 2 ears as the principle causes of the stereophonic effect, neither of which will be seriously affected by a head-positioning error of the magnitude I have assumed possible.

Richard L. Kaye

*Further reply:*

Apparently, the differences in judgment involved in evaluating microphone quality stem from what Mr. Kaye and I consider important in reproduced sound. I am not an engineer by profession or academic background; neither am I a musician. As a result, my qualifications to judge reproduced sound may understandably be challenged by engineers and musicians alike. But to my highly critical ear, the 2 most important aspects of microphone quality are balance and smoothness. I consider extreme high-frequency range to be a secondary characteristic, since it has been my experience that a mike that is *smooth* out to 8,000 cps introduces far less coloration into the sound than does one going out to 20,000 cps, with 1 or 2 peaks in the range below that.

Second, a microphone with a collapsing low end definitely sounds thinner when heard on a really good speaker system, than one which extends to 40 cps or below. For truly full bass response a microphone should be *absolutely* flat to 20 cycles or below, but since this is starry-eyed idealism of the first order, we have to be prepared to accept some compromises. This means that I had to draw the line somewhere; I decided to draw it at 50 cps for Group I in the "Roundup". Don't misunderstand me — I'm not claiming that the Starmaker cannot be used for music recording — only that, within its price class, there are others that seem to be better suited to this application.

Since I don't currently have any staggered-head recorders to work with, I can only speculate as to the difficulties that might result when the head spacing is changed by any given amount. I can, however, point out that changing the head spacing will *not* have the same effect as changing the microphone or speaker spacing laterally. It would, rather, have the same effect as moving one microphone or one speaker *closer* to the sound source or listener, while maintaining the output at its original level. As for the actual quality deterioration that would result, I cannot cite examples of any tests that I have made myself. But certainly there would be *some* disturbance of the original phase characteristics and, since I have found critical ears to be much more sensitive to slight deviations from perfection than is generally realized, I might assume that they could detect this difference.

I wonder if Mr. Kaye has ever actually made an A-B comparison between a recorder playing its own stacked stereo tape (at 7.5 ips), and a second machine, with its head spacing .01 in. different, playing part of the same tape? One major problem that still seems to exist in stereophonic reproduction from 2 speakers is that of "filling" the space between the speakers; making the sound seem to come from an area rather than from 2 separate points. This difficulty may be due to the ear's inability to tolerate phase mixups, or it may stem from some other deficiency in the speakers or microphone technique. I will not commit myself further on this until I have had the opportunity to get hold of a staggered tape player and try it with different head spacings.

Apart from the head spacing from one stereo machine to another, I don't know of any 2 staggered-head units with the same playback equalization characteristic, but then that's another matter altogether.

J. Gordon Holt

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## WOODCRAFTER

*Continued from page 9*

wood which it receives. Now let's check the procedure for making the component parts of this joint:

A. The Mortise.

1) Use a marking gauge to mark off accurately the outline of the mortise (which will, of course, be the same size as the tenon).

2) Within this outline mark a center line lengthwise as a guide for the bit. (In cutting out the mortise a bit and brace is used to remove the bulk of the waste wood.)

3) Use a bit that has the same width as the tenon. If the tenon is ¼ in., use a ¼-inch bit.

4) Bore a series of holes to the depth which the tenon will reach. For assured accuracy use a bit gauge.

5) Using a chisel, clean out the mortise, taking care to stay inside the guide lines. Use a chisel large enough to do a smooth job.

B. The Tenon.

1) Mark the shoulder of the tenon with a knife (Fig. 7A).

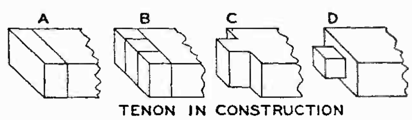



*Fig. 7*

2) Outline the shape of the tenon on the end and sides of the wood with a marking gauge (Fig. 7B).

3) With a back saw, cut out the cheeks and shoulders of the tenon (Fig. 7C).

4) Mark and saw the shoulders on the edges. If necessary, use a chisel for finishing (Fig. 7D).

5) Make a trial assembly.

6) When the fit is perfect, glue and clamp the mortise and tenon together.

*Continued on next page*

## WOODCRAFTER

*Continued from preceding page*

7) Check for squareness.

8) For reinforcement, use a brad or dowel through the side of the mortise.

In summary, here is a list of hints to keep in mind when making a joint:

Use a sharp knife instead of a pencil for marking.

When sawing and chiseling, always work on the waste side of the line.

Make a trial assembly before applying glue.

Always use glue to assure maximum tightness and holding power.

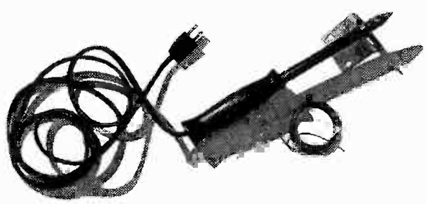
Where possible, reinforce joints with screws, nails, brads, dowels, or corner blocks.

To prevent splitting when using nails in hardwood or plywood, drill pilot holes slightly smaller than the nails and stagger the positions of the nails.

Always clamp your work until the glue hardens.

Check the angle of the work.

Wipe the excess glue off the surface of the wood. If you do this by using



sawdust of the same wood, you'll provide a matching wood filler for joints that might have a slight opening — not that the joints you make will require a filler! However, I must confess that I've found it helpful.

## TAPE NEWS

*Continued from page 14*

recorded without lawsuits from the AFM, then he would undoubtedly get a great deal of use from a stereo recorder. For standard applications, he could use a single channel of the recorder, and he could add both stereo and standard prerecorded tapes to his collection if he felt so inclined. But for someone located out of range of good live performances or stereo broadcasts, a stereophonic recorder would simply be a waste of the extra money except as a playback unit for commercially made tapes.

Accessibility is largely the determining factor, but it is also true that it takes a great deal more of skill and creative ingenuity to make good live tapes monaurally than stereophonically. The recordist who produces good monaural recordings may justly feel very proud of himself, and may validly compare his own recordings with commercial efforts. This may seem like a rather obtuse way of looking at it, but the fact remains that, all other things being equal, a good monaural recording can be a much more rewarding thing, by virtue of the difficulty involved in getting it.

There are, though, other things that can be done with a stereo recorder that would make a monaural recordist think twice before passing one up. If desired, the 2 separate tracks can be used to record related program material, to be mixed together as desired. Or the recorder can be used simply as a half-track unit, with the advantage of being able to start listening to either recording from the beginning of the tape. Then again, it can be used as a half-track unit for 2 parts of a long program, using only one head and turning over the reels at the end of the first half of the recording.

But for a stereophonic recorder to produce as good quality on monaural sources as a standard monaural recorder, the price of the former is likely to be almost twice that of the latter type.

The matter of stacked versus staggered stereo heads is such a bone of contention at the present time that neither can be considered an "industry standard". The choice here *may* be determined by one thing: there is a certain amount of interaction between the 2 halves of a stacked head, and just how significant this is in use will depend upon the application of the recorder. For stereo recording, or for recording 2 programs that are to be mixed together but with no necessity for completely fading one source out, stacked heads seem to me to be the best choice. But when 2 non-simultaneous tracks are to be recorded, staggered heads are absolutely essential.

So far, none of this has touched on the actual performance specifications of recorders, but there's no good reason why this can't be taken up at some later date.

Meanwhile, my recent purchase of a recorder that has provision for A-B switching from the original signal to the playback from the tape has brought up a point that may prove interesting to any other owners or prospective buyers of recorders that use 3 heads for erasing, recording, and playback. The control unit in my system happens to be one of many such that have no provision for taking program from the TAPE OUT

*Continued on next page*

### AUDIO AIDS

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

## TAPE NEWS

*Continued from preceding page*

connection while simultaneously feeding another signal from the recorder *into* the control unit, for monitoring. To add to the inconvenience of it all, there is no readily accessible playback level control on the recorder, so connecting its output directly to my 50-watt power amplifier would be inadvisable, to say the least.

Question. How does one go about connecting a recorder to a control unit so that one can monitor from the tape recorder's output, while recording a signal from the control unit? Judging by the schematic of my particular control unit, it looks as if a conversion would be somewhat difficult, but I'm still mulling at it.

Meanwhile, for the record, I have found 2 commercially available control units that do permit such monitoring, and if I've missed any others I would be interested to hear from their manufacturers for additional listing in the next issue of AUDIOCRAFT. To date, the only units I have turned up that have this monitoring facility are the Altec A-440A and the Bogen PR-100.

## GROUNDED EAR

*Continued from page 5*

entire 3-stage front end of the Williamson, or a little more; it will supply not only the 40 volts drive needed for KT66's, but even the 48 to 50 volts needed by 6550's.

As in the case of the Mullard circuit, the virtues go beyond mere simplicity and economy. One entire stage and one time constant are eliminated; that means less phase shift on both ends. Furthermore, the use of a pentode in the input minimizes the Miller capacitive effect, results in less attenuation at the ultrasonic frequencies and a much lower phase shift in that range. The circuit provides a considerable improvement in stability with a given feedback factor, or permits higher feedback factors for the same stability. Hafler is using the circuit in a new, very compact 50-watt amplifier using a pair of 6CA7's in the output stage and a new transformer he has also designed. With 20 db of feedback the maximum output is delivered at less than 1% IM, and with superb stability and transient response. The output circuit is readily applicable to any Williamson* and I recommend it especially to those who are experiencing instability at either end. Information on the circuit, the kit, and the transformer can be obtained from: The Dyna Company, 5142 Master St., Philadelphia 31, Pa.

*See, for instance, "Modernize Your Williamson Amplifier", AUDIOCRAFT Magazine, January 1956.—Ed.

## TEST INSTRUMENTS

*Continued from page 32*

the single-ended stage as far as trouble shooting is concerned, except that each of the tests must be made for each tube.

Undoubtedly, there are other malfunctions that can occur in your amplifying equipment. Those described in the foregoing paragraphs are typical of the situations one might encounter in measuring operating voltages in equipment that has stopped operating altogether, or at least has stopped operating properly. Most of the causes suggested for various unusual readings are universal in that the factors to which the trouble is attributed are common to practically all power-supply or amplifier circuits, even though the exact circuitry or values of parts may be somewhat different. In other words, lack of plate voltage in an amplifier stage is usually due to an open somewhere in the plate circuit or a short beween the plate and ground, or perhaps a defective tube. The number of resistors in the plate circuit, their values, or the tube type used, have very little to do with this fact. Therefore, the practical information given should be of assistance to you with your own equipment.

Notice too that although a considerable number of tests have been mentioned and directions have been given for locating quite a few different kinds of trouble in a circuit, only the vacuum-tube voltmeter was required for all the test procedures described. A reference was made to having tubes tested, but your local radio repair man will usually be co-operative in this respect. Aside from having tubes tested, the VTVM described and illustrated in previous articles will enable you to make all these measurements, and many others on other types of sound equipment. Certainly, the VTVM is a basic tool; it should probably be the first test item purchased by the home audio builder, experimenter, or maintenance man.

Many of the tests described in this series could have been made with the usual multitester or VOM (volt-ohm-milliammeter). Some of the tests, however (for example, the plate voltage of a voltage amplifier with a high plate-load resistor), do require a VTVM to obtain the proper value. A VOM is fine if one has the knowledge and background to allow for its lower input impedance and sensitivity in interpreting readings. But using a VTVM eliminates this requirement, and readings may be accepted just as they are indicated on the meter. Besides, most manufacturers give voltage measurements for their equipment which have been made with high-impedance VTVM's. You are more likely to duplicate the manufacturers' results if you use the same type of instrument.

## SWITCHING SYSTEMS

*Continued from page 23*

for a portable speaker. To the extent that the speaker is portable and not fixed, there is a corresponding likelihood that the time will come when the speaker is left unconnected to the line being fed by the amplifier.

To prevent a no-load condition, a fixed resistor may be used. This is shown as R in Figs. 2 and 3, where it is assumed that line 2 is used for the portable speaker. When the switch is in position 2, resistor R becomes part of the speaker load. The value of R may be taken as about 3 times that of the highest-impedance speaker to be used on line 2. The amount of power consumed by R is then small enough compared to that taken by the speaker to have no noticeable effect on sound level. Resistors of 10-watt rating should prove adequate for most purposes. A non-inductive resistor should be used.

If desired, R can be replaced by $R_1$, which is put directly across the output

transformer. This affords protection for all lines at all times. $R_1$ has the same value as R. However, the devotee who likes his music strong and is loath to sacrifice one iota of undistorted power available to his main speaker may prefer to use R instead of $R_1$, or possibly neither. Still, the louder one plays his equipment, the more important it is to include the safeguard of either R or $R_1$.

In Fig. 4, R serves a purpose similar to that in Figs. 2 and 3. It provides a load for the amplifier if all 4 lines are in the off position. But it does not provide a safeguard load if all 4 lines are connected to the amplifier and at the same time none of the lines has a speaker connected to it. Although the latter possibility is too remote to be of real concern, it too can be eliminated by using $R_1$ instead of R.

Centralab or Mallory rotary switches are satisfactory for most home applications. While these are not classified as power-handling switches, they can be safely used if audio power does not exceed 30 watts. Thirty watts amplifier output into an average speaker (rated for this amount of power) produce an acoustic power level which very few will wish to hear very often. These few will have to employ conventional power-handling switches or relays.

It may be noted that recommendations for connecting speakers in series have not been made. The reason, again, is amplifier damping: when one speaker is put in series with another, each is denied the advantage of being fed from a low source impedance. There is some disagreement as to whether this is important for speakers covering only high frequencies, but here we have discussed wide-range speakers and speaker systems. In this category only the very finest can be connected in series without ill effects; the same is true of woofers.

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## SOUND SERVICING

*Continued from page 10*

network; some units of well-known brands may be duds.

If none of these measures works you can still reduce the feedback, which might make your amplifier usable at the expense of increased distortion. Try doubling the value of the feedback resistor, which will reduce feedback by 6 db. If your amplifier then performs properly on pulsing tests, let it stay there because stability is more important, all things considered, than the increased distortion it may take to get it.

And, if you feel particularly adept, you might like to play around with a new technique evolved by David Hafler and used in several new, stable Williamson circuits. He puts a "step" in the low-frequency rolloff curve — a step which tends to reverse phase shift at a critical frequency where, otherwise, regeneration would set in. In one of his circuits* Mr. Hafler bridged the 0.25-μfd coupling capacitors between driver and output stages with 1-megohm resistors. That tended to level off low-frequency response at 1.59 cps, and made an amplifier very stable that before was conditionally stable! Of course, if you do this, you put a small amount of positive bias on the output grids, which must be taken into account in total bias considerations. You also reduce total power output slightly — stability again at the price of power.

If you still can't make your amplifier stable, you are, frankly, in need of either the best professional advice available, or one of the new stabilized amplifiers or kits. Amplifier stability is a must for clean reproduction of low-frequency transients, and is becoming increasingly more important as speakers go down lower, with less distortion.

*See "Modernize Your Williamson Amplifier", by David Hafler, AUDIOCRAFT for January 1956.

## AUDIO AIDS

*Continued from page 15*

meter is not used, the variable transformer should be protected with a fuse.

This procedure would be helpful in testing almost any kind of electrical equipment except certain types of synchronous motors that draw excessive current when an attempt is made to run them below the rated voltage.

Henry F. Robbins
New York, N. Y.

### Tape Identification

For positive identification of material recorded on a reel of tape, attach a length of Scotch leader tape to both ends of the recording tape with the necessary information typed or printed on the leader. Typing is preferable since it is clearer and easier to read. For short titles simply attach a short bit of leader to the typewriter roller with 2 pieces of splicing tape. Fasten the leader in the correct position and type away. For typing longer titles, make a jig for the leader by cutting 2 short slots on each side of a piece of standard 8½-inch wide typing paper. Place the jig sheet in the typewriter and run the leader tape in through the slits from right to left. Leave the roll of leader in the box or you may find a lot of tangled tape on your hands. When the leader tape between the slots is filled, return the typewriter carriage, pull through more tape and continue typing. Leave 3 or 4 in. of blank space between titles so that they may be cut apart and spliced to the appropriate recording tape.

T/Sgt. Jerome B. Kmiecik
Loring AFB, Me.

### Detector for Parasitic Oscillations

I have discovered by accident that there is an easy way to detect some inaudible or supersonic oscillations in the output stage of an amplifier. Set the amplifier up in an inverted position and, near it, operate a small AC-DC radio receiver with a built-in loop antenna. If at some point on the dial a frying or buzzing sound can be heard, it may indicate undesirable oscillations in the amplifier.

To determine if the amplifier is really the culprit, hold a metallic wand near each part of the wiring of the amplifier's output stage. If you can grasp the wand without subjecting yourself to shock hazard, your body will help radiate the energy induced. If manipulation of the wand changes the character of the noises heard on the radio, the oscillation is almost certainly coming from your amplifier. You can then experiment, altering tube potentials, feedback, wiring, or other elements, in an effort to suppress the oscillation, using the radio as an indicating device to check your progress. Be sure to make a final check by tuning over the entire dial, as you may only have altered the frequency of the oscillation.

Harry L. Wynn
Derry, Pa.

*This method may not work in all cases. It isn't definite proof that there is no oscillation if no noise is heard.* — Ed.

### Soldering Iron Tip Extension

Ever tried using a standard-size soldering iron tip to make the last few connections in a crowded preamp hookup? Chances are you were not satisfied with the job, or that you waited and borrowed a soldering pencil or gun the following day.

An extension tip to fit on the standard tip of your soldering iron temporarily can easily be made from a small piece of thin copper sheeting. Cut it wide enough to fold around the tip, and long enough to fit solidly on and be formed into a thin, pointed tube projecting about 1 in. from the tip.

This extension tip is inexpensive; it is easily made, mounted, and dismantled; and it can be tinned.

A. Groner.
Kamloops, B. C.

### Stylus Brush

For best performance, the stylus of a modern phonograph cartridge should be freed periodically of accumulated dust and dirt. Wiping the stylus with the fingertip or a cloth is not a very satisfactory method of doing this; a small brush does a more thorough job.

The brush attached to the cap of a small nail-polish bottle makes an ideal stylus brush. Before using the brush, it is necessary to remove the residual nail polish from it and from the bottle in which it is kept — acetone is a good solvent for this purpose. If the polish has hardened inside the bottle and on the brush, let the acetone soak, shaking the bottle occasionally, until the polish is dissolved. At least 4 or 5 rinsings will be required to get brush and bottle thoroughly clean. Use about one eyedropper full for each rinsing.

If the bottle and brush are stored near the tone arm on the turntable mounting board, they will be handy when needed and their presence will act as a reminder to use them. Replacing the cap on the bottle keeps the brush clean when it is not being used.

Eminger Stewart
Stanford University
Stanford, Calif.

# AUDIO NEWS

*Continued from page 7*

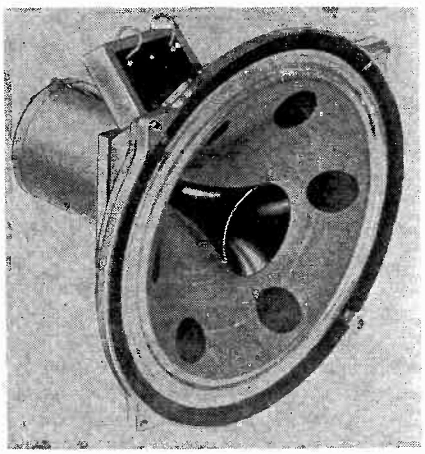
together with an output jack for monitoring with earphones or for playback through an external amplifier. One-, 2-, 3-, and 4-speed models are available utilizing consecutive speeds from 15/16 ips to 15 ips. Models operating at a tape speed of 15 ips meet primary NARTB standards, while models employing a tape speed of 7½ ips meet secondary NARTB standards.

The Magnemite measures 7½ by 9 by 14 in. and weighs 19 lbs. Dry-cell flashlight batteries for the high-gain recording-playback amplifier last 100 operating hours. Flutter over the full winding cycle is said to be 0.1%. Change of speed is accomplished by changing capstans; equalization for different speeds is automatic. The Weathertite VU Magnemite is designed for all applications which subject portable recorders to unusual environmental conditions.

Complete technical specifications and prices may be obtained by writing to Magnemite Division, Amplifier Corporation of America, 398 Broadway, New York 13, N. Y.

## STENTORIAN DUPLEX SPEAKER

A new Stentorian 12-inch duplex (coaxial) loudspeaker has been announced by Beam Instruments Corporation. Utilizing the Stentorian cambric cone construction, this speaker is said to achieve unusually smooth response in the mid-range through the use of fiber stabilizing discs impregnated into the front of the cone. The stabilizing discs are designed to eliminate mechanical break-up in the range from 1,000 to 3,000 cps.

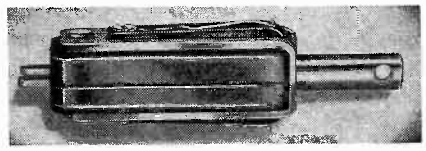


*Mid-range stabilizers on 12-inch Duplex.*

The new Stentorian has two 1½-inch diameter voice coils operating in the field of a series magnet system weighing 11½ lbs. A British "Alcomax 3" ring magnet is used. Specifications supplied by the manufacturer include: magnetic density of HF gap, 17,000 Gauss; magnetic density of LF gap, 14,000 Gauss; bass resonance 35 cps; overall response, 20 to 20,000 cps; impedance, 15 ohms; power rating, 15 watts undistorted. The price is $119.00. The unit also features a built-in, half-pi, 3,000 cps crossover network. For details and complete information, write to Beam Instruments Corp., 350 Fifth Ave., New York 1, N. Y.

## NEW RONETTE PHONO CARTRIDGES

Ronette Acoustical Corporation, importers of Ronette piezo-electric cartridges, pickup arms, and microphones, have announced the introduction of 2 new Ronette *Fonofluid* cartridges. One of these, the Model TO-400, nicknamed the "Slim Jim", is currently available on the market. The other new cartridge is expected to be available very soon. According to the manufacturer, the TO-400 surpasses other models in the Ronette line in compliance, low intermodulation distortion, frequency response, tracking, non-interaction of styli, and constant-velocity response. Like other Ronette cartridges, the 2 new ones will be available with brackets and mountings to fit many types of pickup arms.



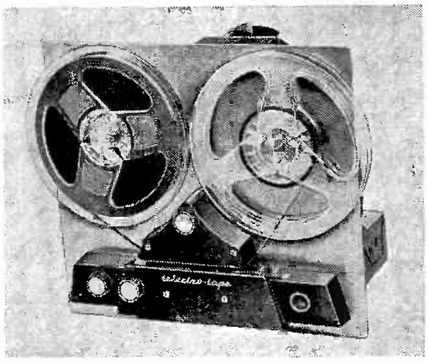
*"Slim Jim" turnover cartridge.*

Complete literature, prices, and technical data are available from Ronette Acoustical Corporation, 135 Front St., New York 5, N. Y.

## TELECTROSONIC TAPE TRANSPORT UNIT

Telectrosonic Corporation announces a new hi-fi tape transport unit, the *Model 220,* for custom installation. The unit is supplied with tape transport, recording amplifier, playback preamplifier, and erase oscillator. It has been designed to provide simplified control of rapid switching from RECORD to PLAY with protection against tape breakage. The unit may be mounted in either the horizontal or in the vertical position.

Operational features include dual speeds of 7½ ips and 3¾ ips; 7-inch diameter reel capacity; single function control for RECORD, IDLE, and PLAY; pre-set recording level; output jack for power amplifier connection; and a magic-eye recording level indicator. Manufacturer's specifications of the Telectro Model 220 tape transport include: output of 1 volt; frequency response from 50 to 12,000 cps, ±3 db at 7½ ips; signal-to-noise ratio of 45



*Model 220 hi-fi tape transport unit.*

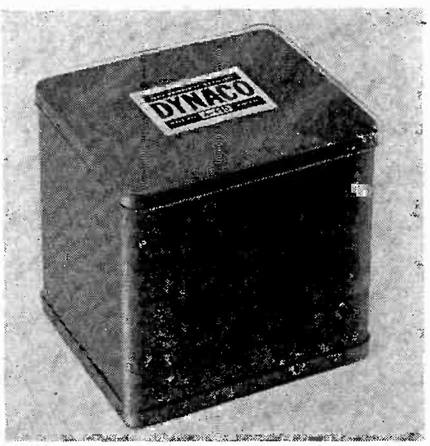
db; less than 2% distortion; flutter and wow of less than 0.3%; low-level input for crystal or dynamic microphones; and high-level input for tuner, TV, radio, or record player.

Additional information about the Model 220 tape transport can be obtained from Telectrosonic Corp., 35-18 37th St., Long Island City 1, N. Y.

### DYNA OUTPUT TRANSFORMER

Dyna Company has introduced the first of its new line of high fidelity audio output transformers. This unit, the A-430, matches 6550 tubes or 6CA7/EL-34 tubes in circuit configurations furnishing from 50 to 100 watts, according to the manufacturer. It is recommended for use in converting Williamson-type amplifiers to 50-watt power capabilities using either of these tube types.

The new transformer is guaranteed by the manufacturer to have a frequency response of ±1 db from 6 to 60,000 cps. Its undistorted power-handling capacity is said to be 50 watts from 20 to




*A-430 high-power output transformer.*

20,000 cps, and 100 watts from 30 to 15,000 cps. It is claimed that the unit has excellent square-wave transmission at all audio frequencies, and that its phase characteristics permit substantial feedback without amplifier instability.

Complete data on the Dynaco A-430 and circuitry for converting Williamson-type amplifiers to high power, high quality use are available on request from Dyna Company, 5142 Master St., Philadelphia 31, Pa.

### BOOKLET ON MAGNETIC TAPE MANUFACTURE

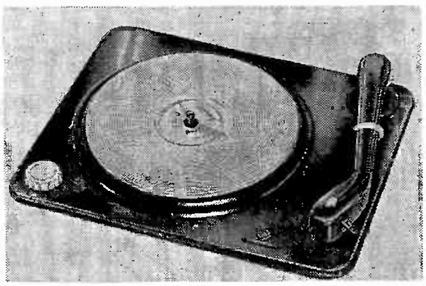
Owners and operators of tape recorders, and others interested in magnetic recording will find a great deal of useful and interesting information about the manufacture and quality control of magnetic recording tape in a new illustrated folder published by ORRadio Industries, Inc.

This new folder, "How Magnetic Tape Is Made", describes and illustrates in 6 photos the manufacture of magnetic recording tape. ORRadio Industries, publishers of the booklet, are manufacturers of the line of *Irish* Brand Ferro-Sheen Process Recording Tapes, and they are offering copies of the booklet free on request.

For your copy, write to ORRadio Industries, Inc., Department 153, Opelika, Ala.

### MIRAPHON XM-110A

Audiogersh Corporation announce that they are now delivering the 1956 model of the Miraphon manual record player.



*New Model XM-110A manual player.*

The new unit is known as the Miraphon XM-110A. The manufacturer claims that, due to a special method of motor mounting, transmission of motor vibration to the chassis has been eliminated. The table moves in a double row of ball bearings, as does the tone arm. The plug-in head will accommodate the user's choice of cartridge.

The unit is started by moving the tone arm to the right, and, at the end of the record, it automatically shuts off. The chassis of the Miraphon XM-110A is exactly the same as that of the Miracord XA-100, so the same base or mounting board can be used for either unit. Adjustment of the tone arm for cartridge weight can be done by means of a thumb screw located under the arm. The unit comes completely assembled with power cord and jack.

For further information about the Miraphon XM-110A, write to Audiogersh Corporation, 23 Park Place, New York 7, N. Y.

### AKG 60K STUDIO MICROPHONE

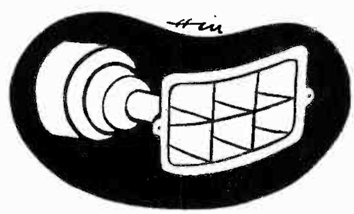
The DYN 60K Studio microphone, manufactured by Akustische & Kinogeraete Ges. M. B. H., Vienna, Austria,

*Continued on page 48*

## Abbreviations

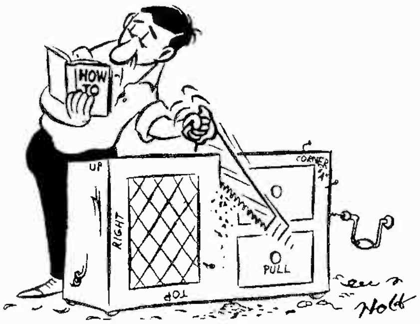
Following is a list of terms commonly used in this magazine, and their abbreviations. The list is arranged in alphabetical order.

| | |
|---|---|
| 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 |
| capacity | C |
| characteristic impedance | Zo |
| 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 |
| 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-ampere | va |
| voltage, or potential difference | E |
| volts, center-tapped | vct |

# Audiophile's Bookshelf

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No. 202 .......... $4.95

**MICROPHONES,** by Engineering Staff of BBC.
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The New **HIGH FIDELITY HANDBOOK,** Irving Greene and James Radcliffe.
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**HIGH FIDELITY RECORD ANNUAL** — A first volume of record reviews — classical music and the spoken word — from HIGH FIDELITY Magazine. Edited by Roland Gelatt. Just off the press.
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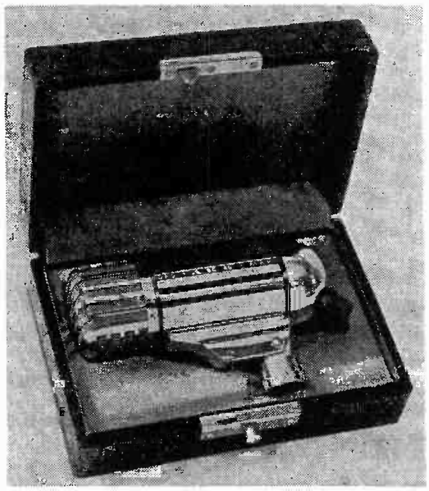
## AUDIO NEWS

*Continued from page 45*

is a pressure receiver of the moving coil type, with an aperiodic spherical response characteristic and uniform sensitivity over a wide range of frequencies. It is mainly intended for studio work



*AKG 60K studio microphone.*

and for all electro-acoustical installations with a demand for very high fidelity reproduction.

Technical data supplied by the manufacturer are as follows:

Frequency range: 50 to 15,000 cps.

Frequency response: at 0°, 90°, or 180° angles of sound incidence, not more than ±2.5 db.

Sensivity: 0.1 mv/$\mu$bar.

Impedance: 50Ω at 1,000 cps.

Weight: 14 oz.

Price: Standard list, $88.00, FOB New York. With additional impedances, $99.50, FOB New York.

Additional information about the DYN 60K Studio Microphone may be obtained in the United States from Electrovert, Inc., 489 Fifth Ave., New York 17, N. Y., and in Canada from Electrovert, Ltd., 265 Craig St., Montreal 1, Que.

### IRC DATA BULLETIN

The International Resistance Company has announced a new catalogue data bulletin on deposited carbon resistors. This bulletin contains comprehensive data on tests, applications, specifications, tolerance, ranges, performance, and dimensions of deposited carbon resistors. For your copy, write to International Resistance Company, 401 North Broad St., Philadelphia 8, Pa., and request Catalogue Data Bulletin B-4a.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