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The Grounded Ear

by Joseph Marshall

B&O Special Cartridge

Some 18 months ago the Fenton Company began to import and distribute the Danish B&O variable-reluctance cartridge, which had excellent frequency response and high output at a modest price. Now there is an additional model called the *B&O Special.* This is somewhat more compact, and accepts a single stylus (either microgroove or 78). The movement has also been improved in compliance and effective moving mass, so that stylus resonance has been pushed upward, thus providing a smoother response. I have been using one for the past month, and I have found that it delivers very satisfying performance. At a price of about $12 (with sapphire stylus) it represents an excellent value.

In Fig. 1 are given response curves I obtained directly from the pickup into a load of 50 K with two test records. On the Cook 10 LP record the stylus resonance produces a peak of about 3 db at 17,000 cps. There is no peak on the Elektra 35 record (of which, more below); the response is about 6 db down at 15,000 and 8 db at 20,000 cps. These are excellent curves. Incidentally, the pickup is surprisingly indifferent to load. Even the 1-megohm load of my AC VTVM, without the 50-K shunt, did not produce any serious peaking. Apparently, the response is excellent with just about any load above 25 K, provided cable capacitance is not excessive. The impedance is low but the output is high (about 25 mv), and hum is no problem.

It mounts easily in any arm or holder which accepts a GE. Like the normal B&O, it has a built-in strip of radioactive foil to minimize static noise and dirt attraction. Tracking is good and distortion is low with a stylus force of 7 grams. There is one disadvantage: the 8-pole magnet is large and heavy; and the magnetic attraction is, in consequence, high. This is not troublesome with nonmagnetic turntables, but will require careful setting of stylus force at exact record-playing position when a ferrous turntable is used.

The cartridge sounds excellent. High frequencies are clean and sweet; it takes a highly overcut record to produce distortion. Chatter is very low. Compliance seems to be just below that of the very finest and most expensive cartridges. All together, it should please a great many audiophiles and offer stiff competition to more expensive cartridges.

The Elektra 35 Test Record

Measuring pickup-cartridge response is complicated by several factors. For one, there is a resonance resulting from the inductance of the cartridge and capacitances of the cable and the preamplifier input circuit. This is more of a problem with variable-reluctance pickups than dynamic cartridges, because of the

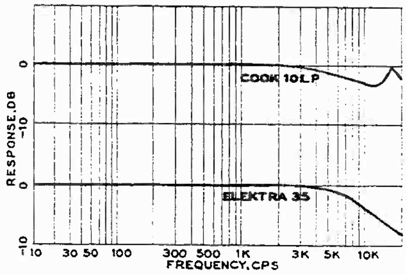


Fig. 1. Pickup response on two records.

much higher inductances of the former. It is quite possible to minimize this by using short cables of minimum capacitance, proper resistive loading of the cartridge, and a preamp or voltmeter with very low input capacitance; but the measured response will obviously be duplicated only with identical capacitance.

Far more difficult to make allowances for or to correct is the mechanical resonance, which is the result of the effective stylus mass and the compliance of the record material. There are no simple ways of eliminating this factor from measured results. All the plastic materials used for modern records result in such resonances, although in some the peaks may be higher up in the frequency range than in others. As a matter of fact, this phenomenon has often been deliberately exploited to widen the response range of a pickup, the resonant peak serving to boost up a response which would normally be sloping. In any event, it has been difficult to obtain measurements which indicate the degree of resonance or the effects of it.

Now the Elektra Corporation has issued a test record (Elektra 35) which has a sweep of 45 discrete frequencies between 18.75 and 20,000 cps. It is pressed from a material which, it is claimed, virtually eliminates the response-measuring problem by moving the normal point of resonance to beyond 20,000 cps. This claim appears to be substantially realized. Tried with a half-dozen cartridges of various types, it has proved to be free of mechanical-resonance peaks. In addition, the record has far more frequency points for measurement than usual, and they are spaced more closely. This leads to greater accuracy, since it provides insurance against missing a sharp narrow peak or dip in response. Finally, the low frequencies especially are freer of distortion than usually is the case, and so are more useful in checking system distortion. The record is a very useful addition to anyone's collection, whether amateur or professional.

It should be emphasized, however, that although this disc is valuable for the purpose of eliminating groove-stylus resonance from response measurements, the frequency response obtained with it is not necessarily representative of the response that will be obtained on commercial musical records. Most, if not all, standard records are pressed from material that produces such resonances within the audible range with most pickups. A cartridge that shows a flat response above 10 Kc on the Elektra 35 would, in consequence, probably have a peak on musical recordings; and one which shows a sloping response on the Elektra 35 might well be reasonably flat on a standard recording. This is indicated by the two curves for the B&O Special in Fig. 1. The resonance-free response on the Elektra 35 begins a consistent gradual decrease above 6 Kc; but with the Cook 10 LP, the response rises again above 14 Kc. Even the Cook record is not representative of commercial pressings, and the pickup response on various music records may well have all sorts of variations from that of the Elektra to a lower slope and higher peak than that of the Cook. Therefore, while the Elektra 35 does solve a hitherto serious problem in

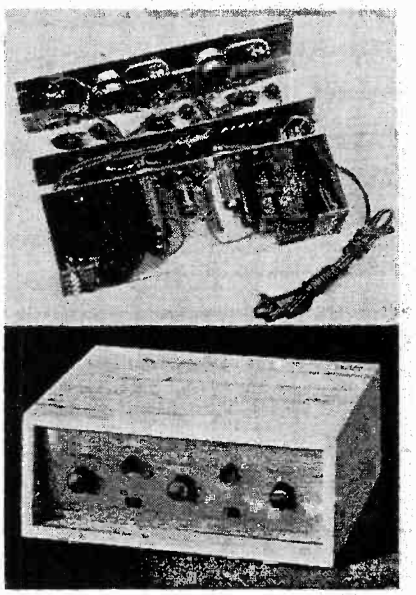
Continued on page 47

Audionews

COLBERT FREQUENCY DIVIDER

Colbert Laboratory has announced the *Model 3-CFD* three-channel electronic frequency divider. The 3-CFD eliminates the necessity for fixed crossover networks by dividing the signal into two or three frequency ranges before amplification. Individual level controls are provided for each sound channel.

Since choice of crossover points depends on such variables as loudspeaker design, enclosure construction, etc., the



Variable electronic dividing network.

Colbert frequency divider provides variable crossover controls for low, middle, and high ranges. This permits the correct adjustment for any combination of components.

The unit includes a 10-watt ultralinear amplifier for use on either the middle- or high-frequency channels. Separate amplifiers are used for the low-frequency channel and whichever of the two higher-frequency channels for which the unit's own amplifier is not used. The Colbert 3-CFD uses the latest nonmicrophonic, low-noise tubes developed by Mullard of England.

Additional information about the Colbert Model 3-CFD frequency divider will be furnished on request.

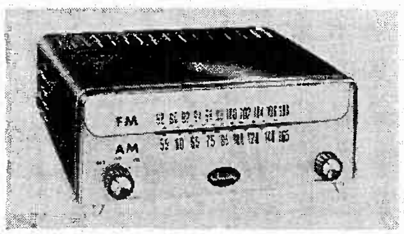
INTERNATIONAL TUBE DATA

Radio Valve Data, Fifth Edition, published by *Wireless World* Magazine, gives in tabular form the characteristics and base connections of approximately 2,500 British and American receiving tubes, over 300 cathode-ray tubes, and 37 transistors. A section of the book is devoted to voltage stabilizers. A complete index and a table of British-American equivalents are provided.

Radio Valve Data is available from British Radio Electronics Ltd., 1833 Jefferson Place, N. W., Washington 6, D. C. The price of the book is $1.50 postpaid.

QUALITY KITS

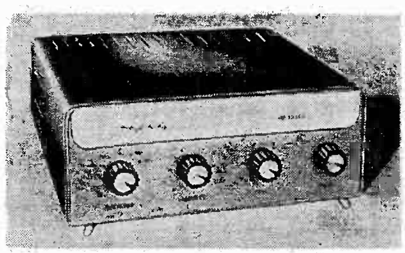
Two kits have been announced by Quality Electronics, Inc. The *Model 1000* FM-AM tuner kit features an Armstrong FM circuit with Foster-Seeley discriminator. A 7-tube-plus-rectifier circuit provides a reported sensitivity on FM



New FM-AM tuner available in kit form.

of 5 μv for 30 db quieting, and a sensitivity of 25 μv on AM. FM bandwidth is said to be 200 Kc, and the AM bandwidth is 8 Kc. The AM section utilizes a Ferrite Loopstick antenna. The Model 1000 contains its own power supply.

A companion piece to the Model 1000 FM-AM tuner is the *Model 2000* amplifier which also is furnished in kit form. This 12-watt unit includes a built-in preamplifier. Output impedances are 4, 8, and 500 ohms, and there



Control amplifier-preamp matches tuner.

are four input jacks on the rear apron of the chassis. The Model 2000 amplifier has three-position equalization.

Both kits, the Model 1000 FM-AM tuner kit and the Model 2000 amplifier kit, come with illustrated, step-by-step instruction manuals.

FAIRCHILD CARTRIDGE AND NEW BOOKLET

Experimental products are made available in limited quantities from time to time by Fairchild Recording Equipment Company in order to provide experimenters with the latest advances in the audio field. The Fairchild Experimental Cartridge, *Model XP-2*, which was demonstrated at recent high-fidelity shows, is the latest in this series.

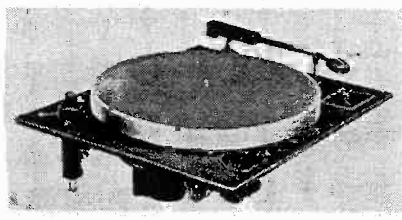
The XP-2 is a hand-constructed cartridge of the moving-coil type incorporating several advanced features, such as a bonded diamond-stylus assembly and an unusual damping system. The cartridge is priced at $60 and is available through regular audio and sound equipment dealers.

A booklet, containing basic design data on various popular types of cartridges and explaining how they are constructed and how they operate, has been published by Fairchild. The booklet is titled *In the Groove* and is available free on request.

CONNOISSEUR TURNTABLE

The 1957 model of the British-made *Connoisseur* turntable is now available in the United States from the Ercona Corporation, importer and distributor of high-fidelity components and systems.

The 12-inch, nonmagnetic turntable of the Connoisseur is a lathe-turned sand casting, custom-fitted to its individual



The latest Connoisseur turntable model.

spindle. The design calls for clearance between the spindle shaft and its bearings sufficient to allow a thin film of oil to be maintained between them preventing metal-to-metal contact.

The unit has a synchronous hysteresis motor, said to be virtually vibrationless, with minimum noise level and hum induction. A 4% speed variation (either 2% faster or 2% slower) is provided at each of the three standard speed positions, without motor braking.

The turntable is equipped with the Mark II Super-lightweight tone arm which, it is claimed, eliminates peaks and provides virtually flat response up to 20,000 cps. The arm is furnished with easily interchangeable plug-in heads for standard and microgroove records.

Complete information about the Connoisseur turntable will be sent on request.

DUOTONE LOUDSPEAKERS

The Duotone Company, manufacturer of Duotone phonograph needles, has announced a new line of loudspeakers. The line is being presented under the

One speaker of Duotone's complete line.

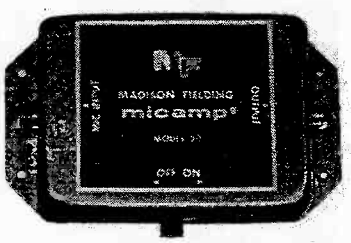
name of *DFF, Duotone Fidelity Focus Loudspeakers.*

The new loudspeaker line is comprised of seven speakers in all, including a 15-inch woofer, 4 coaxials, and 2 tweeters.

Additional information about these loudspeakers will be furnished on request to the Duotone Company, Dept. PR6, Locust Street, Keyport, N. J.

MICAMP

Madison Fielding Corp. has announced production of the *Micamp*, an all-transistorized, impedance-matching preamplifier which is said to permit the direct use of low-impedance, low-gain cartridges and microphones with high-impedance tape-recorders, amplifiers, etc. It is claimed that the Micamp will provide more than 30 db gain, with no



Transistor preamp for low-level uses.

hum pickup, and no distortion at normal levels. Frequency response of the unit is flat within 1.5 db from 20 to 20,000 cps, according to the manufacturer. Input impedance is 50 to 250 ohms; output impedance is 18,000 ohms.

PENTRON STEREO CONVERSION KIT

Stereophonic sound can be added to almost any Pentron tape recorder through the use of a stereophonic conversion kit announced by the Pentron Corporation. The new kit, called *Stereo-Magic*, is reported to permit the addition of stereophonic sound to any Pentron recorder sold during the past five years, with the exception of Models RWN and CT-1.

The Pentron Stereo-Magic kit, said to include everything needed for conversion, is priced at $16.95.

MULLARD TUBES

A new 75-volt stabilizer tube, the *75C1*, has been developed by Mullard to give improved performance in any equipment and any application where low-voltage stabilizers are normally employed.

The Mullard 75C1 combines zirconium electrodes and the sputtered-envelope technique pioneered by Mullard. This design is said to give a combination of high stability and good regulation. Among the features that distinguish this tube may be included the special uranium-oxide coating which ensures that the maximum striking voltage is 110 volts in both daylight and darkness.

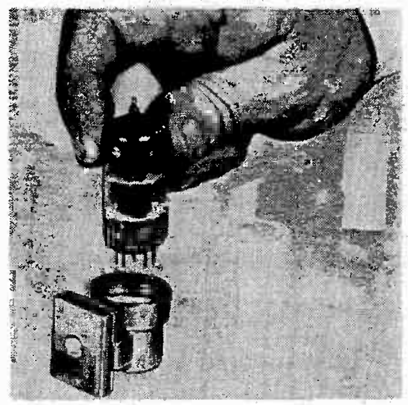
The Mullard 75C1 has a current range of 2 to 60 ma with a regulation of 9 volts and a variation of burning voltage of less than 1% per 1,000 hours. The burning voltage of all Mullard 75C1 tubes at 20 ma is confined within the range of 73 to 79 volts.

Mullard tubes are handled in the United States by International Electronics Corp. who have announced the publication of the *Mullard Preferred Tubes Replacement Guide*. The *Guide* is a comprehensive chart listing types and prices of hundreds of tubes for British and European equipment, particularly for high-fidelity audio amplifiers, FM and AM tuners, tape recorders, and television sets. Cross-reference data are given showing the interchangeability of Mullard tubes with American and European types. Free copies of the *Mullard Preferred Tubes Replacement Guide* are available from International Electronics Corp. on request.

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

MAGNETIC TUBE GUIDE

Altron Products has introduced a new precision-built *Magnetic Tube Guide*, for miniature tubes. The Guide is reported to simplify the arduous task of replacing 7- and 9-pin miniature tubes.



Alignment guide for miniature tube pins.

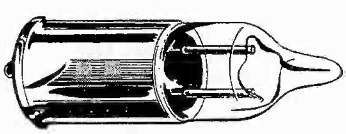
The Guide is held fast to steel chassis by a new-type magnet which is said to afford perfect alignment and which will not deviate from position when new tubes are inserted.

Additional information about the magnetic Tube Guide will be furnished on request.

SUBMINIATURE NEON LAMP

Availability of a new subminiature neon lamp has been announced by the Circon Component Company of Santa Barbara, California. The new lamp, designated the *NE2C*, is the result of a co-operative development effort by the Circon Engineering and Tooling Departments, and the General Electric Miniature Lamp Division. The lamp is designed with a midget flanged base, and it will be interchangeable in many assemblies with existing miniature aircraft lamps of the No. 327 to 338 series. It is substantially smaller than based neon lamps heretofore available.

Electric design of the lamp makes it of interest for a wide variety of indicator and computer applications. Power consumption is reported to be only .04 watt; current drawn is stated as .0003



Neon lamp is smallest made with base.

ampere. The lamp is said to produce practically no heat. It is a long-life unit with average life of over 25,000 hours. Starting voltage is 65 volts AC.

Further information is available from the manufacturer.

TIPS FOR THE WOODCRAFTER

by George Bowe

The Glue Story

In hi-fi cabinet construction, as in all woodworking, proper gluing provides a lasting and inconspicuous bond of unequaled strength. It is one of the oldest materials used to fasten wood — today's museums display examples of early Egyptian and Roman furniture assembled with glue hundreds of years ago and still in good condition. A good joint properly glued is stronger than the wood itself. This can be graphically demonstrated by breaking a narrow strip of plywood and noticing that the glued surfaces remain bonded, causing the wood to splinter and separate at other points. Such great strength is achieved by the glue's seeping into the pores of the wood and uniting the two surfaces.

Assuming that the parts to be glued are well fitted, what constitutes proper gluing? The answer lies in the right kind of glue for the job, its correct application, and the before-and-after treatment of the areas to be joined. There are many types of glues, but let's

Warren Syer

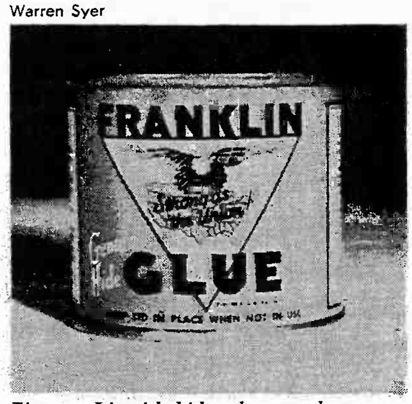
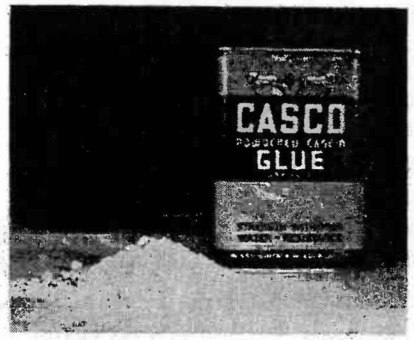


Fig. 1. Liquid hide glue, ready to use.

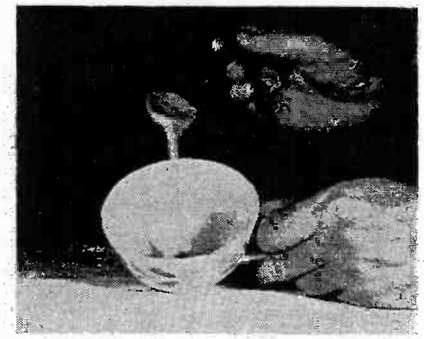
explore those generally available and most commonly used by the home craftsman.

Hot animal glue has been used for hundreds of years. It is made from hide parings, bones, and other waste parts of animals, and is sold in sheets, flakes, and coarse powder. A good grade of animal glue is brittle and transparent. To prepare it for use, it must be placed in a glue pot, covered with clean, cold water, and left to soak overnight. It is then heated to 130°-140° F., as recommended by the manufacturer, and mixed thoroughly. It must be kept hot while in use and, although it can be reheated for later use, it becomes weaker with each reheating. Freshly prepared glue is the best; never

Warren Syer



Warren Syer



Figs. 2 and 3. Casein glue, a milk product, mixes easily and is water-resistant.

mix a new batch with an old one. While not waterproof, hot animal glue is strong. Immediate assembly of the pieces is necessary once the glue is applied, since it sets quickly and loses its holding power as it cools. Because of its fast-drying characteristic, clamps may be removed after 10 or 12 hours.

Animal glue is also available in liquid form. *Liquid animal glue* (Fig. 1) can be used just as it comes from the container, with no mixing or heating and none of the odor associated with the hot type. It is good for all kinds of cabinet and furniture construction, providing strong, light-colored joints, and it does not stain wood. While it is not waterproof, it is mold resistant. On new wood it is best to coat each surface to be joined, wait until the glue becomes tacky, then apply a second thin coating and clamp the pieces together for at least 24 hours.

Made from the tissues and scales of fish, *fish glue* is similar to animal glue in adhesive strength on porous surfaces. It comes ready to use, is easy to apply, and is good for small repairs and all-around home-workshop purposes. It is not waterproof and is more expensive than animal glue.

Casein glue (Figs. 2 and 3) is made from curds of skimmed milk to which alkali is added. The finished product is a fine white or yellow powder which, when mixed with water, becomes a strong, water-resistant adhesive. It can be mixed in large or small quantities as the work requires. It is used cold, and can be applied in temperatures below 70° F., but it cannot be stored once it is mixed. Casein glue is best applied with a stiff brush in a thick coating, and it can accordingly be used on poorly fitting joints. The brush and container should be washed in warm water immediately after each use. This glue is very effective on woods such as spruce and pine and it works well on such oily woods as teak and lemon, but it does not adhere well to oak; a 10% solution of caustic soda should be applied to the joint surfaces of oak and allowed to dry thoroughly before casein glue is used on them. One disadvantage of casein glue is that it stains wood, and work may require bleaching after it is glued.

Continued on page 45

Courtesy United States Plywood Corp.

Fig. 4. Resin glue does not stain wood.

"We're building a

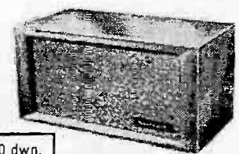
HEATHKIT®...

BECAUSE IT'S SUCH GREAT FUN . . . AND BECAUSE WE GET SO MUCH MORE FOR OUR MONEY!"

Every day more and more people (just like you) are finding out why it's smart to "do-it-yourself" and save by building HEATHKIT high fidelity components. These people have discovered that they get high-quality electronic equipment at approximately one-half the usual cost by dealing directly with the manufacturer, and by doing their own assembly work. It's real fun—and it's real easy too! You don't need a fancy work shop, special tools or special knowledge to put a Heathkit together. You just assemble the individual parts according to complete step-by-step instructions and large picture-diagrams. Anyone can do it!

Heathkit Model SS-1 Speaker System Kit

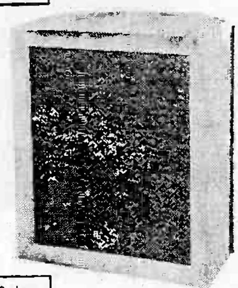
This high fidelity speaker system is designed to operate by itself, or with the range extending unit listed below. It covers the frequency range of 50 to 12,000 CPS within ± 5 db. Two high-quality Jensen speakers are employed. Impedance is 16 ohms, and power rating is 25 watts. Can be built in just one evening. Shpg. Wt. 30 lbs. **$39.95**



$4.00 dwn.
$3.36 mo.

Heathkit Model SS-1B Speaker System Kit

This high fidelity speaker system kit extends the range of the model SS-1 described above. It employs a 15" woofer and a super-tweeter to provide additional bass and treble response. Combined frequency response of both speaker systems is ± 5 db from 35 to 16,000 CPS. Impedance is 16 ohms, and power is 35 watts. Attractive styling matches SS-1. Shpg. Wt. 80 lbs. **$99.95**



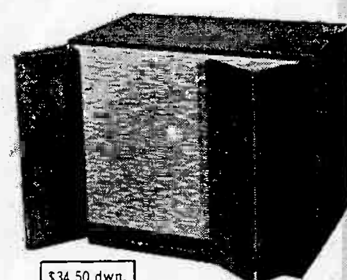
$10.00 dwn.
$8.40 mo.

HEATHKIT "LEGATO" SPEAKER SYSTEM KIT

Months of painstaking engineering by Heath and Altec-Lansing engineers has culminated in the design of the Legato, featuring "CP" (critical phasing) and "LB" (level balance). The result is a *new kind* of high fidelity sound, to satisfy even the most critical audio requirements. Two high-quality 15" theater-type speakers and a high-frequency driver with sectoral horn combine to cover 25 to 20,000 cycles without peaks or valleys. "CP" and "LB" assure you of the smooth, flat audio response so essential to faithful reproduction. Choice of two beautiful cabinet styles below.

"Legato" Traditional Model HH-1-T

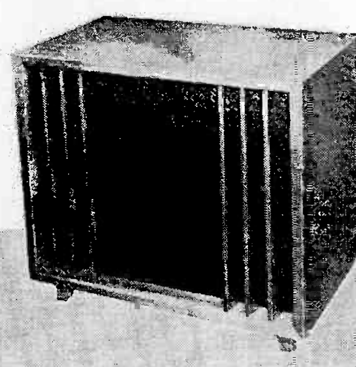
Styled in classic lines to blend with period furniture of all types. Doors attractively paneled. African mahogany for dark finishes unless you specify imported white birch for light finishes. Shpg. Wt. 246 lbs. **$345.00**



$34.50 dwn.
$28.98 mo.

"Legato" Contemporary Model HH-1-C

This fine cabinet features straightforward design to blend with your modern furnishings. Slim, tapered struts run vertically across the grille cloth to produce a strikingly attractive shadowline. Wood parts are precut and predrilled for simple assembly. Supplied in African mahogany for dark finishes unless you specify imported white birch for light finishes. Shpg. Wt. 231 lbs. **$325.00**



$32.50 dwn.
$27.30 mo.

HEATH COMPANY
A Subsidiary of Daystrom, Inc.
BENTON HARBOR 18, MICHIGAN

TAPE NEWS & VIEWS

by J. Gordon Holt

Microphone Technique III

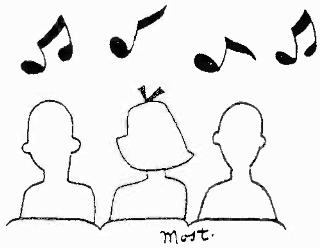
Aural depth perception is an intangible, but readily recognized, characteristic of "live listening" which enables the concertgoer to judge approximately how far he is from each performer in a group even with his eyes closed. Estimating distance by sound alone is a complex process involving, among other things, our preconditioned concepts of how sounds *should* be heard when originating from a certain distance away. We assume, without any conscious cerebration, that a certain ratio of direct sound to reflected sound (or reverberation) implies a certain distance between the sound source and our ears, and when confronted by an atypical direct/reflected sound ratio, we must make a conscious effort to correlate the actual sound sensation with what we believe it should be. A good example of this sort of confusion appears from time to time in class-B motion pictures, when the actor is visibly about 20 ft. behind the screen, yet his voice is *audibly* very close, because of sound-track pickup from a mike directly over his head.

Our ears are less easily fooled by unusual room acoustics when we can listen to stereo recordings, since the brain can more easily analyze the acoustical conditions and make the necessary compensations. But when the medium is a single channel, as in a monaural recording, the effect of varying distance, or acoustical perspective, must be achieved by miking for a "typical" ratio between direct and reflected sound.

The illusion of depth or perspective is obtained by maintaining a fairly large *distance ratio* between the performers at the front and those at the rear of a performing group. The closer is the microphone to a sound source, the closer or more intimate will be the recorded sound, because a close mike receives direct sounds at much higher amplitude than it does echoes. The farther a mike is from the source, the less intensely it receives direct sound, so it picks up relatively more echo, creating an effect of greater distance.

If a mike is placed 10 ft. in front of a performing group 10 ft. deep, the front row will be just half as far from the mike as the back row, giving a distance ratio of ½. Backing the mike off to 20 ft. from the first row will make the distance between the front and rear rows ⅓ of the total mike distance from the rear of the group: a distance ratio of ⅓. The audible result of this 10-foot change in mike distance would be a significant reduction in the apparent depth of the recording, because the *relative* difference between the distances from the mike to the front and to the rear of the group will have been reduced. At the other extreme, a dimensionless and uninterestingly flat tapestry of sound can be produced by placing the mike high in the air directly above the middle of the group, when all the performers will be about the same distance from the mike.

The pickup distance ratio can be utilized to advantage when there are soloists standing at the front of a performing group, because any mike place-



ment in front of the group (rather than directly overhead) will tend to bring the soloists into the slight prominence that should be accorded them.

Smaller performing groups, like jazz combos, chamber groups, and small vocal ensembles, are considerably easier to record well in an auditorium than are larger performing groups, because the intimate miking that suits them tends to minimize the audibility of recording-room acoustics. Recording small groups then becomes almost entirely a matter of obtaining good balance and natural timbres from the performers. If a group is inherently well balanced, microphone balance is obtained with a placement approximately equidistant from all performers at the front of the group. From there on the placement will depend on the kind of mike used. An omnidirectional microphone might best be located about 6 to 8 ft. above the group, directly above or in front of the foremost row. A unidirectional mike (which tends to sound closer than it actually is) might be 5 to 15 ft. away from the group, and about 6 ft. in the air.

If the performing group itself is not too well balanced, adjustments can be made when recording by placing an omnidirectional mike slightly closer to the weakest section of the group, or by aiming a unidirectional mike toward that section (thus giving it the mike's maximum sensitivity).

The timbre of a recording may be modified to some extent by utilizing the high-frequency selectivity exhibited by most mikes. With few exceptions, a microphone's high-frequency response is maximum for sounds arriving perpendicular to its diaphragm. Therefore the crispness of a recording will be maximum when the mike is aimed directly at the sound source. A sweeter or more plush sound can be produced by simply directing the mike's diaphragm to one side of the sound source. With a unidirectional mike, this can be overdone to the point at which the source lies within the mike's reduced-sensitivity area, but there is a fairly wide angle through which its over-all sensitivity remains constant while its high-frequency response varies.

Small groups in small rooms are another problem altogether, because here the acoustics are likely to be far from ideal for recording. The difficulty is further compounded when the playback equipment happens to be in the room where the recording is made, because any of the room's bad characteristics are doubled when the playback is listened to. There are, as usual, exceptions to this rule (and I can cite at least one from personal experience), but it is generally true that a recording made and played in the same room will not sound very good.

Small-room recording almost invariably requires either an ultraclose mike placement or the use of a unidirectional microphone to minimize reflections from walls and ceiling. Best results are often obtained with the performers at one end of the room, in front of heavy drapes covering most of that wall, and with a cardioid microphone 5 to 8 ft. away, aimed toward the group.

When recording a solo performer, with or without piano or instrumental accompaniment, good microphone technique involves a greater degree of attention to naturalness and balance than is practical in larger-group recording. Also, since most of such recording takes place in living rooms, the acoustical shortcomings of small rooms are usually added to help complicate matters. The comments made earlier about small rooms will apply equally here, and the remedial measures are essentially the same as those for small-group recording.

Assuming (perhaps naïvely) that the acoustical problem has been minimized as much as possible, optimum "presence" is the next factor to come under scrutiny. This will depend upon the type of instrument or voice being taped. For instance, a throaty-voiced female singer will benefit by an ultraclose (1- to 3-foot) mike distance, while the window-shaking baritone is often best picked up at an 8- to 15-foot distance.

If there is no accompaniment, the best solo pickup position will be the one to use, but accompaniment adds a second consideration: optimum balance. Once again, the discriminatory properties of a directional mike can be utilized to this end, or, if an omnidirectional mike is used, its relative distances from the soloist and accompanist can be varied.

On rare occasions it is possible to lure a performer into an auditorium for recording purposes, which promptly solves most acoustical problems. Auditorium acoustics are immensely flattering to any voice or instrument, and are almost a necessity for good recordings of powerful projectors such as bravura baritones and brass instruments. Of course, the fact that an auditorium has hugely billowing acoustics does not imply an obligation to produce echoey recordings. Proper mike placement will keep the reverberation under control, and in an ultraclose recording there may not seem to be any audible echo at all; still, the hall acoustics will add to the sound a subtle spaciousness and realism distinctly pleasing to the ear.

With the close miking that will be used for many small groups, the character of the hall's acoustics is less important than when a large group is being recorded. Because the latter calls for appreciable reverberation, the hardness or softness of the echo is a prominent feature of the recording, but close miking can minimize this so that even the harshest acoustics aren't likely to contribute any more than an aura of coldness to the recording.

Thus far we have tacitly assumed that all recordings would be made with but a single microphone, be it omnidirectional, unidirectional, or bidirectional; as far as most amateur recordists are concerned, this is probably a safe assumption. But, despite the ballyhoo

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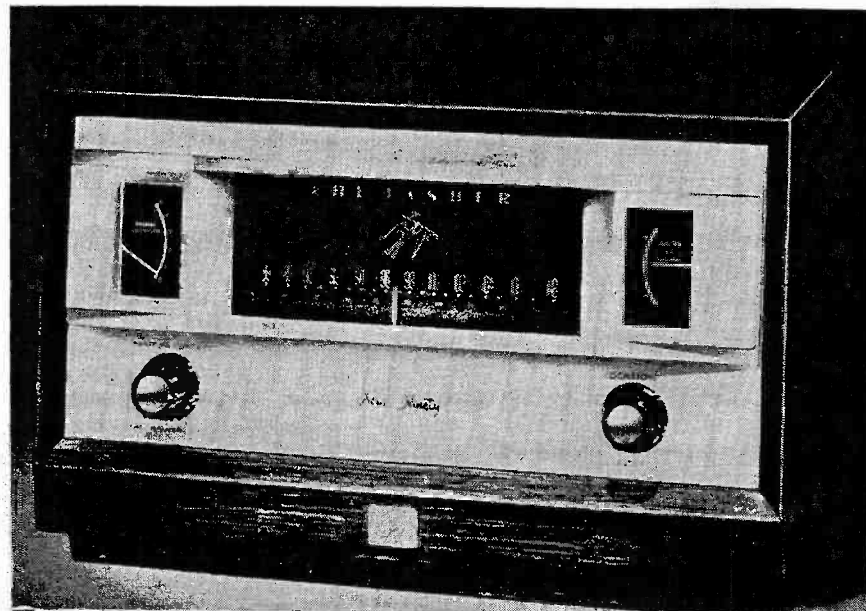

by RICHARD D. KELLER

book reviews

Handbook of Basic Circuits — TV-FM-AM

Matthew Mandl; pub. by Macmillan Co., New York; 365 pages; price $7.50.

This is a handbook of basic circuits — 136 of them — with brief descriptions of their operation and characteristics. It is intended as a reference work for servicemen, commercial and amateur radio operators, instructors, and technicians.

The 136 circuits are presented in alphabetical order for quick reference (i.e., automatic volume control, balanced modulators, bandpass amplifiers, bias clamps, etc.) and they cover the gamut of the basic transmitter and receiver circuits used in the AM, FM, and TV entertainment fields.

The diagrams furnished on each circuit are clear and straightforward, and the descriptions are lucid and non-mathematical. Several transistor circuits are shown, but the transistor RF and IF amplifier circuits are of the common-base configuration seldom used any longer. Most of the other circuits follow current good practice, including a number of color-TV and microwave circuits. The 20 appendixes seem to offer a wealth of useful information including VHF and UHF TV station frequencies and transmission standards, block diagrams and descriptions of complete transmitters and receivers, and information on UHF filters, among other things.

A bit expensive, perhaps, but certainly useful to people working with such circuits.

Mechanix Illustrated Hi-Fi Guide

Donald Carl Hoefler; pub. by Fawcett Publications, Greenwich, Conn.; 144 pages; $0.75; paper bound.

A profusely illustrated guide to the latest in high fidelity, this magazine-book contains a wealth of basic information for the sound-system novice.

Chapters are short and fully illustrated with pictures of the latest equipment. Five rules of thumb, or features, are given by which any system can be identified as high fidelity. These are: 1) a separate speaker cabinet; 2) at least three record-crossover selections (especially if the record collection contains many records made before industry standardization on the RIAA curve last year); 3) separate and independent bass and treble controls; 4) genuine two-way speaker system; and 5) a speaker enclosure of at least 2 cu. ft. (for 8-inch bass speaker — more for larger speakers, of course).

He discusses several packages which meet these specifications and also makes a number of specific component recommendations for complete systems in various price groups, starting at $160 and going on up to his "dream" system, representing the last word in hi fi today regardless of price, for about $800. Practical recommendations on kits are given, complete with many comments drawn from firsthand personal experience.

The author gets in a few jabs at some critics of high fidelity, and I think he comes out on top. Then there are chapters on stereo, tape recording, mobile hi fi, simple maintenance and trouble-shooting hints, and a very interesting panel discussion on loudspeakers by some of the top authorities in that field: Karl Kramer of Jensen, Abraham Cohen of University, Alexis Badmaieff of Altec Lansing, and Col. Paul Klipsch of Klipsch & Associates.

The book winds up with a discography of favorite hi-fi records by the nation's top disc jockeys, and a long list of free items and special bargains available to audiophiles.

Hi-Fi Loudspeakers and Enclosures

Abraham B. Cohen; pub. by John F. Rider, Inc., New York; 360 pages; price $4.60; paper bound.

A complete and well-illustrated volume, this work covers just about every aspect of the subject of loudspeakers and enclosures. The book is divided into three main sections: "The Loudspeaker", "The Enclosure", and "The Room". In the first of these the reader is taken in easy steps from basic loudspeaker principles, construction practices, and materials, to an understanding of the specialized high-quality reproducers of today.

Such items as speaker resonance, impedance, and damping are fully covered. Even the latest electrostatic and ionic speakers are simply, yet understandably,

Continued on page 41

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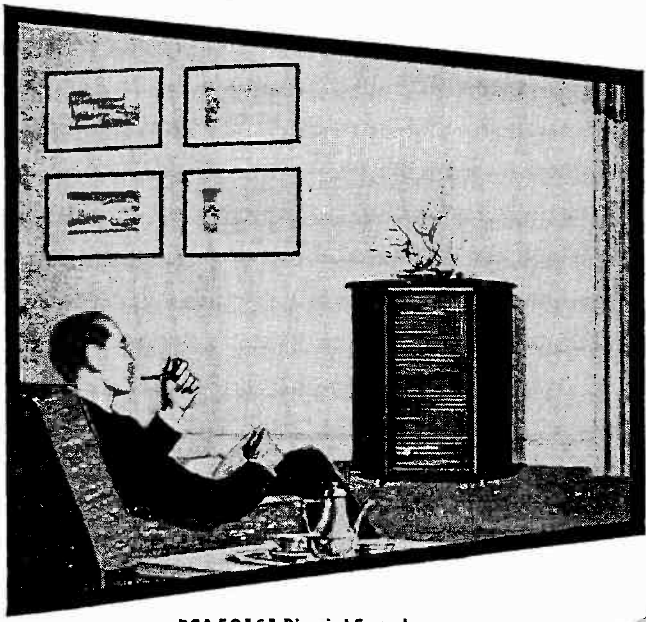

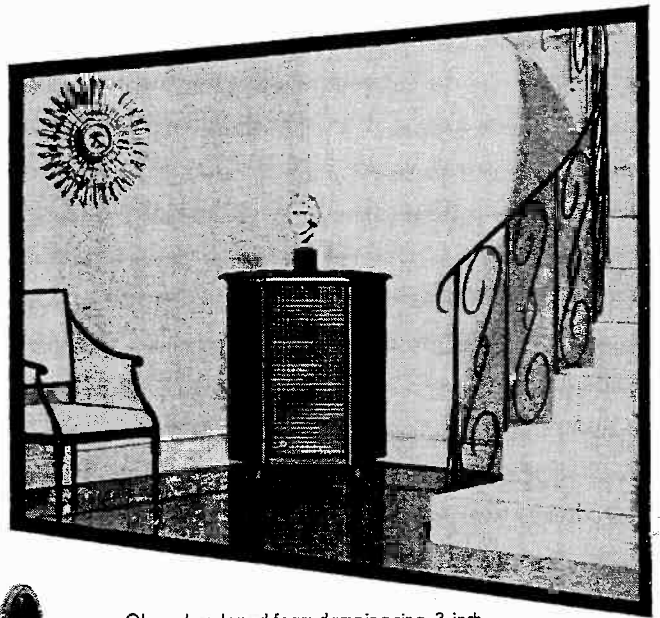

RCA 501S1 Biaxial Speaker—compares with models two or three times the price! Features 14.5 ounce Alnico-V magnet and 8-ohm voice coil. Its 12-inch woofer employs Olson-developed foam damping ring; 3-inch tweeter is mounted off-axis to minimize crossover interference; high frequency response extends beyond 18,000 cps.

Suggested User Price (optional) $33.50

RCA 502S1 Direct Radiator Speaker—outstandingly smooth, unsurpassed in its frequency range to 16,000 cps. Features same construction as 501S1 and incorporates medium-weight curved cone for smoothness, range, damping, and sensitivity to equal sound pressure of higher power speakers using double the power input.

Suggested User Price (optional) $25.25

RCA Enclosures—Beauty in cherry or blonde cabinetry, the *solid Honduras Mahogany* will enhance room decor with the expensive look and feel of fine woods. Built for a lifetime of use, RCA enclosures feature mortise-and-tenon joints, glue-block reinforcements and wood-screw clamping. For 12-inch speakers.

Suggested User Price (optional) 300W1 Cherry **$69.95**, 301W1 Blonde **$74.50**

301X1 Adapter Panel—converts 12-inch enclosures for use with 8-inch speakers.
Suggested User Price (optional) $2.45

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Readers' Forum

Gentlemen:
In the November (1956) issue you carried an article entitled "Rebuild Your Recorder", by Philip C. Geraci, which was quite interesting; so much so that I decided to rebuild an old Brush Soundmirror recorder for binaural sound.

A motor supply house in this area wrote to Eastern Air Devices for a price on the synchronous motor which was described in the article as costing $26.73. A reply was received quoting $45.00 for the motor.

Believing there was an error in the quotation, another letter was written to the company. They informed us that the $26.73 was a quantity price given back in 1953.

At the price of $45.00 for the motor, it would bring the rebuilding cost to well over $110.00. No doubt there are a number of tape recorders torn apart waiting for a motor. May I suggest that in the future you check the validity of prices quoted in your articles.

Raymond H. Ploch, D.C.
Hawthorne, N. J.

We also received a letter directly from Eastern Air Devices, Inc., stating that the price given in Mr. Geraci's article was incorrect, and suggesting that any references to prices be omitted from future articles.

We believe that prices are of interest to readers of construction articles, and are helpful in estimating total costs. This being so, we will continue to publish them whenever it seems desirable to do so. We try to make them as accurate as possible; there seemed no reason to doubt Mr. Geraci's figure on the synchronous motor, however, and (obviously) we didn't check that.

When asked about this matter subsequently, Mr. Geraci sent us 1) a photostatic copy of a quotation from Eastern Air Devices, dated February 2, 1956, in which the price of a single model LH731NCJ synchronous motor was given as $26.73; and 2) the following comment:

"I was amazed at the price of the EAD motor when I bought it. I had shopped all over the country, looking for a precision induction motor or a good sync motor, and prices for both are pretty steep. Many companies refused to quote a price until they knew the

Continued on page 47

EDITORIAL

GOOD news for audiocrafters (and for the rest of the listening and viewing public too) came in the form of two recent announcements from General Electric's Semiconductor Products Department.

First, a new germanium rectifier has been developed for use in TV-set power supplies. This in itself wouldn't be very sensational news; germanium rectifiers with power-handling capacity sufficient for that purpose have been available for at least four years. What *is* important is the new low price, which is supposed to make them directly competitive with vacuum-tube power rectifiers. According to GE, the new germanium units will cost about the same, or even less, than typical rectifiers now used in TV sets.

Failures of rectifier tubes (usually a 5U4 or 5V4 type) now account for about 10% of all television troubles, according to TV repairmen. On the other hand, it is claimed that the life expectancy of a germanium rectifier, properly designed and built, is virtually unlimited; it should certainly last the service life of a television set. Rectifiers have been a major source of trouble in hi-fi equipment also, particularly in power amplifiers. It is a more insidious kind of trouble, because a vacuum-tube rectifier does not often go bad suddenly, but deteriorates gradually. In a TV set this produces a visual effect: a distorted and shrunken picture, reduced brightness, and in some cases increased snow because of degraded RF sensitivity. In a high-fidelity amplifier, unfortunately, the trouble is not so obvious; increasing distortion and reduction of maximum power output are hard to detect when they take place gradually.

Since the same types of rectifiers are used in power amplifiers and TV sets, the new germanium units should be received warmly by all hi-fiers. Mass production is just beginning. They will probably appear in some 1957 TV's, and may be standard for 1958 sets. It won't be long before you'll be able to get them from your local service shop or from mail-order houses, who will have to stock them for TV replacement purposes.

The second announcement concerned a general price cut on 15 types of transistors used in portable radios, phonographs, and high-fidelity equipment. As a result of more efficient production techniques, and continually increasing sales volume, prices of these transistors have been reduced by 9% to 17%. They are now an average of 46% less than they were just a year ago.

The latest reduction, with similar cuts by other companies, now puts transistors in the same price-magnitude order as competitive vacuum tubes, if they aren't yet exactly comparable. With the less costly power supplies needed for transistor circuits, however, the price differential between equipments using transistors and those using tubes may even now be negligible. And future price cuts, which are certain to come with more manufacturing experience, should give transistors a formidable competitive advantage in applications for which they are suited. These applications are becoming more universal as transistor performance and uniformity are improved. Truly, the age of the transistor is at hand.

NEW schedule for High Fidelity Music Shows sponsored by Rigo Enterprises, Inc., for 1957 is as follows: Pittsburgh, March 8-10, Penn-Sheraton Hotel; Philadelphia, April 5-7, Lord Baltimore Hotel; Cincinnati, September 20-22, Sheraton-Gibson Hotel; Miami, October 18-20, McAllister Hotel; and St. Louis, November 22-24, Statler Hotel. These dates are different from those given in our December editorial, and presumably are now fixed. Once again, we urge you to attend and bring your friends if you live in or near one of these cities.

IN a recent letter, Utica (N.Y.) eye specialist John J. Stern, M.D. told us about a local hospital-equipment dealer who doesn't think much of the high-fidelity boom. At least three times a day people walk into his show room, ask to see a microscope, produce a stylus which they inspect carefully, thank him politely, and walk out. — R.A.

ERRATUM

In "Minimizing Pickup Tracking Error", part II, by Dr. John D. Seagrave, AUDIOCRAFT, January 1957, page 26, column 1, equation (27c), a parenthesis was displaced. The correct form of this equation is as follows:

$$D_1 = \frac{\left(\frac{1}{R_1} + \frac{1}{R_2}\right) \sin\beta - \frac{1}{L}}{\left(\frac{1}{R_1^2} + \frac{1}{R_2^2}\right)} \qquad \ldots(27c)$$

Electro-Voice KD-6 Aristocrat

An AUDIOCRAFT *kit report*

SPEAKER-SYSTEM enclosures in the popular Electro-Voice line have been available as kits for some time. Recently, however, the kit program at Buchanan was revised in two important respects: finer woods are now supplied, and several subassemblies are made before shipment.

In the case of the KD-6 Aristocrat kit, korina-veneer plywood is now used for visible exterior panels rather than birch plywood; solid korina molding and trim are furnished rather than poplar. All battens and glue cleats are now factory-installed on the panels, except for two small cleats, and openings for middle- and high-frequency horns, as well as for a 12-inch driver, are cut in the speaker mounting board. The new Aristocrat kit, therefore, is even easier to assemble; it is likely to be assembled more accurately; and it will look better when finished.

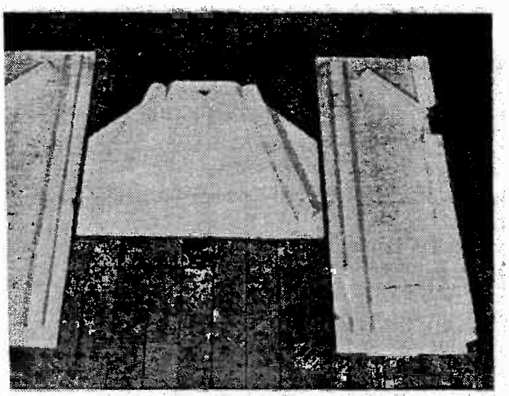

Fig. 1. In first steps of construction, legs are put on bottom panel and small cleat is attached to each side panel.

Construction Notes

The Aristocrat is a rear-loading folded corner horn. There is a resonating chamber directly behind the bass cone that is coupled, by means of a slot at the bottom of the chamber, to a horn that increases exponentially in cross section as it proceeds from the slot, upward and out the back of the enclosure, and into the room corner. The corner completes the horn flare, as it does in all Klipsch-licensed enclosures. Obviously, the Aristocrat must be used at the intersection of two walls and a floor or ceiling in order to preserve its bass response.

Construction is begun by installing legs on the bottom panel, Fig. 1, and the two short slanting cleats on the side panels. Location marks are scribed on the panels at the factory; since it is important to position the two short cleats accurately, this is a real help.

Top and bottom panels are then attached to one side, as shown in Fig. 2. Make sure that no glue has oozed out into the top-panel rabbet from the cleats attached at the factory; if it has, scrape it off with a sharp knife. This will assure tight joints all around the panel. Front edges of these panels should all be aligned, of course. Small brads driven through the cleats into the side panel will hold pieces in position while you insert the screws. The other side panel is then added in a similar way.

Following assembly of the side, top, and bottom panels, the back panels are installed. Fig. 3 shows one of them in place. Follow directions exactly here in order to get a good fit. It may be a good idea to slip the speaker mounting board in place temporarily, while installing the first of these back panels, to insure squareness of the whole assembly.

Fig. 4 is a view of the lower horn-forming internal baffle in position. Space at the bottom is the slot-horn throat. No particular problems here; simply make certain that this baffle is pushed as far back as it will go while still resting on the cleats below it.

The upper horn-forming baffle, trapezoidal in shape, is shown installed in Fig. 5. Its lower edge meets the board added during the preceding step; its sides are screwed to the back panels; its top abuts the top panel. Don't forget to use the shorter ¾-inch screws to attach this panel, and don't overlook the two screws that go into the top panel from the *back* of this board. Fig. 6 shows how this assembly looks from the back.

Photos by Warren Syer

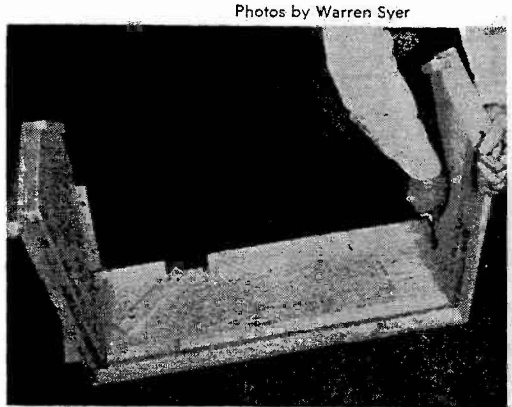

Fig. 2. Top and bottom panels should be attached to one side panel with screws and glue. Other side panel then goes on.

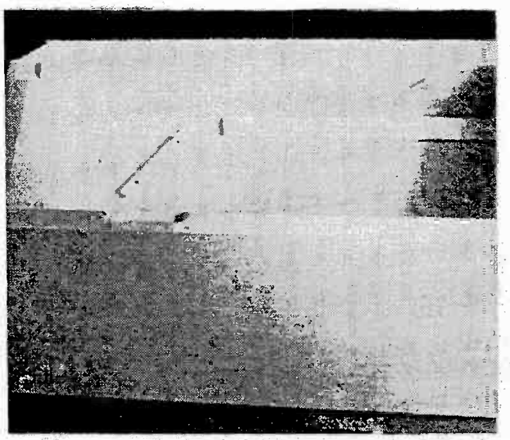

Fig. 3. The back panels are added next. Factory-installed screw cleats make it easy to get precise fits important here.

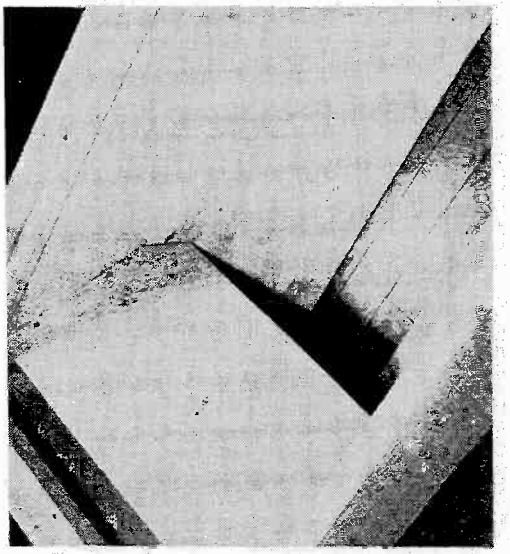

Fig. 4. It may take some effort to get lower horn-shaping piece firmly against back panels and short side-panel cleats.

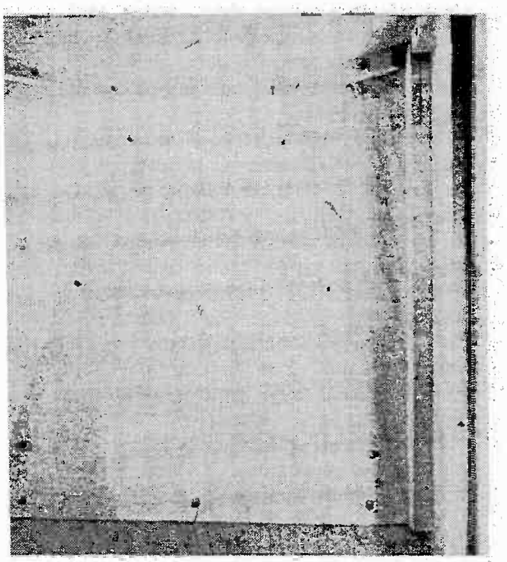

Fig. 5. Looks impossible — but this piece with its crazy angles fits perfectly at each of the panel surfaces it touches.

At this point you install the sound-absorbent pad on the inner surfaces of the speaker cavity. We didn't do this while taking pictures, to keep the surface details as clear as possible. But we did later, and found it fitted so well that a dozen tacks held it in place nicely. You can glue it in if you want to, but it isn't necessary to do so.

The speaker mounting board, Fig. 7, has holes cut for a 12-inch speaker, the Electro-Voice 8-HD middle-range horn, and the T-35 or T-35B tweeter. Cover plates are furnished for both horn cutouts in case you don't use one or both of them. A roll of adhesive-backed rubber gasket strip is supplied, which you cut to length and press onto the speaker-board screw battens as shown. In later tests (with a 12-inch coaxial speaker) we found that the large horn cutout cover plate leaked air and vibrated slightly when the enclosure was driven hard. There was plenty of extra gasket strip, so we used some of it behind the cover plate with a noticeable improvement.

Next you are supposed to tack the grille cloth to the speaker mounting board. Before you do this, install the mounting bolts for your speaker components (and for the cover plates, if you aren't going to use all the cutouts). Also, note that the speaker mounting board is secured to its battens by screws around the edge. When installing the grille cloth, you'll have to make small holes in it directly over these screw holes (not too large holes, or they'll show after the trim frame has been attached). Alternatively, you might use small washers designed for flat-head wood screws when you install the speaker mounting board, to prevent tearing and raveling the grille cloth.

The trim frame comes in four mitered sections which are installed individually, as shown in Fig. 8. This is a clever way of simplifying its assembly and of making it removable in case you have to get at the speakers later. The mitered edges fit together surprisingly well, and are a tribute to the precision with which these kits are manufactured.

You can finish your Aristocrat any way you please; korina is a remarkably versatile wood. It is supplied already sanded, so that only a light rubbing with fine sandpaper is required before you begin the finishing job. It should be noted that Electro-Voice markets complete finishing kits for walnut, cordovan mahogany, fruitwood, cherry, golden oak, and ebony finishes, at $5 each. Kits include a combination stain and filler, shellac sealer, gloss and satin varnishes, solvent, brushes, sandpaper, and instructions.

The Aristocrat kit itself costs $39, and includes everything necessary for its assembly except a phillips-head screw driver and a tack hammer. Even glue

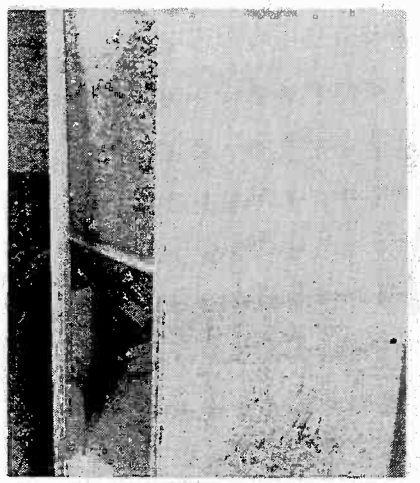

Fig. 6. Here is a back view of the construction so far. Rear-loading horn is completed by expanding room corner.

and tacks are supplied, as well as speaker-mounting hardware and decals. Ready-built versions are priced at $69 (mahogany), $76 (blond), and $79.80 (walnut). With the kit, therefore, you save $30, $37, or $40.80 minus finishing costs, and have available a much wider selection of finishes. The instruction book, No. IB6, is clearly written and illustrated. It contains detailed plans for all parts, so that you can do your own woodworking too if you want to; it costs $1 if ordered separately.

AUDIOCRAFT Test Results

Since the performance of any speaker system is so dependent on the speaker components themselves, it is virtually

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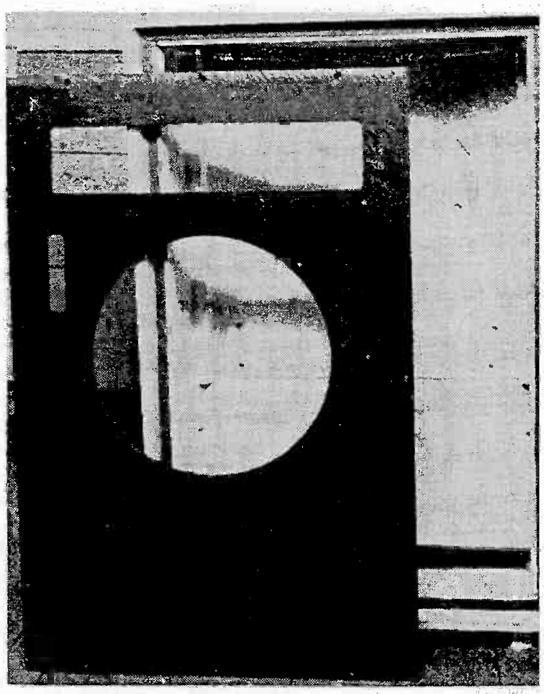

Fig. 7. Speaker mounting panel ready to put in place. Note rubber gasket on the battens; it helps to make air-tight seal.

Designing Your Own Amplifier

by Norman H. Crowhurst

Part VIb: Special Output Circuits

Unity Coupling

A variant of the pentode cathode-follower circuit is the unity-coupled circuit used by McIntosh, and shown schematically in Fig. 11. Half the output load is put in the plate circuit and half in the cathode circuit, using an equal number of turns in each. By cross-coupling the screens, the screen-

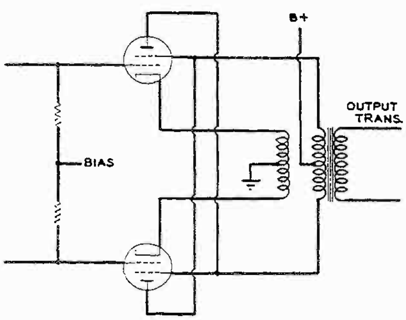

Fig. 11. Schematic of the so-called unity-coupling circuit. Here the tubes are operating as pentodes or tetrodes, with half the amount of feedback achieved by the cathode-follower circuit, Fig. 7.

cathode potential is maintained constant during the audio cycle as with the straight cathode-follower pentode.

It is important that the screen be tightly coupled to its corresponding cathode. For this reason the screen-and-plate winding is wound bifilar with the cathode winding; that is, the turns of both are put on side by side at the same time. Good insulation is needed between these individual turns, since the voltage between them is full B+. DC voltage on the screen will be the same as the plate voltage, as in the ultra-linear method of operation.

The drive swing will be a little more than half that required for the pentode cathode follower, since it will be made up of half the plate plus the full grid swing.

The reduction in harmonic distortion would also be about half that obtained in the cathode follower, resulting in about 0.5% without any over-all feedback. Source resistance will also be about double that of the full cathode-follower pentode circuit.

Cathode-Coupled Ultra-Linear

Another variant may be regarded as an ultra-linear circuit with partial cathode feedback. This is shown schematically in Fig. 12. The proper fraction of swing for the screens is obtained by this cathode winding, and the screens themselves are coupled to a B+ point. This enables a lower B+ to be used on the screens than on the plates, without the necessity for a separate screen winding. The circuit is used in the Bell 2200-C amplifier and also several in the Bogen line.

Circlotron

The last circuit we shall consider in this article is the Electro-Voice Circlotron circuit, shown in Fig. 13. Batteries are shown for the two separate B+ supplies necessary with this arrangement.

This circuit is named for the loop (or circle) consisting of the two output tubes and the two B+ supplies.

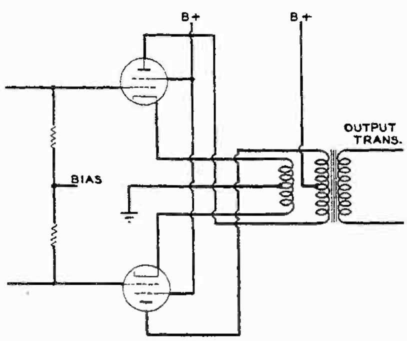

Fig. 12. Another variant is this virtual ultra-linear arrangement, employing partial cathode coupling. For 43% equivalent tapping, the cathode winding should have 43% of the total equivalent primary turns, while the plate winding should have the remaining 57%. This arrangement enables the plates and screen to operate at different DC potentials for increased power, if desired.

When the bias on each output tube is identical, which occurs at the quiescent condition, the current around the entire loop is uniform. But when one grid

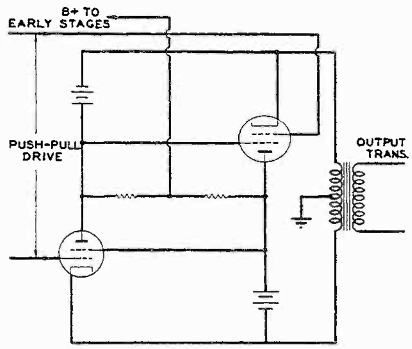

Fig. 13. A simplified schematic of the Circlotron circuit. Batteries have been shown in place of the two separate B+ supplies necessary for the system, both of which are isolated from ground.

is driven positive and the other negative, the currents passed by the two output tubes differ, as do the proportions of total voltage drop across each tube. Then the transformer connected between cathodes (which previously were both at the same potential) carries a load current.

The tubes here are acting as pentodes because cathode and screen are separated in each case by a constant potential. They are virtually in parallel, each cathode being connected to the other plate, with the output transformer being connected across the whole combination. This is where this circuit differs from the normal push-pull arrangement, in which the two tubes virtually feed the load *in series*. The result is that the plate-to-plate (or cathode-to-cathode—whichever you prefer to call it) load has a value one quarter of that proper for the normal push-pull output.

The B+ supply for the earlier part of the amplifier is achieved by putting two resistors in series between the two B+ points and taking the supply from the center point. Ground is provided by the center tap of the output transformer.

If the output-tube grids were returned to ground, the arrangement would be somewhat analogous to a cathode-follower circuit, because the drive at the grids would have to provide the grid swing in addition to the output swing on the cathodes. This is partially offset in the Circlotron cir-

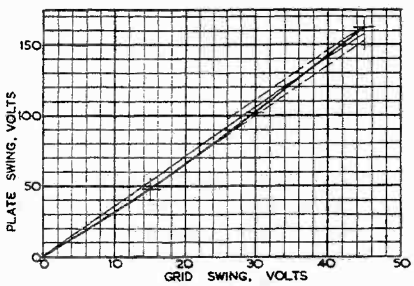

Fig. 14. Estimating the distortion of 5881 tubes as triodes from the transfer characteristic: the peak fundamental is about 158 volts, while the peak harmonic is about 5.5 volts; for this the peak ratio represents 3.5% harmonic.

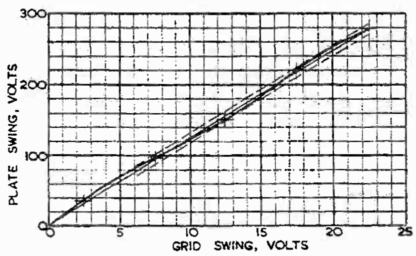

Fig. 15. The distortion estimated for the pentode (tetrode) operation: peak fundamental is 280 volts, and peak harmonic 8 volts, for about 2.85% total.

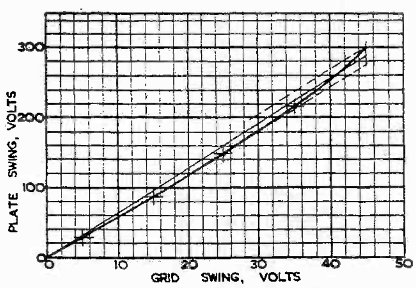

Fig. 16. Estimating distortion for ultra-linear operation, from the transfer characteristics: peak fundamental 288 volts, peak harmonic 11 volts, and total harmonic distortion, 3.8%. These figures give peak summation in each case, and so differ from the RMS text analysis.

cuit by returning the B+ for the push-pull drive stage to the positive voltage from the opposite output tube in each case. It is difficult to know whether this should be called a positive feedback or a reduction in negative feedback.

Estimating Distortion

So much for a number of popular varieties of output circuit. One thing more is needed to complete this article: a graphical way to estimate the distortion. This is illustrated in Figs. 14, 15, and 16 for the triode, pentode, and ultra-linear operation of the 5881 shown in Figs. 2, 6, and 9.

In each case the transfer characteristic is plotted, plate swing vertically against grid swing horizontally. Only one half of the characteristic is plotted because, being push-pull, the two halves are symmetrical. The negative swing would be exactly similar, but to the left and downward from the zero points.

The curve is plotted and then a straight line is drawn from the origin to pass through the curve in such a way that the excursions above and below the straight line (as determined by further dotted lines parallel with it) are equal on both sides.

This method of measurement does not determine the relationship between RMS harmonic and RMS fundamental, but between peak harmonic and peak fundamental. If only one harmonic is present, the two figures will, of course, be identical, but invariably more than one harmonic is present. Then the usual trend is that the peak relationship gives a higher percentage harmonic than the RMS relationship.

Although this means that the figure obtained from the construction will not agree with that obtained by the usual forms of measurement, it does mean that it gives a better practical indication of the annoyance value of the distortion, because multiple harmonics—especially high-order ones—are much more annoying than a single lower-order harmonic.

Fig. 17 is given to aid in identifying various harmonics from the third to the ninth, assuming they are present as separate entities. Here the fundamental is represented as being removed and the deviation shown horizontally.

Compare Fig. 16 with Fig. 17: the point where the curve crosses the center line should be 0.866 of the total excursion for the harmonic to be pure third. In Fig. 16 the crossover point is a little nearer the top than this, meaning that there are some higher-order harmonics although quite small in amplitude.

On the other hand, the crossover point shown for triode characteristics in Fig. 14 is considerably lower than 0.866, indicating in this case that there are probably components of fifth, seventh, and ninth.

The multiple curvature of the transfer characteristic in Fig. 15 can only be explained by the presence of harmonics at least up to the seventh.

Conclusions

Now, having studied different kinds of output circuits somewhat, are we in a better position to assess which kind of output — triode, pentode, or ultra-linear — is the best one to use? Perhaps it would be more accurate to say that now we can see why there is so much confusion on the issue and why different individuals prefer different circuits.

From our previous discussion of amplifier design we learned that the circuit should be designed to give the lowest possible distortion before any feedback is applied (and if possible this distortion should be of low *order*). Comparing the three methods of operation for the 5881:

The pentode seemed to show the lowest distortion, but the dominant components were third and seventh. Additionally, the fact that the load line goes across crooked characteristics means that any load deviation from a straight line of specific value will cause considerable variation in the distortion produced.

The ultra-linear circuit, with 3.3% almost pure third-harmonic distortion, and characteristics whose shape will not cause appreciably increased distortion when the load line deviates from its particular resistance value or goes reactive, is the most practical circuit of the three.

The triode-connected arrangement strikes an intermediate position, with 3.3% third- and 1.25% fifth-harmonic distortion. Variation of resistance value over a limited range will not seriously affect the distortion percentage, but reactive components are a little more likely to do so.

The unity-coupled arrangement

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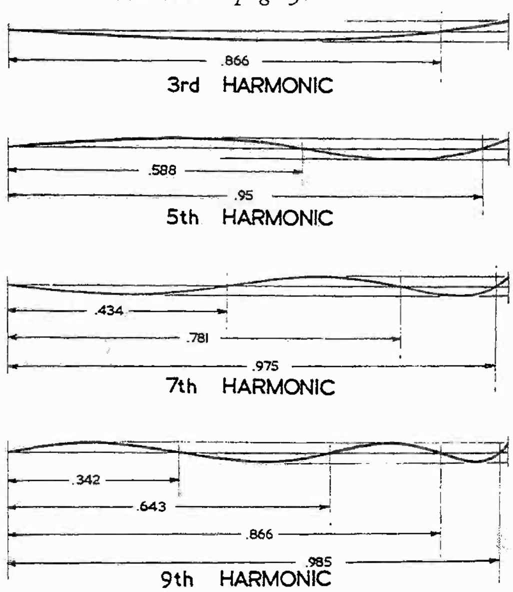

Fig. 17. This chart is to aid in identifying the various odd-order harmonics present from the shape of the transfer characteristic. The fractions give the points at which the curve should cross the straight fundamental line, when only that harmonic is present in signal.

HI FI IN YOUR CAR

by Philip E. Douglas

THE increasing number of high-quality, high-power FM stations, together with improved tuner circuits, now make it practical to have completely noiseless mobile FM reception. You can do it at minimal cost by installing an FM-only tuner in your car, making several changes in the audio system of the standard AM radio. The remarkable quality of FM reception, and the many fine programs broadcast throughout the day, lessen driving tension and make listening to the car radio as enjoyable as listening to a fine high-fidelity system in the home.

Obviously, a very sensitive FM tuner is necessary, since the RF noise level is high and the available signal is often weakened by intervening hills and buildings. In the installation to be described, we used a Harman-Kardon Model A-400 Counterpoint FM tuner, which met the requirements for mobile application very well. The sensitivity of the Counterpoint is high, and the discriminator is of stable design; its physical size is ideal for mounting under the dashboard, and it is electrically accessible for the necessary revisions. Since installing the tuner over five months ago, and after considerable driving, realignment has not been necessary — certainly an important advantage in mobile use, and one that is indicative of the tuner's stability.

To learn what might be expected of FM in a car, we recently took a trip through upper New York State. FM reception was possible at all times by switching from station to station through the mountains where AM reception was, at times, completely dead. In general, FM car reception is solid for the first 40 miles airline, with somewhat noisy (though by no means impossible) reception for the next 10 or 20 miles. This is an average, however; it will vary in different areas.

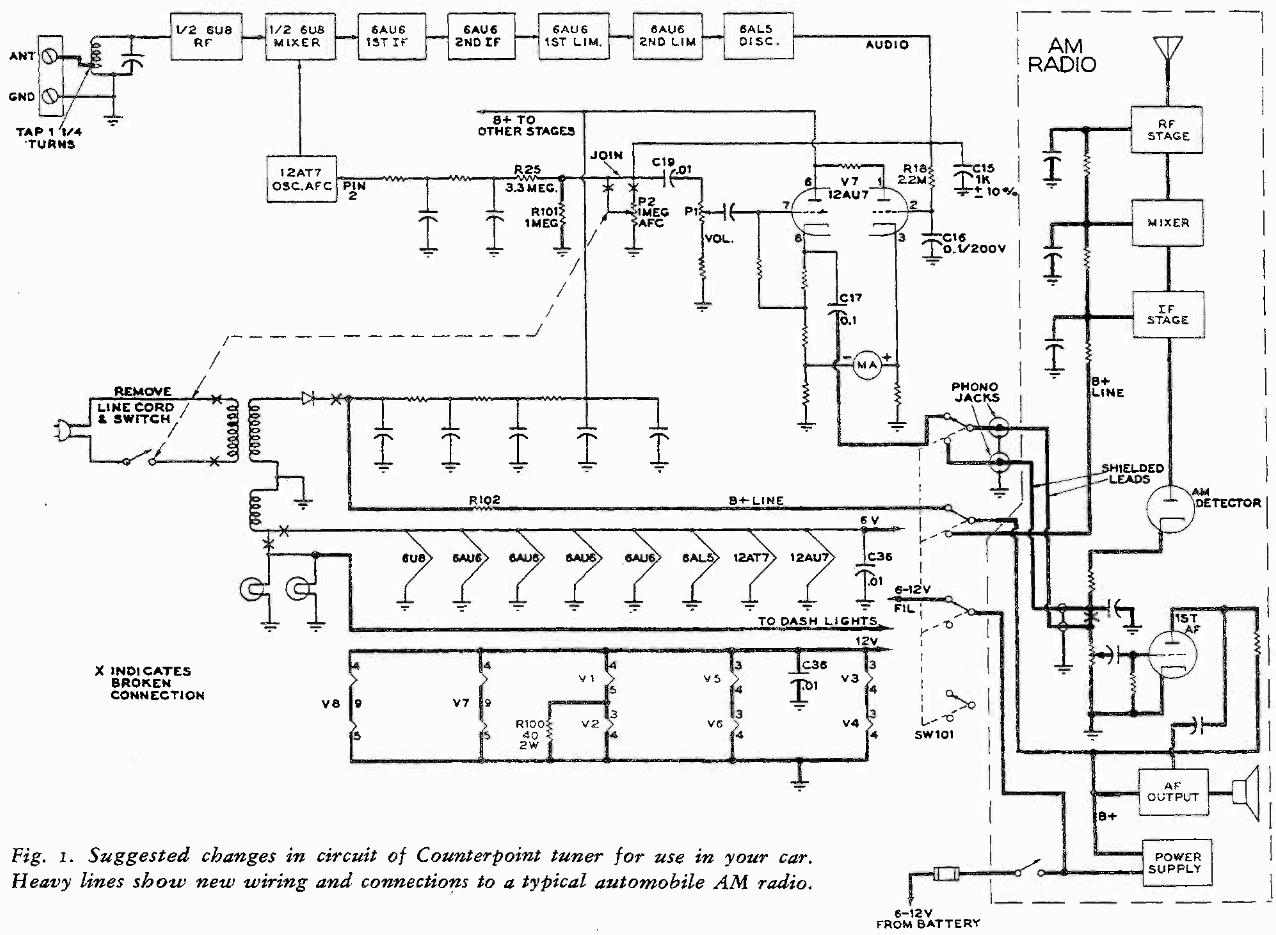

Fig. 1. Suggested changes in circuit of Counterpoint tuner for use in your car. Heavy lines show new wiring and connections to a typical automobile AM radio.

Counterpoint II

A few months ago Harman-Kardon, Inc., superseded the Counterpoint, Model A-400, with its superior Counterpoint II, Model FM-100. Here are instructions for modifying the new model, as given by the manufacturer:

Unsolder yellow wire leading from transformer to board, and tape the end. In its place will go the 6-volt supply lead from the battery. In the case of a 12-volt battery, connect a 25-watt 2-ohm resistor in series with the filament supply.

Unsolder selenium rectifier, being careful to keep a record of its polarity. Connect and solder the high-voltage supply lead from the automobile radio to the hole in the printed board that held the positive terminal of the selenium rectifier, as described in following paragraphs. A voltage-dropping resistor must be connected in series with a high-voltage supply lead; the formula to compute the value of this resistor is given in the article.

Remove all leads from the AC interlock receptacle and the on-off switch on the rear of the AFC control. Tape up the two transformer leads.

Drill a hole in the rear panel to take a 1/4-inch rubber grommet and feed through the hole a 3-wire cable. Connect one wire to the high-voltage circuit, one to the filament circuit, and one to the chassis. Mount a double-pole, single-throw switch to the dash board and connect the far end of the filament and high-voltage wires to it. Ground the far end of the wire connected to the tuner chassis. Complete the wiring by connecting the battery voltage lead and a lead from the high-voltage supply of the automobile radio to the remaining lugs of the switch. When the switch is properly connected it will act as an on-off switch for the tuner.

Tuner Modification

Adapting the Counterpoint for auto use, as can be seen in Fig. 1, is not difficult. First, the filament string must be changed to supply the tubes with the proper heater voltage. If the automobile battery is 6 volts, simply unsolder the green lead from the power transformer to the filament-string tie point, and tape the end. Do not cut this wire; you may want to put your tuner back in the home after the car wears out! In the case of a 12-volt battery, remove the green 6.3-volt AC lead from the filament tie point as before, and modify the filament string in series-parallel as shown in Fig. 1. Equalizing resistor R-100 should be a 2-watt carbon unit. Next, remove the wire from the rectifier output: in this case, the wire from the copper-oxide rectifier to C-32A. This is the point at which you connect high voltage (B+) from the auto-radio power supply. The value of dropping resistor R-102 must be determined by using the formula

$$R = \frac{E_S - E_T}{I_T},$$

where E_S equals the supply voltage of the auto radio, E_T is the high voltage required by the tuner, and I_T is the B+ current drain of the tuner. For the Counterpoint, E_T and I_T are 135 v and .05 a, respectively.

Next, remove the AC line cord. Install a grommet in the line-cord hole. Remove the wires from the on-off switch and AFC control P-2. Remove the knob,

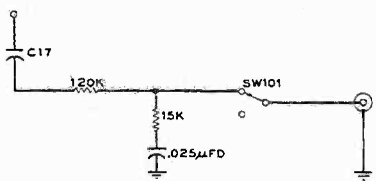

Fig. 2. Optional circuit for bass boost.

pot, and switch. Join the wire that came from the potentiometer's center contact to the wire from the ungrounded end of P-2, and connect a 1-megohm, 1/2-watt resistor, R-101, from the junction thus formed, to ground. This leaves the AFC operative in the maximum position; you won't have to worry about fine tuning in the car!

In place of the control P-2, just removed, install a 4-circuit, 2-position rotary switch, SW-101, Centralab PA-1011, or equivalent. Hook it up as shown in the interconnection diagram, Fig. 1. Disconnect all original wires designated by X.

Tap down the hot antenna-input lead on the RF coil to 1 1/4 turns from the ground end, to give an approximate impedance match with the RG-59/U coaxial cable used in place of twin lead from the antenna.

Cut off the switch shaft to proper length, and install the knob. The circuit shown in Fig. 2 may be added in series with the audio-output lead from C-17 to the switch SW-101, if it is needed, to furnish bass boost. Run all power wires out through the grommet in the line-cord hole. Use phono jacks on the tuner chassis for shielded audio-lead connections.

Mount the tuner under the glove compartment or dash, after all modifications have been made, using angle brackets bent as needed and secured with 8-32 machine screws or sheet-metal screws. Fig. 3 shows our installation.

The Antenna

A good antenna is a modified dipole cut for 98 Mc, approximately 58 in. total length or 29 in. for each element. However, if you want to favor one end or the other of the FM band, you may cut the antenna to the formula

$$\frac{\lambda}{4} = \frac{2820}{f},$$

where f is frequency in Mc and $\lambda/4$ is a quarter wave length in inches. The angle of the V formed by the two quarter-wave elements is 110°. This simply modifies the normal directivity pattern of the dipole to obtain a less pronounced null point off the ends. The antenna does *not* operate as a "V-beam"

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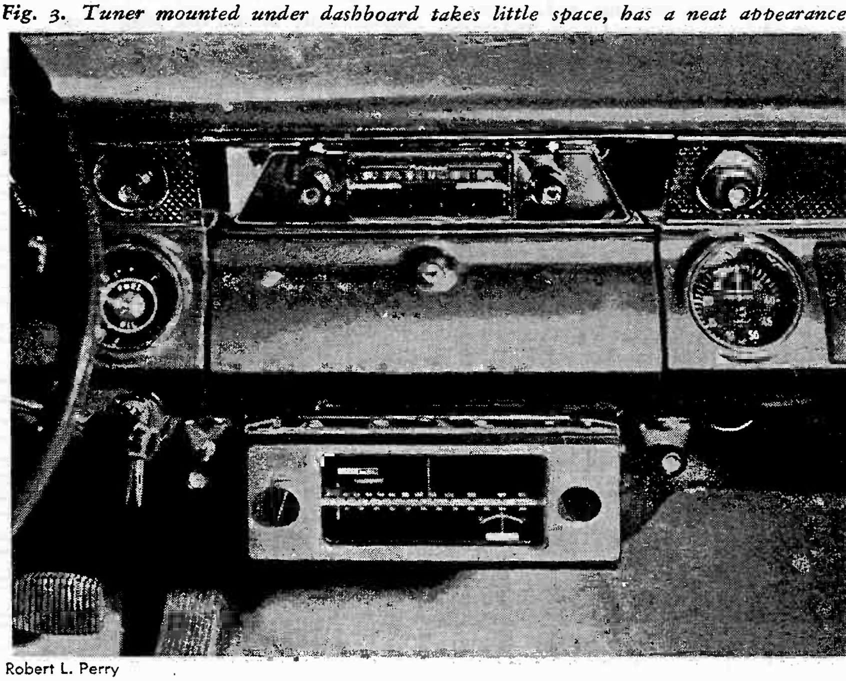
Fig. 3. Tuner mounted under dashboard takes little space, has a neat appearance.



Robert L. Perry

TRANSISTORS *in Audio Circuits*

by PAUL PENFIELD, JR.

IVb: Biasing the Transistor

Optimum Bias

In finding the best bias point for a given transistor in a given application, a number of factors are important. The bias current and voltage must be large enough to accommodate the expected AC peak voltage. The point may be chosen to minimize noise, distortion, or battery power. Bias variables must be small enough so that the dissipation rating is not exceeded at any time.

To be certain that the expected AC signal does not push the transistor into

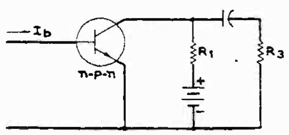


Fig. 11. Resistor R_3 reduces value of AC load without affecting the DC load.

a nonlinear region, we'll have to investigate the path of operation with AC signals.

In Fig. 11, the input signal is the current I_b. This contains a DC component and an alternating part. The DC part is the bias, and the alternating component is the signal of interest. We have already investigated how to locate the quiescent point on the graph; this is labeled *Q* in Fig. 12. But what happens when the input varies in accordance with the applied AC?

If it weren't for the capacitor and the extra load R_3, we would travel up and down the load line *EQ*. When the AC signal happened to be at a maximum, we might be up at point *A*. But with the signal at a minimum (that is, below zero) we would be further down, at some point like *B*.

However, the addition of the capacitor and the extra load changes matters. Now the effective load is smaller, since the AC load consists of two resistors in parallel, R_1 and R_3. The quiescent point, of course, is still determined by ignoring R_3, since the capacitor blocks DC. But the AC load line must be drawn on the basis of R_1 and R_3 in parallel. This corresponds, on the graph, to a steeper line.

Line *CD* represents the AC load line. As the input varies with an AC signal, we travel on the graph up and down this steeper line.

The position of this line can be determined as follows: take the equivalent resistance for the load (in this case the parallel combination of R_1 and R_3) and multiply it by the quiescent collector current. This will give a voltage which we add to the collector quiescent voltage. The point determined by this voltage sum and 0 ma collector current (point *C*) is on the AC load line. The line can now be drawn through points *C* and *Q*.

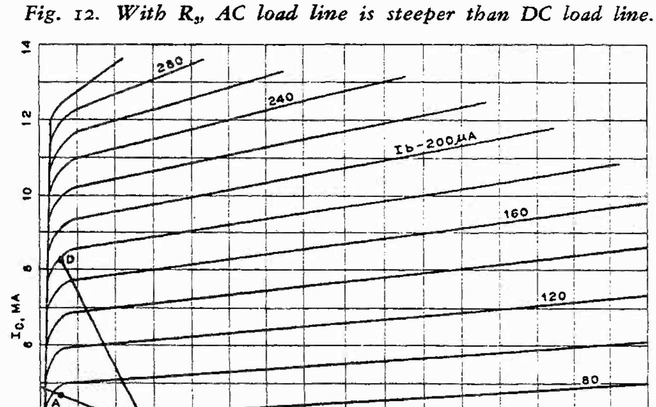
Now notice the lengths of the AC load line between the quiescent point and the nonlinear regions (saturation and cutoff). They are roughly equal. All other things being equal, it is good practice to make these lengths the same. Otherwise, with a large AC signal input, the excursion one way will hit a nonlinearity before the other does.

We must pick a quiescent point such that expected excursions along the AC load line do not take us into a nonlinear region. In addition, it is necessary to choose a quiescent point such that expected excursions along the AC load line do not go into regions of high current or high power dissipation. That is, we should plan to stay within the manufacturer's maximum ratings at all times.

It is simple to see on the graph if we are out of the safe region, and we should plan to stay within it.

So far we've seen two requirements for an optimum bias point. It must be far enough out so that the expected AC excursions do not go into saturation or cutoff. And it must be close enough to the origin so that at no time are the maximum ratings exceeded. Aside from these criteria, the designer is free to choose the quiescent point anywhere. Other factors, however, usually influence the decision. Notably, the circuit builder may want to bias for least power loss, or for least

Fig. 12. With R_3, AC load line is steeper than DC load line.



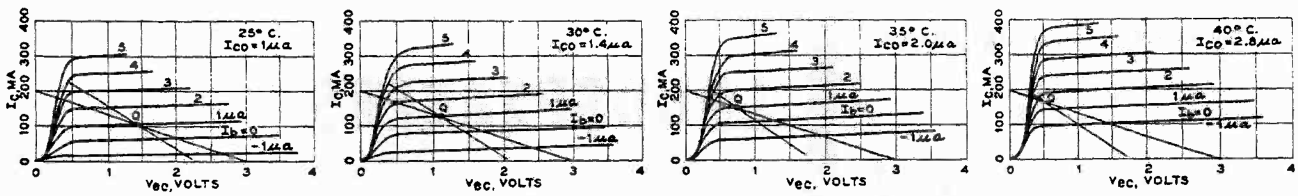


Fig. 13. In poorly designed circuits, relatively small changes in temperature can produce great changes in the bias point.

noise, or perhaps for a compromise between the three.

Temperature Effects

Another important consideration is just how the bias point will change with temperature. In low-power work, bias currents are often less than half a milliampere. A sudden increase in cutoff current, caused by a change in temperature, might be enough to change drastically the quiescent point. Fig. 13A, for example, is the chart of a transistor biased at about 100 μa, with a cutoff current I_{co} of 1 μa. Now suppose the temperature rises about 5° C., so that the cutoff current comes to 1.4 μa (Fig. 13B). We learned last month that all the curves would rise by just $(\beta + 1)$ times as much as the rise in cutoff current, or here, 20 μa. Since the curves have all shifted upward and the load line and base current have remained fixed, the quiescent point moves.

Fig. 13B shows also that the allowable input AC swing is reduced somewhat because of the shift in quiescent point.

If the temperature were to rise another 5° C., the cutoff current would jump to 2.0 μa (Fig. 13C). The quiescent point would be moved still further upward and to the left.

With another 5° change (Fig. 13D) the transistor drifts into saturation, and is effectively out of commission until the temperature comes down.

Fig. 13 shows dramatically, in an extreme case, the value of bias stabilization. It is extreme both because the transistor was biased at such a low current, and because a high DC load was used. A steeper load line would have stayed out of saturation longer.

Temperature had such an effect because of both the cutoff current and the circuit design. Ordinarily the circuit designer has no control over the cutoff current, but he *can* choose circuit values such that the base current decreases with increasing collector current, thus reducing the effect of temperature. These circuits are relatively easy to build once the principles are understood.

Bias Stabilization

The purpose of bias stabilization is to form circuits in which the base current decreases slightly when the collector current increases. In this way, the quiescent point is prevented from changing as much as it normally would.

A useful measure of the amount of bias stability is the so-called stability factor S, introduced by Shea[1]. S is merely the ratio of a small change in collector current to the small change in cutoff current causing it. If the circuit is set up so that the base current is constant, the stability factor S is just $1 + \beta$, since the curves on the collector family all rise by $(1 + \beta)I_{co}$.

It is possible to obtain a formula for S in terms of the circuit constants and the transistor gain, provided certain simplifying assumptions are made. First, we assume that lines of constant base current are horizontal; and second, that the base-to-emitter voltage remains, if not zero or negligible, at least constant.

Using these two assumptions, it is not too difficult to work out the mathematics for any given circuit, and get a number for S. These calculations have

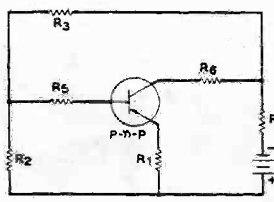


Fig. 14. General bias circuit; see text.

been made for a number of general circuits already[2]. Fig. 14 shows such a circuit. A number of practical bias circuits can be made from this circuit by letting the various resistors take on special values, including 0 and ∞. All the circuits we used this month, for example, can be made this way.

The stability factor for this circuit is approximately

$$S = 1 + \frac{\alpha\left[\frac{R_5}{R_2}(R_3 + R_4) + R_5 + R_3\right]}{(1-\alpha)\left[\frac{R_5}{R_2}(R_3 + R_4) + R_5 + R_3\right] + R_1 + R_4 + \frac{R_1}{R_2}(R_3 + R_1)} \qquad \ldots(3)$$

From this formula the stability of all the circuits we have discussed can be determined. Note that when $R_4 = 0$, $R_2 = \infty$, and $R_1 = 0$, the stability factor is $1/1 - \alpha$, or $\beta + 1$. This is consistent with our previous findings.

A low value of S indicates that the equipment is relatively stable; a high value indicates that problems might occur when the equipment is operated under conditions where the cutoff current changes from time to time.

That's just what the stability factor indicates — the freedom from change induced by changes in the cutoff current. For any one transistor, the cutoff current can be changed by temperature, by illumination, or sometimes by radiation. Further, in a standard circuit of some kind, the cutoff current might change when transistors are replaced or interchanged, because various transistors of the same type number differ widely in cutoff current.

A low value for S keeps all the changes in cutoff current from affecting the collector current appreciably. Just how low a value of S is required is a matter of judgment. The desire for a low S must be compromised with the desire for low power dissipation in the bias network, and the desire for high resistances shunting the signal path. Further, just how low S need be will depend on the normal value of cutoff current, the temperature range encountered, and the effects of shifting bias point. For experimental circuits, values of S up to 30 are not uncommon. For final apparatus, S less than 10 is usually adequate, unless extreme temperature ranges are expected.

Biasing a transistor, then, consists of a number of steps, and a number of pitfalls to avoid. First, the AC excursion that must be designed for is generally known. Second, the maximum ratings on the transistor are presumably known, as is the cutoff current and the temperature range to be encountered.

1) With this information, pick a suitable quiescent point on the graph. It must be chosen so that the expected excursions do not go beyond the transistor ratings, or out of the linear region. If desired, pick it for small distortion or small noise — how to do this will be covered in later installments.

2) To reduce the possibility of

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[1] R. F. Shea, *Principles of Transistor Circuits* (New York, 1953), Chapt. 6.

R. F. Shea, *Transistor Audio Amplifiers* (New York, 1955), pp. 96-99.

[2] Paul Penfield, Jr., "Transistor Bias Stabilization," *Audio*, XL (May 1956), pp. 34, 36, 38, 40-41; (Jul. 1956), pp. 20-21, 41-44.

by George L. Augspurger

LOUDSPEAKERS and ENCLOSURES

V: Speaker Evaluation

THIS is the final article in this series. It will be concerned primarily with the bewildering and often contradictory methods used to test and evaluate speaker systems. Before starting on this, however, I want to backtrack briefly and cover in more detail the question of crossover networks and multiple-channel systems. The statement has been made that two- and three-way systems are very popular, but a more technical discussion was put off until other matters had been taken up.

Now that the various types of speakers and enclosures have been examined, we can return to the possibilities of their use in multichannel systems. That multiple speakers are popular to the point of being identified with high fidelity itself is a rather alarming fact. In mass-produced instruments especially, a minimum of three speakers (total cost: $5.75) is heralded as being necessary for "true panoramic sound". It must be admitted that in these phonographs there is some advantage in the multispeaker idea. Three cheap speakers sound a little better (or a little less bad) than one cheap speaker, and take up less space than a single larger unit. The sad part is that the average audiophile does not seem to be able to make the mental jump from three cheap speakers to one good speaker. But surprisingly, perhaps, some authorities believe the ideal speaker is of necessity a single, wide-range unit!

In any case, multiple-channel systems have their disadvantages as well as good points, and it should be helpful if we can consider the woofer-tweeter design problem objectively. Dividing the spectrum into bands, with each range of frequencies handled by a separate speaker, has a number of points in its favor:

1) *Ease of design.* By using a particular speaker to cover only those frequencies which it reproduces best, many contradictory physical considerations are greatly simplified.

2) *Lower distortion.* Since each speaker reproduces only that part of the audio spectrum which it can handle best, it is easier to keep distortion within reasonable limits. It is in the matter of intermodulation distortion that the multiple-channel system is shown to best advantage. Since intermodulation is usually *measured* as the interaction between high and low frequencies, all one has to do is reproduce highs and lows separately through individual drivers and, presto, the measured distortion has vanished! Although this is highly desirable, the test results are not always indicative of true listening quality. Electro-Voice, in particular, must be complimented for the hesitation with which they publish intermodulation readings, cautioning the buyer that different test conditions (any one of an infinite number of other combinations) might produce a figure much better or much worse than that quoted.

Courtesy Klipsch & Associates

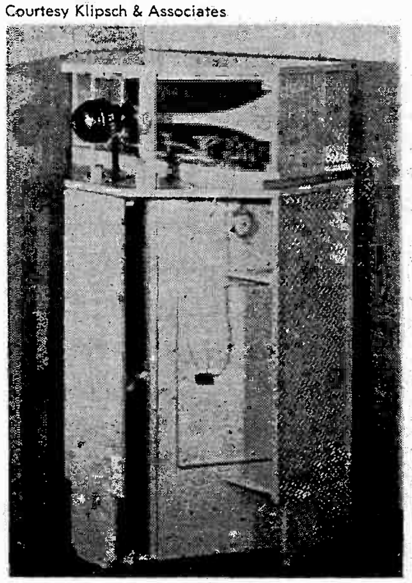

Fig. 1. A 3-way system without controls.

3) *Better directional characteristics.* As we have seen, it is difficult to design a single wide-range speaker in which the highs are not concentrated into a narrow beam. By feeding the treble frequencies to a multiple array of tweeters, or a specially designed horn, satisfactory spatial dispersion can be achieved. Note that the audible effect of such dispersion is *not* to spread the sound source, but to approximate a nondirectional *point source* of sound.

4) *Possible acoustic compensation.* By varying the levels of individual speakers with pads, it is possible to adjust the tonal balance of the system in a way that cannot be done with ordinary amplifier tone controls. Paul Klipsch is vehemently outspoken against pads in speaker lines, so if you want to stay on the safe side of all the authorities, the best solution might be an electronic crossover with separate power amplifiers for each speaker. With this setup, the level on each channel can be adjusted without affecting speaker damping. Klipsch himself uses speakers selected to have similar efficiencies, fed from a three-way dividing network with *no* controls. See Fig. 1.

So much for the points in favor of woofer-tweeter-squawker-honker combinations. What about the disadvantages? "There must be *some* disadvantages, Augspurger," says the wary reader, "or you wouldn't have spent so much time leading up to them." Very well, here are the main drawbacks.

1) *Cost of crossover network.* A good dividing network is not given away, and the cost goes up as the number of channels and cutoff slope increase.

2) *Interference holes.* Woofers and tweeters don't stop operating suddenly above or below their nominal operating ranges. The cutoff slope for most crossover networks is 12 db per octave, which means that the speaker may put out audible energy two octaves away from the theoretical crossover point. Since a separate woofer and tweeter are at least 8 to 12 in. apart, and the two sections of a coaxial speaker (Fig. 2) a few

inches apart, certain frequencies will cancel and others will reinforce around the crossover frequency because of varying phase relations. This effect produces interference holes which are almost impossible to eliminate. Personally, I don't feel that this is too serious in most cases. Our ears are used to diffraction and interference effects occurring with great complexity in live music. Further, the speakers themselves have response variations that are usually just as severe.

3) *Spatial separation.* Since the sound sources for various parts of the spectrum are separated, there can be an audible separation between the fundamental tone of an oboe, let us say, and the upper harmonics of the note. A single instrument playing a run may suddenly seem to shift its position as the speaker crossover frequency is negotiated. Such spatial separation is not pronounced in well designed systems, but when it is noticeable, no matter how slightly, it can be extremely annoying.

4) *Timbre separation.* Not only does a multiple-channel system tend to give audible evidence of its split personality through spatial separation, but the response of the different speakers in the array may sound different in quality. This is an effect which can't be measured, which many manufacturers claim does not exist in *their* designs, but which is easily the most prevalent and most noticeable deficiency of such systems.

Different vibrating materials, different designs, and different baffling arrangements will inevitably give different coloration to the reproduced sound. It is no good saying that individual speakers produce no coloration over their operating ranges. Even Hartley, whose speaker "produces no sound" in the advertising copy, admits in his technical writing that *any* speaker has noticeable coloration, and that the trick is to keep the timbre uniform from one end of the spectrum to the other. This is something that has to be done entirely by ear, and which is difficult and nerve-racking at best.

The effect of this characteristic timbre shift in multiple-channel speaker systems is to change the character of an instrument while it moves from one register to another. The vocal quality of a singer may be suddenly altered as she is electronically parceled out to one speaker and then another. It becomes more noticeable (or less, depending on whose ads you read) as the number of channels is increased. Because of this, some manufacturers draw the line at two channels — on the theory that a good single-channel system is too hard to build, while with more than two channels the potential for realistic sound decreases. A different approach is used by Bozak, who builds three different speakers, each for a different frequency range and different in size, but all designed of the same materials and the same general configuration.

In any case, the debate largely resolves itself into the number of channels which is to be regarded as optimum, because the ideal truly wide-range single-channel system does not exist. Perhaps it will when the electrostatic-speaker designers have produced what they are promising. The most outspoken advocate of the single wide-range speaker is Mr. Hartley, but, while the Hartley 215 represents outstanding design and performance, it uses a segmented cone and what amounts to two voice coils. It cannot really be called a single speaker in the true sense. Hartley has spent years developing a two-way speaker which is as nearly a single unit as he can make it. Other designers prefer to separate the individual units physically and electrically. You have to select whichever sounds best to you.

Courtesy Jensen Mfg. Co.

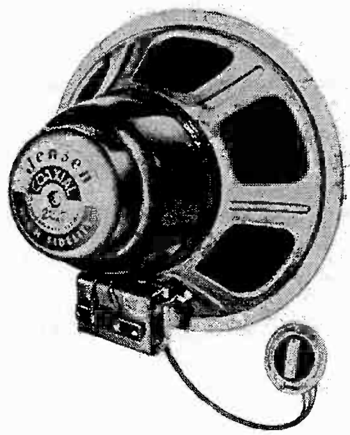

Fig. 2. Coaxial speaker with crossover network and level control for tweeter.

Speaker Test Methods

Let's examine the various laboratory test methods which are in common use and see, if possible, how reliable an index they give to audible results. There are about eight different tests normally used to check speaker systems: free-field response; polar response for frequencies above 2,000 cps; impedance (see the September 1956 installment of this series); harmonic distortion; intermodulation distortion; power-handling ability; transient response; and efficiency.

It seems a pity that even with all these factors specified, the consumer knows hardly more about the speaker than before he waded through the list. Even trained technicians, working with the same material and test equipment for years, develop only a rough correlation between laboratory results and what their ears hear. This is not to say that such measurements have no value; to the experimenter, they often furnish measurable indications of what is happening under certain conditions and give clues as to what should be done to improve a particular unit under test. But the average listener should not buy a speaker on the basis of its specifications. The only case I can think of in which such published curves are helpful is in choosing between different systems made by the same manufacturer, all tested under identical conditions in his own laboratory.

Why is it that so little guidance can be obtained from speaker measurements? Certain factors, it must be admitted, are universally accepted as accurate and useful: the nominal impedance rating, for example. But a plain, ordinary response curve, which is so useful for a preamplifier, becomes most ambiguous for a loudspeaker. First, most tests are run free-field; that is, with no acoustic reflections or resonances cluttering up the curve. Some companies suspend the speaker and microphone over a large body of water. Other companies use an anechoic chamber which is completely absorptive except at low frequencies. Since many speakers are designed to be used in a room corner, the low-frequency response may be run separately in a "simulated living room", to produce a nice response curve which looks surprisingly like a picket fence.

So far it sounds simple; but what do you do when you find that differences of 10 db or so can be achieved at various frequencies simply by moving the microphone two or three inches? And what do you do when you think you have a representative curve of your speaker, but then another laboratory tests it in *its* anechoic chamber and the two curves have nothing in common whatever?

We've barely got started on this. What about clearly audible peaks which show up as dips on the response graph? What power should be used to drive the speaker? What damping factor should be used? Varying the damping factor will affect the bass response of some systems remarkably. And then there is the tacit assumption that a sine wave is the thing to use for testing speakers, but is it? It seems self-evident that only sine-wave testing will give valid information for a frequency-response check; any other wave form contains harmonics, and some would be attenuated more than others.

At the same time, it must be admitted that pure sine waves never occur in speech or music. Perhaps a complex wave form used as the source of a frequency-response test would produce a graph having more meaning, so far as audible response goes, even though it could not be compared with standard laboratory response curves. This is a rather interesting idea, and its possibilities are even more provoking in view of the fact that at least one topnotch engineer has used square waves successfully for speaker testing.

Pro-Plane speaker systems are tested with square waves — the published frequency-response curves are run with a square-wave source. I asked Antony Doschek, the firm's engineering head, to explain, and he replied with the following:

"Speech and music are not sine waves, and the various loudspeaker adulterations and ear distortions which are excited by the complex waves of program material are completely ignored in sine-wave testing. The response curves of Pro-Plane speaker systems are drawn by connecting the check points of a C-major triad through ten octaves, with square-wave excitation. Square waves are used because they have a more uniform and extended harmonic series than other easily generated wave forms, without the strong second harmonic of a sawtooth wave. In short, I feel that the complex response of an enclosure is not excited by sine waves, but is excited by a complex wave. The curves were taken with the speaker suspended 40 ft. above the Allegheny River, and a calibrated microphone was hung 5 ft. in front of the speaker on its axis of projection. The rectangular wave curve thus produced is more definitive to me than any anechoically drawn sine-wave curve that I have tried to correlate with my own hearing."

One more point that should be mentioned is that of extending the measured bass response of a speaker system to match the equally outlandish claims of competitors. The most common method, especially for inexpensive midget systems, is to drive the speaker with about five watts more than it can possibly handle with any degree of fidelity. The oscillator is adjusted to 500 cps, whereupon a raspy 500 cps emerges from the overdriven speaker. The level is duly noted. Then 100 cps is fed to the speaker; a loud buzz consisting almost entirely of distortion components blats forth and the level is marked down. Fifty cps! The cabinet shakes, and rattles and hoots fill the room. The general effect is that of loose shutters on a stormy night. But the loudness is noted on the graph as the 50-cps response.

This sort of thing, needless to say, is not done by the reputable manufacturer. Unfortunately, the opposite extreme produces equally misleading curves. The well equipped test laboratory is set up to read a completely accurate, completely automatic loudspeaker response curve. About .05 watt is fed to the speaker, and the production model is found to have acceptable bass response to 40 cps! Only two things aren't mentioned: the speaker will never produce a 40-cps tone loud enough to hear, since its power-handling ability at that frequency is so minute that only a microphone can sense the delicate wave form; and, while the 40-cps tone is measured accurately (if inaudibly), it occurs at the trailing end of a 12- or 18-db-per-octave slope.

Fig. 3 should demonstrate the second point more clearly. A prominent resonance is deliberately introduced at some frequency around 75 cps. This is actually as low as the speaker can honestly put out decent sound. By blowing up the resonant point to 6 db above the 1,000-cps reference level, the dwindling response at 40 cps is tugged up to only 6 db below the 1-Kc level. Result: response ±6 db from 40 to 15,000 cps. Audible result: a ubiquitous twang at 75 cps, with nothing audible below 60.

The other types of specified measurements, such as harmonic distortion, are useful primarily to the manufacturer, who knows precisely the conditions under which the tests were made.

Since published specs offer no real basis for judging a loudspeaker, what about comparative listening tests? It would seem logical, after all, that if a speaker is to be used for listening, the

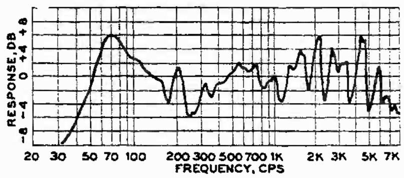

Fig. 3. Typical speaker-response curve.

way to test it is to listen to it. Many authorities won't agree with this, and it must be admitted that there do exist complications not evident in the simple statement, "Listen to it." But to my way of thinking, there is no way of escaping the obvious fact that if a speaker system satisfies me, no amount of testing or arguing will affect my evaluation until I hear something that sounds better.

Most audiophiles read articles telling them how to listen and what to listen for, but they choose according to specifications rather than their own ears. This isn't meant to be critical of such articles; they perform the valuable service of warning the purchaser in advance of defects he may not hear until he has listened to his system for some time. Indeed, excellent papers on subjective evaluation of speakers appear in this magazine. Since so much information is available, there is no need to repeat it here. However, one or two important considerations in the art of judging by ear are sufficiently controversial to point out.

First of all, shall we compare speakers with other speakers or with live music? Unless one is able to attend a public demonstration of live *vs.* recorded sound, he has no opportunity to get a direct comparison between the speaker and a live source. Furthermore, there is the disturbing fact that such live *vs.* canned tests have been used most successfully to promote quite ordinary radios and even acoustic phonographs! Here again, the ability of a certain speaker to reproduce indistinguishably a live oboe played in a small concert hall may give no indication of its worth in reproducing a vocal chorus in your living room.

The important thing, the *really* important thing, is to hear a speaker in your own home. Room acoustics can alter response by 15 or 20 db at some frequencies, and if you get a speaker which peaks at a resonance point of your listening room, you'll be sorry no matter how good it sounded at the audio showroom where you bought it. Conversely (and I have observed this effect more than once), a speaker which *has* a built-in peak *may* compensate for your particular room acoustics to provide more satisfactory results than a truly flat speaker.

Before buying a speaker it is best to try it in your home for at least a week, listening critically with tone and volume controls set as you would normally adjust them. If you can obtain two or three speakers to try simultaneously, so much the better, but adjust your amplifier controls separately to get the best possible sound from each before comparing. Don't use jazzed-up "hi-fi test" recordings to evaluate speakers. Such recordings will sound startling on almost any kind of reproducing system. Organ recordings are likewise extremely misleading; the instrument produces so much throughout the entire bass range that the illusion of tremendous bass can be achieved from even the smallest speaker system.

A few additional tests you can try do not involve listening to recorded music. If the cone of the bass driver is accessible, disconnect it from the amplifier and tap it with your finger. A dull, undefined thud is good; a blatant hard tone is not. If the cone is hidden, connect an ordinary dry cell to the speaker cable. Some speakers will emit a definite short "bong" when the connection is made and broken, but a truly nonresonant unit will make only a sort of undefined click.

In any case, an extended test under actual normal listening conditions is the best test for any piece of sound equipment. It is a sad truth that *no* loudspeaker will reproduce naturally all the recorded material available. Even the best speakers are far short of perfection, and the ideal speaker for you is that one whose good points please you particularly and whose deficiencies irritate you least. Make up your mind which speaker sounds best to you. *Then* read the advertisements and specifications for moral support.

portable hi-fi system

by JOHN K. SMITH

JUST when the idea for a sound system in the classroom occurred to me is difficult to say, but with a background in hi-fi experimenting that extends to pre-LP days, the basic idea was probably in my mind from the beginning of my teaching career. There often arise situations in the classroom that go beyond the scope of ordinary teaching techniques. In some of these situations, it seemed to me, the ability to record and reproduce sound might be of value.

After selling the school board on the idea, my natural course was to investigate the availability of commercially built equipment suitable for classroom use. After careful search, I found that there were record players available that might be adequate. There certainly were PA systems that would suffice. There were tape recorders that would do, and there were radios that could be used. All were individual units, however, and the accompanying duplication of functions in each piece of equipment (if quality was maintained) made their purchase uneconomical, to say the least. It rapidly became clear that what I required could not be bought, and the only apparent solution was to build the unit.

In looking for equipment that would do all of the things deemed important, there evolved certain criteria. It had to be a single mobile unit, able to record and play back tape, play records, record on tape from records, play and record from radio, be usable as a PA system, and be small enough to pass through a door to any room where it might be used. All these properties would have to be provided for in such a manner that the quality would not be impaired. Also, equipment selected had to be well enough engineered to preclude early obsolescence. This requirement alone can cause quite a few gray hairs, considering the rapidly changing picture in hi-fi equipment.

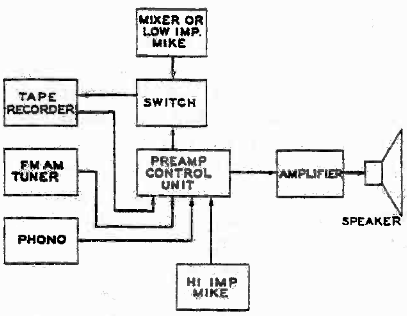

Fig. 1. How equipment is connected.

One of the first things I did was to draw a block diagram that would indicate roughly how things might be accomplished. This done, Fig. 1, the problem then centered around the selection of the individual pieces of equipment. I will list what was finally decided upon rather than subject the reader to the long story of how this selection was made. Needless to say, the ugly head of cost reared itself repeatedly. The rack-mount Ampex 350 was selected as the recorder, with a McIntosh 30-watt amplifier, Fisher FM-AM tuner, Fisher control unit, Rek-O-Kut variable speed turntable, Pickering turnover cartridge, Clarkstan arm, and Electro-Voice SP12 speaker. (A second unit was built for another school in the district, and in this unit a Pickering arm was substituted for the Clarkstan.)

The equipment, less cabinet, came to $1,796. Cabinet cost would be difficult to estimate, since most of the labor was volunteered.

With the equipment chosen, and the hookup roughly diagrammed, the next objective was to design a physical layout that would enable the equipment to be mounted in a single mobile housing which would go through a door. I measured all the doors and almost gave up the job. I was already asking for the moon, and to ask also that the school be remodeled so that this machine could go through the doors would have been the last straw.

But, by judicious juggling and compromising the optimum placement of units, a layout was evolved which would allow exactly 1 in. clearance on each side of the cabinet when going through the smallest door. This is shown in Fig. 2. The final outside dimensions with

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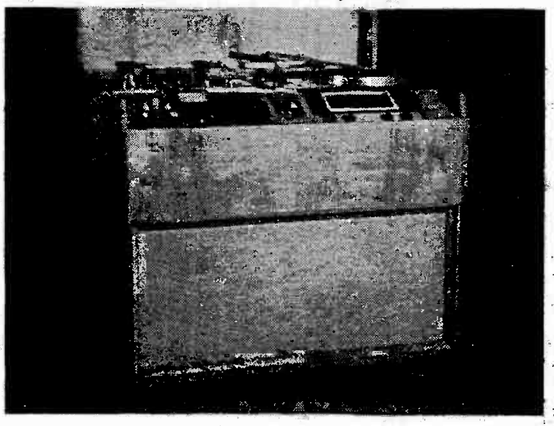
Fig. 2. Cabinet evolved for the system.



BASIC ELECTRONICS

by Roy F. Allison

XVa: Parallel Resonance

IN preceding chapters we have covered all possible series combinations of *R*, *L*, and *C*, and parallel *RL* and *RC* combinations in AC circuits. It remains to discuss *LC* and *RLC* parallel circuits.

A parallel *LC* circuit is shown in Fig. 1, with component values indicated. This is an idealized circuit, since no resistive component is included. It is useful, nevertheless, for the purposes of preliminary illustration, and often is usable with minor modifications for practical circuit calculations.

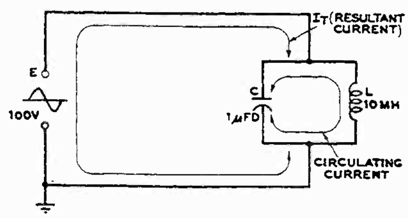


Fig. 1. Parallel LC circuit without R.

We are aware that the current in a capacitor always leads the voltage across it by 90°, and that the current in an inductor lags the applied voltage by 90°. Also, in a circuit such as that of Fig. 1, the magnitudes of these individual currents are determined by source voltage and the individual reactance magnitudes. The reactance magnitudes are determined, in turn, by the capacitance and inductance values, and by the frequency of the AC source. Let's assume at first a source frequency of 637 cps. Applying the usual formulas ($X_L = 2\pi fL$ and $X_C = 1/2\pi fC$), we find that at this frequency X_L is 40 ohms and X_C is 250 ohms. Current in the inductor is, of course, E/X_L: 100/40, or 2.5 a. I_C is E/X_C: 100/250, or 0.4 a. These currents are shown in Fig. 2A, drawn to scale and in proper phase relationships. Since the currents are exactly out of phase with one another, they are directly subtractive; their resultant current, which flows through the source, is the difference in their magnitudes. I_T is, by this reasoning, 2.1 a, and is entirely inductive. The appropriate vector diagram is shown below the wave-form representation.

Now, suppose that the source frequency is changed to 3,980 cps. X_L increases to 250 ohms, and accordingly I_L decreases to 0.4 a. But X_C becomes only 40 ohms at that frequency, and I_C consequently increases to 2.5 a. The situation is exactly reversed, as Fig. 2B shows. The capacitive current is now much larger than the inductive current, but they are still subtractive as far as the source is concerned, because they always flow in opposite directions. Again, the resultant current is 2.1 a, this time purely capacitive.

Somewhere between these two frequencies (at 1,592 cps, to be exact), the capacitive reactance becomes equal to the inductive reactance. At this frequency each reactance is 100 ohms. The current in each, therefore, is 100 v/100 ohms, or 1 a. One leads the voltage by 90° and the other lags it by 90°, so that their resultant is zero, as shown in Fig. 2C.

If we calculate the total impedance of the circuit in each case, dividing the source voltage by the source current (I_T), we find that it is about 48 ohms inductive at 637 cps and 48 ohms capacitive at 3,980 cps. In any other parallel circuit not containing both *L* and *C*, we have found consistently that the total impedance is less than that of any parallel component — but, in this case, it is higher than that of one of the parallel components. If calculations are made for 1,500 cps, the resultant impedance is found to be above 800 ohms (inductive), which is higher than the reactance of *either* parallel component. Further, for exactly 1,592 cps, the resultant current is zero; therefore, the resultant impedance must be infinitely high! At a frequency slightly higher than 1,592 cps, the resultant current is still small, and the total impedance is again very high, but now it is capacitive.

The reason for this seemingly odd behavior is simple: the currents in the two parallel branches *always* flow in opposite directions when the branches are purely capacitive and purely inductive. It follows that the *resultant* current is

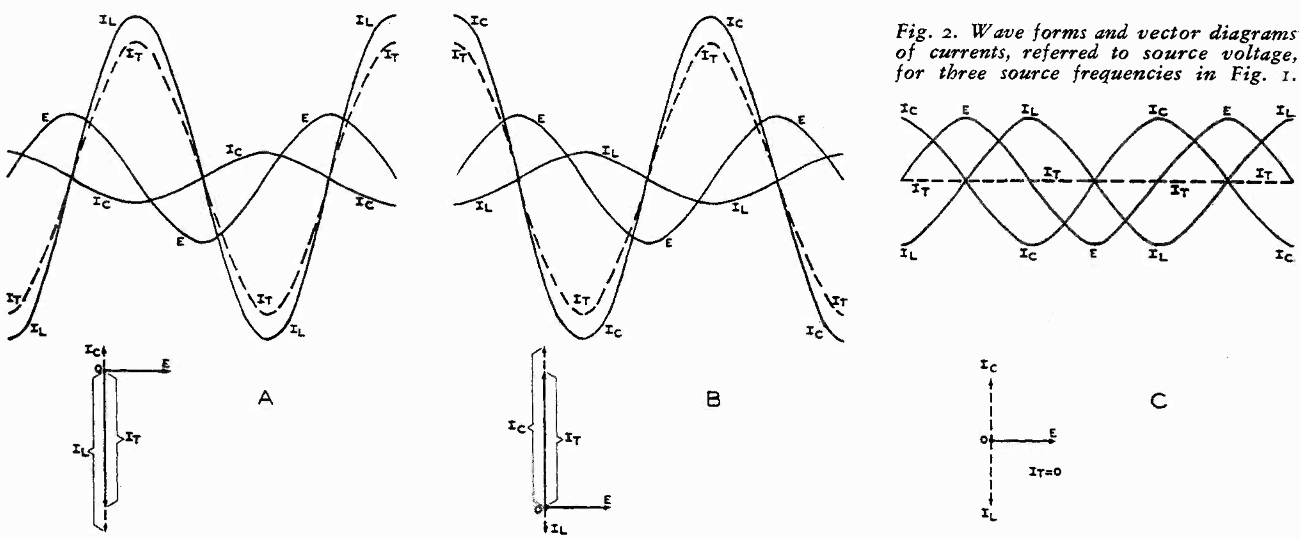


Fig. 2. Wave forms and vector diagrams of currents, referred to source voltage, for three source frequencies in Fig. 1.

always less than that which flows in the branch having least reactance, and, accordingly, the resultant impedance is always greater in magnitude than the branch having least reactance. As the source frequency changes, the magnitudes of the two reactances change in opposite directions. At some frequency the reactances become exactly equal in value; their equal and opposite currents cancel completely, so that no current at all flows in the source. Then the impedance of the combination is infinitely high. At frequencies close to this (the resonant frequency), the two branch currents almost cancel, so that the resultant current is still low, and the impedance very high. As the operating frequency is moved further from the resonant frequency in either direction, the resultant current gradually increases and the total impedance decreases.

For frequencies higher than the resonant frequency, the greater current flows in the capacitive branch; therefore, the resultant current is 90° capacitive, and so is the total impedance. For any frequency lower than the resonant frequency, on the other hand, the greater current flows in the inductance; then the resultant current and the impedance of the combination are 90° inductive. The circuit shifts abruptly and completely from inductive to capacitive (or *vice versa*) as the frequency of the source is varied through resonance.

We have spoken a bit loosely about the branch currents "canceling" one another. They do not actually cancel, although that is the resultant effect on the source. Each current must exist, because each branch represents a certain reactance directly across a voltage. A certain current must flow in that branch, and it is determined entirely by the reactance magnitude and the voltage; you can't cancel Ohm's Law. In fact, the amount of current "canceled" is current that exists in *both* branches simultaneously, flowing in opposite directions: circulating current, as explained for series resonance. The parallel combination is, in effect, a closed series circuit in itself. Circulating current is charge flowing from one plate of the capacitor, through the choke, and into the other capacitor plate, then back in the other direction to the first capacitor plate, then back again in a continuous oscillation (see Fig. 1). The time it takes the capacitor to charge, and the time it takes the choke's magnetic field to build up and collapse, determine the natural period of this oscillation and therefore the resonant frequency. If the source voltage is near this frequency, the circulating current is high in amplitude; when it occurs at the resonant frequency, the circulating current is at its maximum value. The voltage drop of this current at resonance through each branch reactance is equal to the source voltage, and is opposite to it in direction. Consequently, the impedance of the combination is infinite, and the source can force no external current through it.

It is often desirable to be able to calculate the impedance of a parallel *LC* circuit directly, without regard for individual currents and reactances. Such a formula can be evolved easily, as follows:

$$Z = \frac{E}{I_T} = \frac{E}{I_L - I_C}.$$

$$I_L = \frac{E}{X_L} = \frac{E}{2\pi f L}, \text{ and}$$

$$I_C = \frac{E}{X_C} = \frac{E}{\frac{1}{2\pi f C}} = 2\pi f C E.$$

Therefore,

$$Z = \frac{E}{\frac{E}{2\pi f L} - 2\pi f C E} = \frac{E(2\pi f L)}{E - 4\pi^2 f^2 L C E},$$

$$Z = \frac{2\pi f L}{1 - 4\pi^2 f^2 L C}.$$

This formula is valid for a parallel *LC* circuit having any values of *L* and *C*, and at any frequency. It is *not* valid when resistance is present in the circuit. Unfortunately, there is no simple way to obtain the resultant impedance graphically.

We have said that the resonant frequency is that at which the inductive and capacitive branches were equal in reactance. By setting the formulas for inductive and capacitive reactances equal to each other, and solving for f, we obtain the formula for resonant frequency f_o:

$$f_o = \frac{1}{2\pi\sqrt{LC}}.$$

This derivation was carried out in Chapter 13; it is indentical to the formula for series resonance. Again, it is strictly

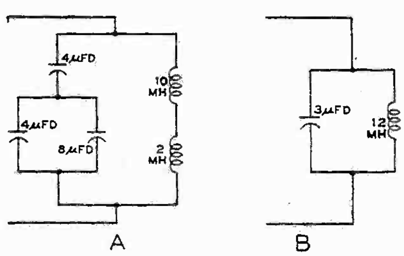


Fig. 3. Circuits A and B are equivalent.

accurate for parallel resonance only if no resistance is involved. We suggest that these two formulas be used to check the results obtained for Fig. 1, calculating resonant frequency and the impedance at various frequencies.

Evidently, both *L* and *C* affect the resonant frequency. Changing the value of either will shift f_o. It follows that, with a given value for one, any desired resonant frequency can be obtained by adjusting the value of the other element. It should be evident also that combinations of inductors and capacitors in any parallel branch should be combined according to the usual rules, and treated as lumped values in these calculations. For example, the circuit in Fig. 3A is exactly equivalent to that in Fig. 3B, so far as the remainder of the circuit is concerned.

In practice, there is always some resistance in the circuit—if nowhere else, in the inductor itself. It is impossible

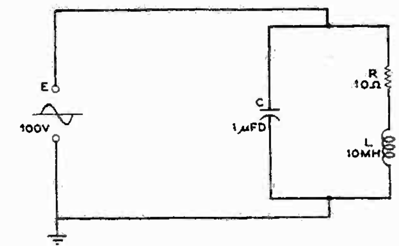


Fig. 4. Practical LC circuit has R in L branch, representing choke resistance.

to make a perfectly resistance-free inductor. The winding resistance of a choke is effectively in series with it, as shown in Fig. 4. This circuit will be recognized as that of Fig. 1, except for the addition of a 10-ohm inductor winding resistance.

Now the current in the inductive branch of the circuit contains a resistive component, so that it does not lag the source voltage by quite 90°. Therefore it is not precisely 180° out of phase with the capacitive-branch current, and the two currents do not "cancel" so effectively. At the former resonant frequency, 1,592 cps (Fig. 2C), the lower vector representing I_L has now rotated slightly counterclockwise, and has become a bit shorter because of the added current-limiting effect of *R* in this branch. The balance is upset; there is an excess of capacitive current. If we consider the resonant frequency of the circuit to be that at which the resultant reactance is zero, it is apparent that this new frequency must be one for which the capacitive current is reduced and the inductive current increased. Such a change will occur if the frequency is shifted downward slightly. The new resonant frequency, for effectively zero total reactance, is given by

$$f_o = \frac{1}{2\pi\sqrt{LC}}\sqrt{1 - \frac{CR^2}{L}}.$$

Thus, unlike the series resonance case, the precise resonant frequency is affected by the circuit resistance. This formula should be recognized as the resonance formula given previously, fol-

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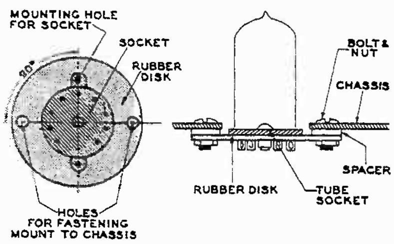
Shock Mounting Input Tubes

Even with today's input tubes, it is sometimes necessary to use a flexible mount in order to avoid microphonics. The following method combines ease of construction with good filtering properties.

1) Remove the socket and enlarge the mounting hole until its diameter is about ¼ to ½ in. larger than the diameter of the base of the tube or of the shield can (if one is used).



How to shock mount an input tube base.

2) Mount the tube socket on a circular piece of flexible rubber or foam rubber ⅛ to ¼ in. thick. The outer diameter should be about ½ in. larger than the diameter of the enlarged chassis hole.

3) Mount the rubber disc with the socket underneath the chassis, as indicated in Figs. 1 and 2. Use spacers to keep ¼ in. distance between the chassis and the rubber mount.

4) Reconnect the socket to the wiring, using flexible leads long enough to permit free movement of the socket.

If it is not possible to enlarge the original mounting hole, the tube may be mounted above the chassis using the same rubber mount.

Peter E. Beckmann
Muenster, Germany

Cabinetmaking Aid

Unless we're excellent cabinetmakers, the task of screwing and gluing perfect seams, even in the precision-cut kits now on the market, is a difficult one. Although a completed speaker cabinet may look fine on the outside after the finish is applied, there may be resonant points in the construction difficult to find without the aid of an audio sweep generator and test speaker.

A good alternative is to buy a quart can of the asphalt paint many photographers use to waterproof do-it-yourself sinks in their darkrooms. A liberal coat or (preferably) two of this thick-base paint applied to all inside seams will not only quiet any unwanted vibration, but it will also make the cabinet nearly 100% airtight.

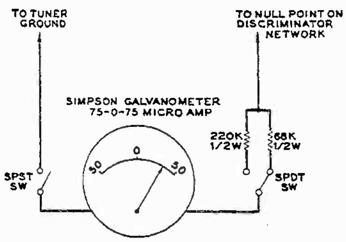
Alex Thien
Milwaukee, Wis.

Null Meter for FM Tuner

A null-indicating meter on an FM tuner, in addition to the signal-strength meter, is valuable for exact tuning of weak signals. Null meters are expensive, but they can be improvised. Here's how the one shown below was made.

A Simpson Galvanometer, 75-0-75 μa, center-indicating meter was mounted on a piece of plastic which was screwed to a wood base. A wire soldered to the null point of the discriminator network and led outside of the tuner chassis allows the meter to be clipped easily into the circuit.

Since exact voltages are unimportant, stock resistors were employed. A double-throw switch allows the choice of two sensitivities; the 68-K resistor provides a voltage reading of about 5 volts, and the 220-K resistor provides one of about



Home-made precision FM tuning meter.

18.5 volts. A switch in the negative lead puts the meter out of commission when desired. The high voltage is used to prevent burning out the meter in the first tuning.

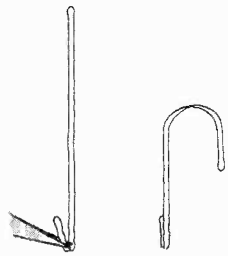
The material for this apparatus, with the exception of the meter, came from the junk box, and the cost was about one fourth the cost of a commercial meter. It has proved very satisfactory.

T. O. Mitman
Jim Thorpe, Pa.

Tone-Arm Finger Lift

There are a number of record players and changers on the market which lack a device to allow easy pickup of the tone arm from a rotating record; i.e., a finger lift.

This omission may easily be remedied with the help of a pair of needle-nose pliers and a woman's bobby pin. First, straighten out the bobby pin, and then bend about a ¼-inch length from one end back on itself, as shown below. This should be done with the thinnest part of the pliers at the point of the bend. The bend should then be squeezed to force the now-parallel parts



Bobby pin used for tone-arm finger lift.

of the bobby pin to touch each other. This forms the part of the fingerlift that fits on the head of the tone arm by friction.

About 1½ in. farther along the pin than the bend just completed, make a larger one, semicircular this time, with a radius of about ½ in. to fit the finger. The completed finger lift is shown at the right in the drawing.

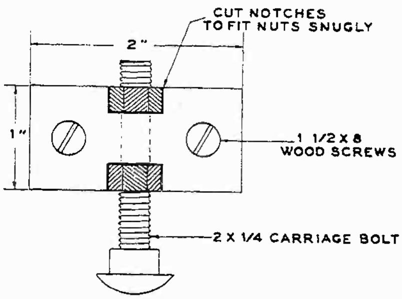
To install the lift, force the sharply bent part up onto the outer wall of the pickup head. Cut off any excess bobby pin remaining on the finger-lift end.

Jeffrey Frey
Ithaca, N. Y.

Leveling Equipment Cabinet

A very ingenious kit is available commercially for leveling turntables and changers or the small cabinets they may be mounted in. But, when putting equipment into a large cabinet with associated equipment, the need to have the cabinet level on a floor that is anything but level requires something heavier. The bill of materials for four heavy-duty leveling devices is as follows: four

¼ × 2-inch carriage bolts, eight nuts (square) for the same, eight 1½ × 8 steel flat-head wood screws, a piece of lumber or plywood ¾ × 8 × 1 in. Cut the lumber into 2-inch lengths. Cut notches in the two sides of the four pieces to fit the nuts, and drill a hole



Leveling device for large cabinets.

in the pieces from top to bottom to connect the notches. Drill and countersink holes on each side of the piece to fasten it to the bottom of the cabinet with the wood screws. Run the carriage bolt through the nuts, placed in the notches, and the cabinet may be leveled easily, using an open-end or adjustable wrench.

Edward T. Dell, Jr.
Roxbury, Mass.

Potentiometer Tapers

Potentiometers used for audio volume controls should be of the logarithmic-tapered type, showing a slow increase in resistance between the center-tap and ground terminals during the first part of the shaft rotation, and a progressively faster increase near the end of the rotation. This makes equal steps in shaft rotation correspond to equal changes in audio loudness since the human ear has a response that is roughly logarithmic.

Linear-taper potentiometers, common in surplus equipment or "junk-box" collections, can be easily converted to an approximation of the log taper by adding a resistor with a value of 1/4 or 1/5 that of the resistance of the pot between the center-tap and ground-end terminals. To find the taper of the potentiometer, measure its resistance between the center and right-hand (ground-end) terminals with the shaft pointing toward you. If the resistance with the shaft in the half-rotated position is about 1/2 that with the shaft all the way clockwise, the potentiometer is linear-tapered. If the resistance is much less than ½ the total resistance, the pot is probably log-tapered. If it is much more than 1/2, it is probably a counterclockwise log taper, and should only be used if you are willing to turn your gain up hind-end first.

The addition of the tapering resistor will lower the input resistance of the pot at maximum gain settings, so that the value of resistance to use in calculating coupling networks and stage gain is that of the parallel combination of pot and added resistor. The loading of the previous tube stage will not change the log-type taper drastically.

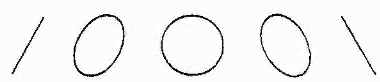
Other tapers can be produced by following similar principles. For instance, an oscilloscope centering control with a fast change of resistance at the ends of the shaft rotation and a slow change of resistance in the center can be made by adding a resistor between each end of the pot and the center terminal, the resistor values to be equal to each other and about 1/5 of the potentiometer resistance.

Donald L. Shirer
Columbus, Ohio

Hum Detection

In tracing hum through amplifiers, it is not always easy to tell its origin, but 60-cps hum and 120-cps hum have different possible causes. In general, 60-cps hum is caused by improper shielding or grounding, and 120-cps hum by poor filtering of the power supply.

An easy method of distinguishing the two is to view the hum on an oscillo-



Lissajous patterns for 60-cycle hum.

scope while the horizontal input is connected to a 60-cps sine-wave source. If the scope has no such source available, use the amplifier's heater leads.

Sixty-cps hum will resemble one of the patterns in Fig. 1, and 120-cps hum

120-cycle pattern shows 2-to-1 ratio.

will resemble one of those in Fig. 2. Once the frequency of the hum is known, steps can be taken to localize it.

Paul Penfield, Jr.
Brookline, Mass.

Amphenol-Connector Adapter

Amphenol microphone connectors, often used for electronic instruments and audio equipment, can easily be adapted to work with phono jacks by means of the five-minute procedure described below. An advantage of using this adapter is that it permits clip-lead connections to be made to the Amphenol-terminated cable.

The adapter requires an Amphenol chassis socket from which the nut, ground lug, and washers are removed; and a phono plug and a piece of tinned wire.

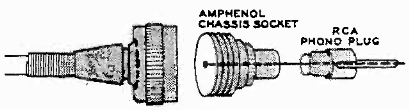
1) Tin the chassis end (the end opposite the eyelet) of the chassis socket inside and out to a depth of about ⅛ in. A 60-watt iron is adequate for the job.

2) Tin the back of the phono plug.

3) Tin a 2-inch piece of No. 20 wire. The diameter suggested is not critical since it is not to be subjected to mechanical stress.

4) Solder the wire to the eyelet of the Amphenol connector, leaving the excess to project through the chassis end of the connector.

5) Slide the phono plug over the wire so that the excess wire projects through the phono-plug pin. Bend this end of the wire to force the two parts firmly together. Insulation may be used,



Microphone plug to phono-plug adapter.

but it is not necessary if the wire is pulled fairly tight as the bend is made.

6) Solder the junction of the two connectors, wiping all the way around with the iron to give a neat and mechanically strong connection.

7) Solder the wire to the phono-plug pin and clip off the excess wire.

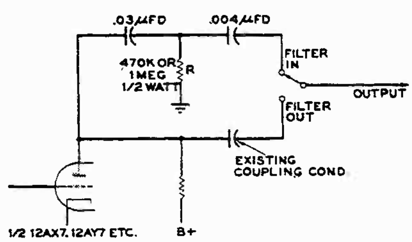
Theodore A. Finger
New York, N. Y.

Rumble Filter

If rumble in your hi-fi system is bothersome, try this simple filter. It can be installed in one evening, and the parts cost less than a dollar.

A single-pole, double-throw switch should be mounted at some convenient spot on the preamplifier control panel near the input stage. Mark one switch position IN, and the other OUT. Connect the parts as shown, keeping all leads as short as possible without running them near the filament wiring. For long runs, use low-loss shielded cable.

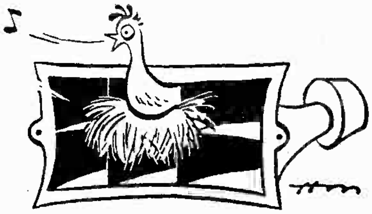
With the switch in the OUT position, the system will function as originally wired; but for exceptionally noisy rec-



Rumble filter can be cut in or out.

ords, placing the switch in the IN position will connect the filter and stop most rumble. Use whichever value of resistance for R that gives most satisfactory results.

L. E. Johnston
Madison, Wis.



Sound-Fanciers' Guide

by R. D. DARRELL

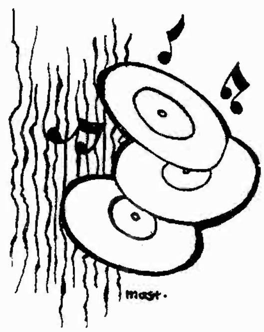
LOOKING back over recent columns in this series, I find that in rapid succession I've rioted in sensational sonic displays, reveled in purely aesthetic delights, sought some middle ground between these extremes, and even questioned the suitability of wide-range technology for certain types of introspective music. I wonder why no one has yet challenged me with the same question that greeted a British-born friend of mine on his arrival in this country. He's never forgotten his first encounter, as a boy of only ten or so, with the mysteries of the American language and temperament. A tough New York youngster stopped him with a blunt "Hey, you!", puzzled over his British school cap, and insisted on knowing, "What team're ya on, buddy?"

I might well be as tongue-tied as he was if I were asked to identify my own allegiances, for to say only that I'm eclectic in my tastes is no satisfactory answer. Probably there isn't a rational way to explain the vagaries of anyone's aural appetites, but if there were, I strongly suspect that it would involve such key factors as *time*, *place*, and *purpose*. I know for certain that my own attitude toward high fidelity at any given moment depends primarily on what I'm listening to and how much attention I'm willing to give it; and secondarily, but significantly, on where I am, who, if anyone, is listening with me, and whether what I've been hearing before has left me in a mood for more of the same or something entirely different.

Extravagant sound concoctions played at high levels may repel me, for example, when heard at hi-fi shows or whenever I'm too distracted or weary to concentrate on them. . . when I crave more substantial musical fare. . . or when I've heard them too often before. On the other hand, when I've been listening for long periods to exclusively "serious" compositions, it's sheer joy (relaxing and stimulating at once) to go uninhibitedly on a kind of sonic binge. This month, for instance, I've been busy for days getting caught up on a long series of recorded tapes, nearly all of them of more or less serious content. And good as most of them are technically, they left me with the realization that as yet the really off-beat sound effects have been largely ignored on tapes — and that a bit of the latter was just what I needed to jolt me back into a state of keen excitement. It's easy enough to poke fun at such sonic *divertissements*, but what most audio purists forget is that they are likely to be harmful only when indulged in too exclusively or extensively. Their best function is to serve as an occasional potent tonic, or aural cocktail, to ginger up jaded listening appetites.

The Ear on the Barnyard Floor

Of course, this is all postrationalization. What I actually did was probably entirely instinctive: shut off the tape deck, switch over to phono, and look around for the zaniest disc oddities I could find. Luckily a good supply had accumulated, topped by one of the screwiest examples to date: the irresponsible E. D. Nunn's second psychiatrists' special, entitled (and the Saukville King of the Transients isn't kidding!) *Adventures in Cacophony* (Audiophile AP 37). Some of its B-side episodes are either a bit anticlimactic or needlessly embellished, but the tugboat whistles and bells are ultrarealistic, while the clock and watch tickings, if continued any longer, would serve as a modern equivalent of the water-drop torture.



Side A is the truly cacophonous one, starring the biggest, toughest, *basso* tomcat that anyone is ever likely to encounter in real life (I hope!), and not neglecting chickens, roosters, pigs, and cows, or even a no less agonizedly squalling human (it says here) infant. The close-to recording is just too good; it sent the local livestock into a frenzy. Any replaying will certainly have to be confined to the winter months; otherwise, with windows and doors open, and my horn-loaded speakers in full roar, the whole neighborhood would be permanently disrupted and every animal within miles thrown into nervous breakdowns by their canned virtuoso colleagues.

But it was great fun for a few minutes, which is more than I can say (as a nonracing sports-car owner) of *Sounds of the Annual International Sports Car* Grand Prix *of Watkins Glen, N. Y.* (Folkways FPX 140). Here the technical interviews with various drivers are, no doubt, first-rate shop talk, the warming-up and racing noises are realistic enough to make one duck every time another car roars by, and the illustrated accompanying notes are highly informative. But one mechanical bumblebee buzzes by much like another, and the picked-up echo effects of multiple PA systems soon become intolerable. I can commend this only to insatiable racing *aficionados*, and for some kind of audio *Grand Prix* for on- as well as off-the-beaten-track documentary material.

Not-so-Steely BWI Bands

After these two shots of straight sonic alcohol, I was stimulated to atone at last for my long neglect of Emory Cook's beloved Steel Bands (made up largely of sawed-off and tuned-up 55-gallon oil tins), which have evolved in recent years in the British West Indies. I suppose most other sound fanciers have long since made the acquaintance of the original *Brute Force Steel Bands* (Cook 1042) and *Steel Band Clash* (Cook 1040), but my belated first encounter with the latter was a surprise, for of course the actual sounds aren't steely or clashing or brutish at all — they're dulcetly enchanting! Catchy as the Latin-American dance materials are, with their incessant but discreetly used maracas and "scratcher" obbligatos, I found these topped by the bubbly marimbalike sound webs woven over such familiar tunes as *In the Mood* and *Over the Waves*.

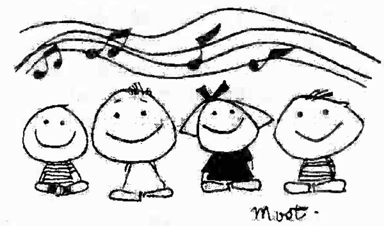
In a more recent batch of releases, the Brute Force band is heard again in *Music to Awaken the Ballroom Beast* (Cook 1048). No less than six other Champion Steel Bands of Trinidad are featured in Cook 1046, illustrating the variety of expressive techniques commanded by individual exponents of this singular medium. Of these six, the one that impressed me most is the Katzenjammers (better subtitled, the "Steel Band with Velvet Gloves"); I recommend most highly of all the disc (Cook 1047) on which it is featured exclusively. For here are not only the same three pieces included in 1046, but some eight others, all as poetically atmospheric, and most of them starring the strange but irresistible "humming-bird" tenor-pan obbligatos of Percy Thomas — one of the most extraordinary musical virtuosos I've ever heard. In case you've been as slow as I have in making this discovery, just listen to *Love is a Many-Splendored Thing, Tiana, Wonderful Illusion*, or *Port of Spain* for something truly new and incomparably satisfying in both sonic and musical experiences.

Anywhere Out of Ulster Co.!

After these moonlit British West Indies evenings and sonorities (complete with nonscheduled indigenous crickets), I had no hankering to return to snow- and ice-bound Ulster County. Physically, I couldn't escape it as I stomped and slid down the lane to my R.D. mailbox, but happily the latter's contents included further tonal passports to warmer or stranger climes. I couldn't wait to unpack two new releases in the Audio Fidelity Mexican series and set them successively spinning, still dripping melted snow, on my turntable. Miguel Dias's Mariachi Orchestra in *Fiesta en Mexico* (AFLP 1816) isn't quite as aurally startling as the Katzenjammers' steel drums, but it's novel enough for its piquant contrasts between energetically sawed or scrabbly strings and brazenly biting cornets (or trumpets?) over steady guitar ostinatos. The high-level recording, too, makes the most of the incisive brass attacks and brisk string pizzicatos (some amazingly steep wave fronts here, surely), and, except for perhaps a little too much singing, I found the light but genuinely folkish music infectiously zestful. It was particularly good to hear such more familiar South-of-the-Border tunes as *El Rancho Grande* and *Cielito Lindo* done, for once, with undiluted stylistic authenticity.

Even more exciting sonically is the *Plaza de Toros*, Vol. 2 of *La Fiesta Brava* (Brave Bulls) series (AFLP 1817). Unlike most sequels, this outdoes the original release (AFLP 1801, reviewed here last October) in ultra-sensational technical brilliance — and also in the marked superiority of its accompanying full-color bullfight picture reproductions. It's hard to resist the contagious lift of the characteristically festive music itself, and quite impossible for any true audiophile to resist the impressive sizzle and solidity with which the Banda Taurina's cymbals and big bass drum emerge here.

After that, what fresh fields remain? There are some if you seek far enough, as I soon discovered when I ventured first into the wild mountains of Bulgaria and then through the whole diversified European folksong repertories. Strange indeed to American ears is the program of exotic and often quite barbaric songs and dances by the Bulgarian Republic Ensemble under Philippe Koutev (Angel 65026), but both performances and music are admirably varied in mood and recorded with the utmost in bold, reverberant effectiveness. On the other hand, the Roger Wagner Chorale's *Folk Songs of the Old World* (Capitol PBR 8345, two 12-inch) exemplifies high fidelity's quieter lyricism. There are no real highs or lows in actual evidence here, yet only extremely wide-range recording equipment ever could have captured these superb voices with such freedom from any hint of distortion. My only complaint is that too many of these lovely melodies have



been gilded by overly sophisticated arrangers or misunderstood by the skilled conductor himself. Even so, at their best, both songs and singing are a caressing balm to one's ears.

Jazz Subtleties; Wurlitzer Cliches

My last three disc *divertissements* this month didn't give me as much pleasure as they should by reason of technological excellences alone, but that may be more my fault than theirs. As a matter of fact, I did get quite a kick from George Wettling's *High Fidelity Rhythms* (Weathers W 5501), for after a rather slow start (and the handicap of the purplest annotation blurbs yet), its relaxed yet intricate performances occasionally burst into real flame in the leader's virtuoso drumming, heard alone in *It Ain't the Humidity*, and alternating with Milt Hinton's imaginative bass bits or Dave Bowman's piano playing, in *Is George Really George?* and *Kettle Blues*. The trouble is that, however crisp and clean this recording may be, I've become so entranced by the advantages stereo sound holds for such small, subtly skilled jazz ensembles that I can no longer relish them with complete satisfaction in single-channel recording, no matter how good. If you have access to stereo equipment, just listen to such tapes as those by Sam Price and the McPartlands (Concert Hall CHT/BN 16 and CHJT/BN 13), Wilbur de Paris (Atlantic AT 7-9 BN), Vic Dickenson with Ruby Braff (A-V 707), and even the less *echt*-jazzy Modernes (Concertapes 508) — and I'm sure you'll understand why.

Hating to reopen old wounds, I hesitate to specify my reactions to Reginald Foort's *Waltz and Ballet* on the Richmond Mosque Wurlitzer (Cook 1058) and Leon Berry's pop and novelty program on the Chicago Hub Rink ditto (Audio Fidelity AFLP 1828). Let me only whisper that, while the former instrument sounded more like an honest pipe organ, it's a poor excuse for an orchestra in materials like the *Coppelia* Ballet, *Nutcracker* Suite, *Rosenkavalier* Waltzes, etc., and that the celebrated Briton's playing seemed as hurried and slap-dash as if he wanted to get it over and done with as badly as I did. As for Berry's monstrously jingled-up "beast", I can only say that in such mercilessly close and brilliant recording as it's given here, it achieves an almost literally ear-splitting stridency such as I've never heard before or ever want to hear again from any so-called musical instrument! I'd admit that this was a record to end all Mighty Wurlitzer records if *Leibert Takes Richmond* (Sonotape SWB 8006) hadn't proved that new dimensions of theater-organ tonalities now exist in stereo — which, incidentally, makes them considerably easier on my nerves, if no more palatable to my taste. (All right, write and tell me I'm prejudiced! I certainly won't deny it, or even claim that I'm ever likely to become less intolerant in this particular domain.)

Tape Reconsiderations

That isn't to say, however, that I *never* change my mind. I often do with certain records that become more listenable, or gradually lose their appeal, upon repeated rehearings. Often, too, I hesitate to accept an immediate unfavorable verdict. This month I've gone back to a number of tapes set aside some time ago for further consideration and, in at least one case, I found my first impressions entirely wrong. Ingrid Haebler's Mozart Piano Concertos, K. 450 and K. 456 (Phonotapes-Sonore PM 129), originally sounded as if something had flawed the piano tone, but I tried it again, this time immediately after demagnetizing my playback head and thoroughly cleaning it and the tape-transport guides.* The trouble vanished and the piano tone emerged perfectly clear and round, if rather on the small side, as it was in the original LP (Vox

*While it isn't my proper function to evaluate equipment and accessories, especially since I seldom attempt to test them objectively, it may be only fair to report (on the basis of subjective observations only) that I have been pleased with the effectiveness of the Audio Devices Head Demagnetizer and the EMC "Long Life" dual cleaner and lubricant fluids I have been using; and more than satisfied by the consistently dependable operation (for nearly a year now) of the Viking Model FF75SU Tape Deck, at least as used with my own home-built preamplifiers.

PL 8300). Of course, nothing could be done about the thinnish string tone of the orchestra itself, so this remains a rather lightweight edition of two superb concertos, charming in many ways, but in no wise comparable interpretatively with the never-to-be-forgotten Elly Ney K. 450 78's.

On the other hand, several older tapes, on which I've withheld unfavorable reports, proved on rehearing to be just the routine performances and undistinguished recordings they seemed at first acquaintance: the Schubert C major Symphony by Schwertfeger (Berkshire B 2105), *Scheherazade* by Haffner (Omegatape OT 3001), and *Caucasian Sketches* and Tchaikovsky String Serenade by Schuster and Haffner (Omegatape OT 3004). Far better indices to these companies' current technical standards are provided by their latest samplers: Berkshire *Highlights*, Vol. 2 (H 2, 5-inch, $1.50) and *Music for High-Fidelity Shows* (Omegatape D 8, 7-inch, $5.75), each of which provides a wide variety of repertory, although in unsatisfactorily short snippets and with vocal announcements and sales pitches which I, at least, could gladly dispense with.

Comme Ci, Comme Ca

Another batch of tapes is of generally satisfactory technical quality (naturally varying considerably, sometimes according to the age of the original recordings), but represents music and/or performances which leave me cold or lukewarm at best. But I certainly don't suggest that you take my highly subjective reactions to these seriously; if you liked the works in their disc versions, you'll certainly enjoy the tapings of Irving's Tchaikovsky *Swan Lake* on RCA Victor DC 12 (Bluebird LBC 1064); Perlea's Tchaikovsky program on Phonotapes-Sonore PM 112 (Vox PL 8700); and a series of Sonotapes featuring Scherchen's Beethoven Overtures on SW 1015 (from Westminster WL 5177 and 5335), Beethoven Fifth Symphony on SW 1001 (from WL 5406), and the same conductor's extremely mannered Liszt *Hungarian* Rhapsodies on SW 1014 (from W-LAB 7003 and 7007); the Barylli-Vienna Beethoven Septet on SW 3007 (WXN 18003); Badura-Skoda's *Moonlight* and *Pathétique* Sonatas on SW 1009 (WL 5184) and — with others — Schubert *Trout* Quintet on SW 3008 (WL 5025); and Farnadi's piano-solo Liszt *Hungarian* Rhapsodies on SW 1030 from WAL 213).

Scherchen is far better represented, to my mind, by his highly vivacious Beethoven Fourth Symphony on Sonotape SW 1039 (from Westminster WL 5406) and his truly chamber-music-scaled Bach *Brandenburg* Concertos on Omegatape OT 8006 and 3010 (no American LP's); while young Badura-Skoda demonstrates his finest qualities in the Schubert Op. 99 Trio, with Fornier and Janigro, on Sonotape SW 1028 (Westminster WL 5188 of 1953 — but still an outstanding example of well balanced and sparkling recording). Another, even older (1951), but also superb chamber-music recording, notable for its rich string sonority as well as exceptionally fine interpretative insight, is the Stuyvesants' Ravel and Debussy Quartets on Phonotapes-Sonore PM 103 (Philharmonia PH 104).

A group of Rodzinski Sonotapes may leave me a bit unmoved interpretatively, but I have to admire the masterly precision of his performances, and I relish immensely the lucid clarity with which he has been recorded in the Bizet *Arlésienne* and *Carmen* Suites on SW 1033 and SW 1018 respectively (W-LAB 7006 and 7005); Franck *Chasseur maudit* on SW 1020 (WL 5311); Borodin *Polovstian* Dances and Ippolitov-Ivanov *Caucasian Sketches* on SW 1029 (W-LAB 7039). Yet even such sonic relish pales beside the profound delights I find in the far less technically distinguished Cluytens Berlioz *L'Enfance du Christ* on Phonotapes-Sonore PM 124, two 7-inch (Vox PL 7122), for here, although the 1951 recording still is satisfactory enough in itself, the supreme attractions are the lovely music heard in an ideally communicative projection.

Finally, there is a special batch of tapes in which I can't find — or want to look for — anything to criticize, however mildly, in either performance or recording: Friskin's Bach Recital on A-V 2002 (from Vanguard BG 543-5); the Wagner Chorale's Gregorian Chants on Omegatape OT 8003; Novaes's Falla *Nights in the Gardens of Spain* on Phonotapes-Sonore PM 5006 (from Vox PL 8520); Scherchen's early triumph with Prokofiev's *Lt. Kije* Suite on Sonotape SW 3005 (Westminster WL 5091); and the same unpredictable genius's latest masterpiece, the remade version of Haydn's *Military* Symphony on Sonotape SW 1042 (Westminster W-LAB 7024). Incidentally, these last two works provide — if you can afford both the tape and disc editions — unequalled opportunities for making direct A-B comparisons between your system facilities for the two mediums. With both reel and disc spinning simultaneously (if not for long in perfect synchronization), I've switched repeatedly from one to the other without being able to detect the slightest change in quality even at the frequency-spectrum extremes. If you can't do the same, I'll be willing to bet the reason lies not in the recordings themselves, but in your own pickup or playback head, or in their associated preamps — most probably in not quite properly adjusted equalization circuits.

HI FI IN YOUR CAR

Continued from page 23

at these frequencies. Several sharp null points are present, though, in this antenna's pattern. They are not a problem, because it is difficult to drive the car straight down a null line. Fig. 4 shows details of an experimental antenna built along these lines. Since it is supported

Robert L. Perry

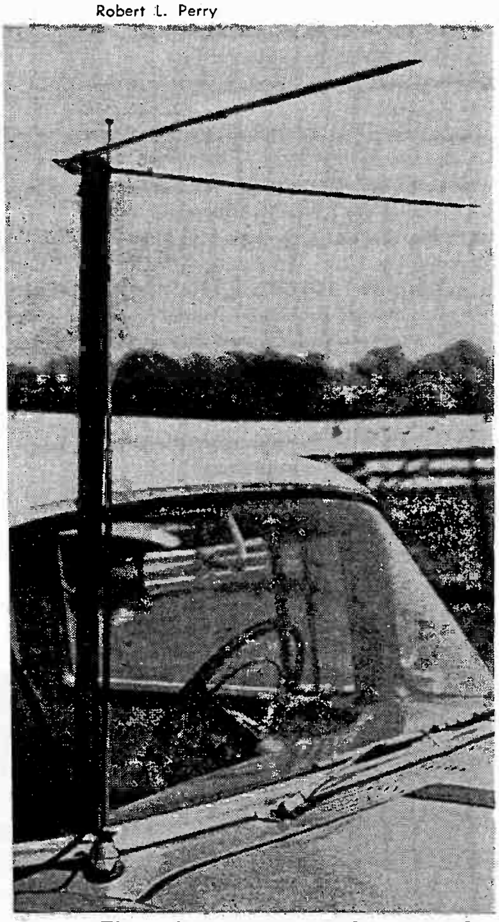


Fig. 4. FM antenna taped to AM whip.

entirely by the AM whip, it isn't necessary to drill holes in the car body. If you want a neater and sturdier job, you may want to use Fig. 5 as an example. In any case, each quarter-wave element should be individually insulated and connected electrically only to the cable shield or inner conductor.

In metropolitan areas your auto-radio whip antenna may be used by telescoping it to ¼ wave length. The range, unfortunately, will be greatly reduced. FM broadcast transmitters radiate horizontally polarized waves and an antenna in a plane within about 10° of vertical will be quite inefficient.

Other Modifications

Fig. 1 shows the tuner used with an arbitrary auto-radio circuit. The exact connections will depend upon the year and make of the AM radio with which

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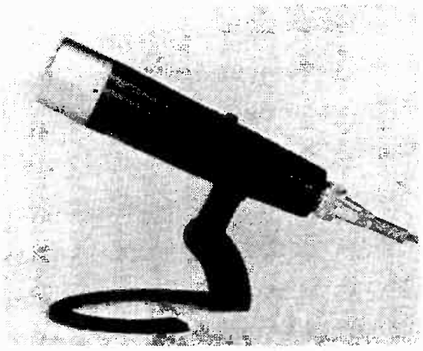

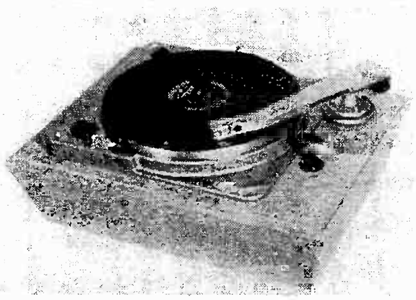

HI FI IN YOUR CAR

Continued from preceding page

the tuner is used. Connection points are fairly well standardized, however, so there should be relatively little trouble. It would be helpful to get a circuit diagram of your auto radio from a parts supplier, and correlate Fig. 1 with your own circuit.

A better but more expensive procedure would be to build up a small

Robert L. Perry

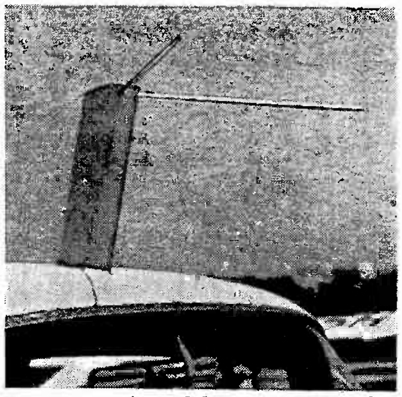

Fig. 5. Roof model is neater, sturdier.

amplifier and vibrator power supply to drive a hi-fi speaker. In most cases, 6 watts at 400 cps will be adequate.

Still another option, which we chose, is to modify the car-radio amplifier. Ours was on a separate chassis with the power supply. By removing the split-inductor phase inverter, and replacing it with a 12AU7 inverter-driver, replacing the output transformer, and adding some 20 db of inverse feedback over the entire audio amplifier, we achieved a fabulous 1 watt at 30 cps. In practice, the sound from our converted system with an Electro-Voice SP8B speaker is essentially flat from 40 cps to 12 Kc. This sounds terrific by contrast with the average AM program over the car radio.

AMPLIFIER DESIGN

Continued from page 21

achieves a high output from a pentode without the same loss of gain that would be necessary when cathode coupling. Because of the extremely tight coupling between cathode and plate-screen circuits, it is possible to operate this amplifier into the grid-current region and still maintain a low order of distortion.

The partially cathode-coupled ultra-linear circuit is another good variant, but requires careful attention to transformer design.

The Circlotron obviates the necessity for highly critical design of the output transformer. What could prove to be a further liability in this arrangement is the fact that the whole of the two B+ supplies constitute part of the output circuit. This means they can contribute to the frequency response or stability characteristics of the entire amplifier.

ARISTOCRAT ENCLOSURE

Continued from page 19

meaningless to assess an enclosure except as concerns bass response — and, even then, the bass-driver characteristics have a great effect. In view of the fact that kit builders of the Aristocrat might have on hand other 12-inch drivers than those of Electro-Voice, we tried out some others too. They were basically similar in price and cone resonance frequency to E-V's 12-inch units recommended for use in this enclosure. All (including the E-V drivers) fell in price between $25 and $75; all had resonance frequencies between 40 and 50 cps.

There were the expected differences in transient response and efficiency, of course, according to magnetic structure and (generally) price. But the bottom response limit correlated only very roughly with cone resonance frequency. On the average, fundamental acoustic loading was maintained well down to about 37 cps, with a quite gradual increase in efficiency up to about 48 cps. There was a noticeable response dip centered around 60 cps, ideally placed

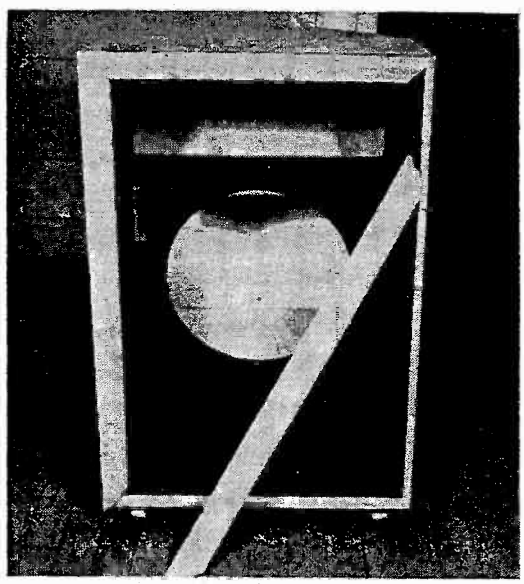

Fig. 8. Frame molding goes on in four separate pieces, but has the appearance of a single unit. Frame is removable.

to subdue hum and audible turntable noises! Any other irregularities in the bass range were not easily detected even by trained listeners. They were minimized experimentally by the addition of a 2-by-2-inch stiffening brace inside the top panel, and another diagonally across the bottom part of the speaker

Continued on page 40

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By SIR DONALD FRANCIS TOVEY. Written by one of the most literate and profound of modern interpreters of music — its theory and history, these essays are all drawn from those articles on music which Tovey prepared for the *Encyclopaedia Britannica.* There are in all twenty-eight articles on different kinds and elements of music — aria, chamber music, concerto, harmony, mass, melody, opera, programme music, sonata, symphony, and others. Paper, $1.35 **239**

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ARISTOCRAT ENCLOSURE

Continued from page 38

mounting board. Generally, bass was full-bodied and, in our test room, well balanced with the back panels about 2 in. from the corner walls; the upper bass could be increased by greater spacing.

Such deep bass is really remarkable when the size of this enclosure is considered: it is less than 30 in. high, 19 in. wide, and about 17 in. deep — well under 4 cu. ft. inside. We should say, on the basis of our experience with the KD-6, that the average builder will find his kit fully as effective as the factory-built Aristocrat.

TRANSISTORS

Continued from page 25

harm from temperature or interchanging transistors, the stability factor *S* must be low, or the DC load line steep, or (preferably) both. Choose a typical bias circuit that, in your estimation, can put the transistor at the desired quiescent point, and calculate the values of resistance necessary. Often several sets of resistances will be possible, and then a choice can be made between the several to get a steep DC load line (small resistance in the collector and emitter leads) and/or a small value for *S*. In general, a low value of *S* is achieved by making the emitter work into a high-resistance external circuit, and the base into a low resistance.

3) Once the proposed bias scheme is designed, the battery power can be calculated. If this is too high, it may be possible to redesign the circuit to get equal or better stability with less battery current.

4) Finally, note that the values of load resistance and the resistors in the base current-dividing circuit should be large, so that not much AC power will be lost in them. Remember that bias circuits are no good if the resistors are of such value that the stage won't amplify.

Designing a bias circuit is, for the beginner, a process of trial and error. With a little experience, however, he will begin to discover shortcuts too subtle and numerous to mention here, and he will find himself more at ease with the circuits. He will develop a feeling for the circuits that will help him select the best one instantly.

This installment should have impressed the reader with some of the important things to look out for in designing a bias circuit, with a few typical circuits to work from. In future installments, the practical amplifiers discussed will of course be biased, and from these diagrams the reader can design more to suit himself.

Any circuit will do to bias a transistor so long as the base and collector currents are proper. With a p-n-p transistor, current normally flows into the transistor by way of the emitter (with the arrow) and *out* of the transistor by way of both the base and the collector. (Conventional plus-to-minus current is assumed in this series.) With an n-p-n transistor, current normally flows into the collector and base, and flows *out of* (with the arrow) the emitter. Therefore, with a p-n-p transistor, the collector and base go to the negative battery terminal, and with an n-p-n transistor they go to the positive terminal.

Next month we will discuss parameters and small-signal equivalent circuits.

Further Reading

Graphical Analysis and Load Lines

Any good book on vacuum-tube amplifiers, for instance:

Crowhurst, N. H. "Designing Your Own Amplifier." AUDIOCRAFT, I (Mar. 1956), pp. 20-23, 43; I (Apr. 1956), pp. 25-27, 41-42; I (May 1956), pp. 21-23, 40-41.

Langford-Smith, F. *Radiotron Designer's Handbook*. Harrison, N. J.: Radio Corporation of America, 1953, pp. 481-517.

Crowhurst, N. H. "More About Load Lines." *Radio-Electronics*, XXVII (Apr. 1956), pp. 40-42.

Crowhust, N. H. "What is a Load Line?" *Radio-Electronics*, XXVI (June 1955), p. 32.

Transistor Load Lines and Biasing

Krugman, L. M. *Fundamentals of Transistors*. New York: Rider, 1954, pp. 71-82.

Lo, A. W., Endres, R. O., Zawels, J., Waldhauer, F. D., and Cheng, C. C. *Transistor Electronics*. Englewood Cliffs, N. J.: Prentice-Hall, 1955, pp. 131-137.

Turner, R. P. *Transistors, Theory and Practice*. New York: Gernsback Publications, 1954, pp. 49-51, 57-58.

Bias Stabilization

Bevitt, W. D. *Transistors Handbook*. Englewood Cliffs, N. J.: Prentice-Hall, 1956, p. 178.

Lin, H. C. and Barco, A. A. "Temperature Effects in Circuits Using Junction Transistors." *Transistors I*. Princeton, N. J.: RCA Laboratories, 1956, pp. 369-402.

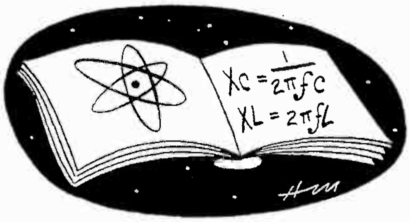
Lo, A. W., Endres, R. O., Zawels, J., Waldhauer, F. D., and Cheng, C. C. *Transistor Electronics*. Englewood Cliffs, N. J.: Prentice-Hall, 1955, pp. 137-154.

Penfield, P., Jr. "Transistor Bias Stabilization." *Audio*, XL (May 1956), pp. 34, 36, 38, 40-41; XL (Jul. 1956), pp. 20-21, 40-44.

Shea, R. F. *Principles of Transistor Circuits*. New York: Wiley, 1953, pp. 97-131.

Shea, R. F. *Transistor Audio Amplifiers*. New York: Wiley, 1955, pp. 96-99.

Small, R. H. "Stabilizing Transistor Amplifiers." *Audio*, XL (Dec. 1956), pp. 47, 76-78.



BOOK REVIEWS

Continued from page 14

explained. Multispeaker-design information is included, along with full information on crossover networks for such speakers, and charts and graphs that permit the hobbyist to build his own networks.

The second section of the book analyzes types of loudspeaker baffles and enclosures from the simplest flat baffle to the most complex folded-horn enclosures. Characteristics of all types are fully covered, and constructional information in the form of graphs and charts is included; with these data the hi-fi

Continued on next page

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Sound Sales Directory

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

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HAymarket 1-6800 — 1-11

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BEverly 8-1067 — 1-11

Voice and Vision, Inc.
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DAvis 8-8822 — 1-11

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BOOK REVIEWS

Continued from preceding page

enthusiast or technician can make his own enclosure or evaluate ready-built or kit enclosures.

In the last section of the book, the listening room is considered as part of the acoustic circuit. Information on enclosure placement and room treatment for best hi-fi reproduction is furnished. The appendix contains 18 complete plans for the construction of typical loudspeaker enclosures of all types and for various speaker sizes.

Material in this book has been compiled from information provided by several of the major loudspeaker manufacturers. The author is a musician and engineer who has been intimately concerned with the fields of acoustics and loudspeaker-enclosure design for many years. His book can be heartily recommended to all who are interested in a deeper understanding of the most highly variable link in all sound systems.

How to Make Good Tape Recordings

C. J. LeBel; pub. by Audio Devices, Inc., New York; 151 pages; price $2.50, cloth cover; $1.50, paper cover.

In this small handbook the author has packed a good deal of basic nontechnical information on tape recording. It is of most value to the beginning home recordist, since it explains in the very simplest terms the nature of sound, how a tape recorder works, the various kinds of tape available, and common faults of tape recordings.

Three of the chapters have been contributed by other specialists in their fields: "Microphone Recording", by Vincent J. Leibler of Columbia Records; "Tape Editing", by A. A. Pulley of the RCA Victor Record Department; and "Use of Sound Effects", by Herman H. Haverkamp of Radio Station WNYC, New York.

The slick-paper pages are well sprinkled with pictures and diagrams. A glossary of tape-recording terms in the back of the book explains very well a good many common terms which the recordist is likely to encounter and with which he should be familiar.

Mr. LeBel is a well known authority in the tape and audio fields. He is vice president of Audio Devices (makers of Audiotape), and secretary of the Audio

Engineering Society, having been its first president.

This book, it might be mentioned, is a companion volume to *How to Make Good Recordings*, which was directed toward disc recordists.

BASIC ELECTRONICS

Continued from page 31

lowed by a correction factor. For the circuit in Fig. 4 the correction factor is 0.995, and the resonant frequency is 1,584 cps: a downward shift of 8 cps, or 0.5%.

This subject will be continued (and concluded) in the second part of Chapter 15, which will appear in the next issue.

PORTABLE HI FI

Continued from page 29

the lid closed were 46 × 40 × 29 in.

The next problem, and one of no mean importance, was designing the speaker enclosure. It had been my previous experience that if all spurious cabinet vibrations were eliminated, if sufficient volume and damping material were provided, and if proper tuning to the specific speaker were accomplished, then the bass-reflex enclosure was not only desirable but pretty hard to beat. My problem was how to use the bass reflex and still have the thing go through a door. By using a 12-inch speaker, to help hold down the necessary internal volume, I found that it could be done. Another consideration in the selection of the speaker was its power-handling capacity. Since the amplifier was rated at 30 watts, the speaker had to be able to handle at least that to preclude separation of the cone from its suspension by an inexperienced operator. The cabinet material was ¾-inch birch plywood with 2 × 2 and 2 × 4 framing, and all joints were glued and screwed. All large areas of the plywood were broken up by 2 × 2 bracing strips installed at erratic angles. Using an audio oscillator, the port was tuned to match the speaker.

I found that one of the easiest places to make a mistake that could seriously affect the fidelity of the whole system was in setting up the switching circuits between units. In a panel on the slanted front of the cabinet, between the Ampex recording amplifier and the Fisher FM-AM tuner (Fig. 3), there are mounted a double-pole double-throw switch, a standard phone-jack mike receptacle as used on most high-impedance mikes, a Cannon mike receptacle, and a Jones plug power takeoff for an external mixer. The points where mistakes are easy to make are in the plug connections to the double-pole double-throw switch (from mike to recorder through the switch, and from the control unit to the recorder through the switch). The connections must be correctly wired, Fig. 4, since the input lines from the mike and mixer are low-impedance two-con-

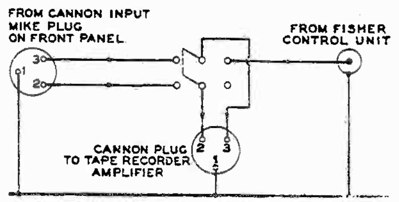


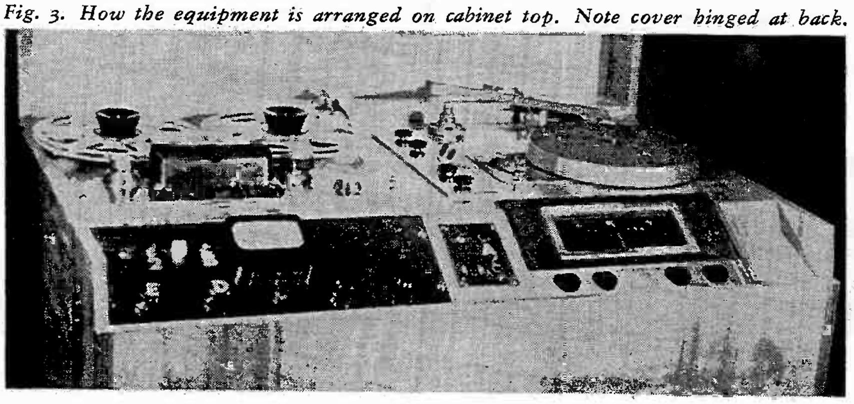
Fig. 4. Wiring of recorder input switch.

ductor shielded cables, and the input line from the Fisher control unit is a cathode-follower single shielded cable.

It is necessary for the input switch on the recording amplifier to be set at the unbalanced bridge position when the recorded signal is coming from the control unit; this may be fed by the record player, FM-AM tuner, or a high-impedance mike. The input switch on the Fisher control unit must be set to coincide with the input desired: the tape recorder, high-impedance mike, FM-AM tuner, or the phono pickup, all of which go through the control unit. Because the tape-output jack on the Fisher control unit is connected before the volume control, it is possible to record a radio program while it is being listened to, or, by turning the gain down on the control unit, it can be recorded without audible monitoring. This is true also for phonograph records and for high-im-

Continued on next page

Fig. 3. How the equipment is arranged on cabinet top. Note cover hinged at back.



PORTABLE HI FI

Continued from preceding page

pedance mikes. The input switch on the recording amplifier must be turned to the microphone position to provide for the low-impedance line from the mixer or from a low-impedance mike.

Since this machine was intended for general use, and not just for one trained operator, it was necessary to make up a list of operating instructions. These instructions were pretty much cookbook style, and they necessitated labeling all the important controls of the machine. This was done by using large white decals. Both letters and numerals were used because, with the large number of controls, we wanted to avoid as much confusion as possible. The problem of allowing inexperienced persons to operate the unit did not prove a serious one, and routine maintenance keeps everything in top operating condition.

There is one problem that cannot be solved with a slide rule in the realization of such projects as this. Schools are financed with public funds, which makes it necessary to justify each expenditure in terms of educational advantage to the pupil. This is as it should be. In case some readers may want to get such a machine authorized for use in their local schools, I will list a few of the benefits that may be derived.

First, it can be a motivator to stimulate creative thought. These creative thoughts may be put in written form, such as scripts. This brings up opportunities for teaching composition, penmanship, and spelling. All this, remember, was initiated by the tape recorder. Listening to one's own efforts at a later time allows for more objective self-evaluation. Co-operation is learned through projects requiring group effort.

Developing appreciation of good music becomes a much simpler problem when the music sounds as the composer and interpreter intended it to. An instrumental-music program is assisted immeasurably, since progress is immediately obvious and encouraging to the student. This growth in accomplishment can be the finest type of inspiration to the pupil. There are also, of course, the readily apparent uses in the improvement of reading and speech.

All these advantages are purely academic. In addition, exposure of students to quality sound systems can do much for the advancement of high fidelity. When Johnny goes home and tells dad that his hi-fi table model sounds terrible compared to the system at school, you can be sure that dad will end up with a better system soon.

WOODCRAFTER

Continued from page 8

The basis of *plastic resin glue* (Fig. 4) is formaldehyde, uric acid, and other chemicals. Like casein glue, it is available as a fine white powder which must be mixed with water before it is ready to be used. However, plastic resin glue should be spread in a thin coat; this means the joint must fit snugly. It has many applications in cabinet work, veneering, building construction, boats, aircraft, and other projects, indoors and out. Since it does not stain, it lends itself especially to use on light-colored woods. Working temperature should

Courtesy United States Plywood Corp.

Fig. 5. A completely waterproof glue.

never be less than 70° F., and higher temperatures will speed the setting of the glue. Light work can be handled after 6 or 7 hours under pressure; heavy work requires 10 hours or more.

Where a completely waterproof glue is needed, *resorcinol resin glue* is the adhesive to use. It comes in two-part cans, a liquid resin in one and a powder catalyst in the other (Fig. 5). The two are mixed when the work is ready to be glued. It is particularly suited for use on outdoor furniture, boats, and other equipment constantly exposed to weather. Since heat speeds up the setting of the glue, place the container of mixed glue in cold water to retard the setting while you are using it in warm weather.

Polyvinyl glue (Fig. 6) is more popularly known as "white glue" because of its color. It is strong, stainless, and dries clear, requiring only half an hour for setting on most woods at a temperature of 70° F. With a thin coat applied to each surface, the work should be clamped until the glue sets. Its range of application also includes paper, cloth, leather, china, hardboard, hard plastics, and other materials. The glue is packaged ready to use in tubes, jars, and

Courtesy Paisley Products, Inc.

Fig. 6. Quick-setting ready-mixed glue.

handy plastic squeeze bottles, and is gaining new friends every day as a good all-around adhesive. However, where a water-resistant glue is needed, a plastic resin or casein glue should be used.

Clamps

The woodworking clamp provides the pressure essential to a good glue joint. The choice of clamp is determined by the type and size of the job.

C-clamps or *carriage clamps* (Fig. 7) are available in sizes from 2 in. to 20

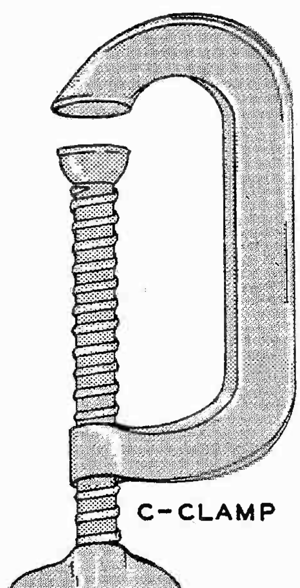


Fig. 7. Popular home-workshop clamp.

in.; practical sizes for a home workshop are 6 in. to 10 in. The C-clamp is useful for clamping pieces of irregular shape, for holding jigs on power tools, clamping metal parts together, and many other chores. When clamping wood, use

Continued on next page

WOODCRAFTER

Continued from preceding page

small plywood squares to keep the jaws of the clamp from damaging the work.

For home workshop use, the 6-inch to 12-inch sizes of *hand-screw clamp* (Fig. 8) are recommended. For larger work they can be bought with openings up to 20 in. They can be used for many clamping operations such as on tapered surfaces, since the angle of pressure is adjustable.

Bar clamps or *pipe clamps* are used for holding wide assemblies together. Use wood blocks to prevent marring

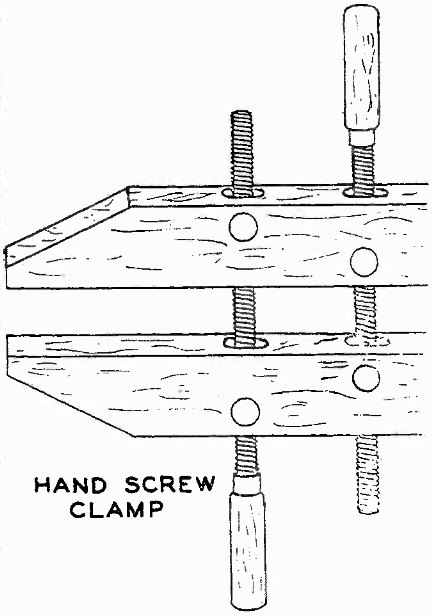


Fig. 8. Can be used on tapered surfaces.

the stock and alternate the clamps on opposite sides of the work. If the wood tends to buckle, flatten it with boards across each end of the work held fast with C-clamps or hand-screw clamps. In gluing up a table top, there will be less chance of future warping if narrow strips of lumber are used rather than wide boards. Another precaution is to alternate the direction of the end grain of the boards.

Gluing Hints

1) Mark surfaces to show where they are to be joined.

2) Have clamps properly adjusted in advance to the right opening.

3) Check your work by clamping pieces together without glue to see if all joints fit correctly.

4) Be sure the surfaces to be glued are dry and clean. Old glue must be completely removed.

5) Use the correct glue, mixing only the amount necessary and following the manufacturer's directions.

6) A rough surface has greater holding power than a smooth surface. Use a rasp or coarse sandpaper to open the pores of the surfaces to be glued.

7) Both glue and wood should be at least at normal room temperature for proper absorption of glue by the wood pores.

8) Apply glue with a stick or brush, covering the surface without overloading it.

9) When gluing the porous end grain of wood, first apply a thin coat and, when it becomes tacky, apply a second coat before joining the parts.

10) In clamping, do not apply so much pressure that the parts buckle or the edges become marred. Doing so can result in a glue-starved joint.

11) Remove excess glue before it dries by rubbing with sawdust.

12) Allow assembled piece to dry according to manufacturer's directions.

Twenty-four hours is a safe time limit usually.

13) If excess glue has dried on the work, remove it with a chisel or knife; never use a plane.

14) Using glue wherever wood is brought together by nails or screws adds strength to the joint. In such cases the nails or screws supplant the need for clamps.

GROUNDED EAR

Continued from page 4

measurement, it will not serve alone as a true measure of cartridge response on typical commercial recordings.

Fairchild XP-2 Cartridge

If I seem to be devoting this issue entirely to cartridge news it is because, by coincidence, I have received two that are well worth discussing. For the second, Fairchild has announced an addition to its commercial line. Those who have visited audio shows in the past year have no doubt heard Fairchild's XP-2, an experimental handmade modification of its regular 225A. I hope shortly to present some measured data on the XP-2. For the moment, let it suffice to say that many others who have had the opportunity to hear and try it share my opinion that it is one of the very finest cartridges so far developed. Especially notable is the extremely low IM distortion.

The XP-2 has hitherto been made available only to a few, for the purpose of obtaining user experience and reaction. It is now available through regular audio dealers at a price of $60. I am not sure that most people would find the difference in price between the XP-2 and the extremely fine production-model 225A audible or justifiable, but a few who are chasing perfection may consider the premium worth while.

Tuner Note

The trend toward wide-band FM detectors continues. Now Fisher announces a new tuner, the FM-90, which has wide-band detection and what is claimed to be an unusually effective limiter for noise reduction. If the FM-90 lives up to previous Fisher performance in other respects, these improvements will make it a hot item indeed.

READERS' FORUM

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quantity, and others said an order for a single motor would be impossible.

"On this basis, $45.00 is still a good price. It is, however, a new price to me, as the enclosed photostats will show. . . . There was nothing in my request for motor prices or in their reply and invoice to indicate that the price I paid was other than normal, and I proceeded on this assumption. Hence the figure I gave you."

We apologize for whatever inconvenience may have been caused readers by the incorrect price quotation. Whatever the reason for the mixup, it is a consoling fact that, as Mr. Geraci says, the Eastern Air Devices motor is a very good buy even at $45.00.

TRADERS' MARKETPLACE

Here's the place to buy, swap, or sell audio equipment. Rates are only 20¢ a word (excluding address) and your advertisement will reach 20,000 to 35,000 readers. Remittance must accompany copy and insertion instructions.

TAPE NEWS

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for single-mike recording technique by several record companies, multimike recording is more versatile and often gives better results than can a single mike.

The requisite items for multimike recording are an input mixer, two or more microphones, and a pair of high-fidelity headphones. Mixers, which provide independent control of volume from two or more input sources, are available at a cost of from $5 to $1,000 or more. There are two basic types, sometimes termed "wet" and "dry", on the basis of their circuitry.

Although there are enough unclassifiable mixers on the market to confound any attempt at distinguishing wet from dry mixers, it is usually deemed acceptable to consider as "wet" any mixer that has an independent preamplifier stage for each input channel. Dry mixers, then, are those which do their mixing *before* preamplification, thus having a single preamplifier stage serving all inputs, or which have no preamplifier stages at all. This dubious distinction will be challenged by those who feel that the important difference lies in *how* signals are mixed (through tubes or directly from the controls) rather than where they are mixed, but the distinction is not really important except as it affects mixer performance.

There is some insertion loss introduced by all dry-mixer circuits, so if a dry mixer is inserted between the mikes and the preamp input, it will reduce the system signal-to-noise ratio. Also, some dry mixers introduce serious high-frequency losses because of their very high-impedance circuits, although their generally low cost makes them attractive to budget-conscious recordists. They run from about $5 to $75, whereas the cheapest wet mixer is likely to be priced at around $20. The best wet mixers, with low-impedance input transformers, step-type attenuator controls, a db meter, and balanced 600-ohm output, start at around $200 and go beyond $1,000. Such deluxe instruments are fine, but most home recordists can get along with something more modest.

Wet mixers are generally quieter and more positive in operation than dry types, and their isolated input stages permit the use of different input impedances in different channels. Cost is their most important drawback as far as the amateur recordist is concerned, although some excellent ones can be home-constructed for a fairly modest sum.

With a mixer providing independent control of several microphones, many of the compromises necessitated by single miking can be largely avoided and all sorts of special effects can be achieved. Comprehensive multimiking has, in fact, been the basis for some of the most spectacular "hi-fi" recordings on the market today, but it is also a source of musical mayhem.

While it is possible to get improved results from multimiking a large group, it is also easy to produce very much *worse* recordings than could ever result from single-mike pickup. The imbalance and sonic mish-mash of a badly multimiked tape can sow seeds of despair in even the most ardent recordist, but the realism of a good multimiking job can be worth 25 failures.

It cannot be too strongly emphasized, though, that anyone contemplating a multimike recording should be confident that he has learned how to handle single mikes, and should be thoroughly familiar with the individual characteristics of each microphone he intends to use. It may be difficult to learn how to analyze the causes of an unsatisfactory single-mike recording, but even a topnotch recording engineer can grow gray hairs trying to analyze multimiked sound. However, for anyone who feels ready for it, and is willing to part with the necessary cash for a couple of additional microphones, a good input mixer, and a pair of high-fidelity headphones, multimiking can open up a whole new field of experimentation and perplexity. It can also make the difference between good recordings and excellent ones.

ADVERTISING INDEX