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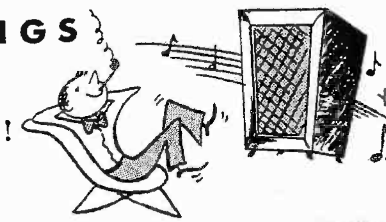

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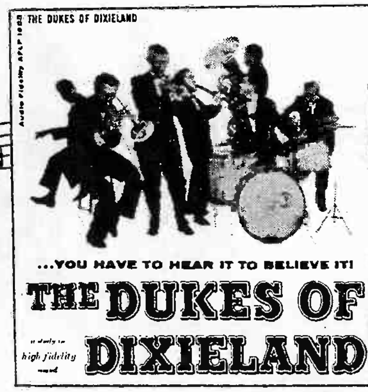

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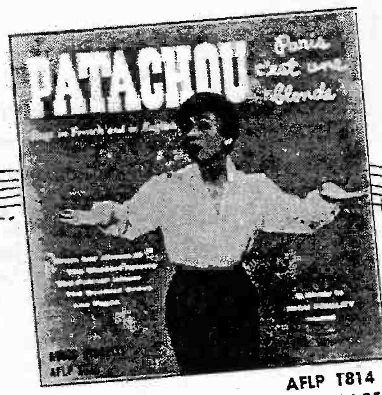

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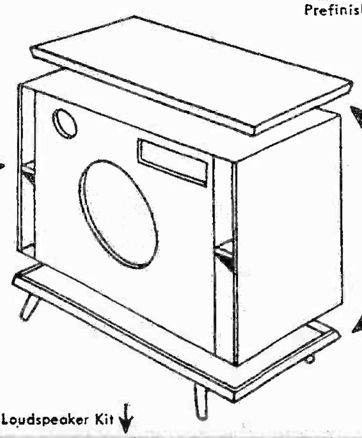

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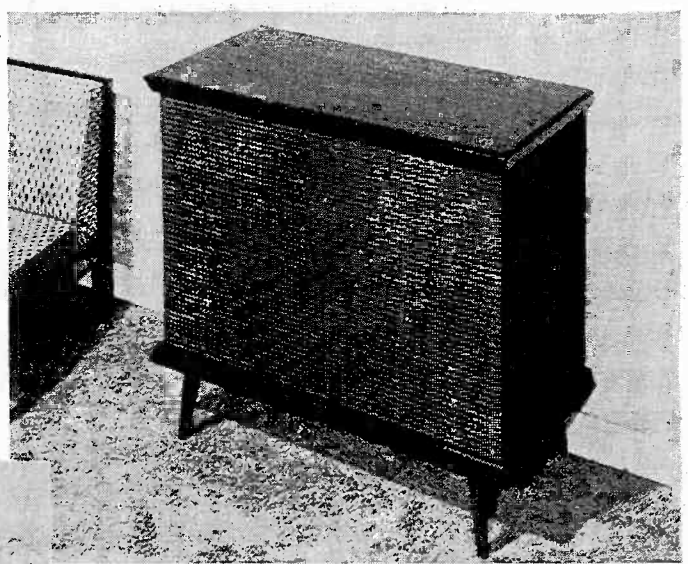
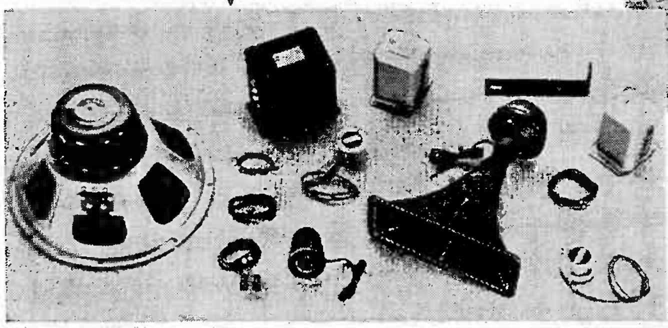


Jensen KT-32 Loudspeaker Kit



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System Type	"Woofer" Size	Equivalent Jensen Reproducer	Jensen Speaker Kit Model	Jensen Speaker Kit Price	Type	Cabinart Cabinet Kits: Basic Cabinet Kit	Price	Dress Kit‡	Price
3-Way	15"	Imperial	KT-31	$184.50	Corner* Horn	K-101	$89.00	P-201	$54.00
3-Way	15"	Triplex	KT-32	169.50	Corner* Bass-Ultraflex	K-103	48.00	P-203	39.00
3-Way	15"	Triplex	KT-32	169.50	Low Boy Bass-Ultraflex	K-105	48.00	P-205	39.00
2-Way†	15"	——	KT-21	99.50	Corner* Bass-Ultraflex	K-103	48.00	P-203	39.00
2-Way†	15"	——	KT-21	99.50	Low Boy Bass-Ultraflex	K-105	48.00	P-205	39.00
2-Way†	12"	Concerto	KT-22	73.00	Corner* Bass-Ultraflex	K-107	39.00	P-207	36.00
2-Way†	12"	Concerto	KT-22	73.00	Low Boy Bass-Ultraflex	K-109	39.00	P-209	36.00
2-Way†	8"	Contemporary	KDU-10	24.75	Corner* Bass-Ultraflex	K-111	23.00	P-211	25.00
2-Way	8"	Duette Treasure Chest	KDU-10	24.75	Duette	K-113	18.00	P-213	21.00

* Gives excellent results against sidewall. Bass-Ultraflex is a Jensen trademark.
† Cabinet provides for expansion to 3-way system at any time with Jensen KTX-1 Range Extender Supertweeter Kit, price $43.75.
‡ Available in Mahogany or Korina Blonde.

For information about Jensen Speaker Kits write:

Jensen
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Division of The Muter Company
6601 SOUTH LARAMIE, CHICAGO 38, ILLINOIS
In Canada: Copper Wire Products Co., Ltd.

For information about Cabinart Cabinet Kits write:

CABINART
The Pioneers in High Fidelity Radio Furniture
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January 1957

Volume 2 Number 1

audiocraft

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

We have only one author in this issue who is contributing to AUDIOCRAFT for the first time. **Russell J. Tinkham** is an engineer and Manager of Audio Custom Engineering for Ampex Corporation. He lives in Palo Alto, a short (for California, anyway) drive to Ampex's Redwood City plant.

Many owners of home stereo playback equipment have discovered to their consternation that, with the usually recommended speaker separation of 8 to 10 ft., there is an obvious and disturbing "hole" between the speakers with some types of stereo material. With such a hole much of stereo's realism is lost, for it is difficult to retain an illusion of source breadth when the sound is obviously coming from two distinct sources with nothing between them. And, with such wide separation, the dual-source effect is even more pronounced when both speakers are used for monaural reproduction.

This problem was of vital concern to Ampex, of course, since the company was one of the pioneers in stereo tape recording, and now manufactures home as well as commercial stereo tape machines. It has accumulated a great deal of experience in large-scale stereo demonstrations; recently, it undertook an extensive series of tests on stereo reproduction in small rooms—with most interesting results. They are described by Mr. Tinkham in his article beginning on page 16.

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The Grounded Ear

by Joseph Marshall

Wall-Mounted Multiple Speakers

Fully 50% of my mail from readers — especially from newcomers to high fidelity — concerns speaker systems. The general preoccupation with the subject is reflected also in AUDIOCRAFT articles, as readers have no doubt noted. Apparently, most people seem to know what pickup, preamp, amplifier, and turntable they want, but not many are completely sure about the speaker system. Many others obviously are disturbed by the high cost of speakers capable of the performance that the rest of their systems can deliver.

There is one method of achieving excellent speaker performance at relatively low cost (provided circumstances permit) that is commonly neglected; since several of my readers have asked specifically about it, while others might find it a means of solving their problems, this month I shall try to summarize in a few words the benefits and the techniques of using wall-mounted speakers.

Many have tried *single* speakers in wall mountings and, while some have liked the resulting sound, others have been disappointed. The trouble with a single wall-mounted speaker is that it is inefficient in bass response as compared with a horn or bass-reflex enclosure. A horn achieves, in effect, a bass boost by uniform air loading; a bass-reflex system reinforces the bass by inverting the phase of the back wave and radiating it so that it adds to the front wave. On the other hand, when a single speaker is mounted in a wall, the back wave is suppressed (or radiated into another room) and does not augment the front wave. Furthermore, a single piston moving into a 180° air space has a tough time moving enough air at low frequencies unless it is specifically designed for the purpose.

What is needed is to bring up the efficiency of wall mounting in the bass end. One way would be to mount it in a corner where the adjacent walls and floor provide images and permit the single piston to radiate more efficiently at low frequencies. Since most reflex baffles or horns are also placed in corners, the single wall-mounted speaker will still be far less efficient for bass.

The answer is to increase the piston area by adding additional speakers close-coupled to the first. The gain in efficiency at very low bass frequencies is marked; theoretically it is equal to the square of the increase in piston area. That is, two identical speakers close-coupled in a wall will be not merely twice as efficient but four times as efficient at extremely low frequencies as either one of them would be alone; four speakers are 16 times more efficient than any one of them would be. The theoretical efficiency comes reasonably close to being achieved in practice.

Furthermore, along with the increase in efficiency there is achieved a similar decrease in distortion (at bass frequencies) and a smaller but very significant decrease also in middle- and high-frequency distortion.

So far we have been talking about identical speakers. Additional advantages are achieved by using dissimilar speakers. For example, the bass response at the low end may be extended smoothly by using speakers with staggered resonance frequencies. Also, no speaker has a flat curve above resonance. In combinations of dissimilar speakers the response is flattened out because the

peaks and dips are in different portions of the spectrum; frequently, the peaks of one fall into the dips of another, and, in any case, the individual aberrations are far less noticeable.

Finally, whereas horns and other bass-augmenting enclosures have a steep cutoff slope below the system resonance, wall-mounted speakers have a more gentle slope, so that there is more usable response in the final octave or two which costs the most to achieve.

In addition to these benefits multiple wall-mounted speakers share the advantage of a single wall-mounted speaker — the lack of enclosure resonance and, in comparison with a horn system, the absence of problems in maintaining homogeneity of sound source. Wall mounting produces a characteristic dull, nonresonant sound. When several speakers are used, the sound source is widened and the hole-in-the-wall effect is minimized. If the multiple speakers are mounted one above the other vertically, there is some small sacrifice in bass efficiency over that provided by mounting them in a square or rectangular cluster, but the horizontal angle of radiation is widened.

So much for the benefits. How do you do it? The best place for a set of wall-mounted speakers is in the partition or wall between two rooms. The exact location is not critical. The longer the distance from the front of the speakers around to the back, the better the low-frequency response. A front-to-back distance of at least 25 ft., by the shortest path, will give good results, and greater distances are even better. A location in a corner of one or both rooms will increase bass efficiency, but if more than two speakers are used this increase might well be too great for some rooms; therefore, middle-of-the-wall positions work out nicely with three or four speakers.

When a partition or wall between two rooms is used, the sound will radiate into both rooms. This may or may not be an advantage. On the rear side of the speakers high-frequency losses can be corrected if necessary by using a tweeter facing the back-side room.

The back radiation can be exhausted also into a closet, a stair well, an attic, or into the garage.

Assuming that your home permits any of these locations and you can cut into the walls, the actual mounting is not difficult. Cut the wall away on both sides between two of the studs from the floor to the height sufficient to accommodate the number of speakers you want to use. Put pieces of two-by-four across the top and bottom of the desired aperture between the studs. Now mount your speakers on a piece of ¾-inch plywood, one above the other, with the rims 3 or 4 in. apart.

The plywood should be about 4 in. wider and 2 in. higher than the stud framing, so it will overlap the surface of the wall in one room and permit screwing the edges to the two-by-fours of the wall. To minimize vibration of the wall and resulting loss of low-frequency power, glue strips of sponge rubber, automobile door seal, or weather stripping to the back side of the board at the edge, so that the rubber will fit between board and wall, sealing air leaks when drawn tight. Screw the speaker panel to the studs with long wood screws. Since walls are usually only 4 in. thick, but speakers are much deeper, the backs of the speakers will protrude beyond the surface of one wall. You can make a frame deep enough to clear the speakers and screw this or nail it firmly to the wall. Then cover front and back with grille cloth, repairing plaster where necessary.

The better the individual speakers, the better the sound; but a special virtue of the method is that even inexpensive speakers produce a notably good sound when multiplied this way. I have tried many combinations and have finally settled on the following set of four: a pair of Hartleys at the top, an RCA LC1A at the bottom nearest the floor, and an RCA 515-S2 between the LC1A and the two Hartleys. Axiom 80's are well suited for wall mounting, although I would use no more than two, and add some other type to increase the number. The lower the resonance points of the speakers, the better the response below 40 cps. Speakers intended for use in infinite baffles are, of course, excellent.

The speakers can be hooked up in series-parallel combinations so as to achieve a given resultant impedance or to divide the input to each speaker in proportion to its capabilities or character. My four are wired very simply this way: the two 4-ohm Hartleys in series to make a pair with an 8-ohm impedance; the two RCA's in parallel to make a pair with an impedance of 8 ohms. The two pairs are then paralleled to make a total impedance of 4 ohms. I do not like complex crossover networks and would avoid speakers that use them, although when one or more (for example) triaxials with integral networks are used with other speakers not having such networks (or simply a capacitor before the tweeter to roll off bass input), the effects of the network are minimized.

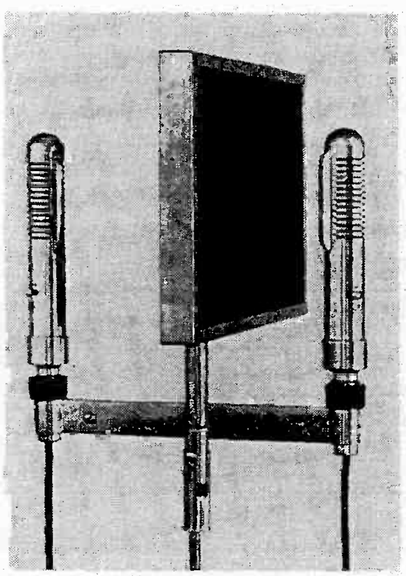
I realize that not everyone is able to cut a hole in the wall and that, even in households where this can be done, there will be competition between his plan for speakers in a given place and her idea for a breakfront or a refrigerator in the same spot. But where the method *can* be used, a given amount

Continued on page 45

Audionews

NEW B&O 53 MICROPHONE & "BINOR" STEREO RIG

The Fenton Company has introduced a new version of its popular B&O 50 microphone — the *B&O 53*. The B&O 53 uses an identical magnet and ribbon assembly. However, it has a multi-tapped output transformer with output impedances of 50, 250, and 40,000 ohms. It is housed in a cast-iron body for better shielding. The B&O 53 is slightly heavier than the B&O 50 — approximately 19 oz. instead of 15 oz. — and is available either in matte-finish "TV-grey" or satin-finish chrome. It is supplied with 20 ft. of cable, female



"Binor" assembly for stereo recording.

microphone connector, and a snap-on bayonet adaptor to ⅝ by 27 thread stand. The B&O 53 has the same ball swivel as the B&O 50, and the same three-position voice, music, and off switch.

The output is said to be only slightly lower than that of conventional moving-coil microphones, while the hum sensitivity is reported to be considerably below that of other conventional types of microphone. The impedance-selection switch is screw-driver operated, while the voice, music, and off switch is finger operated.

Since stereo is becoming more important to the public in high-fidelity reproduction, there is a definite need for some simple and foolproof method of positioning two microphones for stereo recording, rather than using the normal 9-foot separation between the microphones and becoming involved in the consequent difficulties of the amateur in locating proper placement. Experimentation resulted in what is now called the *Fen-Tone B&O Binor Rig*. This rig utilizes a little-known facet of the polar characteristics of a velocity microphone; that is, a spiral and logarithmic increase in sensitivity from the side towards the front or back. Actually the particular portion of this curve which is usable in stereo recording is located midway between the front and the side, so that, if two microphones are placed approximately 10 in. apart and each faces 45° away from the front of the rig, this particular portion of the polar pattern is utilized. The addition of an acoustical separator between the two microphones completes the stereo rig. Results achieved with this method are said to be indistinguishable from those achieved with standard microphone positioning. The working principle actually evolves from the angle of divergence of an apparent sound source as the function of the difference in level between two sources. The portion of the polar curve of a velocity microphone used very closely matches this angle of divergence curve.

In addition to other benefits, the Binor rig is said to overcome one of the primary objections to wide microphone spacing — that of a "hole" between the two microphones. If a small source (solo instrument or soloist) is positioned mid-way between the two microphones and on the axis between the two microphones, it stands approximately at the null point of sensitivity between them. Binor is said to eliminate this entirely.

The Binor rig can be used only with two B&O 50 or two B&O 53 microphones.

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.

NEW KLIPSCH SHORTHORN

A new corner-horn loudspeaker system has been introduced by Klipsch & Associates. Designed by Paul W. Klipsch and known as the *Shorthorn Model "T"*, it may be used to provide full-range sound reproduction for television, record and tape players, and radio. Its dimensions permit placement of a table-model television set on top of it.

According to the manufacturer, the range of the new unit extends from below 40 cps to above 22,000 cps, with substantial efficiency from 45 to over 16,000 cps.

Like the Model "S" Shorthorn, the Model "T" is available with the Klipsch K-ORTHO three-way drive system. This system is made up of elements selected by means of listening and engineering tests, ensemble tested by Paul W. Klipsch. It includes three drivers, with a choice of a 12- or 15-inch bass cone driver,

The new and shorter Klipsch Shorthorn.

and crossover network. The bulk of the Model "T" is the bass horn, and, mounted inside of it, are the mid-range and high-frequency horns.

While it achieves its maximum range when used in corners, the Model "T" has a "built-in" corner which permits it to be used in any part of the room with a sacrifice of part of the bass range. It is equipped with casters as a convenience for those who may wish to move it from one place to another.

Further information on the Klipsch Shorthorn Model "T" is available from the manufacturer.

RECORD-INDEXING SYSTEM

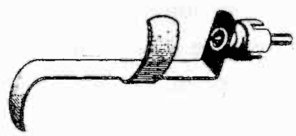
If you're thinking of indexing your record or tape collection, you'll be interested in a new filing sytsem developed by the Old Colony Sound Lab of Roxbury, Mass.

This system, centering around the use of a specially designed stamp, is foolproof and exceptionally simple to operate.

Full particulars about the Old Colony System will be furnished on request to Old Colony Sound Lab, P. O. Box 91, Roxbury 20, Mass.

SOLDERLESS PHONO PLUG

The Solderless Phono Plug, *Model PP*, manufactured by Workman TV of Teaneck, N. J., is a new type of plug especially designed to eliminate soldering. It is reported to be easily attached in



This phono plug requires no soldering.

one minute, and can be used with any coaxial cable or shielded wire commonly used in audio.

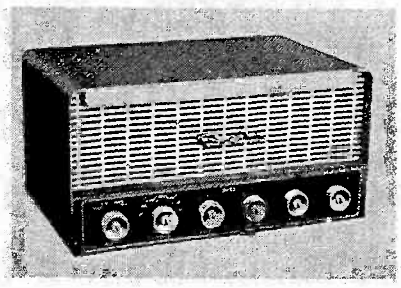
The curved finger pull allows easy insertion or removal of the phono plug without undue stress on either the plug or the attached cable, thus eliminating broken wires or pulled-out pins. Installation is accomplished by first forcing the center conductor of the cable or wire onto the sharp pin of the Solderless Phono Plug and then tightly crimping the side tab of the phono plug over the exposed shield braid of the cable.

BELL SOUND STEREO LINE

Bell Sound Systems, Inc., have introduced three new pieces of equipment designed to make high-fidelity stereophonic sound available in the home at moderate prices.

The Bell *Model 3-DTG* two-channel amplifier provides complete preamplification and power amplification for stereo signals from any source — tape deck or recorder (with or without preamplification), phonograph, or AM-FM tuners. Single-knob controls on this one-chassis amplifier affect both channels equally and simultaneously to permit stereophonic balance of the two speaker systems powered by the amplifier's dual outputs. The output of the amplifier is rated by the manufacturer at 10 watts on each channel. Monaural program material can also be reproduced through the two channels and speakers.

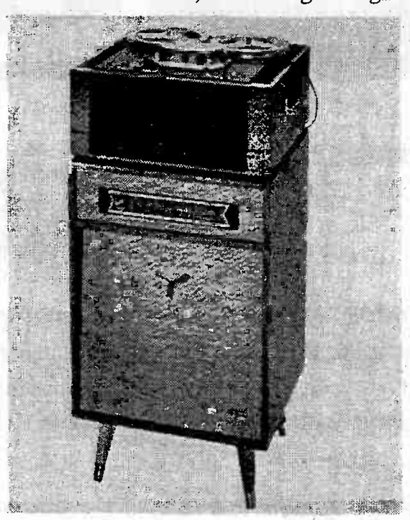
Portability and moderate price are features of the second new Bell item,



Bell model 3-DTG stereo amplifier.

the *BT-76* tape recorder with stereophonic playback. This unit offers high-quality monaural recording and playback, with stereophonic playback provided by a second head and preamplifier. Staggered stereo heads feed separate equalizing preamplifiers, with these signals channeled for No. 1 head into the recorder's self-contained power amplifier and speaker, and from No. 2 head (by furnished cable) to any radio, TV, or other amplifier with phono input, to utilize it for a second power amplifier and speaker. Various adaptations are possible; an external speaker can be attached to the recorder, output from either or both heads can feed other amplifiers, or both heads can feed the Bell Model 3-DTG stereophonic amplifier. The BT-76, alone, retails for less than $200.

The third new Bell item is an I. W. Simons-designed matching console cabinet for the BT-76, containing a high-

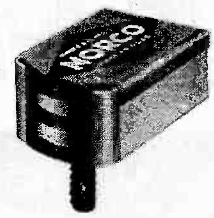


Console amplifier-speaker unit; BT-76 monaural recorder with stereo playback.

fidelity amplifier and a fully finished pull-out speaker enclosure for optional use at a remote location. This provides the second power amplifier and remote speaker necessary to utilize the stereophonic potentialities of the BT-76. The unit can also be used by itself as a high-fidelity amplifier with speaker.

NORCO RECORDING HEAD

The Nortronics Company of Minneapolis has announced a *Model TLD* in-line magnetic head for low-cost, high-quality recording and reproduction in stereophonic sound applications. The head, it is stated, can be compensated for flat response between 30 and 10,000 cps at 7.5 ips. It is compact and is reported to provide long wear, negligible oxide accumulation, good rejection of surrounding fields, and uniformity of frequency and amplitude response. The new head features precision-ground and lapped gap, balanced electric and magnetic structure, high output, and precise colinear alignment. The active tape surfaces do not pass over any epoxy resin



New high-quality stacked stereo head.

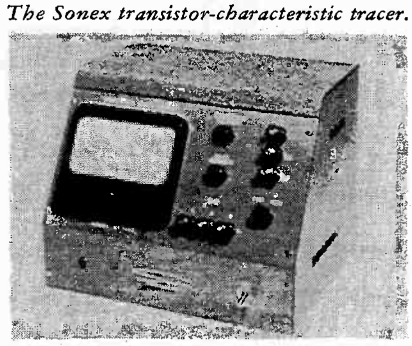
or plastic surfaces. The head is reported to be suited for use in new equipment design, for replacement, and for conversion of existing tape recorders to stereophonic operation. Detailed dimensional drawings and specifications will be furnished upon request.

TRANSISTOR CURVE TRACER

The Sonex *Transistor Curve Tracer*, when used with an oscilloscope, presents one curve at a time of the collector family (V_c *vs.* I_c), with I_b held constant. This may be done on all n-p-n, p-n-p, surface barrier, grown or diffused-junction transistors. The base current is indicated at all times on a 4-inch panel meter and can be varied from 0 to 500 ma while the limits of collector voltage being swept are controlled. Calibrated co-ordinate axes are displayed at all times.

Another use for the curve tracer is the presentation of the forward and reverse characteristics of crystal diodes with the calibrated axes. The voltage *vs.* current characteristics of thermistors and varistors can also be displayed simultaneously with calibrated axes.

The Sonex transistor-characteristic tracer.



Heathkit Model FM-3A High Fidelity FM Tuner Kit
Features A.G.C., and stabilized, temperature-compensated oscillator. Ten uv sensitivity for 20 DB of quieting. Covers standard FM band from 88 to 108 mc. Ratio detector for efficient hi-fi performance. Power supply built in. Illuminated slide rule dial. Pre-aligned coils and front end tuning unit.
$25.95* (With Cabinet) *Shpg. Wt. 7 Lbs.*

Heathkit Model BC-1 Broadband AM Tuner Kit
Special AM tuner circuit features broad band width, high sensitivity and good selectivity. Employs special detector for minimum signal distortion. Covers 550 to 1600 kc. RF and IF coils pre-aligned. Power supply is built in.
$25.95* (With Cabinet) *Shpg. Wt. 8 Lbs.*

Heathkit Model WA-P2 High Fidelity Preamplifier Kit
Provides 5 inputs, each with individual level controls. Tone controls provide 18 DB boost and 12 DB cut at 50 CPS and 15 DB boost and 20 DB cut at 15,000 CPS. Features four-position turnover and roll-off controls. Derives operating power from the main amplifier, requiring only 6.3 VAC at 1 a. and 300 VDC at 10 ma.
$19.75 (With Cabinet) *Shpg. Wt. 7 Lbs.*

Heathkit Model W-5M Advanced-Design High Fidelity Amplifier Kit
This 25-watt unit is our finest high-fidelity amplifier. Employs KT-66 output tubes and a Peerless output transformer. Frequency response ± 1 DB from 5 to 160,000 CPS at one watt. Harmonic distortion less than 1% at 25 watts, and IM distortion less than 1% at 20 watts. Hum and noise are 99 DB below 25 watts. Output impedance is 4, 8 or 16 ohms. Must be heard to be fully appreciated.
$59.75 *Shpg. Wt. 31 Lbs. Express Only*

MODEL W-5: Consists of Model W-5M above *plus* Model WA-P2 preamplifier. $81.50* *Shpg. Wt. 38 Lbs. Express only*

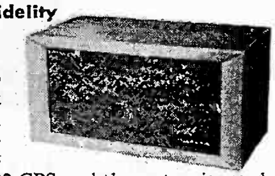
Heathkit Model W-3M Dual-Chassis High Fidelity Amplifier Kit
This 20-watt Williamson Type amplifier employs the famous Acrosound Model TO-300 "ultra linear" output transformer and uses 5881 output tubes. Two-chassis construction provides additional flexibility in mounting. Frequency response is ± 1 DB from 6 CPS to 150 kc at 1 watt. Harmonic distortion only 1% at 21 watts, and IM distortion only 1.3% at 20 watts. Output impedance is 4, 8 or 16 ohms. Hum and noise are 88 DB below 20 watts.
$49.75 *Shpg. Wt. 29 Lbs. Express only*

MODEL W-3: Consists of Model W-3M above *plus* Model WA-P2 preamplifier. $71.50* *Shpg. Wt. 37 Lbs. Express only*

HEATHKIT SPEAKER SYSTEM KITS

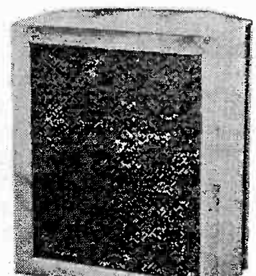
These speaker systems are a very vocal demonstration of what can be done with high-quality speakers in enclosures that are designed especially to receive them. Notice, too, that these two enclosures are designed to work together, as your high-fidelity system expands.

Heathkit Model SS-1 High Fidelity Speaker System Kit

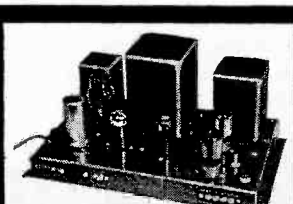


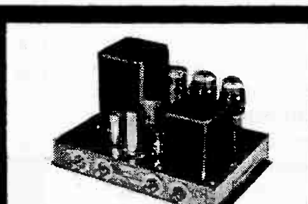
Employing two Jensen speakers, the Model SS-1 covers 50 to 12,000 CPS within ± 5 DB. It can fulfill your present needs, and still provide for future expansion through use of the SS-1B. Cross-over frequency is 1600 CPS and the system is rated at 25 watts. Impedance is 16 ohms. Cabinet is a ducted-port bass-reflex type, and is most attractively styled. Kit includes all components, pre-cut and pre-drilled, for assembly.
$39.95 *Shpg. Wt. 30 Lbs.*

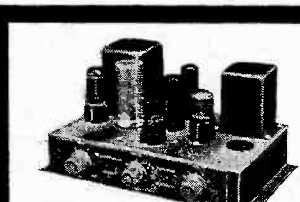
Heathkit Model SS-1B Range Extending Speaker System Kit

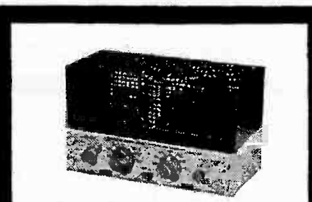


This range extending unit uses a 15″ woofer and a super-tweeter to cover 35 to 600 CPS and 4000 to 16,000 CPS. Used with the Model SS-1, it completes the audio spectrum for combined coverage of 35 to 16,000 CPS within ± 5 DB. Made of top-quality furniture-grade plywood. All parts are pre-cut and pre-drilled, ready for assembly and the finish of your choice. Components for cross-over circuit included with kit. Power rating is 35 watts, impedance is 16 ohms.
$99.95 *Shpg. Wt. 80 Lbs.*


MODEL W-4AM


MODEL A-9B


MODEL A-7D


MODEL XO-1

Heathkit Model W-4AM Single-Chassis High Fidelity Amplifier Kit
The 20-watt Model W-4AM Williamson type amplifier combines high performance with economy. Employs special-design output transformer by Chicago Standard, and 5881 output tubes. Frequency response is ± 1 DB from 10 CPS to 100 kc at 1 watt. Harmonic distortion only 1.5%, and IM distortion only 2.7% at this same level. Output impedance 4, 8 or 16 ohms. Hum and noise 95 DB below 20 watts.
$39.75 *Shpg. Wt. 28 Lbs.*

MODEL W-4A: Consists of Model W-4AM above *plus* Model WA-P2 preamplifier. $61.50* *Shpg. Wt. 35 Lbs. Express only*

Heathkit Model A-9B 20-Watt High Fidelity Amplifier Kit
Features full 20 watt output using push-pull 6L6 tubes. Built-in preamplifier provides four separate inputs. Separate bass and treble tone controls provided, and output transformer is tapped at 4, 8, 16 and 500 ohms. Designed for home use, but also fine for public address work. Response is ± 1 DB from 20 to 20,000 CPS. Harmonic distortion less than 1% at 3 DB below rated output.
$35.50 *Shpg. Wt. 23 Lbs.*

Heathkit Model A-7D 7-Watt High Fidelity Amplifier Kit
Qualifies for high-fidelity even though more limited in power than other Heathkit models. Frequency response is ± 1½ DB from 20 to 20,000 CPS. Push-pull output, and separate bass and treble tone controls.
$17.95* *Shpg. Wt. 10 Lbs.*

MODEL A-7E: Same, except that a 12SL7 permits preamplification, two inputs, RIAA compensation, and extra gain. $19.95* *Shpg. Wt. 10 Lbs.*

Heathkit Model XO-1 Electronic Cross-Over Kit
Separates high and low frequencies electronically, so they may be fed to separate amplifiers and separate speakers. Selectable cross-over frequencies are 100, 200, 400, 700, 1200, 2000, and 35,000 CPS. Separate level control for high and low frequency channels. Minimizes intermodulation distortion. Attenuation is 12 DB per octave. Handles unlimited power.
$18.95 *Shpg. Wt. 6 Lbs.*

*Price includes Fed. Excise tax where applicable.

HOW TO ORDER:

It's simple—just identify the kit you desire by its model number and send your order to the address listed below. Or, if you would rather budget your purchase, send for details of the HEATH TIME-PAYMENT PLAN! All prices subject to change without notice.

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TIPS FOR THE WOODCRAFTER

by George Bowe

Sanding Materials

The term *sandpaper* is, without a doubt, one of today's greatest misnomers because of the lack of sand in sandpaper. Many years ago sand, applied with a piece of wet hemp, actually was used as an abrasive for wood and metal; later pulverized glass glued to a paper backing proved even more effective. Down through the years various minerals, natural and synthetic, have added their abrasive qualities to the material which we still know today as sandpaper.

An old method of producing sandpaper involved dropping the abrasive on paper previously coated with glue. Today, the backing of cloth or paper, the abrasive, and the adhesive are all brought together in one machine. A charge of electricity passes through the abrasive particles as they are dropped on the backing causing them to stand on end, the best position for maximum cutting efficiency.

Courtesy Minnesota Mining & Manufacturing Co.

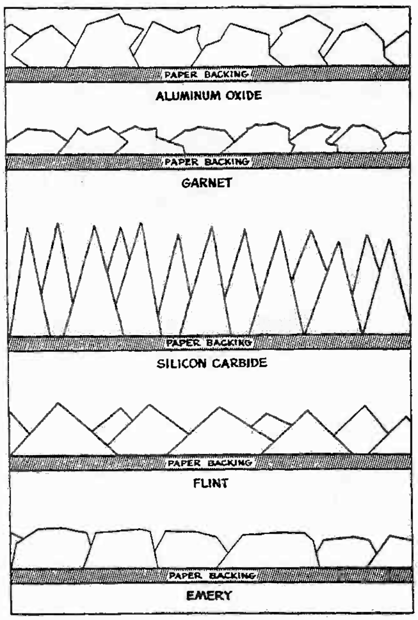


Fig. 1. The five types of sandpaper.

I like to think of sandpaper as a tool — a tool that has great bearing on the finished appearance of a project. No matter the craftsmanship that has gone into construction, selection of the right kind of sandpaper and the proper grit size often means the difference between a job that looks homemade and a home-workshop project that has an eye-catching, professional-looking finish. You have a choice of five different mineral abrasives: flint, garnet, emery, aluminum oxide, or silicon carbide. The first three are natural minerals taken from the ground in mining operations; the latter two are synthetic and are those most used in industry today. Each of the five has its own distinct characteristics (Fig. 1).

Courtesy Minnesota Mining & Manufacturing Co.

Fig. 2. How to sand a flat surface.

Flint is best known by home users because of its general availability. It is found in New Hampshire and Maryland, and resembles real sand. It is a relatively dull mineral compared to the synthetics, and its softness gives it a short cutting life. It serves best on simple sanding jobs such as smoothing heavily painted surfaces, gummy woods, etc.

Garnet is a reddish-colored rock found in the Adirondack Mountains in New York State. It has good, medium-hard cutting edges and is one of the best wood-finishing abrasives — excellent for hand sanding and for some types of power sanders.

Emery is imported from Turkey and the Grecian Islands. Dull black in color, it is a hard and durable finishing abrasive, slow cutting and used primarily for cleaning or polishing metal.

Aluminum oxide, one of man's creations, has bauxite in its chemical makeup. Of a brownish color, it is hard, sharp, and long-lasting. It might well be termed the best all-around sandpaper for household or workshop use on bare wood or metal, producing high-quality finishes with hand or power sanding. It is sold under such trade names as "3M-Ite", "Aloxite", and "Alundum".

Silicon carbide is another of the manufactured abrasives produced in high-temperature furnaces. Shiny black in color, it is used extensively in fine wood finishing, machine woodworking, and leather finishing. The cutting characteristics of this mineral make it ideal for imparting extra-smooth, professional-looking finishes to varnish or lacquer coats. Silicon-carbide sandpaper is made in two forms — one for dry sanding, and the other as a waterproof paper for sanding with lubricants. Since the waterproof type can be used either wet or dry, this type is recommended for home use. When used with light oil as a lubricant, silicon carbide in the finer grits (extra fine, super fine) gives lacquer a lustrous satin appearance. In various brands it is called "Tri-M-Ite", "Red-I-Cut", "Durite", and "Cristolon".

Until recent years, selecting the proper grit size in sandpaper was a somewhat bewildering experience for the average person. The back of each sheet bore such designations as 3/0 or 120, or even 000, depending upon the manufacturer, and only someone with a fair knowledge of sandpaper could tell whether the figures indicated a coarse, medium, or fine paper. The selection was usually reached after the customer had rubbed his thumb over the surface of the paper and decided it was "about rough enough" or "smooth enough" to do the job. Today, however, all that is changed — thanks to the initial move of the Minnesota Mining and Manufacturing Company in simplifying the grit system for home users. Now, most sandpaper manufacturers are replacing or supplementing the grit symbols and mesh numbers with such descriptions as *fine*, *medium*, and *coarse*. This is the way Minnesota Mining and Manufacturing's simplified grit system compares with industrial grit markings:

Courtesy Minnesota Mining & Manufacturing Co.

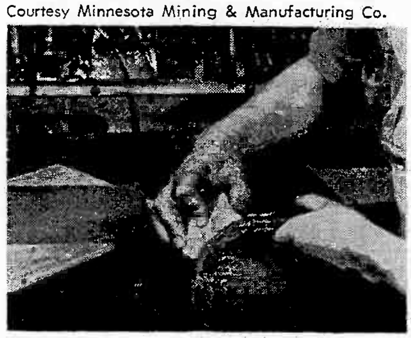
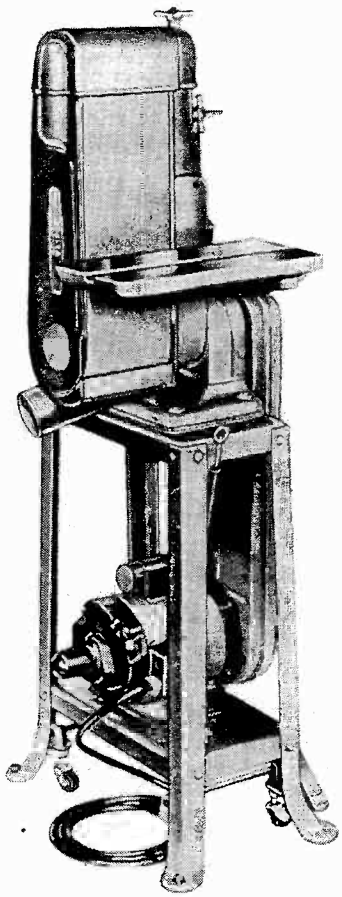


Fig. 3. Rubber sanding block for curves.

Courtesy Rockwell Manufacturing Co., Delta Power Tool Division.



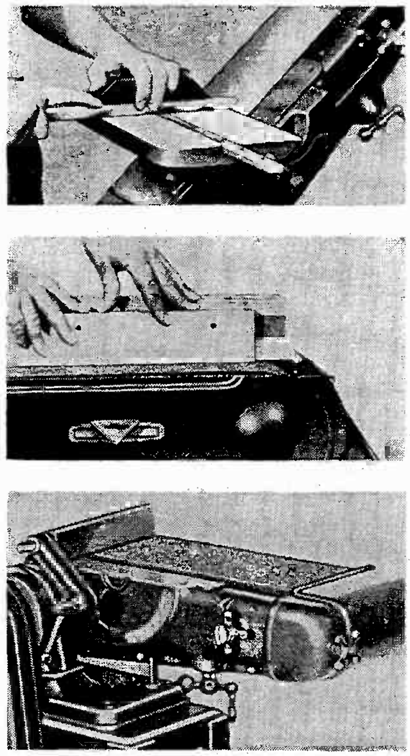


Fig. 4. Four views of the Delta 6-inch belt sander demonstrate its versatility.

	Simplified Grits	Mesh Numbers	Grit Symbols
		600	
		500	
	Super Fine	400	10/0
	Extra Fine	320	9/0
Finishing		280	8/0
		240	7/0
	Very Fine	220	6/0
		180	5/0
		150	4/0
	Fine	120	3/0
		100	2/0
	Medium	80	1/0
Bare Wood Sanding		60	1/2
	Coarse	50	1
		40	1½
	Very Coarse	36	2
		30	2½
		24	3
		20	3½
		16	4
		12	4½

Hand Sanding Flat Surfaces

When working on flat surfaces, use a rubber sanding block (Fig. 2) or a wood block with a felt pad or a piece of sponge rubber glued to the base. The block should be about 1 in. thick, 3 in. wide, and 4½ in. long to fit one-quarter of a sheet of sandpaper. To tear the sandpaper to size, hold it down with an old hacksaw blade and tear it as you would wax paper from its box; or use a piece of board or the back edge of your handsaw as a straightedge. On new wood surfaces usually a medium paper (1/2) is satisfactory for the first sanding, followed by a fine grit (2/0 or 3/0), and concluding with very fine (5/0) for a final rubdown.

Sand along the edges first, working toward the center. Keep the block flat to avoid rounding edges that are supposed to be squared. Exert uniform pressure and always sand with the grain; sanding across the grain or in a circular motion causes scratches which, in many cases, cannot be removed or are not noticed until a stained finish reveals their ugliness.

Sanding Edges and Ends

It is just as important to sand an edge or an end squarely as it is to plane it squarely. Use a wooden sanding block and hold it true with both hands. On end grain, sand in *one direction only* — the result will be a glass-smooth surface that will take a finish equal to that on a surface *with* the grain. Sanding in both directions on end grain serves only to smooth the grain down with one stroke and raise it again on the return stroke.

On rounded edges, dispense with the sanding block and pad the sandpaper with the palm of the hand which will cushion itself nicely against the contour of the wood. When sanding a concave surface or an inside curve, wrap the sandpaper around a tool handle or any round piece of wood and sand with the grain.

For rounding off sharp edges and corners, use a sanding block padded with rubber or felt. As pressure is applied, the softness of the pad yields and the sharp edge is transformed into one smoothly rounded. An old blackboard eraser wrapped in sandapaper lends itself most effectively to the performance of this operation.

For curved surfaces a rubber sanding block is ideal since it flexes sufficiently to allow the paper to follow the contour of the stock (Fig. 3).

Power Sanding

Today's craftsman has a choice of machines to spare himself the effort and

Continued on page 46

Courtesy Rockwell Manufacturing Co., Delta Power Tool Division.

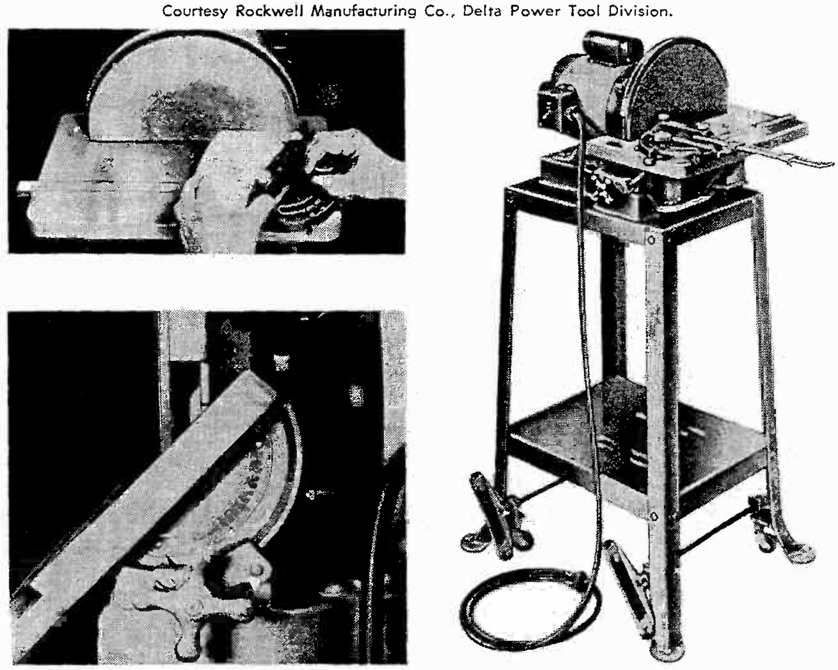


Fig. 5. The 12-inch Delta disc sander is also adjustable for many types of work.

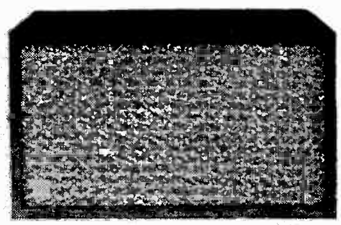
AR-2

The AR-2 speaker system uses the same ***acoustic suspension*** *principle as the AR-1. Because of this fact it is able to achieve a performance quality which, by pre-****acoustic suspension*** *standards, is associated with a price range several times higher than its 96.00.**

**in birch or mahogany; other finishes 89.00 and 102.00*

SUGGESTED PRICE RANGE FOR INSTALLATIONS USING THE AR-2

COMPONENT		PRICE
AMPLIFIER (10-30 clean watts, complete with controls)	*Install on shelves, or in cabinet*	$75 - - $125
RECORD PLAYER (changer or manual)		$40 - - $60
CARTRIDGE(S) (diamond needle for LP)		$20 - - $45
TUNER		$70 - - $100
AR-2 SPEAKER SYSTEM (Complete with enclosure; size 13½" x 11⅜" x 24")		$89 - - $102
		$294 - - $432
		(phonograph only, $224 - - $332)



Literature on request from:

ACOUSTIC RESEARCH, INC. 24 Thorndike Street Cambridge 41, Mass.

TAPE NEWS & VIEWS

by J. Gordon Holt

Microphone Technique I

Mercury records scored an instant hit several years ago with the first of their Living-Presence recordings, Moussorgsky's *Pictures at an Exhibition.* The disc was so well received by critics and record buyers that Mercury launched a full-scale campaign to commit everything to disc via the single Telefunken microphone suspended over the conductor's head. This has produced many fine recordings, and some not so fine.

The exceedingly crisp and rather dry Mercury sound is ideal for the bombast and robustness of *Pictures at an Exhibition*, the *1812* Overture, and *British Band Classics*, but it just doesn't come off with the Debussy Nocturnes and Ravel *Pavane*, for which an ethereal quality and sweetness of tone are required. Such anomalies are not the fault of the orchestras or even of the Living-Presence technique itself; they are just the result of failure to apply musical judgment to recording technique. Compare, for instance, Mercury's Nocturnes recording (MG-50005) with London's disc of the same thing (LL-530).

Certain types of music are supposed to sound bright and brassy; others simply aren't supposed to be that way. Some music sounds best enveloped in rolling, sustained reverberation, while other music calls for more dead, intimate acoustical surroundings. It is impossible to generalize about "the best" recording technique, except possibly to state that the best technique is the most flexible one.

Generalizations can, however, be made about the type of sound that usually (I repeat, *usually*) suits certain types of program material.

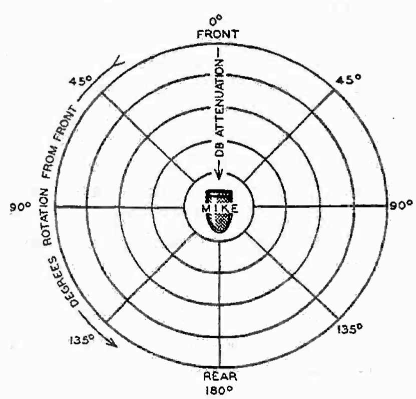
Music that is supposed to make your hair stand on end (Berlioz, Bruckner, Wagner, Saint-Saëns, and most works for large chorus) demands as powerful-sounding a recording as can be obtained, along with a large amount of hall reverberation to convey the illusion of size and spaciousness.

Chamber music and instrumental works which were originally intended for performance in rooms rather than auditoriums require a drier, more intimate sound with very much less reverberation.

Organ music generally calls for plenty of echo, with the actual closeness depending on the complexity of the music. A Bach fugue, for instance, should be fairly close-miked to preserve the details in its structure, whereas a slower, heavier work such as Franck's *Pièce héroïque* needs greater distance and "bigger" sound.

Romantic music having a neutral character, such as most of Beethoven's, Brahms's, and Tchaikovsky's orchestral works, should be miked neutrally. The sound should be sufficiently distant to provide good blending, yet close enough for adequate detail. Reverberation should be fairly prominent, yet not overwhelming.

Impressionistic music (Ravel, Debussy, Griffes) should sound airy, light, and misty; that is, it should be miked from a more distant point than most other types, but reverberation should be kept well under control. These are actually conflicting requirements; still, both can be obtained by judicious use of microphone characteristics.



Omnidirectional microphone is equally sensitive to sounds from any direction.

Determining the *type* of sound that a recording should have is the first step toward producing a tape that is both technically and musically satisfying; the next step is to achieve that sound, and this is where microphone technique comes into the picture.

Little need be said at this point about the quality of the microphones a recordist should use, except for the obvious admonishment to use the best that his finances will bear. But the *way* a microphone will be used depends more on its directional characteristic or pickup pattern than on its absolute quality.

Three general types of pickup pattern are to be found: omnidirectional, unidirectional, and bidirectional. Omnidirectional (or nondirectional) microphones have an essentially spherical pickup pattern; they pick up with equal sensitivity sounds orginating all around them. Unidirectional microphones are most sensitive to sound originating in front, and are less and less sensitive to a sound source as it circles toward the rear of the mike. Bidirectional mikes have a figure-8 pickup pattern, being most sensitive to sounds at the front and back and least sensitive at the sides.

There are variations of these three basic directional patterns, and some microphones are difficult to classify because they seem to fall mid-way between two types. But most will be found to have one or another of the basic patterns.

There is also usually some variation of pattern in every microphone with respect to frequency, so that no matter what basic pattern a microphone has, it has it even more so at high frequencies. A unidirectional microphone having essentially a heart-shaped (cardioid) pickup pattern will show maximum high-frequency response when aimed directly at the source of the sound. As it is turned to one side of the source, pickup of the very highest frequencies drops off faster than low-frequency response. Omnidirectional microphones (with at least one exception: the Altec 21-B) are truly omnidirectional only up to a certain frequency, above which they become increasingly directional. Their maximum high-frequency response is to sounds arriving perpendicular to the plane of the diaphragm surface. This tendency for increasing directionality at high frequencies is markedly less noticeable in slim microphones, since the diameter of the mike's case (more accurately, the effective diameter of the diaphragm surface) is directly related to frequency discrimination.

The importance of a microphone's directional characteristic to a recordist is that it enables him to control to a large extent the relative volume of sounds coming from different directions.

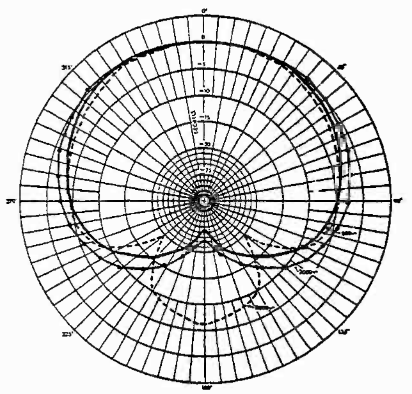
At any given spot in a concert hall, sounds of a performing group arrive from two different sources: directly,

Continued on next page

TAPE NEWS

Continued from preceding page

from the instruments on stage, and indirectly, as reflections from the walls and ceiling of the hall. Direct sound radiates outward from the instruments as do light rays from a candle; so as far as a microphone is concerned, they all approach from pretty much the same direction. The echoes that comprise indirect sound are, however, bouncing back and forth from the rear of the hall, the ceiling, the side walls, and the back of the stage. They arrive at the microphone from all directions, and they are likely to be equally loud throughout the entire hall. Direct sounds, though,



Pickup pattern for undirectional mike.

are not equally loud from one end of the hall to the other; they diminish rapidly in strength as they get farther away from their source, just as the light from a candle diminishes with increasing distance.

Since the microphone distance affects the intensity of direct sounds but not that of indirect sounds, we have a handy way of controlling the relative amount of echo or reverberation in our recordings. The farther away we place the microphone, the more the volume of direct sound approaches that of the indirect sound, and the closer we get, the *less* echo we seem to pick up because the direct sound is so much stronger.

The second factor which determines optimum microphone placement is the amount of definition and blending required. Definition is that attribute of recorded sound which enables a listener to separate readily, identify, and follow each instrument while the entire group is playing. It is a quality of crispness, sharpness, and the ability to hear the mechanical noises that accompany the musical sounds of instruments.

Blending is the opposite of definition. It is a merger of all the instrumental sounds into an integrated texture, with the individuality of instruments lost in the pattern of sound. In its extreme form, when practically all definition has been lost, it is referred to as blurring — but it should never be carried to an extreme in recording, any more than should definition.

Both these qualities are primarily functions of microphone distance. As the microphone moves away from the sound, definition decreases and blending increases. The optimum distance is, as before, that which produces the effect most appropriate to the music.

Those are two ways in which a simple microphone can be positioned to control the quality of recorded sound. In theory we should be able to determine immediately the mike placement that will give us the definition, blending, and reverberation qualities that we want. Unfortunately, though, it doesn't always work out that simply.

Auditoriums differ widely in reverberation characteristics, as anyone who tilts an attentive ear to hall acoustics well knows. Not only does the time required for an echo to die out (reverberation time) vary widely from one hall to another, but the nature of the echo also varies depending upon the texture of the walls and furnishings in the hall. Some places have soft, subdued-sounding acoustics; others have hard, brittle acoustics (the kind that makes you want to whisper and walk softly).

As far as the recordist is concerned, the best recording location has a fairly long reverberation time and a soft acoustical character. It will come as little satisfaction to him to find that this is a rare combination of virtues, because the padded seats and heavy drapes that soften the echoes also swallow them up, shortening the reverberation time. The best type of auditorium for recording seems to be one that is very large and long, with plenty of soft padded seats, plenty of intricate plaster moldings around the walls, and irregular wall surfaces. It usually turns out that old vaudeville halls and ornate churches are ideal for recording — while the decorations soften the sound, the immensity of the buildings lengthens the reverberation time.

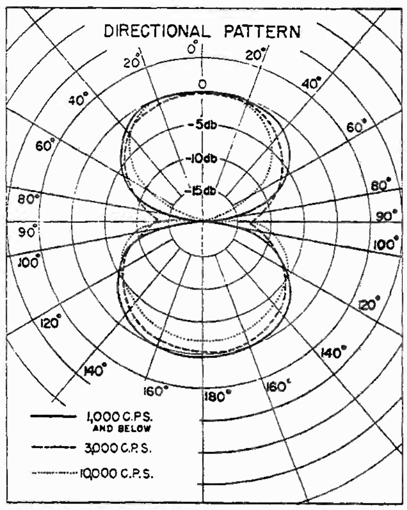
Few of us can pick our recording location, however, having rather to record where performances take place. We have to make the best of less-than-ideal acoustical conditions that are likely to exist in the local high-school auditorium or gymnasium. In such halls, the usual problems are a little too much echo and an acoustic quality that is too hard.

At this point it is worthwhile to analyze this business of sound reflection. A flat, hard, polished surface behaves much like a mirror in that it reflects all the sound waves directed at it. Rough, soft surfaces absorb some of the sound that strikes them, and since high frequencies consist of short-wavelength vibrations, these are lost more readily than lower tones.

The frequency of a tone also determines how easily it is deflected in its path of travel. An object intervening between the sound source and the microphone will completely block all the direct high frequencies, but tones having longer wavelengths will simply flow around the object and continue on to the mike. For this reason, *all instruments to be recorded should be visible at the microphone site.*

This frequency-selective absorption is what accounts for the soft sound of some auditoriums; the drapes and padded chairs and gingerbread adornments swallow up and diffuse highs more readily than the lower ranges. Add an audience to any auditorum, though, and you have one of the most effective sound absorbents known. And since reflected high frequencies travel in straight lines from the walls just as they do from the original sound source, whereas the audience reflects few high tones, the "softest" reflections in a live auditorium will come from the audience itself.

Remembering how directional microphones behave, we can begin to see some solution to the problem of ob-



Pattern for a bidirectional microphone.

taining the right combination of direct sound and reverberation.

Let's take a specific example of a knotty recording problem and see how these principles might be applied. The occasion is a public performance of a community orchestra which we feel is good enough to commit to tape. The performance is to be given in the high-school gymnasium, which has typically hard acoustics and a fairly long reverberation time when empty. The major work on the program is Berlioz's *Symphonie Fantastique* (this is an unusual community orchestra), so this is the work to which the miking will be tailored.

Continued on page 47

Readers' Forum

Gentlemen:
Stromberg-Carlson appreciates very much any reference to its products, as, for example, the reference to our speaker housing having a folded path for sound travel in AUDIOCRAFT for October 1956, page 22. We are sorry to note, however, the manner in which you use "Acoustical Labyrinth". The term "Acoustical Labyrinth" is not a name for a speaker housing as you imply in your article, but is our trademark for "speaker housings having a folded path for sound travel". See, for example, U. S. Trademark Registration No. 329,-319. Therefore, a proper usage would have been "'Acoustical Labyrinth', a speaker housing having a folded path for sound travel." In view of the trademark nature of "Acoustical Labyrinth", it is our request that you write "Acoustical Labyrinth" with initial capital letters and quotation marks, together with the generic name "a speaker housing having a folded path for sound travel", or otherwise indicate the trademark nature of "Acoustical Labyrinth" so that our trademark will not pass into the general English language as the common or generic name of the speaker housing, thereby depriving the public of the assurance of uniform high quality that the mark "Acoustical Labyrinth" indicates.

We would appreciate it if you would make a notation in some appropriate place in the next issue of the Magazine AUDIOCRAFT stating that "Acoustical Labyrinth" is a trademark of Stromberg-Carlson and thereby indicates a product manufactured by Stromberg-Carlson, a division of General Dynamics Corporation, and that the common or generic name of the particular type of device that we manufacture is a "speaker housing having a folded path for sound travel", which is a term that may be used by anyone.

In closing, let me thank you again for your interest in our product.

J. W. Whitesel
Patent Department
Stromberg-Carlson Company
Rochester, N. Y.

Gentlemen:
The government publishes a technical manual, TM 11-662, *Basic Theory and Application of Electron Tubes*, for $1.00 that, in my opinion, is outstandingly good. It is simply written for the beginner and gives a thorough treatment

Continued on page 45

EDITORIAL

WE'VE just received a report on an informal survey made at the New York audio show last September. Paul Penfield, Jr., our major contributor on transistor matters, asked representatives of each exhibiting manufacturer 1) if they had any products now utilizing transistors, and 2) if they had any *definite* plans for such products in the near future. The following comments are those of Mr. Penfield:

"Besides transistorized hearing aids, portable radios, broadcast equipment, and public-address amplifiers, the survey turned up a few *bona fide* hi-fi products.

"The Fisher TR-1 all-transistor preamp was the best known of these, having been released some months before the show. This unit has, indeed, become so popular that Fisher has discontinued the corresponding vacuum-tube model. General Electric exhibited an experimental hybrid preamp using one tube and one transistor. Presto's line of tape recorders featured transistors in the playback amplifier input stage; in this application, a heavy and costly input transformer is eliminated. Magnecord indicated that transistors would be applied in their equipment in the near future.

"Almost all manufacturers said that they were vitally interested in transistors, but not necessarily with respect to any particular product now.

"The conclusion from all this is that most manufacturers believe the transistor does not yet have quite enough advantages for hi-fi equipment to offset the higher initial cost. A major exception is Fisher, with a very successful preamp. As far as a few other manufacturers too are concerned, the transistor is of age. More can be expected to concur with this opinion soon."

ON October 3rd a concert of live and recorded music, entitled "Sound Reproduction", was given at Carnegie Hall under the joint auspices of Wharfedale Wireless Works, Ltd. and British Industries Corporation. G. A. Briggs served as Director and master of ceremonies; collaborating and operating the equipment was H. J. Leak. Performing artists were E. Power Biggs, organist; Morton Gould, conductor of a percussion ensemble with tap dancer Danny Daniels; and Ferrante and Teicher, duo pianists. They alternated with (and in some cases supplemented) disc and tape recordings reproduced by high-fidelity equipment. A few of the recordings were made especially for the program, although most were standard commercially available discs.

Equipment used for reproduction of the recorded parts of the concert was Wharfedale loudspeaker systems, Garrard turntables, Leak pickups and arms, Leak amplifiers, and an Ampex tape recorder, all available for home use.

Critics agreed as to the success of this well attended enterprise. Probably much of its success can be attributed to the clocklike precision and smoothness with which it was conducted. Mr. Briggs has presented several such demonstrations in England, and this was his second at Carnegie Hall; his experience was well applied and quite evident, when contrasted with the similar hi-fi concert in Hartford on October 19. — R.A.

Morton Gould conducts, Danny Daniels dances at the Carnegie Hall hi-fi concert.

by russell j tinkham

placement speaker stereo

IN the constant search for fidelity (i.e., realism) in sound reproduction, stereophony appears to be the next logical step forward after the hi-fi phonograph. Developments in this art since the Bell Labs experiments of 1933[1], and particularly within the past five years, have assured the music lover that this type of sound reproduction is now practical for the home as well as the theater. Excellent home systems are now within the reach of those with but modest means.

One of the basic problems is to arrange the loudspeakers within the listening enclosure — be it concert hall, theater, or living room — in such a manner that the reproduced sound closely approximates the original sound in its original environment. It goes without saying, at least theoretically, that exact reproduction can never be achieved except when such reproduction takes place in the original environment. But it is possible to achieve excellent and subjectively satisfying results with modern equipment in the home as well as the theater. This is true only if the recordings have been made correctly, and if the speakers have been properly disposed with respect to the reproducing room and the listener.

Let it be established that the final criterion is the subjective reaction of the listener himself; there is no other possible. Since people are not all alike, each has his own subjective reaction, depending on his past experiences. Fortunately, most of us fall into groups whose subjective reactions are sufficiently similar that we are more or less satisfied with the same things. How unfortunate it would be if each of us wanted particular shades of red and green on his new two-tone automobile, instead of the standard red and green available at the factory! The question is, then, can a sound-reproducing system be put together which will give satisfaction to a great many people? The answer appears to be "yes".

Until recently, all our music-reproducing systems were single-channel. That is, all sound was recorded in a single groove on the record (some records by Cook, *et al.*, excepted) or over a single radio or TV channel, or on a single-channel tape. All the improvements in reality from the first Edison cylinders to the modern LP were achieved step by step in widening the frequency response; making the frequency response more uniform; and, finally, by reducing distortion and noise. In the LP discs issued since World War II we have come a long way toward realism. On a good system we can hear the rosin scrape of the violin bow, the breathy rush of air in the flute, and the groans of the string bass with its overtones. Sometimes we can distinguish the English horn from the oboe, a neat trick even at a concert. What's more, we can tell whether the artist is in the back of the room ("off-mike") or swallowing the mike for that intimate feeling. In other words, with modern single-channel systems we can get reasonably accurate reproduction of depth, dynamic range, and undistorted frequency range. But we have no way to obtain accurate information as to the location or size of the original source; our sound seems to come through a hole in the wall from another room.

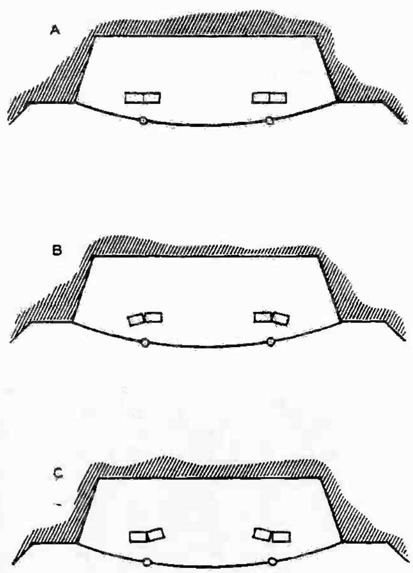


Fig. 1. Stereo speaker pairs; see text.

Attempts to remedy this hole-in-the-wall situation have been many and ingenious. They have ranged all the way from more than one speaker, fed simultaneously from the same source, to two speakers being faced diagonally *into* the corners of a room. The attempt is, of course, to "fill the room with sound". It works surprisingly well. But, to take an extreme example, compare the recording of a ping-pong game reproduced over a single-channel system of this sort with a stereophonic recording of the same ping-pong game reproduced over a stereophonic system. This is like comparing a black and white drug-store photo with a stereo-camera color slide.

This leads those of us who have heard good stereophony[2] to wonder how to situate the loudspeakers so as to achieve the best results. We must make one assumption: the recording technique employed by the engineers was correct to begin with.[3] If we plan to use two-channel (binaural) headphones to listen with, then the two microphones used to make the recording should have been spaced about six inches apart. But for the home loudspeaker system, the two microphones should have been spaced several feet apart. There is a constantly growing library of professionally recorded two-channel stereophonic tapes becoming available currently, with more still to be released, and it is with this in mind that we continue the discussion.

The Bell Labs experimenters of 1933 concluded that one should have an infinite number of loudspeakers disposed vertically and horizontally in a plane, each fed simultaneously from separate microphones and transmission systems, to achieve optimum results. Recognizing the economic impracticality of such

[1]"Acoustic Perspective; A Symposium." *Electrical Engineering*, LIII (Jan. 1934).

[2]Accurate stereophonic reproduction is possible only from two or more separate sound tracks recorded simultaneously from two or more microphones, through separate amplifiers to separately recorded tracks on tape or separate grooves on a record, and reproduced simultaneously through separate amplifiers and separate loudspeakers.

[3]R. J. Tinkham. "Binaural or Stereophonic?", *Audio Engineering* (Jan. 1953).

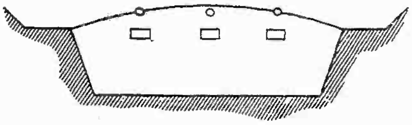
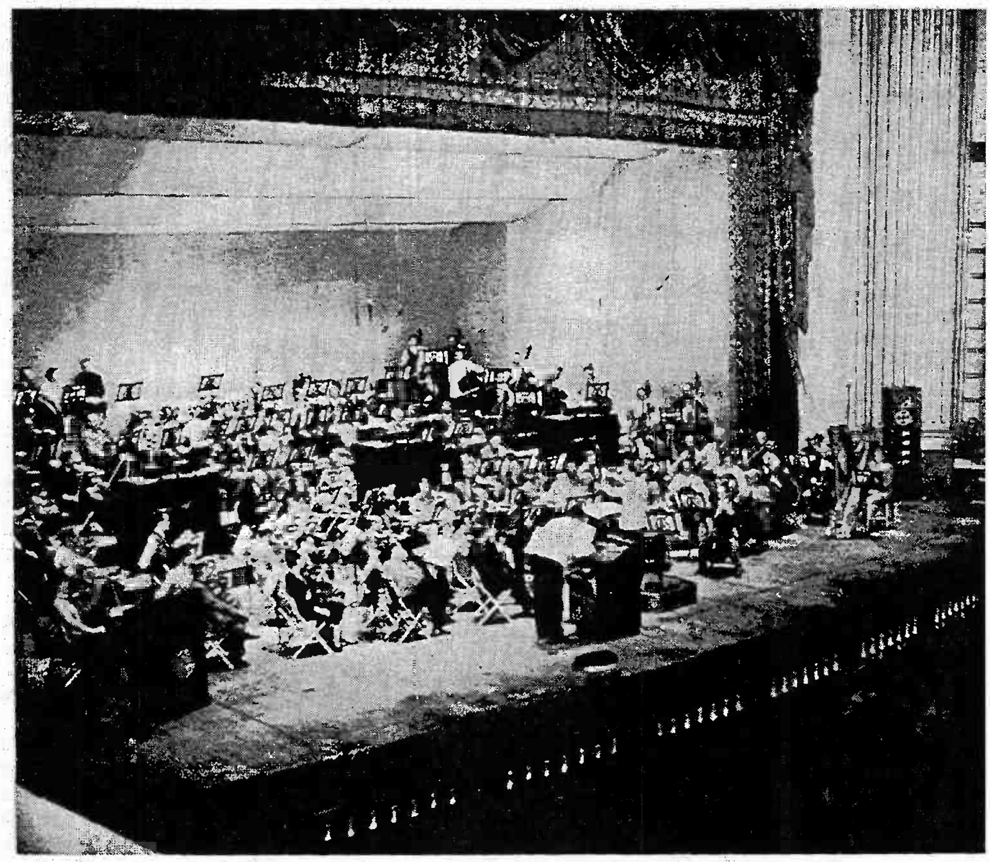


Fig. 2. Three-channel stereo speakers at San Francisco concert last March. Right: SF Symphony under Enrico Jorda rehearses Ussachevsky's Concerto for Tape Recorder and Symphony Orchestra.



a system, they went further and stated that good results in small rooms, and even in moderately large rooms, could be obtained with two channels.

During the past several years members of the Ampex sales and engineering staff have conducted a number of experiments in stereophony. The objective was to define a practical stereophonic reproducing system within the economic means of the home. Some of the earlier experiments were previously reported.[4] Experience in building CinemaScope theater systems and all the Todd-AO theater systems yielded more background. Within the past six months further experiments have produced additional data of interest.

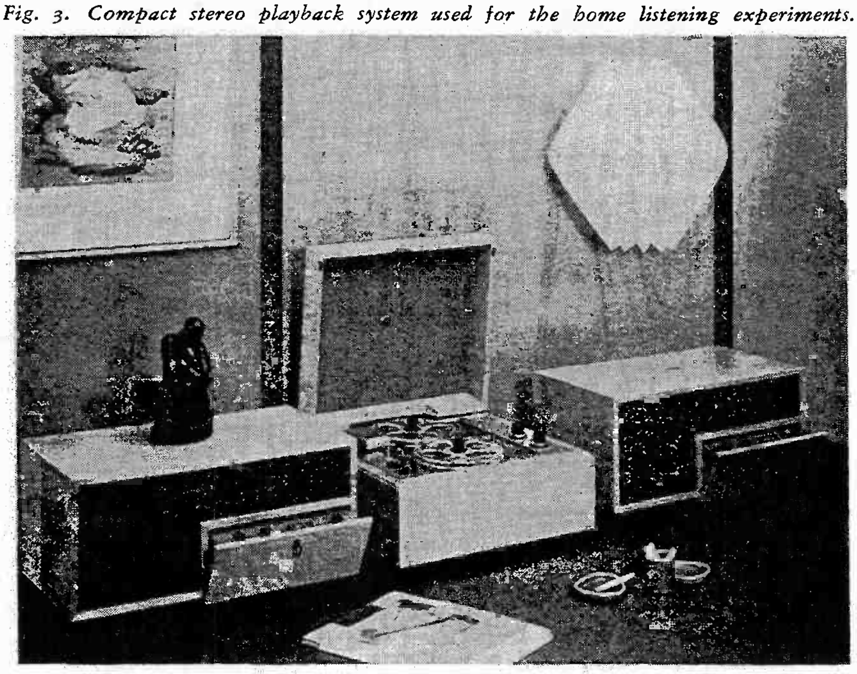
In February, 1956, I was privileged to co-operate with WGMS (Washington, D. C.), Fisher, Jensen, and Eugene Ormandy and the Philadelphia Orchestra, in recording and reproducing on-the-spot an evening concert to a full house at the Academy of Music, in Philadelphia.[5] A two-channel system was used here, consisting of Telefunken mikes, Fisher amplifiers, Ampex 350-2P recorder, and Jensen speakers. A pair of speakers was used for each channel, because single-speaker assemblies of the type used did not have sufficient power-handling capacity to equal the orchestral power in this same room. The pair on the left was fed from one channel and the pair on the right from the other.

At first, the faces of the speaker enclosures were placed in the same plane across the stage, as shown in Fig. 1A. This produced interference patterns from the closely spaced speakers in each pair, even though they were properly phased, which caused phase cancellations or dead spots. Moreover, coverage at the sides of the hall toward the front was not as good as desired. Tilting the two outer speakers toward the center, Fig. 1B, accentuated the interference patterns. The results were not natural. Tilting the two inner speakers toward the center and keeping the two outer speakers in a plane across the stage, as in Fig. 1C, reduced the interference between each pair, provided adequate power-handling ability, and sounded more natural. Furthermore, the sound distribution at the sides of the hall was enhanced, because some of the directionality of the speakers themselves was overcome. This was the arrangement used during the performance, and audience comments in the lobby following the concert were favorable.

In March 1956, the San Francisco Symphony Association and Ampex sponsored an evening concert at the San Francisco Memorial Opera House, with stereophonic reproduction using three channels (three were used because of the wide stage and wide, shallow auditorium). The system consisted of Altec "lipstick" mikes, Ampex Model 300-3R special (½-inch tape at 30 ips) recorder, three Ampex theater amplifiers, and three Ampex theater speakers. Equalization was used for the auditorium where the performance took place, and was the result of a consensus of the musical judgments of

[4] R. J. Tinkham. *Op. cit.*
[5] Adrian Siegel. "Stereo Soundorama", *Tape Recording Magazine* (Apr. 1956).

Fig. 3. Compact stereo playback system used for the home listening experiments.



Mr. Jorda, the conductor, and others. The orchestra's sound level was measured in the auditorium with a sound-level meter, and reproduced sound was adjusted to the same level. Frequency response of the total system, limited only by the loudspeakers, was compensated to be flat from 50 to 12,000 cps. Tape hiss was better than 65 db below full output.

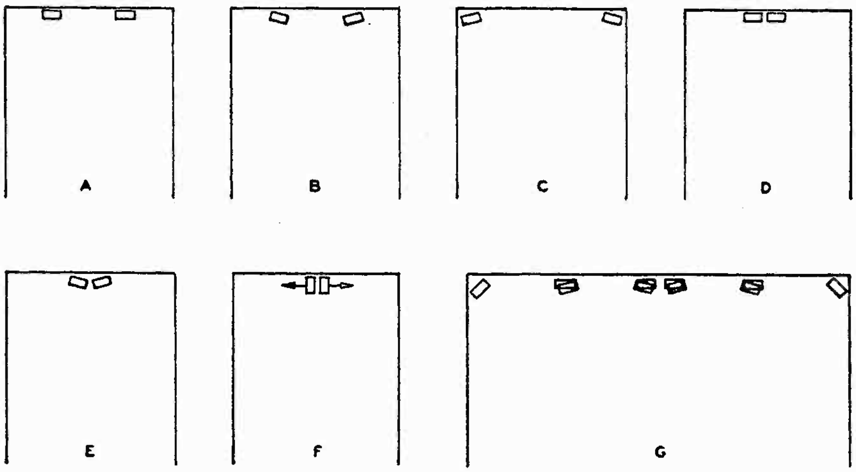


Fig. 4. Experimental positions for stereo speakers. Placement E was judged best.

The three speakers were of the horn-loaded woofer and horn-loaded tweeter type used in CinemaScope and Todd-AO theater applications. They were set back of the violins in the midst of the orchestra, in a plane across the stage, as shown in Fig. 2. The proof of the naturalness of the system is best illustrated by a sequence of events relating to the first number on the program. The orchestra had come on stage, Mr. Jorda walked to the podium, raised his baton, and the orchestra started. About halfway through the number, members of the orchestra laid down their instruments — yet the music continued to be heard without an appreciable break. The audience applauded spontaneously in compliment to the technical excellence of the similarity between the recording and the orchestra itself.

It was not until later, however, that the full impact of what they'd really heard became apparent. The commentator explained, after intermission, that the orchestra had faked their part of the first number; the tape recorder had originated the entire selection, which had been recorded at the rehearsal two days before. The audience gave an audible gasp, a laugh at themselves, and applauded again.

A third group of experiments relating to this general subject was conducted during the months of April and May 1956. The Ampex 612 two-channel home stereo system, with its associated movable speakers, had been on sale for some months at that time. Letters were received from many customers requesting more specific information on the best speaker placement, or taking issue with suggestions that were made to them. And some people didn't like the idea of all the bits and pieces necessary to this flexible system. Moreover, many music lovers were used to having only one cabinet to arrange with their furniture, and weren't given to experimenting on their own; they just liked to listen, not push the furniture around. Was it possible to put the whole arrangement into one cabinet in order to end some of the arguments? Was there any common denominator?

A series of experiments was organized in an attempt to discover a common denominator for the home system. These tests involved subjective reactions to stereo speaker placement in several different types of home music rooms. The Ampex 612 system and its associated 620 speakers, Fig. 3, were used to make the tests. Only commercially available stereo tapes were used in the experiments. A carefully selected group of six people, three men and three women, of widely varying temperaments, musical experience, and taste were used. Represented were a hepcat, a long-hair, a reasonably accomplished musician, a really truly golden ear, and a couple of average, normal human beings who liked music, and at first didn't even know what was happening to them. The rooms used included a typical rumpus room with asphalt-tile floor; a wall-to-wall carpeted, overstuffed living room of average dimensions; a sparsely furnished living room; a huge 18 by 42-foot living room; and a side patio out in the open.

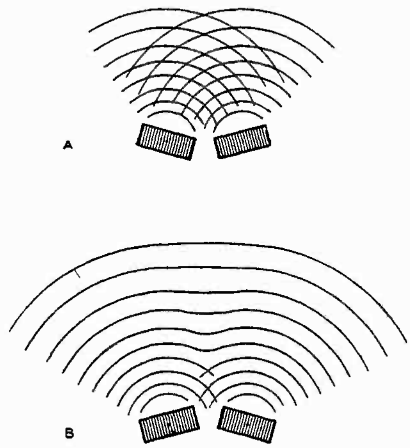


Fig. 5. For monaural sources, divergent speaker positioning was most pleasing.

The equipment used made it possible to switch from two-channel stereo reproduction to a single-channel composite of the same selection reproduced simultaneously over both speakers. A note should be made here regarding some of the phenomena attendant to reproducing music in such a manner. If one listens to the two speakers reproducing the same signal, assuming they are electrically phased and are delivering the same acoustic output, and if one is close to them, the sound will appear to be originating at a point midway between the two speakers (see Figs. 4A and 4D). An excellent example of this is the new Motorola TV model with "stereosonic" sound; the sound appears to issue from the picture itself, since the picture tube is flanked on each side by a speaker being fed from the same single-channel signal. This is an aural illusion. It is caused by the interference patterns similar to those experienced at Philadelphia, mentioned previously. When the speakers, still reproducing the same single-channel signal, are faced slightly outward, as in Figs. 4B and 4E, this interference is reduced. There is instead a welcome enhancement of a single-channel sound. See Figs. 5A and 5B.

If one suddenly switches from single-channel reproduction over these two speakers to stereo reproduction, the change may or may not be apparent. Whether it is noticeable or not depends on several related factors, assuming equivalent microphone placement: a) spacing between the speakers, b) subtended angle from listener to speakers, c) room-reverberation time, d) character of sound, and e) subjective listening experience.

If the speakers themselves are widely spaced, as in Fig. 4A or more particularly Fig. 4C, the change between one channel reproduced over both speakers to two-channel stereo is readily apparent to nearly everyone, including the inexperienced listener. This is the "dramatic" type of presentation which is useful in showing the inexperienced listener "just what stereophonic sound really is".

If one is standing close to the speakers so that the subtended angle between him and the speakers is, say, 45° to 90° or better, the shift from "single" to "stereo" is also noticeable.

Continued on page 41

TRANSISTORS *in Audio Circuits*

by PAUL PENFIELD, JR.

IIIb: Junction Transistor Characteristics

Other Curve Families

In part IIIa we discussed the collector curve family of the transistor operated in a grounded-emitter circuit. This form of data is most useful for the grounded-emitter circuit, since it shows the output (collector voltage and current) as a function of the input (base current). There are several other ways of presenting the same data not quite so clearly; one common way is to plot collector current *vs.* collector voltage for constant emitter-current lines — the so-called grounded-base collector family. See Fig. 13. The grounded-emitter curves described here are, however, the easiest to interpret.

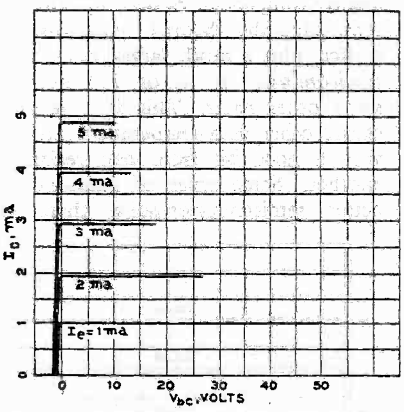


Fig. 13. Grounded-base collector curves.

The one set of curves discussed does not exhaust the useful information available about the transistor that can be obtained from volt-ampere characteristics. No mention was made of the base-to-emitter voltage all this time. Information about this variable is normally presented in either of two ways.

The first, Fig. 14, is a plot of base voltage *vs.* base current, with the collector voltage used as the running parameter. Since these lines all fall very close together normally, one such line is sufficient for practical purposes, at least with low-power transistors.

The second way is to plot collector current *vs.* collector voltage, using base-to-emitter voltage as the running parameter. This is shown in Fig. 15.

Either of these, together with Fig. 6, completely specify the volt-ampere characteristics of the regions of operation they cover. Any other set of graphs may be derived, more or less accurately, from either of these two sets.

Since Fig. 15 and Fig. 6 have the same co-ordinates, the two can be superimposed, and sometimes are. Fig. 16 is the result, limited to the region within the maximum ratings.

Often it is convenient to operate transistors very close to zero current and zero voltage. This corresponds to a region very near the origin of Fig. 6. Fig. 17 shows the low-power region blown up in size for close inspection. Whenever it is necessary to operate a transistor with a minimum of battery power, it is advisable to consult a graph such as Fig. 17, if one is available.

Variations in Curves

The published characteristic curves may not fit any given transistor for any or all of three reasons. First, there are differences between various transistors of the same type number. Second, the temperature of the collector junction is important. Third, any given transistor will age in use, and its characteristics will change.

Not all transistors of the same type number are the same. Minor differences in manufacture result in considerable variations in characteristics from one transistor to another. They are functions of β, which is $\frac{\alpha}{1-\alpha}$. Since α is very close to one, practically insignificant changes in α from one transistor to the next produce large fluctuations in β. For this reason, the graphs published by manufacturers for typical transistors should be used for general guidance only. Actual transistors taken off the shelf will undoubtedly produce curves that vary considerably from those published.

The electrical volt-ampere characteristics for any given unit will change when external conditions change. The most important condition of this sort is the temperature of the collector junction. Less important conditions are humidity (negligible in effect, except with open experimental transistors), illumination (negligible with metal-encased units), and electromagnetic fields (sometimes important when transistors are used near broadcast stations, induction heaters, etc.).

Temperature changes the collector family through action on the cutoff current I_{co}. Remember that I_{co} is strongly temperature-dependent, doubling every 10° C. or so.

Upon an increase in temperature the bottom line in Fig. 10, representing I_{co}, will rise. The line denoting zero base current will rise just $(\beta + 1)$ times as much, and all lines of constant base current will go up roughly the same amount — that is, $(\beta + 1)$. Thus a large increase in I_{co} will, in general, push any point on the family up roughly β times as far. In some cases this shifting of the curves is quite important, and must be taken into account.

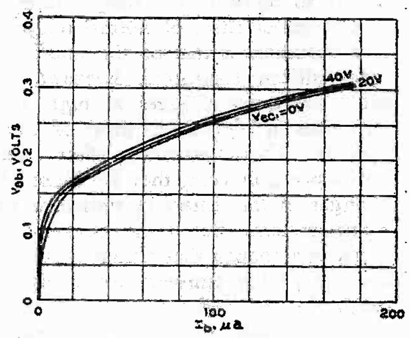


Fig. 14. Curves of the emitter family.

Changes in Fig. 13 (the grounded-base collector family) due to temperature shifts will be slight. The changes that occur with temperature in any other set of curves can be figured out on the basis of the physical picture of the transistor. For example, in Fig. 13, all the lines of constant emitter current rise by an amount just equal (or nearly so) to the change in I_{co}.

A secondary effect of high temperatures is of interest also. The curve

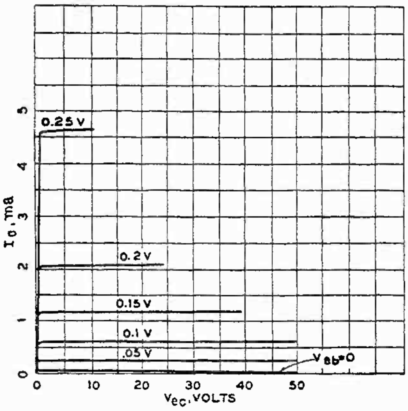


Fig. 15. V_{eb} is running parameter here.

families do not shift together as a body exactly, but rather some rearrangement of the lines occurs. At elevated temperatures the slope of the lines will become more pronounced, and the spacing between the lines will change slightly. These effects, however, are usually less important in circuit design than the changes caused by increased cutoff current.

The fact that transistor curve families are displaced by temperature changes indicates that germanium transistors, with their high normal cutoff current, will exhibit more temperature effect than silicon transistors. This is true, so far as this analysis goes. Silicon *is* useful to much higher temperatures than germanium and some of the compound semiconductors are expected also to be useful to much higher temperatures.

The major effect of normal temperature variations is that on the cutoff current; silicon transistors, however, exhibit parameter changes at high temperatures in addition to those of cutoff current. These secondary effects need not concern us here; they are caused by changes in the material's resistance and other factors. But these changes are often bad enough that silicon transistors are not used at temperatures as high as might be expected.

As a matter of interest, transistors operate very well at low temperatures. Except for mechanical failures (from stresses in the housing or leads, for example), transistors will give good results immersed in liquid nitrogen. For any practical purposes low-temperature problems can be ignored. The gain of the transistor (β) does drop off below 0° C., and transistor action does stop altogether at a *very* low temperature, but in general these problems are not too serious.

Aside from temperature effects, transistor characteristics change with age. Transistors do not have an infinite life span, but deteriorate in use. This deterioration takes the form of an increased cutoff current, lowered current gain, and more erratic operation. Storage or operation at high temperatures hastens old-age failures.

The best modern transistors can be reasonably expected to function many years if not run at too high temperatures. The worst, however, are no better than vacuum tubes in this respect, and cannot be expected to last as long as other components.

Transistor Noise

The smallest signal that a transistor can amplify usefully is determined by the noise generated in the transistor. If the input signal is reduced continuously, sooner or later the inherent noise level of the transistor will become louder than the signal; this lower limit cannot be removed. From this standpoint, noise

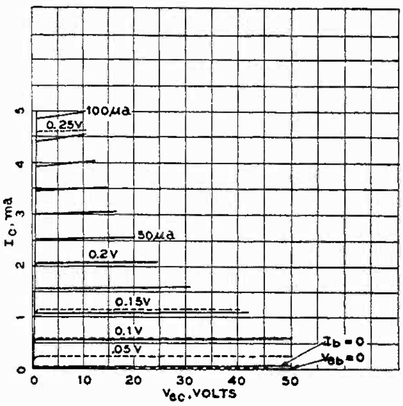


Fig. 16. Collector curve family, with both V_{eb} and I_b as running parameters.

is the electrical engineer's worst enemy.

In audio work, noise forms an incessant *hiss* underneath the program. In television, noise appears as a speckled pattern known as *snow*. In precise scientific work, the limits of accuracy are always determined ultimately by noise.

When transistors are used to amplify small signals, care must be taken to provide for low-noise operation. Some of this responsibility rests with the transistor manufacturer (some transistor types are made especially for low noise), and some with the circuit designer (transistor noise depends on the method of operation, bias values, etc.). The designer's part in noise reduction will be covered in a future installment.

The absolute low limit for noise is so-called *thermal noise*, or *Johnson noise*. This is shown by resistors even when no current passes through them. The higher the temperature, the higher is this value of noise. Thermal noise is caused by electrons in the resistor, transistor, or whatever, bouncing around. The amount of thermal noise present depends only on the temperature, and not on the type of resistor involved.

Any noise may be thought of as being a sum of a large number of randomly varying sine waves, each of a different frequency. These sine waves cover the entire frequency range from direct current on up to infinite frequency. It is convenient to consider the relative magnitudes of the various frequency components — that is, the so-called frequency spectrum of the noise.

Thermal noise has the characteristic that the component at any one frequency is, on the average, just as strong as the component at any other frequency. That is, it has a so-called *flat* spectrum, independent of frequency. Such a spectrum is shown in Fig. 18. Other types of noise may or may not exhibit this behavior.

Thermal noise is actually quite low in magnitude. It is seldom that circuit designers have to worry about it; usually other types of noise associated with the circuits are far more important.

When current flows through the resistor we used a moment ago as an example, then it is found that the total noise voltage across the resistor has a steady component caused by the current, plus the thermal noise discussed earlier, plus a much larger noise called *semiconductor noise*, or *1/f noise*, or some other name. This is a result of causes other than temperature, and will vary a good bit from one resistor to another. Some types of resistors have better semiconductor-noise characteris-

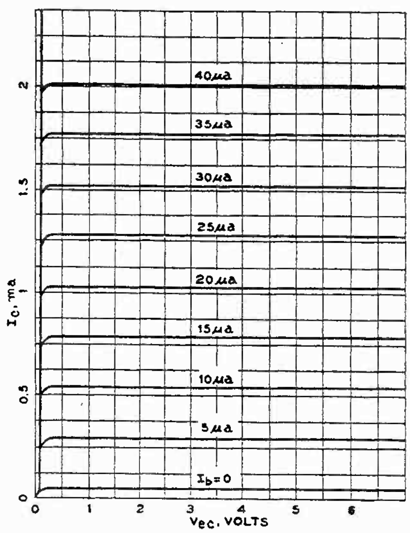


Fig. 17. The low-power collector family.

tics than others, although all have the same thermal-noise characteristics.

Semiconductor noise rises at low frequencies, and drops off to nothing at high frequencies. Over a large range it is approximately proportional to $1/f$, where f is the frequency, and so it is called $1/f$ noise. At high frequencies it becomes negligible compared with the thermal noise and other noises. Fig. 19 shows how this noise varies with frequency.

So much for noise in a resistor. How about transistor noise?

Modern junction transistors vary in their noise characteristics. The best of them are superior to fine-quality vacuum tubes. The worst transistors (that is, those with the highest noise) have a $1/f$ noise spectrum over the entire audio frequency range, because of semiconductor noise.

Transistor noise level at any given frequency is measured as compared to thermal noise in a given circuit. It is then expressed in decibel notation as so many db above thermal noise at some frequency. The *noise figure* of a transistor is the value of the transistor noise in some standard circuit above the thermal noise present, customarily measured at 1,000 cps, right in the middle of the audio range.

It has been found that the better low-noise transistors have noise characteristics similar to those shown in Fig. 20 and Fig. 21. That is, at the high-frequency end, the noise has a flat spectrum or a rising spectrum, and at the low-frequency end the semiconductor noise becomes important. It can be seen, incidentally, by comparing Figs. 20 and 21 that, although Fig. 21 has over the audio range less noise, their rated noise figures will be the same, since the noise figure is measured at 1,000 cps.

This fact has brought criticism to the customary definition of noise figure for a transistor. Some manufacturers are trying a lower frequency for the spot

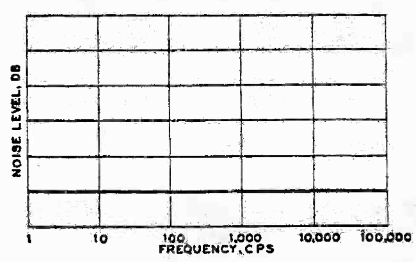


Fig. 18. A flat noise characteristic.

figure, and some are giving an average figure over the audio range, or other range of interest. Most, however, still use the 1,000-cps spot figure, and we will use it here.

The best low-noise transistors have a spot figure of 4 db, sometimes less. That is 4 db above thermal noise— really quite small in comparison with normal audio signals. The best vacuum tubes fall within this range also, although a strict comparison is not possible between the two.

The subject of transistor noise gets very complicated when we try to explain it on the basis of a physical picture. Even the experts do not agree on either the physical picture or the electrical equivalent circuit to represent transistor noise. The best we can do here is to accept the noise figure as presented by the manufacturer, or as measured, and try to design our circuits to reduce the effect as much as possible. Biasing for

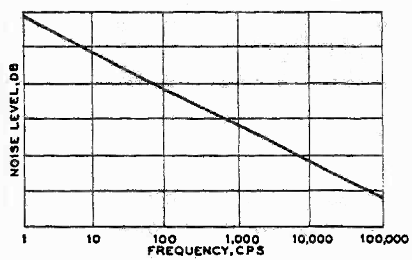


Fig. 19. Pure 1/f semiconductor noise.

least noise will be covered in a later installment.

High-Frequency Limitations

Another inherent limitation of transistors, besides the nonlinearities, the noise, and the temperature effects, is the high-frequency response.

If a transistor is set to amplifying a sine wave of some frequency, and this frequency keeps rising, the transistor's amplification will eventually begin to decrease. This is caused by a number of factors working together, of which the base width and the collector capacitance are the most important. Junction transistors available today often have serious limitations in high-frequency response, and in high-fidelity work this is sometimes troublesome.

Holes injected into the base from the emitter of a p-n-p junction transistor do not travel across the base instantaneously, but rather tend to drift across under the action of the collector bias voltage. This may take a few ten-millionths of a second or more, depending on just how thick the transistor base is. It should be clear that if this *transit time* is longer than one cycle of the frequency of interest, that frequency is not going to be amplified very well. So it is that the base width limits the high-frequency response of a transistor.

Another factor is also at work. Often the associated circuit appears to the

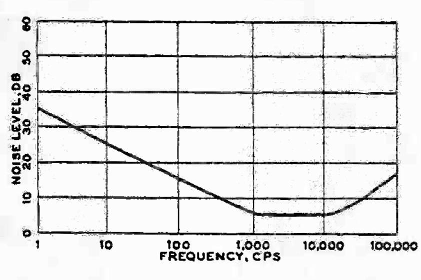


Fig. 20. Noise in fairly good transistor.

collector junction as a rather high resistance. The collector region, being close to the base region but separated by the junction, tends to act like a capacitor, and, in conjuction with the high external resistance, tends to smooth out fast changes in the collector voltage, the same way *any* capacitor will. In simple terms, this merely means that the high-frequency response will be diminished because of the *collector capacitance.*

The reader is cautioned against imagining a little capacitor as existing inside the transistor near the junction. Although the behavior is nearly identical to an actual capacitor, the size of the *equivalent capacitance* varies with such factors as temperature, impressed voltage, and so on. Under any given combination of such conditions, however, the collector capacitance can be determined, and under these conditions it explains the behavior of the transistor fairly well.

In part II we discussed several types of transistors which were designed for

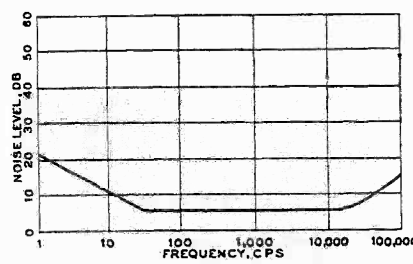


Fig. 21. Noise in very good transistor.

high-frequency work. Most involved reduced base width, or lowered collector capacitance, or both. This discussion should point up why these two factors, along with various other minor ones, are important in determining the high-frequency response.

If a transistor is operated common-base, the current gain, remember, will be just α. The high-frequency effects can be taken into account by considering α as a function of frequency— close to 1 at DC and low frequencies, and dropping off at higher frequencies. The frequency at which α is down to 0.7 (or $\frac{1}{2}\sqrt{2}$), its low-frequency value, is called the *alpha cutoff frequency.* This is the best single number characterizing low-level transistor frequency response. Typical values are 1 Mc and up.

Unfortunately, when the transistor is operated in the grounded-emitter configuration (this is in practice the most common), the frequency falls off just about β times as fast, so the gain is down at a far lower frequency. For this reason, most junction transistors made today cannot be relied upon for operation at frequencies higher than about 100 Kc.

Lack of adequate high-frequency response can be a problem in designing wide-range equipment, such as high-fidelity amplifiers. This problem will be taken up again in the installment on distortion.

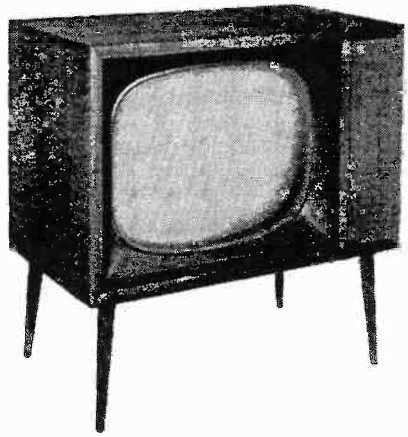
Continued on page 40

An AUDIOCRAFT *kit report*

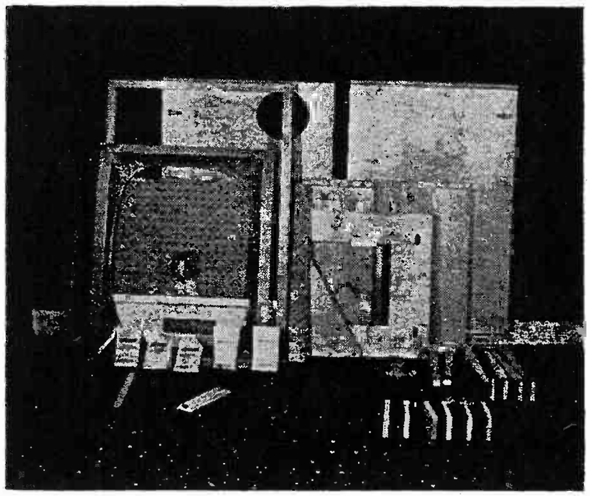
Cabinet Kit for Custom TV

by Charles Fowler

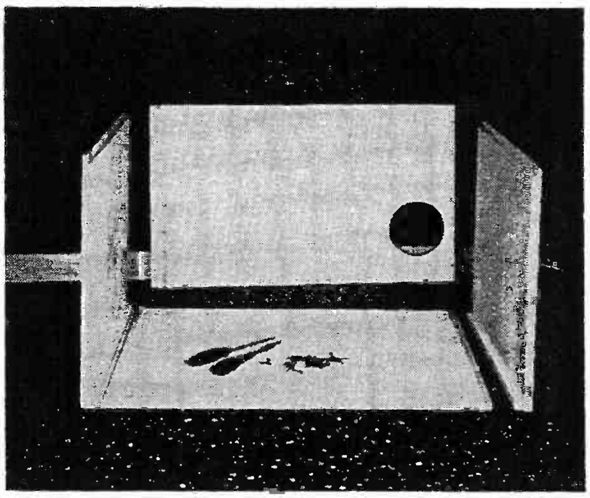
Few things are so unattractive — or so dangerous — around a house as a naked television chassis . . . and so often they balance precariously on the oldest card table available. Yet building them in, or designing and building a cabinet, is a tedious and difficult job for most of us. If it would be for you, then Conrac has an answer: a cabinet kit — practical and attractive! This is the picture story of one we assembled.



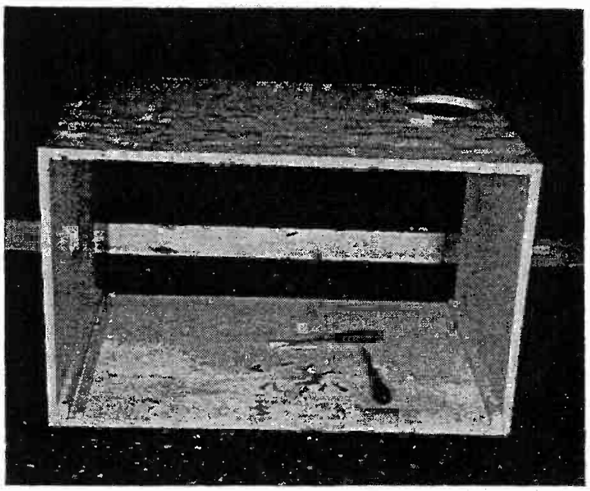
PHOTOS BY THE AUTHOR



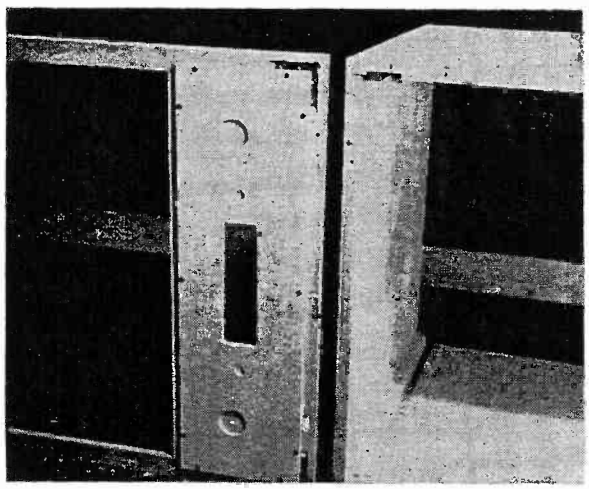
1. First step is to check the parts list, not only to see if everything is there (it was), but also to help identification by counting noses — in this case, screws. For example: "38 8 × 1¼". Whatever you have 38 of are the "8 × 1¼" units.



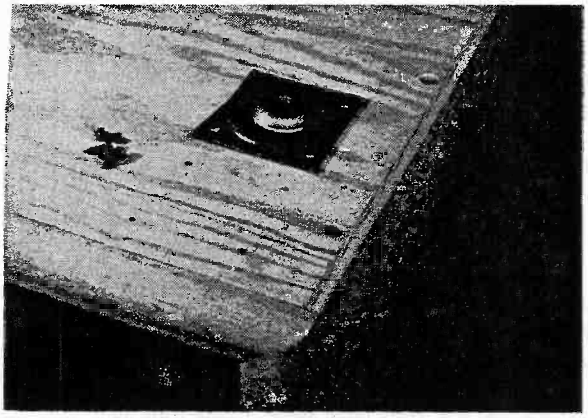
2. Put the top down on a rug, or other clean surface, and attach side pieces. Be sure all pieces face the same way; you can tell by the holes along one edge, which will be used later for front frame screws. Piece against wall is bottom.



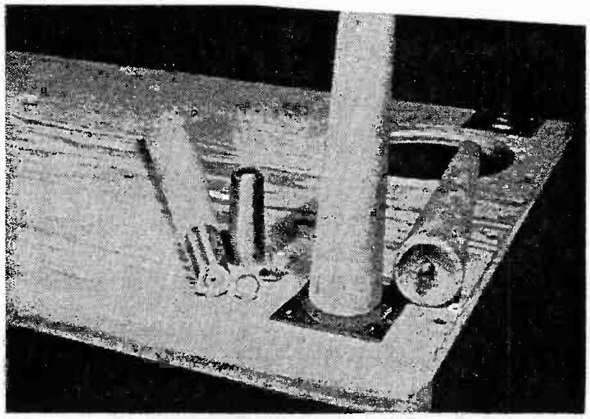
3. Attach bottom to sides. All holes are cleanly drilled and countersunk, and pilot holes are drilled in matching pieces; this is double assurance of correct assembly. Large hole in bottom (upper right in picture) is for an 8-inch speaker.



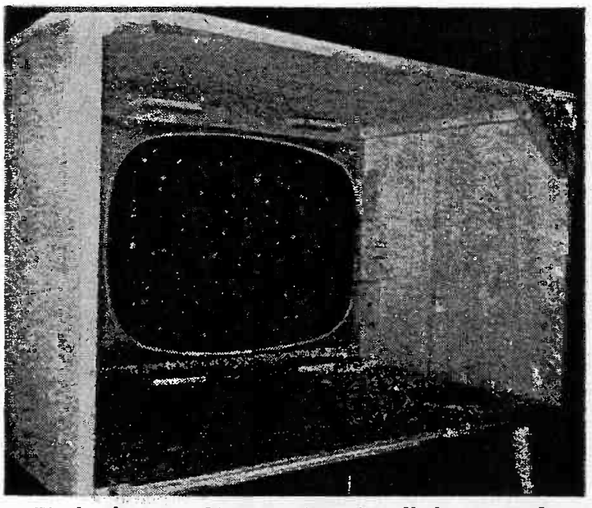
4. Next, attach the front frame. Careful here: it can go either way. The controls end goes opposite the speaker-hole end. Match dowels and dowel holes with care. For Fleetwood nonremote chassis, control holes are knocked out.



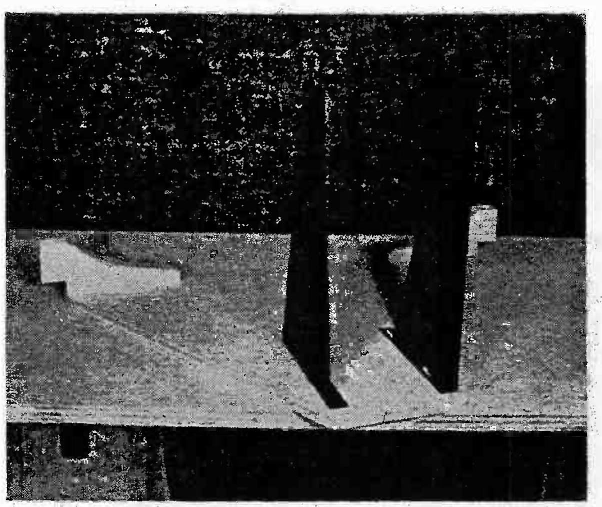
5. Attach the leg plates to the bottom. The legs must aim outward; the plates therefore tip slightly. Double check before screwing down; the cabinet is heavy and you will not want to up-end it to redo a leg plate! Note pilot holes.



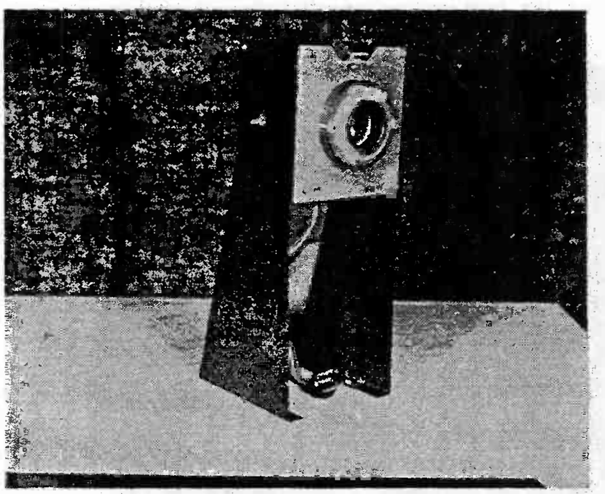
6. Leg assembly comprises several pieces. Slip the ferrule over end of leg and drive tack home before screwing leg to plate; it's easier to hammer the tack that way. If your kit is short a ferrule, look again. They nest quite snugly.



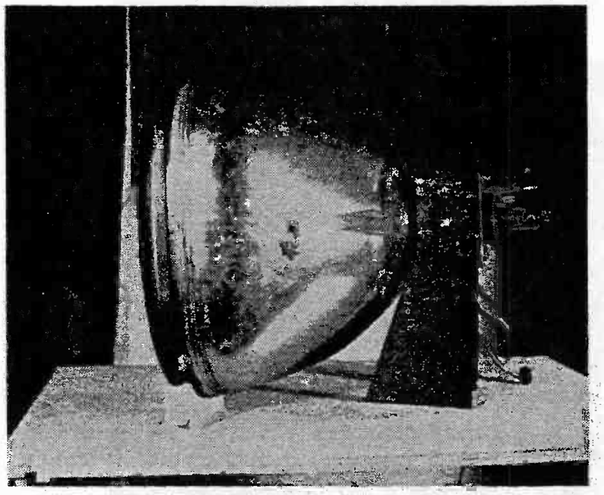
7. Final cabinet-making step is to install the corner brace (be sure you get the right screws) and then the safety glass and mask. Clean the inside surface of the safety glass before installation — and then keep your smudgy fingers off it!



8. Sometime soon you're going to have to start disassembling your TV set, including the tube — a job we always find terrifying. For the moment, mount the yoke supports and picture-tube supports on the triangular tube pallet.



9. Well, if you are still with us, your picture tube didn't implode and you must have remembered to ground the high voltage lead. So: mount this yoke assembly. Snug up, but do not tighten the two wing nuts. You need some play.



10. Hold your breath and skid the tube in place, neck through the yoke . . . gently, but gently. To the front, the forward ridge near the face should center on the wood blocks. Adjust so tube face is vertical, yoke against neck.

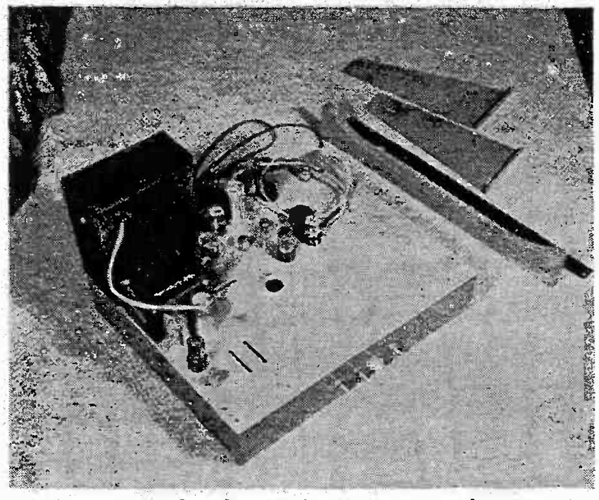
We are glad to report that we had no problems in putting together the Conrac TV cabinet kit. The instructions are clear and to the point, but read them carefully, and be sure of each step.

We fouled up on the corner brace; used the wrong screws and started to come through the top of the beautifully sanded top panel. Hence the warning in step 7.

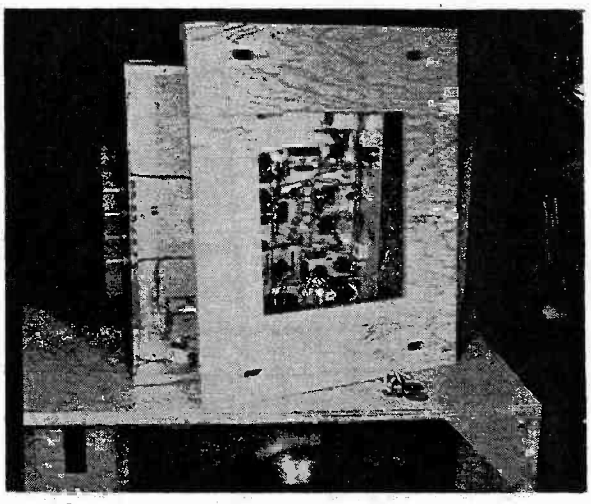
Some assemblers may have wire trouble. The leads on our set which run from the tube base, the yoke, and the high-voltage lead, were not long enough to reach the extra few inches required by the rearrangement of tube in relation to chassis. We cut the wires and spliced in short extra lengths, with an appropriate assortment of cuss words. Three days later we stopped in for odds and ends at our local radio supply house, and there on the counter was a complete assortment of extension cables! [*Sounds like a TV plot.* — ED.]

While the Conrac kit is designed primarily for the company's own TV tuners — they make both true remote and nonremote units — we can see no reason why the kit could not be used successfully with any make of television receiver. For the remote-control Fleetwood units, the little removable wood patch covers semifixed controls such as vertical hold and so forth. For nonremote units, precut knockouts are provided to match the main operating controls. For sets of other makes, some care may be necessary in drilling the right shaft holes. As can be seen from illustration 4, the large panel (with the cutout in the illustration) is screwed on from the back of the front frame. If you're working with a non-Conrac set, we'd suggest mounting the chassis on its pallet right after step 3. Then, with the chassis in place in the slots, the front frame can be trial-fitted and drilled as necessary. After you are certain everything fits, remove the chassis and proceed with step 4.

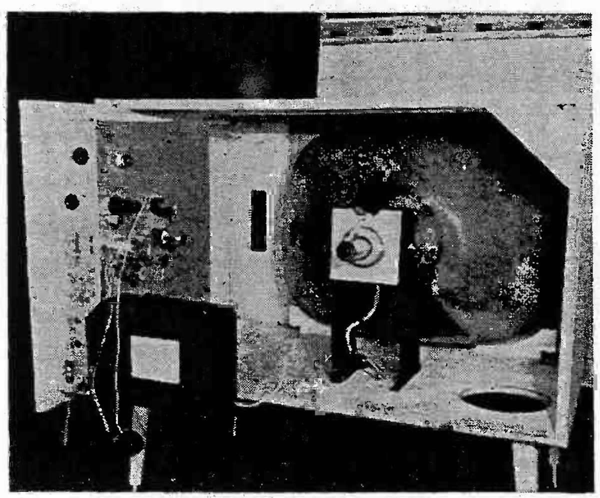
We have used a Fleetwood remote unit for nearly three years (with complete satisfaction, by the way). Now, at last — it is in a cabinet!



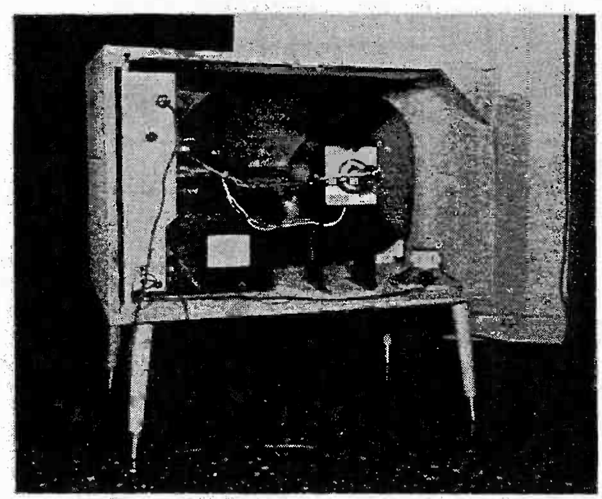
11. Now strip the chassis of unnecessary tube mounting brackets and hardware. For our Fleetwood, this meant removing the front tube and yoke mounting supports; other types of set may require somewhat different treatment.



12. There is a mounting pallet for the tube and another, shown above, for the chassis. — At this point, our habit of never throwing away anything paid off: we were finally able to find the four original chassis bolts and washers.



13. Slide the tube up against the mask; check to be sure it touches all the way around; tighten yoke screws and then screw down the tube pallet. Next, slide the chassis pallet into the two slots. It is a perfect fit but awkward to handle.



14. Everything done! Tube in, chassis in, speaker mounted. Check all wires, turn it on, tune it up; after final picture-tube and tuning adjustment have been made, the protective back piece can be added. Cabinet finish? Suit yourself!

by Dr. JOHN D. SEAGRAVE Part II

MINIMIZING PICKUP TRACKING ERROR

Approximating Tracking Error

Since we are interested in $\alpha/R = (\phi - \beta)/R$, it is tempting to set $\sin\phi \doteq \phi$ in Eqn. (12), and to ignore the small term in D^2. This procedure can lead to errors of a degree in angle or 5% in overhang. The results of this approximation have been worked out.[3] The results of a much better approximation given below are no more complex. We may make use of the well-known trigonometric identity $\sin(A + B) = \sin A \cos B + \cos A \sin B$, taking $A = \phi - \beta$ and $B = \beta$. Then

$$\sin\phi = \sin(\phi - \beta)\cos\beta + \cos(\phi - \beta)\sin\beta \doteq (\phi - \beta)\cos\beta + \sin\beta, \quad \ldots(13)$$

where now the very much better approximations $\sin(\phi - \beta) \doteq \phi - \beta$ and $\cos(\phi - \beta) \doteq 1$ can be made, since $\phi - \beta = \alpha$ will be only a few degrees for cases of interest. Transposing Eqn. (13), we then have

$$\alpha = \phi - \beta \doteq (\sin\phi - \sin\beta)/\cos\beta. \quad \ldots(14)$$

It will simplify the notation if we introduce

$$D_1 = D[\,1 - \tfrac{1}{2}(D/L)\,], \quad \ldots(15)$$

so that Eqn.(11) reads

$$\sin\phi = \tfrac{1}{2}(R/L) + D_1/L. \quad \ldots(16)$$

Note that this is not an approximation, merely a shorthand. When we have first solved for D_1, then

$$D \doteq D_1[\,1 + \tfrac{1}{2}(D_1/L)\,], \quad \ldots(17)$$

to a sufficient degree of approximation. We will also introduce the symbol $m = \alpha/R$ for the distortion index. Then from Eqns. (14) and (16),

$$m = \frac{\phi - \beta}{R} = \frac{57.3}{\cos\beta}\left(-\frac{1}{2L} + \frac{D_1}{R^2} - \frac{\sin\beta}{R}\right). \quad \ldots(18)$$

The number 57.3 is the factor required to convert from radians to degrees. The value of m at the endpoints R_1 and R_2 is given in Eqn. (18) by substituting R_1 or R_2 for R. The index has a minimum value (most negative) when $R = 2D_1/\sin\beta$, a result which is obtained by setting $dm/dR = 0$ and solving for R. It may be verified by direct substitution. The corresponding minimum value is

$$m_{min} = \frac{57.3}{\cos\beta}\left[\frac{1}{2L} - \frac{\sin^2\beta}{4D_1}\right]. \quad \ldots(19)$$

[3]B. B. Bauer, *Electronics* 18, (March, 1945) 110.

Optimum Solution

We have seen that $m_1 = m_2 = -m_{min}$ for the "optimum" solution. If $m_1 = m_2$, from Eqn. (18), we have the relation

$$\frac{D_1}{R_1^2} - \frac{\sin\beta}{R_1} = \frac{D_1}{R_2^2} - \frac{\sin\beta}{R_2},$$

which can be simplified to read

$$\sin\beta = D_1\left(\frac{R_1 + R_2}{R_1 R_2}\right). \quad \ldots(20)$$

But we also want $m_1 = -m_{min}$, so from Eqns. (18) and (19),

$$-\frac{1}{2L} + \frac{\sin^2\beta}{4D_1} = \frac{1}{2L} + \frac{D_1}{R_1^2} - \frac{\sin\beta}{R_1}. \quad \ldots(21)$$

From these two equations in $\sin\beta$ and D_1, we may solve for

$$\sin\beta = \frac{R_1 R_2 (R_1 + R_2)}{L[\,\tfrac{1}{4}(R_1 + R_2)^2 + R_1 R_2\,]} \quad \ldots(22)$$

and

$$D_1 = \frac{R_1^2 R_2^2}{L[\,\tfrac{1}{4}(R_1 + R_2)^2 + R_1 R_2\,]}. \quad \ldots(23)$$

This pair of equations, or either one together with Eqn. (20), represents the optimum solution for an arm of length L and limiting radii R_1 and R_2. The corresponding value of $m_{opt} = m_1 = m_2 = -m_{min}$, which may be taken as a standard of comparison for non-optimum designs, is

$$m_{opt} = \frac{57.3}{L\cos\beta}\left[\frac{1}{2} - \frac{1}{1 + \tfrac{1}{4}(R_1 + R_2)^2/(R_1 R_2)}\right] \quad \ldots(24)$$

Using the values $R_1 = 2.40$ in. and $R_2 = 5.70$ in. discussed above, we may evaluate Eqns. (22), (23), and (24) numerically for LP's as

$$\sin\beta_0 = 3.68/L; \quad \ldots(25a)$$
$$D_1 = 6.22/L; \quad \ldots(25b)$$
$$m_0 = 2.60/L\cos\beta. \quad \ldots(25c)$$

The actual overhang D is to be calculated from the value of D_1 given, by using Eqn. (17). For transcriptions, the numerical values in Eqn. (25) are: 5.38, 13.59, and 1.85, while for 78's they are 3.23, 4.46, and 4.18, respectively.

Eqns. (25) are illustrated in Fig. 4, where the optimum offset and overhang for LP's and the corresponding minimum distortion index are plotted as a function of arm length. For accurate numerical values, use Eqns. (25).

Best Overhang for Given Offset

In many cases it will not be possible to modify the offset angle β, but for β fixed, there is still a "best" value of D which leads to a maximum distortion index that is, of course, larger than m_0, but the smallest it can be for the given nonoptimum value of β. The procedure is similar to that employed above,

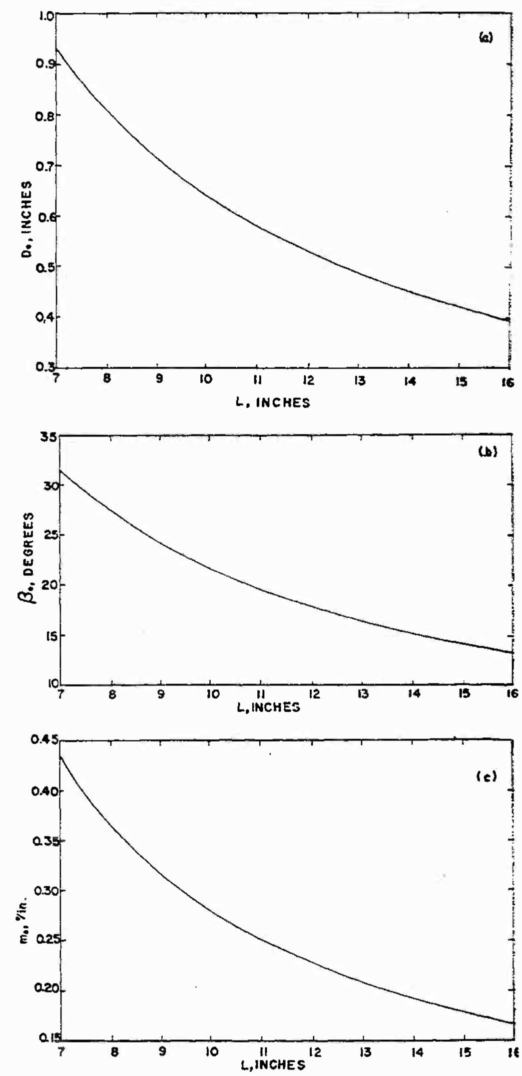


Fig. 4 Optimum design parameters for LP's as a function of arm length. a) optimum overhang D_0; b) optimum offset angle β_0; c) minimum distortion index m_0.

and will not be carried out in detail. A complication arises in that the form of the solution depends on how the given value of β compares with β_0 and β_1. β_0 is the optimum value given by Eqns. (22) or (25), and β_1 is a special value for which the point of minimum m occurs at R_1, that is, for $m_1 = m_{min}$.

It is given by

$$\sin\beta_1 = \frac{R_1}{L[1 - \tfrac{1}{2}(1 - R_1/R_2)^2]} = \frac{2.88}{L}, \quad \ldots(26)$$

with the previous values for R_1 and R_2. For transcriptions and 78's, $L\sin\beta_1$ becomes 4.32 and 2.42, respectively.

If β is greater than β_0, we must set $m_1 = -m_{min}$, and

$$D_1 = \tfrac{1}{2}R_1\{[(\sin\beta - R_1/L)^2 + \sin^2\beta]^{\frac{1}{2}} + (\sin\beta - R_1/L)\}. \quad \ldots(27a)$$

But if β lies between β_1 and β_0, we set $m_2 = -m_{min}$, and

$$D_1 = \tfrac{1}{2}R_2\ \{[(R_2/L - \sin\beta)^2 + \sin^2\beta]^{\frac{1}{2}} - (R_2/L - \sin\beta)\}. \quad \ldots(27b)$$

Finally, if β is less than β_1, we take $m_2 = -m_1$, and

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

As before, D is obtained from D_1 using Eqn. (17). These results are given elsewhere[4] together with best values of β when D is fixed. D and β of Ref. 4 should be replaced by D_1 and $\sin\beta$ for a better approximation.

Commercial Examples

In this section four representative commercial arms will be analyzed numerically to illustrate the use of the equations developed. The values of L and β used are from Ref. 1, and may differ from those appropriate to a particular unit due to production variations, changes, or numerical errors.

GE A1-500. For this arm, $L = 8.72$ in., $\beta = 17.5°$. From Eqn. (25a), $\beta_0 = 25.0°$, and from Eqn. (26), $\beta_1 = 19.3°$. Since β is less than β_1, we know that for this arm $m = 0$ only once, and the best placement is $m_2 = -m_1$, corresponding to Eqn. (27c), which gives $D_1 = 0.308$ in., or $D = 0.314$ when Eqn. (17) is used to obtain D from D_1. The situation is illustrated in Fig. 5a. The maximum distortion index is given by evaluating Eqn. (18) at either R_1 or R_2. $m_{max} = 0.85°$/in. (numerically). If the same setting is used for 78-rpm records, m will drop to -0.99 [Eqn.(19)] and come back to -0.86 at R_1 (78). Because of the record-speed factor, the distortion will not be as serious on 78's, however. Also shown in Fig. 5a is the performance to be expected with optimum offset and overhang for this length of arm. In that case $m_{max} = 0.33°$/in. for LP's.

[4] B. B. Bauer, *Electronics* 22, (June, 1949) 87.

GE A1-501. This is the "transcription" version of the above arm, with $L = 12$ in. and $\beta = 19°$. For 16-inch transcriptions the performance is similar to that of the smaller version on LP's, since for transcriptions $\beta_0 = 26.6°$, and $\beta_1 = 21.1°$. For 12-inch LP's, however, $\beta_0 = 17.8°$, so that β is greater than β_0 and we must use Eqn. (27a) to determine D_1 and hence D. The result is $D = 0.584$ in., and $m_1 = -m_{min} = 0.30°$/in. This case is drawn in Fig. 5b. It is to be noted that this is only slightly better than the figure 0.33°/in. obtainable with an optimum design of the much shorter arm.

Fairchild 280A. This arm has an average length $L = 9.13$ in. (specimens vary), and $\beta = 20.5°$. For such a length $\beta_0 = 23.7°$ and $\beta_1 = 18.3°$. Since β is between β_1 and β_0, Eqn. (27b) is used to find D_1. $D = 0.501$ in., and $m_{max} = 0.50°$/in., which is to be compared with $m_0 = 0.31°$/in. This arm is very nearly optimum for 78's, for which $\beta_0 = 20.8°$. Since β is less than β_0, the same formulae apply, and the only difference is that m passes through zero and reverses sign between R_1 (33) and R_1 (78). See Fig. 5c. The "transcription" version of this arm, the 281A, has an offset greater than optimum for LP's, but the best placement leads to the same value of $m_{max} = 0.50°$/in. Since $m_1 = -m_{min}$, the larger arm does not offer such good performance on 78's.

Audax Studio. An extremely long length $L = 14.63$ in., with an offset $\beta = 14.5°$ (equal to the optimum for LP's) combine to give $m_0 = 0.19°$/in. for $D = 0.43$ in. This arm is not at all optimum for transcriptions, however, for which $\beta_0 = 21.6°$ and $\beta_1 = 17.2°$. If used for transcriptions only, with $D = 0.356$ in., $m_{max} = 0.456$, which, as can be seen from Fig. 5d, is not much better than with LP-optimum overhang, for which $m_{max} = m_2 = 0.50°$/in. Because of the arm's great length, m is rising sharply at R_1; even though it is only 0.19°/in. at R_1 (33), it rises to 1.36°/in. at R_1 (78), which leads to the same distortion as 0.58°/in. at 33.3-rpm.

Multipurpose Compromises

We have seen that if one is willing to confine the problem to a particular size and speed of record, it is easy to specify optimum offset and overhang as a function of length, but very few commercial arms come even close to this ideal. Some compromises are of course necessary when the arm must serve more than one purpose. The case of playing both 78-rpm and LP records with the same arm and plug-in or turn-around cartridge has already been illustrated in the examples. For reasonable values of m on 78's, β may be slightly less than β_0 for LP's, but a degree or so is sufficient; some arms have β as much as 8° less than β_0. For changer use, further compromises may be necessary in connection with the mechanism. D, for instance, is a function of the number of records on the table. Since the friction of tracking is directed at an angle ϕ to the line from stylus to pivot, there is an inward radial force on the groove wall proportional to $\tan\phi$ and a similar outward force on the stylus. For vinyl LP's and lightweight arms this is less than a gram, or comparable to bearing drag, though it may be more serious for old shellac 78's. Due to the 45° inclination of the groove wall, this same force appears as a reduction in the tracking force, but by a small fraction of its magnitude. At best, this argument is an argument for long arms and correspondingly small values of β_0, rather than for making β much less than β_0. Because of the changing tracking error α, the stylus is continually being reground; so, for uniform distribution of wear, the average of α along the track, or the radial average of $R\alpha$, should be zero. That condition is extremely well satisfied for optimum designs, and poorly for many commercial arms. In short, we find no justification for "high-fidelity" arms with β more than a degree different from β_0.

Mounting Tolerances; Measurements

In view of the sharp dependence of m on D, and production variations in L and β (ranges of ¼ in. and ¾° are noted in Ref. 1), mounting instructions which may state "the pivot flange should be

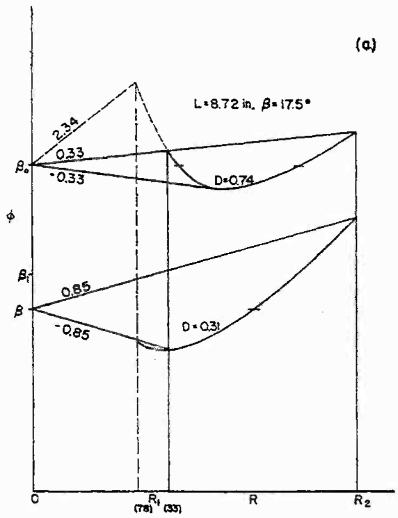


Fig. 5. Analysis of four commercial arms. a) GE A1-500; b) GE A1-501; c) Fairchild 280A; d) Audax Studio.

mounted with its center 9 in. from the center of the turntable" are ludicrously inadequate. This can lead to values[5] as high as 2.5°/in. Such information may be useful for rough positioning to see about cabinet clearance, but for good results, the values of L and β should be measured (see below), and the value of D calculated from the appropriate formula or determined graphically from Fig. 3. (Even if you use the formulae, it pays to plot the results as a precaution against numerical errors.) In some cases, an error of 1/32 in. in D can double the maximum distortion. Instead of juggling an arm into "position" and then screwing it down to the mounting board, it is better procedure to locate the position roughly and then drill *oversize* holes for the mounting screws long enough to pass through the board into a block or plate beneath, so final adjustment can be made by sliding the whole assembly in the holes. Leveling schemes which involve tilting the flange can be accommodated by providing an intermediate thin plate of metal (drilled for clearance holes) which goes between flange and mounting board.

Length. L is the distance directly from the horizontal pivot to the stylus. In some arms, the pivot is hidden, but is coaxial with the center of the mounting flange, in which case L is the average of the distances to the nearest and farthest points of the flange. If the pivot is offset, measurement can be made to the end of the bearing mounting screw. An error of as much as 1/16 in. in the measurement of L is not serious, though of course the best value measurable should be used.

Offset Angle. Since β is the angle between the pickup axis of symmetry and the line by which L is measured, it should be measured at the same time. Note that it is *not* the "elbow" angle of sharply bent arms. A plastic protractor, preferably divided in half-degrees, is easy to use. Align a thin ruler from pivot to stylus, then line up the protractor axis (0 — 180°) with the cartridge axis and read the offset angle against the ruler. Greater accuracy will result if it is arranged to read the scale in the direction of the pivot. Take care not to damage the stylus. Several tries should be made, and the median taken for β. The construction of some pickups makes it difficult to see the axis at a glance. It is helpful to mark a point on the cartridge at the end opposite the stylus which is equidistant from the sides. Turn-around cartridges like the GE have a mark built in — the other stylus. An effort should be made to determine β well within a half-degree.

Overhang. Unlike L and β, which are best measured with the arm upside down, and require only reasonable care in their measurement, D must be measured with the arm mounted on the table, preferably leveled and adjusted to proper height. Moreover, the most precise measurement it is feasible to make with simple scales (say 0.01 in.) is none too good. An error of 0.1 in. may change m by over 1°/in., as may be seen from Fig. 3 considered as being drawn for a 10-inch arm with curves every 0.1 in. A machinist's scale divided in 64ths or 100ths of an inch is suitable. A special tool for this measurement is illustrated in Ref. 1. It consists of a 9-inch bar of metal marked with a centerline, with a short section of scale attached at one end, one side of the centerline. A hole is bored through both bar and scale to fit the table centerpin. Ideally, this hole should be 0.285 in., but the nearest drill size (Letter L = 0.290 in.) may suffice. In use, the centerline is aimed at the pivot, the arm carried over the pin, and D read on the scale. For a changer, replace the spindle with a short, closer-fitting plug. If you have saved a worn stylus, it is well to substitute it for the good one while making such measurements. A similar idea can be developed in wood or plastic, either to measure D by scratching the bar with the stylus and then measuring the scratch with the arm out of the way, or to set D by making a mark first, and then moving the arm until the stylus corresponds. Some plastic spindle-adapters for 45-rpm records can be used in this manner. No special jig is needed if you have a micrometer and a set of spring calipers, though care should be taken to measure in line with the pivot.

Modifications

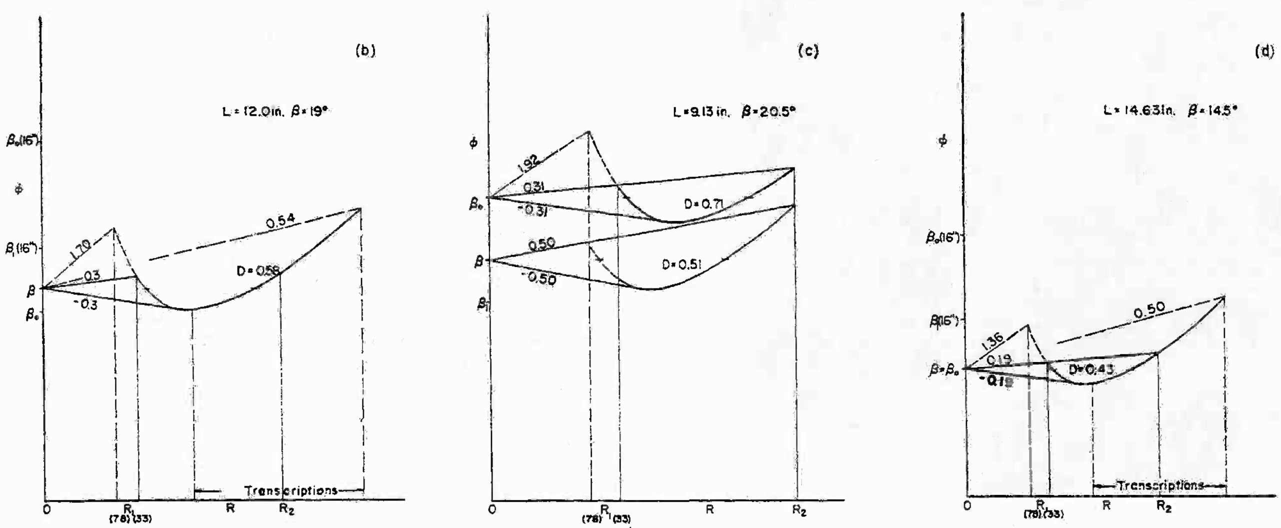
If you measure up your present arm, chances are high that it will have an overhang significantly different from the "best" value. The simplest modification is to change the overhang. Some pickup mountings may permit a small amount of shimming to increase D, but it is probably best to provide sufficient adjustment in the arm mounting flange as described above. Some plug-in cartridge slides allow enough clearance for the cartridge that if the mounting holes are enlarged, the value of β may be changed several degrees. Since one has to measure β anyway, the resulting sloppy fit is all right provided the screws can still be tightened. Lock-washers may be helpful.

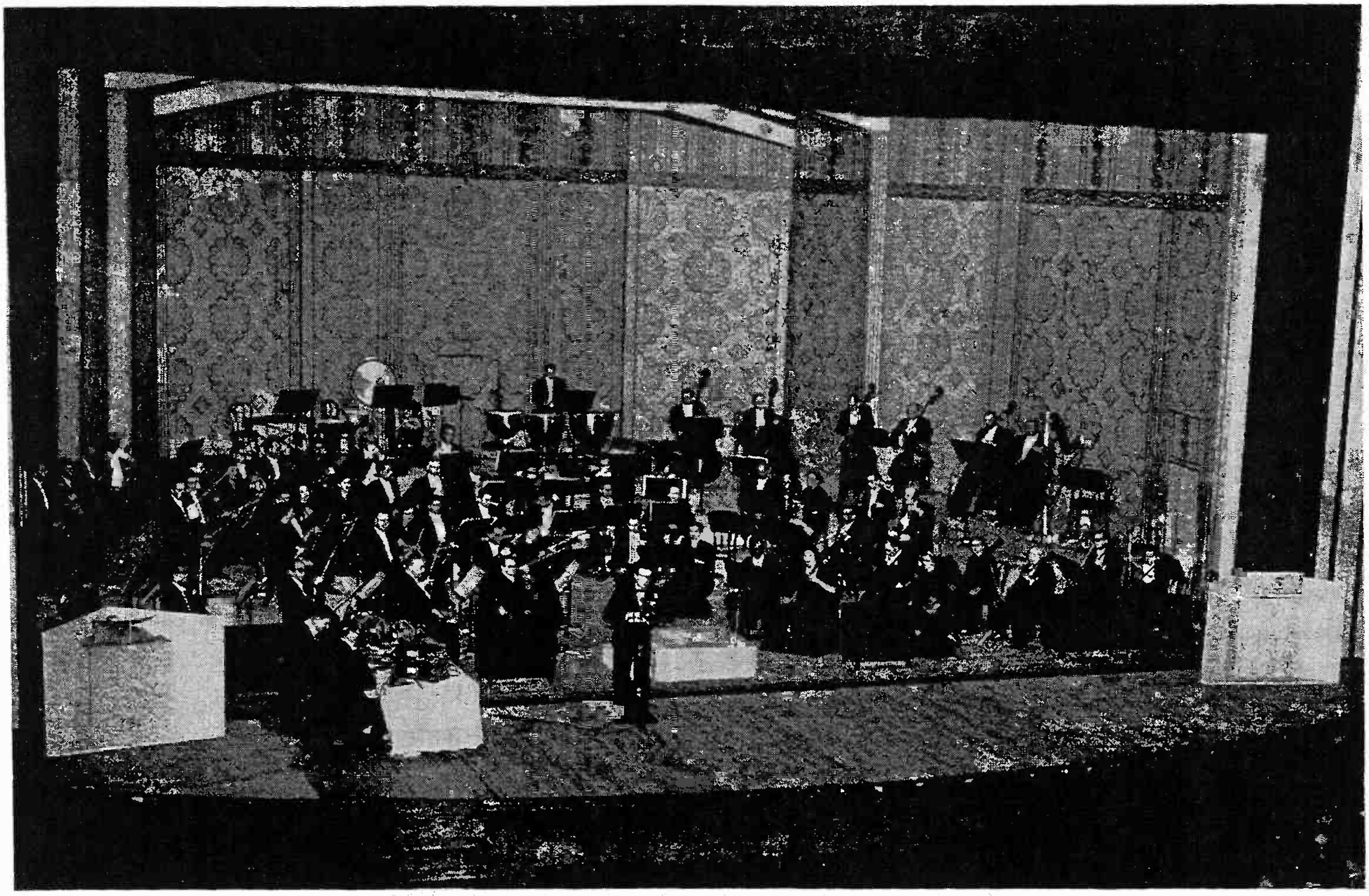
If you are planning to buy a new arm or cartridge, consideration of such flexibility is desirable. The tabulation in Ref. 1, and the associated discussion of other factors affecting arm performance — such as lateral and vertical inertia, warp-induced wow sensitivity, and arm resonances — will be found very useful.

As a final example, let us consider some possible modifications of the Fairchild 280A arm used as a previous example. It was noted before that this arm is nearly optimum for 78's,

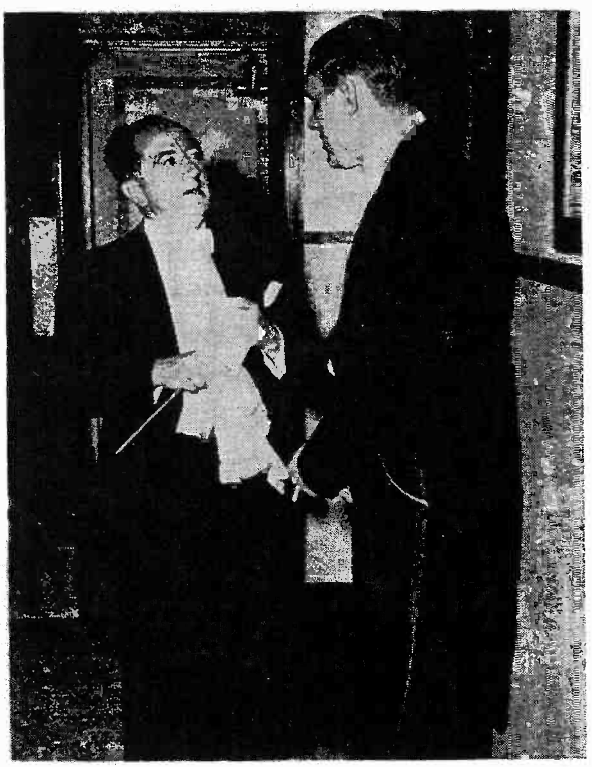
Continued on page 40

[5] "Hi-Fi Pickup Arms", *Consumer Reports* 21, (May 1956) 245-251.





Relative positions on stage of the orchestra, Klipschorns for binaural reproduction, and Gray speakers for monaural part of concert.



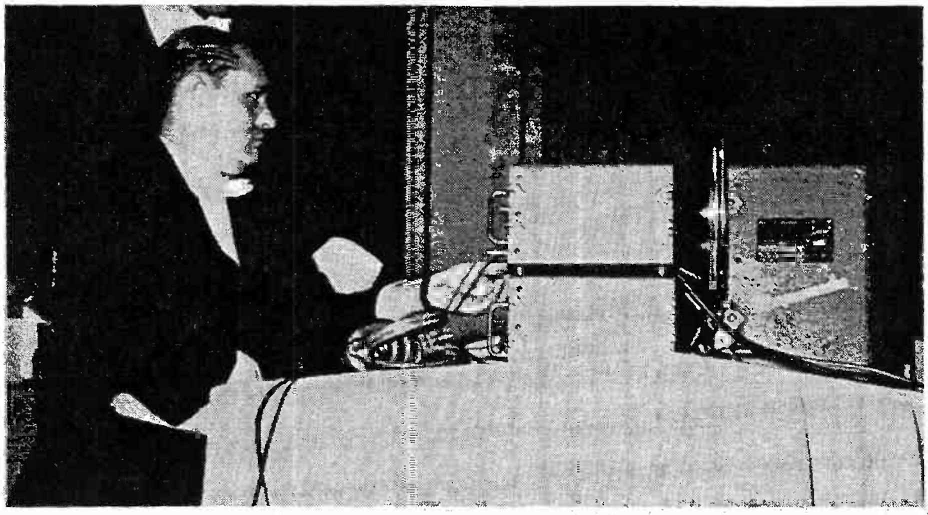
Fritz Mahler, conductor, with Magnecord's Charles Bailey.

Hi Fi in Hartford

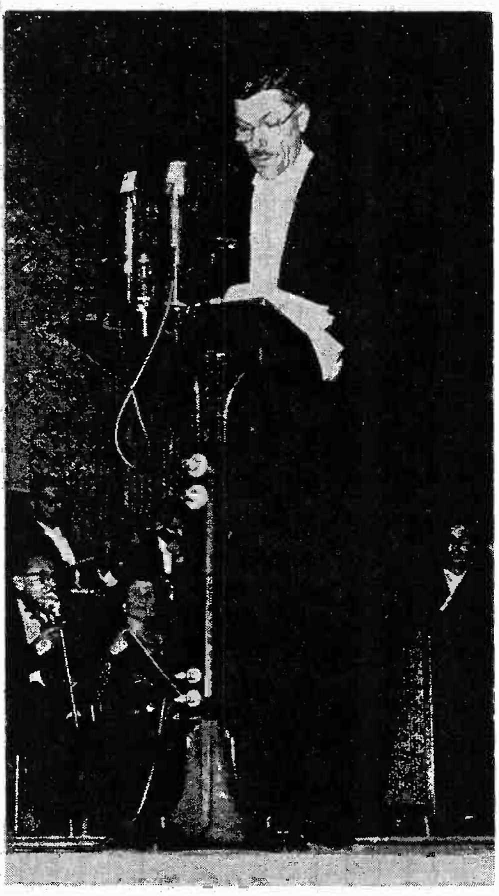
THESE pictures were taken October 9 in Bushnell Memorial Hall, at a special concert entitled *The Hartford Symphony in High Fidelity.* Gray Research and Development Company and The Audio Workshop of West Hartford were cosponsors of the event; it marked the first time in Connecticut that a live orchestra had been compared directly with reproduced sound.

Three types of sound reproduction were scheduled. The Hartford Symphony Orchestra, under Fritz Mahler, had been taped during rehearsal on a Magnecord binaural recorder. The tapes were played back at the concert (and the sound compared with the orchestra) through Gray 50-watt amplifiers feeding a pair of Klipschorns located at the ends of the stage. Rehearsal tapes were also to be played and broadcast by two FM stations (WFMQ, Hartford and WTMH, Providence) at the proper time, picked up by a receiving system at Bushnell, and reproduced during the concert as an indication of FM quality. Unfortunately, this attempt failed because of technical difficulties. The third type of reproduction was from a disc record, played by a standard Gray phonograph console through four Gray speaker systems spread across the stage.

Paul W. Klipsch, of Klipsch and Associates, was speaker and commentator during the program. Tape recording and playback was supervised by Charles Bailey of Magnecord, Inc. Master of Ceremonies was Theodore H. Parker, music critic for the *Hartford Courant.* The program included Beethoven's *Egmont* Overture, two movements from Brahms' Symphony No. 2, Britten's *Young Person's Guide to the Orchestra,* and the final movement of Tchaikovsky's Symphony No. 4.



Riding playback gain to cue in tape recording for A-B comparison test.



Paul Klipsch speaking on phonograph history.

Cutaway Klipschorn in the exhibit room was a center of interest.

Hi-fi equipment exhibit by Audio Workshop collects curious crowd.

BASIC ELECTRONICS

by Roy F. Allison

XIII: Series RLC circuits; Resonance.

CHAPTERS XI and XII* of this series were concerned with the behavior of inductors and capacitors, alone and with series resistance, in AC circuits. It was demonstrated that the AC voltage across an inductor *leads* the current through it by 90°, and that the AC voltage across a capacitor *lags* the current by 90°. An inductor or capaci-

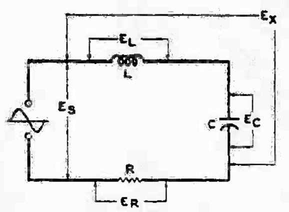


Fig. 1. A simple RLC series circuit.

tor has a current-limiting ability that depends on its *reactance*; this is measured in ohms, just as the resistance of a resistor is. Reactive ohms, however, are frequency-dependent. Inductive reactance (X_L) is *directly* proportional to frequency and the value of inductance:

$$X_L = 2\pi fL,$$

where f is the AC frequency in cps, and L is inductance in henries. Capacitive reactance (X_C) is *inversely* proportional to frequency and the value of capacitance:

$$X_C = \frac{1}{2\pi fC},$$

where, again, f is the AC frequency in cps, and C is capacitance in farads. When a resistance is put in series with a reactance across an AC source, it is found that the voltages developed across them, although proportional to the individual ohmic values, have a simple total that is higher than the source voltage. This is because the voltages are out of phase with one another—that is, their peak values do not occur at the same time. Therefore they cannot be added directly. Since they are 90° out of phase, they can be represented by perpendicular vectors whose resultant is the hypotenuse of a right triangle formed by the individual vectors. Thus,

$$E_S = \sqrt{E_R^2 + E_X^2},$$

where E_S is the total voltage across the combination, E_R is voltage across the resistance, and E_X is voltage across the reactance. Finally, since greater voltages exist in the circuits than would be obtained with purely resistive ohms, a greater current must flow for any given value of source voltage. This implies that the total current-limiting ability of a reactance and a resistance in series must be less, in ohms, than the simple sum of their individual ohmic values. The resultant ohmic value (the *impedance*) of a resistance and a reactance in series can be calculated by a formula similar to that for voltage relationships:

$$Z = \sqrt{R^2 + X^2},$$

where Z is impedance in ohms, R is resistance in ohms, and X is reactance (either inductive or capacitive) in ohms.

With the basic relationships reviewed above well in mind, we can proceed to slightly more complex circuits. Consider, for example, that in Fig. 1: an inductor, a capacitor, and a resistor in series across an AC source. Before doing anything on a quantitative basis, it will be instructive to examine the voltage and current wave forms for this circuit. These are shown in Fig. 2. The relative amplitudes are adjusted arbitrarily; still, it should be apparent that we could obtain exactly these conditions by choosing appropriate values of R, L, C, and f.

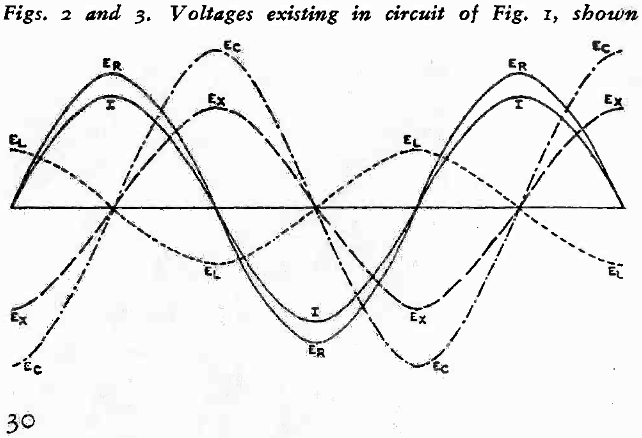
Since the current is identical in all parts of the circuit, we choose it as reference, or zero, phase. The chart begins at an instant when current is going through zero in the positive direction, continuing through 1½ cycles. The voltage across R, E_R, is in phase with the current and is so drawn. The voltage across L, E_L, leads the current by 90°; it goes through corresponding parts of its development ¼ cycle ahead of the current, so it is drawn ¼ cycle to the left of the current wave form. (Remember that the time base begins at the left.) In a similar way the voltage across C, E_C, is drawn ¼ cycle to the right of the current wave form because it lags 90° behind it. E_C is drawn with alternate dashes and dots; E_L with dots only.

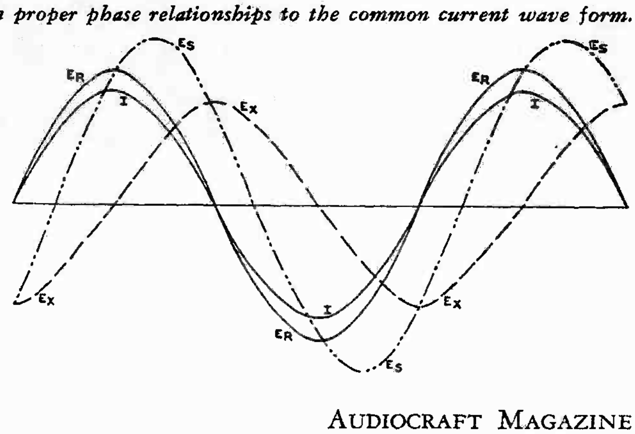
It is apparent immediately that E_C and E_L are continuously in opposition—that is, they are exactly 180° out of phase. Each exists independently of the other, to be sure, but if we were to measure their sum (E_X, Fig. 1) we would find that they are subtractive. The total voltage would be the larger minus the smaller; this is drawn in Fig. 2 as a dashed line. So far as the rest of the circuit is concerned, E_X represents accurately the sum of the two reactive voltages.

Fig. 3 shows the result of combining all three voltages to obtain the source voltage E_S. The two opposing reactive voltages are represented by E_X. Because

*AUDIOCRAFT, I (October 1956), pp. 34-35, 48-49; (November 1956), pp. 28-29, 40-41.

Figs. 2 and 3. Voltages existing in circuit of Fig. 1, shown in proper phase relationships to the common current wave form.





E_C is larger than E_L, the total reactance is capacitive; if E_L had been larger than E_C, then the circuit as a whole would have been inductive. As would be expected, since E_R is slightly larger than the net capacitive voltage E_X, the resultant or source voltage lags the current by something less than 45°, and it is less than 1.414 times E_R.

The complementary vector diagram is shown in Fig. 4. E_R, in phase with the current, is plotted to scale at zero

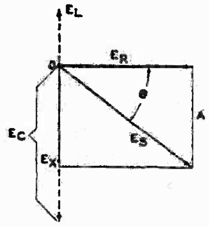


Fig. 4. Voltage vectors for the circuit.

degrees (directly right from the origin). E_L, leading the current by 90°, is plotted at +90°, directly upward from the origin. E_C, because it lags the current by 90°, is plotted at −90°, or directly downward from the origin. Again, it is clear that the two reactive voltages oppose one another; the length of their resultant, E_X, is obtained by subtracting E_L from E_C. Then E_S and θ are found as usual by determining the resultant of E_R and E_X.

Vector E_R and construction line A are the two sides of a right triangle, and vector E_S is the hypotenuse. The length of line A is that of E_X. Therefore

$$E_S = \sqrt{E_R^2 + E_X^2} = \sqrt{E_R^2 + (E_L - E_C)^2},$$

and it isn't necessary to draw out the wave forms or a vector diagram to solve this sort of circuit problem either. The angle θ can be found, without actual measurement, from the obvious trigonometric formulas applicable; for example:

$$\sin\theta = E_X/E_S = (E_L - E_C)/E_S.$$
$$\tan\theta = E_X/E_R = (E_L - E_C)/E_R.$$

In any of these formulas, if E_C is always assumed to be minus in sign—that is, if it is always subtracted from E_L, no matter which is larger—the sign of the angle will be the proper one.

In a series circuit such as this we know that voltage drops are directly proportional to the ohmic values of the individual circuit elements, since the same current is common to all. If AC voltages developed across one type of reactance tend to cancel those across the other type, and the net result is smaller than the largest one, then it follows that we must treat the ohmic values in the same way, in order to preserve the validity of Ohm's Law. We must use the same type of vector diagram and the same types of formulas when working directly with impedances rather than voltages. As shown in Fig. 5, resistance values are plotted to the right from the origin; inductive reactance is plotted at +90°; capacitive reactance is plotted downward at −90°. To find the total reactance X we take the difference between X_L and X_C. The total circuit impedance, Z, is then the resultant of R and X. This procedure leads to the same sort of mathematical solutions as for voltages:

$$Z = \sqrt{R^2 + X^2} = \sqrt{R^2 + (X_L - X_C)^2}.$$
$$\sin\theta = X/Z = (X_L - X_C)/Z.$$
$$\tan\theta = X/R = (X_L - X_C)/R.$$

To avoid confusion it should be pointed out that the subtractive process applies only to unlike reactances and reactive voltages. When multiple inductances or capacitances are involved they should be combined normally, if it is necessary to do so for computation, to obtain the total circuit inductance and the total circuit capacitance. Then the subtraction is applied to the lumped values.

Perhaps a more concrete example would be appropriate at this point. Let the circuit and values given in Fig. 6 be assumed. What is the current

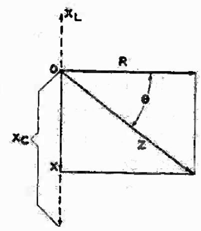


Fig. 5. Impedance vectors for Fig. 1.

through the source, the voltage across each element, and the phase angle of the source voltage with respect to current? First, we calculate the total inductance and total capacitance in the circuit. L_1 and L_2 are inductors in parallel; to find their resultant value we apply the formula

$$L = \frac{L_1 L_2}{L_1 + L_2}.$$

$$L = \frac{3 \times 6}{3 + 6} = \frac{18}{9} = 2 \text{ mh.}$$

This combination, whose resultant is 2 mh, is in series with another inductor L_3, of 3 mh. Series inductor values add directly; therefore the total effective series inductance in the circuit is 5 mh.

There are two capacitors in the circuit, C_1 and C_2, of 7 μfd and 1 μfd respectively. They are in parallel, so the total capacitance is their numerical sum: 8 μfd.

At the frequency of the source voltage (1,000 cps) the total inductive reactance is $X_L = 2\pi fL = 2 \times 3.14 \times 1{,}000 \times .005 = 31.4$ ohms. The total capacitive reactance is

$$X_C = \frac{1}{2\pi fC} = \frac{1}{2 \times 3.14 \times 1{,}000 \times 8 \times 10^{-6}}$$
$$X_C = 19.9 \text{ ohms.}$$

Total impedance in the circuit is

$$Z = \sqrt{R^2 + (X_L - X_C)^2} = \sqrt{(5)^2 + (31.4 - 19.9)^2} = \sqrt{25 + 132.25}$$
$$Z = \sqrt{157.25} = 12.54 \text{ ohms.}$$

Current through the source can now be found easily:

$$I = \frac{E_S}{Z} = \frac{20}{12.54} = 1.6 \text{ a.}$$

Total voltage across all inductive components: $E_L = IX_L = 1.6 \times 31.4 = 50.2$ v. This is divided across the 2-mh combination of L_1 and L_2, and 3-mh L_3, in direct proportion to their values. Therefore, the voltage across the parallel combination is $2/5 \times 50.2 = 20.1$ v. Voltage across L_3: $3/5 \times 50.2 = 30.1$ v.

Voltage across the capacitors: $E_C = IX_C = 1.6 \times 19.9 = 31.8$ v.

Voltage across the resistor: $E_R = IR = 1.6 \times 5 = 8$ v.

Phase angle between source voltage and current: $\sin\theta = (X_L - X_C)/Z = (31.4 - 19.9)/12.54 = 11.5/12.54 = 0.91707$. From a math table, $\theta = 66.5°$. It is a positive angle because the total reactance is inductive.

Resonance

It was probably observed that several of the voltages within the circuit were actually greater than the source voltage. This is because the circuit is fairly close to its *resonant frequency.*

We are all familiar with mechanical resonance; the classical example is the string under tension. Once set in motion, the string keeps vibrating because it possesses mass (which gives it inertia) and an elastic, or compliant, restoring force. The more compliant is the restoring force, the slower will be the vibration; the greater the mass, the slower will be the vibration. If there is

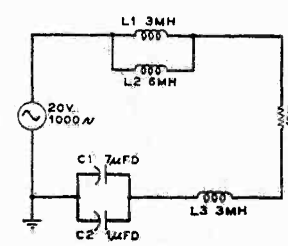


Fig. 6. Circuit for the problem example.

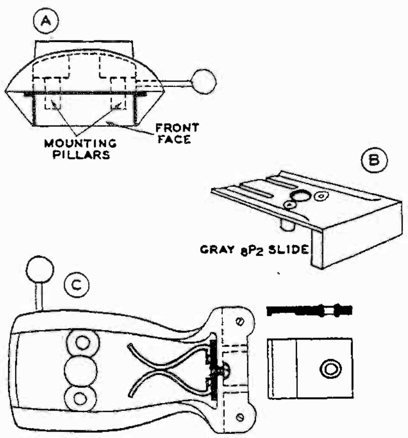
very little friction in the system the vibrations will continue for a long while after initial stimulation, and it will take only a little push at the proper moment during each cycle to sustain the vibrations. But if there is an appreciable amount of friction—if the system is

Continued on page 44

Adapting the B-J Tone Arm

Owners of the B-J tone arm may be well pleased with its tracking ability, but rather put out, as I was, by the inaccessibility of the cartridge and the inconvenient method of counterbalancing (by adding or subtracting weights at the rear of the arm, using bolts to hold the weights in place). Once the balancing is accomplished, of course, there is no further need for adjustment, provided only one cartridge is used, so this is only an initial problem. But as for removal of the cartridge for inspection or cleaning, the business of having to remove the arm from its base to get at it conveniently, and subsequently of having to realign the whole assembly for best tracking performance, are recurring headaches.

Since there are a number of tone arms on the market featuring slide-in cartridges, automatic counterbalancing, and shorting of leads to prevent hum when a cartridge is removed, I decided to modify the B-J arm to include these features.

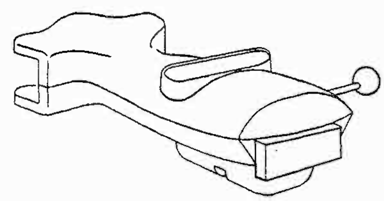


Modifying B-J arm for slide cartridges.

After removing the cartridge, arm and all (hoping it would be the last time), I took out the two cartridge-mounting pillars. Using a hack-saw blade, I cut out the front face of the head (bold line in drawing A), making the two vertical cuts first, cutting just up to the seam where the curved top and angled sides meet. Then two cuts were made diagonally from the center to the corners, leaving a triangular piece which was removed with a file, working to a clean edge at the seam. With the hack-saw blade I then cut two shallow grooves on the inside of the head along the seam, sawing a little at a time, keeping the grooves the same depth on each side and measuring the depth needed by occasional comparison and trial insertions of the cartridge slide (a Gray 8P2) into the grooves until the slide could be inserted all the way, smoothly but firmly. The thickness of the hack-saw blade and the slightly crimped edges of the slide assured a snug fit. The groove may have to be widened slightly to admit the slide, but care should be taken not to enlarge it to such an extent that the slide will not sit firmly once it is in place.

If only one cartridge is to be used, the conversion can end here. The cartridge should be mounted directly on the slide (mounting pillars are already on the slide) and counterbalancing corrected for the added weight of the slide by adding another weight or two at the rear of the arm in the usual manner. Existing connection sleeves at the ends of the cable leads in the head may be used, since the leads are long enough to allow for removal of the slide cartridge and the sleeves are easily slipped on or off the cartridge pins.

For those who plan to use a second cartridge, the following approach may be found useful. To a piece of insulation board 5/8 by 7/8 in. (a section of screw-terminal strip will do very well, since one of the holes having a threaded metal eye can be used to fasten the finished assembly to the inside back of the head), attach two strips of copper 3/16 in. wide and 1 3/8 in. long (see drawing C). To seat the strips firmly, file a slight groove in the insulation board the width and thickness of the strips and fasten the strips to the insulation board with small screws or rivets. Make sure the ends of the strips at the base do not touch. Bend the strips as shown, so that they make good contact at the bend under their own pressure. Connection sleeves can now be removed and the lead ends soldered to the bases of the spring strips. After drilling through the rear of the head a hole of the size needed to pass a screw from the terminal, the complete assembly can be mounted as shown.



Appearance of modified tone-arm head.

In the drawing at C, the signal lead to the phono input of the preamp is shorted out so no loud hum occurs when the cartridge is removed. When the slide cartridge is inserted, the pins separate the copper spring strips, opening the circuit and making firm contact.

To standardize the weight needed to counterbalance cartridges of different weights, mount the cartridges in the Gray 8P2 slides with the weights supplied by Gray specifically for each make of cartridge. Insert one of them into the tone arm and add weights to the rear of the tone arm until the proper pressure, as measured with a good stylus pressure gauge, is arrived at. The counterbalance need never be readjusted, since the weighted slide and cartridge automatically compensate.

Harold Kristiansen
Queens Village, N. Y.

AUDIO AIDS WANTED

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

Wire Brush Cleans Salvaged Parts

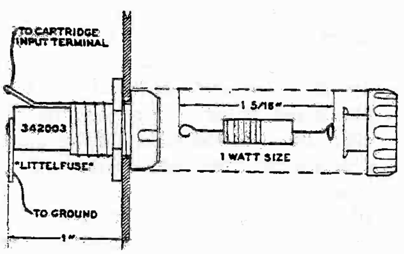
To clean up switches, controls, sockets, or other items salvaged from old equipment, try using a wire brush like the ones auto mechanics use to remove scale and rust from steel parts. These brushes look pretty stiff and cumbersome at first glance, but the bristles are quite flexible. A few passes over a tube socket, for example, will remove all dirt and resin from both the terminals and the base material. Then, to remove the solder blobs, heat the terminals one at a time with a soldering iron and brush the solder away while it is still liquid. Presto — you have a part that is as good as new.

L. E. Johnston
Madison, Wis.

Plug-in Load Resistors

Here is an easy way to provide plug-in facilities for any value of load resistor one would wish to use in an amplifier or preamp. Ordinarily it is a lot of trouble to remove the preamp cover and solder in a new resistor. With this modification a simple substitution of various resistors will determine the optimum load for your system. Not only that — when cartridge comparisons are made, the proper resistor can be inserted at the same time.

The original plug-in was mounted on the back of a Heathkit WA-P2 preamp. As an ESL Concert cartridge was used,



Quick-change provision for pickup load.

the lead lengths were immaterial (because of its low impedance); generally speaking, however, it should be mounted as close to the phono input as possible and the lead lengths kept short to minimize the possibility of hum pickup.

Any fuse extractor post can be used. The Littelfuse 342003 is recommended because it extends only one inch behind the panel. Drill a half-inch hole through the panel and mount the fuse retainer. Turn the binding nut *finger-tight* only; this will be firm enough, and the use of a wrench may strip the threads or even break the bakelite retainer in two. Solder a lead from one terminal to audio ground and a lead from the other terminal to the audio input terminal (where the existing load resistor is connected). Snip out the existing load resistor, leaving the pigtails as long as possible; it may be wanted later.

Use 1-watt resistors for plug-in purposes, rather than the usual ½-watt size. The pigtails are heavier and will take more end pressure without bending or buckling.

Referring to the diagram, snip and loop the pigtails at each end of the selected resistor, so that the total length is about 1 5/16 in. The inner end loop fits in the socket inside the fuse retainer. The loop on the other end should form a flat circle so as to engage the spring in the fuse retainer cap. Insert the resistor, turn and lock the cap, and it's ready to operate.

A good idea would be to prepare at one time (and label, if desired) all the probable resistors that would ever be used. These should cover just about any eventuality: 200 ohms (for ESL with 201 transformer), 27 K, 33 K, 40 K, 47 K, 100 K, and 120 K.

Don H. Brooks
Trenton, Ont.

Reinforcing Record Covers

The backs of LP-record jackets are only paper and often the record breaks through them. I have found that they may be easily repaired or reinforced by placing a strip of Scotch electrical tape along the back of the jacket. The tape forms a strong back that will take a lot of abuse.

Adhesive tape can also be used for this purpose, but masking tape and cellophane tape are not strong enough and should not be used.

Norman Worth
Oakland, Calif.

Adjusting Stylus Force

One of the many problems encountered in installing an audio system is the proper weighting of the tone arm with the cartridge installed. Since a new penny weighs about 3.1 grams, the simplest type of counterbalance scale is sufficient to weigh the arm accurately. I use a drafting triangle balanced on a triangular ruler.

For a GE or a Pickering cartridge, two pennies, placed so that they are the same distance from the fulcrum of the scale as the stylus of the cartridge is, are about the right weight. For other values, the relation "distance (from center of penny to fulcrum) times weight (of penny) equals distance (from stylus tip to fulcrum) times weight (of tone arm)" will yield good results. Be careful that your scale is balanced before pennies and stylus are placed on it.

John C. Alderman, Jr.
Washington, D. C.

One precaution: make certain that the stylus force is adjusted with the arm at exactly the same height as it normally is when playing a record. — Ed.

Hex-Nut Guide

When your fingertips are too big to guide a crucial hex nut to its place in a compact assembly, and even tweezers can't find a direct route, try this. Attach a piece of hookup wire to one of the hex nut's flat edges with a dab of solder. Bend the wire as required and, using it as a handle, place the nut in position and tighten the bolt. Then detach the wire.

If a little tugging and wiggling won't work, heat applied to the wire as near the nut as possible will soften the solder sufficiently to free the wire.

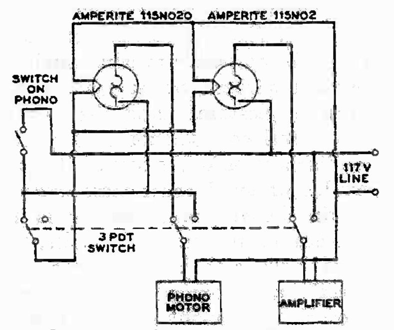
Hugh Kenner
Santa Barbara, Calif.

Automatic Delay Circuit

One of the problems which seem to concern many audiophiles is the question of switching their equipment automatically with a record changer. Several solutions have been offered in "Audio Aids".

A somewhat more complex switching circuit is shown in the diagram. With the switch thrown to the right, the system operates normally; with it thrown to the left, the automatic switching is in effect.

In this position the phono switch operates only two thermal relays, less than .05 amp, in addition to the phono motor. Thus, there is no problem of overloading the contacts. The relay



A phono/amplifier switching circuit.

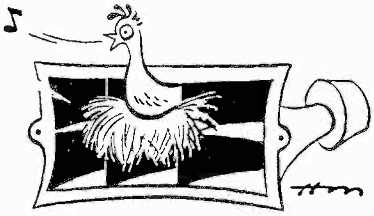
contacts are rated at 3 amps. Since they are thermally operated, there is no strong magnetic field.

The relay operating the amplifier has only a two-second delay (the shortest available), while the relay operating the phonograph is chosen to match the warm-up time of the amplifier. Thus the changer is not started until the amplifier is ready for the music.

When the switch on the changer opens, the changer shuts off immediately, and after a few seconds the amplifier shuts off. On my system, this delay is 10 to 15 seconds, time enough to turn the record over or to add another one without the amplifier's being shut off.

To use the system with this delay circuit, one merely turns on the changer and flips the reject-starting knob. The circuit turns on the amplifier, gives it time to warm up, and starts the changer. When the record is finished, it shuts off the changer and gives you a chance to put on another record. If you do not choose to do so, it will then shut off the amplifier too.

James J. Schmidt
Omaha, Nebr.



Sound-Fanciers' Guide

by R. D. DARRELL

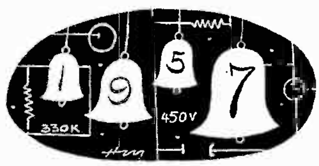
EVERY time I complete a full-cycle swing between the magnetic poles of hi fi's most blatant and most gentle attractions, I pause for a moment on the neutral zero line to wonder about the possibilities of any combination of these two usually conflicting appeals. Generally this dispassionate interval of equilibrium (or is it indecision?) is only an instantaneous hesitation preliminary to renewed cycling from one extreme to the other — for, temperamentally, I'm disposed to agree with Oscar Wilde that "Nothing succeeds like excess!" Like so many other enthusiasts, amateur or pro, my liveliest listening delights sometimes lie in going all out (as reported in this column for last November) for sheer sonic displays; at others (last month, for example), I succumb completely to more intellectual, perhaps spiritual, but at any rate purely musical, pleasures. And I've never yet found real contradiction in any similar readiness to make the best of all possible worlds of aural experience: *everything* in the domains of sound should be of concern to the true audiophile, and I have only pity both for the "music lover" who puritanically abjures all sensuous, sonic-only enticements, and the rabid hi-fi fanatic who just as intolerantly remains deaf to disembodied aesthetic charms which whisper persuasively to the inner ear rather than dramatically assault the outer ones.

Nevertheless, I still try to convince myself that there must be some middle ground where extremists of both types can join on equal terms in mutual enjoyment. This month I've been investigating some of the most likely areas of meeting, of which the most promising seemed to be an extension of our earlier explorations of organ and percussion repertories into the realm of historically significant yet less familiar instruments, or those less frequently heard in solo or small-ensemble isolation. But for once I find it difficult to make either a clearly positive or negative report: the odder the instruments themselves are, the more intriguing it is to discover and study them; but, at the same time, the more quickly their fascinations, both sonic and musically expressive, seem susceptible of exhaustion. Consequently, I often hesitate to recommend such highly specialized recordings for inclusion in most permanent record collections, even when I feel very strongly that they richly warrant at least one attentive hearing. Each listener-reader will have to make up his own mind on this point; the best the following reports can do is to provide some useful evidence on which a final decision — always as colored by personal predilections — can be fairly based.

On The Rim

The most curious of all these instrumental oddities is that passionate favorite of eighteenth-century sensibility, known almost entirely by repute only to present-day listeners: the Musical Glasses, on which Gluck as a young man once performed in London; which, as ingeniously mechanized by none other than Benjamin Franklin, became known as the Glass Harmonica; and for which Mozart among many others composed special music. I first became interested in it from the references in Mozart's letters, and became even more deeply absorbed when I had to do research for annotations for an LP of his Adagio and Rondo, K. 617, originally written for glass harmonica with flute, oboe, viola, and cello, but now usually played (as in this disc, Vox PL 8550) with a celesta substitute in the starring role. It's a gravely beautiful little work and brightly played, in a somewhat hard, however clean, recording; but the delicate tinging of the celesta whetted my appetite all the more for hearing exactly what the more keening and ethereal tones of the original instrument sound like.

Well, as far as I've been able to discover, the mechanically rotated glasses of Franklin's devising, with or without



the keyboard which others added later, have never been recorded. But at least something close to their authentic quality is captured by Bruno Hoffmann's "Glass Harp", which may be heard in the Archive LP (ARC 3044) of not only K. 617 but also the solo Adagio in C, K. 617a. The accompanying notes are rather ambiguous on the exact nature of Hoffmann's instrument, but I gather that it is a set of some 40 water-filled tuned glasses (each with its own wooden resonator) which are edge-rubbed in the immemorial manner by the player's moistened finger tips. At any rate, the sounds produced are certainly unique in their fragile and antique, if somewhat strained, qualities. Hoffmann and his colleagues play the Andante and Rondo considerably slower and with more obvious expression than the Vox group, but the replacement of struck steel bars by rubbed glass rims makes for an entirely different sonic magic, and their peculiar timbres (at times resembling those of piano-tone tapes played backward) can be studied and relished even more effectively in the unaccompanied Adagio. This disc surely can be recommended for a place in your permanent library, for if the novel attractions of the glass harp are not enough in themselves, there is a wealth of too seldom-heard but wholly Mozartian music — not only the pieces cited, but (on the other side of the LP) a series of some unaccompanied vocal canons, in both serious and highly ribald veins, which also present unfamiliar facets of the composer's genius and, like the instrumental works, they are recorded with immaculate tonal purity.

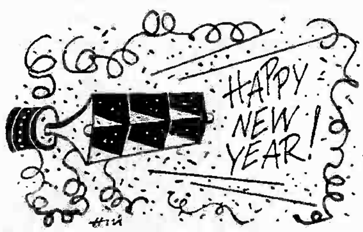
However, I doubt that anyone except a highly specialist collector would want to hear more than once a very different LP of a similar set of some 27 musical glasses, played this time by Einar Hansen in an interminable medley of *Christmas Music Around the World* (MGM E 3277). Some years ago Hansen put out a ten-inch LP (Banner 2000) of unspecified but presumably glass-harmonica pieces by Mozart, Gossec, and Beethoven. I was never able to secure a copy, and so was delighted to discover the more recent release. But, alas, my anticipation was disappointed: apart from the rather wispy charms of the tunes themselves, drawn from 37 countries ranging from Argentina to Wales, Hansen seems to have less secure control over his instrument than Hoffmann; or, perhaps, it is the fault of the extremely close microphone placement and excessive amplification that the background-noise level is so high here and every extraneous squeak of the glass rubbing so prominent. There are occasional wondrously pure and expressive high tones, but too often they are shrilly penetrating, and the sense of effort in the playing (noticeable even in Hoffmann's performances) here leaves one uncomfortably uncertain whether the plodding soloist won't break down in despair before the end. At any rate, even for devotees of the musical glasses, a little — especially without Mozart to help out — goes a long way!

Carillon, Clock, Carousel, — and Pedals

The same weakness in lasting appeal seems to be characteristic of several other instrumental curios, to which I have listened with no small fascination, but always wind up wondering whether I'll ever want to hear them again. Luckily, some of their other attractions may be more decisive. In the case of the Charles T. Chapman *Carillon Concert* (McIntosh MM 102), the obvious sales pitch is that for a memento of a visit to the famous Singing Tower of Luray, Virginia, and secondarily for the Christmas-tune selections on the second side. But I could dispense entirely with the quasi-folk and light-classic airs on side 1, which strike me as too flimsy for carillon performance (or at least for such naïvely arranged and ornamented versions as these), and I'd gladly swap most of the flowery notes on the Singing Tower and its setting for a few words of information about how the recording itself was made. For I must admit that I've never heard a set of tower bells (and this is a genuine, not electronically simulated, carillon) captured more cleanly on discs, or with less background and extraneous noise distractions. For sonic-study purposes, then, this is a real gem, no matter how musically synthetic and tasteless it may seem to me.

In the next case, that of the *Musical Organ Clock* recital (Vanguard VRS 7020), played on an instrument, ascribed to 1787, that is in the private collection of Rudolf Huebner in Vienna, the contents are musically more diversified, if lightweight in different fashion; but they make for a musicologist's nightmare with their quite spurious "Mozart" pieces, and those by such mysterious composers as "Liszt" (not Franz certainly; maybe Anton Liste?), "Sterl" (Sterkel?), Polledro, and of course the eternally fecund "Anon." Moreover, some of these (like the one-time hit tune from Haydn's *Creation*) are of later date than that credited to the mechanical instrument itself — although its cylinder repertory not implausibly may have been freshened by new releases in later years. Yet, despite all this, and despite too the wheeziness of the little pipes which Mozart disdained as "too high-pitched" and "too childish" when he was writing music for a similar organ clock in 1790, I must say that most of the naïve little *divertissements* here are amusing; some are even quite charming; and one of them, the "Liszt" Allegro, provides some startling intimations of much later silent-movie accompaniment idioms. The recording itself is first-rate, even to the merciless realism with which it exposes every intonation uncertainty, tone-production distortion, and mechanism noise obbligato.

Another prize example of ultrarealistic recording (exposing this time even more mechanism clankings) is Audio Fidelity's *Merry-Go-Round Music* (AFLP 901, 10 in.) of an old-time circus carousel preserved in Bethpage, L. I. This instrument strikes me as more authentic, if tonally less interesting, than other recorded carousels I've heard; indeed, there's almost the irreducible minimum of actual wind tone; yet this very thinness of sonority, as contrasted with the accompanying percussive tinkles, gives the silly little tunes played and their even sillier ornamentations a singularly intriguing and nostalgic innocence. I was amused, too, by



the piquant syncopations between the basic steady beat and the sometimes slow-to-speak pipes; but here again it's primarily as a documentary that this disc makes any substantial bid for ownership.

The last LP in this group (Cook 1131, 10 in.) *should* be the most important, for it is the first (as far as I know) to be devoted to the pedal harpsichord, and moreover includes well-known good-sized works by Bach and Mozart. The instrument itself is not a baroque original, but a far larger (and louder) product of the celebrated Hans Neupert of Nürnberg, who built it to the design specifications of the present player, Bruce Prince-Joseph. It's hard for me to decide whether the lack of crispness in its reproduced performances is a fault of the close, dry, but very clean recording, or — more likely — of the instrument itself. At any rate, there is no lack of weight in the recorded tones: there are thunderous moments such as I have never before heard from an ordinary harpsichord, or even from the modern (pedal-less) Erard of Mme. Landowska. There also are some extremely interesting examples of the widely varied registrations commanded by a big harpsichord. But my pleasure in all this is ruthlessly suffocated by the really brutal heaviness of Prince-Joseph's performances, which among other crimes achieves the minor miracle of making Bach sound intolerably pedestrian. And I see no excuse whatever for playing the popular C major Mozart Sonata, K. 545, on a double-keyboard-*cum*-pedalboard instrument. But I'd still like to hear more of it — exclusively in music which can properly utilize its powerful pedal tones, and preferably under the fingers and toes of a more spirited and communicative executant.

Field, Ceremonial, and Parade Music

Returning to more orthodox instruments, if not always in their most familiar combinations, I was delighted at first by the boldly recorded piccolos and drums of Frederick Fennell's *Spirit of '76* (Mercury MG 50111), a valuable historic documentary of U. S. Army fife-and-drum field music. But after a few minutes (and despite all the fine tunes, rhythmic steadiness, and amusing "pawky piping") the monotony was enough to send at least one listener over the hill in a hurry. Happily, however, Fennell's companion disc of *Ruffles and Flourishes* (Mercury MG 50112), for trumpets and drums, not only has equal documentary value, but far more varied sonic and musical attractions. And it is here that the full power and authenticity of the recording itself is clearly revealed: I may have heard more brilliant cymbal clashes and crashes elsewhere, but I never have heard any that sounded more honestly and metallically like the real thing! But there are many other aural highlights as well to make this disc a joy to the audiophile's ears and a rigorous test of over-all sound system transient response.

I can skip hastily over *The Trombone*, Vol. 1 (London LS 989, 10 in.), for this coupling of solos by Gabriel Masson and ensemble pieces by the *Quatuor de Trombones de Paris*, while skillfully played and recorded, offers examples of little-known modern French composers which (except perhaps for the quartet by one Dondayne) will interest trombonists only — if, indeed, it doesn't bore even them.

I got infinitely more satisfaction out of *The Golden Age of Brass* (Unicorn UN 1003), which stars the far more brilliant trumpeter, Roger Voisin, and other Bostonian brass virtuosos in an astonishing variety of British, Italian, and German early seventeenth-century compositions. Here, my only complaints are piddling: that authentic-score Cornetts or *Zinks* (which are nothing like cornets, remember!) were not used, but the more plangent and sharply focused modern trumpets were; and that all the performances (instead of just the Gabrieli Canzona and Bonelli Toccata) weren't recorded in the spacious acoustical environment of Boston's Symphony Hall. The rest are good small-studio recordings, certainly, but much of the glowing warmth, as well as of the ceremonial impressiveness, of such brass-ensemble playing demands more open acoustics. These are only minor flaws in an otherwise fine disc, one to be cherished for far more than a single hearing, and one which leaves me eagerly anticipating its just announced companion, *The Modern Age of Brass* (Unicorn UN 1031), which was recorded by Peter Bartók in the new MIT Kresge Auditorium.

After this dip into the far-off Renaissance, the search for further varieties of brass and wood-wind sonorities leads back into the more familiar and more obviously tuneful marching-band repertories, represented at their best by the latest release in the long Deutschmeister Band series (*Marches of Many Nations*, Westminster W-LAB 7037 or Sonotape SW 1034), and the second release by the Scots Guards (*On Parade*, Angel 35337). Both are the electrifying real McCoy in march music: the former characterized by rather gruff and sometimes even coarse tone production, very broadly and brilliantly recorded; the latter by smoother, richer qualities (except of course in the brief sections for reedy bagpipes alone) and by the same superbly open-air acoustic spaciousness that made the earlier Scots Guards LP (Angel 35271, reviewed here July 1956) the finest recording of its kind I know. Unfortunately, however, the music in the Scotsmen's new program, for all its lustiness, is less immediately appealing — or else, as in so many sequels, it comes as less of an overwhelming surprise. So if you can afford only a single example, by all means go back to the incredibly effective 35271.

Pops With & Without Gimmicks

Another approach to the practical solution of the sound *vs.* music problem is to minimize or even largely disregard the aesthetic significance of the music involved by drawing on the popular song-and-dance repertories for light but ingratiating materials which can be arranged and recorded for optimum sonic novelty without running any risk of artistic tastelessness or stylistic heresies.

The most sensational current example of this approach (and a worthy successor to earlier ingenious technological experimentations in the domain of pops records) is Ferrante and Teicher's gimmicked two-piano playing in *Soundproof* (Westminster WP 6014). This is the latest high in gadgetry run wild, for, in addition to "preparing" the pianos themselves, all kinds of odd performance techniques, tape-editing tricks, and no less than 17 mike channels are utilized. The results are (not unexpectedly) highly mixed themselves, but at their best they are indeed far out of this world. I was let down a bit when the players exhibited the wrong feeling for a couple of their tunes (as in an overly antique and listless *Greensleeves* and a dragged *Someone to Watch Over Me*), and at other points where the distinctive sound qualities struck me as having been anticipated by normal instruments or by science-fiction-film sound effects. But for the rest, there is a wide enough variety of novel colors and timbres here to satisfy the most insatiable seeker after new sounds, and a verve and point to the performances sufficient to make even the strangest of these expressively significant. Perhaps no experimental disc of this kind could hope to live up wholly to all the ballyhoo with which it has been promoted, and I'm not at all sure how well it will wear with familiarity — but *Soundproof* is definitely not to be missed.

Since the Paul Severson Quartet's *Sounds . . . Crazy* (Jazztape Stereo ST 4016) hasn't been given any advertising prepublicity that I've seen, it comes in some ways as an even more exhilarating surprise. It is mild enough, and by no means hot, jazzical entertainment; its

real distinction never would be suspected in single-channel reproduction. But in stereo, Severson has endowed his own buoyant little tunes with uncommon grace and effectiveness by scoring them as antiphonal duos for widely spaced trombone and sax. The contrapuntal interplay between these two instruments, against an easy-going rhythm accompaniment, exploits deliciously a stereo-sound potentiality which no other composer yet has had the wit — and skill — to explore as ingeniously. Only the title is wrong: this sounds not at all crazy, but mighty sensible and sportive, to me.

After these happy experiments, it's hard to be either excited or satisfied by the Sauter-Finegan reworkings of their once-novel percussion melodramatics. If you haven't heard them before, then perhaps *On the Outer Fringe* (RCA Victor LPM 1240) will please you better than it did me. There certainly is plenty of maniacal clattering, tinkling, banging, and throbbing here (and the recording is a masterpiece of effortless and crystalline transient transcription), but too often the music making itself is little more than half-hearted tonal doodling. Even the more spirited pieces never really get anywhere, and the portentous "prunes-and-prisms" precision of Ruth Yorke's recitation of a Karl Shapiro poem sounds incongruously out of place here (or perhaps anywhere).

For me, there was a vast sense of relief when I turned back either to honest bluff-and-hearty straight hot jazz or to the finest vintages of night-club tonal champagne. In the former, I got a real kick out of the rowdy but jubilant drive of Kid Ory's Band in *The Legendary Kid* (Good Time Jazz L 12016), recorded with a perfect combination of tonal solidity and brilliant definition; and the easy-swinging virtuosity of Buddy Collette in *Man of Many Parts* (Contemporary C 3522), nicely but less extraordinarily recorded, and sonically most interesting for Collette's ability to adapt his individual style to the distinctive characteristics of alto and tenor sax, flute, and clarinet.

For the latter, Pearl Chertok's rather wispy harping in *Strings of Pearl* (Audio Fidelity AFLP 1805), recorded rather too closely for more than pastelish tone coloring, left me pretty lukewarm except when it was suddenly galvanized into life by Johnny Rodriguez's magnificent bongo drumming. But the same company's highly reverberant (echo chamber?) recording of the husky-voiced, multilingual *chanteuse*, Patachou (AFLP 1814) gave me a thrill — at least in such French songs as *Boum*, *Un Jour tu verras*, and *Sous le ciel de Paris* — which approached, if it never could quite match, that with which I heard incredulously for the first time, too many years ago, the still incomparable song projections of Marlene Dietrich.

But for less exciting yet even more graciously satisfying just-listening pleasure, I find myself always returning eventually to Georges Feyer, who continues to prove his supreme mastery of light piano-medley domains. His imagination for creating just the right treatments of both tunes and bridges is as inexhaustible as it is ingenious. The famous tunes from the Rodgers and Hammerstein *King and I* and *Carousel* never sounded better or fresher to me than in Feyer's versions (Vox PL 21.300); while his *Echoes of Spain* (Vox VX 25.070 or Phonotapes-Sonore PM 5005) demonstrates anew his gift for capturing the quintessence of musical local color. And, for that matter, the infectious rhythmic pulse he brings to such war horses as the Falla *Fire Dance* and Chabrier *España* might put many a far more celebrated "serious" pianist and conductor to shame. Here, if ever, the reconciliation between captivating musical appeals (however light) and those of transparently pure reproduced sound (however unsensational) is flawlessly (and seemingly effortlessly) achieved.

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Book Reviews

by Richard D. Keller

Introduction to Printed Circuits

Robert L. Swiggett; pub. by John F. Rider, Inc., New York; 101 pages; $2.70; paper bound.

This book discusses the history, development, and application of printed circuits.

As the author points out, the trend in the electronics industry is toward more and more involved circuitry, giving rise to extremely complex new equipments, computers, and radar and control systems. It is necessary, at the same time, to miniaturize and lighten these systems as much as possible for airborne applications and reasonable portability on the ground.

The old wiring and cabling techniques with their tedious layout and soldering problems, and metal chassis, were incompatible with such requirements. Although production of *components* has been improved by highly efficient and automatic methods, electronics-assembly work, until very recently, has not been mechanized as much as the average layman would suppose. Mass production of radio and television sets has utilized long tables and hundreds of women, each wielding a soldering iron.

One of the major factors that deterred mechanization of electronics assembly was simply the physical form of the products. No assembly machine, no matter how well engineered, could be expected to cope with the maze of wires and components characteristic of the point-to-point wiring method.

Printed circuits provide the answer — a system of making electrical interconnections without wires. They provide a supporting medium for components in a form that is readily adaptable to machine handling and machine assembly. They have made possible the revolution now under way in the factories of electronics-equipment assemblers.

This book describes each of the types of printed circuits encountered in electronic equipment today, discussing their characteristics and functions, how they are made, and their effects on techniques of servicing devices that contain them. The purpose of the book is to provide the reader with a broad knowledge of the printed-circuit field and to equip the serviceman or technician with the specific know-how that he needs when he encounters printed-circuit assemblies. Although it is not intended to be a complete handbook of engineering information, design engineers should find here a comprehensive introduction to the field.

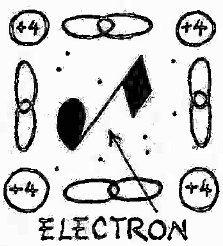
Electronics Made Easy

Lothar Stern; pub. by Popular Mechanics Press, Chicago; 192 pages; $2.95.

Here is a book for the rank beginner interested in electronics. It is primarily a collection of how-to-build-it articles with construction data on various kits of radios, amplifiers, and intercom systems.

There are also complete chapters devoted to radio theory, radio servicing, high fidelity, and transistors. These are written simply and illustrated profusely, often in two colors.

The build-it-yourself projects start with a crystal set and proceed up through 1-, 3-, and 5-tube receivers, a



short-wave converter, a 3-tube wireless broadcaster, a 1- and a 2-tube phono amplifier, an AC-DC and a battery-operated intercom, and several switching control devices such as a capacity-operated relay, a humidity-controlled switch, and an "electronic-eye" switch circuit. Several modifications of existing equipment (an intercom from a table radio, PA system from a car radio, earphones added to a TV set, clock timer for a home radio, etc.) are given.

Finally, several hi-fi amplifier and preamplifier kits are described, as well as a number of simple transistor circuits and two portable transistor-radio kits.

Electronic Engineering

Samuel Seely; pub. by McGraw-Hill Book Co., New York; 524 pages; $8.00.

This engineering text covers a large variety of electronic subjects.

It begins with a general introduction to tubes and tube-circuit principles, and then proceeds into a detailed physical and mathematical analysis of the more important types of electron-tube circuits. Examples and problems are used to relate the theoretical developments with practical situations.

Applications in such fields as radar, television, electronic control and instrumentation, computers, and power rectification are included in the framework of the book, as well as chapters on solid-state theory and transistor application. The transistor information here is quite limited since it is based entirely on early work in the field, but the tube-circuit information and mathematical analyses are quite detailed and thorough.

Transistors Handbook

William D. Bevitt; pub. by Prentice-Hall, Englewood Cliffs, N. J.; 410 pages; $9.00.

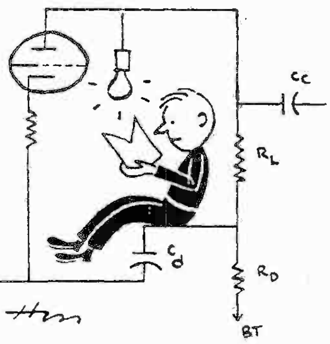
A reference work on transistor circuits, applications, and characteristics, this book concerns itself mainly with the point-contact and early junction types of transistors.

Technical information on point-contact transistors, although interesting from a historical or background point of view, is more or less antiquated now that junction types have proved their superiority for most applications. Greater stability, lower noise figures, and higher gains have established the junction types in just about all applications except some switching circuits.

This points up one difficulty in an area growing as rapidly as the transistor field. By the time information is compiled, written up, and passed through the long publishing cycle to book form, the state of the art has moved considerably forward. Such is the case here.

One of the main features of the book is that a large number of practical circuits are shown. However, these often use now-unavailable point-contact transistors (such as the G11) or else the cheapest of the junction types — transistors which naturally have extremely wide parameter variations. Most of the basic transistor material deals with r parameters and little mention is made of h parameters which are now more generally used. And the commercial-transistor characteristics chart in the appendix lists many early experimental point-contact types, but it shows no higher than the 2N57 in its compilation of types, although RETMA listings up to around 2N190 have been commercially available for over a year and registered types are up in the 2N250 area now.

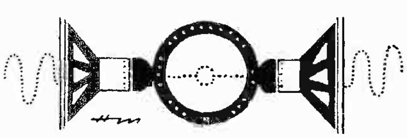
In short, although the publishing date of the book is 1956, the material in it appears to be of 1951 to 1954 vintage.



Miscellany

Vol. 8 of the Howard W. Sams *Photofact* publication, *Automatic Record Changer and Tape Recorder Service Manual*, has just been released by the publisher. Treated in this volume are the following models: Aircastle 795-880; Berlant BR-1; Collaro RC-54 (Mark II); Columbia 461; Continental 220; Crestwood 404; Dixie-Land 110; Dukane 11A200; Ekotape 212; F-M-E (Federal) 37-C, 47-A; Knight 96RX-635; Masco 500; Pentron CT-1; RCA SRT-403 (MI-15917); Silvertone 4072, 6072; Telectro-Tape 556; V-M 954-B; and Webcor E-2020, 2021, 2022, 2023, 2024. Price is $3.95.

Vol. 7 of the Howard W. Sams *Photofact* publication, *Audio Amplifiers and Associated Equipment*, has also been released. Parts data, operating information, and schematic diagrams are furnished for the following amplifiers and tuners: Bell 2122C, 2199B, 2256, 2255; Brociner Mark 12; Browning RJ-43, RJ-49; Craftsmen C350, C550, C1000; David Bogen DB15G, DO30A, J15, FM400A, R750; Electro-Voice A-20C; Espey 500A, 501, 201, 301; Fairchild 260; Fisher FM-80; Grommes LJ-3, 55PG; Knight SX8L727, SX11L719, SX14L721, SX19L720; Masco C5-6P-3; McGohan MG-25B, WA-310; McIntosh MC-30; Pedersen PCP-20, PRT-1B, PRT-1LC; Pilot AA-420, AA-903, AA-904, AF-860, FM-607; RCA SVT-1; Rauland AA-7; Regency HF-80, HF-200, HF-350P; H. H. Scott 232, 232A; and Stromberg-Carlson AP-60, AU-58. Price is $3.50.



TRANSISTORS

Continued from page 21

Further Reading

The following books will be referred to by the author's name in the categorical reading suggestions below:

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

Kiver, M. S. *Transistors in Radio and Television.* New York: McGraw-Hill, 1955.

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

Scott, T. R. *Transistors and Other Crystal Valves.* London: Macdonald and Evans (Distr. in U. S. by Essential Books, Inc.), 1955.

Shea, R. F. *Principles of Transistor Circuits.* New York: Wiley, 1953.

Shea, R. F. *Transistor Audio Amplifiers.* New York: Wiley, 1955.

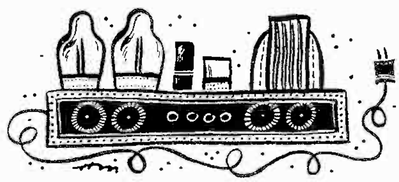
Curve Families

Kiver, pp. 74-79.

Lo, Chapt. 2.

Shea, *Transistor Audio Amplifiers*, pp. 4-9.

Webster, W. M. "On the Variation of Junction-Transistor Current-Amplifi-



cation Factor with Emitter Current." *Proc. I. R. E.*, XLII (June 1954), p. 914.

Low-Power Curves

Bevitt, pp. 38-40.

Shea, *Transistor Audio Amplifiers*, pp. 4-9.

Variations in the Curves — Temperature

Bevitt, pp. 194-202.

Aging and Life

Bevitt, p. 46.

Kiver, pp. 63-64.

Scott, pp. 129-133.

Noise — General

Bennett, W. R. "Methods of Solving Noise Problems." *Proc. I. R. E.*, XLIV (May 1956), p. 609.

Bennett, W. R. "Noise." *Electronics*: Characteristics and Origins of Noise, (Mar. 1956) p. 154; Equipment for Generating Noise, (Apr. 1956) p. 134; Techniques for Measuring Noise, (May 1956) p. 162; Designing Low-Noise Equipment, (June 1956) p. 154; Reducing Noise in Communications Systems, (Jul. 1956) p. 148.

Pierce, J. R. "Physical Sources of Noise." *Proc. I. R. E.*, XLIV (May 1956), p. 601.

van der Ziel, A. *Noise.* Englewood Cliffs, N. J.: Prentice Hall, 1954.

Noise — Transistor

Bargellini, P. M., and Herscher, M. B. "Investigations of Noise in Audio Frequency Amplifiers Using Junction Transistors." *Proc. I. R. E.*, XLIII (Feb. 1955), p. 217.

Bevitt, pp. 185-194.

Lo, pp. 122-130.

Shea, *Principles of Transistor Circuits*, Chapt. 20.

Shea, *Transistor Audio Amplifiers*, pp. 124-129.

Starke, H. F. *The Measurement of Transistor Noise.* Newton, Mass.: Raytheon Manufacturing Co. Pamphlet, 1955.

High-Frequency Limitations

Bevitt, pp. 44-46.

Lo, Chapts. 7 and 8.

Shea, *Principles of Transistor Circuits*, Chapt. 9.

TRACKING ERROR

Continued from page 27

but is 3¼° less than optimum for LP's. The *distortion* level for 78's is thus much less than for LP's. The author uses separate GE cartridges for 33 and 78-rpm records, and there is sufficient clearance to increase β to β_0 for LP's as described above. There are several possibilities: 1) increase β to β_0 for LP's for both cartridges, using the corresponding D. Then m_1 (78) = 1.92°/in., equivalent to 0.82 for an LP; 2) change the values of β to their respective values of β_0, thus obtaining optimum performance in both cases. This is an attractive possibility, but it involves accommodating the difference in optimum overhangs, which amounts to 0.21 in.; 3) if $\beta = 22.5°$ and $D = 0.62$ in., m_1 (33) = 0, and $m_2 = -m_{min}$ = 0.39°/in, and m_1 (78) = 1.42°/in., equivalent to 0.6; 4) if we require m_1 (78) = $-$ 2.35 m_{min} (33), so that maximum *distortion* on 78's equals that on LP's, the solution is $\beta = 21.8°$ and $D = 0.57$ in., for which $m_2 = -m_{min} = 0.44$ for LP's; 5) if we change β to β_0 (33) with $D = D_0 = 0.71$ in., then with the same value of D but with $\beta = 25.6°$ for 78's, m_1 (78) = $-m_{min}$ = 0.91, corresponding to 0.39, while m_{max} (33) = m_0 = 0.31°/in.

Clearly there are as many different compromises as there are ways of evaluating the relative importance of conflicting considerations. The author will consider his purpose achieved if he has succeeded in acquainting the reader with the nature and importance of tracking distortion, and in providing the means to minimize it.

STEREO SPEAKER

Continued from page 18

If, however, the speakers are rather widely spaced, as in Fig. 4C, or along the long wall of the room, as in Fig. 4G, and the listener is well back in the room (subtended angle still around 45° or so) he will be less likely to distinguish between single and stereo. This is because the reverberation characteristics of the average room "mix" or diffuse the sound, and the reflected sound from the walls, floor, and ceiling is nearly as loud as the direct sound from the speakers, thus helping to confuse the situation. (Author's note: try cupping your hands behind both ears in any listening room. You will suddenly cut out much of the reflected sound and hear primarily the direct sound from the speaker, or speakers.)

The character of the sound may give a clue as to whether it is single-channel or stereo. The recording of the ping-pong game will show up the difference between single and stereo under almost any conditions, while the difference between the single and stereo recording of a glee club may tax even the most experienced listener's judgment.

What was it we were driving at? Just this: realism in reproduced sound. If we were to place the speakers in the corners of the room, as in Fig. 4C, there is no question that we'd get ping-pong separation, and in large doses. But is it natural? Not even for ping-pong! Such separation, with respect to the room and to the listener, gives us two individual point sources, just the kind of source that we've been trying to get away from all the while. And such wide separation leaves a "hole" in the middle of our orchestra, too. Conversely, if we move the speakers too close together, as in Fig. 4D, we may as well stick to single-channel hi-fi, because the two sources become almost one. The reader by this time has either given up, or reached the same conclusions our committee did: the answer lies somewhere between these two extremes. After trying all the combinations and angles we could think of from flat against the wall to 90°, from close spacing to wide, we built a single cabinet with the two speakers mounted at about a 12° angle therein, as in Fig. 4E, and spaced about 3 ft. on centers. Each wide-range speaker was housed in its own separate enclosure within the main cabinet. This gave the dispersion and blending to enhance single-channel reproduction from both speakers, and it also gave reasonable spacing for definition, yet blending between the two stereo channels. Such spacing avoided the hole-in-the-center effect, and facilitated construction of a single furniture unit in which could be housed all the usual signal sources: tuner, record changer, and tape recorder, plus a connection for TV sound. It is shown in Fig. 6. A cabinet with 4-foot spacing between speakers was also tried.

These experimental cabinets were placed in the various living-room environments and were shown to a wider selection of listeners. The difference in sound quality between the 3-foot and 4-foot speaker spacing was questionable. The size of the cabinet then became the major consideration. These conclusions on stereo speaker placement led to the introduction of the Ampex Model A432, first shown in 1956.

In conclusion, it can be stated that if one wishes to experience the ultimate in sound reproduction, at the present state of the art, he can be assured of an emotional experience far beyond present day hi-fi if he equips his home with true two-channel stereo sound. While one would expect to pay more for such a system than for a simple single-channel system, it will be found that a good stereo system is a definite economic possibility.

The best (optimum) placement of speakers within any given room is necessarily a problem of subjective determination on the part of the expert listener, and will depend mutually on his good taste, judgment, skill, and the furnishings of his room. But for the great body of music listeners a single stereo cabinet, properly designed to fit the conditions usually found in most homes, will give many hours of enjoyment with single-channel sound from radio and records, as well as the sounds of today and tomorrow on recorded stereophonic tapes.

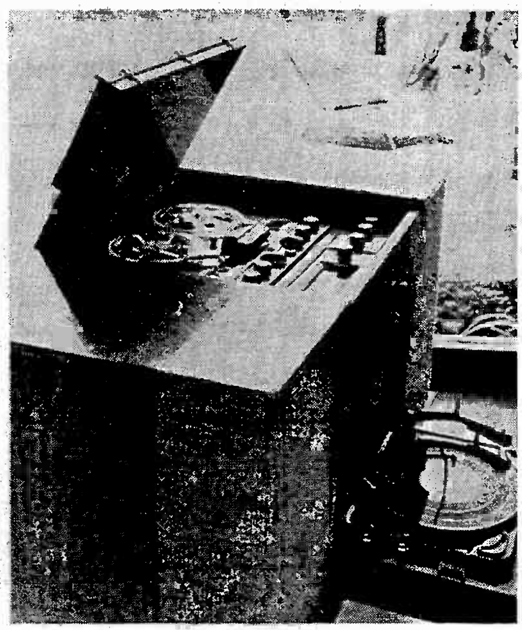


Fig. 6. Complete packaged stereo system.

The RUMBLE *Seat*

Gentlemen:

I should like to point out some errors in Mr. Crowhurst's article in AUDIOCRAFT for October, and also to point out some engineering flaws in the amplifier design that he uses for an illustration.

On page 28, he equates output-impedance reduction with gain reduction. Actually the gain reduction is much less than the impedance reduction. In the case stated, the impedance reduction is nearer to 50. I refer you to pages 310 and 311 of the *Radiotron Designer's Handbook*, 4th Edition. On page 30, it is stated that the 9,000-ohm plate-to-plate load resistance is the equivalent of 2,250 ohms from each plate to ground. Actually it is the equivalent of 4,500 ohms from each plate to ground, or 2,250 ohms from one plate to ground.

It is not good engineering to have a low-frequency rolloff inside the over-all feedback loop which appreciably reduces the gain within the desired pass band. Such a gain reduction results in less feedback and, therefore, more distortion. Although in the amplifier used as an example there is a great deal of feedback left at 50 cps, most of it is from the primary of the output transformer where it is of no use in reducing the distortion caused by the output transformer itself. The highly desirable, sharp rolloff below 20 cps should be obtained outside the main feedback loop.

The brute-force stabilization method used for the high-frequency end also reduces the feedback available at frequencies above a few kilocycles with the resultant increase of distortion. Some of these deficiencies might be tolerated if they were necessary, but they are not. A well-designed Williamson-type will out-perform the amplifier illustrated and will be much easier to stabilize.

Of course, it may be stated that the circuit used is meant only to demonstrate the method of application of feedback. However, it is very likely that a number of your readers will consider that it is the last word in a well-designed amplifier and will construct one. I think that they should be protected from this.

W. B. Bernard
Arlington, Va.

Reply:

One thing about Mr. Bernard's letter pleases me; it gives me the opportunity to clarify something about my design that appeared in the October issue. While I strongly contest what he raises as errors and flaws, he is right about one thing. I did select the circuit to demonstrate design method, rather than because it is the last word in amplifier design. For this reason, I chose a circuit that would illustrate a representative cross section of design problems— not one intended to be the ultimate in quality. Now to answer his so-called flaws.

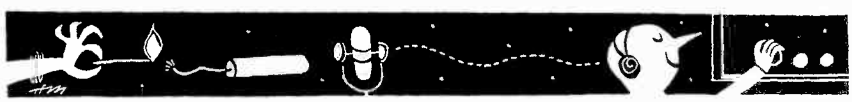
Radiotron Designer's Handbook says nothing contrary to my article. It depends what gain reduction and what impedance reduction we are talking about. We must adhere to the same terms of reference. If we refer to the impedance *and* gain reduction with resistance load connected, then, with any pentode-type output stage, 20 db of over-all feedback will result in an output impedance of approximately a tenth of the load impedance — *because the load impedance is a principal factor in determining the gain without feedback.*

This, it is true, *can* be regarded as a reduction in source impedance by a factor of about 50, but note that this is *source* impedance, not the *circuit* impedance. It will be the circuit impedance when the load is removed. This is precisely why my approach is important; the load we normally use is not the nominal constant resistance of the design.

His next criticism gives the impression he hasn't read parts I to IV. Academically, I agree it would be better to word the sentence in question, "from *one* plate to ground", but my wording should not cause difficulty to anyone who read the article dealing with composite load lines. One could also say "from *both* plates to ground", except that involves a visual difficulty because the plates are connected to opposite ends of the transformer primary.

I'm afraid I must disagree very definitely with what Mr. Bernard defines as good engineering. Had he used the term *amplifier promotion*, I would agree, because the stress on ever wider response is largely promulgated by manufacturers' promotion people, not engineers.

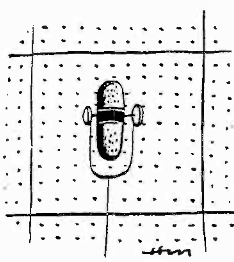
The latter part of his third paragraph shows that he shares a die-hard erroneous impression. Reduction of low-frequency output-transformer distortion



does not depend on taking feedback from its secondary. Also an adequate, sharp rolloff outside the feedback is more difficult (and more expensive) to obtain. It lacks, too, some of the performance safeguards provided by including it in the over-all design.

His last sentence in paragraph 4 is impressive, but vague: what does he mean by *outperform*? What are the determining criteria? Tests with resistance load, or practical listening tests? It is true that the Williamson can be stabilized according to the load used, but the circuit I developed, and any modern circuit (Mr. Williamson never claimed his circuit to be the last word), should not need to be stabilized.

His reference to distortion shows he does not know certain facts about amplifier performance. These I cannot deal with in detail here, but I will give a brief statement to satisfy readers who may wonder what the answer is. More *harmonic* distortion of the lower frequencies will not be as distressing (provided, of course, that it is kept within bounds by good design) as some of the things caused by preserving full feedback to 20 cps: certain forms of IM, for example, that do not show up on test, but are *very* audible. My method avoids appreciable increase of measurable IM between low and high frequencies, which usually accompanies the increased harmonics, without giving rise to the other kinds. As for the high-frequency end, who wants to make sure that frequencies only the birds can hear don't get distorted? Why bother trying



to reproduce them? They aren't on the recordings. If they do appear, they are spurious anyway. Unnecessary extension of the high end causes other troubles too; particularly increased IM of the "mushy" variety that spoils program reproduction, but does not show up on test.

Finally, he regards my method as unnecessary and the "time-honored" approach (still copying the Williamson) as normal. I venture to predict that this point of view will be obsolete in a few years. Sound design, based on optimum criteria, will become the normal practice, while extension of response by trick design methods will be seen unnecessary.

Norman H. Crowhurst
New York, N. Y.

AUDIOCRAFT
Sound Sales Directory

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

KEY TO PRODUCTS HANDLED
1 Audio system components
2 Speakers and enclosures
3 Records and record accessories
4 Tape recorders
5 Pre-recorded tape
6 Radio hardware
7 Tools, wood
8 Audio parts
9 Microphones
10 Books
11 Test equipment
A series of items numbered consecutively is identified by a hyphen between the first and last numbers. Thus, 1-6 indicates 1 through 6; 8-11 indicates 8, 9, 10, and 11.

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

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

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

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

GEORGIA
Atlanta
Baker Fidelity Corp.
1140 Peachtree St., N. E.
TRinity 5-2156 — **1-6, 8, 9**

ILLINOIS
Chicago
Allied Radio Corp.
100 N. Western Ave.
HAymarket 1-6800 — **1-11**

Evergreen Allied High Fidelity, Inc.
2025 W. 95th St.
BEverly 8-1067 — **1-11**

Evanston
Allied High Fidelity Stores, Inc.
602 Davis St.
DAvis 8-8822 — **1-11**

Oak Park
West Suburban Allied High Fidelity, Inc.
7055 W. North Ave.
EStebrook 9-4281 — **1-11**

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

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

MISSOURI
St. Louis
The High Fidelity Showroom
6383 Clayton Rd.
Parkview 1-6500 — **15, 9**

NEW JERSEY
New Brunswick
Monmouth Music House
215 Bevier Road
CHarter 9-5130 — **1, 2, 4, 5, 9**

NEW YORK
Albany
Audio-Video Corporation
324 Central Ave.
3-1167 — **1-5, 8-10**

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

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

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

Ten Cate Associates
6128 Morton St.
Germantown 8-8500 — **1-5, 8-10**

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

WEST VIRGINIA
Charleston
Electronic Specialty Co.
Virginia Street W., At Park Avenue
3-5656 — **1-11**

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

BASIC ELECTRONICS

Continued from page 31

heavily damped — the vibrations will soon die out of themselves, or it will take a lot of work to keep them going.

In electrical circuits an inductor has a property similar to mechanical mass. It tends to prevent rapid changes in charge flow, just as a heavy object is resistant to sudden changes in its position when at rest, or in its velocity when in motion. A capacitor is analogous to a compliance (such as a spring), which has very little resistance to compression or expansion at first, but whose resistance to further compression increases proportionally to the amount of compression. An uncharged capacitor can be charged very easily at first, but less and less easily as the amount of charge increases.

When both inductance and capacitance are present in a circuit, therefore, an electrical resonance occurs that is closely analogous to mechanical resonance. If a charge of electrons is set circulating in the circuit and the source voltage is short-circuited, the inductor will tend to keep electrons flowing in the same direction. This draws electrons from one capacitor plate and piles them up on the other plate, until the capacitor is charged so fully that the collapsing flux in the inductor cannot charge it any more. After the charge stops flowing the capacitor begins to discharge, forcing the electrons through the choke in the other direction and building up its flux field again. When the capacitor is fully discharged the instantaneous charge flow has reached a maximum rate; again, the choke tends to keep it flowing in the same direction, which charges the capacitor to an opposite polarity. When this charge has built up to maximum, the electron flow ceases once more, and then the capacitor begins to discharge through the inductor. The result is a continuously shifting flow of charge from one capacitor plate to the other, in a slowly subsiding oscillation. If either the capacitor or the inductor is increased in value, the oscillation will be slower — because it will take the capacitor longer to charge, or longer to build up electron flow through the inductor. The smaller the circuit resistance, the longer the oscillations will continue: each time the charge flows through the resistance, some energy is consumed, and a large resistance will consume more energy per cycle. Therefore electrical resistance is analogous to mechanical friction, and it serves a similar damping function.

When we reconnect the voltage source, it may or may not be of appropriate frequency to stimulate the natural oscillatory period. If it is at or near the resonant frequency, the applied voltage will reinforce strongly this oscillation, and the circulating current may become very high if there is little resistance in the circuit. Maximum circulating current is achieved, and maximum voltages exist across the reactive components when the inductive reactance completely cancels the capacitive reactance — that is, when the two are equal. Then the current is limited only by the resistance. This resonant frequency, f_o, is that at which

$$2\pi fL = \frac{1}{2\pi fC},$$

$$f^2 = \frac{1}{4\pi^2 LC},$$

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

Extremely high currents and reactive voltages can be obtained when the source resistance and circuit resistance is kept low. There is a limit beyond which it is impossible to go, however, because the inductor winding must have some resistance. The *ratio* of inductor reactance at resonance to its DC resistance is known as its quality factor, or Q. It is sometimes known as the magnification factor, because it is the number by which, in a resonant circuit, the source voltage is multiplied to obtain the voltage across either reactance.

GROUNDED EAR

Continued from page 5

invested in good speakers and a few hours of simple carpentry can yield extremely good sound.

New B&O Mike

The B&O microphone which Gordon Holt and I both liked now comes in an improved form, with an internal switch for three different input impedances: 50 ohms, 250 ohms, and Hi-Z. It also has more thorough shielding to provide a lower hum level, especially in the Hi-Z position. The changes should make this excellent mike even more useful.

There is also a binaural mount with a sound baffle for two B&O mikes separated by less than a foot, for a new type of stereophonic recording and reproducing technique.

READERS' FORUM

Continued from page 15

of elementary theory and application. I think many of your readers would be intersted in it. The book can be ordered from the Superintendent of Documents, Government Printing Office, Washington 25, D. C.

Virgil McCarley
Spring Valley, Calif.

Gentlemen:

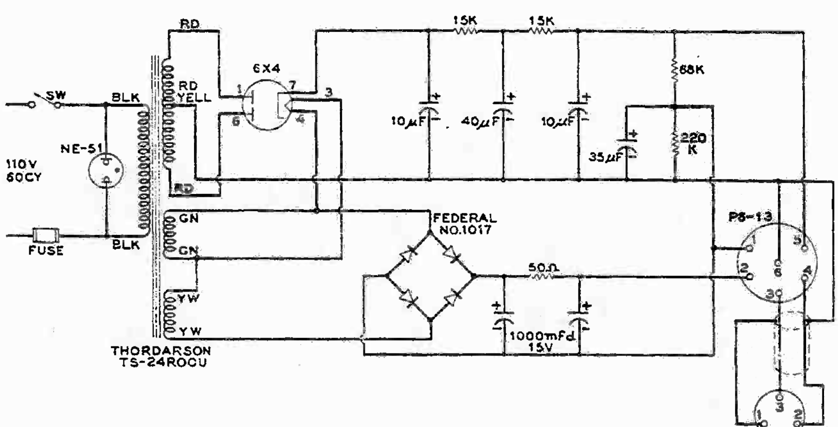
I would like to add my hearty "second" to Mr. Holt's praises of the Capps condenser microphone in the October issue of AUDIOCRAFT. We have been using a Capps CM-2250-A (250-ohm model) in our studio for just about a year now and are still amazed with the ultra-natural quality it produces.

For the serious audiophile who would like to own a Capps condenser system but can't meet the $225 price, home construction of the power supply and interconnecting cable will produce a saving in the neighborhood of $70. The microphone alone (head and preamp) can be purchased for $114. The power supply (see diagram) can be home-built for around $25. The interconnecting cable with connectors will run approximately $13.

Those who might be interested in actually hearing the fine performance of this unit can do so by listening to any Cook record or Audiosphere tape. Both of these companies use the Capps exclusively.

As for the mike's breaking up from overloading, etc., I doubt that this would occur as I understand it has even been used for checking the noise level of jet engines.

Tom H. Jones
Rochester Transcription Service
Rochester, Minn.



Schematic diagram of the Capps condenser microphone power supply.

WOODCRAFTER

Continued from page 11

time consumed in hand sanding. Perhaps his first taste of power sanding comes with the sanding disc for his portable electric drill. While this is limited in its application, he will find a variety of motorized equipment designed specifically for sanding. Here are a few examples:

Belt Sander (Fig. 4). In this machine a continuous abrasive belt travels over a fixed table and can be operated vertically, horizontally, or at any degree in between. It is excellent for sanding flat work and is extensively used for finish-

Courtesy Skil Corporation.

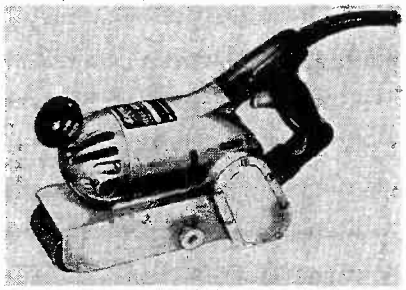


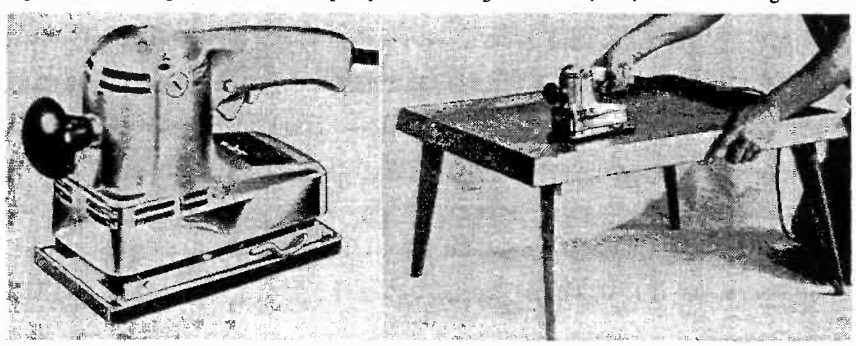
Fig. 6. The portable belt sander.

ing, mitering, and chamfering operations on both wood and metal.

Disc Sander (Fig. 5). One of the most tedious jobs to do by hand—finishing plywood edges—is a simple task for this type of sander which also smooths end grain and squares up small stock. In fact, it does much the same work as a belt sander with one exception—it is not as effective on flat surfaces.

Portable Belt Sander (Fig. 6). This is a most convenient sander for cabinet

Fig. 7. Finishing sander has rapidly oscillating motion for fine smoothing work.



Courtesy Skil Corporation.

TRADERS' MARKETPLACE

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work since you can take the tool to the work. It comes in various sizes with sanding bases from 2 in. to 4½ in.; the 3-inch portable belt sander is quite satisfactory for the average home workshop. Belts are obtainable in all standard abrasives and grit sizes.

Finishing Sander (Fig. 7). As its name implies, this tool is used primarily for a fine finish after a project has been completed. Of the many types manufactured, most operate on the oscillating principle and, with the addition of special pads, can be used for rubbing finishes. It is possible to convert a portable electric drill into a finishing sander with an attachment now on the market.

There are many other accessories available which make it possible to perform certain sanding operations with other power tools in the workshop. For the drill press, there are sanding drums for sanding the edges of curved or straight stock. For the scroll saw, there are sanding sleeves for finishing concave, convex, or flat surfaces — excellent for sanding inside curves.

Whether your sanding job is done by power or hand, give it the attention it deserves and your finished efforts will be richly rewarded with compliments and self-satisfaction.

In the next issue Mr. Bowe will discuss cabinet hardware.

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The music is highly romantic, ranging from heavy lushness through a lilting waltz to hair-raising bombast. The sound should be rather heavy and sumptuous, with a high degree of definition and yet a moderate amount of echo. The trouble, of course, is that by the time the audience gets seated there won't be any excess echo, and getting close enough to the orchestra for the requisite definition means reducing the echo too much.

For those who own multi-channel microphone input mixers, the solution is fairly simple: a single unidirectional microphone suspended well out in the hall at a height of about 15 ft., aimed at the stage, and an omnidirectional microphone about 15 ft. directly over the front of the orchestra. The rear microphone then does most of the pick-up, giving the requisite echo and blending, while the close microphone is mixed in at a lower volume level which is just adequate to provide crispness. Tilting the rear mike downward slightly so that it aims at the heads of the front-row patrons will minimize pickup of the reflected highs from walls and ceiling that give the hall its hard sound.

For the single-mike recordist, the problem is a little more difficult, but here again a unidirectional mike will probably provide the best means for a solution. Closeness is the only thing that will give definition to the sound, yet closeness normally means that the direct sound will overpower the echo, which we don't want in this case. The unidirectional microphone permits us to get both closeness *and* a good reflected-to-direct sound ratio. With the microphone suspended over the conductor's head and aimed *out at the audience*, the mike's rear discrimination will reduce the level of the orchestra to a point at which it is comparable to what it would be farther out in the hall. Yet the close proximity of the mike to the instruments will maintain the definition that is needed, while the slight rolling off of highs that will result (because of frequency-selective directivity) will help add lushness to the sound. And since the microphone's diaphragm is aimed squarely into the audience (from whence cometh the least reflected highs), the acoustic character of the hall will seem to be greatly improved in the recording.

When setting up microphone coverage for a live performance, don't forget that the addition of an audience will make a tremendous difference in both

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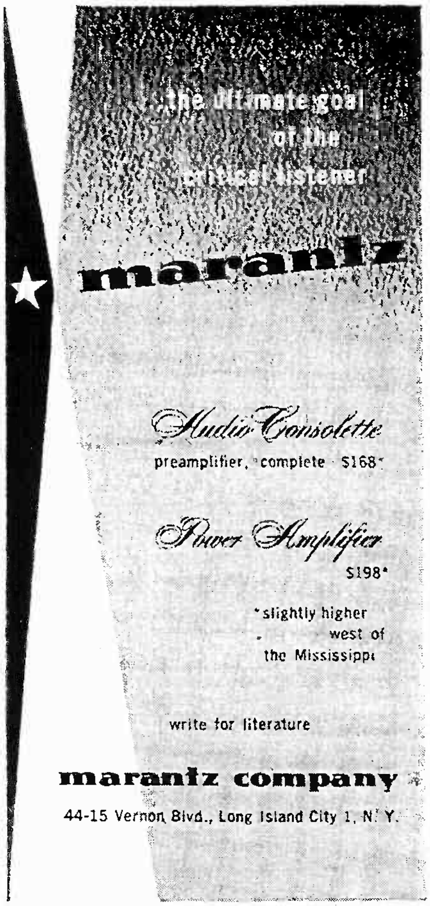

TAPE NEWS

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the reverberation time and the acoustical nature of the hall. The audience invariably reduces the reverberation and softens the sound, so make allowances for this when setting up in the empty hall. It is always a good idea when a recording job like this is approaching to attend a few of the rehearsals and run off test recordings with the mike(s) in several different locations. It takes only a few inches or a few degrees difference (with directional mikes) to effect a large change in the sound pick-up; make a careful note of each trial position so the best one can be duplicated at the performance.

I'll have more to say about mike technique next month. Meanwhile, I'm still indignant about a letter I received from some bewildered individual who reports that a dealer told him his binaural tape recorder was obsolete because everyone was currently using stereophonic recorders. Being of a cautious sort, the man dropped us a line about it before he traded his "obsolete" recorder for a modern stereo job, and was promptly advised that said dealer was evidently confused (or ingeniously clever), because a binaural recorder and a stereo recorder are one and the same animal.

The only difference between binaural and stereo sound is in the way they are miked and played back. Binaural recording makes use of two microphones spaced the same distance apart as the human ears, and they are often mounted on both sides of a dummy head to further simulate the human hearing mechanism. Two parallel simultaneous recordings are made, and they are played back *through earphones* to maintain total separation of one channel from the other. The effect is startlingly realistic, since it amounts to transporting the listener right into the concert hall. Binaural sound has never enjoyed much popularity, because listening to it is a singularly antisocial form of entertainment and often produces sore ear lobes.

A stereo recording utilizes two microphones spaced 10 to 30 ft. apart (depending upon the size of the performing group), and is intended for playback through two loudspeakers spaced several feet apart. It, in effect, transports the concert hall into the living room.

Binaural recordings are not supposed to be played on a stereo system (because the source separation was not great enough at the recording end), nor are stereo recordings intended for headphone listening (because the wide mike separation exaggerates the original spatial relationships), but they are both made and played on the same type of recording machine.

ADVERTISING INDEX