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audiocraft

THE MAGAZINE FOR THE HI-FI HOBBYIST

THIS MONTH'S COVER: an audio kit builder inserts tubes in his newly completed Heath Audio Analyzer Kit. Photo by Phil Geraci

NOVEMBER 1957
Volume 2 *Number 11*

ADVERTISING

Main Office — Claire Eddings, The Publishing House, Great Barrington, Mass. Telephone: Great Barrington 1300.
New York — Frederick H. Wallin, 280 Madison Ave., New York, N.Y. Telephone: Murray Hill 3-4449, 3-4450.
Chicago — John R. Rutherford & Associates, Inc., 230 East Ohio St., Chicago, Ill. Telephone: Whitehall 4-6715.
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Audiocraft Magazine is published monthly by Audiocom, Inc., in Great Barrington, Mass. Telephone: Great Barrington 1300. Editorial, publication, and circulation offices at: The Publishing House, Great Barrington, Mass. Subscriptions: $4.00 per year in the United States and Canada. Single copies: 35 cents each. Editorial contributions will be welcomed by the editor. Payment for articles accepted will be arranged prior to publication. Unsolicited manuscripts should be accompanied by return postage. Entered as second-class matter October 1, 1955, at the post office, Great Barrington, Mass., under the act of March 3, 1879. Additional entry at the post office, Pittsfield, Mass. Printed in the U. S. A. by the Ben Franklin Press, Pittsfield, Mass.

The Grounded Ear

by Joseph Marshall

Shure Dynetic Pickup

It is not easy to be original in any field today, but it is especially difficult in the field of phonograph pickups, where it almost seems as if all possibilities had been explored and developed. It is even more remarkable for an organization, hitherto identified with low-priced components of good but not superb quality, to make the jump into the class of "best available" hi-fi components. That is exactly what Shure has done with its Dynetic pickup, which is not only ingenious in principle but represents a high degree of engineering refinement and an extraordinarily close approach to the ideals of phono reproduction.

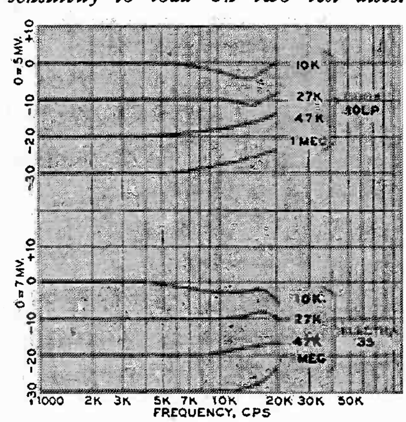
The Dynetic is based on the "moving magnet"; the stylus moves a *magnet* within a coil, inducing in the coil a current proportional to the stylus's movement. With the high-power magnets available today, the movement can be made extremely light and sensitive. The stylus is coupled to the magnet through a short cantilever arm damped in an elastomer composition. The stylus, incidentally, is easily removed and the design is such that, despite the ease of changing the stylus, its placement can be very precise. The simplicity of the design makes possible one of the highest compliances and lowest stylus masses achieved so far.

Having worked out a successful adaptation of the new principle, it must have been a big temptation to rush the design into a commercial cartridge. If so, Shure resisted it, using the new design simply as the foundation for an integrated system of unusual characteristics and extremely high performance.

For one thing, the Dynetic reproducer will track most records on good turntables with a stylus force of only 1 gram. Thus, for example, with a force of 1 gram on my Scott turntable, the Dynetic tracks the new Cook test record as well as any other pickup I have used except the Ferranti, which is still tops for high-amplitude low frequencies. Increasing the force to 1½ or 2 grams improves tracking, but mostly on records heavily overcut in the middle and high portions. Only one other pickup, the Weathers FM, has hitherto permitted the use of such low tracking force, and in that instance a brush mounted alongside the stylus, the hair riding the grooves, has been necessary to help maintain tracking. The Dynetic, on the other hand, tracks stably, even when the turntable is tilted off level as much as 30°. Vibration and shock can cause the stylus to jump, but, if the turntable is damped either by floating springs or a rubber cushion, no trouble should be experienced. If vibration is troublesome, increasing the tracking force to 2 grams will provide higher stability. There is no tendency for the pickup to skate, even with a turntable off level or with a warped record.

The force of 1 gram would approach or exceed the theoretical ideal for a 1-mil stylus on vinyl material from the standpoint of groove deformation. However, Shure has chosen to take advantage of the low tracking force by using a stylus of 0.7-mil radius to obtain better tracing of the high frequencies. This is an excellent compromise between 1 mil and ½ mil. Theoretically, a ½-mil stylus requires one-fourth the force of a 1-mil stylus for the same deformation pressure; the 0.7-mil requires only half the force. Thus the Dynetic with the 0.7-mil stylus and 1 gram of force should produce considerably less record damage than the conventional cartridge with a 1-mil stylus and 4 or 5 grams of force; and the smaller stylus produces a superior high-frequency transient response.

High-end response of the Dynetic shows sensitivity to load on two test discs.



The frequency response of the Dynetic is right up with that of the best available pickups. It is quite sensitive to loading up to 50 K. Shure recommends a load of 10 K and supplies a resistor to provide such a load. Increasing the load increases the response above 5,000 cps. The graph curves were measured with a 3-foot length of shielded phono cable attached to the twisted pair supplied with the pickup. Under these particular circumstances, it will be noted, a load of 27 K turned out to provide the flattest curve. I might point out that this pickup is extremely sensitive to even the slightest variation in amplitude. The Cook is within +1 db up to 10 Kc; variations as small as ¼ db are clearly noticeable with the Dynetic. These are obviously fine curves; the low end is absolutely flat to 20 cps. Shure says that there is considerable response below 20 cps and above 20,000 cps. The good low-end response is indicated by the fact that, with a speaker system which goes down to 20 cps or lower, it is possible to feel vibration due to wow, rumble, or LF flutter in the subaudible range below the usual rumble frequencies. There is no resonance whatever, however, down to 18 cps — as low as I can measure. It is notable also that there appears to be no peak of any significance due to groove-needle resonance below 20,000 cps.

A change in loading produces a smooth rise in response above 8 or 10 Kc, rather like that achievable with a good tone control or equalizer. Thus the use of a variable pot for a load would provide a means of adjusting the response to compensate for lead length and preamp input characteristics, with little danger of producing a resonant peak in the useful audible range.

I noted that the pickup is sensitive to both minor variations in amplitude and to subsonic rumble, flutter, or shock vibration. It appears also to be somewhat more sensitive than most to surface and electrostatic noise. Indeed, the sharpness of the reproduction of a periodic surface or electrostatic noise indicates fine transient response. Greater care will be required to keep records clean and destaticized; but this is a low price to pay for improved transient response.

The distortion is extremely low—on better records it is completely inaudible even at very high modulation levels—and the tendency of noise to ride the modulation (a type of IM distortion) is also low. The Dynetic pickup is not easily overdriven—quite the contrary—but there *are* records which will overdrive it even when it is adjusted for 2 grams force.

The arm is a refinement of the old Pickering, and represents one of the most successful designs yet. As in the Pickering, the head is hinged for vertical motion, but the pivots are jeweled. Tracking force is adjusted with a small counterweight attached to the head. The arm is adjustable for vertical height at the rear post, which revolves on jeweled pivot bearings. There is a counterweight at the end of the arm, but it is damped with a plastic cushion. The friction both vertically and horizontally is just correct for the cartridge. Horizontal friction, for example, is low enough to permit tracking at low stylus force, and yet not so low that the arm is actuated by the side-to-side motion of the stylus. The vertical compliance of the pickup and arm is also very good. There is no audible needle chatter, and very warped records are playable without any audible increase in distortion.

It is difficult to position an arm as light as this one on a record. Shure solves the problem with a button on top of the arm; the button is pressed to lift the pickup cartridge. I don't consider this a completely successful solution, though what could be done to improve it I cannot say offhand—a larger and slightly concave button top might help. It might also be helpful if the button could be locked in the "down" position; it is quite a strain for a nervous hand to hold the button down and move the arm into exact position. However, this imperfection is made harmless by the fact that with 1 gram of stylus force it is practically impossible to damage the record—even skidding the pickup across the record produces no visible damage.

Shure provides an excellent template for positioning the arm. The height is adjustable over a wide range to accommodate various turntables. However, the vertical lift of the head is not very great, and if the height is adjusted for the proper angle on a thick Microfusion disc, a warped thin disc will be unplayable. Hence, it is sometimes necessary to readjust the height of the arm. This is easy to do but is something of a nuisance. The arm is held at rest by magnetic attraction, and, because the arm is smaller and lighter than the Pickering, this system is thoroughly satisfactory.

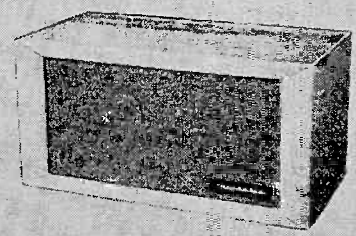
As for the sound, the highs are smooth and the low end, very clean; transient response is excellent. The Dynetic pickup is a fitting companion for the finest preamplifiers, amplifiers, and speakers.

HEATHKIT "BASIC RANGE" HIGH FIDELITY SPEAKER SYSTEM KIT

The very popular model SS-1 Speaker System provides amazing high fidelity performance for its size because it uses high-quality speakers, in an enclosure especially designed to receive them.

It features an 8" mid-range-woofer to cover from 50 to 1600 CPS, and a compression-type tweeter with flared horn to cover from 1600 to 12,000 CPS. Both speakers are by Jensen. The enclosure itself is a ducted-port bass-reflex unit, measuring 11½" H x 23" W x 11¾" D and is constructed of veneer-surfaced plywood, ½" thick. All parts are precut and predrilled for quick assembly.

Total frequency range is 50 to 12,000 CPS, within ±5 db. Impedance is 16 ohms. Operates with the "Range Extending" (SS-1B) speaker system kit later, if greater frequency range is desired. Shpg. Wt. 30 lbs. MODEL SS-1 $39.95

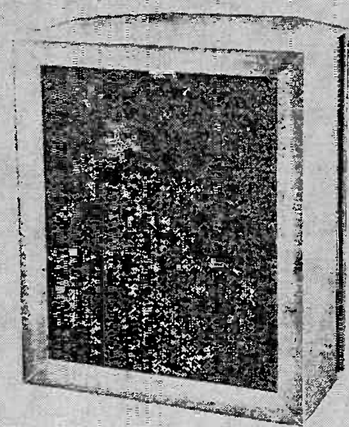


"BASIC" SPEAKER SYSTEM

HEATHKIT "RANGE EXTENDING" HIGH FIDELITY SPEAKER SYSTEM KIT

The SS-1B uses a 15" woofer and a small super-tweeter to supply very high and very low frequencies and fill out the response of the "Basic" (SS-1) speaker system at each end of the audio spectrum. The SS-1 and SS-1B, combined, provide an overall response of ±5 db from 35 to 16,000 CPS Kit includes circuit for crossover at 600, 1600 and 4000 CPS Impedance is 16 ohms, and power rating is 35 watts. Measures 29" H x 23" W x 17½" D, and is constructed of veneer-surfaced plywood, ¾" thick. Easy to build! Shpg. Wt. 80 lbs.

MODEL SS-1B $99.95



RANGE EXTENDER

...*and save!*

HEATHKITS

World's finest electronic equipment in kit form...

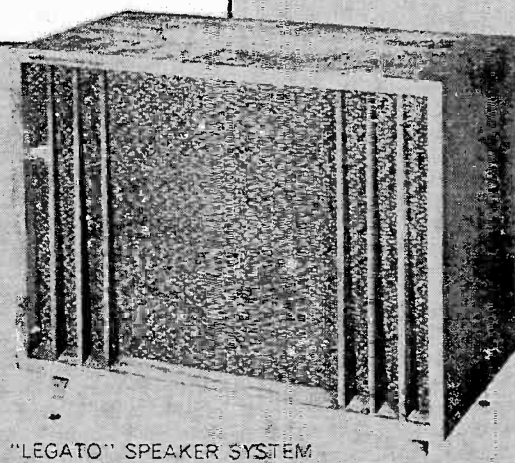
HEATHKIT "LEGATO" HIGH FIDELITY SPEAKER SYSTEM KIT

The fine quality of the Legato Speaker System Kit is matched only in the most expensive speaker systems available. The listening experience it can bring to you approaches the ultimate in esthetic satisfaction.

Frequency response is ±5 db 25 to 20,000 CPS. Two 15" theater-type Altec Lansing speakers cover 25 to 500 CPS, and an Altec Lansing high frequency driver with sectoral horn covers 500 to 20,000 CPS. A precise amount of phase shift in the crossover network brings the high-frequency channel into phase with the low-frequency channel to eliminate peaks or valleys at the crossover point. This is one reason for the mid-range "presence" so evident in this system design.

The attractively styled "contemporary" enclosure emphasizes simplicity of line and form to blend with all furnishings. Cabinet parts are precut and predrilled from ¾" veneer-surfaced plywood for easy assembly at home. Impedance is 16 ohms. Power rating is 50 watts for program material. Full, smooth frequency response assures you of outstanding high fidelity performance, and an unforgettable listening experience. Order HH-1-C (birch) for light finishes, or HH-1-CM (mahogany) for dark finishes. Shpg. Wt. 195 lbs.

MODELS HH-1-C or HH-1-CM $325.00 each



"LEGATO" SPEAKER SYSTEM

Pioneer in "do-it-yourself" electronics

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A subsidiary of Daystrom, Inc.
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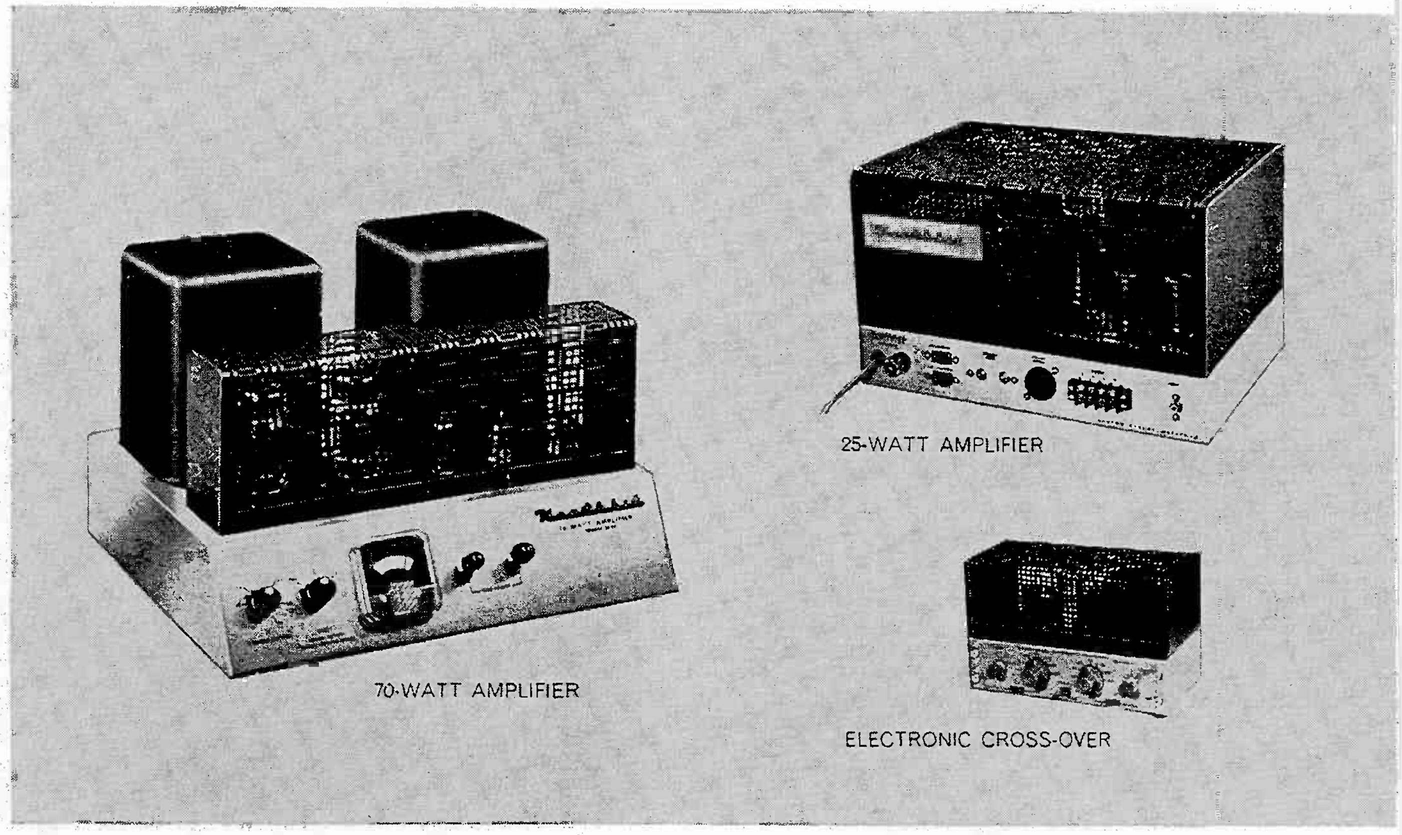


easy-to-build designs by HEATH *insure*

You get more comprehensive assembly instructions, higher quality circuit components, and more advanced design features, when you buy HEATH hi-fi!

HEATHKIT 70-WATT HIGH FIDELITY AMPLIFIER KIT

This new amplifier features extra power reserve, metered balance circuit, variable damping, and silicon-diode rectifiers, replacing vacuum tube rectifiers. A pair of 6550 tubes produce full 70-watt output with a special-design Peerless output transformer. A quick-change plug selects 4, 8 and 16 ohm or 70 volt output, and the correct feedback resistance. Variable damping optimizes performance for the speaker system of your choice. Frequency response at 1 watt is ±1 db from 5 CPS to 80 KC with controlled HF rolloff above 100 KC. Harmonic distortion at full output less than 2%, 20 to 20,000 CPS, and intermodulation distortion below 1% at this same level. Hum and noise are 88 db below full output. Variable damping from .5 to 10. Designed to use WA-P2 preamplifier. Express only. Shpg. Wt. 50 lbs. MODEL W-6M $109.95

HEATHKIT 25-WATT HIGH FIDELITY AMPLIFIER KIT

The 25-watt Heathkit model W-5M is rated "best buy" in its power class by independent critics! Faithful sound reproduction is assured with response of ±1 db from 5 to 160,000 CPS at 1 watt, and harmonic distortion below 1% at 25 watts, and IM distortion below 1% at 20 watts. Hum and noise are 99 db below rated output, assuring quiet, hum-free operation. Output taps are 4, 8 and 16 ohms. Employs KT66 tubes and Peerless output transformer. Designed to use WA-P2 preamplifier. Express only. Shpg. Wt. 31 lbs. MODEL W-5M $59.75

HEATHKIT ELECTRONIC CROSS-OVER KIT

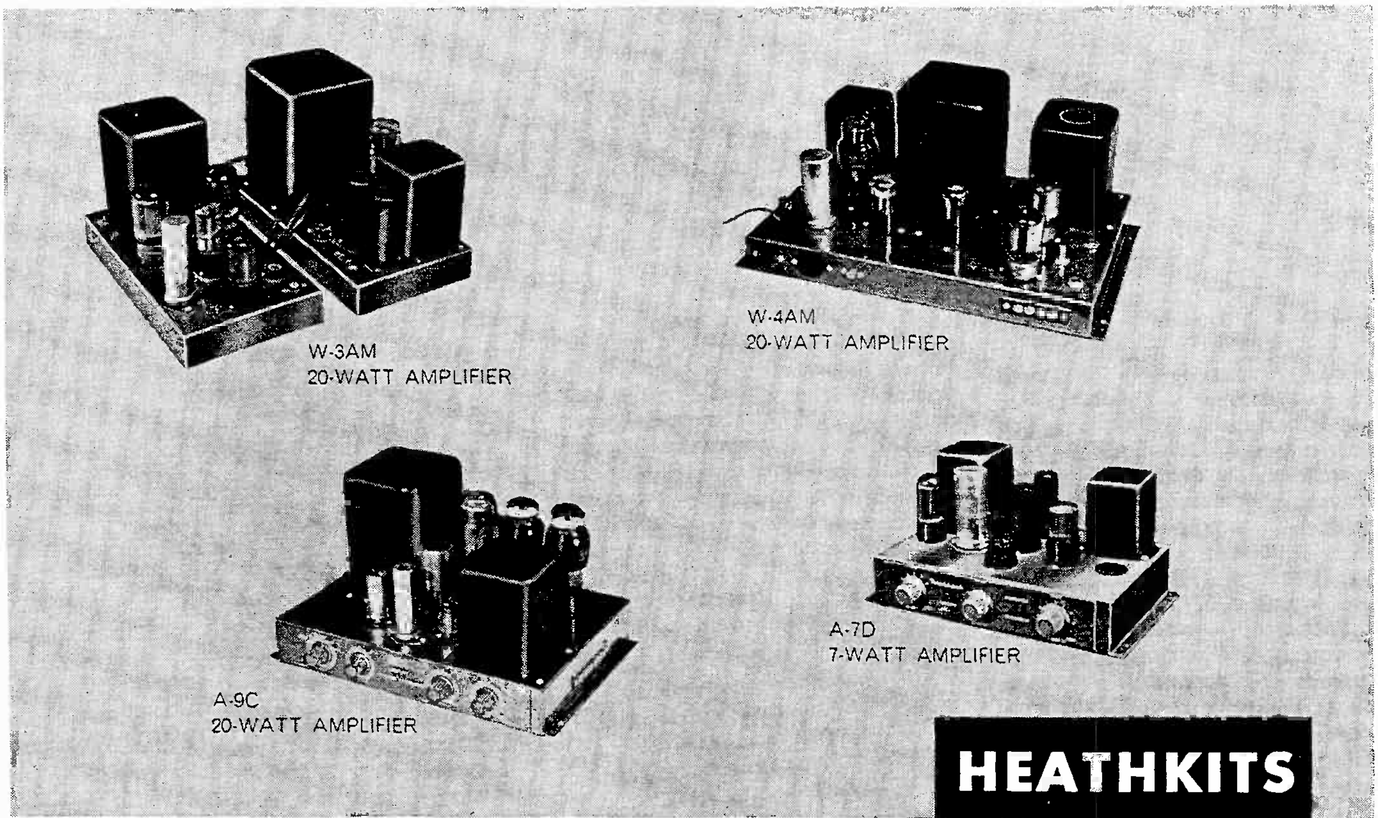
This device separates high and low frequencies electronically, so they may be fed through two separate amplifiers driving separate speakers. The XO-1 is used between the preamplifier and the main amplifiers. Separate amplification of high and low frequencies minimizes IM distortion. Crossover frequencies are selectable at 100, 200, 400, 700, 1200, 2000, and 3500 CPS. Separate level controls for high and low frequency channels. Attenuation is 12 db per octave. Shpg. Wt. 6 lbs.
MODEL XO-1 $18.95

HEATHKIT W-3AM HIGH FIDELITY AMPLIFIER KIT

Features of this fine Williamson-type amplifier include the famous Acrosound model TO-300 "ultralinear" transformer, and 5881 tubes for broad frequency response, low distortion, and low hum level. Response is ±1 db from 6 CPS to 150 KC at 1 watt. Harmonic distortion is below 1% and IM distortion below 1.3% at 20 watts. Hum and noise are 88 db below 20 watts. Provides output taps of 4, 8 or 16 ohms impedance. Designed to use WA-P2 preamplifier. Shpg. Wt. 29 lbs. MODEL W-3AM $49.75

HEATHKIT W-4AM HIGH FIDELITY AMPLIFIER KIT

A true Williamson-type circuit, featuring extended frequency response, low distortion, and low hum levels, this amplifier can give you fine listening enjoyment with a minimum investment. Uses 5881 tubes and a Chicago-standard output transformer. Frequency response is ±1 db from 10 CPS to 100 KC at 1 watt. Less than 1.5% harmonic distortion and 2.7% intermodulation at full 20 watt output. Hum and noise are 95 db below full output. Transformer tapped at 4, 8 or 16 ohms. Designed to use WA-P2 preamplifier. Shipped express only. Shpg. Wt. 28 lbs. MODEL W-4AM $39.75



HEATHKITS

World's finest electronic equipment in kit form...

...*top* HI-FI *performance*

HEATHKIT A-9C HIGH FIDELITY AMPLIFIER KIT

This amplifier incorporates its own preamplifier for self-contained operation. Provides 20 watt output using push-pull 6L6 tubes. True high fidelity for the home, or for PA applications. Four separate inputs—separate bass and treble controls—and volume control. Covers 20 to 20,000 CPS within ±1 db. Output transformer tapped at 4, 8, 16 and 500 ohms. Harmonic distortion less than 1% at 3 db below rated output. High quality sound at low cost!

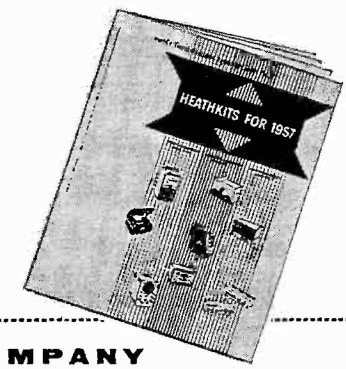
Shpg. Wt. 23 lbs. **MODEL A-9C $35.50**

HEATHKIT A-7D HIGH FIDELITY AMPLIFIER KIT

This is a true high fidelity amplifier even though its power is somewhat limited. Built-in preamplifier has separate bass and treble controls, and volume control. Frequency response is ±1½ db from 20 to 20,000 CPS, and distortion is held to surprisingly low level. Output transformer tapped at 4, 8 or 16 ohms. Easy to build, and a fine 7-watt performer for one just becoming interested in high fidelity. Shpg. Wt. 10 lbs. **MODEL A-7D $17.95**

Model A-7E: Same as the above except with extra tube stage for added preamplification. Two switch-selected inputs, RIAA compensation, and plenty of gain for low-level cartridges. Shpg. Wt. 10 lbs. **$19.95**

HOW TO ORDER . . .
Just identify the kit you desire by its model number and send check or money order to address below. Don't hesitate to ask about HEATH TIME PAYMENT PLAN.



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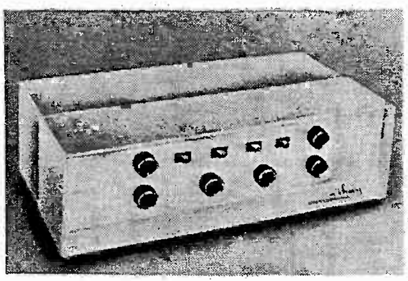
Audionews

ARKAY LINE

A new stereo-binaural high-fidelity component line available in kit form was announced by Arkay recently. The new line consists of the *Model SA-25* stereo preamp-amplifier, the *Model SP-6* stereo preamp, and the *Model ST-11* stereo FM-AM tuner.

The Arkay SA-25 features a dual-channel preamplifier-amplifier which drives its own 25-watt, linear, Williamson-type amplifier in conjunction with any present amplifier.

The SP-6 stereo preamp has the same features as the Model SA-25, less the amplifier. The controls of both models include equalization for all records, tuner, NARTB tape heads, and auxiliary equipment. A two-position lo-cut and hi-cut filter enables the user to filter out undesirable frequencies at either end of the band. Frequency response for



New Arkay stereo preamp-amplifier.

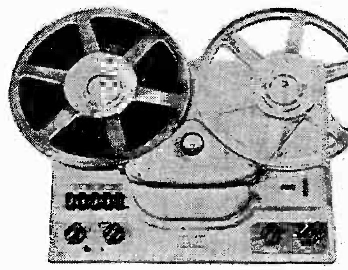
both models is said to be from 20 to 20,000 cps. The Model ST-11 stereo tuner may be used monaurally (AM or FM), if desired.

Retail prices for the Arkay stereo-binaural line are: Model SA-25 kit $59.95, wired and tested $89.95; Model ST-11 kit $47.95, wired and tested $69.95; Model SP-6 kit $34.95, wired and tested $47.95.

SWISS TAPE RECORDERS

Recently introduced were two new Swiss tape recorders, the *ReVox B-36-1* (single track) and *B-36-2* (dual track). Each of these machines features three motors; separate recording and playback heads; separate recording and playback amplifiers, permitting simultaneous and continuous monitoring; and 8-inch coaxial speaker. The price of the B-36-1 is $469; the B-36-2 is $449.

Speeds of these two ReVox recorders are 3¾ ips and 7½ ips. At 7½ ips, wow and flutter are said to be 0.15 to 0.20% RMS max., measured at either 5,000 or 3,000 cps. Frequency response at 7½ ips (1,000 cps reference) is reported to be 40 to 12,000 cps +1, —2 db, and rolloff to 15,000 cps, under —5 db. Over-all size, with carrying case,

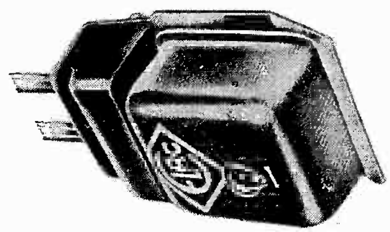


Swiss recorder has low flutter and wow.

is 18⅛ in. by 11 1/16 in. by 13⅜ in. A rack console is also available for custom installation.

MIRATWIN CARTRIDGE

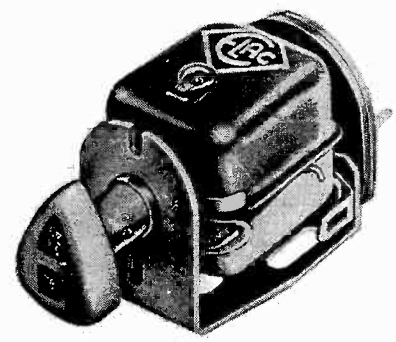
Miratwin Cartridges are now available in two types, the *MST-2* turnover cartridge and the *MST-1* single cartridge. The Miratwin is a variable-reluctance, magnetic cartridge which will accept all types of styli. Stylus replacement is accomplished simply by removing the old stylus with the fingertip and inserting the new one.



Miratwin styli can be replaced easily.

The MST-2 consists of two independent units mounted back to back in a turnover mount. It is claimed that there is no magnetic attraction between the two sections, and the one in use is said not to be affected by the other.

The Miratwin has output voltage of 55 mv for a stylus velocity of 10 cm/sec for microgroove records, and 45 mv for standard groove. Frequency response is said to be flat within 2 db from 30



Model MST-2 is Miratwin turnover unit.

to 20,000 cps at 33⅓ rpm, and within 4 db from 30 to 22,500 cps at 78 rpm.

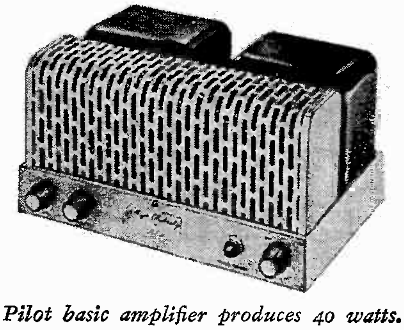
The recommended tracking force for the Miratwin cartridge when used in a record changer is 6 to 8 grams, but the unit is said to track well at low stylus force when used in transcription-type tone arms.

A short time ago, Audiogersh Corp. announced substantial price reductions in the prices of Miratwin cartridges. The manufacturer explained that the reductions were the result of increased demand for the cartridge and improved production methods.

NEW PILOT AMPLIFIER

Pilot Radio Corporation has announced the addition of a new basic amplifier, the *Model AA-908*, to its line of high-fidelity components.

Power output of the AA-908 is said to be 40 watts continuous and 80 watts peak. Frequency response is stated to be flat, ±0.1 db, from 20 to 20,000 cps with the speaker compensation switch in the FLAT position. Harmonic distortion is 0.5%, and hum level 90 db below 40 watts, according to the manufacturer. IM distortion is said to be 0.9% at 40 watts, 0.4% at 20 watts,



Pilot basic amplifier produces 40 watts.

and 0.15% at 10 watts. Output impedances are 0, 8, and 16 ohms. The unit has a detachable control escutcheon for convenient custom installation on a panel ⅜ in. thick, or less.

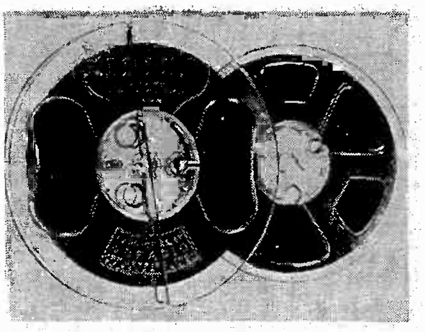
Retail selling price of the AA-908 amplifier is $125. West-coast prices are slightly higher.

NEW IRISH TAPE REEL

ORRadio Industries, Inc., has introduced a new 5¾-inch tape reel which is said to offer many advantages over the standard 5-inch reel. The new reel will carry the same footage as the 5-inch reel, according to the manufacturer.

The 5¾-inch reel has a professional-type hub 2¼ in. in diameter, the same size hub as the 7-inch reel has. This is said to equalize tension on the tape and make for smoother, more efficient operation of the recorder.

Easier access to the threading eye is provided by a larger opening in the reel. The reel also has the Irish "No-Spill" feature — a rubber band in notches keeps the tape from spilling loose on the reel.



Irish "No-Spill" reel is easily threaded.

The 5¾-inch reel is available with 600-foot lengths in Irish No. 195 Brown Band, No. 211 Green Band, No. 300 Shamrock, and No. 220 Sound Plate. It carries 900-foot lengths of Irish No. 600 Long Play (Mylar base) and No. 600-AB Long Play (acetate base). In the Irish Double Play tape, the reel carries 1,200 ft.

VIKING "CUSTOMER SERVICE"

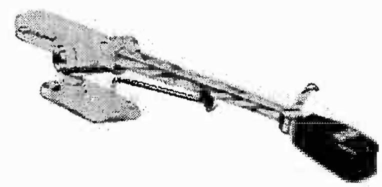
Viking of Minneapolis, through its Customer Service Department, now provides a new service to users of Viking tape equipment. Customers may request specific information as to equipment recommended, and the interconnections possible to permit use of Viking tape equipment to best advantage with their present or intended music systems.

Those using this service should include with their inquiries complete information about the model and year of their preamplifier, mixer, tuner, power amplifiers, etc.; and state the types of tape operation desired: i.e., monaural recording and playback, stereo play, stereo play and monaural erase-record, etc. It is requested that rough diagrams showing electrical connectors (inputs and outputs) to preamplifiers, tuners, and power amplifiers should be included.

Address inquiries to Viking of Minneapolis, c/o Customer Service Department, 9600 Aldrich Avenue South, Minneapolis 20, Minn.

TRANSCRIPTION TONE ARM

Garrard has recently introduced the *Model TPA/10*, said to be the first transcription tone arm adjustable for length, as well as for tracking angle, stylus pressure, and mounting height.



Garrard arm permits instant adjustment.

A special patented protractor lays out the recommended angle on which to align the cartridge for any arm length being used. The protractor enables the user to set the tracking angle at any desired radius.

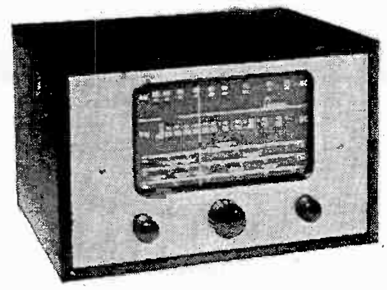
The Model TPA/10 is supplied as a complete packaged unit, ready for installation. All necessary mounting hardware is included, together with special templates which show the exact mounting location. The price of the TPA/10 is $24.50.

FM-AM-SW TUNER

Available to American audiophiles for the first time, the British Chapman *Model S-5E Globemaster* FM-AM-SW tuner is said to provide high sensitivity and selectivity with drift-free tuning on FM, broadcast, and short-wave bands. The Chapman S-5E features the new Mullard EM-81 tuning eye, a variable selectivity control, and a preset volume control at the rear of the chassis. Tuning ranges cover 1.1 to 3.3 Mc (90 to 250 meters), 3 to 8.5 Mc (35 to 100 meters), 8.1 to 23 Mc (12.5 to 37 meters), 88 to 108 Mc, and 545 to 1,600 Kc. The tuner is available in both chassis and cabinet versions.

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. Make use of this special service; save postage and the trouble of making individual inquiries to a number of different addresses.

Chapman unit tunes FM, AM, and SW.



MIDDLE-RANGE SPEAKER

The Racon Electric Company's *Model 6-M* middle-range speaker is only 6 in. in diameter. It has a heavy cast-aluminum frame and body which encloses and seals the speaker's magnetic structure. Its cone is fabricated from linen impregnated with bakelite to act as a very stiff piston. Response of this middle-range unit is said to be from 300 to 6,000 cps, ±3 db.

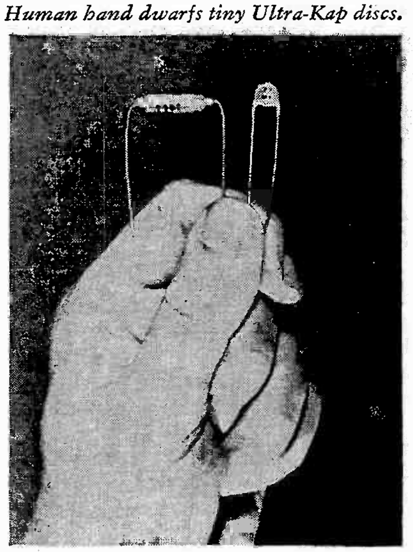
The Model 6-M is reported to be able to handle 25 watts of program material. Its impedance is 8 ohms, or 15 ohms on request. The audiophile net price is $35.00.

MINIATURE DISC CAPACITOR

A new micro-miniature disc capacitor designed to meet the small-size, high-capacitance demands of transistor circuitry has been introduced by Centralab Division of Globe-Union Inc. This capacitor, trademarked *Ultra-Kap*, is intended to meet stringent demands of space, performance, and economy.

Additional information about the Ultra-Kap will be furnished on request.

Human hand dwarfs tiny Ultra-Kap discs.



by RICHARD D. KELLER

book reviews

Acoustical Engineering

Harry F. Olson; pub. by D. Van Nostrand Co., Inc., Princeton, N.J.; 718 pages; $13.50.

This large volume is a tour de force, a standard reference for acoustical engineers. The material, based on the author's earlier work, *Elements of Acoustical Engineering*, has been considerably expanded and updated. An idea of the scope of the book is evident in the author's definition of the science of acoustics as "the generation, transmission, reception, absorption, conversion, detection, reproduction, and control of sound." He has delved deeply into each of these phases of the over-all picture.

After providing a background coverage of sound waves, acoustical radiating systems, mechanical vibrating systems, dynamic analogies, and acoustical elements, Mr. Olson tackles the practical applications of these basic principles in direct radiator and horn-loaded loudspeakers and enclosures, microphones, phonograph pickups and miscellaneous transducers, measurement techniques, architectural acoustics, speech, music, and hearing, sound reproduction systems, underwater sound, and ultrasonics. Pertinent mathematical formulas are given, but their lengthy derivations are generally omitted.

This book is primarily a reference handbook for the sound-equipment and architectural design engineer rather than a college-level textbook.

Acoustics

Joseph L. Hunter; pub. by Prentice-Hall, Inc., Englewood Cliffs, N.J.; 405 pages; $8.50.

Primarily a college-level text, this volume is designed to introduce the field of acoustics to the student and provide him with the background necessary for an understanding of acoustical writing such as that found in the *Journal of the Acoustical Society of America.* Rather than cover extensively the many subdivisions of this science, the author has attempted to develop an insight into the basic principles, and a familiarity with the mathematical tools and techniques of acoustics. Previous knowledge of general physics, electric-current theory, and calculus through partial differentiation is almost mandatory.

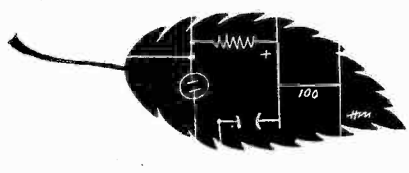
The first half of the book deals with the basic mathematical concepts of oscillation, vibrating strings, plane and spherical waves, and radiation. The second half provides a general mathematical treatment of loudspeakers, microphones, recording and reproduction, speech, hearing, noise and intelligibility, architectural acoustics, measurements, ultrasonics, and underwater sound, with an appendix of Bessel functions. Problems are given at the end of each chapter, with solutions presumedly available to course instructors.

Acoustics for the Architect

Harold Burris-Meyer and Lewis Goodfriend; pub. by Reinhold Publishing Corp., New York; 126 pages; $10.00.

Here is a well-illustrated large-page acoustical guide for the architect done in the clear layout style of *The Architectural Forum.* It's a do-it-yourself handbook which avoids like a plague the formalized mathematics of the classic approach. Instead, it provides descriptions, graphs, and tables of various materials and their acoustic properties, and general design principles for rooms and interiors. It should be a real boon to any architect, whether he's designing residences, theaters, schools, factories, or arenas.

Sound and music, and the clear transmission of intelligence play such a vital part in our lives that acoustic factors in our dwellings and working areas must not be overlooked. Published in July, *Acoustics for the Architect* is up-to-date with the latest materials and techniques. Unlike many other books on acoustics, this volume avoids not only mathematics, but practically any mention of sound equipment such as loudspeakers and enclosures and microphones, dealing strictly with the architectural materials and room shapes best used to control sounds once they are *in* the air.



Antennas

Ed. by Alexander Schure; pub. by John F. Rider Publisher, Inc., New York; 79 pages; $1.50, paper-bound.

Another in the Electronic Technology Series published by Rider, this booklet continues the high standards of its predecessors.

Written for students, technicians, experimenters, hams, trainees in electronic technology, and others interested in communications and related electronics, the book explains fundamental antenna principles clearly and concisely. It illustrates with drawings, graphs, and analogies, many important points concerning polarization, antenna tuning, feeding, matching, gain, and directivity, which are often left nebulous and hazy by a more formalized mathematical approach. For this reason, I strongly recommend this book (and the others in the series such as *Resonant Circuits*, *RC/RL Time Constants*, and *Inverse Feedback*) to all serious students of electrical and electronics engineering, as well as hams and others, whose enjoyment and success in this field are proportional to their knowledge of these basic principles and their ability to apply such knowledge to their professions and hobbies.

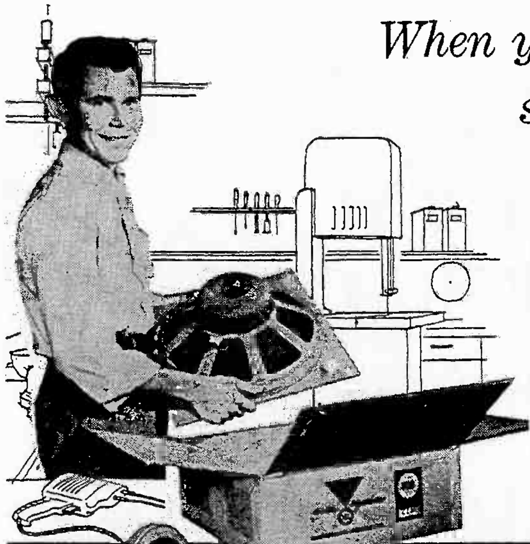
Electronics for Everyone

Monroe Upton; pub. by New American Library, New York; 302 pages; $.50, paper-bound.

This is fascinating reading for the layman (and entertaining even for the engineer, with its easy coverage of the historical aspects and background of his profession).

First, it covers the fundamental history, events, and names which provided the basis for our present-day electronics terminology. The second half of the book proceeds to show how these ideas have been developed into present-day radio, FM, television, radar, computers, etc. More than 100 drawings and a lucid easy-going style make this almost as much fun to read as a good novel. Lives up well to its title.

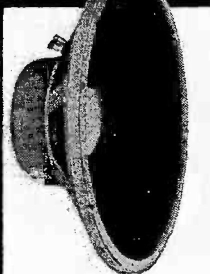
When you build your High Fidelity sound system, use **THE VERY BEST LOUDSPEAKERS YOU CAN GET**



You are planning to build, or improve, your high fidelity sound system. Unstintingly, you will pour out your enthusiasm, time, and energy to get the finest music reproduction you can bring into your home. Get a loudspeaker that will do full credit to your handiwork . . . Install a JBL Signature Extended Range Loudspeaker, or two-way speaker system, in your enclosure.

JBL Signature Loudspeakers are made with the same careful craftsmanship, the same precision forming and fitting that you yourself would use if you set out to make the finest loudspeaker the world had ever heard. JBL Signature precision speakers are the most efficient loudspeakers made.

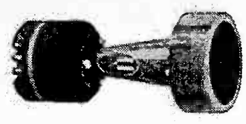
With a JBL Signature Loudspeaker in your high fidelity system, you can exhibit your components with pride, confident that those you have made yourself are being demonstrated in the most effective way possible.



MODEL D130—15" extended range loudspeaker The only 15" extended range speaker made with a 4" voice coil is the world-famous JBL Signature D130. The large voice coil stiffens the cone for crisp, clean bass; smooth, extended highs. Your basic speaker, the D130 works alone at first, later becomes a low frequency driver when you add a JBL Signature high frequency unit and dividing network to achieve the ultimate excellence of a JBL Signature two-way system.

MODEL D123—12" extended range loudspeaker With outstanding "presence" and clean response throughout the entire audio spectrum, the D123 features an unusual shallow construction. Only 3⅝" deep, it is designed to mount flush with the wall, between studding, in any standard wall or partition. Frequently, the D123 is used in multiples in "infinite baffle" wall installations. In this case the JBL Signature 075 is a logical high frequency unit to add when you advance to a two-way system.

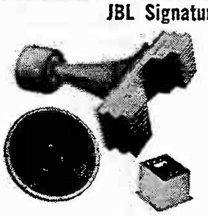
MODEL D208—8" extended range loudspeaker **A precision transducer** in every sense of the word, the famed JBL Signature 8" D208 is made with the same care and precision as the larger units in the James B. Lansing Sound, Inc. line. If space and cost are major considerations, the D208, properly enclosed, provides the most lastingly satisfactory sound you can get. It is widely used in top quality systems where extension speakers are desired for areas other than the main listening room.



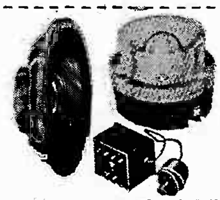
MODEL 175DLH high frequency assembly The acoustical lens is only available on JBL Signature high frequency units. The 14 element lens on the 175DLH disperses sound within the listening area over a 90° solid angle, smoothly, with equal intensity regardless of frequency. The acoustical lens is the greatest contribution to lifelike high frequency reproduction in 20 years, and it was developed for use with high fidelity equipment by James B. Lansing Sound, Inc. In addition to the lens, the 175DLH consists of a high precision driver with complex phasing plug and a machined aluminum exponential horn. Designed for crossover at 1200 cycles with the JBL Signature N1200 Network.

MODEL 075 high frequency unit Another exclusive for James B. Lansing Sound, Inc. is the ring radiator in the JBL Signature 075 high frequency unit. A ring, rather than a diaphragm, radiates into the annular throat of an exponential horn. The result is high frequency reproduction of unmatched smoothness and clarity, absolutely free of resonances and strident peaks. The horn is beautifully machined from aluminum, the entire unit a gratifying, solid piece of fine craftsmanship. Designed for crossover at 2500 cycles with the JBL Signature N2500 Network.

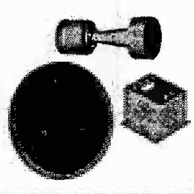
JBL Signature two-way systems are available as kits



086 KIT This two-way system is made up of units which have been acclaimed by impartial authorities as the finest available anywhere today. Included in the kit are the 150-4C Low Frequency Driver, N500H Network, 375 High Frequency Driver, 537-509 Horn-Lens Assembly. These are the same units—including the serpentine acoustical lens—which are used in The Hartsfield . . . units designed originally for installation in the most modern theaters in the world.



002 KIT Including some of the newest speakers made, the JBL Signature 002 Kit includes a D123 for low frequency reproduction, N2500 Network, 075 High Frequency Unit. The 002 Kit is moderately priced, yet gives the user all the advantages of a two-way system made with independent drivers.



001 KIT Probably the most popular high quality two-way system on the market, the JBL Signature 001 system consists of a 130A Low Frequency Driver, N1200 Network, 175DLH High Frequency Assembly. The D130 may be substituted for the 130A without disturbing the balance or coverage of the system.

There are many more kits and loudspeakers in the JBL Signature line. Whatever your needs, you will find exactly the right unit or system in the complete JBL Signature catalog. Send for your free copy. A limited number of technical bulletins are also available. Please ask only for those in which you are vitally interested.

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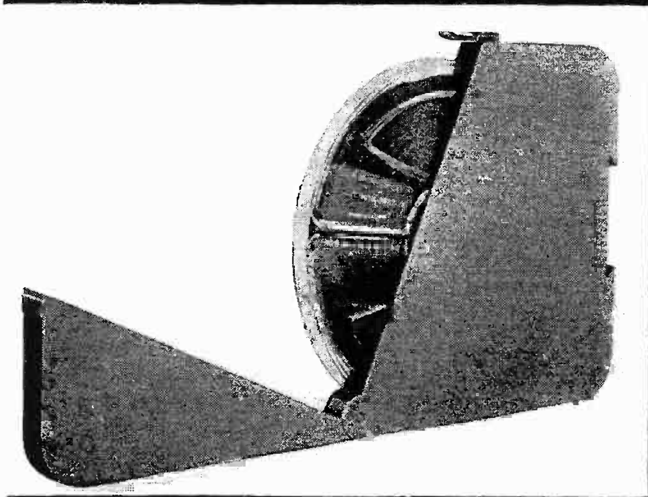


THE QUALITY RECORDING TAPE IN THE NEW PERMANENT PLASTIC CONTAINER

SONORAMIC

Here is an extraordinary new product designed to protect, preserve and facilitate storage of your Sonoramic Wide Latitude Recording Tape. It's the exclusive NEW Sonoramic *permanent plastic tape container*. Sonoramic's fine quality magnetic recording tape PLUS the new container makes this your best buy in recording tape.

Here's the story on the container:

- *Protects tape against dust and dirt.*
- *Made of high-impact, shatter-proof, polystyrene plastic in handsome decorator color.*
- *Opens at flick of finger pushing tape forward for easy access.*
- *Stacks neatly on shelf, bookcase, or table.*
- *Dovetail strip (available from company) lets you hang a row of tape containers on wall.*
- *Unique Sonoramic indexing system on pressure sensitive labels included free in every package. Permits you to keep tabs on all recordings.*
- *Tape time ruler on carton permits accurate measurement of elapsed and remaining time.*

Inside the container...

...is Sonoramic Wide Latitude Recording Tape, a superb new miracle of recording tape engineering. From the selection of raw materials, to coating, slitting and packaging—this tape reflects the care and precision it takes to make a quality product. Here's the story on the tape:

- *Distortion-free recordings guaranteed by exclusive time-temperature dispersing techniques.*
- *Broad-Plateau Bias assures maximum performance regardless of make of recorder, line voltage fluctuations, tube age, head condition.*
- *High resistance to abrasion, print-through and cupping.*
- *Life-time lubrication eliminates squeal, layer-to-layer adhesion, and deposits on heads.*

There are three tapes designed for all uses—all on 7" reels. These include: *Standard Play*, 1½ mil acetate, 1200 feet, meets rigid requirements for both professional and home use. *Long Play*, 1 mil mylar,* 1800 feet, a premium quality tape designed for maximum strength and immunity against heat, humidity and other weather conditions. *Extra Long Play*, ½ mil mylar,* 2400 feet, a high quality tape useful for extra recording time, and where tape tension is not excessive.

When you buy your next reel of tape remember these facts: not only do you get the excellent quality of Sonoramic Wide Latitude Recording Tape—but every reel comes in its own handsome permanent plastic container.

NOTE: To the first 50 people who write in requesting it—we'll be happy to send out a free Sonoramic tape container. Please remember: we can only do this with the first 50 requests: *Write to Dept.* D-101. *Ferrodynamics Corporation, Lodi, New Jersey.*

Store on table...

...or on wall...

...or in bookcase.

**A DuPont trade mark.*

SONORAMIC IS A PRODUCT OF THE *Ferrodynamics* CORPORATION • LODI, NEW JERSEY

Readers' Forum

Gentlemen:
Thanks to Mr. Marshall for his latest (August 1957) "Grounded Ear." His loudness control is a fine achievement. For about $12 worth of pieces and with some 20 hours of work, I made this instrument, which proved very efficacious right at the first trying. I am using it as a separate unit following my preamplifier. The improvement it affords is absolutely unquestionable, and, as a bonus, it eliminates the last trace of hum in the whole system.

Such contributions as Mr. Marshall's make a subscription to your magazine a fine bargain indeed.

Roger Maltais
Sherbrooke, Que.

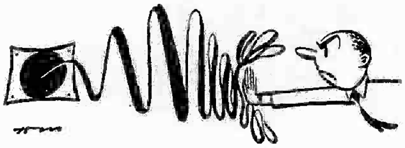
Gentlemen:
Regarding the Troy, New York, reader's complaint (AUDIOCRAFT, August 1957, p. 13) that your present covers look "cheap, gaudy, and commercialized . . ." I would like to say that it is a matter of personal opinion and my personal opinion follows: I see nothing cheap, gaudy, or undignified about them.

As for their being commercialized, I feel that your covers, at a glance, give the prospective reader the main facts. The artistically arranged title plate tells at once that this is a magazine for the "Hi-Fi Hobbyist." The brief headings and picture should be enough, if the prospective reader is interested in hi fi, to intrigue him into buying a copy and investigating its contents.

Effective packaging is one of the essentials of modern selling, and all good advertising should 1) attract favorable attention to the product or service offered; 2) create a desire on the part of the prospective buyer for the product or service advertised; 3) produce action on the part of the prospective buyer to purchase and use the product or service advertised. I think your present covers fulfill all of these requirements.

But let's not quibble over the cover. After all, "proof of the pudding is in the eating"; it is the contents of the magazine that count.

De Witt H. Thompson
Chicago, Ill.



EDITORIAL

THE matter of amplifier power requirements has received a great deal of attention recently in the technical and semitechnical press, as well as in consumer publications devoted in whole or in part to high fidelity. It is a stimulating subject — an important one. But so much of the published information has been purposely or inadvertently misleading that we consider it our duty to comment editorially on the subject.

First, what is amplifier power needed for anyway? To produce, with a loudspeaker system, sufficient acoustic power in a listening room to satisfy the listeners therein. Simple enough. What determines how much acoustic power is needed in any given circumstance? Eliminating the question of "personal preference," we are left with the following three factors:

1) *Room size.* The larger the listening room, obviously, the more acoustic power is needed to create a given intensity level, other factors remaining equal. The proportion is practically direct.

2) *Surface absorption.* The rate at which room furnishings and surfaces absorb sound energy has a direct bearing on the rate at which acoustic power must be generated in order to maintain a given intensity level. A heavily draped, carpeted, and padded room will accordingly demand much more acoustic power than a room of the same size having predominantly "hard" surfaces; the power needed is directly proportional to the absorption.

3) *Type of program material.* Music for large orchestral and choral works naturally produces peaks of sound intensity, at a typical listening location, that are higher than the peaks produced by a solo instrument or a small ensemble.

Now, the measured peak intensity levels of full orchestras at central auditorium seats average about 95 db above the threshold of hearing. This is much lower than the level *in the orchestra.* Since many listeners prefer closer seats, we should adjust this upward by at least 5 db for them; therefore, a realistic basis for an intensity level that will satisfy any listener may be taken as 100 db — not even close to the threshold of feeling. According to Olson,* such a level will be produced in a room 8½ by 12 by 15 ft., of average absorption characteristics, by acoustic power of about ¼ watt. That would be considered a small room. If a loudspeaker system 5% efficient in converting electrical power to acoustical power were used, an amplifier would have to deliver 5 watts to the speaker in order to obtain the required 100-db sound level.

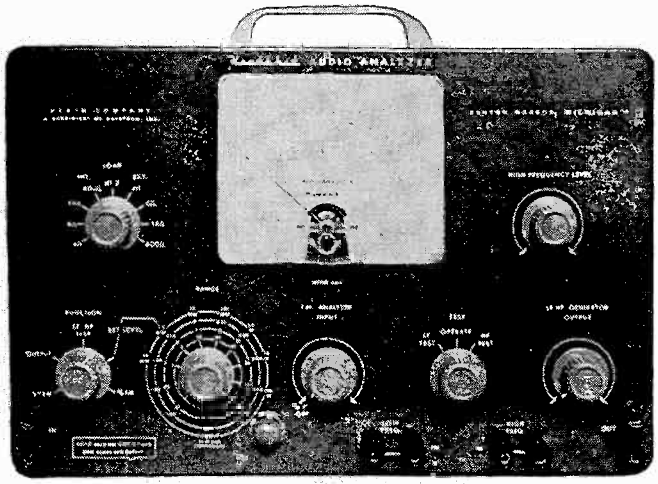
If the room is smaller or larger, this figure has to be adjusted in proportion to the relative room volume. A room 10 by 15 by 30 ft. (not uncommonly large) contains three times the volume of air, and accordingly a 15-watt amplifier is needed to produce the same level. If the room is on the dead side acoustically, or has some normally open doorways (which represent 100% absorption over their area), again the amplifier power requirement is increased, possibly by a factor of two or more. If the room is smaller or less absorptive than normal, compensatory decreases in the estimated power requirement should be made.

These adjustments, large as they may be, are small compared to that for loudspeaker efficiency. This factor may vary from less than 1% to nearly 50%. Very large loudspeakers, fully horn-loaded over the entire frequency range, average from 15% upward; some combination direct-radiator and rear-loading horn systems may be around 10% efficient; bass-reflex and similar systems vary between 3% and 7%; large infinite-baffle units are usually 2% or slightly more; small, totally enclosed high-quality units average less than 2%. There are exceptions in each group, but they are rare. The important thing is that, if you use a 40% efficient speaker in an average small room, the amplifier need deliver only ⅝ watt for a 100-db sound level, rather than 5 watts. But if you use a 1% efficient speaker, the electrical power requirement is 25 watts. And if your room is three times as large, and is twice as absorptive as normal, then you need 150 watts into a 1% efficient speaker!

Note, too, that all these figures are based on undistorted power *into the speaker.* Loudspeaker systems, unfortunately, are not the perfectly uniform impedance loads upon which amplifier power ratings are based. Impedances of even the best systems vary by factors of four or more in maxima to minima.

It is apparent that individual circumstances (primarily the speaker system's efficiency) affect the required amplifier power to such a great extent that no one can say flatly that 10 watts, or 30 watts, or even 100 watts, is enough. It is perfectly possible that 1 watt may be plenty, or 200 watts inadequate. We hope that the information given here will serve as a rough guide to determine how much power *you* need. — R.A.

*H. F. Olson, *Acoustical Engineering* (Princeton, N.J., 1957), p. 524.



Heathkit Audio Analyzer

OF all of the instruments devoted to high-fidelity design, testing, and repair, none can be more useful than a distortion analyzer. In the design of new circuits, tube tables and characteristic curves can indicate theoretical values for circuit components and certainly are indispensable, but if the circuits in question can later be tested for distortion, and the ultimate values zeroed in by using a distortion meter, the builder is likely to come closer to the practical ideal.

Such a meter is the Heathkit Audio Analyzer, and it is not called a "distortion meter" because it is much more than just that. It is indeed an analyzer of audio devices, because it is at once a generator of audio frequencies used in intermodulation testing (60 and 6,000 cps); a frequency-selective filter coupled to a high-sensitivity meter which measures intermodulation directly in percentages; a high-sensitivity AC VTVM which will measure as low as 10 mv full scale; and, finally, it contains built-in load resistors of 4, 8, 16, and 600 ohms. These are switch-selected and rated at 25 watts continuous, and 50 watts intermittent. Note the compact mounting position in Fig. 1.

Such a listing of capabilities suggests an infinite number of variations. The generator frequencies can be altered in level to provide virtually any ratio desired, by the use of two potentiometers which are combined in a unique fashion so that, once the high-frequency to low-frequency ratio is set, altering the low-frequency pot not only sets the output level but maintains the ratio already determined. The output voltage is as high as 10 v, and as low as 3 mv (although that is a bit tricky to set). One setting of the input selector switch connects the generator output directly to the meter, so that the frequency ratio and output level may be determined without the fuss of test-lead switching.

One further aspect of the Audio Analyzer which attests to its versatility is the fact that the variable output settings permit the generator to work through an equalizer. With the meter switch set to VTVM, the signal is applied to the device containing an equalizer, and the ratio set at the output. Since the generator can be set to extremely low levels, this feature assures convenience in checking phono and tape preamps. In Fig. 2, components are shown before assembly.

The provision of built-in resistors

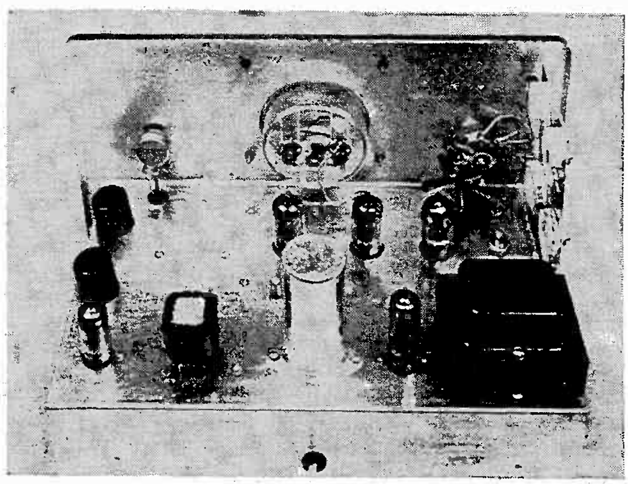


Fig. 1. Top view of the completed Analyzer. Load resistors and load selector switch are in upper right corner of photo.

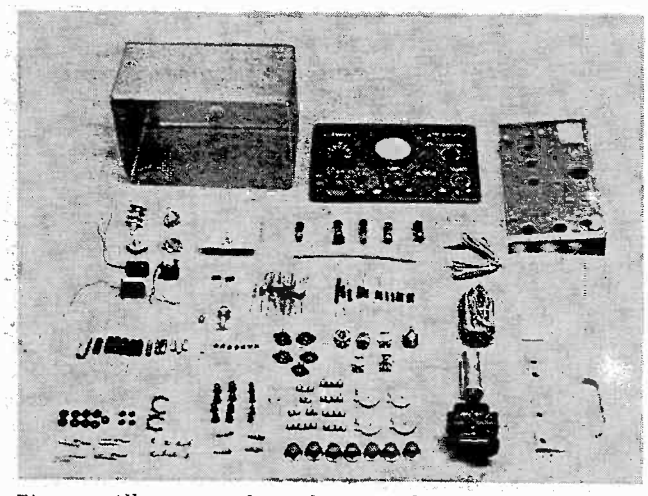


Fig. 2. All parts used in the assembly of the Analyzer kit are shown. The builder must provide only tools and solder.

for amplifier loading is an excellent idea, as anyone who has not similarly equipped himself will appreciate. The switch that selects the internal load also provides for an external load if one is used (such as a loudspeaker, perhaps) and has a high-impedance position for use when testing voltage amplifiers. A close-up of the switch appears in Fig. 3.

The meter itself has full-scale readings of 1 and 3 which, when translated to the proper voltage or IM percentage, can be .01 v up to 300 v full scale, and 1 up to 100% IM full scale. In addition, two sets of decibel markings are on the meter, with a 10-db spacing between the two 0-db indications. The range selector is graduated in 10 db per step, and it is a simple matter to determine the signal-to-noise ratio of an electronic device by noting the output reading at maximum signal, killing the input signal, and simply decading down with the range switch.

All in all, the Heathkit Audio Analyzer lives up to its name superbly. In one package, for $49.95, it provides facilities for checking the accuracy with which an amplifying device will pass audio signals, and at what amplitude. When used in conjunction with an audio oscillator, the pair combined will permit the inspection of 99% of all amplifier performance capabilities. Although the analyzer provides its own 60-cps frequency for IM testing, the characteristic of the low-pass filter is such that any low frequency from a separate oscillator may be used, up to about 600 cps, and mixed with the analyzer's 6,000-cps signal.

Circuit Description

The heart of the instrument is the AC vacuum-tube voltmeter (Fig. 4). In order to provide sufficient gain for a 10-mv full-scale reading, two stages of amplification are required. A 12AT7 tube is connected as a cascode amplifier, and its output fed to one half of a 12AU7. This is in turn coupled to the meter rectifier — two diodes in a half-wave bridge circuit — which provides DC current for the 200-μa meter. Inverse feedback is applied from the bridge back to the cathode of the input stage. A calibration control varies the resistance in the input-stage cathode, thus changing the amount of feedback and determining the gain.

In order to provide proper decading — that is, correct voltage readings at all settings of the range switch — an additional diode and a variable resistor are shunted across the meter, furnishing a means for adjusting the meter linearity.

As an AC VTVM, the black meter scales are read directly as the range switch is rotated. Precision resistors are used in the range switch, thereby providing accurate voltage division in all settings. To measure power levels, precision compensating networks are incorporated which correct the voltage relative to the power dissipated in the load resistors. The output from this network is coupled to the range switch and the VTVM, indicated on a red scale.

In the measurement of intermodulation distortion, mixed high and low frequencies from the generator are injected into an amplifier or other device under test, and the device's output fed to the analyzer input. The high frequencies are amplified, and low frequencies not modulating the higher frequency are rejected. The modulated

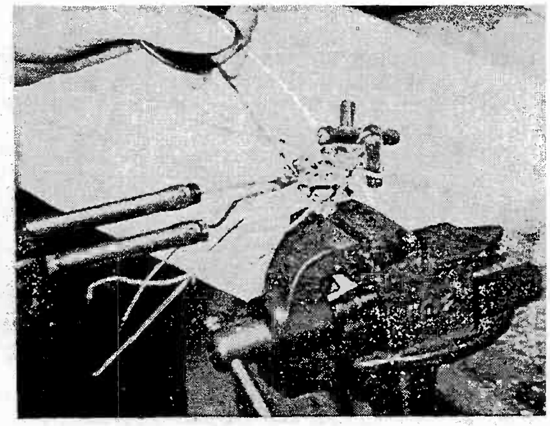
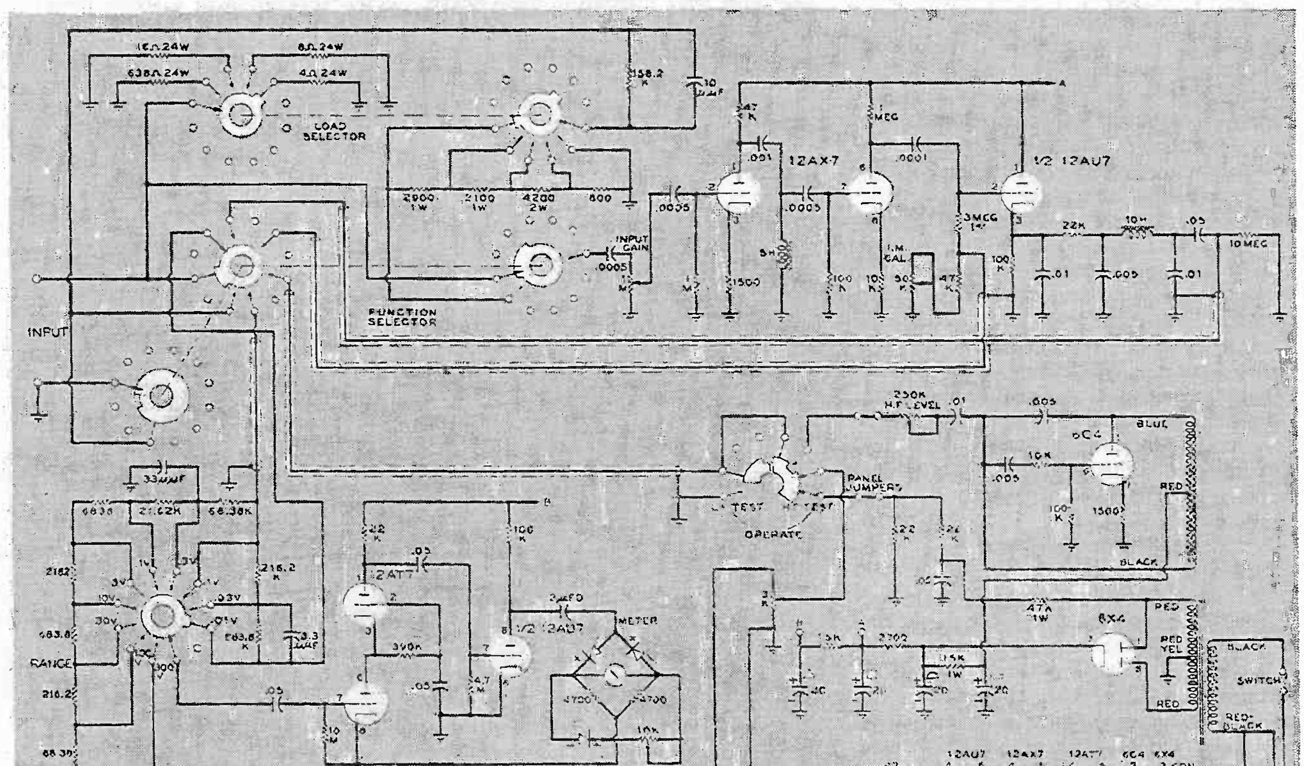


Fig. 3. Holding the switches in a vise greatly simplifies soldering operation.

Fig. 4. Schematic of the Heath Audio Analyzer kit. In addition to this easy-to-follow schematic diagram, the kit contains a booklet with simplified drawings showing several stages of assembly, and step-by-step instructions which are complete and easy to follow.



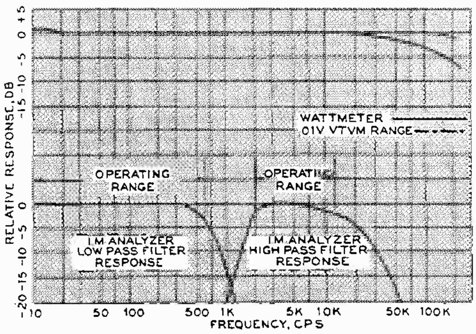


Fig. 5. Frequency response of the VTVM, and IM-filter operating ranges.

higher frequency is set to a predetermined level and demodulated. The remaining low frequency is passed through a low-pass filter to remove any residual high-frequency component, then sent to the meter where the remaining signal is indicated directly in IM percentage.

In the analyzer the mixed 60- and 6,000-cps signal is fed from the input switch to a level control through a small capacitor — which attenuates the low frequency — and thence to the first half of a 12AX7 IM amplifier. The amplified high frequency goes through a high-pass filter and is further amplified by the second half of the 12AX7, then sent to the detector. A potentiometer which with two resistors makes up the grid load for the detector is used to calibrate the analyzer. The calibrated portion of the output signal is fed to the VTVM for setting the reference level for IM measurement.

The detector is one half of a 12AU7 cathode follower. When the signal appears at the grid, any modulation which is present is evident at the cathode. A small amount of the high-frequency test signal remains, but this is filtered out in the low-pass filter which follows the cathode, and the remaining low frequency sent to the VTVM for reading. Fig. 5 illustrates the response of the VTVM and filter sections.

The power supply is conventional, with a 6X4 rectifier. Voltage for the low-frequency signal is supplied by the power transformer though a filter which removes all harmonic content and provides a pure 60-cps wave form. The 6,000-cps oscillator is a 6C4, which has a separate filtering system to insure complete isolation.

Construction Notes

The Heathkit Audio Analyzer is not a simple kit, and can't be built in a rush. It took us slightly longer to assemble the analyzer than we spent on the Heath oscilloscope — all told, about 10 hours. On the other hand, the oscilloscope utilizes a printed-circuit board, which speeds up things considerably. The analyzer has no printed-circuit wiring at all — everything is point-to-point, and some of the wiring around the tube sockets, as things begin to get a bit cramped near the end of the process, requires care. But it's not at all troublesome, and in fact is simpler than a great many chassis we have worked on (Fig. 6).

The only tools we needed for our assembly were a soldering gun, a screw driver, and an adjustable wrench. The kit, of course, contains all parts except solder. At the end of the project, we had exactly 3 in. of wire, four lock washers, and one 6-32 screw left over!

Following the advice of the instruction manual, we proceeded step by step, made no changes in the wiring procedure, and checked off each step as it was completed. The reward for strict obedience to instructions was an instrument which worked perfectly as soon as completed.

Although the range switch appears to be the kit's most complicated component to assemble, in actuality it was extremely simple. To hold it steady while applying solder and heat simultaneously, a small bench vice was tightened around the shaft (see Fig. 3). This same procedure worked admirably with all the switches that required wiring before being fastened to the chassis. To avoid overheating the precision resistors, an alligator clip was clamped on each lead near the resistor body before solder was applied. After soldering, we waited a few seconds before removing the clip, to let the joint cool.

After the wiring had been completed, and tubes were inserted in their sockets, an ohmmeter was used to check for shorts in the B+ line. The meter pointer soon zoomed up over 20,000 ohms, indicating that everything was all right in the power supply.

The line cord was inserted and the switch thrown. All tubes glowed normally, and smoke and acrid smells were, happily, absent.

Calibration

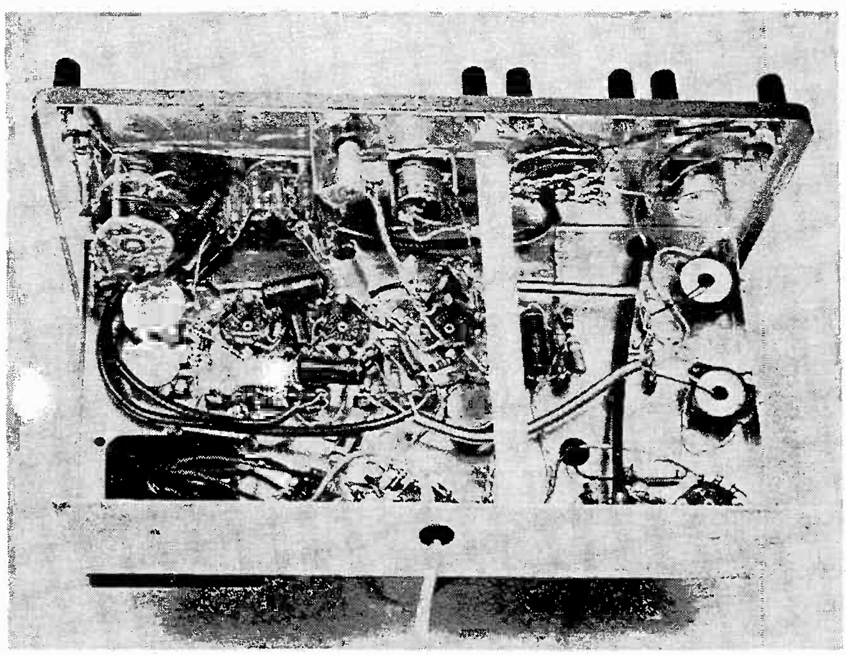
The VTVM is calibrated by adjusting the CALIBRATE CONTROL until the meter pointer indicates the exact value of a known voltage applied to the input terminals. In our case, we relied on the line voltage, which our trusty multimeter had shown to read exactly 117 v, for calibration. This operation was executed carefully, by attaching the alligator clips on the analyzer leads to a short piece of line cord clamped in the vise. Only after the clips were securely in place was the cord plugged into the AC outlet, and it was removed as soon as the calibration had been completed. Careless handling of bare line cords can be dangerous!

The meter linearity adjustment was also very simple. Using the low-frequency output of the internal generator, the RANGE switch was set to the 10-v position, and the GENERATOR OUTPUT control adjusted until the meter pointer indicated precisely 10 v. Then the RANGE switch was rotated to higher voltage ranges, and the LINEARITY control adjusted until the pointer indicated the same numerical value on all ranges. Only a slight degree of correction was necessary.

Calibration of the IM analyzer section was a bit more difficult, and involved the application of a signal containing a known quantity of intermodulation and adjustment of the CALIBRATE control until the meter indicated this value. The instruction manual describes a method of calibration using nothing more than

Continued on page 44

Fig. 6. The wiring of the completed Audio Analyzer is shown in this photograph of the inverted chassis. Wiring operations are explained in detail in the instructions.



TRANSISTORS *in Audio Circuits*

by PAUL PENFIELD, JR.

VIIb: Transistor Input Stages

Phonograph Transducers

Probably of greater interest than microphones are the various phono cartridges. Piezoelectric and magnetic types are the most important.

Before we get into circuit details, a word about the output from cartridges is in order. The subject of equlization is tied up with the load the cartridge looks into, and why this is of importance in transistor circuits but not in vacuum-tube circuits will become apparent.

Roughly speaking, we can say that magnetic cartridges (including variable reluctance types) are magnetic in nature, and crystal and ceramic cartridges are capacitive. That is, the internal impedance shown in Fig. 1 (AUDIOCRAFT, October 1957, p. 18) for each case includes something besides pure resistance.

Consequently, the open-circuit output voltage from a cartridge and its short-circuit output current will by no means be the same, and we must take account of this.

Modern records are recorded with a treble pre-emphasis, and a bass attenuation, relative to a *constant-velocity* recording. That is, the stylus velocity as a function of frequency appears as Fig. 9. Tipping this graph upside-down of course yields the equalization required in the preamp (RIAA) provided our cartridge responds to stylus velocity uniformly over the frequency range. Magnetic cartridges operating open-circuited do this.

However, because of the relationship between stylus velocity and *stylus amplitude* (or displacement), the latter curve, as it exists for modern records, appears as in Fig. 10. Roughly speaking, modern records are cut constant-amplitude, except for the small treble dip and bass boost. Consequently, any cartridge that responds uniformly to stylus amplitude will not need so much equalization. Open-circuited crystal cartridges respond in this way, and, as is well known, do not require major equalization.

The situation is changed when we are interested in short-circuit output current. When we short a crystal cartridge, the capacitive nature of the device attenuates the low-frequency current at just the rate of 6 db per octave — exactly the difference between Figs. 9 and 10. Consequently, we can express this fact by saying that on short circuit, a crystal cartridge responds to stylus *velocity*. Hence if we short it (a grounded-emitter input resistance is generally a good short even at the high frequencies) and amplify the resulting current, we shall have to provide for the customary equalization — the standard RIAA de-emphasis curve.

On the other hand, if we operate a magnetic cartridge short-circuited it will, because of its magnetic nature, respond to stylus *amplitude*. Thus there will be no need to apply the normal RIAA equalization and, in fact, in noncritical cases we can sometimes get by with no equalization whatever. This of course is possible only when we really short the cartridge; in practical cases it is not always easy.

These points are not usually brought up, because vacuum tubes seldom offer impedances which can be considered as short circuits for phono cartridges. However, transistors can and do.

Crystal and ceramic cartridges: as mentioned before, crystal cartridges, when shorted, require normal equalization, or close to it. If a circuit like that in Fig. 11 is used, the mere 1,000 ohms or so offered by the amplifier is small as far as the cartridge is concerned, and is effectively a short.

The equalization required is some-

Fig. 9. Treble pre-emphasis and bass attenuation curve (RIAA).

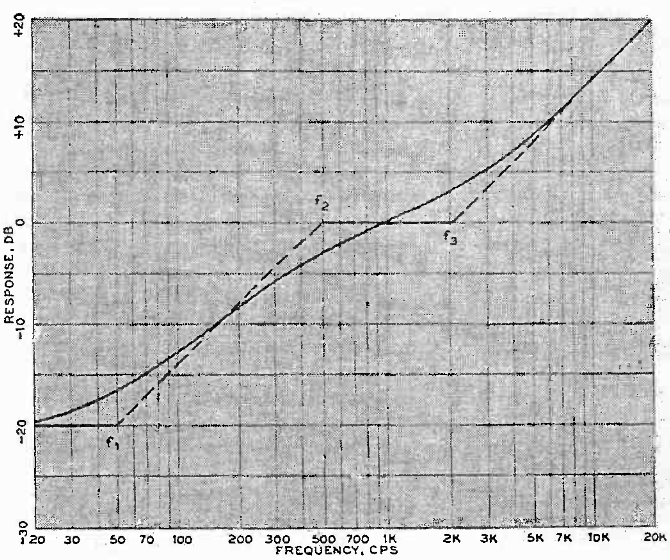
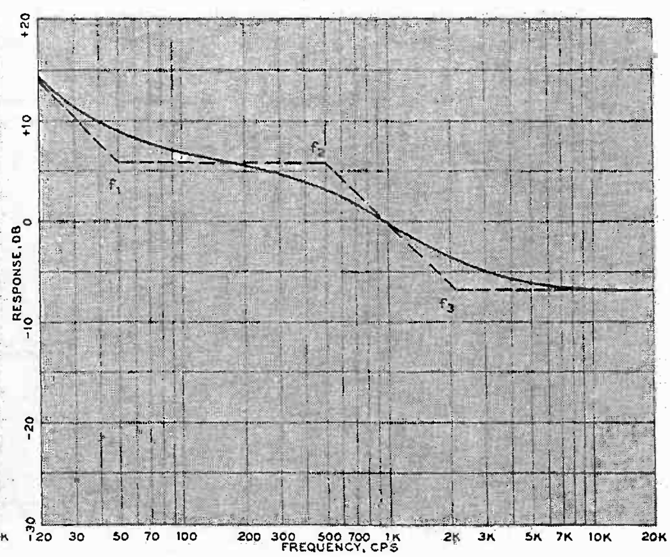


Fig. 10. Response of constant-amplitude cartridge to RIAA curve.



what harder to achieve with transistors than with vacuum tubes, for reasons that will be covered in Part VIII of this series.

Crystal cartridges can also be operated essentially open-circuited by adding an

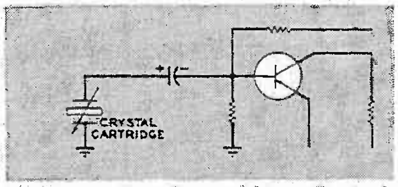


Fig. 11. Crystal cartridge effectively short circuited with 1000-ohm resistor.

input resistor, as was done with crystal microphones, or an input transformer can be used.

Furthermore, it is possible to equalize the treble automatically by using the correct value of resistor. The correct value will make the cartridge essentially short-circuited at frequencies below the frequency f_3, and essentially open-circuited above this. The composite output curve would resemble Fig. 12 if exactly the right value were chosen, leaving only the bass boost to worry about.

Magnetic cartridges: open-circuited magnetic cartridges require equalization — and it is not hard with transistor circuits to terminate the cartridge in the recommended resistance and still achieve moderately high signal-to-noise ratios. This is commonly done.

Table I

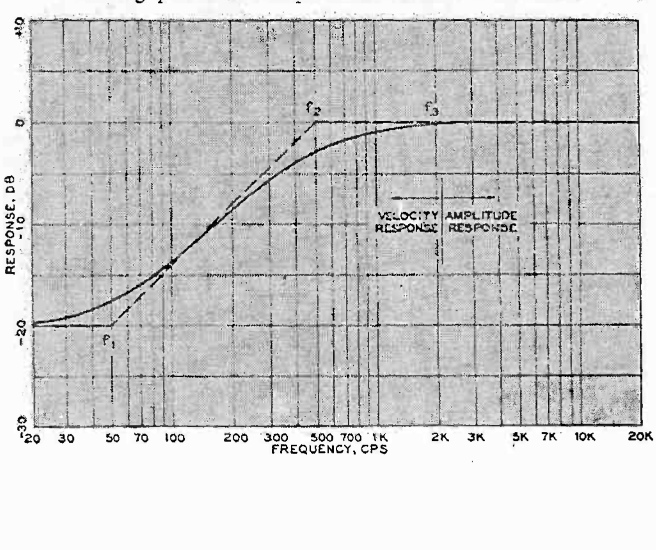
Cartridge	Output Voltage E, mv	Inductance L, mh	Internal Resistance R, ohms	Recommended Terminating Resistance R_t, ohms	Automatic Rolloff Resistance R_p, ohms
Audak Hi-Q7	20	720	560	100,000	9,100
Connoisseur	25*	20	400	10,000 to 40,000	[1]
ELAC MST-1 Miratwin:					
microgroove	55*	320	1,400	47,000	2,860
standard	45*	220	1,000	47,000	1,940
ELAC MST-2 Microgroove	55*	320	1,400	50,000	2,860
ELAC MST-2 Standard	45*	220	1,000	50,000	1,940
ESL Soloist	2.2*	.007	1.5	not critical	[1]
ESL Concert	1.7*	.007	2.0	not critical	[1]
ESL Professional	2.2*	.007	2.0	not critical	[1]
Fairchild 225A	7.1*	2.6	195	above 5,000; optimum 47,000	[1]
Fairchild 215 [2]	2.0	[3]	70	above 5,000	—
Fairchild 220 [2]	2.5	[3]	170	above 5,000	—
Fenton B&O-72	35*	[3]	2.8	10,000	—
Fenton B&O-350	70*	80	5.0	100,000	1,060
Garrard GMC 5	.013*	0.1	3	not critical	—
General Electric RPX-041A, RPX-061A, RPX-063A, RPX-050A, RPX-052A, RPX-053A, RPX-040A	22*	520	420	6,200 for automatic rolloff; 62,000 for open circuit	6,500
Pickering 350, 194	15*	325	3,600	27,000	730
Pickering 220, 240 260 (both sides)	30*	125	800	1,500 for automatic rolloff; 27,000 for open circuit	870
Recoton 500	10	[3]	1,500	47,000	—
Shure M-1	14*	130	180	10,000	1,550

Notes:
Output Voltage *E* is the average expected from a microgroove recording, except as indicated otherwise.
*Referred to 10 cm/sec, the American standard. Average LP output will range from ½ to ⅔ this value, with peaks as high as 4 times this value.
[1] Cannot be done, because of high internal resistance.
[2] Obsolete model. Data included for reference only.
[3] This information not available.

It is best to keep DC out of magnetic cartridges, so a blocking capacitor is normally used, as shown in Fig. 13. Here the sum of the series resistor and the transistor input resistance should be adjusted to the recommended terminating resistance (see Table I). Again, the grounded-emitter circuit offers the most gain, and is always used. Many preamps using essentially this type of input stage have been described in the literature.

Shorting a magnetic cartridge to make it responsive to amplitude is difficult, but sometimes possible. The criterion is that the internal inductance (in henries) must be large compared with the sum of the cartridge internal resistance and the amplifier input resistance, divided by six times the lowest frequency of interest. For hi-fi applications requiring response down to 20 cps, this condition may be pretty hard to meet with present cartridges. But in the future cartridges will be designed specifically for use with such circuits. Then the advantages of only minor equalization and very high signal-to-noise ratios will be realized.

Fig. 12. Output of a crystal cartridge with proper loading produces composite curve illustrated here.



Again, it is possible to equalize partially by terminating the cartridge in the proper resistance: the time constant of the upper break point (rolloff, f_3), 75 μsec, should be equal to the ratio of the cartridge inductance in henries to the sum of cartridge winding resistance and terminating resistance in ohms. In that case, the output will look just like Fig. 12, and only the low-frequency end will need boosting.

Essential data on many common magnetic cartridges are given in Table I, supplied by courtesy of the various manufacturers. The symbols *E*, *L*, and *R* refer to the AC representation of Fig. 14 for the magnetic cartridge, which is valid throughout the audio frequencies. The value of recommended terminating resistance R_t is for constant-velocity response, while the terminating resistance R_p was calculated by the author to provide automatic high-frequency rolloff, as indicated in Fig. 12.

Other Input Transducers

Although microphones and phono cartridges are the most important transducers for audio work, there are many others.

Magnetic tape head: coupling to a magnetic tape head is of considerable interest. The standard circuit shown in

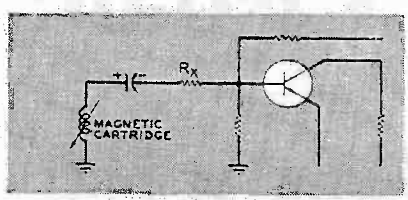


Fig. 13. Circuit for magnetic cartridge requires series capacitor to block DC.

Fig. 15 is as simple as it is useful. As contrasted with vacuum-tube circuits, a transformer is eliminated, with, usually, an improvement in noise. The advantages are just as great here as with dynamic microphones; indeed, the same sort of considerations hold. There is no real reason why all tape-playback amplifiers should not have a transistorized input stage, at the present state of the art.

A grounded-base circuit can also be used, but there is no real advantage except possibly bias stabilization (which is very easy to achieve using two batteries and the grounded-base configuration).

Photocells: the high-impedance phototubes, gas phototubes, or photomul-

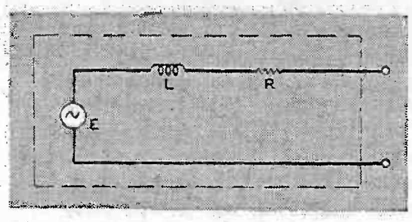


Fig. 14. Equivalent circuit for magnetic cartridge helps find termination value.

tipliers, are better used with vacuum-tube amplifiers, for the same reason as condenser microphones. However, the new and smaller semiconductor photocells can be used very well with transistor amplifiers.

The photoresistive type of cell, using germanium, cadmium sulfide, or lead sulfide, can be used directly with transistor amplifiers. Since these units work in a fashion similar to the carbon microphone, it is possible merely to hook them up with a battery in series with one of

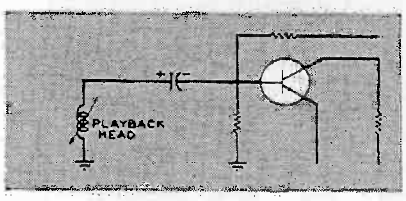


Fig. 15. Coupling a tape-playback head requires standard circuit shown here.

the transistor leads. It is only necessary to be sure that the current expected will not take the transistor out of the linear range. Various photoresistors have different current capacities, sensitivities, etc.

Photodiodes and phototransistors can be connected in the same way.

If the signal of interest is at an audio frequency (as for example in light-beam communications) then capacitors can be used to couple the photocell to the first stage, so each can be biased separately.

Self-generating photocells can often be put in the base lead directly — either to provide the sole base bias, or to change an already existing base bias. No circuits will be shown, since there are as many circuits as there are types of photocells and applications. Any coupling that works is of course permissible, so long as the maximum ratings of the various devices are not exceeded.

Pickup loops: it is possible to transmit audio frequencies over some distance by means of setting up magnetic fields corresponding to the audio signals. This is done merely by feeding the output from an amplifier into a large loop. The loop might be made, for example, to broadcast to a large hall or building. The signal can be picked up by a small loop, and a transistor amplifier used. One typical circuit for coupling the pickup loop to a transistor stage appears in Fig. 16.

This same arrangement can be used with a pickup loop under a telephone set, to listen in on conversations without making a direct connection. Or the loop might be used to pick up other information, such as existence of hum fields.

Line bridging: it is often necessary in audio work to bridge across a low-impedance line. This is done without a transformer by using a transistor. To be a good bridge, the so-called bridging impedance should be at least ten times the line impedance. In Fig. 17, for example, if the line is rated at 600 ohms, then a suitable bridging impedance might be 6,000 ohms or higher. For an unbalanced line there is no trouble — Fig. 17, with the series resistor equal to 6,000 ohms, will surely not load the line.

With a balanced line, it is not possible to take the signal between one side and ground, because this would unbalance the line; also, the line may be used for a phantom circuit, in which case any such tap would pick up the phantom signal as well. With balanced lines it is usually better to use a bridging transformer, unless the whole amplifier and its output can be not only balanced but isolated from ground as well. On the secondary of the bridging transformer, any arrangement such as that shown in Fig. 5 (AUDIOCRAFT, October 1957, p. 19) for the crystal microphone can be used.

Summary

This concludes the discussion on coupling of important audio transducers to transistor circuits. All such circuits really boil down to three different types: 1) feed in through a capacitor, using a series resistor if necessary; 2) feed in through a transformer; and 3) feed in directly, when the transducer can be biased with the same current as flows in the transistor.

For response down to 20 cps, the input capacitor should be in the range of 20 to 50 μfd, unless there is a resistor in series with it, in which case it can be lower. Almost invariably best gain and lowest noise result from the grounded-emitter circuit, although sometimes the grounded-base configuration is preferred for bias-stability reasons.

The bias point chosen is a compromise between least noise and low power

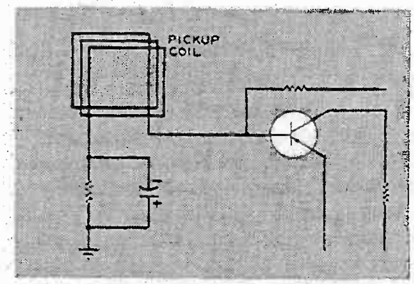


Fig. 16. This circuit can be used to couple pickup coil to transistor stage.

loss, and a desire to have good stability. The smallest input signal that can be handled is determined, at audio frequencies, by transistor noise, and at DC by temperature drift of the operating point. The greatest signal input is determined by how close the bias point lies to transistor nonlinearities.

The transistor is at a disadvantage compared to the vacuum tube for transducers which require a high loading resistance; in such cases, noise will be

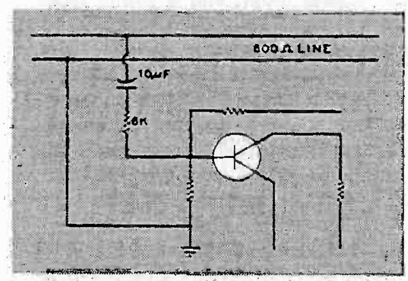


Fig. 17. Transistor circuit for bridging low-impedance line replaces transformer.

much worse. The advantages show up, however, when we consider the low-impedance transducers such as magnetic devices — recording heads, microphones, cartridges, etc.

Further Reading

Transducers and Sensitivity

Langford-Smith, F. *Radiotron Designer's Handbook*. Harrison, N.J.: Radio Corporation of America, 1953, pp. 701-723, 775-782.

Cleland, A. L. "Flattening Response of Crystal Pickups." *Electronics*, XXX (May 1, 1957), pp. 181-183.

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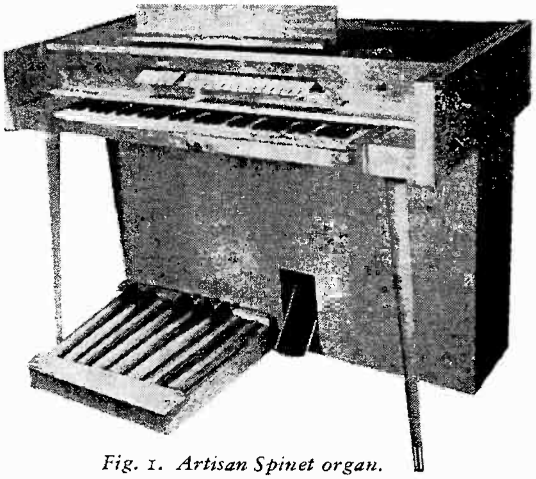


Fig. 1. Artisan Spinet organ.

The Electronic Organ: King of Kits

by FRANK R. WRIGHT

PART II: Assembling the electronic components

PIPE organs, while never having lacked devotees, have never achieved very great popularity as household musical instruments. They are just too big. More suited to the temper of the times is the electronic organ, although no full-fledged pipe-organ buff will admit the newcomer as a substitute for his cherished instrument. The electronic organ, however, must be accepted as a proper musical instrument in its own right, and, even as a compromise between the ideal and the practical, it has its place. Certainly there is much interest in electronic organs today, and a prime factor which prevents many people from owning one is the high price.

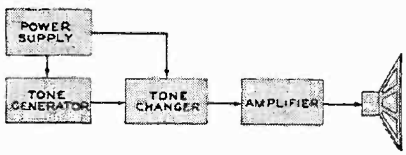


Fig. 2. Block diagram showing essential components of the Artisan Spinet organ.

The classic way to beat high prices is to build one's own. There *are* individuals courageous enough, persistent enough, and talented enough to build an electronic organ from the plans up; but the practical alternative for most of us is a kit. With a kit, the ordinary mortal can accomplish miracles. With an organ kit, the builder does not have to be an expert in electronic-organ design; he need know very little about the theory of the instrument; yet, without too many tears, he can turn out an instrument of which he will be justly proud. Many kits nowadays have been simplified to the point at which there is no longer an element of adventure in putting them together. Parts are dropped into place in a printed-circuit board, a touch of solder is applied, and she's ready to go. No thought necessary, and very little manual dexterity! The organ kit comes at just the right moment to save the day for all red-blooded, dyed-in-the-wool kit builders.

Fig. 3A. Power supply furnishes voltages for the tone generators and tone changer.

The Artisan Spinet organ (Fig. 1), discussed in this article, is manufactured by Electronic Organ Arts of Los Angeles, California. Artisan organs are true electronic instruments, in that their tone is generated by electronic oscillators employing vacuum tubes. Kits range in size from the single-manual Spinet to the Classic Model J-4 with four manuals. Altogether there are 14 different Artisan models, with prices ranging from $995 for the Spinet kit to several thousand dollars for the larger instruments. The cash saving in buying an organ as a kit, rather than as a finished instrument, runs in the neighborhood of 50%.

Artisan kits offer several definite advantages. For one, they are sold à la carte; that is, they can be purchased piecemeal. If your ambitions run to a mammoth, three-manual instrument, but your purse can't meet the price all at once, it is possible to buy a few parts at a time and build the organ up manual by manual. As each stage of the organ is finished, it can be mounted in the console and played while the next stage is still under construction. Or, suppose you began organ building on a modest scale with the Spinet, and then decided after a year or so to go on to something more elaborate. All your work will not have been wasted as far as the larger organ is concerned. Artisan organs are all built up from the same parts, so that the tone generators, manual tone changer, and power supplies of the Spinet can be transferred to a larger Artisan instrument, and Electronic Organ Arts will take the smaller console in trade toward a larger one.

Photos by Warren B. Syer

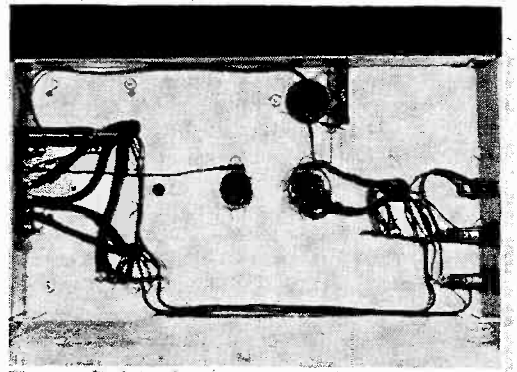
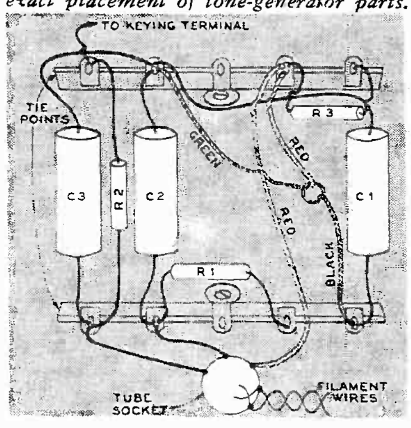


Fig. 3B. Below-the-chassis view of power supply illustrated in Fig. 3A. (below).

The essential parts of the Artisan electronic organ are shown in the block diagram, Fig. 2. The presentation is greatly simplified, but it will serve to make the organ somewhat less formidable to anyone not already familiar with the basic components of the instrument. The amplifier and loudspeaker system are not part of the kit, although Electronic Organ Arts can supply them. I used an amplifier and speaker system I had on hand and they worked out very well. Power requirements for the amplifier are not particularly demanding, and 20 watts will usually suffice for a home installation. It is axiomatic, of course, that the better the amplifier and loud-

Fig. 4A. Pictorial diagram showing the exact placement of tone-generator parts.



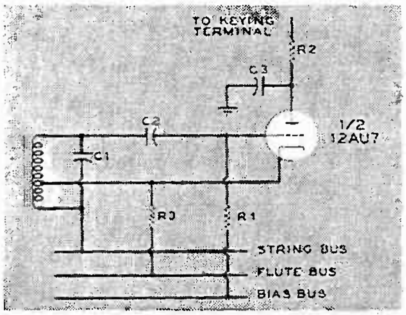


Fig. 4B. Schematic diagram illustrating electrical characteristics of Fig. 4A.

speaker system, the better the organ will sound.

The power supply in the block diagram is the high-voltage power supply (Figs. 3A and 3B). This furnishes all the necessary power for the filaments and plates of the vacuum tubes in the tone generators and tone changer. The output of these units is limited, and so the number employed will depend on the size of the organ. For the Spinet, one high-voltage supply is sufficient; a two-manual instrument would require two high-voltage supplies.

Artisan organs have a separate tone-generating oscillator for each note. The form of the oscillator is an L-C tank, and the pitch of the note generated is determined by the size of the coil and the capacitor. Tuning is accomplished by changing the air gap of the coil. The complete circuit for a one-note tone generator is given in Figs. 4A and 4B. Keying voltage is applied to the plate of the 12AU7 through R2, which, in conjunction with C3, prevents key clicks and provides a slight time delay to imitate the effect of a pipe organ. To keep the tank oscillating, a portion of the signal is fed back through the capacitor C2. Twelve tone generators, covering one octave, are mounted on each tone-generator chassis (Figs. 5A and 5B). Seven octaves are employed, the lowest note being 32 cps.

The tone changer (Figs. 6A and 6B) colors the tones produced by the tone generators. Each manual, or keyboard, of the organ has its individual tone changer to provide the stops for that manual; the pedal section also has its own tone changer. Even though the Spinet has only one manual, the keys can be divided electrically so that the left hand can be playing an accompaniment on the flutes, with the right hand playing the melody on a string or diapason stop. Where a greater variety of tone is wanted, two tone changers can be connected to a single manual, but two is the practical limit.

The voicing of the stops can be changed radically by changing the values of the components in the various sections of the changer circuit, or by adjusting the tone reactor chokes. With a little experimentation, the builder can alter the voices so that they are entirely different from the ones originally provided. The manual tone changer of the Spinet furnishes seven speaking stops: *Violin Diapason*, *Harmonic Flute*, *Stopped Diapason*, *Salicional*, *Vox Humana*, *Oboe*, *Gamba*; and six couplers. An electronic vibrato mounted on the manual tone changer provides a combination amplitude and frequency modulation. This is adjustable both as to speed and depth. The pedal changer (Figs. 7A and 7B) has four speaking stops: *Diapason*, *Dulciana*, *Bourdon*, and *Gedekt*. Since each stop operates its own preamp, an additive effect is obtained as each stop is switched on.

There is a low-voltage power supply (Figs. 8A and 8B) not shown in the block diagram. It furnishes the power to operate the manual coupler slides, the manual-to-pedal coupler switch, and the pilot light. AC power for this power supply is furnished by the filament transformer.

So much for the parts of the organ and what they do. Now let's see how to put them together and how difficult a job it is.

Man Working

To tell the truth, it looked like a very difficult job when the kit arrived and I was confronted with boxes and boxes of parts. There was even a time of panic when I was tempted to forget the whole business and let the magazine's honor look after itself. Rather than give way, however, I steadied myself by mobilizing my tools for the work ahead.

The tool requirements are not exten-

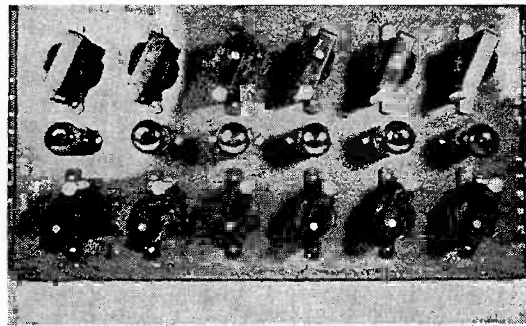


Fig. 5A. This chassis contains twelve tone generators which cover one octave.

sive, but it is important to have the right ones so that no mistakes will be made out of sheer clumsiness. I found that two soldering irons were a great convenience: a medium-size, 100-watt iron for the heavier soldering jobs, and a small, 25-watt pencil iron for making the more delicate connections. Soldering guns are fine for them that likes 'em, but I prefer an old-fashioned iron for this kind of work. Other tools required are a pair of diagonal wire cutters, a

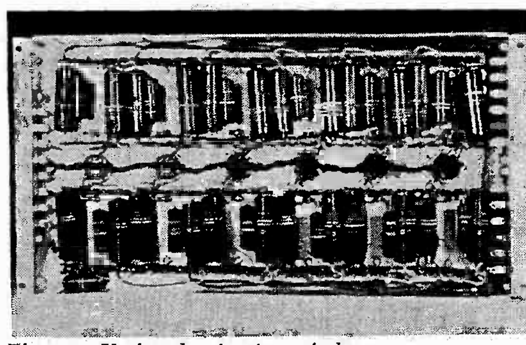


Fig. 5B. Under-chassis view of the tone generator assembly illustrated above.

medium-size phillips screw driver, a selection of three or four different sizes of regular screw driver, a 5/16-inch speed wrench, and a pair of long-nosed pliers. Besides these tools, it is helpful to have a pair of needle-nose pliers and a spin-on wiring tool. A VTVM will come in handy too, if it is necessary to do any trouble shooting later on. Armed with these, you should be ready to take on any Artisan organ kit yet devised.

In constructing the Spinet, I took advantage of a couple of short cuts which Electronic Organ Arts offers to its cus-

Fig. 6A. This is a top side view of the tone changer which colors all of the tones produced by the tone generators.

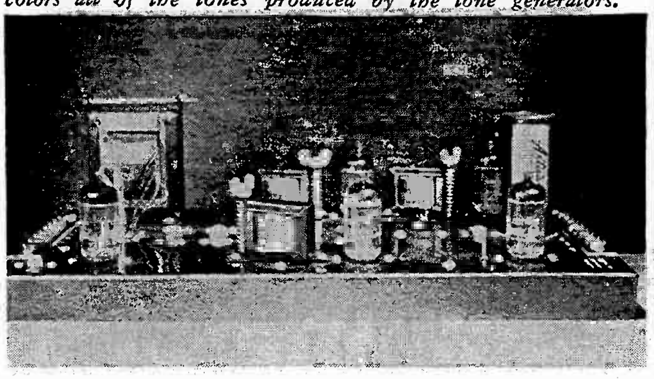
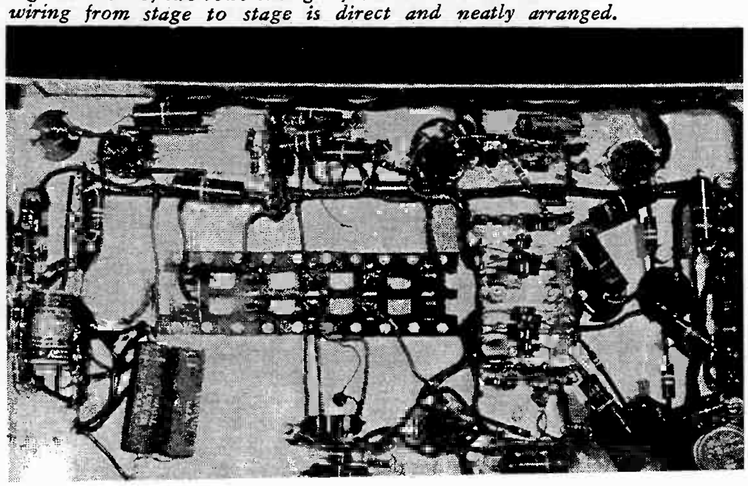
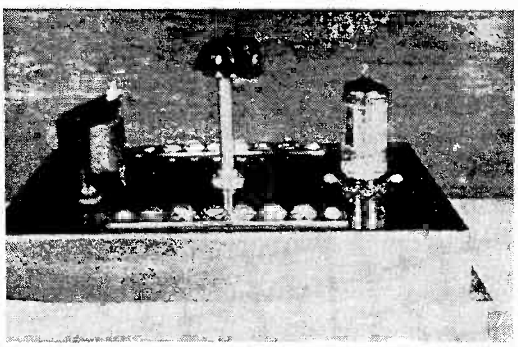
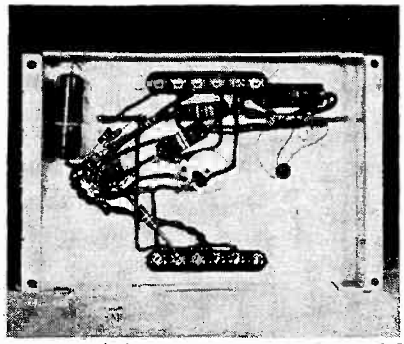


Fig. 6B. View of the tone changer from underneath shows how wiring from stage to stage is direct and neatly arranged.







Figs. 7A and 7B. These photographs show top and bottom views of the pedal changer which controls stops for pedal section and contains top pedal note.

tomers. For one thing, the organ console (cabinet) was finished at the factory, saving me a good many hours of hard work. It also adds $100 to the cost of the kit. I used ready-made cables in wiring the organ, and these would have added another $40 or so to the price. Even with these time savers, it took me about 120 hours to assemble the organ from start to finish.

It has been said many times in the pages of this magazine, and it bears repeating, that the first thing to do in constructing any kit is to check the parts against the parts list. This caution holds true for the organ kit, and before each component is assembled, the parts should be taken out of the box and checked against the parts list. Each component is boxed separately and comes with its own set of instructions, so there is no possibility of turning out hybrid components.

Because, when I began work on the Spinet, I was almost totally unfamiliar with the workings of electronic organs, I thought the safest starting point would be one of the components least strange to me. I would recommend this procedure to anyone else in the same position. I assembled the high-voltage power supply (Fig. 9) first. As can be seen from the schematic, this unit is simple and straightforward, and it's easy to check out when it's finished. It took about four hours to wire the chassis and check it.

The second component assembled was the low-voltage supply (Fig. 10). The low-voltage supply is the simplest component of the organ, and assembly time was a mere hour and a quarter.

Having whetted my appetite on two easy assembly jobs, I felt ready to tackle some of the more mysterious parts, starting with the pedal tone changer (Fig. 11), which turned out upon closer acquaintance not to be so mysterious after all. The pedal tone changer provides the four pedal stops on the Spinet, and also furnishes the top note for the 13-note pedal tone-generator system. Its output is fed directly into the organ's amplifier. Construction time for the pedal changer was about four hours.

I recall feeling, at this point, that the organ wasn't going to be nearly so difficult as it had appeared at the beginning. I seemed to be making satisfactory progress and had encountered nothing very difficult. All I had actually done was to get the simplest jobs out of the way, and I slowed down when I encountered the manual tone changer (Fig. 12). It took 11½ hours to put it together. The manual tone changer provides the speaking stops and couplers for the Spinet's manual keyboard. Despite the fact that I tried to check my work carefully after each unit was completed, a mistake in the wiring of the tone changer slipped through so that the *Harmonic Flute* and *Stopped Diapason* stops failed to become silenced when their switches were closed. The error was easily corrected when it was discovered later, but it points up the necessity of double-checking all work as you go along.

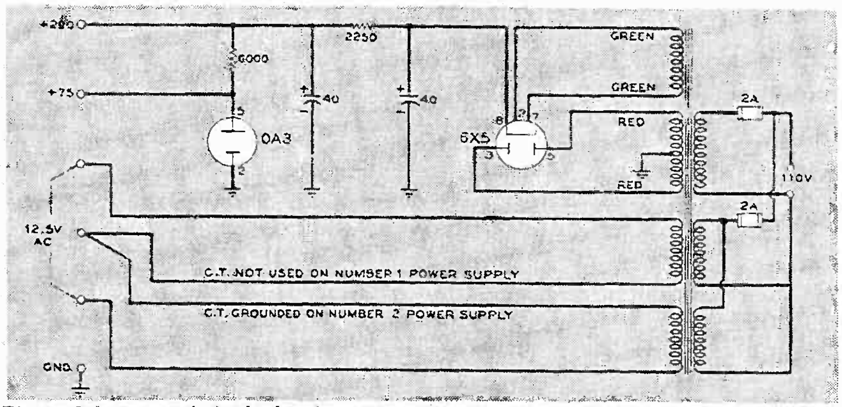


Fig. 9. Schematic of the high-voltage power supply. The author recommends starting with this component to get the feel of wiring before attempting other assemblies.

I had saved the tone generators until last because there were so many of them and, except for the values of the parts, they were identical. It seemed logical to suppose that, if I could do the same job several times over, I would develop a system of sorts and order the work to save as much time as possible. This was true, but, alas, I never became as proficient at assembling tone generators as the instruction book said I should. A good man, according to the book, can assemble a one-octave tone-generator chassis in six hours. I never even came

Fig. 8A. Top view of low-voltage power supply which powers manual coupler slides and manual-to-pedal coupler switch.

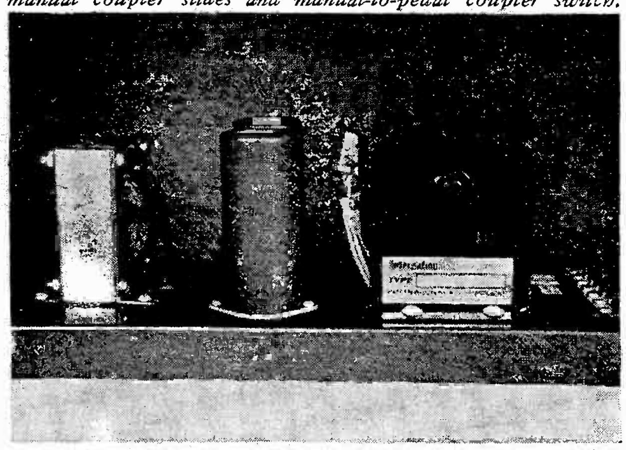
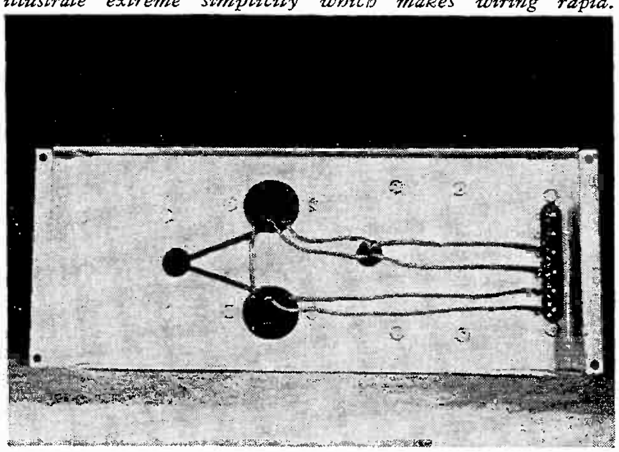


Fig. 8B. Mounting details of the low-voltage power supply illustrate extreme simplicity which makes wiring rapid.



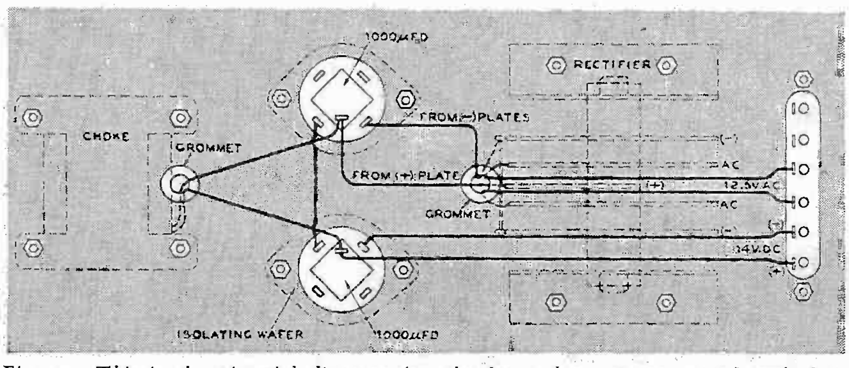


Fig. 10. This is the pictorial diagram for the low-voltage power supply which is shown photographically in Fig. 8A and 8B. Wiring this component took 75 minutes.

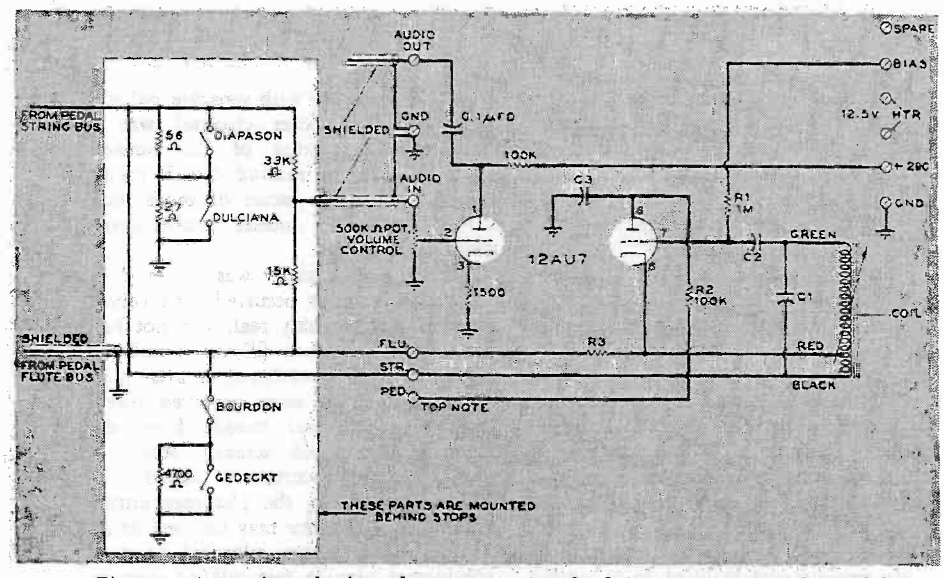


Fig. 11. Approximately four hours were involved in constructing the pedal tone changer shown here in a schematic drawing. See Fig. 7A and 7B for photographs.

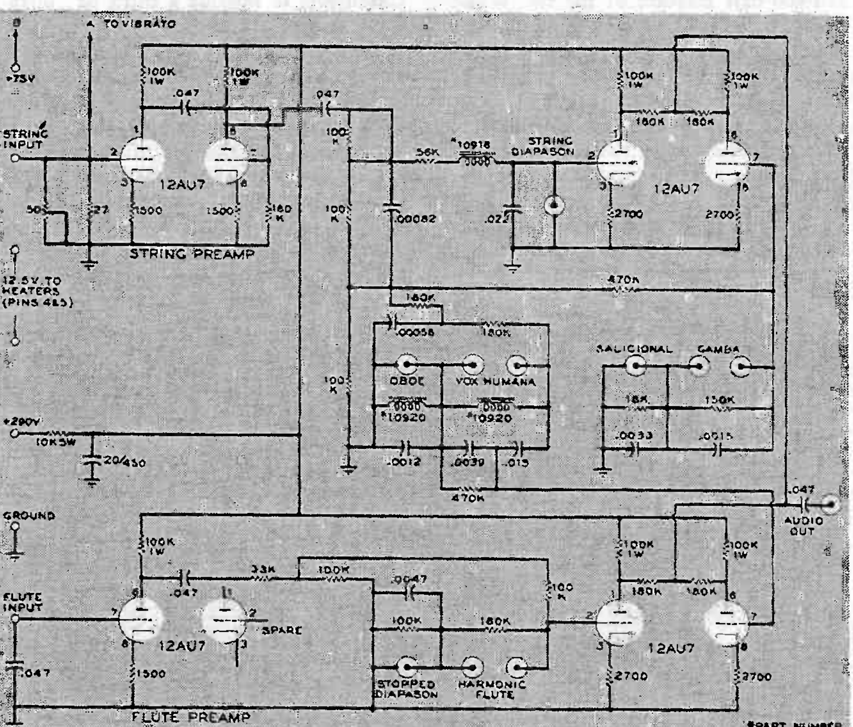


Fig. 12. The manual tone changer. The author states that this was the most difficult assembly of the entire kit, and construction time was about 11½ hours.

close. The first one I tried took 9 hours and 40 minutes; the second, 8 hours and 20 minutes; and, after that, they worked out to about 7½ hours each*. The technique I evolved, though, is included here for the benefit of any other tyro organ builder who may wish to try it.

First, put the tuning clips, bolts, springs, and nuts on all the oscillator coils for the octave under construction. Then, mount all the hardware such as tube sockets, terminal strips, and so on, on the chassis. The terminal strips should be positioned so that the prongs point toward the chassis bottom. The oscillator coils are put on next, with the tie points mounted on the under side of the chassis on the same bolts that are used to mount the coils. This work goes more easily if the side of the coil nearest the tube sockets is bolted first, but not tightened until after the outer side of the coil is bolted. The tube filaments are then wired.

The schematic of the tone-generator circuit (Fig. 4B, page 23) shows that the center tap of the oscillator coil is connected to the cathode of the 12AU7 tube. In the actual wiring, this lead does not go directly to the cathode pin of the tube socket, but is connected to it by a lead from one of the terminals of a tie point (see pictorial diagram, 4A, page 22). It is a good plan to put this lead in place right after wiring the filaments. Then it can be pushed down next to the chassis where it will be out of the way when the other components of the circuit are mounted. Other leads in proper order for wiring are the bias lead, string lead, and flute lead. The leads from the oscillator coil, starting with the red lead from the coil tap, are next. The flute bus resistor and capacitor, mounted in one corner of the chassis, should be wired in, and then the capacitors, in the order of C1, C3, and C2, for each note. After that, you're on your own. The logical next step is to put in the resistors for each note and follow with the rest of the connections from tube sockets, but I couldn't see that it made much difference what order this work was done in.

With the last of the electronic components assembled, a different type of work is in order. The next phase of the operation is to mount the parts in the organ console and wire them together. This step and the subject of putting the finishing touches on the completed instrument will be taken up next month.

*The Artisan factory offers printed- or etched-circuit tone-generator chassis for those who want to cut assembly time for these units in half. The printed-circuit chassis are the same price and same size, and are interchangeable with the regular aluminum chassis.

Two-Track, Three-Channel Stereo

by PAUL W. KLIPSCH

Two-channel stereo playback occasionally suffers from an apparent void in the central region between the playback speakers. This becomes noticeable, for example, if a soloist is in the center of the recording stage. While the two-channel playback system can be balanced or focused for any one listener so that the soloist is sensed to be in the proper location, a listener to the right or left may hear the soloist on the right or left of center.

Closer speaker spacing tends to bring center-stage sounds into focus, but in the process some of the breadth and separation is lost which stereo should convey. In this article Mr. Klipsch discusses a new technique that seems especially promising: employment of a third playback amplifier fed by both left- and right-hand stereo tracks, driving a third speaker placed between the two main stereo speakers. This would be a two-track three-channel system, the third channel being a phantom derived from a combination of the other two.

It is expected that a more formal paper on this subject will be published later in one of the professional journals, but, in view of the evident importance and wide applicability of the technique, Mr. Klipsch has kindly consented to present the following advance discussion in AUDIOCRAFT. Home experimenters will find that it contains all the information necessary for duplication of the 2T3C system.
—ED.

THE first published information on stereophonic sound using three channels appeared in 1934[1] (as far as this writer is aware), in the form of a symposium.

The part of this symposium entitled, "Physical Factors," by J. C. Steinberg and W. B. Snow is of interest to this discussion. They showed five different combinations of microphones, channels, and speakers, including one using two microphones and three speakers, mixing the two microphone outputs at —6 db to feed the center speakers.

Harlan Thompson[2] set two speakers in the center of the wall and used two flanking corner speakers, feeding one of the center speakers and one corner speaker from one of two stereo tape channels and the other center and corner speakers from the other tape channel. He reports obtaining a balance of sound across the room, retaining stereophonic effects of the two channels, but eliminating the bifurcated effect one sometimes gets with two channels alone.

My experiments resulted in a workable system with the center channel a half-and-half mixture of the two sound tracks, and the flanking channels using corner speakers fed from the two sound tracks with 3 db attenuation relative to the center channel.

Several unsuccessful experiments were tried, culminating in using two corner-type speakers in the room corners and a center speaker, all having substantially the same middle-range and upper-range efficiencies, and each driven by its appropriate amplifier with separate volume controls. The center channel was a half-and-half mixture of the outside channels, and the mixing circuit maintained a cross-talk factor of more than 20 db between the outside speaker channels.

When success finally was achieved in balance, a surprise occurred: the center channel was perfectly real, and not just a simulated effect to fill up a hole in space. Sounds remembered as arising in the center of the stage occurred there; one ceased to hear sounds from the three speakers, and actually sensed a spread across the curtain of sound.

For explanation, the phantom circuit of telephone practice may be used as an analogy, wherein two physical pairs of conductors provide two talking circuits, and the pairs themselves are used as conductors to provide a third or "phantom" talking circuit.

One immediately jumps to the conclusion that a three-sound-track stereo system should provide at least two, possibly even three, phantoms; and if one is going to the expense of three sound tracks, the only way to take full advantage thereof would be to plan the recording and playback geometry for as many phantoms as are practicable.

Fig. 1 shows the two-track three-channel system with one phantom. As in

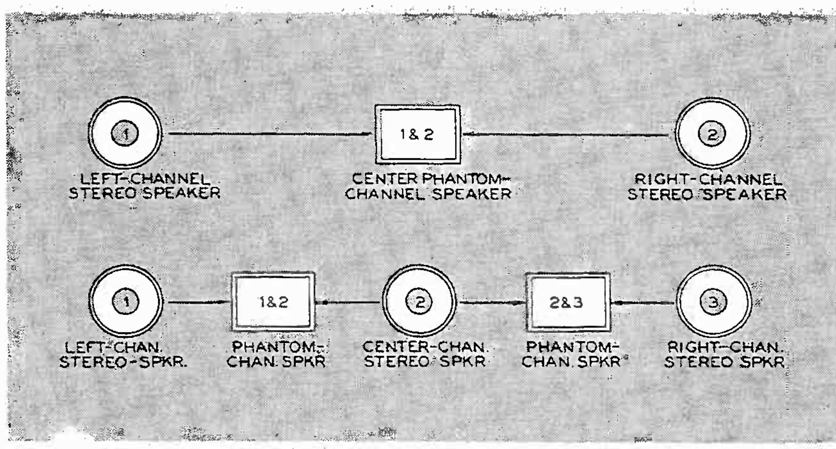


Fig. 1 (above). Diagram showing one phantom with a two-track, three-channel system.
Fig. 2. Customary three-track stereo in which two phantom channels have been used.

[1]Bell Laboratory Staff, "Symposium on Auditory Perspective," *Trans. AIEE*, LIII (1934). See also *Electrical Engineering*, LIII (Jan. 1934), pp. 9-32, 214-219.

[2]Personal communication. Mr. Thompson was then with Berlant-Concertone.

telephone circuits, there can be $n-1$ phantoms for n physical tracks.

Fig. 2 shows the customary three-track stereo with two ($n-1$) phantoms added. Experience suggests little is to be gained with this addition of channels.

Fig. 3 shows how three physical tracks might be employed to real advantage to cover a domain of two actual space dimensions. It is fascinating to conjecture how one might attain a true "3D" or three-space dimensional effect with a practical small finite number of tracks and channels. But the purpose of this paper is to discuss the single lineal dimension afforded by the two-track three-channel array.

Amplitude for Center Channel

Some guessing was done as to the level to be fed the center channel, and the guesses were all wrong. Experiment led to better thinking, and a theoretical basis was arrived at and corroborated. To divorce the concept from physical circuitry seems to complicate rather than simplify; therefore, consider Fig. 4.

Suppose each track source to be of negligibly low impedance, so that the inputs to the output-channel amplifiers are of high impedance compared to R_1. Then assume that track 2 contains no signal. The phantom receives half the voltage of track 1. If track 2 contains a signal and track 1 does not, the phantom receives half the signal of track 2. If tracks 1 and 2 contain signals equal in intensity and phase, the phantom receives the same signal.

But assume that tracks 1 and 2 are not equal, as would be expected when two microphones are used 30 ft. apart. Now the center channel does not receive a signal represented by $a/2 + b/2$ as a scalar quantity; its average effective signal will be $\sqrt{2}/2$, approximately 0.71, or 3 db down from the outer channels. Thus the flanking channels likewise should be attenuated 3 db.

Fig. 5 shows the circuit finally developed to achieve balance.

The 82-K and 220-K pad elements provide 3 db attenuation for the outside channels. The 33-K resistors perform the mixing function. The source of signal was a Berlant 30 with added pad attenuators of about 5 K output impedance, and the amplifiers fed by the mixing box in Fig. 5 presented 500-K load impedances to the box. Therefore, the cross talk from track 1 to channel* 3 would be approximately 5,000/66,000, or slightly better than 20 db, which is deemed to be more than adequate. The 500-K amplifier inputs constitute negligible loads for the impedances within the box.

The center amplifier was provided

Continued on page 45

*Note that "track" implies a source: there are two sound tracks on the tape, thus two sources; "channel" denotes speaker locations, of which there are three.

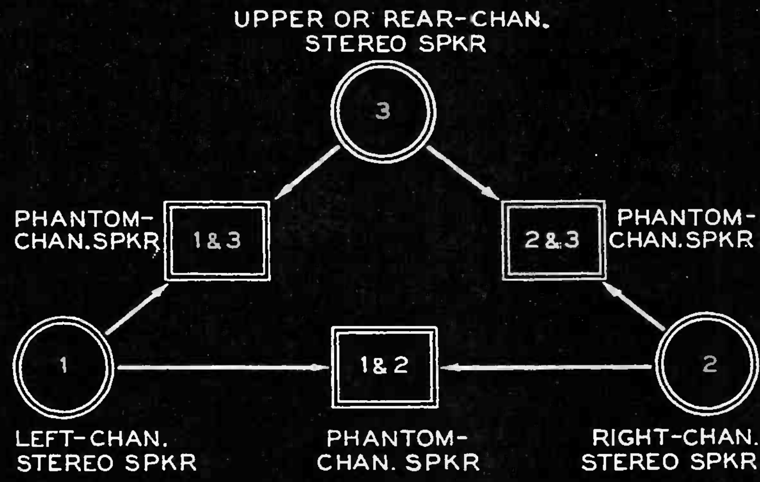


Fig. 3. Theoretical positioning of three phantom channels with three-track stereo.

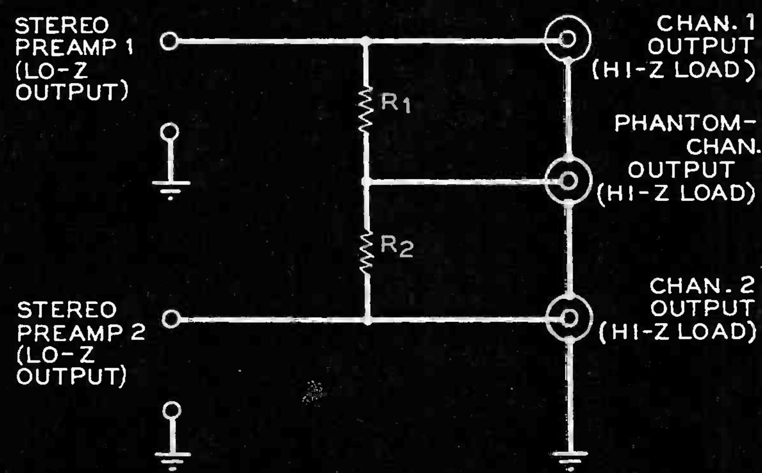


Fig. 4. Method by which phantom channel is derived from two-track stereo input.

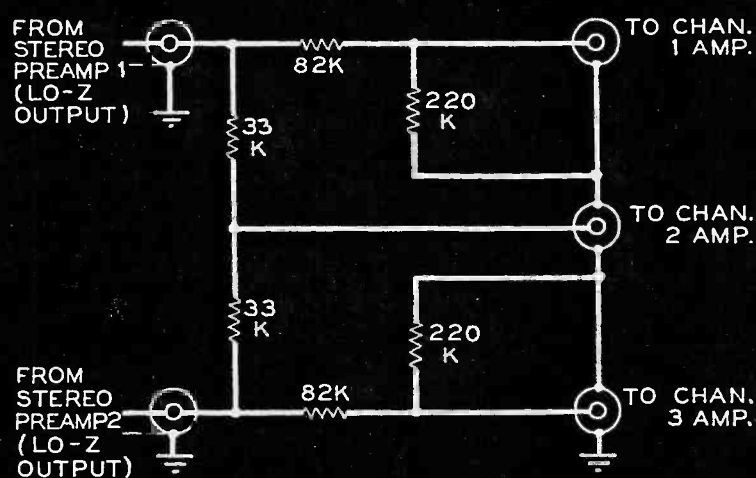
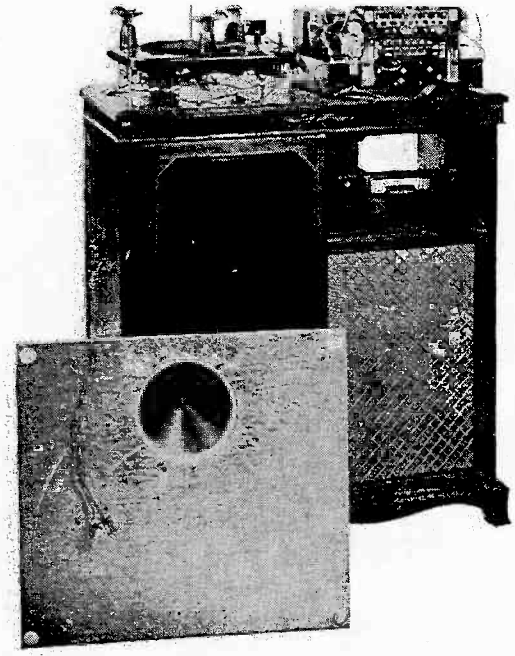


Fig. 5. Circuit as ultimately developed with values which will produce balance.



SOUND in a Sealed Package

HOW large is *your* hi fi?

If it's like most, chances are it's either spread out in separate components extending from one end of the living room to the other, or else it's housed in a massive cabinet, or series of cabinets, the sum total of which probably takes up one entire side of your listening room. Where's the amplifier? In another room? So there — your system takes up *two* rooms.

As a matter of fact, so do most of the sets we know of, but the current trend in high fidelity — aside from the continuing battle against distortion and uninvited noise — is toward something which, for want of a better definition, might be termed "smallerization."

And, in that vein, here is one approach to the trend as advocated by the Permoflux Products Company of Glendale, California. It's a system which the Permoflux engineers like to call "Sealed Package Sound." The gist of the idea hinges on the fact that a speaker enclosure, when fully varnished, grille-clothed, and otherwise prettied up, can be a downright costly box. The Permoflux people feel that, in many cases, something near half of the cost of a ready-built speaker enclosure goes to pay the carpenters who made it a piece of furniture.

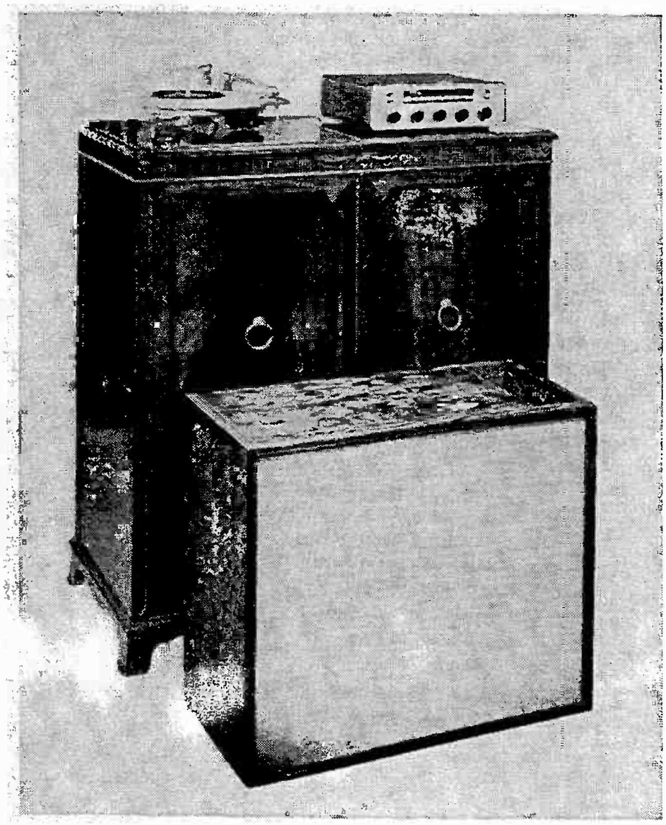
Not that Permoflux has anything against pretty furniture. They just feel that the average do-it-yourselfer would rather put the finishing touches on a ready-built basic enclosure himself and pocket the difference. So, with only good, clean sound in mind, they have designed two speaker enclosures, equipped them with speakers, built them with sturdy plywood, and painted them black.

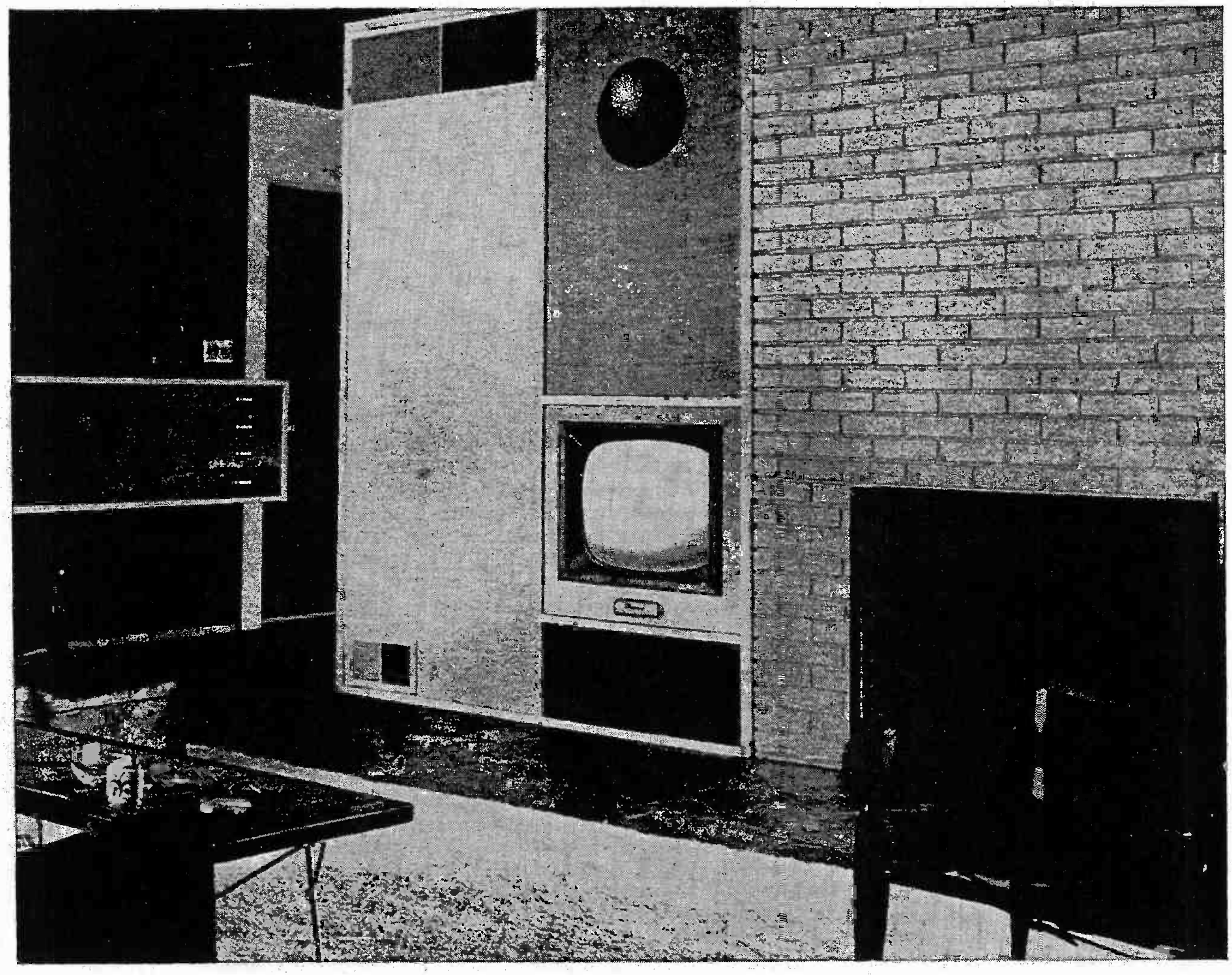
They call their system "Sealed Package Sound," because the sound of the speaker enclosure is independent of the cabinet that you, the builder, put it in. It's sealed in, and the unit is adjusted and ready to play as it arrives. All you have to do is hunt around your attic for an unused cabinet or what-have-you — of the proper size, of course — tear out the innards, and hook in the Permoflux "Sealed Package Sound" speaker system.

Or, the Permoflux people hasten to add, you can build your unit into a bookshelf, a closet door, antique furniture, a desk, an old radio cabinet, a bar, a room divider or planter, a ceiling or wall, or just about anywhere your wife will permit.

And then, tucking the dollars you have saved back into your wallet, sit back and listen. This, Permoflux says, will be your second big thrill.

After modification, the equipment cabinet (above, left) takes on a new appearance as "Sealed Package Sound," combined tuner-amplifier, and modern changer replace older-model equipment.





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BASIC ELECTRONICS

by Roy F. Allison

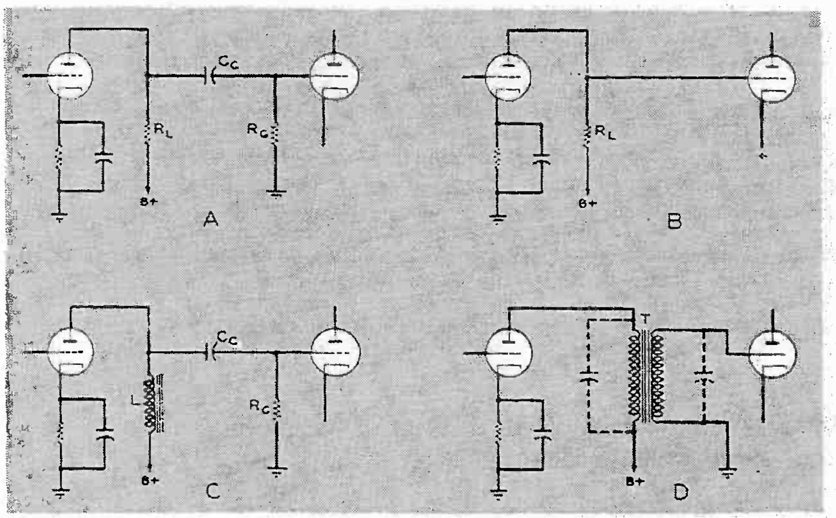
XX: Operating class; coupling methods

IN preceding chapters we have described in considerable detail the process of voltage amplification in vacuum tubes, and the kinds of tubes commonly used in audio circuits. Signal voltages applied to the control grid in each case produce corresponding changes in plate current; the plate-current variations are responsible for variations in voltage drop across the plate load; and these plate-voltage variations constitute the output signal, which ideally is an enlarged replica of the input. It will be recalled also that positive changes in grid voltage produce increased plate current, and negative changes in grid voltage, decreased plate current. Thus the plate-current variations follow the signal at the grid, or are in phase with it. But increasing plate current reduces the plate voltage; therefore, the output voltage variations representing the signal are opposite in phase with the input signal. The tube reverses the phase of the signal, on a voltage basis, besides amplifying it.

This phase reversal is not comparable to the phase shift encountered in reactive circuits. Reactive phase shift involves a definite shift in time, whereas an amplifying tube simply turns the signal upside-down almost instantaneously. Coupling circuits between successive amplifying stages usually employ capacitance or inductance, however, and reactive phase shift occurs in these circuits at frequencies for which the circuit reactance becomes comparable to the resistive component. At such frequencies the signal also begins to be imperfectly passed by the coupling circuit; that is, the circuit begins to attenuate the signal. Without attenuation there is no reactive phase shift and, since coupling circuits are designed to pass the frequency range of interest without appreciable attenuation, it follows that phase shift within the useful frequency range is slight. It is of no importance in practical audio amplifiers, therefore, except when the circuit is enclosed within a feedback loop. This subject will be developed further in another chapter.

For the moment, let us look more closely at coupling circuits and their purpose. In all our illustrations so far we have used an *RC coupling* network, as shown in Fig. 1A. The output voltage of the first tube is developed across R_L. This AC signal is superimposed on the positive DC plate voltage. If the grid of the following stage were simply connected directly to the plate of the first stage, the AC signal would be coupled as desired, but the high DC voltage would also be applied to the grid. Since the grid must be at a DC potential slightly ***negative*** with respect to the cathode, in order to achieve a proper bias, a direct connection isn't often practical. The coupling capacitor keeps the DC plate voltage off the grid; but, provided the time constant of C_G and R_G is long enough, the AC signal applied to the two in series is developed primarily across R_G, and in that way "coupled" to the grid.

Fig. 1. Conventional methods of coupling between stages. Diagrams show RC coupling (A), direct coupling (B), impedance coupling (C), and transformer coupling (D).



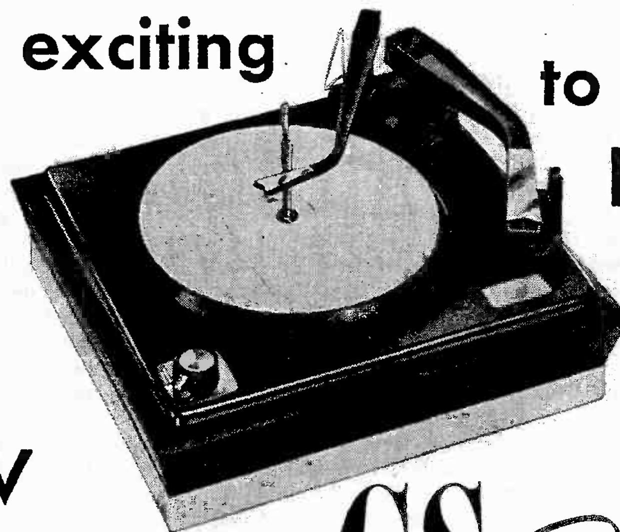
The DC potential at the bottom of the grid resistor is the same as that at the grid, because there is no DC current flow in the resistor. If the grid resistor is grounded, as it often is, then the grid is at DC ground potential; bias is obtained by making the cathode slightly positive, either by applying a fixed voltage to it or by means of a cathode bias resistor. Alternatively (although this is less common), the grid resistor may be connected at the bottom end to a small negative voltage for bias, and the cathode grounded directly.

There are a few circumstances in which *direct coupling*, Fig. 1B, may be used to advantage. In any such application the circuit must be set up so that the DC potential of the following stage cathode is slightly higher than the preceding plate's DC voltage *before* the connection is made, so that the driven tube's normal-bias operation will not be disturbed by the connection. The only important audio circuit in which this is practical is the split-load phase inverter direct-coupled to the preceding voltage amplifier stage, as in the Williamson amplifier. Direct coupling is occasionally employed from cathode-follower drivers to power output stages, and in a few other special cases. These are relatively rare, however.

Fig. 1C shows the *LC coupling* method, sometimes called impedance coupling. This is basically the same as RC coupling except that the signal is developed across an inductance in the plate circuit rather than across a resistance. The inductance is made large

Continued on page 48

what could be more exciting to a high fidelity enthusiast?



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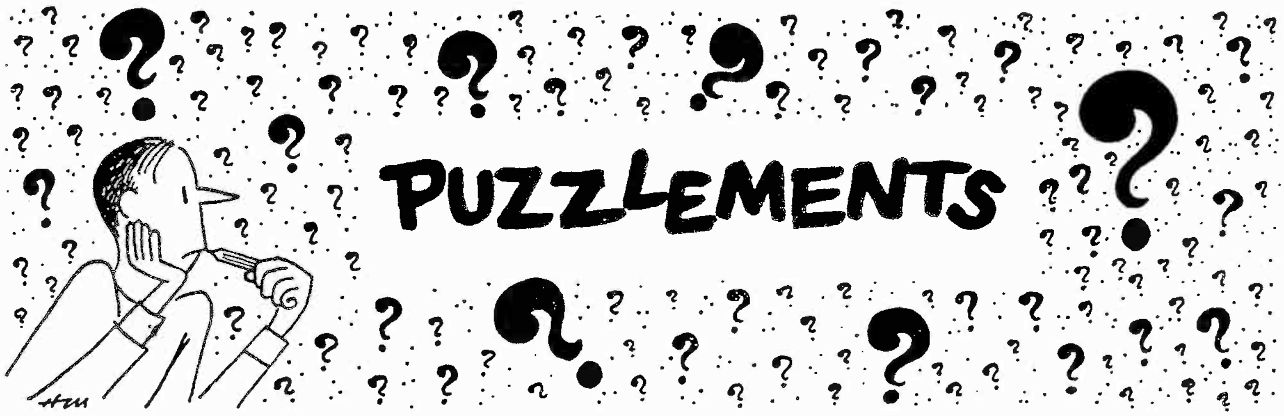

PUZZLEMENTS

by NORMAN H. CROWHURST

IN the musical, *The King and I*, there is a song ("Puzzlements") sung by the King in which he laments losing the certainty he had as a youth. Then, as he recalled, what was supposed to be so was certainly so; what was supposed to be not so was actually not so, and no nonsense about it. Now that he has gained years and experience, he finds that there are few things he is absolutely sure of.

This description well fits many people interested in high-fidelity reproduction. We start by thinking the facts are relatively simple, even though they pose some tough problems. For example, if we can get a smooth, flat frequency response in an amplifier, a pickup, a loudspeaker, and all the other components in the audio chain, we assume that we shall have virtually perfect reproduction — imperfect only in the degree that the components fail to meet the required standards.

This "fact" seems to be so simple and obvious as not to require proof. Then, years later, when we get to know a little more about it, or start listening to different combinations, we suddenly find that these things are only "nearly so."

One example of this came to light at a recent audio show. A friend of mine, who was interested in acquiring an electrostatic tweeter, went round to several exhibits in which one was used, with the idea of getting an impression of how good the tweeter sounded and what it would do for his own system at home. In one room he heard an electrostatic tweeter which seemed to complement the dynamic woofer and the middle-range units beautifully, so as to give a very smooth, sweet-sounding tone.

In another room, however, an electrostatic tweeter of the same type was rendering the extreme highs quite poorly. It sounded good on percussion, but on instrumental material it was somewhat harsh. Also, it was very noticeable that background noise, such as hiss from the needle riding in the groove, was particularly pronounced. As one person in this second room put it, "It seems that all the tweeter does is to give more hiss."

Why did the electrostatic tweeter work so wonderfully in one room, while in the other room it seemed so poor? My friend who raised this question got the exhibitors interested in solving the problem.

Each of them checked his equipment and found that the amplifier and other elements in the chain were performing up to standard. On the face of it, it seemed that the second exhibitor must have somehow gotten a poor sample of the electrostatic tweeter. So they decided to try exchanging units. both of them still had the same results: the second exhibitor's new tweeter was a lemon while exhibitor No. 1 was still getting good performance.

This seemed to prove that there was nothing wrong with the tweeter, so what about the amplifier? A quick check of voltage and a frequency run with a borrowed oscillator showed nothing measurably wrong with the amplifier. Then someone had the bright idea of comparing both amplifiers using a different kind of tweeter.

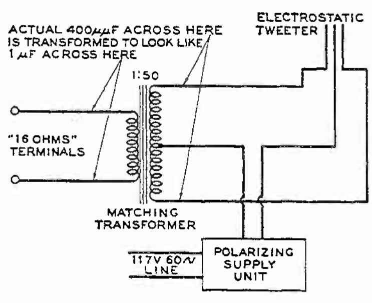

Fig. 1. Connecting transformer directly changes tweeter's effective capacitance.

Checking the amplifiers with several other tweeters (not electrostatics) there was no difference. The amplifiers used by both exhibitors seemed to give unimpeachable performance. More critical measurements made later showed that each amplifier gave a frequency response indistinguishable from flat (audibly, at any rate) and with distortion which also should be inaudible. What, then, had caused the harsh reproduction from the second exhibitor's combination (and which returned when the electrostatic unit was reconnected), while exhibitor No. 1 achieved what apparently should happen in theory?

There was a puzzlement.

This particular puzzlement was eventually solved by applying a capacitor on the output of the amplifier of the same value as the reflected capacitance produced by the tweeter. The tweeter is supplied with a matching transformer which gives it a *nominal* impedance of 16 ohms. Use of this built-in transformer means it should readily connect directly in parallel (or through a crossover network) with the woofer and middle-range units of like impedance — in this case, 16 ohms. My friend had been taught to believe that a matching transformer "transforms" one impedance to another, so he imagined that putting a matching transformer between an electrostatic tweeter and the amplifier converted its impedance into an *actual* 16 ohms.

This definitely is not true.

The impedance actually produced depends on just what the manufacturer does between the transformer and the tweeter. If the transformer is connected directly to the tweeter, with a polarizing supply but no resistance (Fig. 1), all the transformer does is to convert the tweeter's effective capacitance to a different value.

Suppose the transformer produces a 50-to-1 step-up; this is an impedance ratio of 2,500 to 1. So the effective impedance due to the capacitance at the primary side of the transformer will be the actual impedance on the secondary divided by 2,500. Because the imped-

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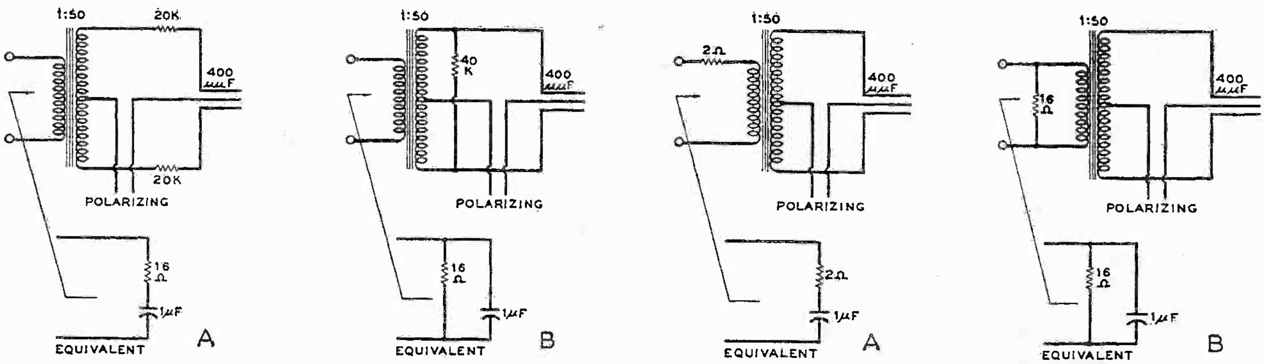

Fig. 2 (A and B, left) and Fig. 3 (A and B, right). The method of adding resistance to the tweeter circuit in series (Fig. 2A) and parallel (Fig. 2B). Resistance can also be added to the primary circuit of the matching transformer (Fig. 3A, and 3B).

PUZZLEMENTS

Continued from page 32

ance, or reactance, of a capacitance is inversely proportional to its capacitance value, the effective capacitance at the primary will be 2,500 times the capacitance of the electrostatic tweeter.

Assume the electrostatic tweeter is about 400 $\mu\mu$fd in capacitance. The transformer will have the effect of multiplying this value by 2,500, so the primary side of the transformer "looks like" a capacitor of 1 μfd. The amplifier then performs as if a 1-μfd capacitor were connected across the 16-ohm output terminals. The effect of adding resistance in the tweeter circuit is shown in Fig. 2.

The "padding" resistance can also be connected in the primary circuit of the matching transformer, as shown in Fig. 3. The arrangement at Fig. 3A is a simplified equivalent for the JansZen unit, connected for 16 ohms use. The remaining components in the JansZen circuit, not shown here, serve to avoid a resonant peak due to interaction between the inductance of the woofer and the capacitance of the tweeter—an effect more noticeable with some amplifiers than others, but which is apt to get blamed on the tweeter "because it was not there before."

When we tried measuring the response of the two amplifiers in question with a 16-ohm resistor to represent the low- and middle-range speakers, in parallel with a 1-μfd capacitor to represent the tweeter, the reason for the peculiar performance became evident. The first amplifier still gave a smooth frequency response, with a slight rolloff up in the region of 20,000 cps, while No. 2 amplifier produced a colossal peak in the region of 13,000 cps. Occasionally, when the oscillator input was switched or changed rapidly, it was noticed that there was a parasitic oscillation on the output wave form when looked at on the scope. So the amplifier worked very well with a 16-ohm resistive load and also did quite well when connected to a wholly dynamic loudspeaker load, but when the electrostatic capacitive load of about 1 μfd was connected, the amplifier

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WEBSTER'S *New Collegiate Dictionary* (current edition) defines noise as "sound without agreeable musical quality." This is perhaps an unfortunate definition as far as we are concerned; anyone familiar with high-fidelity recording knows that a great deal of the *desired* sound on recordings is strictly nonmusical, even though it may often be agreeable. The cacophonous thunder of a steam locomotive à la Emory Cook may be music to the ears of some people, as may the nervous extreme of contemporary classical music, but it is clear that the noise we refer to when we complain about tape noise or preamplifier noise is obviously noise of another genre.

Actually, what we mean is *unwanted* noise or, even, unwanted sound (if we wish to include the background of audience mutterings and coughs at a concert performance). So let's just call it extraneous noise and let it go at that.

Poring through the available literature on the subject, I gather that there are about four different kinds of noise that annoy tape recordists. These are acoustical noise, electrical interference, hum, and hiss.

We're going to narrow the field down even more and consider in detail only the hum and hiss that arise as a direct result of using tape as a recording medium, rather than film or discs. Unfortunately, hum and hiss that are audible from tape may be coming from the tape, or they may just as well be coming from somewhere else. On the other hand, they may not, and that's where there is likely to be some confusion unless we succeed in localizing the source of the noise.

Let's look at hum for a moment (on a scope, if we wish). Hum can be 60 cps, 120 cps, or both, with or without other harmonics added. As a rule, 60-cps hum is caused by leakage from the amplifier's heater supply; 120-cps hum is the result of inadequate filtering in the B+ supply; and so-called harmonic hum (which is noted for its peculiarly musical and sometimes wavering sound) generally comes from capacitive coupling between an unshielded signal lead and a

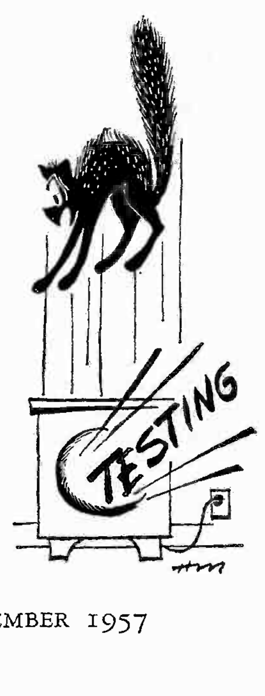
TAPE NEWS & VIEWS

Minimizing Tape Hiss

conveniently close source of AC radiation, such as a power transformer.

Now that we all know where hum in an amplifier comes from, we can delve into any one of a number of excellent books on servicing or design, and solve the problem for ourselves. The only reason I mentioned hum at all was to emphasize the fact that, if you've got it on your tapes, something is wrong with the recorder — not with the tape medium. Actually, there has been so much written about the elimination of hum in amplifiers and preamplifiers, that it hardly seems worth while going into it again. On the other hand, service manuals and design texts are almost universally silent when it comes to the hum problems that are unique to tape recorders, so I shall attempt to fill in the gaps.

There are only two ways in which hum can conceivably get *onto* a tape. In rare cases, it has been known to originate from the ultrasonic bias supply, in which inadequate B+ smoothing or a drastic heater-cathode leak in the oscillator tube has managed to superimpose a 60- or 120-cps modulation onto the bias signal. The solution to this is obvious: add some more B+ smoothing, or replace the tube, depending on the frequency of the hum. About 99% of hum-ridden tapes have become so because of preceding stages in the recorder — either from the preamplifier or amplifier stages themselves, or from hum pickup in the microphone, phono pickup, or interconnecting cables. On the other hand, the hum you hear may *not* be on the tape itself, but may be coming from the playback head.



So, let's make a test. Set the recorder and everything up as if in preparation for playing a tape. Turn everything on, adjust the volume control to the usual playback setting, but don't put any tape on the recorder. If it has any safety relays on it that would normally shut the transport off when the tape runs out, prop these open. Now start the unit running, and listen for hum. If the hum level is much lower than is normally heard from recorded tapes, two possibilities are suggested; the excessive hum is coming from the bias oscillator, or it is coming from a part of the amplifier's external circuitry that is not connected when playing tapes. In either case, conventional servicing techniques will solve the mystery, given enough ingenuity, patience, and the will and stamina to fight against overwhelming odds such as tightly wrapped resistor pigtails, buried components, and the lack of a legible schematic for the recorder's electronic section.

Let's assume, however, that the test described above has turned up a nice case of hum. Now we have several interesting possibilities, depending upon the type of recorder that is acting up.

If the recorder is of the usual variety, having a single head for recording and playback, chances are that all or most of the amplifier circuitry is common to both the record and playback functions. If this is so, it is possible that hum heard in playback (without tape) is also going onto any tapes that are made on the recorder, simply because a humming amplifier will feed its hum equally to a record head or a playback loudspeaker. If the recorder uses separate heads and

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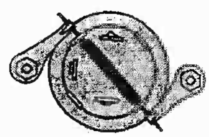
Temporary Capacitor Mountings

In making experimental or temporary hookups, one frequently needs to use an upright can-type electrolytic capacitor. Damaging the twist prongs of the capacitor would render it incapable of being mounted on permanent equipment, however. A way to get around the difficulty, yet still mount the capacitor vertically on a metal chassis, is to twist only one or two of the prongs, depending on whether the total number of prongs is three or four. If they are damaged, these prongs can be cut off later, leaving two good ones for the final mounting.

The drawing shows a better method which can be used where grounding of the can is permissible. In mounting the wafer base, put a long (7/8-inch) soldering lug under each bolt, as shown, and pry them up slightly away from the chassis. Slip the capacitor into the wafer and run a 2-inch length of bare No. 10 copper wire across the bottom of the capacitor, bending it if necessary to avoid live contact lugs (slip a piece of spaghetti over the center portion of the wire for safety). The wire goes through the holes in the prongs on opposite sides, and is soldered to the lugs. The capacitor is now held firmly, and it can be removed unharmed for further use.



Temporary mounting for can capacitors.

Harry L. Wynn
Derry, Pa.

Improving an Old Record Changer

After it had been in use for more than eight years, an originally fine rim-drive record changer had quite a pronounced flutter. Replacement of the idler wheel produced no noticeable improvement, so the entire motor and transmission was raised about 3/32 in. with respect to the frame and turntable. Additional washers placed under the three rubber-cushion legs that support the motor platform raised the level of the idler wheel so that it made better contact with the inside rim of the turntable. Flutter was virtually eliminated.

Later, new rubber legs were obtained. Then one thin (3/64 in.) additional washer under each leg was sufficient. The rubber cushions had become compressed through the years, allowing the whole motor assembly to drop, thereby producing the disagreeable flutter.

John E. Hodge
Peoria, Ill.

Paint-Brush Care

A brush used to apply furniture finishes must be as clean as possible, and a brand new brush should be used once or twice before it can be considered to be ready to apply finish coats. Loose bristles and dust will usually come out of a brush the first time it is used. Every time a brush is cleaned and allowed to dry out, it is likely to get in the same dusty condition as a new brush. To avoid this problem, a brush used for finishing is left in the varnish or other finish being used so it will not dry out. For the hobbyist working from a small can, it may be more practical to store the brushes in the plastic bags sold for the purpose at most paint stores. Aluminum foil can also be used, and it will last for several coats.

After applying a coat of finish, a little thinner should be added to the brush. The brush should then be wrapped as tightly as possible to keep air away from it. Brushes wrapped in this manner will stay clean and soft for several days.

Daniel B. DeBra
Mountain View, Calif.

Plexiglass Panels

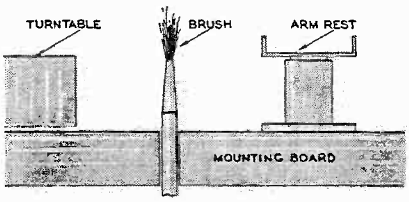
Whenever an old radio-phonograph cabinet is converted to house hi-fi equipment, the problem of adding a panel to match the cabinet's finish invariably comes up. I have found that beautiful effects can be obtained by using colored transparent plexiglass sheets — dark red for mahogany, amber for walnut, etc.

Panels 1/4 in. thick can be used, or, for lower cost, use 1/8-inch plexiglass backed by Masonite or hardboard sheeting which can be painted for varied background shades. To enhance the effect, pilot lights can be arranged to show through the color by making appropriate cutouts in the Masonite.

Harold E. Foss
Coulee Dam, Wash.

Stylus Brush

An effective stylus brush can be made from an inexpensive artist's brush with soft bristles. First, drill a hole in the mounting board the same diameter as the brush's handle. This hole should be



Inexpensive stylus brush arrangement.

so positioned that the stylus will pass directly over it when the arm is moved from the arm rest to the record. The handle of the brush is pushed into the hole and the height adjusted so the brush will touch the stylus. A little glue can be used to hold the brush in position in the hole.

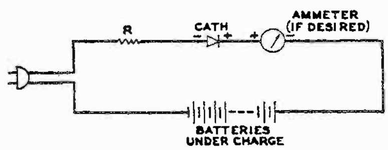
If the handle of the brush extends too far below the mounting board, it can be broken off.

A. Michael Noll
Newark, N.J.

Battery Charger

It is often necessary for the audiophile to work with storage batteries of one sort or another, perhaps for lighting filaments with DC, or for experimenting with transistor circuits. Charging these batteries can be a nuisance, and it is costly besides, unless the owner has a charger on hand. A simple, effective, and inexpensive battery charger is described here.

Merely take any rectifier with a peak inverse-voltage rating greater than 150 volts, and place it in series with a current-limiting resistor, the battery to be charged, and, if desired, an ammeter.



Circuit for charging storage batteries.

Any type of rectifier will work, but semiconductor types are simpler and therefore to be preferred.

Connect this series string to the 110-volt house-current line, as shown in the diagram. The charger will charge at a constant rate, as opposed to some commercial chargers which taper off the current as the battery approaches a fully

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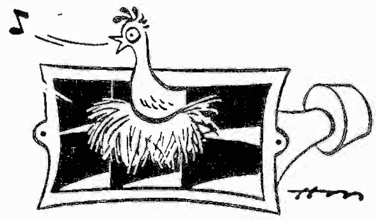

Sound-Fanciers'

Stereo Aids

Originally I had planned to crib intact my neighboring "Audio Aids" column's caption for this month's discussion of "tool" recordings — that is, tapes and discs which serve conveniently in adjusting home sound systems for optimum operation, as well as in choosing suitable permanent-library investments. But now that my time clock has automatically switched me over from listening and card filing to actual copy preparation, I find that most of my pertinent materials (apart from two guides to single-channel tape-recorder operation and the latest LP samplers) are only generically *audio* aids; more specifically they are tools, some well-nigh essential, but all immensely useful, for a better understanding of *stereo*-sound techniques and their most profitable home utilization.

In an earlier piece ("Sneaking up on Stereo" in the SFG column of last June) I outlined some of my personal, nonorthodox stereo views and surveyed the then-available repertory of test, sampler, and demonstration releases. But at that time I could only announce, rather than report from experience, the initial major contributions to the first of these categories: Stereophony's *Test Tape for Stereo Balancing* (T 50) and Sonotape's *Stereophonic Alignment Tape* (SWB AL 101).

The former has proved to be a real (reel) bargain at its modest price of only $1.98, despite its limited scope — some four minutes of running time, which, however, allows for the inclusion of a brief channel-identification tone, alternating-channel 3-Kc tone bursts for level balancing, and a series of spoken passages (delivered from different announcing positions) for determining both speaker locations and levels for optimum stereo spread and minimum center-hole effect.

The latter costs much more ($11.95), but runs for some 24 minutes and includes vastly more plentiful and varied materials. The "technical test" Section I features first (at —20-db level) a 15-Kc tone for head azimuth alignment, a 250-cps reference-level tone, and 12 check frequencies from 50 cps to 15 Kc; then (at 0-db operating level) another 250-cps reference tone, a frequency sweep from 15 Kc to 30 cps, a 3-Kc tone for wow and flutter testing, 440-cps "A" for speed checking against an accompanying tuning fork, and an intermittent tone for checking track synchronization. The "demonstration test" Section II begins with voices on alternate channels and a Stravinsky tango in normal transposed-channel recording; alternating-channel performances of single-channel music for loudness balancing; and a series of speaker-placement tests. Next, there is a series of equalization checks using extreme-range musical materials contrasted with examples of 8-Kc high and 200-cps low cutoffs, and breaking-glass and percussion-battery transient-response demonstrations. Finally, a purported test of maximum-possible undistorted output level is given, which combines the Finale of Tchaikovsky's Fourth with a N.Y. subway train — both running at full blast.

This last bit may be of dubiously meaningful "test" value, but it is the most amazing example of sonic montage I've yet encountered — a hi-fi fanatic's dream (or schizophrenic nightmare!) of aural melodrama. Yet, if it is only audiophile horseplay, the rest of the tape is consistently informative. And not the least of the complete work's attractions are the imaginative inclusion of an actual tuning fork and Kurt List's illuminating notes — both on utilizing the tape itself and on stereo techniques in general. (There are also notes for the Stereophony tape, included in a booklet covering this firm's first 13-reel release list, but here I find myself violently disagreeing both with the "necessity" of strictly symmetrical speaker systems and the general "philosophy" of stereo acoustics and speaker placements.)

Stereo Demos/Samplers/Studies

The list of current newcomers to stereo-tape publication is headed by Capitol, Columbia, Mercury, Urania, Vanguard, and Verve (plus various smaller companies) and is scheduled to be augmented — before this appears in print — by Audio Fidelity and Expériences Anonymes, possibly also by Angel and still others. It is significant that all these are issuing *stereo* tapes only, and for the most part exclusively in stacked-head versions. Only Capitol and Urania, however, have included demonstration samplers in their debut catalogues, but the former has two of these, ingeniously devised to exploit separately the

Guide

by R. D. DARRELL

main potentialities of general programs and specific "showcase" reels.

Thus the *Intro to Stereo* (Capitol ZA 1) is primarily intended to acquaint the novice with both the new medium's realistic reproduction of moving sound sources and its enhanced spaciousness of strictly musical performances, while *A Study in Stereo* (ZH 2) is more of an orthodox sampling of works and artists represented in the first (or forthcoming) Capitol tapes. The former has vocal narration (by Art Gilmore) and in its effectively brief first half dramatizes, after the fashion of Concertapes' famous *Sound in the Round* series, diesel engines, subway trains, holiday crowds, etc., plus a more novel bowling-alley skit and ferryboat trip. The second (musical) half is more uneven in interest (pops and jazz as well as symphonic highlights), but at its best (in single-channel vs. stereo comparisons and an unaccompanied carol by the Wagner Chorale) it is mightily persuasive stereo propaganda indeed.

The latter is a longer and more expensive (26 vs. 12½ minutes; $16.95 vs. $9.95) program of unannounced longer musical pieces or excerpts (except for one "Railway Crossing Episode"). These are of highly diverse appeals; the most impressive — to me — are the Waring Pennsylvanian's *Dry Bones*, Wagner Chorale's *Were You There?*, and Slatkin's percussion episodes from Britten's *Young Person's Guide*. Most of the others strike me as Hollywoodian schmaltz or fanciness, but apart from this my only serious criticism is of the extraordinarily high recording level, apparently just on the safe side of tape saturation, but running serious overloading risks for many smaller home

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SOUND FANCIER

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systems. I am sorry too that while this longer tape is accompanied by an excellent booklet (including a first-rate essay on stereo by Edward Tatnall Canby), there is none for the shorter reel.

Urania's *Stereo Demonstration* (UST D2) has the merits of low price ($3.98), no vocal commentary, introductory alternating-channel 250-cps balancing tones, and extremely crisp and clean recording qualities. Unfortunately, however, only the brilliant Strauss *Egyptian* March and Varèse *Ionisation* are complete, while the scarcely less impressive Haydn concerto, Tchaikovsky and Saint-Saëns symphony excerpts are so disconcertingly fragmentary that they provide inadequate indexes to the notable attractions of the complete tapes from which they are drawn, especially the first truly satisfactory recording of the grandiose *Organ* Symphony (UST 1201) and the ultrasensational *Breaking the Sound Barrier* all-percussion program (UST 1204), both of which I hope to review in more detail in some later column.

Two other stereo samplers, announced without hearing in my June 1957 listings, can be more briefly noted: Omegatape's *Stereo Holiday* (STD 10, $5.95) and Stereophony's *Sampler Vol. 1* (C 80, $4.95), both full reels with around 10 complete selections or extensive excerpts representing the former company's latest and the latter company's first releases. The Omegatape is characterized by strong, high-level recording, no announcements, and mostly pop and light music, topped by a rousing choral and orchestral section from Kálmán's *Gypsy Princess*. The Stereophony tape has brief, low-pressure announcements, less exaggeratedly brilliant but beautifully balanced recordings, and all pops or jazz pieces, topped by two exhilaratingly rowdy contributions by "Doc" Evans's Dixieland Band.

Outside the strict sampler category, Sonotape's *Stereophonic Study in Double Choruses* (SWB 8020) proves to be a unique aid to understanding both the new medium itself and home speaker-placement principles by virtue of its illustrative spoken dialogues (and booklet) wherein the conductor, David Randolph, and recording director, Kurt List, describe—*in situ*—the various choral layouts used for a series of unaccompanied works by Schütz, Allegri, Lotti, Lassus, and Monteverdi. The inclusion of these dialogues, to say nothing of a lack of professional polish in the singing itself, handicaps the repetition of this tape for listening pleasure alone, but for the primarily technically interested student it provides an invaluable education in stereo technology.

So, in a quite different way, does the unnarrated (but elaborately booklet-annotated) exposition of *The Orchestra* by Stokowski (Capitol ZH 8), wherein the various symphonic choirs are heard first separately, then in various combinations, and finally all together. Particularly notable for its Farberman *Evolution* (percussion only) and climactic Mussorgsky-Ravel *Pictures at an Exhibition* excerpts, this is available as well on LP with comparable elegant packaging (SAL 8385), where it also serves as a revelatory document of orchestral scoring, superbly controlled virtuoso performance, and ultrabrilliant, ultracrystalline contemporary recording. But once one has been lucky—or unlucky—enough to hear the stereo edition, even such topnotch merits pale beside the same ones as incredibly enhanced by stereo's spaciousness and overwhelming dramatic impact.

Tape-Recorder Guides

While a wide choice of good—and not-so-good—printed books is available for the enlightenment of prospective and practicing tape-recorder owners, these all suffer from the lack of actual

sonic illustrations of distortion, wow, head magnetization, etc., and the effects of various mike types and locations. The first attempts to fill this lack are an instructional disc, *How to Use Your Tape Recorder* (Golden Crest CR 3005), prepared by Hal Michael, and a 7-inch reel, Jack Bayha's *All About Tape on Tape* (*Tape Recording Magazine*).

The former contains much too much talk (needlessly so, since it's all duplicated in an accompanying booklet) and far too few — and too amateurish — sonic illustrations for my taste. But the latter, while including rather more talk than might be strictly necessary, is far better organized and more practically helpful, while its sonic illustrations (augmented by visual ones in the accompanying booklet), particularly in the "demonstrated glossary" of common tape terms, are marvelously illuminating. Any tape-recorder operator can profit by knowing it; for the brand-new owner it strikes me as well-nigh indispensable.

LP Samplers and *Divertissements*

I see that I haven't left myself much space for the latest disc samplers (tape is really taking over these days!), but in most of the present instances the titles alone are almost-sufficient enticements — and, in any case, even passing mention is much more than such works get in most LP-review columns, where they too often are deliberately or carelessly ignored despite what to my mind is their incalculable value, both as fascinatingly diversified representations of new and old recording techniques and as appetizing stimulators for full-length programs of many of the artists involved.

The two "Voxamples" in particular practically sell themselves since *This is Feyer* and *This Is Novaes* (Vox SFP 1 and SNP 1, $1.98 each) are definite "musts" for every previous admirer of the finest night-club pianist's *Echoes* series and the most gracious of all romantic keyboard interpreters, whose extensive repertories are liberally illustrated here, as well as ideal introductions to two artists, each of whom is quite incomparable in his and her own special musical domains. Emory Cook's first samplers ($1.98 each) also are not to be missed for other reasons: the *Calypso Jazz* (XX 2) as a diverting 14-selection introduction to no less than nine Cook's Caribbean Tours, which probably only specialist collectors as yet know *in extenso* (although I've been reviewing as many as I can get hold of in these pages); the *Audio Follies* (XX 1) as capsulated representations of some 15 famous Cook "follies" and triumphs (everything from locomotives to Lizzie Miles, calliope to zither, steel bands to symphony orchestra) which perhaps only the most fanatical audiophiles already have in the original full-program LP's. The pressings are by the Microfusion process, natch!, while the tough-plastic-envelope format is something brand new — and likely soon to find many imitators.

Unicorn also hits the jackpot with *Music at M.I.T.* and *Unicorn* samplers,

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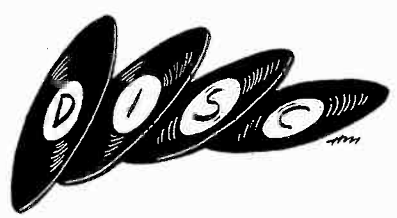
(UNSR 1 and 2, $1.98 each). The latter, with 13 complete pieces or movements from old composers (Vivaldi, Palestrina, Gabrieli, Handel, Sullivan, Schubert, Sweelinck) on one side and comtemporaries (Bartók, Ives, Stravinsky, Randall Thompson, Roland-Manuel, Cowell) on the other, is musically the most catholic and substantial of any disc sampler yet, if technically somewhat uneven as a result of its inclusion of some less striking European recordings. What Unicorn — and Peter Bartók — can do on their home grounds is best illustrated in the former LP, with its three complete big works (Levy's Beethoven Op. 109 Piano Sonata, Moe's and Liepmann's Handel *Organ* Concerto, and the Voisin Brass Ensemble's Berezowski Suite) which demonstrate anew the superb acoustics of the M.I.T. Kresge Auditorium and the miraculously (for single channel) "stereogenic" recordings that can be achieved there.

In addition, Vanguard's *Hi-Fi Hi-Jinks with Strauss* (SRV 104, $1.98) is an enticingly varied and brightly recorded selection from Anton Paulik's Vienna State Opera Orchestral best-selling waltz, polka, etc., programs. . . Elktra's Sampler No. 3 (SMP 3, $2.00) is sheer joy for its A-side folksong artists; or more specialized, but still lively, interest for its six-item pops and jazz program overside . . . while Debut's *Autobiography in Jazz* (DEB 198, $1.98), with 14 performances representing its 1952-1954 activities, is markedly more limited in both jazzical imagination and recording quality.

That brings us up-to-date on samplers, but I can't switch off without expressing my uninhibited relish of two other combined audiophile and musical adventures, both of which rank in their own distinctively individual ways right up with the *Music at M.I.T.* and Stokowski's Capitol LP as outstanding examples of the finest contemporary engineering. However, only the grimmest hi-fi fan is likely to appreciate these discs purely technologically, since *The Music of*



Leroy Anderson (with Fennell and the Eastman-Rochester Symphony, Mercury MG 50103) is so delectably ingratiating in itself, as well as revealing — in the *Irish* Suite — scoring and arranging felicities hitherto commanded only by Robert Russell Bennett — and since *Port Said* (starring Mohammed El-Bakkar with his Oriental Ensemble, Audio Fidelity AFLP 1833) is as much out of this world as it is out of Suez hot spots. Already a "sleeper" hit on Broadway (perhaps as much for its dancing-girl pin-up cover as for its sonic qualities), this may feature more singing than some audiophiles, entranced by its piquant variety of exotic instrumental timbres (and particularly those of its near-ultransonic finger- or antique-cymbals' glitters) would ordinarily approve. But there is at least one all-instrumental piece, the rhapsodic *Al Jazayair*, and even the rest, vocals and all, are super-hootchy-cootchy fare, at once provocatively hypnotic and stimulating — irresistibly catchy offbeat entertainment which can be safely guaranteed to galvanize even its staidest listeners into dance gyrations of their own!

AUDIO ANALYZER

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voltages available in the analyzer itself, and we tried this method to see how accurate it really was. As it turned out, the method is accurate to within less than 5%.

Ultimately, the analyzer was zeroed in using an oscillator and three calibration resistors (which, incidentally, are supplied in the kit) and following the instructions in the manual. The 5% that the meter was off in actuality made almost no change in percentage indication when an amplifier generating 0.2% or 0.3% IM was tested — which merely proves that calibration of the analyzer using the internal method is sufficiently accurate for all practical purposes.

Audiocraft Test Results

Testing a test instrument is a nebulous proposition, since most test devices — and this one is no exception — provide adjustable controls which are used for calibration from time to time to keep the instrument up to snuff.

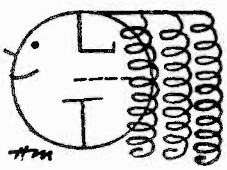
We checked the VTVM meter indications against a number of known voltages, using several transformers which produced voltages as low as 5 v and as high as 300 (the top limit for full-scale meter indications). Furthermore, we checked the voltage indications of our analyzer against a multipurpose VTVM and against a VOM which has served infallibly over many years. Finally, it was checked against another analyzer which has been in use for quite some time. In every instance, our unit was right on the beam.

The multitude of uses for which the Audio Analyzer is suited make it an electronic jack-of-all-trades. The internally selected combination of functions limits the number of test leads used for checking a single amplifier to two.

We replaced the generator output leads with a length of shielded cable, and soldered an RCA plug to the end. Not only does this shield the test signal from stray magnetic fields, but it provides a solid ground connection, so that the analyzer's input ground can be ignored for most applications. With such an arrangement, however, it is important to make certain the output transformer in an amplifier under test *is* grounded on one side before attaching a cable only to the tap to be used for testing.

The analyzer provides the most rapid means we know of checking an amplifier for maximum power output, distortion, and signal-to-noise ratio, without making any changes in the test leads. When used for design work, the generator and VTVM provide a speedy method for checking stage gain, and the IM analyzer is readily available for double-checking distortion as various bias values are tested.

If design and/or testing of high-fidelity equipment are uppermost in your mind, the Heathkit Audio Analyzer can find an indispensable place on your workbench. It will save you time and trouble, and will provide an accurate and rapid means of checking the results of your handiwork.



STEREO

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with an input gain control adjustable by means of a screw driver. Much listening has been conducted with different adjustments, and the fact that the best adjustment is full-on appears to corroborate the validity of the theoretical basis involved in making the box.

Listening Results

At an orchestra rehearsal, the conductor did a lot of talking. Using the customary two-track two-channel stereo, there was seldom any definite sense of location of events taking place in the middle of the stage. Adding the phantom center channel resulted in bringing those central-area events into focus. The director talked from the center; a conversation between him and the harpist assumed the proper space geometry. The tuba sounded off from somewhere between the center and right-flank speakers. Percussion could be pointed out with fair accuracy. A soloist was as accurately focused as would appear possible with a third physical track. It appears to afford

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all, or at least most, of the advantages of a three-track three-channel system at substantially less expense. In the present state of the art three-track stereo is being practiced on tape wider than 1/4 in., which takes it out of the practical class for home use.

Microphone Placement

Most of the tapes studied so far were recorded with the idea of obtaining maximum results from two-track two-channel stereo. With three-channel playback, the microphone placement appeared to be good. One tape was made by a young recordist (who was unaware of the necessity for wide microphone placement) using only 10 ft. spacing for a full band and small choral group; on three-channel playback the band was inadequately differentiated, and the choral group was spread too far.

Two symphony orchestras were then recorded with 22 and 26 ft. microphone spacing. Both spacings gave good results. The 22-foot separation was judged adequate, but capable of improvement. A jazz group spread out about 20 ft. was recorded with microphones spaced 15 ft.; another jazz group spread out about 14 ft. was recorded with 10-foot spacing. These gave good results on two-channel playback, and the improvement with three-channel playback suggested that the mike placement was nearly optimum.

On a two-piano duet with a microphone over each piano, three-channel playback added little, but neither did it detract. This seems to confirm further the correctness of the idea of three-channel playback from only two sound tracks.

"Longitudinal Stereophonic,"[4] wherein one microphone is placed close to and the other remote from a solo source such as single piano, or organ with single pipe loft, has been practiced with two-track two-channel technique, and the addition of the center channel appeared neither to help nor hurt.

Some "Chronolateral Stereophonic"[5] recordings, which involve recording one channel at one time, and the other channel later with the same or a different performer, were likewise benefited but little and hurt none by the additional channel.

Speaker Placement

When I started experimenting with two-channel stereo, experience at various demonstrations was unsatisfactory; insufficient separation or sharply focused

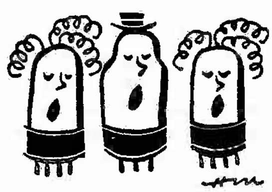
[4]Paul W. Klipsch, "Experiences in Stereophony," *Audio*, XXXIX (Jul. 1955), pp. 16-17, 41-42.
[5]Paul W. Klipsch, "Making Stereophonic Tapes," *Music at Home*, II (Nov.-Dec. 1955), pp. 54-56, 68, 72.

point-source effects marred the demonstration. Accordingly, my approach was made on a basis of wide microphone separation and corner placement of speakers. It was found that, usually, placement of corner speakers against the long wall of a room was better than against a short wall. Thus in a living room of 20 by 30 ft., the performance was better with the speakers 30 ft. apart than 20 ft. But the wider the separation the more apt there is to be a "hole" in the middle, and this has been the subject of occasional adverse criticism.

The phantom center channel has not merely filled this hole, it has satisfied the ear to the extent that there appears to be a solid curtain of sound rather than a group of point sources. Probably part of the over-all effect has been due to lucky microphone placement, and it is this writer's personal opinion that the corner placement of the flanking speakers is a contributing factor to the stereo effect as well as to the individual speaker performance per occupied unit space.

As practiced, the right-side speaker is a Klipschorn, the left is a Shorthorn, and the center a Rebel V, all being three-way systems using the same type middle-range and tweeter driver units. The bass efficiencies differ slightly, and the bass cutoffs differ considerably; but the ear is not sensitive to direction for frequencies below 100 cps, so the differing bass response is not sensed. Since all three speakers are corner types and the center unit has no corner, it is at a disadvantage which, again, is not noticed. It is proposed to design a noncorner speaker specifically for this application, although it is doubted that its use will be so confined. Since this writer has been the advocate of corner speakers exclusively, this noncorner design has been proposed to be called "Klipsch's Heresy."

The mention of speakers is made to show that the three speakers need not all be of the most expensive type, but they should afford a full treble range



and a smooth response within the range they cover. At least one of them should afford a full bass range. Since the bass range is the expensive part, it will be seen that the cost of two of the speakers can be reduced drastically without impairment of the sound quality even to a small degree.

To give an idea of the characteristics of the three speakers, the large unit on the right is 10 db down at approximate-

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ly 32 cps; the smaller unit at the left is down 10 db at 50 cps; and the small center unit is down 10 db at about 90 cps. Substitution of speakers with more extended bass range did not improve performance even on pipe-organ program material, where fundamentals down to 32.7 cps were frequent. All three speaker systems are substantially flat to 13,000 cps.

Conclusion

The cost of going from a single-channel playback system to two-channel stereo is something like half the initial cost, assuming one already has a qualified tape machine which is convertible to stereo. The cost of going from two-track two-channel playback to two-track three-channel playback is only that of an amplifier and the smallest of the three speakers; this cost is almost marginal, but the improvement is almost as great as the step provided in going from monaural to two-track stereo.

PUZZLEMENTS

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didn't like it. This happened because the capacitive reactance interfered with the stability of the amplifier's "negative" feedback. At some high frequency the feedback turned positive, and at one point was almost enough to cause oscillation — in this case, at about 13,000 cps.

The equivalent circuits we have discussed have not shown the effect of radiation resistance and other acoustic results of the fact that the diaphragm *moves*. In the basic unit (assumed perfect) the radiation resistance can be regarded as part of the resistance in parallel with the capacitance. In practical tweeters, the component due to motion of the diaphragm is much more complicated. But it does not change the basic kind of impedance; it only produces small additional components.

Like the political problems that were such a puzzlement to the King of Siam, this situation poses problems for hi-fi equipment manufacturers. Who is to blame for the distortion? Most often the uninformed consumer will blame the tweeter, because the amplifier worked perfectly with his previous tweeter. But there is nothing wrong with the tweeter, as No. 2 amplifier demonstrated.

Manufacturers of amplifiers having this trouble may resent the advent of the electrostatic tweeter, and complain of it as an abnormal load to connect to an amplifier. But the electrostatic tweeter is a new member of the hi-fi component community, so everyone must — even if grudgingly — grant him work space.

BASIC ELECTRONICS

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enough so that its reactance is many times the magnitude of the following grid resistor over the required frequency range. Therefore the total AC load impedance, determined primarily by the grid resistor, is practically constant — so the gain of the stage is uniform over the operating frequency range. The main advantage of LC coupling, compared to RC coupling, is that there is very little DC voltage drop in the inductor; accordingly, the full B+ supply voltage can be utilized at the plate, which means that a larger signal output voltage can be produced before distortion becomes significant. The circuit is often used at radio frequencies. At audio frequencies, however, high-quality chokes of suitable value are quite expensive. It is usually more satisfactory to obtain equivalent performance in other ways; consequently, LC coupling circuits are seldom employed at low frequencies.

The final coupling method we shall consider is *transformer coupling*. A circuit is shown in Fig. 1D. Here the plate load is the primary winding of a transformer; the signal is coupled to the following stage, or to an external load, via the transformer secondary winding. This has the same advantage as the LC-coupled circuit: it makes available the full B+ voltage on the plate of the preceding tube, since the DC voltage drop in the primary is negligible. In addition, by varying the relative number of turns in the two windings, a voltage step-up or step-down can be obtained on the secondary side. Finally, in the case of a power output stage, the load impedance can be effectively transformed by a proper turns ratio choice to the optimal value of load for the tube.

Unfortunately, it is difficult — and costly — to make a transformer having uniform response over a frequency range wide enough for use in high-fidelity audio amplifiers. A transformer is inherently a narrow-band device. It is, again, cheaper and more satisfactory to use RC coupling exclusively for interstage applications in hi-fi amplifiers; interstage audio transformers, once used extensively, are virtually never employed now. Transformers are used almost without exception, however, between the power output stage and the load.

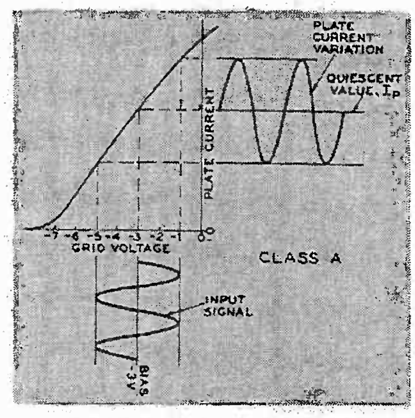
For high-frequency (radio and television) tuners, amplifying stages with very narrow frequency pass bands are not only acceptable but desirable, and extremely high gain is needed as well. Transformer coupling is ideally suited to this application. The primary or secondary winding, or both, are commonly made sharply resonant at the desired radio frequency by adding tuning capacitors, as shown by the dashed lines in Fig. 1D. This practice makes the stage gain very high for the desired frequency, and very low for any undesired frequency.

Operating Class

One of the more important methods of classification for vacuum-tube amplifier stages is by the amount of bias, which determines the operating point of the tube. In Class-A operation, for example, the tube is biased approximately halfway between zero and cutoff.

Fig. 2 shows a typical grid characteristic curve for a triode, modified to include the effect of variations in plate voltage with changing plate current, for a given value of plate load. Thus, we

Fig. 2. A typical grid characteristic curve drawn for a triode vacuum tube.



can assume that the curve gives a true indication of the effect of grid voltage on plate current, under practical operating conditions.

Note that the tube is biased at —3 v, which is approximately half the distance down the linear portion of the curve from the zero-bias point. If we now apply an input signal of 4 v peak-to-peak, the grid voltage will vary 2 v above and 2 v below the bias point: from —1 to —5 v. This variation is pictured below the E_G-I_P curve, and the extreme values of the input-signal variation are projected upward to the curve. These points on the curve represent the extreme values of plate-current variation produced by the signal swings. Accordingly, they represent the inverse of the plate-voltage variation extremes and, projected to the right, give an inverse picture of the output voltage signal.

Since the bias is so chosen that operation is entirely within the most linear portion of the E_G-I_P curve, the output signal is a reasonably accurate facsimile of the input signal. If the input signal were increased to 6 v peak-to-peak, it would produce a grid-voltage variation from zero to —6 v. The curve is still fairly linear over this range; but, if the input signal were increased still further, it would enter increasingly nonlinear portions of the curve simultaneously

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at both excursion peaks. Within the normal operating range, however, plate current flows during all parts of the input signal swing. The essence of Class-A bias is that it provides for maximum linearity and minimum distortion, not for maximum power output nor efficiency. For that reason it is invariably employed in audio voltage-amplifier stages, and very often in power output stages also.

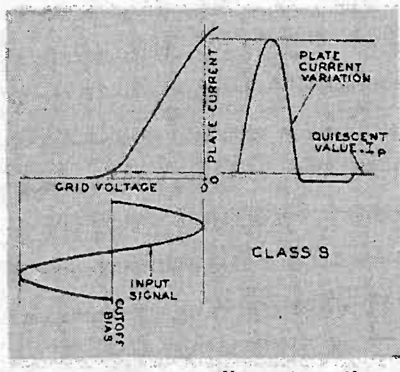

Fig. 3A. Diagram illustrating Class-B operation with bias near cut-off point.

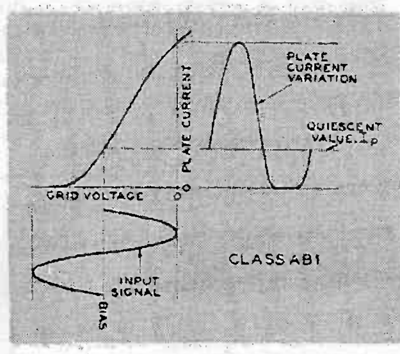

Fig. 3B. In Class AB1, grid will never become positive with respect to cathode.

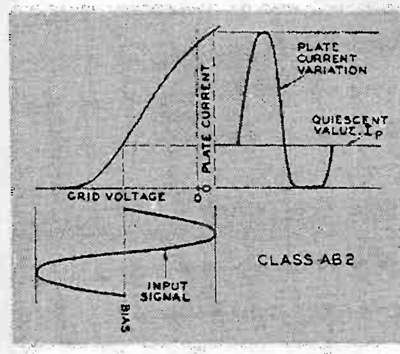

Fig. 3C. If grid becomes positive the operating class is known as Class AB2.

In Class-B operation, diagrammed in Fig. 3A, the tube is biased at (or very near to) the cutoff point. Only on the positive halves of the input signal swings does the tube conduct current, and only the upper halves of the signal wave form are reproduced. The negative halves are simply clipped off. As before, the output wave form shown is that of the current; the output voltage wave form is inverted.

Fig. 3B shows an intermediate bias point, in which the tube is biased more than halfway to but not at cutoff. Logically, this method of operation is known as Class AB. The positive swings of the input signal are completely reproduced. The negative swings are completely reproduced if the input signal is small enough so that it does not drive the grid to cutoff; if it is large enough so that the negative swings do exceed the cutoff point, as in Fig. 3B, then they are partially reproduced. For small input signals, then, this is a Class-A stage, and for large input signals it approaches Class-B operation.

Note that the input signal is shown just large enough to reach zero grid voltage on its positive swings. If this represents the largest signal that will be accommodated — in other words, if the grid is never to go positive with respect to the cathode, and hence will not conduct current — the operating method is known as Class AB_1. But if the signal source is so designed that it can drive the grid positive without severe distortion of the peak, as shown in Fig. 3C, then the designation is Class AB_2. It is obvious again that for small signals that do not drive the grid to cutoff, the equivalent of Class-A operation may be obtained. A similar distinction can be made for Class-B stages, Class B_1 indicating operation restricted to zero grid voltage or below, and Class B_2 indicating that grid drive into the positive region is permissible.

Still another bias point is possible. Although not illustrated, Class-C stages are biased well below the cutoff point, so that only the positive peaks of the input wave form cause conduction. Class-C bias is employed primarily in radio transmitters with resonant circuits as the plate loads. The peaks of conduction replenish the oscillating currents in the plate "tank" (resonant circuit); the output wave form of such a circuit is a sine wave, which is desired, and a badly distorted input wave form is not important.

Efficiency and maximum power output both increase as the bias point is lowered. Thus, a power output tube operated Class AB can produce more power than the same tube operated Class A, and if it is operated Class B still more power can be obtained. This fact makes AB and B operation appealing for audio amplifier power output stages. The very bad distortion of the negative input-signal swings can be overcome, with varying degrees of success, by employing a push-pull output circuit. Basically, a push-pull circuit is one in which two tubes are used, driven by oppositely phased input signals. One tube is driven in the positive direction while the other is driven negative; when one tube is cut

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off, the other is conducting. The plate currents are added to produce a composite output. This will be discussed more fully in another chapter.

Before leaving the subject of operating class, however, it should be mentioned that only in Class-A operation is the average plate current anywhere nearly uniform regardless of the size of the input signal. It follows that cathode bias is usable only in Class-A stages. For any other method of operation, it is necessary that fixed bias be supplied — a disadvantage that may weigh heavily in audio amplifier design, and another reason for the higher cost of high-power amplifiers.

TAPE NEWS

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amplifiers for recording and playback, the hum is restricted to the playback amplifier, and probably will not have contaminated any tapes. Either way, the job is simply a matter of conventional trouble shooting.

Of course, the hum might not really be in the amplifier itself; it might be coming from the playback head, or from the shielded lead between the head and the preamplifier stage. We can find out which is the case by soldering a short heavy lead across the preamplifier input tube's grid resistor. The shorting lead should be no longer than necessary to get from one grid-resistor connection to the other, and when installed, the recorder should be reassembled and put through the same test as before. If the hum is killed or is markedly reduced, it was originating in the head or the connections thereto. A ground loop or dry-soldered connection may be the cause of the trouble. Alternatively, the head may be acting as an inductive-pickup device and receiving AC magnetic radiation from the drive motor or amplifier power transformer. Check by substituting new units. Finally, if all else has failed, try reorienting the drive motor, rotating it about its axis, and try relegating the power transformer to a separate chassis located a few feet away from the recorder. If the hum still persists, pick up a high-fidelity equipment catalogue, turn to the pages listing tape recorders, and start saving money for a new unit.

The hum problem has been pretty well licked in most current models of half-decent tape recorders, but the story of tape hiss is thus far not quite so inspiring. Hiss is by far the most common annoyance encountered by amateur magnetographers,[1] and although many extreme cases of it are directly attributable to poor playback loudspeakers,[2] it is frequently a problem in some of the best-regulated setups. Unlike hum, though, tape hiss seems to be inherent in the tape medium itself. Hiss is never entirely absent from a tape recording, and although it may on the best machines be well below the limit of audibility, constant care and maintenance are requried to keep it there. Low tape hiss is a rather critical condition of delicate balance, and practically any malfunction of the record or playback transducers can spoil things and bring hiss into the foreground.

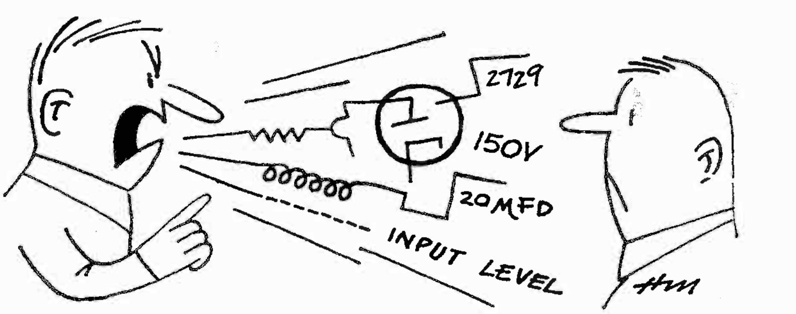
Tape hiss can be controlled, however, often with a minimum of effort on the operator's part — so let's be fundamental about it and consider what causes tape hiss in the first place. Then we may dwell on the cures.

To begin with, a tape-playback head does not actually respond to magnetism per se, any more than a phono pickup responds to a groove. There must be motion of the record groove before the pickup will transduce, and there must be *change* of magnetism before a playback head will transduce. If a powerful DC magnet is held perfectly still near a tape playback head, the head will produce nothing. Moving the magnet will create an impulse through the head, and moving it rapidly back and forth will generate a wave train, or an audible tone.

Because of this, we should expect a tape that is *uniformly* magnetized to play just as silently as one which carries no magnetization at all.

The tape with the DC magnetization provides no more magnetic variations than the tape that is devoid of magnetization, so it should not produce any more sound than the blank tape. In practice it does, simply because magnetic

[1] "Readers' Forum," AUDIOCRAFT, II (Sept. 1957), p. 48.
[2] J. Gordon Holt, "Tape News and Views," AUDIOCRAFT, II (May 1957), pp. 10-11, 46-47.



tape is not quite so homogeneous as it may look.

A magnetic tape coating consists of billions of tiny particles of iron oxide, each one capable of becoming an individual magnet, with its own north and south poles. Thus it is immediately evident that the coating can never be imbued with a true DC magnetic charge; we could magnetize the surface ends of all the particles longitudinally so that we had a string of north and south poles lined up from end to end along the tape. But we cannot produce a continuous magnet from one end of the tape to the other, with north and south poles 2,400 ft. apart. In practice, the closest approach we can get to a DC magnetized tape is one whose *average* magnetization is uniform along its length, and this is where DC "erasure" fails to produce low hiss levels. The average throughout the depth of the tape may be uniform, but the playback head is going to respond most strongly to those particles coming in *direct* contact with the playback-gap pole pieces. If the particles were absolutely identical in shape, size, and orientation, we might still get fairly low hiss. But since they aren't, the law of averages works to bring slightly more positive poles than negative poles across the gap at a given instant, and perhaps more negative poles the next instant. There is consequently an inevitable and random shifting of the average magnetic force, and the result is random sound — or hiss.

For minimum background noise, then, the ideal state for a magnetic tape is absolute neutrality, or absence of magnetism. If anything acts to add DC magnetization to the tape, or otherwise to sway the tape to a state of nonneutrality, its background hiss level will increase proportionally.

Tape is erased by subjecting it to a magnetic field of sufficient intensity to magnetize it to the point at which it cannot retain any more magnetization. We can bring tape to this saturation point by running it over the poles of a permanent magnet, but we have already seen what this does to the hiss level. So to erase a tape and then minimize residual hiss, we must first saturate it and then neutralize it. This is best done by means of an alternating magnetic field. The field is first applied at its full intensity so that it swings the tape's magnetism back and forth through positive and negative saturation. Then, as the field is gradually diminished in intensity, the tape's magnetism follows it through successively diminishing alternations until, with the field tapered off to nothing, the tape is almost perfectly free of magnetism.

This is the operating principle of the bulk eraser, which can wipe out an entire reel of tape within seconds, without un-

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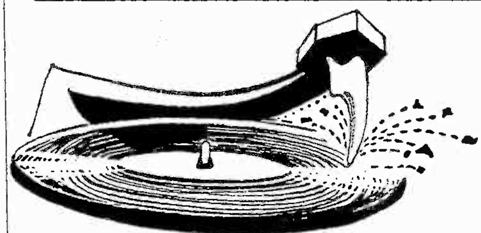

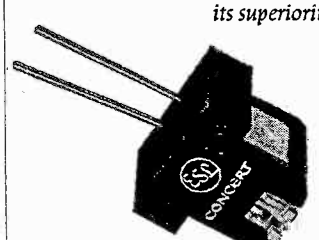

TAPE NEWS

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winding the tape. It is also the principle of the erase head on a recorder — only the details of operation are different. A bulk eraser provides its saturation intensity when held close to the reel of tape, and the diminishing field is created by withdrawing it slowly from the vicinity of the reel. An erase head, however, must be in constant contact with the tape. It produces saturation of the tape passing over its pole pieces, and the diminishing magnetic lines of force radiating outward from the head gap provide the diminishing field as the tape moves past the gap. Since the recorder must supply an ultrasonic bias signal to the recording head, to insure low-distortion tapes, it is convenient to use this tone for erasure also.

That's the theory of noiseless, hissless tape recording, and it's so perfect and foolproof and idealistic that it almost brings tears to my eyes. Unfortunately, there are some problems, a few of which I shall enumerate forthwith.

The erase signal and the bias signal must have perfectly sinusoidal wave forms, otherwise the magnetic reversals on the tape will not be in precise opposition to one another, and the tape hiss will be increased.

There must be no trace of DC leakage from the recorder's output stage into the record head. If there is, the head will carry DC magnetism and will permanently increase the hiss level of any tape, old or new, that passes across it.

There must be no way in which DC current can get to the playback head (if this is separate from the record head). If it does — tape hiss.

If the recorder is switched from the record mode at *any* time, it may cut off during a positive or negative peak of the bias signal. One such impulse won't hurt, but if the law of averages should decree that the switch contacts break on positive peaks more often than negative ones (or vice versa) residual DC magnetism will build up on heads.

Be reassured, though; there are several solutions to the tape-hiss bugaboo. I'll reserve word about them for next month.

PROFESSIONAL DIRECTORY

TRANSISTORS

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[2]For further references to noise, see Paul Penfield, "Transistors in Audio Circuits," Part III, AUDIOCRAFT, II (Jan. 1957), p. 40.

STATEMENT OF THE OWNERSHIP, MANAGEMENT, CIRCULATION, ETC., REQUIRED BY THE ACTS OF CONGRESS OF AUGUST 24, 1912, AND MARCH 3, 1933

Of Audiocraft, published monthly at Great Barrington, Massachusetts, for October 1, 1957

1. The names and addresses of the publisher, editor, managing editor, and business manager are: Publisher, Charles Fowler, Egremont, Mass.; Editor, Roy F. Allison, North Egremont, Mass.; Managing Editor, Frank R. Wright, New Marlboro, Mass.; Business Manager, Warren B. Syer, New Marlboro, Mass.

2. The owner is: Audiocom, Inc., Great Barrington, Mass.; R. F. Allison, North Egremont, Mass.; C. G. Burke, Ghent, N. Y.; J. M. Conly, Great Barrington, Mass.; S. Q. Curtiss, Sheffield, Mass.; C. Fowler, Egremont, Mass.; R. H. Hoopes, Jr., Chevy Chase, Md.; R. Lindstrom, North Egremont, Mass.; F. C. Michalove, Englewood, N.J.; W. B. Syer, New Marlboro, Mass.; H. R. Sykes, Pittsfield, Mass.; F. R. Wright, New Marlboro, Mass.

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(Signed) Charles Fowler

Sworn to and subscribed before me this Sixth day of September, 1957.
(Seal) Lillian F. Bendross, Notary Public
Commission expires June 24, 1961.

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AUDIO AIDS

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The value of the current-limiting resistor, *R*, is determined experimentally to furnish the desired charging rate.

If the desired charging rate is higher than the rating on the rectifiers available, two or more may be put in parallel using separate dropping resistors for each. Consider the total charging current as the sum of the two charging currents, one from each rectifier.

This charger should only be used as described here when there is no danger of a short circuit from the 110-volt line. If such danger is likely to exist, an isolation transformer in the line will be necessary.

Paul Penfield, Jr.
Cambridge, Mass.

Preamp Pilot Light

In "Audio Aids" in August 1956 there was an item showing how to install a pilot light on the Heathkit preamp. This item recommended the use of a No. 110 Flushlite neon.

I was unable to get around to installing a pilot light on my own Heathkit for some time. When I did get around to it, I was unable to get the Flushlite. As a substitute, I used a No. 1010 Omni-Glow, which is manufactured by Industrial Devices, Inc., of Edgewater, N.J.

I believe the Omni-Glow is easier to install than the Flushlite. It is a tubular affair with an over-all diameter of 5/8 in. and requires a ½-inch hole for mounting. The speed nut supplied with

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AUDIO AIDS

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the light holds it sufficiently tightly to the panel.

Not having a drill large enough to make a ½-inch hole, I drilled a ¼-inch hole and reamed out the balance.

The leads supplied with the light are long enough to reach the ground terminal of the hum control. The lead to the filter-capacitor terminal marked with the rectangular oblong was dressed through the grommet at the end of the chassis side. I had to use an extra piece of wire to make it reach.

William Adler
Brooklyn, N.Y.

Hold Tape in Reel

A simple but effective method of keeping the loose end of a reel of recorded tape in place is to use a short piece of rubber tubing. Use ordinary rubber tubing with a diameter somewhat larger than the space between the flanges of the tape reel, cutting off sections of the tubing about 1 in. in length. Then split each section lengthwise and remove a strip about ⅛ in. wide. The sections of tubing will then fit snugly between the reel's flanges, either along a diameter of a partially filled reel or along the circumference of a full reel.

My own experience has shown that tubing with an outside diameter of ⅜ in. and an inside diameter of ¼ in. works well.

Herbert J. Friedman
Morgantown, W. Va.

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